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Faculty Of Engineering

Department Of Electrical & Electronic Engineering

ERROR DETECTION & CORRECTION

Graduation Project
EE- 400

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ACKNOWLEDGMENT

I am deeply indebted to my parents for their love and support. They have always encouraged me to pursue my interest and ambitions throughout my life.
To my teacher Dr. Jamal FATHI who has helped me to finish and realize this difficult task.
My deep gratitude's and thanks to Prof. Dr. Fakhreddin MAMEDOV who is dean of Engineering Faculty to all their participate.
Also thanks for all my teachers for their advices.
To my sisters (Basma, Arwa, Tanweer) and brothers who give me all love and respect.
To all my friends especially Asim YOUNIS, Allam HIJAWI,
To all of them, all my love.

CONTENTS

INTRODUCTION

As we used to use too many types of equipment, which depends on transmitting and receiving the signals like telephones, TVs, etc. we used to have some noises and miss getting the exact output. In deeply the signals we are getting or receiving or transmitting, can be corrupted. For reliable communication, errors must be detected and corrected.

So as we will detect and correct the errors, we have to study the signals, which mostly gets the error, analog signal and digital, and we also facing multiplexing systems, which is the set of techniques that allows the simultaneous transmission of multi signals across a single data link. Transmission media has to be included throughout our studying, the next step is to investigate the transmission process itself. Information-processing equipment such as PCs generates encoded signals but ordinarily require assistance to transmit those signals over a communication link. For example, a PC generates a digital signal but needs an additional device to modulate a carrier frequency before it is sent over a telephone line.

Error many types, which effects in our transmitted message, we have detected three of them in this topics by detection methods, and we used a method of correcting them.

1. SIGNALS

1.1 Analog and Digital

Both data and the signals that represent them can take either analog or digital form. Analog refers to something that is continuous- a set of specific points of data and all possible points between. Digital refers to something that is discrete. Time is an analog quantity. It is a continuous stream that can be divided up into quarters, hundredths, thousandths, and so on. The measurement of time, however, can be either analog or digital. The hands of a traditional, or analog, clock do not jump from minute to minute or hour to hour; they move smoothly through all possible intermediate subdivisions of a 12-hour period.

Information can be analog or digital. Analog information is continuous. Digital information is discrete.

Digital and analog information can be distinguished by how we think about and refer to them. Analog quantities are generally described using various units of measure, while digital quantities are counted.

We use measuring units for analog quantities; for example, the length of a room can be 12 feet. We count digital quantities; for example, the number of students in a class can be 56.

Like the information they represent, signals can be either analog or digital. An analog signal is a continuous wave form that changes smoothly over time. As the wave moves from value A to value B, it passes through and includes an infinite number of values along its path. A digital signal, on the other hand, is discrete. It can have only a limited number of defined values, often as simple as 1 and 0. The transition of a digital signal from value is instantaneous, like a light being switched on and off.

We usually illustrate signals by plotting them on a pair of perpendicular axes. The vertical axis represents the value or strength of a signal. The horizontal axis represents the passage of time. Figure 1.1 illustrates an analog and a digital signal. The curve representing the analog signal is smooth and continuous, passing through an infinite number of points. The vertical lines of the digital signal, however, demonstrate the sudden jump the signal makes from value to value; and its flat highs and lows indicate that those values are fixed. Another way to express the difference is that the analog signal changes continuously with respect to time, while the digital signal changes instantaneously.

Signals can be analog or digital. Analog signals can have any value in a range; digital signals can have only a limited number of values.

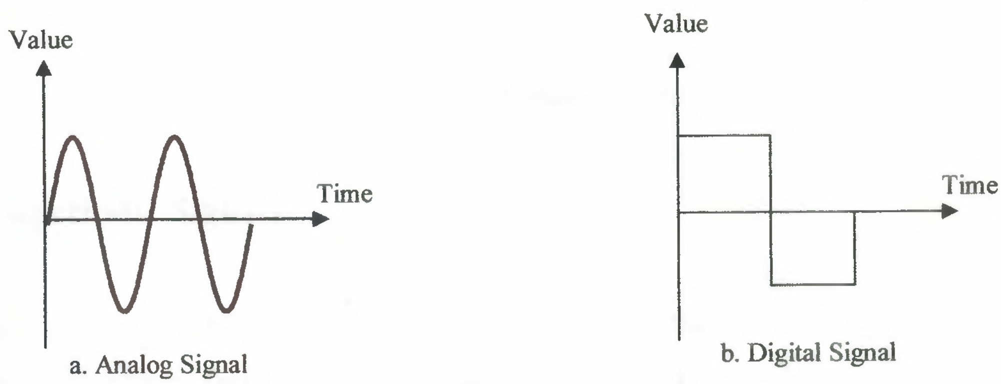


Figure 1.1 Comparison of analog and digital signals

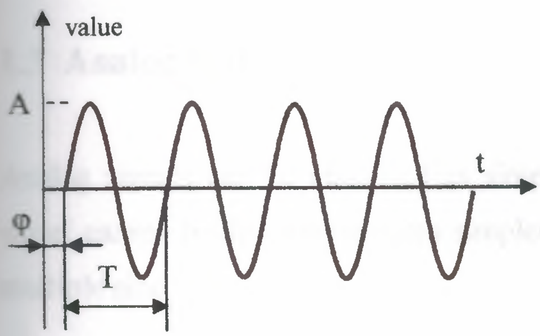
1.2 Aperiodic and Periodic Signals

Both analog and digital signals can be of two forms: periodic and aperiodic.

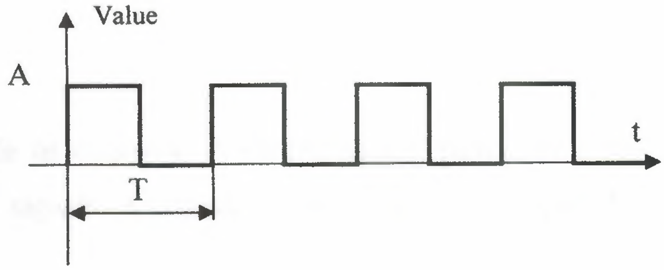
1.2.1 Periodic Signals

A signal is periodic if it completes a pattern within a measurable time frame, called a period, and repeats that pattern over identical subsequent periods. The completion of one full pattern is called a cycle. A period is defined as the amount of time (expressed in seconds) required to complete one full cycle. The duration of a period, represented by T, may be different for each signal, but is constant for any given periodic signal. Figure 1.2 illustrates hypothetical periodic signal.

A periodic signal consists of a continuously repeated pattern. The period of a signal (T) is expressed in seconds.



a. Analog



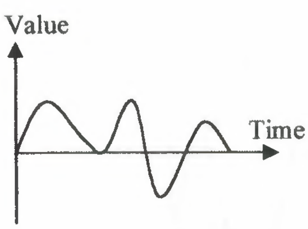
b. Digital

Figure 1.2 Examples of periodic signals

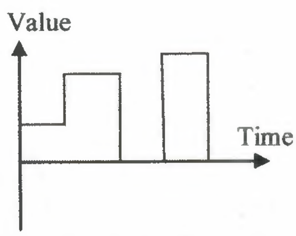
1.2.2 Aperiodic Signals

An aperiodic, or nonperiodic, signal changes constantly without exhibiting a pattern or cycle that repeats over time. Figure 1.3 shows examples of aperiodic signals.

An aperiodic, or nonperiodic, signal has no repetitive pattern.



a. Analog



b. Digital

Figure 1.3 Examples of Aperiodic Signals

An aperiodic signal can be decomposed into an infinite number of periodic signals. A sine wave is the simplest periodic signal.

1.3 Analog Signals

Analog signals can be classified as simple or complex. A simple analog signal, or a sine wave, cannot be decomposed into simpler signals. A complex analog signal is composed of multiple sine waves.

1.3.1 Simple Analog Signals

The sine wave is the most fundamental form of a periodic analog signal. Visualized as a simple oscillating curve, its change over the course of a cycle is smooth and consistent, a continuous, rolling flow. Figure 1.4 shows a sine wave. Each cycle consists of a single arc above the time axis followed by a single arc below it. Sine waves can be fully described by three characteristics:

A. Amplitude
B. Period or frequency
C. Phase

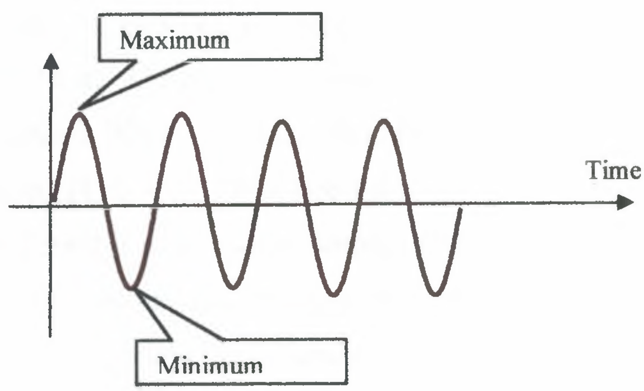


Figure 1.3 A sine wave

A. Amplitude

On a graph, the amplitude of a signal is the value of the signal at any point on the wave. It is equal to the vertical distance from a given point on the wave form to the horizontal axis. The maximum amplitude of a sine wave is equal to the highest value it reaches on the vertical axis.

Amplitude is measured in either volts, amperes, or watts, depending on the type of signal. Volts refers to voltage; amperes refers to current; and watts refers to power.

Amplitude refers to the height of the signal. The unit for amplitude depends on the type of the signal.

B. Period and frequency

Period refers to the amount of time, a signal needs to complete one cycle. Frequency refers to the number of periods a signal makes over the course of one second. The frequency of a signal is its number of cycle per second. Mathematically, the relationship between frequency and period is that they are the inverse of each other, if one is given, the other can be derived.

Frequency = 1/period Period = 1/frequency

Period is the amount of time it makes a signal to complete one cycle; frequency is the number of cycle per second. Frequency and period are inverse of each other: $f = 1/T$ *and* $T = 1/f$.

Unit of Frequency Frequency is expressed in Hertz (Hz), after the German physicist Heinrich Rudolf Hertz. The communication industry uses five units to measure frequency: Hertz (Hz), Kilohertz (KHz = 10^3 Hz), Megahertz (MHz = 10^6 Hz), Gigahertz (GHz = 10^9 Hz), and Terahertz (THz = 10^{12} Hz). See Table 1.1.

Unit of period Period is expressed in second. The communication industry uses five units to measure period: second (s), millisecond (ms = 10^{-3} s), microsecond (m = 10^{-6} s), nanosecond (ns = 10^{-9} s), and picosecond (ps = 10^{-12} s). See table 1.1.

Table 1.1 Unit of frequency and period

Frequency		Period	
Unit	Equivalent	Unit	Equivalent
Hertz (Hz)	1 Hz	Second (s)	1s
Kilohertz (KHz)	10^3 Hz	Millisecond (ms)	10^{-3} s
Megahertz (MHz)`	10^6 Hz	Microsecond (ms)	10^{-6} s
Gigahertz (GHz)	10^9 Hz	Nanosecond (ns)	10^{-9} s
Terahertz (THz)	10^{12} Hz	Pico second (Ps)	10^{-12} s

Example 1.1

A sine wave has a frequency of 8 KHz. What is its period?

Solution

Let T be period and f be the frequency. Then,

$$T = 1/f = 1/8{,}000 = 0.000125 = 125 \text{ m}$$

Example 1.1

A sine wave complete one cycle in 25 m. What is its frequency?

Solution

Let T be the period and f be the frequency?

$$f = 1/T = 1/(25¥\ 10^{-6}) = 40{,}000 = 40 \text{ KHz}$$

C. Phase

The term phase describes the position of the waveform relative to time zero. If we think of the wave as something that can be shifted backward or forward along the time axis, phase describes the amount of that shift. It indicates the status of the first cycle.

Phase is measure in degree or radians (360 degree is 2**p** radians). A phase shift of 360 degree corresponds to a shift of complete period; a phase shift of 180 degree corresponds to a shift of a half a period; and a phase shift of 90 degree corresponds to a shift of a quarter of a period (see figure 1.4).

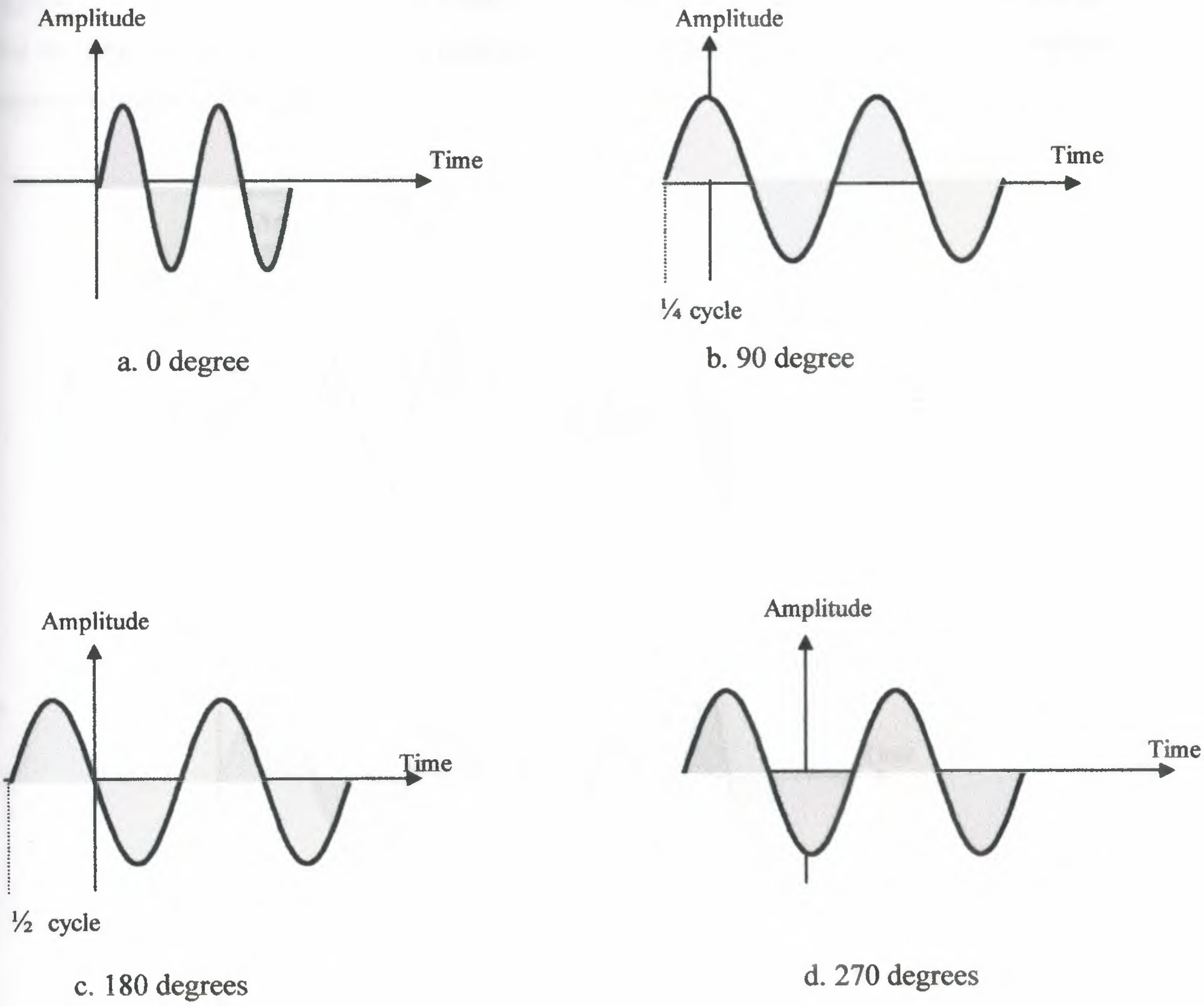


Figure 1.4 Relationship between different phases

A visual comparison of amplitude, frequency, and phase provides a reference useful for understanding their function. Change in all three attributes can be introduced into a signal and controlled electronically.

1.3.2 More about Frequency

We know already that frequency is the relationship of a signal to time, and that the frequency of a waveform is the number of cycle it completes per second. But another way to look at frequency is as a measurement of the rate of change. Electromagnetic signal are oscillating waveforms; that is, they fluctuate continuously and predictably above and below a mean energy level. The rate at which a sine wave moves from its lowest to its highest level is its frequency. A 40 Hz signal has half the frequency of an 80 Hz signal: it completes one cycle in twice the time of the 80 Hz signal, so each cycle also takes twice as long to change from its lowest to its highest voltage levels.

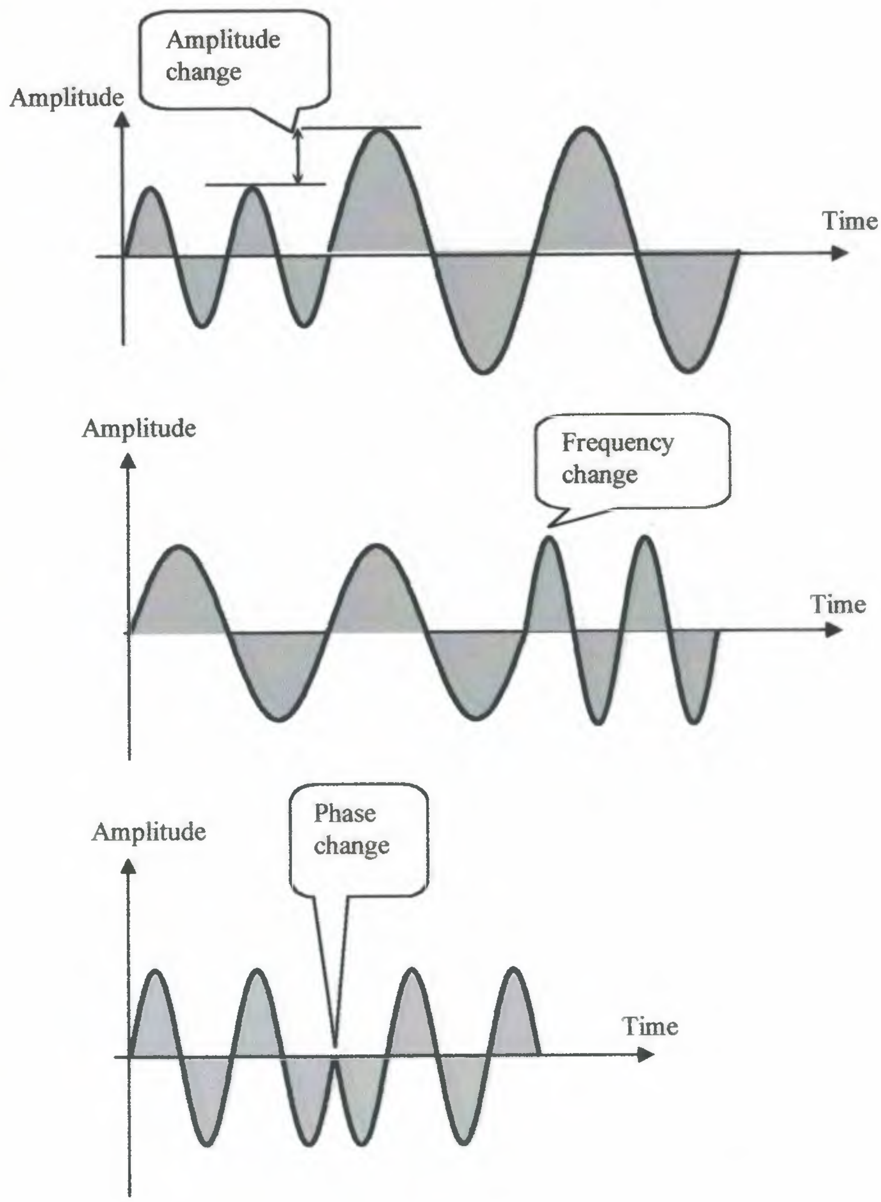


Figure 1.5 Amplitude, frequency, and phase changes

Frequency, therefore, though described in cycles per second (HZ), is a general measurement of change of a signal with respect to time.

