CHAPTER

1 Basic Communication Systems

1.1 INTRODUCTION

It is a natural desire for mankind to communicate with each other. Scientists and engineers working in this field are continuously developing new resources and techniques for better communication. The era of electrical communication began with the successful demonstration of sending a message through a telegraph line running between Baltimoe, MD and Washington, D.C. by Morse in 1844. The invention of vacuum tube triode amplifier by Forest in 1906 revolutionized completely the field of electrical communication and opened the door for wireless communication.

Many remarkable inventions were made in the field of communications during the period 1906–1947. We can say that this was the first phase in the development of electronics and communication technologies. The said era witnessed the development of wireless telegraphy and telephony radio and television broadcasting, teletypewriter, radar and first electronic digital computer, electronic numerical integrator and computer (ENIAC). The discovery of transistor by Brattain, Bardeen and Shockley in 1947, at Bell Laboratories started the second phase of advancement in electronics and communication technologies and communication systems. Some most remarkable communication technologies and related systems that were developed during this phase are:

- Microwave communication links
- Color television
- Satellite communication
- Stereo FM broadcast
- Digital communication
- Cable television system
- Optical fiber communication

The breakthrough of the single chip microprocessor, Intel 4004 by Intel Corporation in 1971 started the golden era of computers. Computers and very large-scale integration (VLSI) technology, which is used in fabricate integrated circuits (ICs), introduced diverse modes of communication systems, such as:

- Fax
- Optical fiber communication
- Cellular mobile communication
- Personal mobile communication
- Computer networks
- Cordless phones

- Paging
- The internet
- Advanced Optical Communication Systems and Networks: Wavelength Division Multiplexing Technology
- 5G and 6G wireless technologies

The present era can be termed as the era of modern communication systems and technologies. "From the smallest personal items to the largest continents, everything, everywhere will be digitally connected and responsive to our wants and likes" is a typical vision for the future of communication networks as stated in "The World in 2025: 10 Predictions of Innovation" (Thomson Reuters). In particular, wireless communications are expected to dominate everything, everywhere, and transform everyday life, mainly by means of revolutionary fifth generation (5G) and sixth generation (6G) technologies characterized by concepts such as:

Cell-less architectures, massive spatial processing, tactile response times, big data processing and virtualization, to name a few. The first 5G networks have just been rolled out in a handful of cities, and they promise massive improvements over 4G. Where 4G speeds have plateaued around 45 megabits per second even the first 5G networks are capable of up to 600 megabits per second. They are even faster than your home broadband connection. Toward this end, research, performance evaluation, experimentation, and standardization activities have started. Researchers and engineers have recently focussed their attention on setting the requirements, exploring the system concept characterstics, and investigating the enabling technologies for 5G.

Nowadays, scientists and researchers are working about 6G. Although no standards have been set so far, researchers at the Jacobs university, Bermen in Germany forsee 6G networks with downloads of around 1 terabit per second—1667 times faster than 5G, and 22, 223 times faster than 4G.

It is a mind boggling speed; what will we use it for? Researchers are of the view that high data transfer speeds will be needed to make the most of futures Artificial Intelligence (AI) systems. The faster speeds of sixth-generation networks will be necessary for several other fields.

In this chapter, we study the basics of communication systems, which is essential for understanding of communication technologies.

1.2 MODERN COMMUNICATION SYSTEM

Basically, communication is the science and practice of transmitting information. Communication engineering and technologies also deals with the techniques of transmitting information. In a specific sense, communication engineering means electrical communication, in which information is transmitted through electrical signals. In this process, the information or message, such as spoken words, photographs, live scenes, and sound is first converted into electrical signals and then transmitted through electrical links. We can say that electrical communication is a process by which the information/ message is transmitted from one point to another, from one person to another, or from place to another in the form of electrical signals, through some communication link.

We may note that a communication system provides a link between the information source and its destination. The process of electrical communication involves sending, receiving, and processing information in electrical form. A basic communication system consists of certain units, called constituents, subsystems, or stages. Further, the information to be transmitted passes through a number of stages of the communication system before it reaches the destination.

A schematic block diagram of the most general form of basic communication system is shown in Fig. 1.1.