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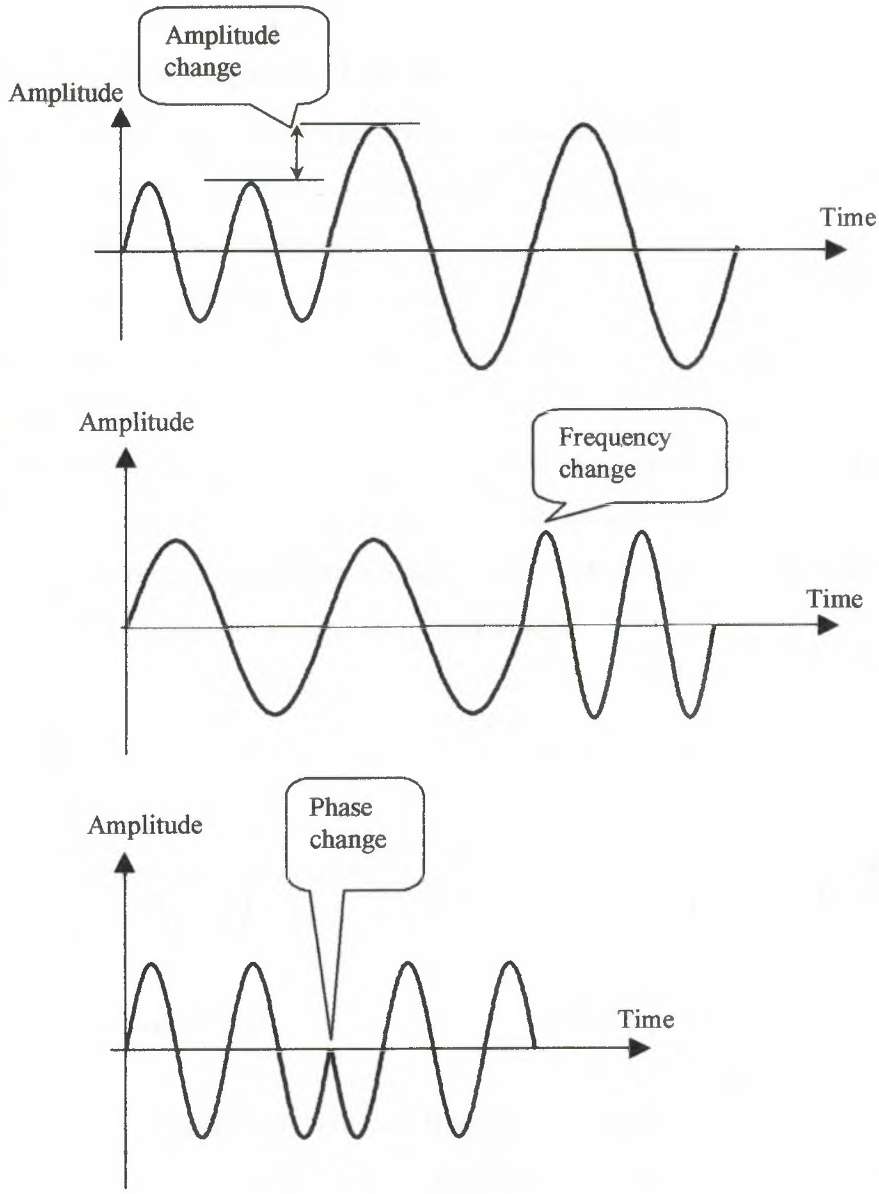


Figure 1.5 Amplitude, frequency, and phase changes

Frequency, therefore, though described in cycles per second (HZ), is a general measurement of change of a signal with respect to time.

Frequency is rate of change with respect to time. Change in a short span of time means high frequency. Change in a long span of time means low frequency.

If the value of a signal changes over a very short span of time, its frequency is high. If it changes over a long span of time, its frequency is low.

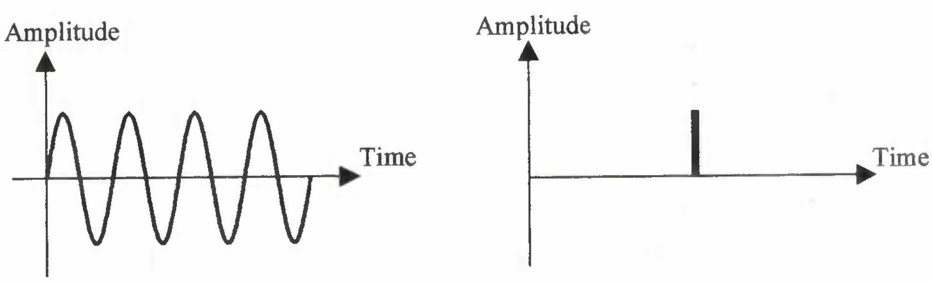
1.3.3 Time versus Frequency Domain

A sine wave is comprehensively defined by its amplitude, frequency, and phase. To show the relationship between the three characteristics (amplitude, frequency, and phase), we can use what is called a frequency-domain plot.

There are two types of frequency-domain plots:

1. Maximum amplitude versus frequency
2. Phase versus frequency.

The first type of frequency-domain plot (maximum amplitude versus frequency) is more common in data communications than the second (phase versus frequency). Figure 1.6 compares the time domain (instantaneous amplitude with respect to time) and the frequency domain (maximum amplitude with respect to frequency).



a. Time domain

b. Frequency domain

Figure 1.6 Time and frequency domains

Figure 1.7 gives examples of both time-domain and frequency-domain plots of three signals with varying frequencies and amplitudes. Compare the models within each pair to see which sort of information is best suited to convey.

A low-frequency signal in the frequency domain corresponds to a signal with a long period in the time domain and vice versa. A signal that changes rapidly in the time domain corresponds to high frequencies in the frequency domain.

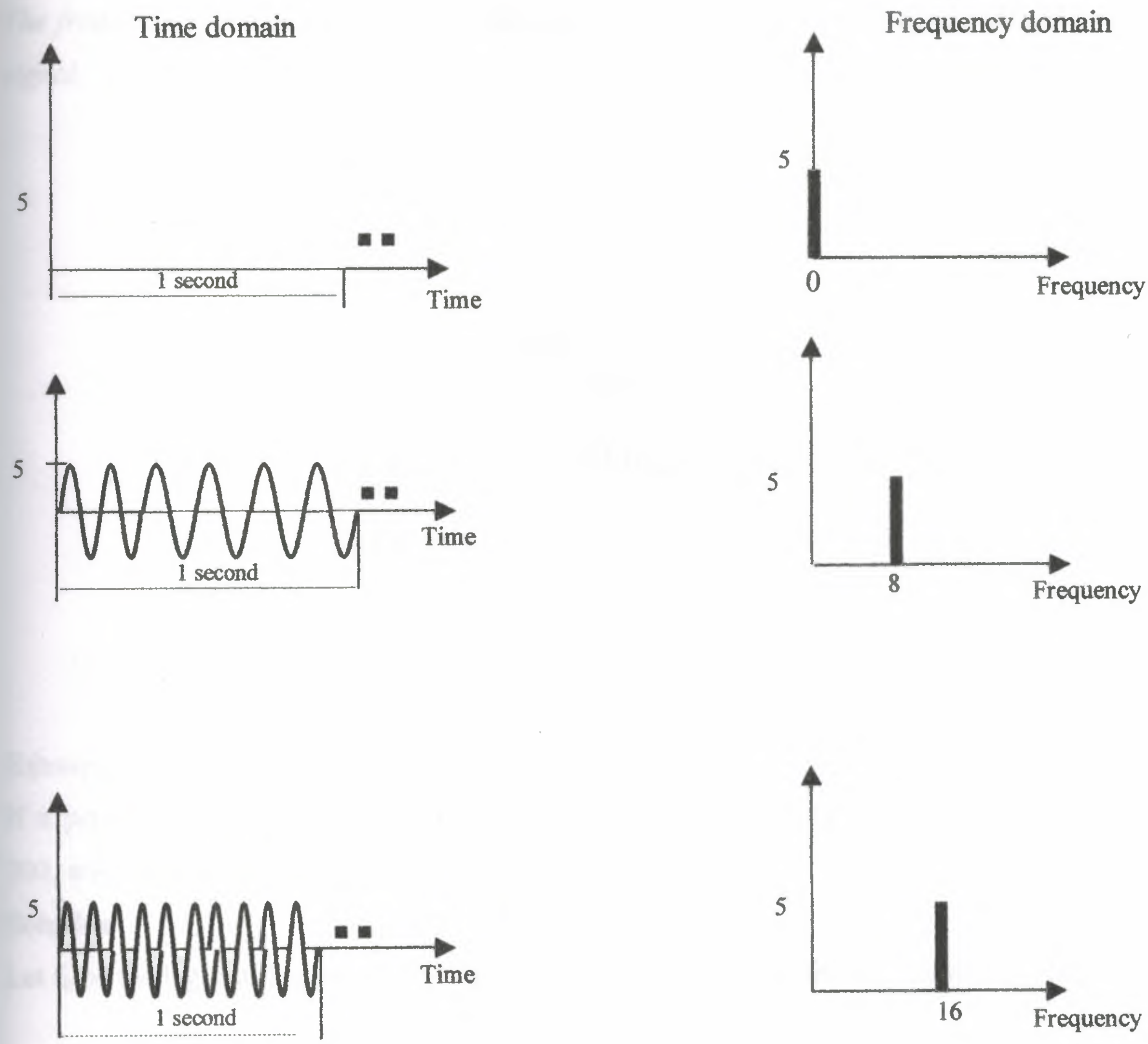


Figure 1.7 Time and frequency domains for different signals

1.3.4 Frequency Spectrum and Bandwidth

Two terms need mentioning here: spectrum and bandwidth. The frequency spectrum of a signal is the collection of all the component frequencies it contains and is shown using a frequency domain graph. The bandwidth of a signal is the width of the frequency spectrum (see figure 1.8). In other words, bandwidth refers to the range of component frequencies, and frequency spectrum refers to the elements within that range. To calculate the bandwidth, subtract the lowest frequency from the highest frequency of the range.

The frequency spectrum of a signal is the combination of all sine wave signals that make that signal.

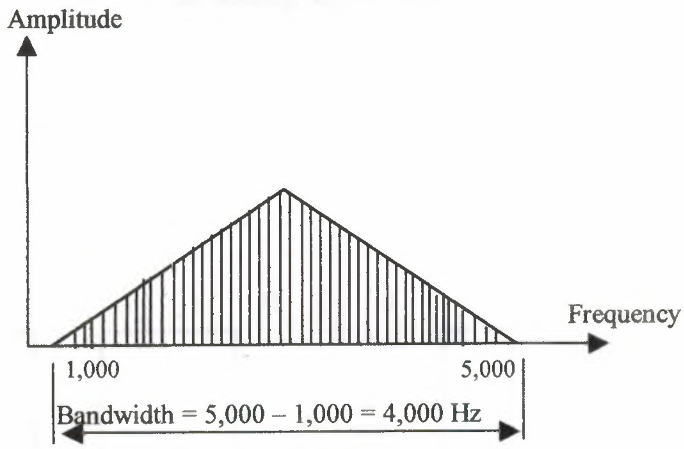


Figure 1.8 Bandwidth

Example 1.3

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is the bandwidth?

Solution

Let f_h be the highest frequency, f_l be the lowest frequency, and B be the bandwidth. Then,

$$B = f_h - f_l = 900 - 100 = 800\text{Hz}$$

Example 1.4

A signal has a bandwidth of 30Hz. The highest frequency is 60 KHz. What is the lowest frequency?

Solution

Let f_h be the highest frequency, f_l be the lowest frequency, and B be the bandwidth. Then,

$$B = f_h - f_l$$

$$20 = 60 - f_l$$

$$f_l = 60 - 20 = 40 \text{ KHz}$$

1.4 Digital Signals

In addition to being represented by an analog signal, data can also be represented by a digital signal. See Figure 1.9.

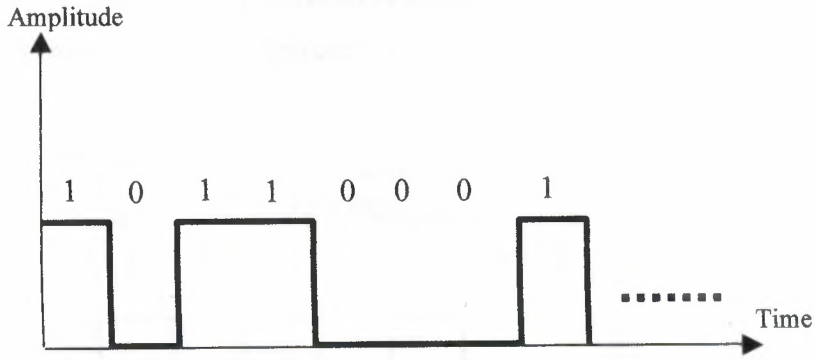


Figure 1.9 A digital signal

1.4.1 Amplitude, Period, and Phase

The three characteristics of periodic analog signals (amplitude, period, and phase) can be redefined for a periodic digital signal. (see Figure 1.10).

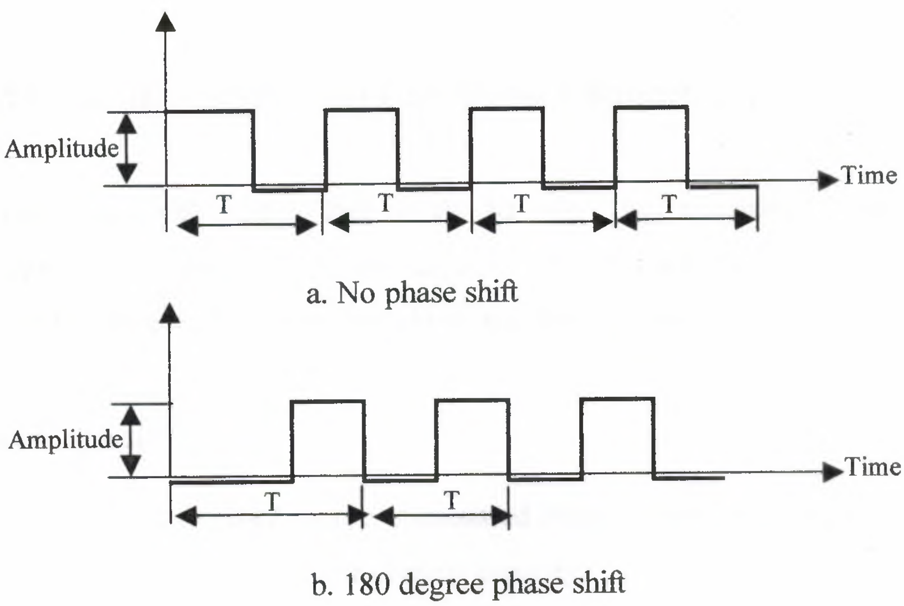


Figure 1.10 Amplitude, period, and phase for a periodic digital signal

1.4.2 Bit Interval and Bit Rate

Most digital signals are aperiodic and thus period or frequency is not appropriate. Two new terms, bit interval (instead of period) and bit rate (instead of frequency) are used to describe digital signals. The bit interval is the time required to send one single bit. The bit rate is the number of bit interval per second. This means that the bit rate is the number of bits sent in one second, usually expressed in bps. See Figure 1.11.

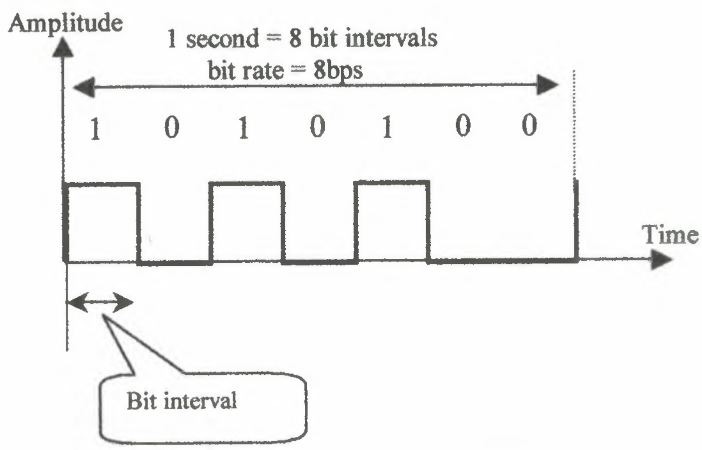


Figure 1.11 Bit rate and bit interval

1.4.3 Medium Bandwidth and Data Rate: Channel Capacity

The medium bandwidth puts a limit on the bit rate. The maximum bit rate a transmission medium can transfer is called channel capacity of the medium. The capacity of a channel depends on the type of encoding technique and the signal-to-noise ratio of the system (see Figure 1.12).

Channel Capacity

The rate at which data can be transmitted over a given communication channel, under given conditions, is referred to as the channel capacity.

There are four concepts here that we are trying to relate to one another.

- Data rate: This is the rate, in bits per second (bps), at which data can be transmitted.
- Bandwidth: This is the bandwidth of the transmitted signal as constrains by the transmitter and the nature of the transmission medium, expressed by Hertz.

- Noise: The average level of noise over the communications path.
- Error rate: The rate at which errors occur, where an error is the reception of a 1 when a 0 was transmitted or the reception of a 0 when a 1 was transmitted.

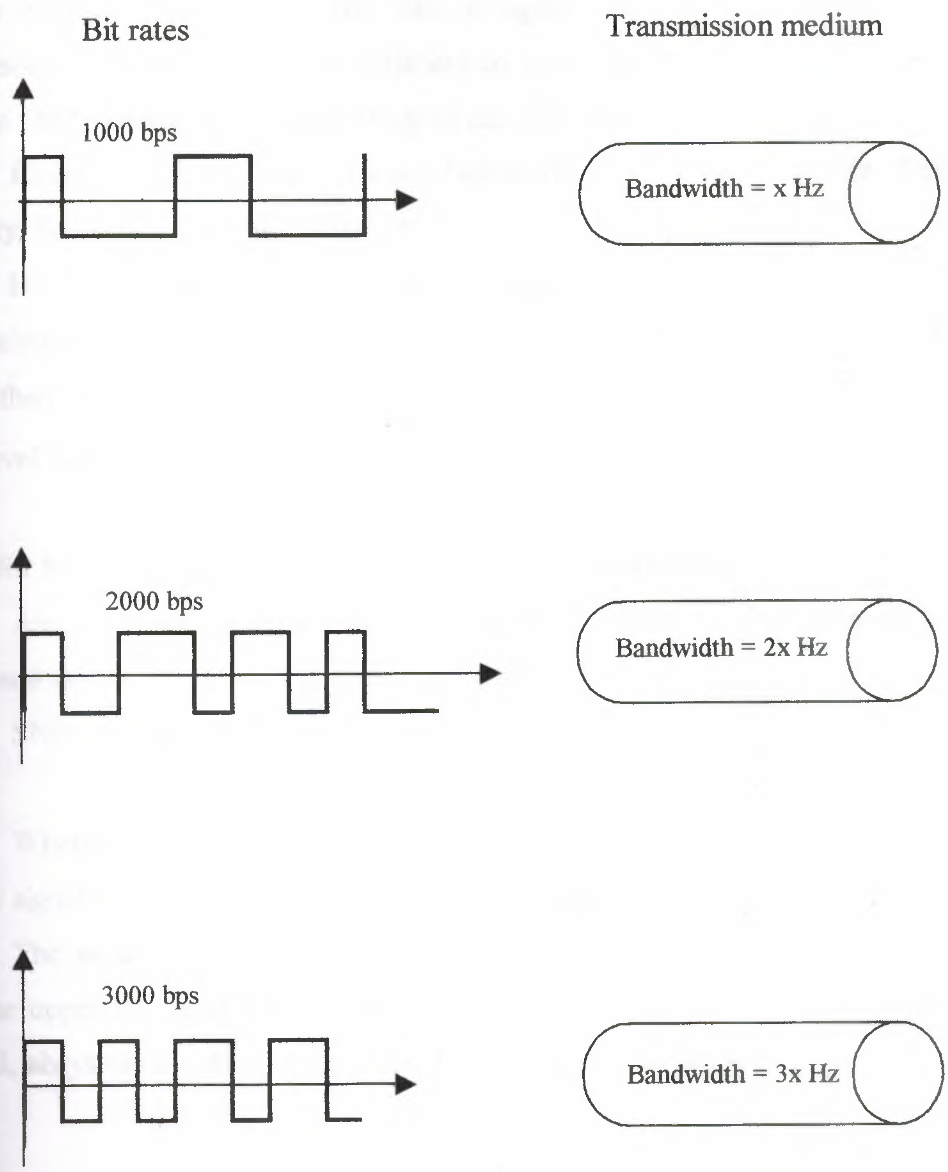


Figure 1.12 Medium bandwidth and data rate

Communication facilities are expensive and, in general, the greater the bandwidth of a facility the greater the cost. Furthermore, all transmission channels of any practical interest are of limited bandwidth. The limitations arise from the physical properties of the transmission medium or from deliberate limitations at the transmitter on the bandwidth to prevent interference from other sources. Accordingly, we would like to make as efficient use as possible of a given bandwidth.

Let us consider the case of a channel that is noise-free. In this environment, the limitation on data rate is simply the bandwidth of the signal. A formulation of this limitation, due to Nyquist, states that if the rate of signal transmission is 2W, then a signal with frequencies no greater than W is sufficient to carry the data rate. The conserve is also true: Given a bandwidth of W, the highest signal rate that can be carried is 2W.

Example 2.2. A voice channel bandwidth is of W = 3100 Hz. Find the channel capacity. Solution: $C = 2\,W = 6200$ bps.

However, as we shall see signals with more than two levels can be used; that is each signal element can represent more than one bit. For example; if M possible voltage levels are used, then each signal element can be represented by $n = \log_2 M$ numbers of bits. With multilevel signaling, the Nyquist formulation becomes.

$$C = 2\,W \log_2 M$$

Thus, for M = 8, a value used with some modems, C becomes 18600 bps.

An important parameter associated with a channel is a signal-to-noise ratio (SNR) expressed as

$$SNR = 10\log_{10} (S/N)\ dB$$

Where S/N – signal –to- noise powers ratio. Clearly a high S/N will mean a high - quality signal and a low number of required intermediate repeaters.

The signal - to noise ratio is important in the transmission of digital data because it sets the upper bound on the achievable data rate. The maximum channel capacity, in bits per second, obeys the equation attributed as the Shannon – Hartley law

$$C = W \log_2 (1+ S/N)$$

Example 4.5

Consider a voice channel with bandwidth of 3000 Hz. A typical value of S/N for a telephone line is 20 dB.

Solution

$$S/N = W \log_{10}(1+S/N) = 3{,}32\ W\log_{10} (1+S/N),\ S/N = 100$$

$$C = 3000 \log_2 (1 + 100) = 19963 \text{ bps}$$

2.MULTIPLEXING

Multiplexing

Whenever the transmission capacity of a medium linking two devices is greater than the transmission needs of the devices, the link can be shared, much as a larger water pipe can carry water several separate houses at once. Multiplexing is the set of techniques that allows the simultaneous transmission of multi signals across multiple signals across a single data link.

As a data and telecommunications usage increases, so does the traffic. We can accommodate this by continuing to add individual lines each time a new channel is needed, or we can install higher capacity links and use each to carry multiple signals. Nowadays technology includes high-bandwidth media such as coaxial cable, optical fiber, and terrestrial and satellite microwaves.

Each of these has a carrying capacity of a link is greater than the transmission needs of devices is connected to it, the access capacity is wasted. An efficient system maximizes the multizations of all facilities. In addition, the expensive technology involved often becomes cost-effective only when links are shared.

Figure 2.1 shows two possible ways of linking four pairs of devices. In figure 2.1*a* each pair has its own link. If the full capacity of each link is not being utilized, a portion of that capacity is being wasted. In figure 2.1b transmissions between the pairs are multiplexed; the same four pairs share the capacity of a single link.

2.1 Many to One/One to Many

In a multiplexed system, n devices share the capacity of one link. In figure 2.1*b* the basic of multiplexed system. The four devices on left direct their transmission streams to Multiplexer (MUX), which combines them into single stream (many to one). At the receiving end, that stream is fed into a Demultiplexes (DEMUX), which separates the stream back into its component transmissions (one to many) and directs them to their receiving devices.

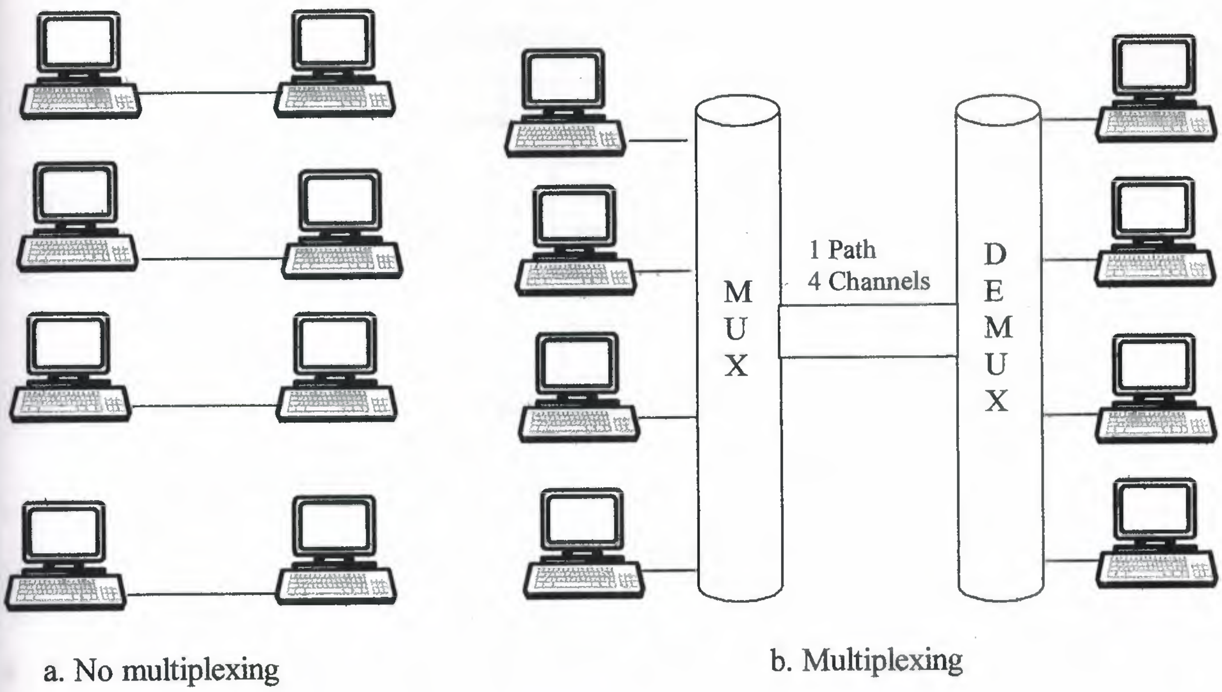


Figure 2.1 Multiplexing versus no multiplexing

In figure 2.1*b* the word *path* refers to the physical link. The word *channel* refers to a portion of a path that carries transmission between a given pair of devices. One path can have many (n) channels.

2.2 Types of Multiplexing

Signals are multiplied using two basic techniques: frequency-division multiplexing **(FDM)** and time-division multiplexing **TDM** (usually called **TDM**) and synchronous **TDM**, also called statistical **TDM** or concentrator (see figure 2.2).

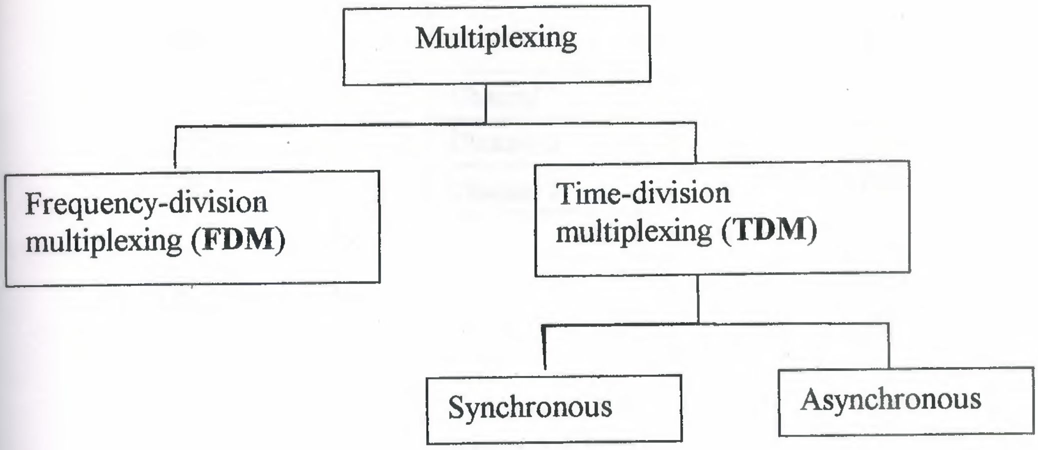


Figure 2.2 Categories of multiplexing

2.2.1 Frequency-division multiplexing (FDM)

Frequency-division multiplexing **(FDM)** is an analog technique that can be applied when the bandwidth of a link greater than the combined bandwidth of the signals to be transmitted.

In **FDM**, signals generated by each sending device modulate different carrier frequencies. These modulated signals then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by enough bandwidth to accommodate the modulated signal.

These bandwidth ranges are the channels through which the various signals travel. Channels must be separated by strips of unused bandwidth (guard bands) to prevent signals from overlapping.

In addition, carrier frequencies must not interfere with the original data frequencies. Failure to adhere to either condition can result in Unrecoverability of the original signals.

Figure 2.3 gives conceptual view of **FDM**. In this installation, the transmission path is divided into three parts, each representing a channel to carry one transmission.

As an analogy, imagine a point where three narrow streets merge to form three–lane highway. Each car merging into the highway from one of the streets still has its own lane and can travel without interfering with cars in other lanes.

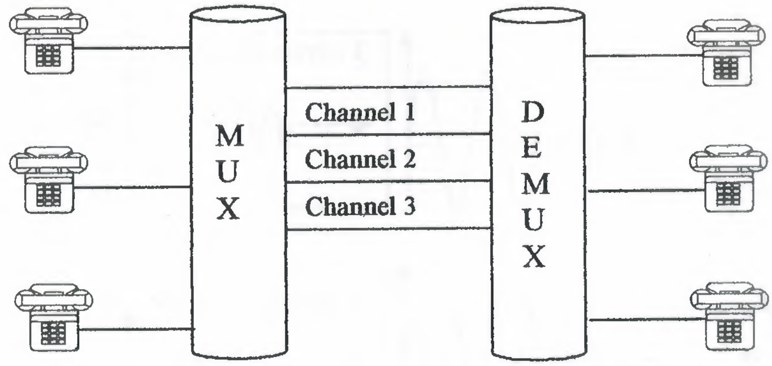


Figure 2.3 FDM

Keep in mind that although figure 2.3 shows the path as divided spatially into seperate channels, actual channel divisions are achieved by frequency rather than by space.

2.2.1.1 The FDM process

Figure 2.4 is a conceptual time-domain illustration of multiplexing process. FDM is an analog process and we show it here using telephone as the input and output devices. Each telephone generates a signal of similar frequency range. Inside the Multiplexer, these similar signals are modulated onto different carrier frequencies($f_1 f_2$, and f_3).

The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.

Figure 2.5 is the frequency-domain illustration for the same concept. (note that the horizontal axis of this figure denotes frequency, not time. All three-carrier frequencies exist the same time within the bandwidth.) In FDM, signals are modulated onto seperate carrier frequencies (f_1,$f_2$and f_3) using either **AM** or **FM** modulation.

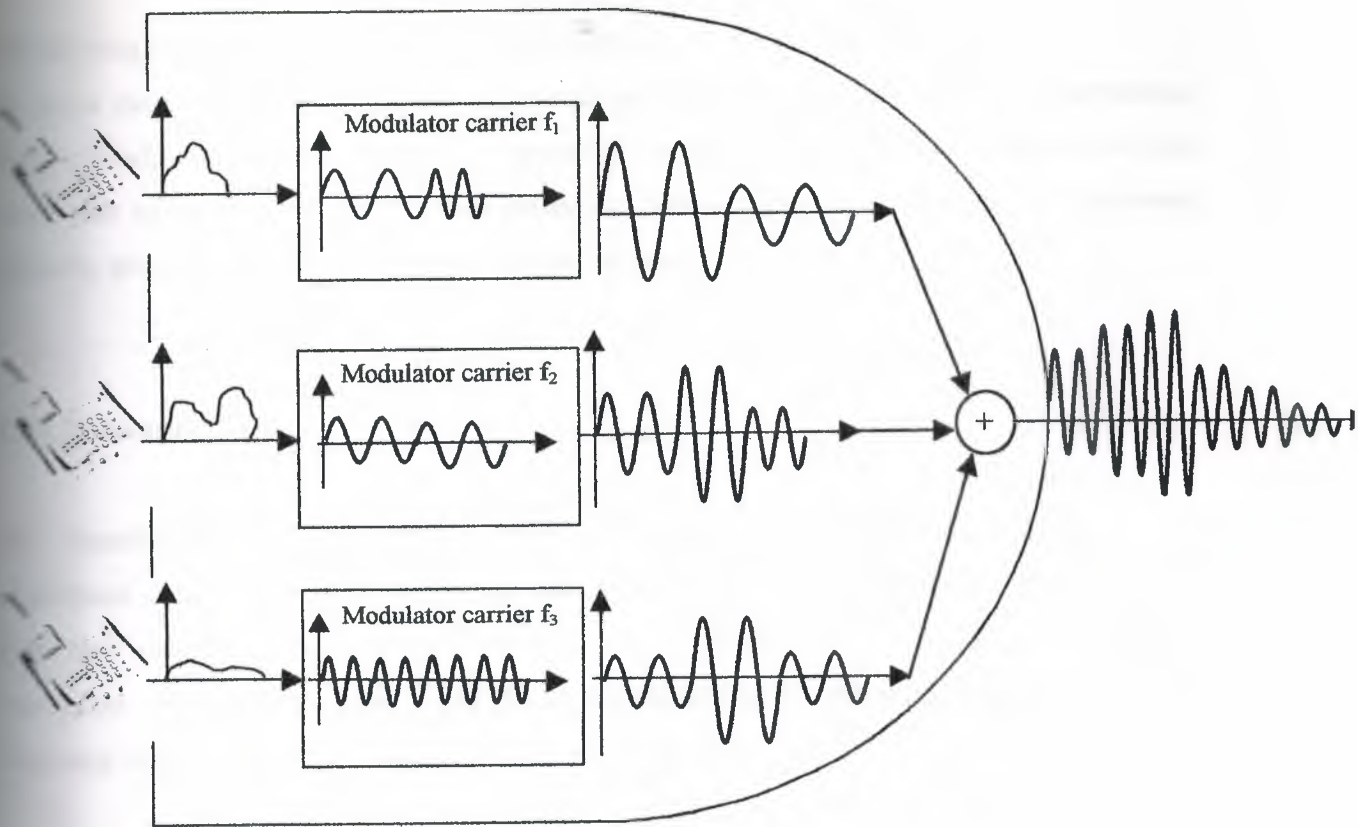


Figure 2.4 FDM multiplexing process, Time-domain

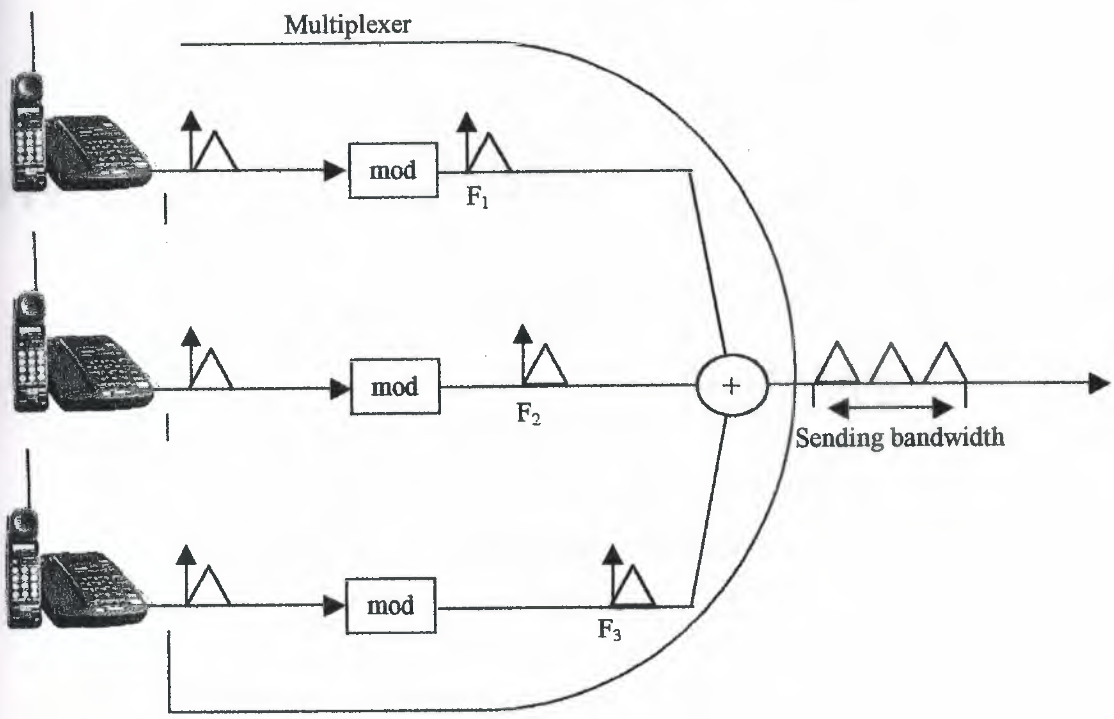


Figure 2.5 FDM multiplexing process, frequency-domain

Modulating one signal to another results in bandwidth of at least twice the original. To allow more efficient use of the path, the actual bandwidth can be lowered by suppressing half the band, using techniques that are beyond the scope of third book. In this illustration, the bandwidth of each input signal: three times the bandwidth to accommodate the necessary channels, plus extra bandwidth to allow for the necessary guard bands.

2.2.1.2 Demultiplexing

The Demultiplexer uses a series of filters to decompose the multiplexed signal into its constituent signals. The individual signals are then passed to demodulator that separates them from their carriers and passes them to the waiting receivers, figure 2.6 is a time–domain illustration multiplexing, again using three telephones as a communication devices. The frequency-domain of the same example is shown in figure 2.7.

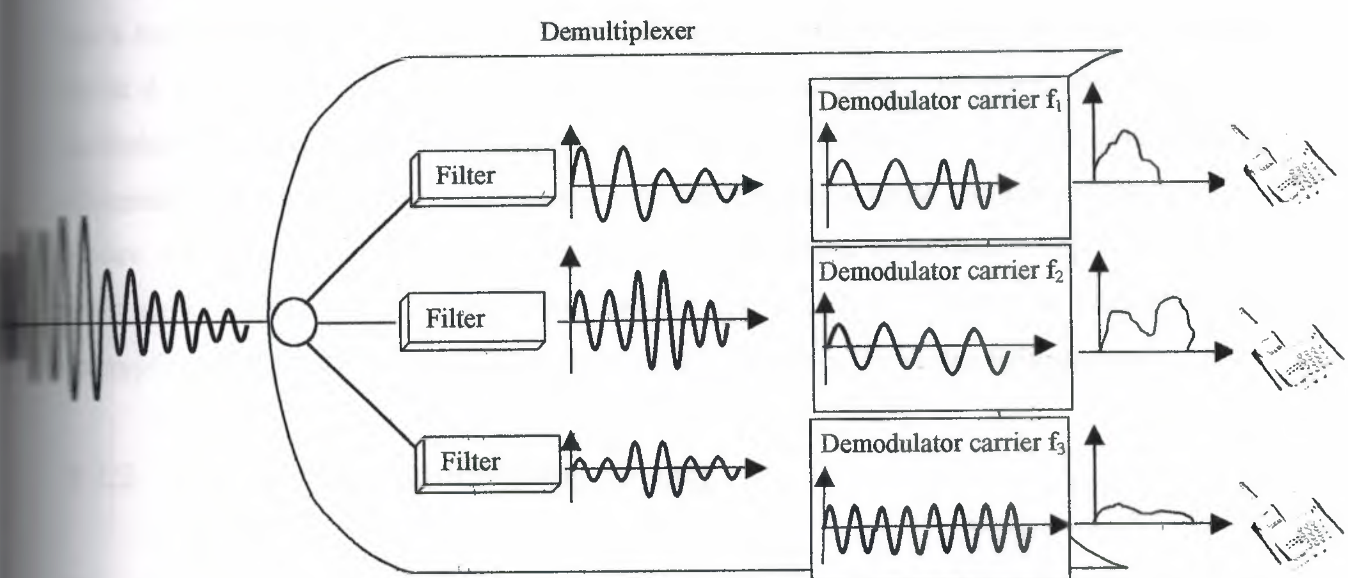


Figure 2.6 FDM demultiplexing process, time-domain

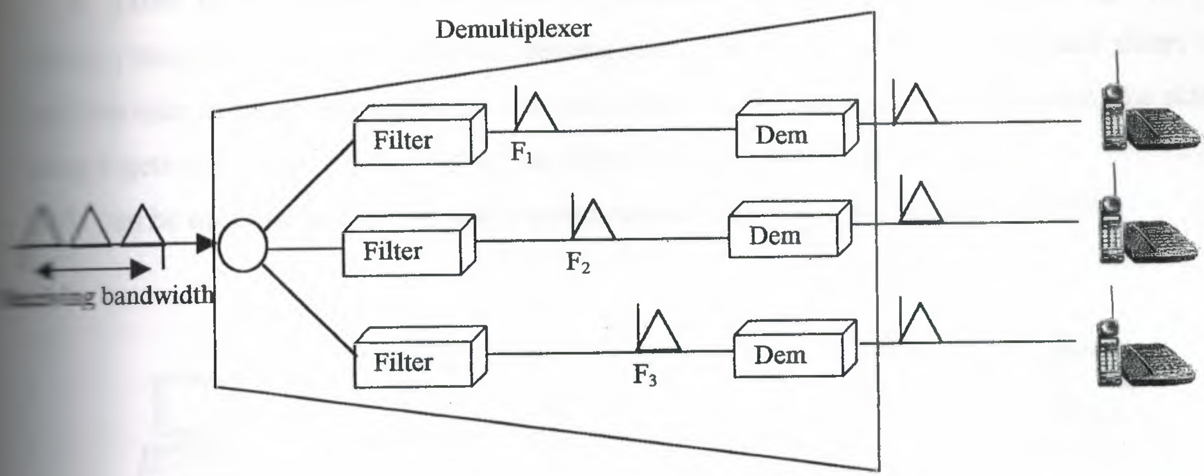


Figure 2.7 FDM Demultiplexing, frequency-domain

Example2.1 : cable television

A familiar application of FDM is cable television. The coaxial used in cable television system has a bandwidth system of approximately 500 MHz. An individual television channel requires about 6 MHz of bandwidth for transmission. The coaxial cable, therefore, can carry many multiplexed channels (theoretically 83 channels, but actually fewer to allow for guard bands). A demultipexer at your television allows you to select which of those you wish to receive. Today, a new and more efficient method is being developed to implement FDM over fiber-optic cable. Called wavelength division multiplexing (WDM), it uses essentially the same concepts as FDM but incorporates the range of frequencies in the visible light spectrum.

2.2.2 Time-division multiplexing (TDM)

Time-division multiplexing (**TDM**) is a digital process that can be applied when the data rate of transmission medium is greater than the data required by the sending and receiving devices. In such a case, multiple transmission can occupy a single link by subdividing them and interleaving the portions.

Figure 2.8 gives a conceptual view of **TDM**. Note that the same link is used as in the FDM; here, however, the link is shown sectioned by time rather than frequency.

In **TDM** figure, portions of signals 1,2,3 and 4 occupy the link sequentially. As an analogy, imagine a ski lift that serves several runs. Each run has its own line and skiers in each line take turns getting on the lift. As each chair reaches the top of the mountain, the skier riding it gets off and skis down the run for which he or she waited in line.

TDM can be implemented in two ways: **synchronous TDM** and **asynchronous TDM.**

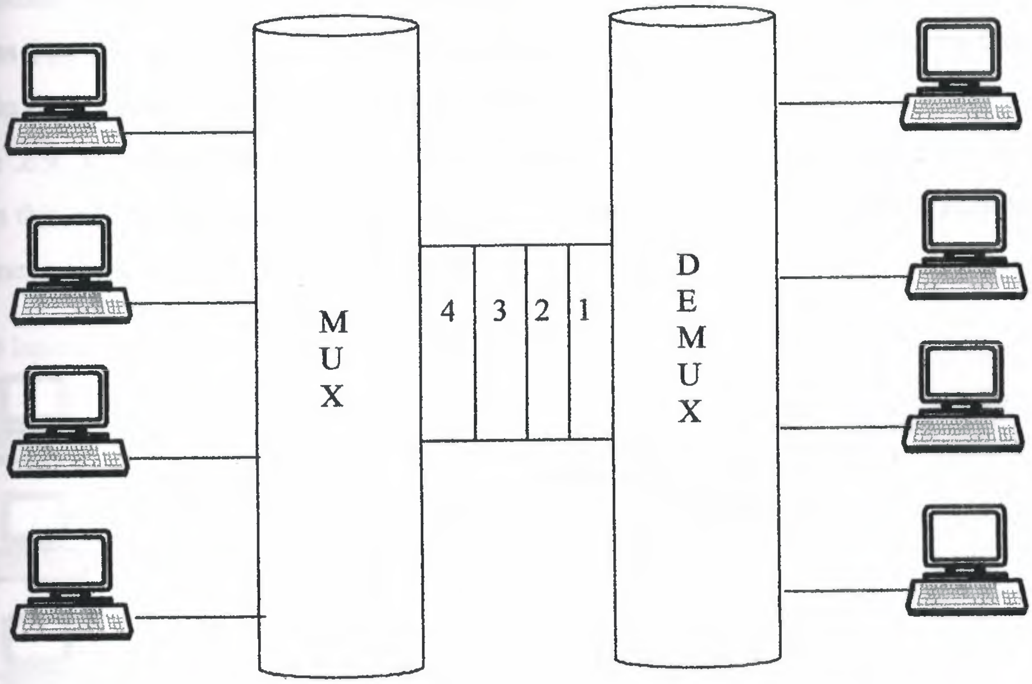


Figure2.8 TDM

2.2.2.1 Synchronic TDM

In time-division multiplexing, the term *synchronous* has a different meaning from that used in areas of telecommunications. Here synchronous means that the Multiplexer allocates exactly the same slot to each device at all times, whether or not a device has anything to transmit. Time slot, for example, is assigned to device A alone and cannot be used by any other device. Each time its allocated time slot comes up, a device has the opportunity to send a portion of its data. If a device is unable to transmit or does not have data to send, its time slot remains empty.

Frames time slot are grouped into frames. A frame consists of one complete cycle time slots, including one or more slots dedicated to each sending device, plus framing bits (see figure 2.12). in a system with n input lines, each frame has at least n slots, with each slot allocated to carry data from a specific input line. If all the input devices sharing the link are transmitting at the same data rate, each device has one time slot per frame. However, it is possible to accommodate varying data rates. A transmission with two slots per frame will arrive twice quickly as one slot per frame. The time slots dedicated to a given device occupy the same location in each frame constitute that device's channel.

In figure 2.9, we show five input lines multiplexed onto a single path using synchronous TDM. In this example, all of the inputs have the same data rate, so the number of time slots in each frame is equal to the number of input lines.

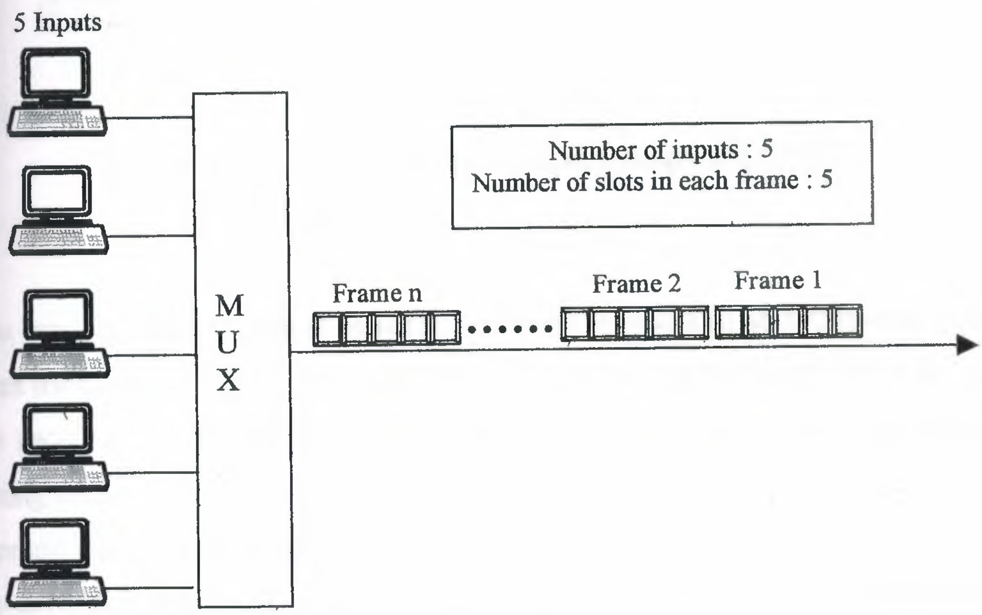


Figure 2.9 Synchronous TDM

Interleaving synchronous TDM can be compared to a very fast switch. As the switch opens in front of a device, that device has the opportunity to send a specified amount(x bits) of data onto the path. The switch moves from device to device at constant rate and in a fixed order. This process is called interleaving.

Interleaving can be done by bit, by byte, or by any other data unit. In other words, the Multiplexer can take one byte from each device, then another byte from each device, and so on. In a given system, he interleaved units will be always the same size.

Figure 2.10 shows interleaving and frame building. In the example, we interleave the various transmissions by character (equal to one byte each), but the concept is the same for data units of any limit.

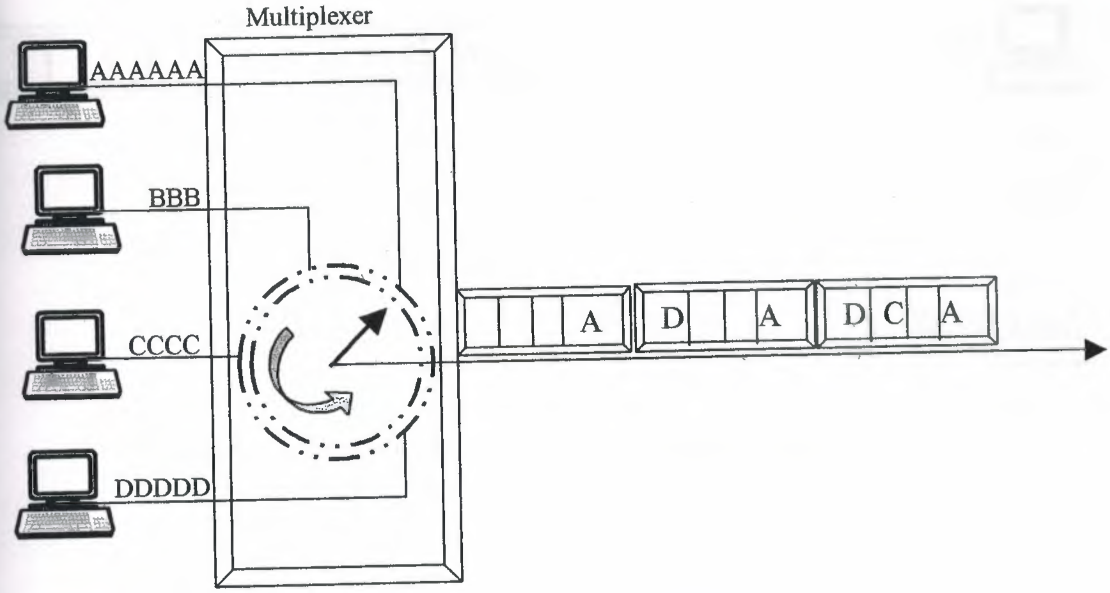


Figure 2.10 Synchronous TDM, multiplexing process

As you can see, each device is sending a different message. The Multiplexer interleaves the different messages and forms them into frames before putting them onto the link.

At the receiver the Multiplexer decomposes each frame by discarding the framing bits and extracting each character in turn. As a character is removed from the frame, it is passed to the appropriate receiving device (see figure 2.11).

Figure 2.10 and figure 2.11 also point out the major weakness of synchronous TDM. By assigning each time slot to a specific input line, we end up with empty slots whenever not all the lines are active. In figure 2.10, only the first three frames are completely filled.

The last three frames have a collective six empty slots. Having 6 empty slots out of 24 means that a quarter of capacity of the link is being wasted.

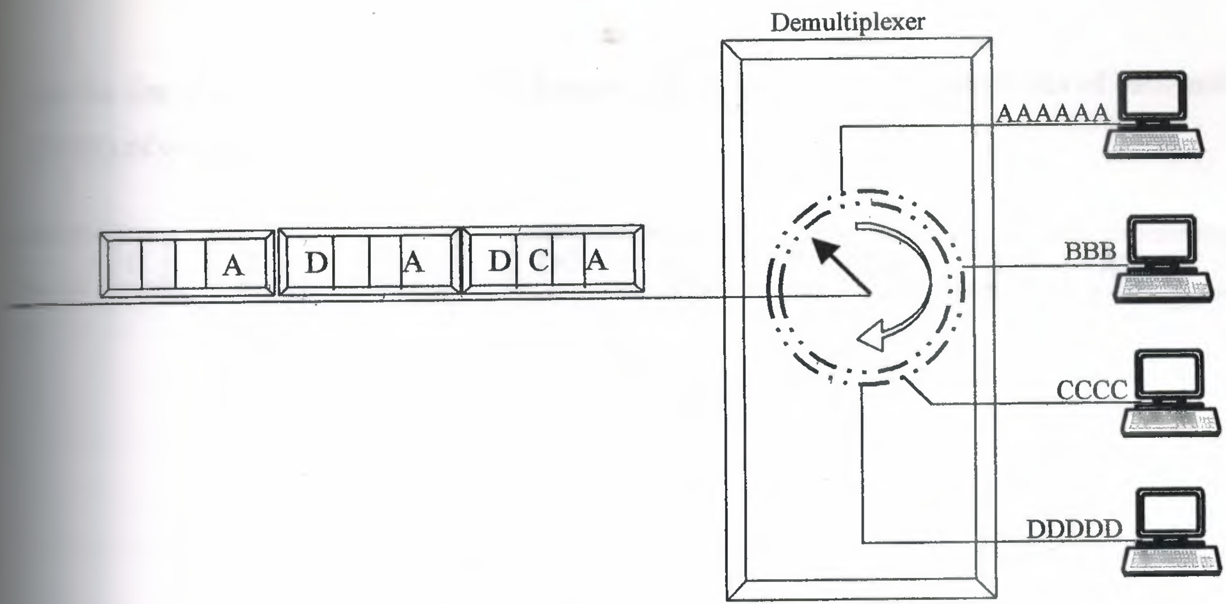


Figure 2.11 Synchronous TDM, Demultiplexing process

Framing bits

Because the time slot order in a synchronous TDM system does not vary from frame to frame, very little overhead information needs to be included in each frame. The order of receipt tells the Demultiplexer where to direct each time slot, so no addressing is necessary. Various factors, however, can cause timing inconsistencies. For this reason, one or more synchronization bits are usually added to the beginning of each frame. These bits, called framing bits, follow a pattern, frame to frame, that allows the Demultiplexing to synchronize with the incoming stream so that it can separate the time slots accurately. In most cases, this synchronization information consists of one bit per frame, alternating between 0 and 1 (01010101010), as shown in figure 2.12.

Synchronous TDM Example 2.2:

Imagine that we have four input sources on a synchronous TDM link, where transmissions are interleaved by character. If each source is creating250 characters per second, and each frame is carrying 1 character from each source, the transmission path must be able to carry 250 frames per second (see figure 2.13).

If we assume that each character consists of eight bits, then each frame is 33 bits long: 32 bits for the four characters plus 1 framing bit. Looking at the bit relationships, we see that each device is creating 200 bps (250 characters with 8 bits per character), that the line is carrying 8250 bps (250 frames with 33 bits per frame): 8000 bits of data and 250 bits of overhead.

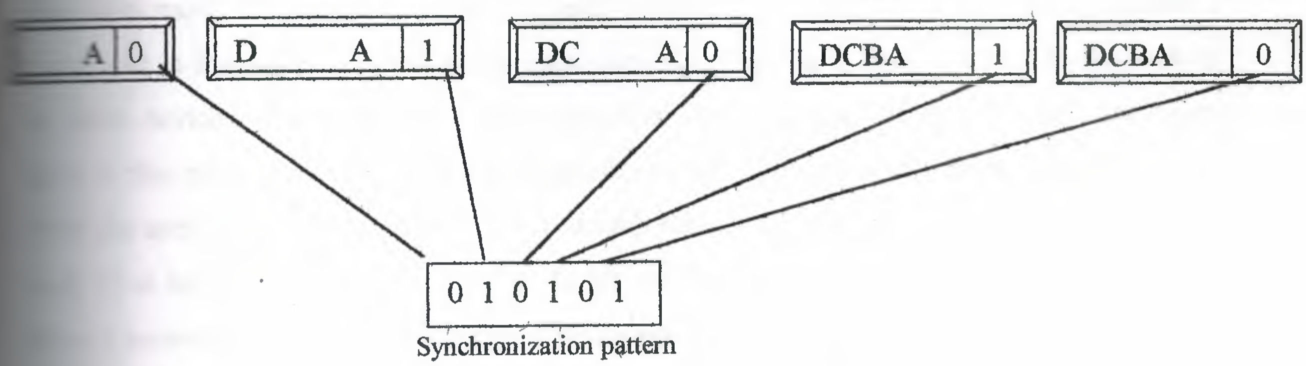


Figure 2.12 Framing bits

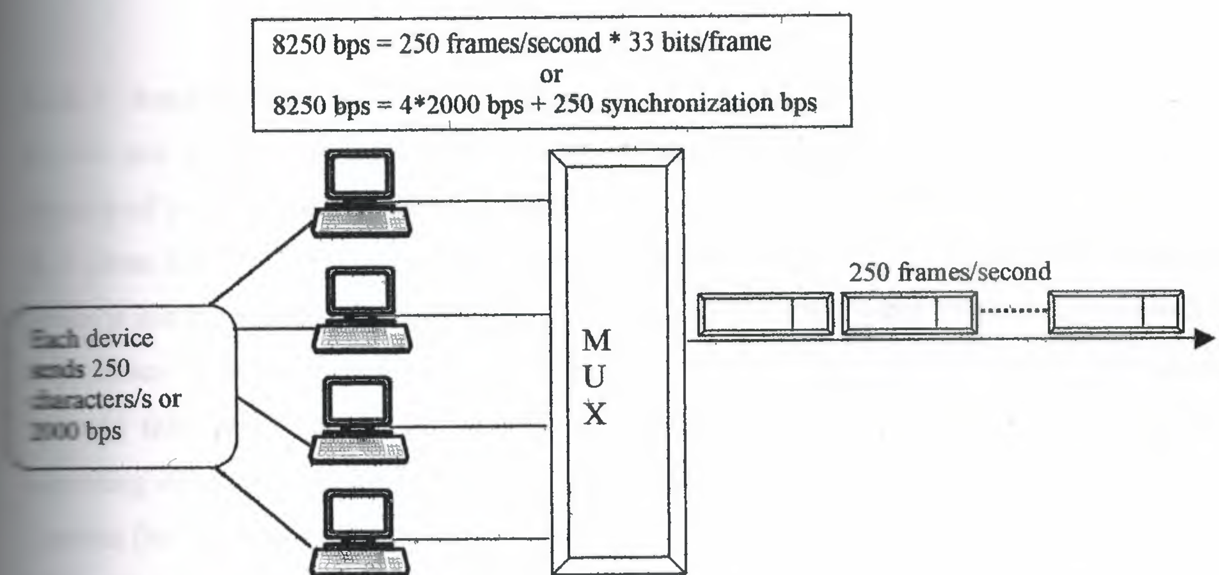


Figure 2.13 Data rate calculation for frames

Bit Stuffing

As noted above, it is possible to connect derives of different data rates to a synchronous TDM. For example, device A uses one time slot, while the faster device B uses two. The number of slots in a frame and the input lines to which they are assigned remains fixed throughout a given system;

and devices of different data rates may control different numbers of those slots. But remember that time slot length is fixed. For this technique to work, therefore, the different data rates must be integer multiples of each other. For example, we can accommodate a device that is five times faster than the other devices by giving it five slots to one for each of the other devices. We cannot, however, accommodate a device that is five and a half times faster by this method, because we cannot introduce half a time slot into a frame.

When the speeds are not integer multiples of each other, they can be made to behave as if they were, by a technique called bit stuffing. In bit stuffing, the Multiplexer adds extra bits to a device's source stream to force the speed relationships among the various devices into integer multiples of each other. For example, if we have one device with a bit rate of 2.75 times that of the other devices, we can add enough bits to raise the rate to 3 times that of the others. The Demultiplexer then discards the extra bits.

2.2.2 Asynchronous TDM

As we saw in the previous section, synchronous TDM does not guarantee that the full capacity of a link is used. In fact, it is more likely that only a portion of the time slots is in use at a given instant. Because the time slots are reassigned and fixed, whenever a connected device is not transmitting the corresponding slot is empty and that much of the path is wasted.

Asynchromous time-division multiplexing, or statistical time- division multiplexing is designed to avoid this type of waste. As with the term *synchronous*, the term *asynchronous* means something different in multiplexing than it means in other areas of data communications. Here it means flexible or not fixed.

Like synchronous TDM, asynchronous TDM allows a number of lower speed input lines to be multiplexed to a single higher speed line. Unlike synchronous TDM, however, in synchronous TDM the total speed of the input lines can be greater than the capacity of the path. In synchronous system , if we have n input lines, the frame contains a fixed number of at least n time slots. In asynchronous system, if we have n input lines, the frame contains no more than m slots, with m less than n (see figure 2.14). In this way, asynchronous TDM supports the same number of input lines as asynchronous TDM with a lower capacity link. Or, given the same link, asynchronous TDM can support more devices than synchronous TDM.

The number of time slots in an asynchronous TDM frame (m) is based on a statistical analysis of the number of input lines that are likely to be transmitting at any given time.

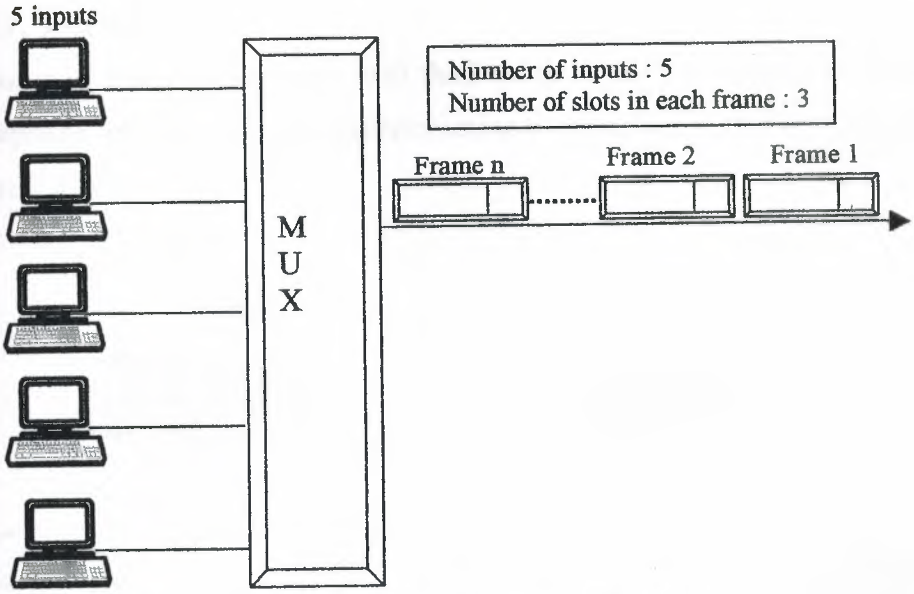


Figure 2.14 Asynchronous TDM

Rather than being reassigned, each slot is available to any of the attached input lines that has data to send. The Multiplexer scans the input lines, accepts portions of data until a frame is filled, and then sends the frame across the link. If there are not enough data to fill all the slots in a frame, the frame is transmitted only partially filled; thus full link capacity may not be used 100 percent of the time. But the ability to allocate time slots dynamically, coupled with the lower ratio of time slots to input lines, greatly reduces the likelihood and degree of waste.

2.3 Multiplexing Application: The Telephone System

Multiplexing has long been an essential tool of the telephone industry. A look at some telephone company basics can help us understand the application of both FDM and TDM in the field. Of course, different parts of the world use different systems. We will concentrate only on the system used in North America.

The North American telephone system includes many common carriers that offer local and long-distance services to subscribers.

These carriers include local companies, such as Pacific Bell, and long-distance providers, such as AT&T, MCI, and Sprint.

For the purposes of this discussion, we will think of these various carriers as a single entity called the telephone network, and the line connecting a subscriber to that network as a service line (see figure 2.15).

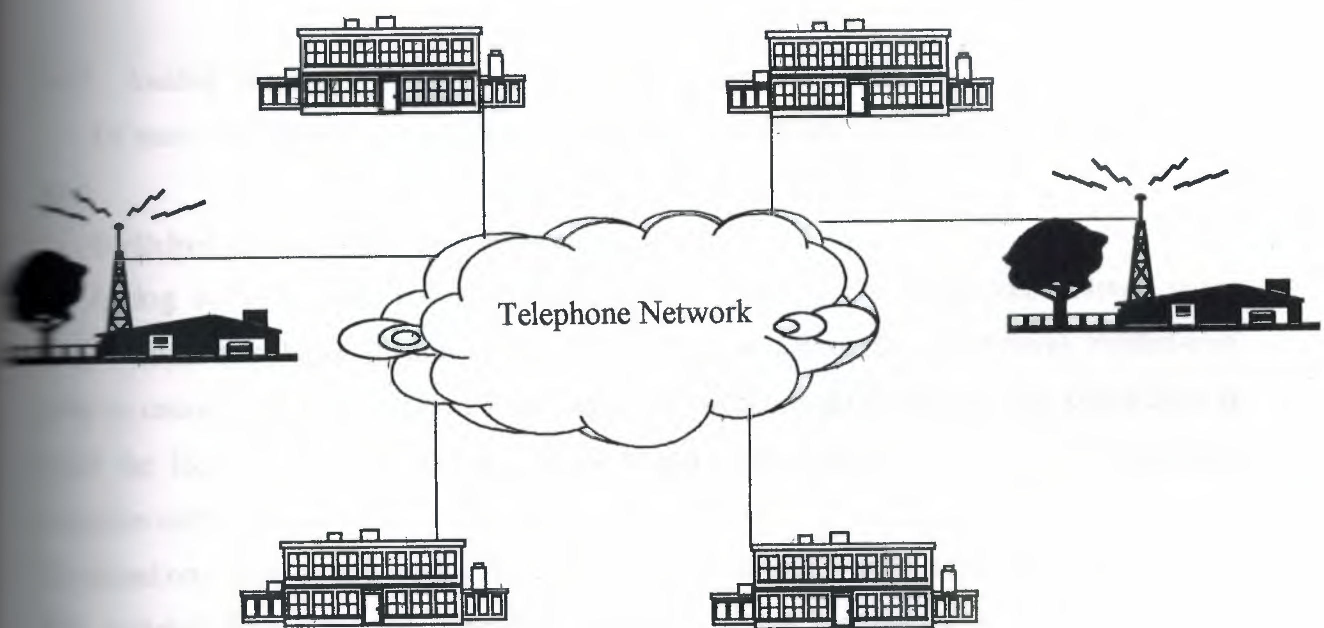


Figure 2.15 Telephone network

2.4 Common Carrier Services and Hierarchies

Telephone companies began by providing their subscribers with analog services that used analog networks. Later technology allowed the introduction of digital services and networks. Today, North American providers are in the process of changing even their service lines from analog to digital. It is anticipated that soon the entire network will be digital. For now, however, both types of services are available and both FDM and TDM are in use (see figure 2.16).

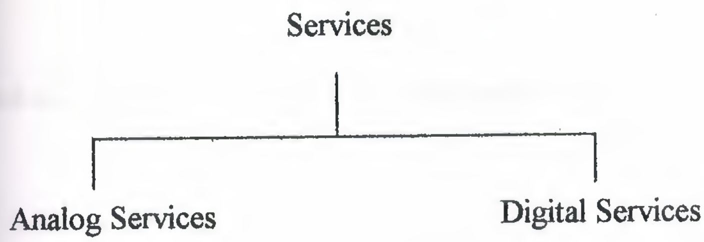


Figure 2.16 Categories of telephone services

2.4.1 Analog Services

Of many analog services available to subscribers, two are particularly to our discussion here:

Switched services and **leased** services.

Analog switched service is the familiar dial-up service most often encountered when using a home telephone. It uses two-wire (or, for specialized uses, four wire) twisted-pair cable to connect the subscriber's handset to the network via an exchange. This connection is called the local loop. The network it joins is sometimes referred to as a public switched telephone network (PSTN).

The signal on a local loop is analog, and the bandwidth is usually between 0 and 4000 Hz.

With switched lines, when the caller dials a number, the call is conveyed to a switch, or series of switches, at the exchange. The appropriate switches are then activated to link the caller's line to that of the person being called.

The switch connects the two lines for the duration of the call (see figure 2.17).

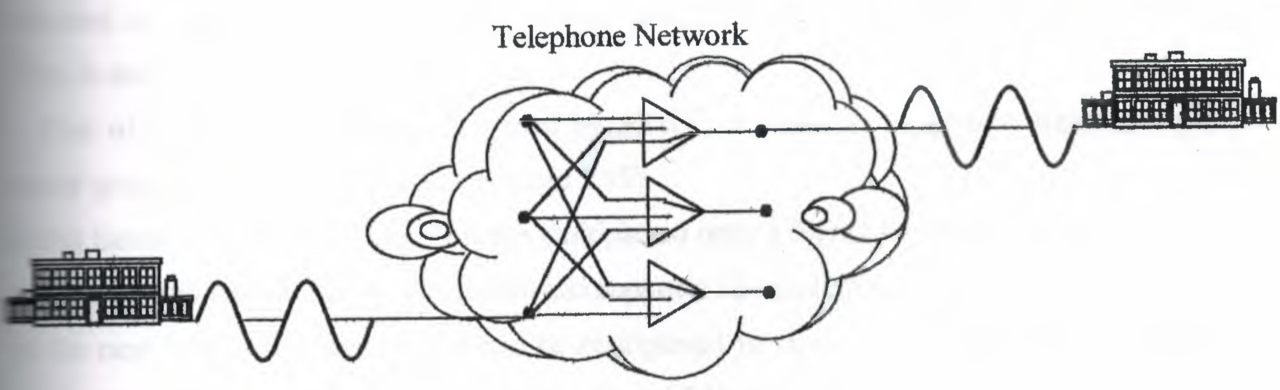


Figure 2.17 Analog switched service

The leased connection is determined by one direct connection (see fig. 2.18).

Figure 2.18 Analog leased service

Conditioned lines

Telephone carriers also offer a service called conditioning. Conditioning means improving the quality of a line by lessening attenuation, signal distortion, or delay distortion. Conditioned lines are analog, but their quality makes them usable for digital data communication if they are connected to modems.

The Analog Hierarchy

To maximize the efficiency of their infrastructure, telephone companies have traditionally multiplexed signals from lower bandwidth lines to higher bandwidth lines. In this many switched or leased lines can be combined into fewer but bigger channels. For analog lines, FDM is used.

One of these hierarchical systems used by AT&T, is made up of groups, super groups, master groups, and jumbo groups (see figure 2.19).

In this hierarchicy, 12 voice channels are multiplexed onto a higher bandwidth line to create a group. A group has 48 KHz of bandwidth and supports 12 voice channels.

At the next level, up to five groups can be multiplexed to create a composite signal called a super group. A super group has a bandwidth of 240 KHz and supports up to 60 voice channels.

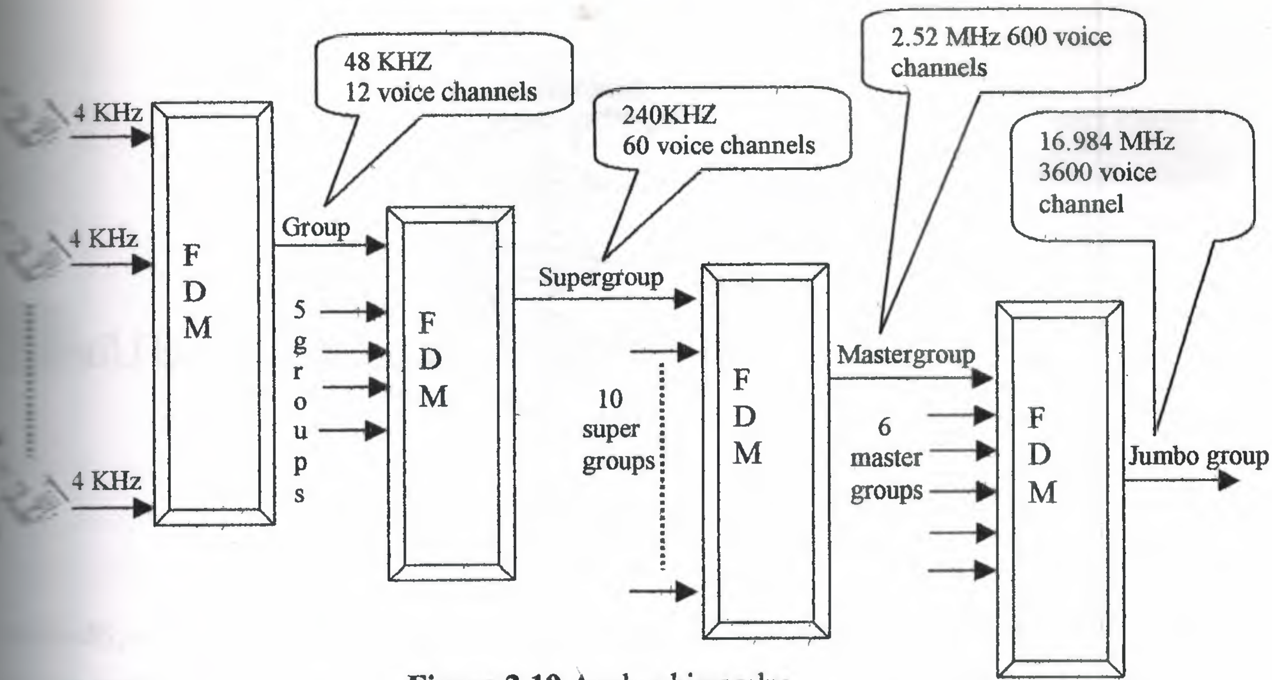


Figure 2.19 Analog hierarchy

Supergroups can be made up of either five groups or 60 independent voice channels. At the next level, 10 supergroups are multiplexed to create a master group. A master group must have 2.40 MHz of bandwidth, but the need for guard bands between the channels increases the necessary bandwidth to 2.52 MHz. Master groups support up to 600 voice channels.

Finally, six master groups can be combined into jumbo group. A jumbo group must have 15.12 MHz (6*2.52 MHz), but is augmented to 6.984 MHz to allow for guard bands between the master groups.

There are many variations of this hierarchy in the telecommunications industry. However, because this analog hierarchy will be replaced by digital services in the near future, we will limit our discussion to the system above.

2.4.2 Digital Services

Recently telephone companies began offering digital services to their subscribers. One advantage is that digital services are less sensitive than analog services to noise and other forms of interference. A telephone line acts like an antenna and will pick up noise during both analog and digital transmission. In analog transmissions, both signal and noise are analog and cannot be easily separated. In digital transmission, on the other hand, the signal is digital but

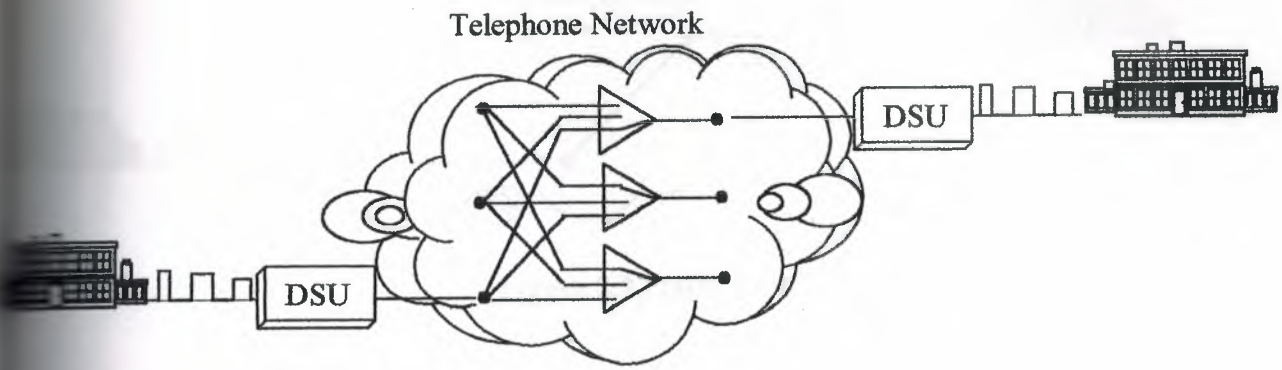


Figure 2.21 Switched/56 service

Ironically, a DSU is more expensive than a modem. So why would a subscriber elect to pay for the switched/56 service and DSU? Because the digital line has better speed, better quality, and less susceptibility to noise than an equivalent analog line.

Bandwidth on Demand

Switched/56 supports bandwidth on demand, allowing subscribers to obtain higher speeds by using more than one line. This option allows switched/56 to support video conferencing, fast facsmile, multimedia, and fast data transfer, among other feature.

2.4.2.2 Digital Data Service (DDS)

Digital data service (DDS) is the digital version of an analog leased line; it is a digital leased line.

The maximum speed available over DDS is 56 Kbps. However, a subscriber can choose among five actual rates: 2.4, 4.8, 9.6, 19.2, or 56 Kbps. Once the speed is chosen by the subscriber, it is set by the telephone company and must be observed.

Like switched/56, DDS requires the use of a DSU. The DSU for this service is cheaper than that required for switched/56, however, because it does not need a dial pad (see figure 2.22).

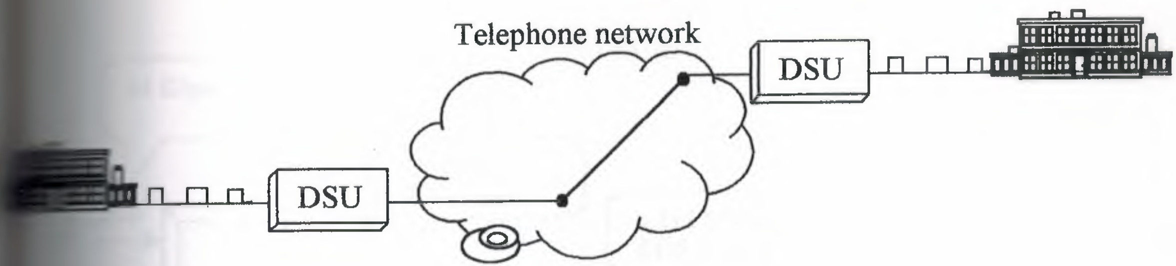


Figure 2.22 DDS service

2.4.2.3 Digital Signal Service (DS)

After offering switched/56 and DDS services, the telephone companies saw a need to develop a hierarchy of digital services much like that used for analog services. The next step was digital signal (DS) service. DS is a hierarchy of digital signals. Figure 2.23 shows the data rates supported by each level.

1. A DS-0 service resembles DDS. It is a single digital channel of 64 Kbps.
2. DS-1 is a 1.544 Mbps service; 1.544 Mbps is 24 times 64 Kbps plus 8 Kbps of overhead. It can be used as a single service for 1.544 Mbps transmissions, or it can be used to multiplex 24 DS-0 channels or to carry any other combination desired by the user that can fit within its 1.544 Mbps capacity.
3. DS-2 is a 6.312 Mbps service; 6.312 Mbps is 96 times 64 Kbps plus 168 Kbps of overhead. It can be used as a single service for 6.312 Mbps transmissions, or it can also be used to multiplex four DS-1 channels, 96 DS-0 channels, or a combination of these service types.

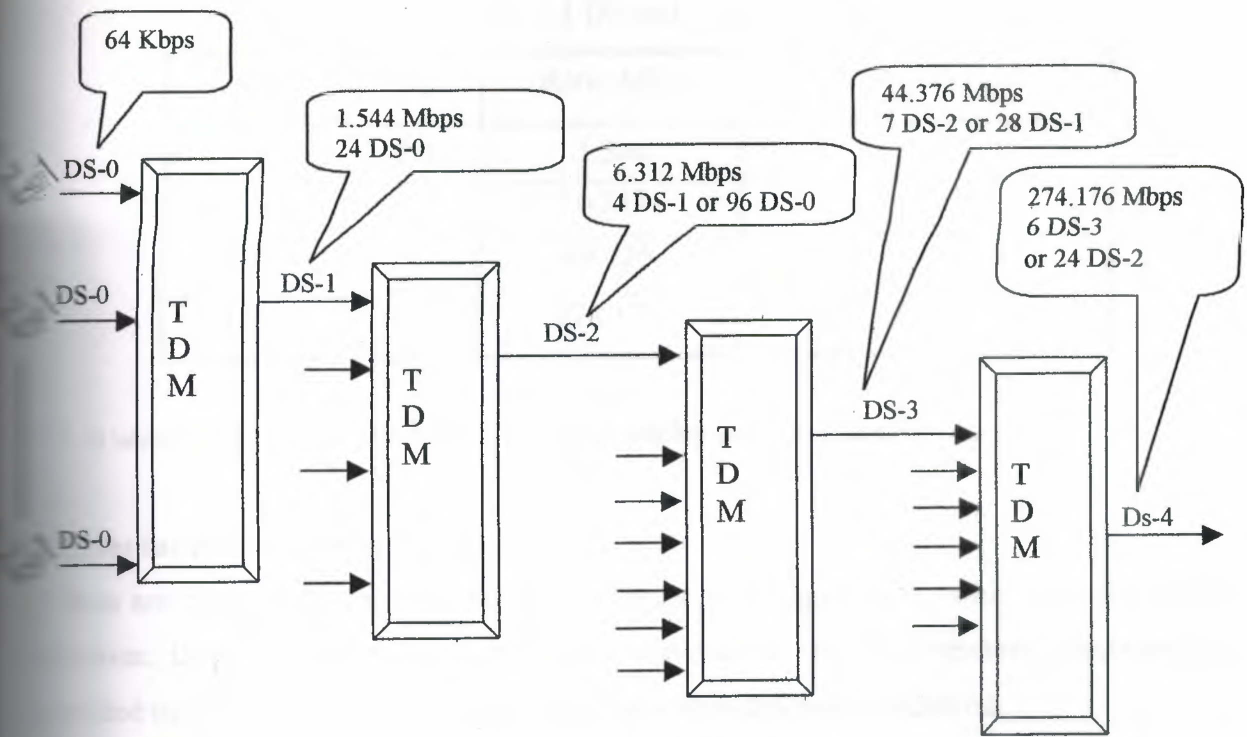


Figure 2.23 DS Hierarch

4. DS-3 is a 44.376 Mbps service; 44.376 Mbps is 672 times 64 Kbps plus 1.368 Mbps of overhead. It can be used as a single service for 44.376 Mbps transmissions, or it can be used to multiplex seven DS-2 channels, 28 DS-1 channels, 672 DS-0 channels, or a combination of these service types.
5. DS-4 is a 274.176 Mbps service; 274.176 Mbps is 4032 times 64 Kbps plus 16.128 Mbps of overhead. It can be used to multiplex six DS-3 channels, 42 DS-2 channels, 168 DS-1 channels, 4032 DS-0 channels, or a combination of these service types.

T Lines

DS-0, DS-1, and so on are the names of services. To implement those services, the telephone companies use T lines (T-1 to T-4). These are lines whose capacities are precisely matched to the data rates of the DS-1 to DS-4 services (see table 2.1).

Table 2.1 DS and T line rates

Service	Line	Rate (Mbps)	Voice Channels
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

T-1 is used to implement DS-1, T-2 is used to implement DS-2, and so on.

T Lines for Analog Transmission

T lines are digital lines designed for the transmission of digital data, voice, or audio signals. However, they can also be used for analog transmission (regular telephone transmissions), provided the analog signals are sampled first, then time-division multiplexed.

The possibility of using T lines as analog carriers opened up a new generation of services for the telephone companies. Earlier, when an organization wanted 24 separate telephone lines, it needed to run 24 twisted-pair cables from the company to the central exchange. Today, that same organization can combine the 24 lines into one T-1 line and run only the T-1 line to the exchange. Figure 2.24 shows how 24 voice channels can be multiplexed onto one T-1 line.

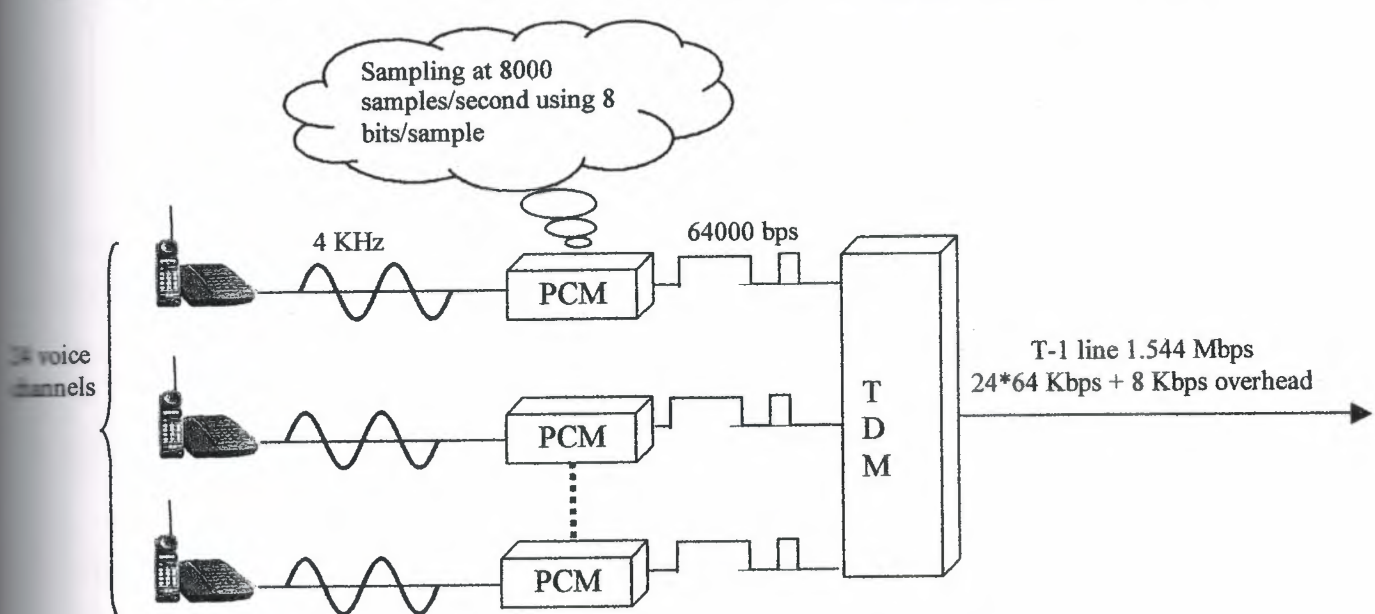


Figure 2.24 T-1 line for multiplexing telephone lines

E Lines

Europeans use a version of T lines called E lines. The two systems are conceptually identical, but their capacities differ. Table 2.2 shows the E lines and their capacities.

Table 2.2 E line rates

Line	Rate (Mbps)	Voice Channels
E-1	2.048	30
E-2	8.448	120
E-3	34.368	480
E-4	139.264	1920

3. TRANSMISSION MEDIA

3.1 Mathematical Models for Communication Channels

In the design of communication systems we find it convenient to construct mathematical models that reflect the most important characteristics of the transmission medium. Below, we provide a brief description of the channel models that are frequently used to characterize many of the physical channels that we encounter in practice.

The additive noise channel. The simplest mathematical model for a communication channel is the additive noise channel, illustrated in Figure 3.1.

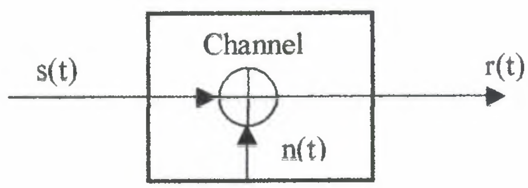


Figure 3.1 mathematical models for communication channel.

In this model, the transmitted signal s *(t)* is corrupted by an additive random noise process *n(t)*. Physically, the additive noise process may arise from electronic components and amplifiers at the receiver of the communication system, or from interference encountered in transmission as in the case of radio signal transmission.

If primarily electronic components and amplifiers at the receiver introduce the noise, it may be characterized as thermal noise. This type of noise is characterized statistically as a *Gaussian noise process.* Hence, the resulting mathematical model for the channel is usually called the *additive Gaussian noise channel.* In this case the received signal is

$$r(t) = \alpha s(t) + n(t)$$

Where α represents the attenuation factor.

The linear filter channel. In some physical channels such as wire-line telephone channels, filters are used to ensure that the transmitted signals do not exceed specified bandwidth limitations and thus do not interfere with one another. Such channel (Figure 3.2) output can be characterized as

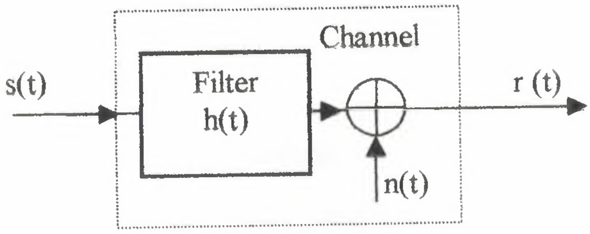


Figure 3.2 Output channel characterization

$$r(t) = s(t) * h(t) + n(t) = \int_{-\infty}^{\infty} h(\tau)s(t-\tau)d\tau + n(t)$$

Where h(t) is the impulse response of the linear filter and symbol * denotes convolution.

The linear time-variant filter channel. Physical channels such as underwater acoustic channels and ionosphere radio channels, which result in time-variant multi-path propagation of the transmitted signal, may be characterized mathematically as time-variant linear filters. A time-variant channel characterizes such system with impulse response h (τ; t) filters (Figure 3.3). For an input signal s(t), the channel output is

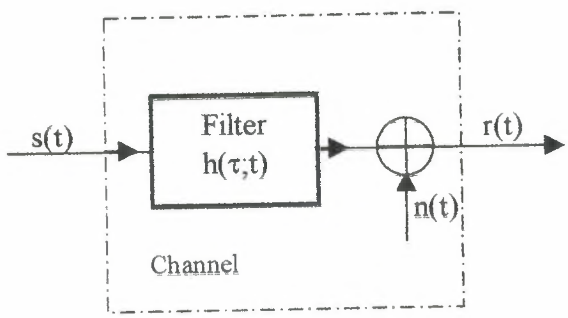


Figure 3.3 Time-variant channel with impulse response.

$$r(t) = s(t) * h(\tau ; t) + n(t)$$

The three mathematical models described above adequately characterize a majority of physical channels encountered in practice.

3.2 Transmission Impairments

The transmission medium is the physical path between transmitter and receiver. The characteristics and quality of data transmission are determined both by the nature of the signal and the nature of the medium.

With any communication system, it must be recognized that the signal that is received will differ from the signal that is transmitted due to various transmission impairments. For analog signals, these impairments introduce various random modifications that degrade the signal quality. For digital signals, bit errors are introduced: a binary 1 is transformed into a binary 0 and vice versa. The most significant impairments are: Attenuation, Delay distortion and Noise. The various impairment effects that can degrade a signal during transmission are shown in Figure 3.4.

3.2.1 Attenuation

As a signal propagates along a transmission medium its amplitude decreases. This is known as signal attenuation. To compensate the attenuation, amplifiers are inserted at intervals along the cable to restore the received signal to its original level. Signal attenuation increases as a function of frequency. To overcome this problem, the amplifiers are designed to amplify different frequency by varying gains of amplifications. These devices are known as **equalizer**. For guided media (Twisted wires, Coaxial cables and Fiber optic cables) attenuation, is generally logarithmic and it is typically expressed as a constant number of decibels per unit distance.

$$N, dB = 10\log\frac{P_2}{P_1}, \quad \text{Where N – number of decibels}$$

P_1, P_2 – input and output powers.

Example 3.1. A signal with power 10 mW is inserted into a transmission line. The power measured some distance is 5 mW. Find the loss.

$$Loss = 10\log_{10}\frac{5}{10} = 10(-0.3) = -3dB$$

Taking into account that power is proportional to the square of voltage:

$P_1 = U_1^2/R$; $P2 = U_2^2/R$ and

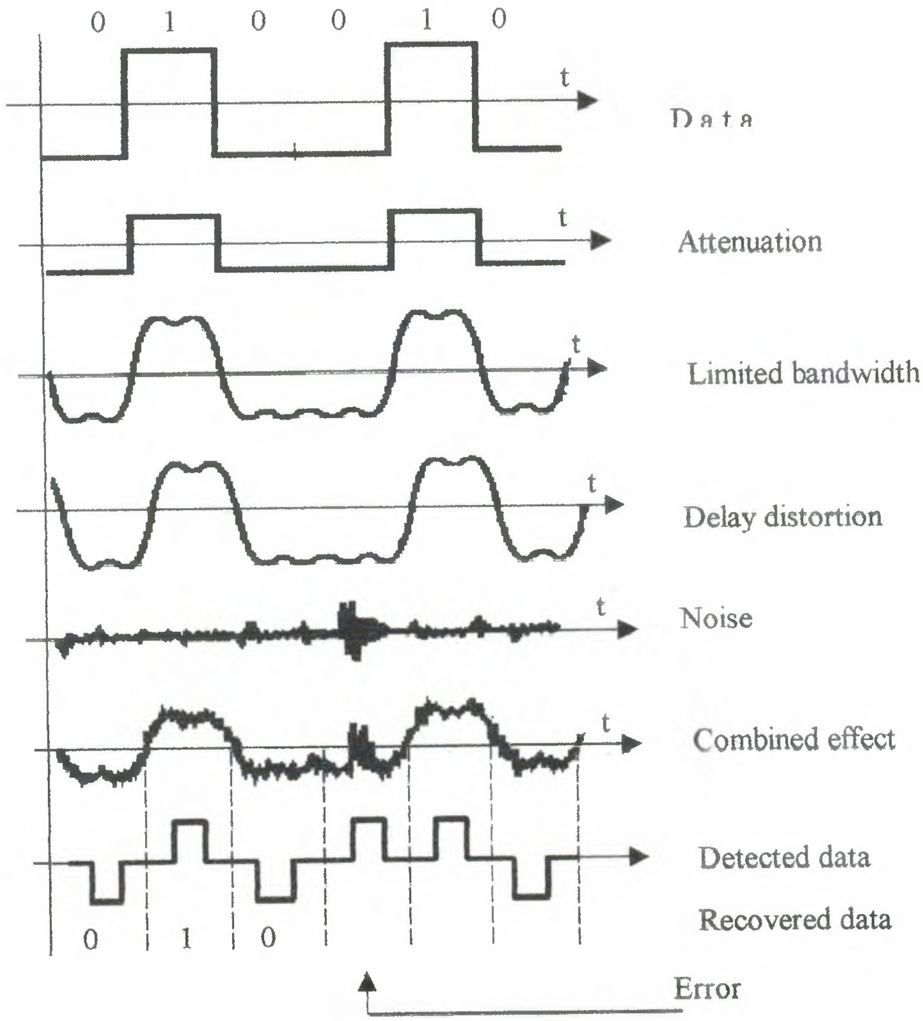


Figure 3.4 Various Impairment Effects

$$N, dB = 20\log\frac{U_2}{U_1}$$

For unguided media attenuation is a more complex function of distance and the make-up of the atmosphere. An example is shown in Figure 3.5, which shows attenuation as a function of frequency for a typical wire line. In Figure 3.5, attenuation is measured relative to the attenuation at 1000 Hz. Positive values on the y-axis represent attenuation greater than

That at 1000 Hz. For any other frequency f, the relative attenuation in decibels is $N_f = 10 \log_{10} P_f / P_{1000}$ .The solid line in Figure shows attenuation without equalization. The dashed line shows the effects of equalization.

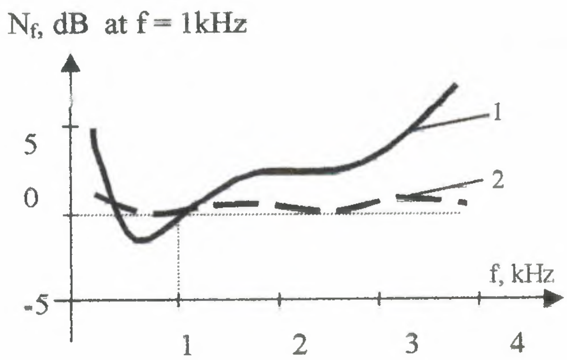


Figure 3.5. Attenuation without Equalization.

3.2.2 Delay Distortion

Delay distortion is a phenomenon peculiar to guided transmission media. The distortion is caused by the fact that the velocity of propagation of a signal through a guided medium varies with frequency. This effect is referred to as delay distortion, since the received signal is distorted due to variable delay in its components. Delay distortion is particularly critical for digital data. Consider that a sequence of bits is being transmitted, using either analog or digital signals. Because of delay distortion, some of the signal components of one bit position will spill over into other bit positions, causing inter-symbol interference, which is a major limitation to maximum bit rate over a transmission control. Equalizing techniques can also be used for delay distortion.

3.2.3 Noise

For any data transmission, the received signal will consist of the transmitted signal, modified by the various distortions imposed by the transmission system, plus additional unwanted signals that are inserted somewhere between transmission and reception. These undesired signals are referred to *Noise* and can be divided into four categories: **Thermal noise**, **Inter-modulation noise**, **Cross talk** and **Impulse noise**.

The Thermal noise is due to thermal agitation of electrons in a conductor. It is present in all electronic devices and transmission media and is a function of temperature. Thermal noise is uniformly distributed across the frequency spectrum and hence is often referred to as *white noise*. Thermal noise cannot be eliminated and therefore places an upper bound on communications system performance. This noise is assumed to be independent of frequency. The thermal noise in watts present in a bandwidth of W-hertz can be expressed as

$$N = kTW$$

Or, in decibel-watts:

$$N = 10 \log k + 10 \log T + 10 \log W$$

$$N = -228.6 \text{ (dbW)} + 10 \log T + 10 \log W$$

Where No - noise power density, watts/hertz;

k - Boltzmann's constant $k = 1.3803 \times 10^{-23}$ J/^{0}K; T - temperature, degrees Kelvin When signals at different frequencies share the same transmission medium, the result may be *inter-modulation noise*. The effect of inter-modulation noise is to produce signals at a frequency, which is the sum or difference of the two original frequencies or multiples of those frequencies. For example, the mixing of signals at frequencies f_1 and f_2 might produce energy at the frequency $f_1 + f_2$. This derived signal could interfere with an intended signal at the frequency $f_1 + f_2$.
Inter-modulation noise is produced when there is some non-linearity in the transmitter, receiver, or interviewing transmission system.

Cross talk has been experienced by anyone who, while using the telephone, he/she is able to hear another conversation: it is an unwanted coupling between signal paths. It can occur by electrical coupling between nearby twisted pair or rarely coaxial cable lines carrying multiple signals. Among several types of cross-talk the most limiting impairment for data communication systems is near-end cross-talk (self-cross-talk or echo), since it is caused by the strong signal output by the transmitter output being coupled with much weaker signal at the input of the local receiver circuit. Adaptive noise canceller is used to overcome this type of impairment.

A ***Impulse noise***, has short duration and have relatively high amplitude. It is generated from a variety of causes, including external electromagnetic disturbances, such as lightning, electrical impulses associated with the switching circuits used in the telephone exchange.

Impulse noise is generally only a minor annoyance for analog data. For example, voice transmission can be corrupted by short clicks and crackles with no loss of intelligibility. However, impulse noise is the primary source of error in digital data communication. For example, impulse noise of 0.01 s duration would not destroy any voice data, but would wash out about 50 bits of data is being transmitted at 4800 bps.

3.3 Channel Capacity

The rate at which data can be transmitted over a given communication channel, under given conditions, is referred to as the channel capacity.

There are four concepts here that we are trying to relate to one another.

- Data rate: This is the rate, in bits per second (bps), at which data can be transmitted.
- Bandwidth: This is the bandwidth of the transmitted signal as constrains by the transmitter and the nature of the transmission medium, expressed by Hertz.
- Noise: The average level of noise over the communications path.
- Error rate: The rate at which errors occur, where an error is the reception of a 1 when a 0 was transmitted or the reception of a 0 when a 1 was transmitted.

Communication facilities are expensive and, in general, the greater the bandwidth of a facility the greater the cost. Furthermore, all transmission channels of any practical interest are of limited bandwidth. The limitations arise from the physical properties of the transmission medium or from deliberate limitations at the transmitter on the bandwidth to prevent interference from other sources. Accordingly, we would like to make as efficient use as possible of a given bandwidth.

Let us consider the case of a channel that is noise-free. In this environment, the limitation on data rate is simply the bandwidth of the signal. A formulation of this limitation, due to

Nyquist, states that if the rate of signal transmission is 2W, then a signal with frequencies no greater than W is sufficient to carry the data rate. The conserve is also true: Given a bandwidth of W, the highest signal rate that can be carried is 2W.

Example 3.2. A voice channel bandwidth is of W = 3100 Hz. Find the channel capacity.
Solution: C = 2 W = 6200 bps.

However, as we shall see in chapter 3, signals with more than two levels can be used; that is each signal element can represent more than one bit. For example; if M possible voltage levels are used, then each signal element can be represented by $n = \log_2 M$ numbers of bits. With multilevel signaling, the Nyquist formulation becomes

$$C = 2\ W \log_2 M$$

Thus, for M = 8, a value used with some modems, C becomes 18600 bps.

An important parameter associated with a channel is a signal-to-noise ratio (SNR) expressed as $\quad SNR = 10\log_{10} (S/N)\ dB$

Where S/N – signal –to- noise powers ratio. Clearly a high S/N will mean a high - quality signal and a low number of required intermediate repeaters.

The signal - to noise ratio is important in the transmission of digital data because it sets the upper bound on the achievable data rate. The maximum channel capacity, in bits per second, obeys the equation attributed as the Shannon – Hartley law

$$C = W \log_2 (1+ S/N) \approx 3{,}32\ W\log_{10} (1+S/N),$$

Example 3.3. Consider a voice channel with bandwidth of 3000 Hz. A typical value of S/N for a telephone line is 20 dB.

$$SNR = 10 \log_{10}(S/N),\ S/N = 10^2 = 100;\ C = 3000 \log_2 (1 + 100) = 19963\ bps$$

3.4 Guided Media

The guided media includes: twisted pair, coaxial cable and fiber-optic cable (see Figure 3.6).

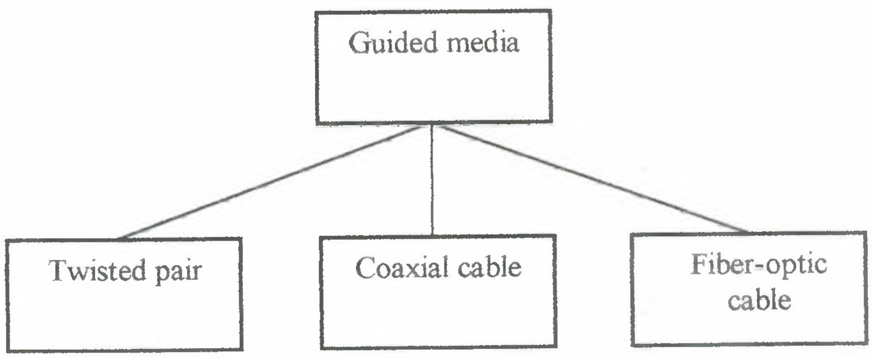


Figure 3.6 Categories of Guided Media

Table 3.1 contains the typical characteristics for guided media

Table 3.1 Typical characteristics for guided media

Medium Transmission	Total Data Rate	Bandwidth	Repeater Spacing
Twisted pair	1-100 Mbps	100Hz -5 MHz	2 - 10 km
Coaxial cable	1Mbps-1 Gbps	100 Hz – 500 MHz	1 - 10 km
Optical fiber	2 Gbps	2 GHz	10- 10 0 km

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In the past two parallel flat wires were used for communications. Each wire is insulated from the other and both are open to free space. This type of line is used for connecting equipment that is up to 50 m apart using moderate rate (less than 20 kbps). The signal, typically a voltage or current level relative to some ground reference is applied to one wire while the ground reference is applied to the other. Although a two wire open line can be used to connect two computers directly, it is used mainly for connecting computers with modems. As shown in Figure 3.7 two simple wires more sensitive to noise interference.

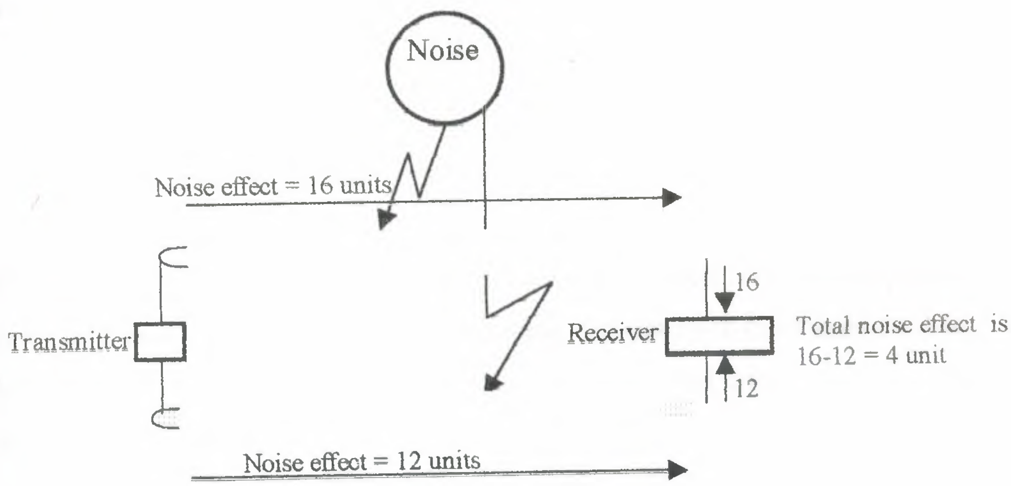

Figure 3.7 Effect of noise in parallel lines.

3.4.1 Twisted Pair

A twisted pair consists of two insulated copper wires. Over longer distances, cables may contain hundreds of pairs. The twisting of the individual pairs minimizes electromagnetic interference between the pairs (see Figure 3.8).

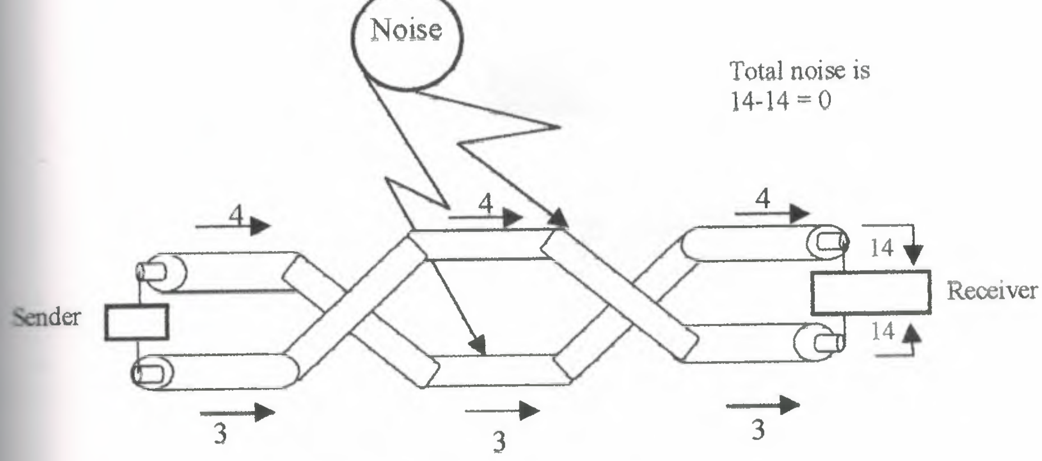

Figure 3.8 Effect of noise on twisted-pair lines

Wire pairs can be used to transmit both analog and digital signals. For analog signals, amplifiers are required about every 5 to 6 km. For digital signals, repeaters are used at every 2 or 3 km. It is the backbone of the telephone system as well as the low - cost microcomputer local network within a building. In the telephone system, individual telephone sets are connected to the local telephone exchange or "end office" by twisted - pair wire. These are referred to as "local loops". Within an office building, telephone service is often provided by means of a Private Branch Exchange (PBX). For modern digital PBX systems, data rate is about 64 kbps. Local loop connections typically require a modem, with a maximum data rate of 9600 bps. However, twisted pair is used for long - distance trucking applications and data rates of 100 Mbps or more may be achieved.

The twisted pair comes in two forms: shielded (STP) and unshielded (UTP). Figure 3.9 shows STP (a) and UTP (b, c). The metal casing prevents the penetration of electromagnetic noise and eliminates cross-talk. Materials and manufacturing requirements make STP more expensive than UTP but less susceptible to noise. UTP is cheap, flexible, and easy to use.

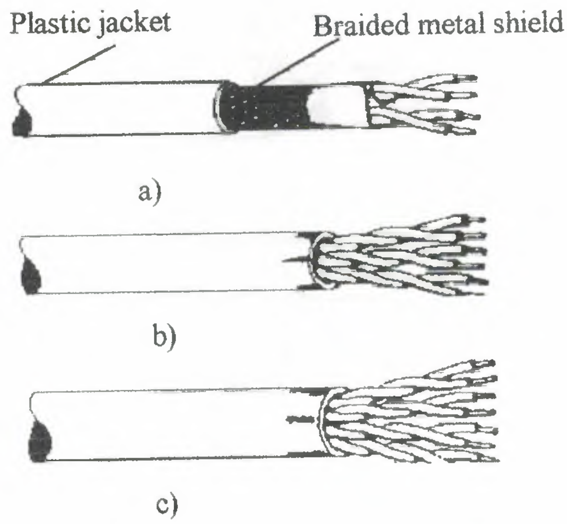

Figure 3.9 shows STP (a) and UTP (b, c).

3.4.2 Coaxial Cable

The main limiting factors of a twisted pair line are its capacity and a phenomenon known as the skin effect. As the bit rate increases, the current flowing in the wires tends to flow only on the outer surface of the wire, thus using the less available cross-section. This increases the electrical resistance of the wires for higher frequency signals, leading to the attenuation In addition, at higher frequencies, more signal power is lost as a result of radiation effect.

Coaxial cables, like twisted pairs, consist of two conductors, but are constructed differently to permit it to operate over a wider range of frequencies. Coaxial cables have been perhaps the most versatile transmission medium and enjoying increasing utilizing in a wide variety of applications. The most important of these are long-distance telephone and television transmission, television distribution, and short-range connections between devices and local area networks. In Figure 3.10 are shown the constructions of the coaxial cables. Using frequency-division multiplexing a coaxial cable can carry over 10,000 voice channels simultaneously. Coaxial cables are used to transmit both analog and digital signals.

The principal constraints on performance are attention, thermal noise, and intermodulation noise.

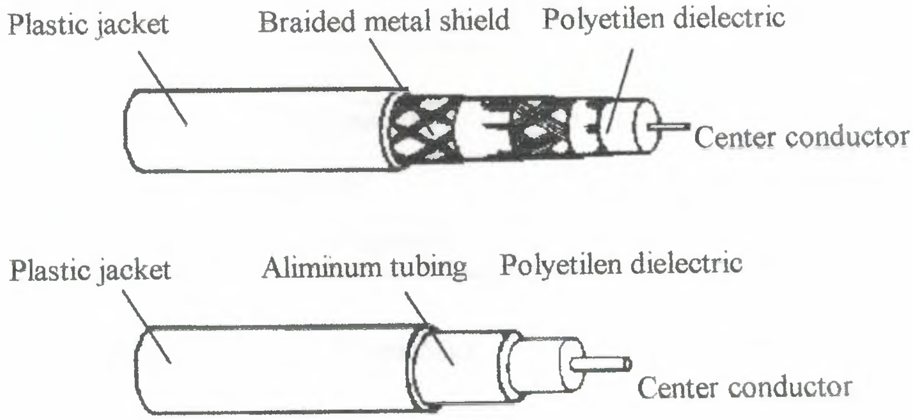

Figure 3.10 Coaxial Cable

3.5 Unguided Media

There are three basic modes of getting a radio wave from the transmitting to receiving antenna: ground wave, space wave, sky wave proportions (Figure 3.11)

The subdivision of the electromagnetic frequency range is given in the Table 3.2

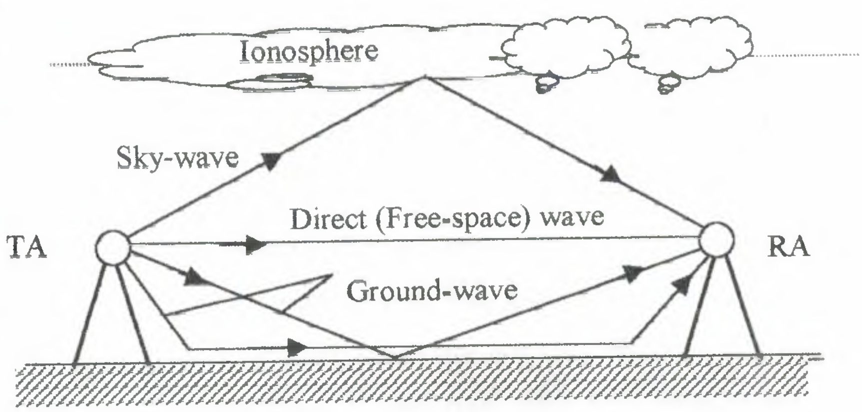

Figure 3.11 Sky Wave Proportion

Table 3.2 Frequency Range For Wireless Communication

Frequency Band	Name	Data rate	Principal applications
30 – 300 kHz	LF (Low Frequency)	0.1 – 100 bps	Navigation, Submarine
300 kHz - 3000 kHz	MF (Medium Frequency)	10 – 1000 bps	AM radio
3 - 30 MHz	HF (High Frequency)	10 – 3000 bps	Shortwave radio, CB radio
30 – 300 MHz	VHF (Very High Frequency)	To 100 kbps	VHF Television, FM radio
300 – 3000 MHz	UHF(Upper High Frequency)	To 10 Mbps	UHF Television Mobile communication Terrestrial Microwave
3 - 30 GHz	SHF (Super High Frequency)	To 100 Mbps	Satellite and Terrestrial microwaves, Radar

The frequency of the radio wave is of primary importance in considering the performance of each type of propagation.

Ground – Wave Propagation

A ground wave is a radio wave that travels along the earth's surface. It is sometimes referred to as a *surface wave*. Attenuation of ground waves is directly related to the surface impedance of the earth. This impedance is a function of conductivity and frequency. If the earth's surface is highly conductive, the absorption of wave energy, and thus its attenuation, will be reduced. Ground–wave propagation is much better over water (especially salt water) than say a very dry (poor conductivity) desert terrain. The ground losses increase rapidly with increasing frequency. For these reasons ground waves are not very effective at frequencies above 2 MHz.

Ground- wave propagation is the only way to communicate into the ocean with submarines (about 100 miles distance). To minimise the attenuation of seawater, extremely low frequency (ELF) propagation is utilised. A typically used frequency is 100 Hz, the attenuation is about 0.3 dB/m.

Space – Wave (line-of-site propagation) Propagation

The two types of space waves are shown in Figure 3.13. They are the direct wave and ground reflected wave. Do not confuse these with the ground wave just discussed. The direct wave is by far the most widely used mode of antenna communications. The propagated wave is directed from transmitting to receiving antenna and does not travel along the ground. The earth's surface, therefore, does not attenuate it. The direct space wave has one severe limitation – it is basically limited to so called line–of –sight transmission distances. Thus, the antenna height and the curvature of the earth are the limiting factors. The actual radio horizon is about 1/3 times greater then the geometric line of sight due to diffraction effects and is empirically predicted by the following approximation:

$$d = \sqrt{2h_T} + \sqrt{2h_R}$$

Where d - radio horizon (mi); h_T- transmitting antenna height (ft);
h_R - receiving antenna height (ft)

Ghosting in TV reception. Any tall or massive objects obstruct space waves. This results in diffraction (and subsequent shadow zones) and reflections. Reflections pose a specific problem since, for example, reception of a TV signal may be the combined result of a direct space wave and a reflected space waves. This condition results in ghosting, which manifests itself in the form of a double – image distortion. This is due to the two signals arriving at the receiver at two different times. A possible solution to the ghosting problem is to retune the receiving antenna orientation so that the reflected wave is too weak to be displayed.

Sky Wave Propagation

The sky wave has the ability to strike the ionosphere. It can be refracted from it to the ground, strike the ground, be reflected back toward the ionosphere, and so on. A frequency occurring problem is *signal multipath*. The multipath occurs when the transmitted signal arrives at the receiver via multipath paths at different delays. Signal multipath results intersymbol interference in a digital communication system The signal components arriving via different propagation paths may add destructively, resulting in a phenomenon called *signal fading*. Sky wave propagation ceases to exist at frequencies above 30 MHz. However it is possible to have atmospheric scatter propagation at the range of 30 MHz and troposphere scattering at 40 MHz to 300MHz.

Microwaves

Two general ranges of frequencies are of interest in discussion.

1. Microwave frequencies that cover a range of about 3 to 30 GHz. At these frequencies, highly directional beams are possible, and microwave is quite suitable for point-to-point transmission.
2. Radio waves that cover a range of about 30 MHz to 1 GHz. At these frequencies, omni directional transmission is possible, and microwave is quite suitable for broadcasting.

We will refer to signals in the range 30 MHz to 1 GHz as radio waves.

Omni directional transmission is used and signals at these frequencies are suitable for broadcast applications. The most common type of microwave antenna is the parabolic "dish". A typical size is about 10 ft in diameter. The antenna is fixed rigidly and focuses a narrow beam to achieve line-of-sight transmission to the receiving antenna. Microwave antennas are usually located at substantial heights above ground level in order to extend the range between antennas and to be able to transmit over intervening obstacles.

The primary use for terrestrial microwave systems is in long-haul telecommunications service, as an alternative to coaxial cable for transmitting television and voice. Like coaxial cable, microwave can support high data rates over long distances. The microwave facility requires far fewer amplifiers or repeaters than coaxial cable for the same distance, but requires line of sight transmission.

Another increasingly common use of microwave is for short point- to point links between buildings. This can be used for closed- circuit TV or as a data link between local networks. Finally, a potential use for terrestrial microwave is to provide digital data transmission in small regions (radius < 10 km). This concept has been termed as "local data distribution" and would provide an alternative to phone lines for digital networking.

The microwave transmission covers a substantial portion of the spectrum. Common frequencies used for transmission are in the range 2 to 40 GHz. The higher the frequency used the higher the potential bandwidth, and therefore, the higher the potential data rate.

As with any transmission system, a main source of loss for microwave is attenuation. For microwave (and radio frequency), the loss can be expressed as

$$L = 10\mathrm{Log}(\frac{4\pi d}{\lambda})^2 \text{ dB}; \quad \lambda[m] = \frac{3.10^8}{f[Hz]}$$

Where d is the distance and λ is the wavelength in the same units.

Thus loss varies as the square of the distance. This is in contrast to twisted pair and coaxial cable where the loss varies logarithmically with distance (linear in decibels). Thus repeaters or amplifiers may be placed farther apart for microwave systems – 10 to 100 km is typical. Attenuation is increased with rainfall. Another source of impairment for microwave is interference.

3.6 Fiber Optic Cable

The fiber-optic is defined as branch of optics that deals with the transmission of light through ultra pure glass, plastic or some other form of transparent media. One of first noted experiment that demonstrated the transmission of light through a dielectric medium has been created to John Tyndall. In 1854 John Tyndall demonstrated that light could be guided through stream of water based on the principle of total internal reflection.

In1880 Alexander Graham Bell invented the photo phone, a device that transmits voice signals over a beam of light.

Great interest in communication at optical frequencies was created in 1958 with the invention of the laser by Charles H. Townes.

In 1966 Charles K. Kao and George Hockham of Standard Telecommunications Laboratories of England performed several experiments to prove that, if glass could be made more transparent by reducing its impurities, light loss could be minimized. Their research led to a publication in which they predicted that optical fiber could be made pure enough to transmit light several kilometers. In the next two decades researchers worked intensively to reduce the attenuation to 0.16 dB/km.

In1988 the Synchronous Optical Network (SONET) was published by the American National Standards Institute (ANSI).

1995 Multimedia applications for business have become the major impetus for increased use of optical fiber within the LAN, MAN, and WAN environment.

Just as standard electric cables come in a variety of sizes, shapes, and types, fiber optic cables are available in different configurations. The simplest cable is just a single strand of fiber, whereas complex cables are made up of multiple fibers with different layers and other elements.

The portion of a fiber optic cable (core) that carries the light is made from either glass or plastic. Another name for glass is silica. Special techniques have been developed to create nearly perfect optical glass or plastic, which is transparent to light. Such materials can carry light over a long distance. Glass has superior optical characteristics over plastic. However, glass is far more expensive and more fragile than plastic. Although the plastic is less expensive and more flexible, its attenuation of light is greater. For a given intensity, light will travel a greater distance in glass than in plastic. For very long distance transmission, glass is certainly preferred. For shorter distances, plastic is much more practical.

All fibers consist of a number of substructures including (see Figure 3.16):

- A core, which carries most of the light, surrounded by
- A cladding, which bends the light and confines it to the core, surrounded by
- A substrate layer (in some fibers) of glass which does not carry light, but adds to the diameter and strength of the fiber, covered by
- A primary buffer coating, which provides the first layer of mechanical protection, covered by
- A secondary buffer coating, which protects the relatively fragile primary coating.

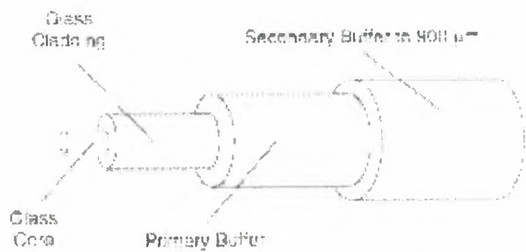

Figure 3.16 Fiber Optic Cable

The cladding is also made of glass or plastic but has a lower index of refraction. This ensures that the proper interface is achieved so that the light waves remain within the core. In addition to protecting the fiber core from nicks and scratches, the cladding adds strength. Some fiber optic cables have a glass core with a glass cladding. Others have a plastic core with a plastic cladding.

Another common arrangement is a glass core with a plastic cladding. It is called plastic-clad silica (PCS) cable.

3.6.1 Theory of Light

In the seventeenth and eighteenth centuries, there were two schools of thought regarding the nature of light. Sir Isaac Newton and his followers believed that light consisted of rapidly moving particles (or corpuscles), whereas Dutch physicist Christian Huygens regarded light as being a series of waves.

The wave theory was strongly supported by an English doctor named Thomas Young. By 1905, quantum theory, introduced by Clark Maxwell, showed that when light is emitted or absorbed it is not only as a wave, but also as an electromagnetic particle called a photon. Photon is said to possess energy that is proportional to its frequency. This is known as Planck's law, which states:

$$E = h \times v$$

where E = photon's energy (J);

h = Planck's constants, 6.63×10^{-34} (J-s);

v = frequency of the photon (Hz).

Using the particle theory, Einstein and Planck were able to explain photoelectric effect: when visible light or electromagnetic radiation of a hire frequency shines on a metallic surface, electrons are emitted, which is turning an electric current.

Electromagnetic Spectrum

Fundamentally, light has been accepted as a form of electromagnetic radiation that can be categorized into a portion of the entire electromagnetic spectrum, as shown in Table 3.2. In addition, each frequency can be specified in terms of its equivalent wavelength. Frequency or wavelength are directly related to the speed of light.

$$C = f \times \lambda$$

Where c - speed of light in a vacuum or free space, 3×10^{8} (m/s);

f - frequency (Hz); λ-wavelength (m).

Example 3.2 Compute the wavelength for a frequency of 7.20 MHz.

Solution: $c = f \times \lambda$

Therefore, $F = c/\lambda = (3 \times 10^{8}) / (7.2 \times 10^{6}) = 41.7$ m

Table 3.3 Electromagnetic Spectrum

Range of wavelength, nm	Name of wavelength	
$10^{6} - 770$	*Infrared*	
770 – 662	Red	*Visible*
662 – 597	Orange	
597 – 577	Yellow	
577 – 492	Green	
492 – 455	Blue	
455 – 390	Violet	
390 – 10	*Ultraviolet*	

The portion of the electromagnetic spectrum regarded as light has been expanded in Table 3.3 to illustrate three basic categories of light:

1. ***Infrared***: that portion of the electromagnetic spectrum having a wavelength ranging from 770 to 10^{6} nm. Fiber optic systems operate in this range.

2. ***Visible:*** that portion of the electromagnetic spectrum having a wavelength ranging from 390 to 770 nm. The human eye, responding to these wavelengths allows us to see the colours ranging from violet to red, respectively.
3. ***Ultraviolet:*** that portion of the electromagnetic spectrum ranging from 10 to 390 nm.

The light that we use for most fiber optic systems occupies a wavelength range from 800 to 1600 nm. This is slightly larger than visible red light and falls within the infrared portion of the spectrum.

Snell's Law : Total Interval Reflection

For light to propagate in any medium, the medium must be transparent to some degree. The degree of transparency determines how far light will propagate. Transparent materials can be in the form of a liquid, gas, or a solid. Some examples are glass, plastic, air, and water.

One of the most fundamental principles of light is that when it strikes the interface between two transparent mediums, such as air and water, a portion of the light energy is reflected back into the first medium and a portion is transmitted into the second medium. The path in which light travels from one point to another is commonly referred to as the ray. Figure 3.12 illustrates the classic example of a ray of light incident upon the surface of water. Notice that part of the light is *reflected* off the surface of water and part of it penetrates the water. The ray penetrating to water is said to be *refracted* or bent toward the normal. The amount of refracted light is determined by the medium's index of refraction, generally denoted by the letter n. Index of refraction is the ratio of the speed of light in a vacuum - c, to the speed of light in the given medium - v. This relationship is given by the equation: $n = c / v$. Since the speed of light is lower in mediums other than a vacuum, the index of refraction in such mediums is always greater than 1.

Example for air $n = 1.003$, for water $n = 1.33$, for fiber-optic $n = 1.6$.

In 1621, the Dutch mathematician Willebrard Snell established that rays of light could be traced as they propagate from one medium to another based on their indices of refraction. Snell's low is stated by the equation:

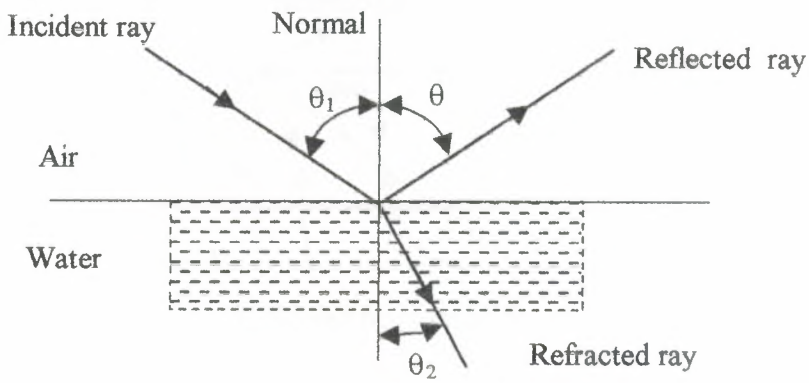

Figure 3.12 Ray Of Light Incident Upon The Surface Of Water.

$$\frac{n_1}{n_2} = \frac{\sin\theta_2}{\sin\theta_1}; \qquad n_1 \sin\theta_1 = n_2 \sin\theta_2$$

where n_1 - refractive index of material 1; θ_1 - angle of incidence; θ_2 - angle of refraction; n_2 - refractive index of material 2. When the angle of incidence, θ_1, becomes large enough to cause the sine of the refraction angle, θ_2, to exceed the value of 1, total internal reflection occurs. This angle is called the critical angle, θ_c. The critical angle, θ_c, can be derived from Snell's law as follows

$$n_1 \sin\theta_1 = n_2 \sin\theta_2$$

$$\sin\theta_1 = n_2 \sin\theta_2/n_1$$

When $\sin\theta_1 = \sin\theta_2$, then $\sin\theta_1 = n2 / n1$. Therefore, critical angle: $\theta_c = \sin^{-1} (n2 / n1)$

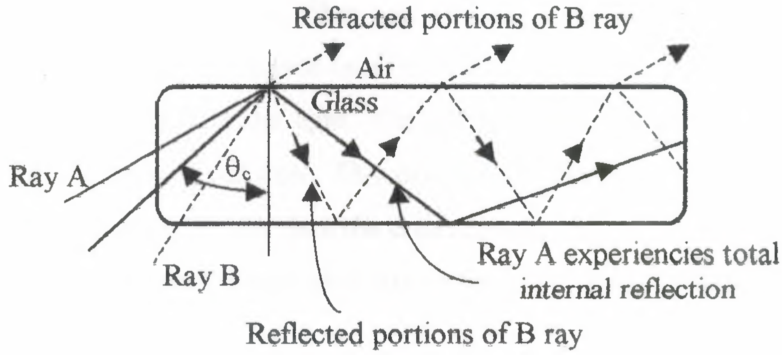

Figure 3.13 Ray A penetrates the glass-air interface at an angle exceeding the critical angle, θc.

For glass, n = 1.5; for air n = 1.0 and $\theta_o = \sin^{-1}(n2 / n1) = \sin^{-1}(1.0 / 1.5) = 41.8^0$.

By surrounding glass with material whose refraction index is less than that of the glass, total internal reflection can be achieved. This is illustrated in Figure 3.13. Ray A penetrates the glass-air interface at an angle exceeding the critical angle, θc, and therefore experiences total internal reflection. On the other hand, Ray B penetrates the glass air interface at an angle less than the critical angle. Total internal reflection does not occur. Instead, a portion of ray B escapes the glass and is refracted away from the normal as it enters the less dense medium of air. A portion is also reflected back into the glass. Ray B diminished in magnitude as it bounces back and forth between

The glass-air interface. The foregoing principle is the basis for guiding light through optical fibers.

Two key elements that permit light guiding through optical fibers are its core and its cladding. The fiber's core is manufactured of ultra pure glass (silicon dioxide) or plastic. Surrounding the core is a material called cladding. A fiber cladding is also made of glass or plastic.

Its index of refraction, however, it is typically 1% less than that of its core. This permits total internal reflection of rays entering the fiber and striking the core-cladding interface above the critical angle of approximately 82-degree ($\sin^{-1}$ (1/1.01). The core of the fiber therefore guides the light and the cladding contains the light. The cladding material is much less transparent than the glass making up the core of the fiber. This causes light rays to be absorbed if they strike the core-cladding interface at an angle less than the critical angle.

In Total internal reflection occurs as it strikes the lower index cladding material.

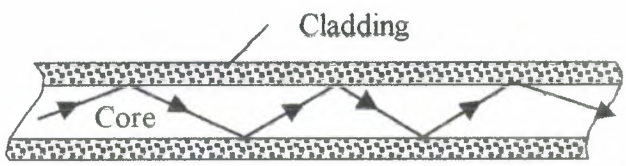

Figure 3.14 A Light Ray Is Transmitted Into The Core Of An Optical Fiber.

3.6.2 Block Diagram of the FOS

One of the main limitations of communication systems is their restricted information carrying capabilities. In more specific terms what this means is that the communications medium can only carry so many messages. And, as you have seen, this information-handling ability is directly proportional to the bandwidth of the communications channel. In telephone systems, the bandwidth is limited by the characteristics of the cable used to carry the signals. As the demand for telephones has increased, better cables and wiring systems have been developed. Further, multiplexing techniques have been used to transmit multiple telephone conversations over a single cable.

In radio communication systems, the information modulates a high frequency carrier. The modulation produces sidebands, and therefore, the signal occupies a narrow portion of the RF spectrum. However, the RF spectrum is finite. There is only so much space for radio signals. To increase the information capacity of a channel, the bandwidth of the channel must be increased. This reduces available spectrum space. Multiplexing techniques are used to send more signals in a given channel bandwidth, and methods have been developed to transmit more information in less bandwidth.

The information-carrying capacity of the radio signal can be increased tremendously if higher carrier frequencies are used. As the demand for increased communications capacity has gone up over the years, higher and higher RFs are being used. Today, microwaves are the preferred radio channels for this reason, but it is more complex and expensive to use these higher frequencies because of the special equipment required.

One way to expand communications capability further is to use light as the transmission medium. Instead of using an electrical signal traveling over a cable or electromagnetic waves traveling through space, the information is put on a light beam and transmitted through space or through a special cable. In the late nineteenth century, Alexander Graham Bell, the inventor of the telephone, demonstrated that information could be transmitted by light.

Light beam communication was made more practical with the invention of the laser. The laser is a special high-intensity, single frequency light source. It produces a very narrow beam of brilliant light of a specific wavelength (color). Because of its great intensity, the laser beam can penetrate atmospheric obstacles better than other types of light, thereby making light-beam communication more reliable over longer distances. The primary problem with such free-space light beam communication is that the transmitter and receiver must be perfectly aligned with one another.

Instead of using free space, some type of light carrying cable can also be used. For centuries it has been known that light is easily transmitted through various types of transparent media such as glass and water, but it wasn't until the early in 1900s that scientist were able to develop practical light carrying media. By the mid-1950s glass fibers were developed that permitted long light carrying cables to be constructed. Over the years, these glass fibers have been perfected. Further, low cost plastic fiber cable also developed. Developments in these cables permitted them to be made longer with less attenuation of the light.

Today the fiber optic cables have been highly refined. Cables many miles long can be constructed and interconnected for the purpose of transmitting information on a light beam over very long distances. Its great advantage is that light beams have an incredible information carrying capacity. Whereas hundreds of telephone conversations may be transmitted simultaneously at microwave frequencies, many thousands of signals can be carried on a light beam through a fiber optic cable. Using multiplexing techniques similar to those used in telephone and radio systems, fiber optic communications systems have an almost limitless capacity for information transfer.

The components of a typical fiber optic communications system are illustrated in Figure 3.15.

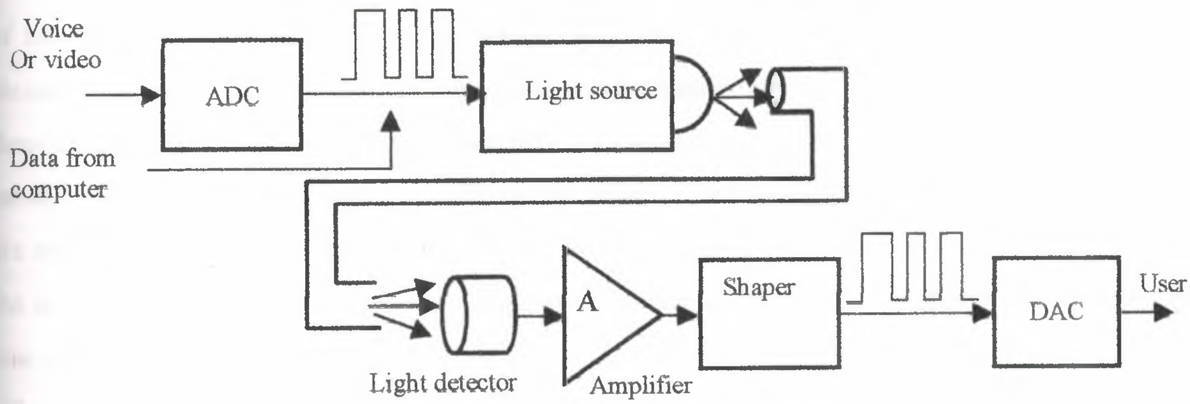

Figure 3.15 Typical Fiber Optic Communications System

The information signal to be transmitted may be voice, video, or computer data. The first step is to convert the information into a form compatible with the communications medium. This is usually done by converting continuous analog signals such as voice and video (TV) signals into a series of digital pulses. An Analog-to-Digital Converter (ADC) is used for this purpose. Computer data is already in digital form. These digital pulses are then used to flash a powerful light source off and on very rapidly. In simple low cost systems that transmit over short distances, the light source is usually a light-emitting diode (LED). This is a semiconductor device that puts out a low intensity red light beam. Other colors are also used. Infrared beams like those used in TV remote controls are also used in transmission. Another commonly used light source is the laser emitting diode. This is also a semiconductor device that generates an extremely intense single frequency light beam.

The light beam pulses are then fed into a fiber optic cable where they are transmitted over long distances. At the receiving end, a light sensitive device known as a photocell or light detector is used to detect the light pulses. This photocell or photo detector converts the light pulses into an electrical signal. The electrical pulses are amplified and reshaped back into digital form. They are fed to a decoder, such as a Digital-to-Analog Converter (DAC), where the original voice or video is recovered for user.

3.6.3 Basic Construction of the Fiber-Optic Cables

There are two basic ways of classifying fiber optic cables. The first way is an indication of how the index of refraction varies across the cross section of the cable. The second way of classification is by mode. Mode refers to the various paths that the light rays can take in passing through the fiber. Usually these two methods of classification are combined to define the types of cable. There are two basic ways of defining the index of refraction variation across a cable. These are step index and graded index. Step index refers to the fact that there is a sharply defined step in the index of refraction where the fiber core and the cladding interface. It means that the core has one constant index of refraction N1, while the cladding has another constant index of refraction N2.

The other type of cable has a graded index. In this type of cable, the index of refraction of the core is not constant. Instead, the index of refraction varies smoothly and continuously over the diameter of the core. As you get closer to the center of the core, the index of refraction gradually increases, reaching a peak at the center and then declining as the other outer edge of the core is reached. The index of refraction of the cladding is constant.

Mode refers to the number of paths for the light rays in the cable. There are two classifications: single mode and multimode. In single mode, light follows a single path through the core. In multimode, the light takes many paths through the core.

Each type of fiber optic cable is classified by one of these methods of rating the index or mode. In practice, there are three commonly used types of fiber optic cable: ***multimode step index***, ***single mode step index*** and ***multimode graded index*** cables.

3.6.4 Advantages and Disadvantages of the FOS

a) Advantages

The major advantages are:

Bandwidth One of the most significant advantages that fiber has over copper or other transmission media is a bandwidth. Bandwidth is directly related to the amount of information that can be transmitted per unit time. Today's advanced fiber optic systems are capable of transmitting several gigabits per second over hundreds of kilometers. Ten thousands of voice channels can now be multiplexed together and sent over a single fiber strand.

Less Lose. Currently, fiber is being manufactured to exhibit less than a few tenths of a decibel of loss per kilometer.

Less Weight and Volume. Fiber optic cables are substantially lighter in weight and occupy much less volume than copper cables with the same information capacity. For example, a 3-in. diameter telephone cable consisting of 900 twisted-pair wires can be replaced with a single fiber strand 0.005 inch in diameter (approximately the diameter of a hair strand) and retain the same information-carrying capacity. Even with a rugged protective jacket surrounding the fiber, it occupies enormously less space and weights considerably less.

Security. Since light does not radiate from a fiber optic cable, it is nearly impossible to secretly tap into it without detection. For this reason, several applications requiring communications security employ fiber-optic systems. Military information, for example, can be transmitted over fiber to prevent eavesdropping. In addition, metal detectors cannot detect fiber-optic cables unless they are manufactured with steel reinforcement for strength.

Flexibility. The surface of glass fiber is much more refined than ordinary glass. This, coupled with its small diameter, allows it to be flexible enough to wrap around a pencil. In terms of strength, a 0.005-in. strand of fiber is strong enough to cut one's finger before it breaks, if enough pressure is applied against it.

Economics. Presently, the cost of fiber is comparable to copper at approximately \$0.20 to \$0.50 per yard and is expected to drop as it becomes more widely used. Since transmission losses are considerably less than for coaxial cable, expensive repeaters can be spaced farther apart. Fewer repeaters mean a reduction in overall system cost and enhanced reliability.

Reliability. Once installed, a longer life span is expected with fiber over its metallic counterparts since it is more resistant to corrosion caused by environmental extremes such as temperature, corrosive gases, and liquids.

b) Disadvantages

In spite of the numerous advantages that fiber optic systems have over conventional methods of transmission, there are some disadvantages, particularly because of its newness. Many of these disadvantages are being overcome with new and competitive technology.

Interfacing costs. Electronic facilities must be converted to optics in order to interface to fiber. Often these costs are initially overlooked. Fiber-optic transmitter, receiver, couplers, and connectors, for example, must be employed as part of the communication system. Test and repair equipment is costly. If the fiber optic cable breaks, splicing can be a costly and tedious task

Strength. Fiber, by itself, has a tensile strength of approximately 1 lb, as compared the coaxial cable at 180 1b (RG59U) surrounding the fiber with stranded Kevlar and a protective PCV jacket can increase the pulling strength up to 500 1b. Installations requiring greater tensile strengths can be achieved with steel reinforcement.

Remote Powering of Devices. Occasionally it is necessary to provide electrical power to a remote device. Since this cannot be achieved through the fiber, metallic conductors are often included in the cable assembly. Several manufacturers now offer a complete line of cable types, including cables manufactured with both copper wire and fiber.

4. ERROR DETECTION AND CORRECTION

4.1 Overview

Networks must be able to transfer data from one device to another with complete accuracy. A system that cannot guarantee that the data received by one device are identical to the data transmitted by another device is essentially useless. Yet anytime data are transmitted from source to destination, they can become corrupted ia passage. In fact, it is more likely that some part of a message will be altered in transit than that the entire contents will arrive intact. Many factors, including line noise, can alter or wipe out one or more bits of a given data unit. Reliable systems must have a mechanism for detecting and correcting such errors.

Data can be corrupted during transmission. For reliable communication, errors must be detected and corrected.

Error detection and correction are implemented either at data link layer or the transport layer of the OSI model.

4.2 Types of Errors

Whenever an electromagnetic signal flows from one point to another, it is subject to unpredictable interference from heat, magnetism, and other forms of electricity. This interference can change the shape or timing of the signal. If the signal is carrying encoded binary data, such changes can alter the meaning of the data, changing 0 to 1 or 1 to 0. bits can be changed singly or in clumps. For example, a 0.01 second burst of impulse noise on a transmission with a data rate of 1200 bps might change 12 bits of information. Other circumstances can alter just one bit of a data unit, or the first and third bits but not the second. These errors, though seemingly less significant, can make the data just as unreadable as wiping out 12 bits. So it is important to understand all three types of errors and how to detect