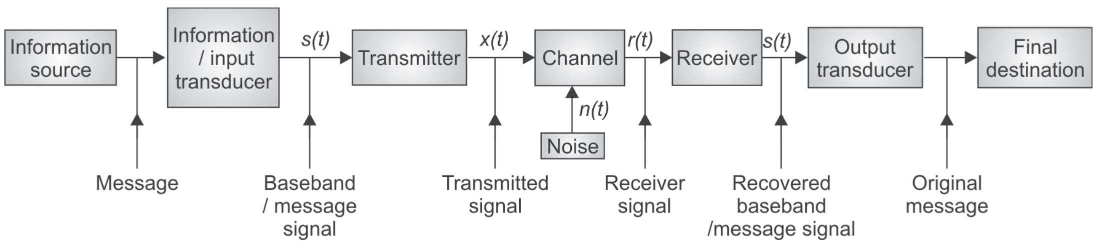

Fig. 1.1: Schematic block diagram of a basic communication system

As shown in Fig. 1.1, the main constituents of a basic communication system are:

- Information source and input transducer
- Transmitter
- Channel or medium
- Noise
- Receiver
- Output transducer and final destination.

There are many types of communication systems, such as analog, digital, radio, and line communication systems. Each type of communication system comprises the constituents is shown in Fig. 1.1. We may note that different communication systems apply different principles of operation and physical appearance to each constituent, in accordance with its type.

We now briefly describe each of the constituents or subsystems, explain the correlation between the subsystems, and provide an overview of the working of a basic communication system.

1.2.1 Information Source of Input Transducer

Information source is the first stage of a communication system because a communication system transmits information from an information source to a destination. In fact, information is not a physical quantity and the physical form of information is represented by a message that is originated by an information source, e.g. a sentence spoken by a person a message that contains some *information*. The person, in this case, is the information source. Other familiar examples are music, voice, live scenes, written text, and e-mail.

A communication system transmits information in the form of electrical signal or signals. If the information produced by the source is not in an electric form, one must use a device, known as a *transducer*, to convert the information into electrical form.

A transducer is a device that converts a non-electrical energy into its corresponding electrical energy, called signal and vice versa, e.g. during a telephone conversation, the words spoken by a person are in the form of sound energy. This has to converted to its equivalent electrical form before it is transmitted. A microphone is an example of a transducer.

The information produced by the Information source is applied to the next stage, termed as the *information* or *input transducer*. This in turn, produces an electrical signal corresponding to the information as output. This electrical signal is called the *baseband* signal. It is also called a *message signal*, an information signal, or an *envelope*. The baseband signal is usually designated as $s(t)$.

1.2.2 Types of Signals

There are two types of signals: *Analog* and *digital*.

(i) Analog Signal

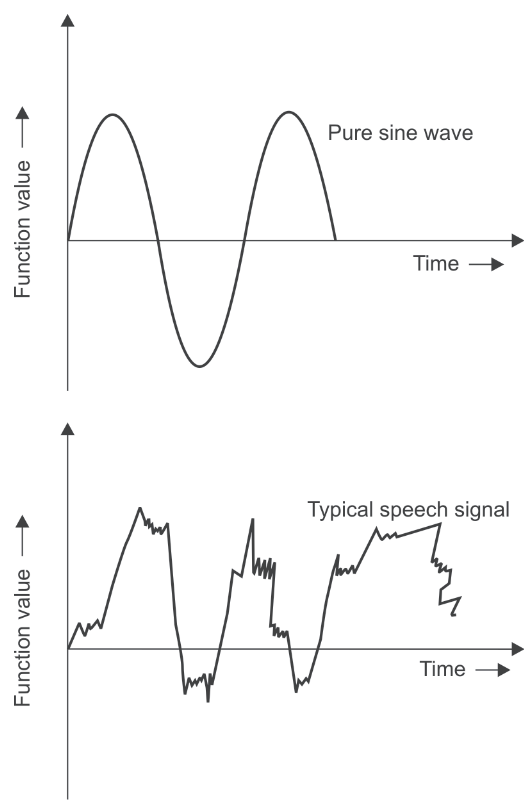

Fig. 1.2: Analog signals

An analog signal is a function of time, and has a continuous range of values. Further, there is a definite function value of the analog signal at each point of time.

An example of analog signal or analog waveform is pure sine wave. A voice signal is a practical example of an analog signal. When a voice signal is converted to electrical form by a microphone, one obtains a corresponding electrical analog singal. One can see this electrical signal on an oscilloscope. Bring an analog signal, the waveform has definite values at all points of times. Analog signals are shown in Fig. 1.2.

(ii) Digital Signal

A digital signal does not have continuous function values on a time scale. A digital singal is discrete in nature, i.e. it has some values at discrete timings. In between two consecutive values, the signal value is either zero or different value, e.g. sound signal produced by drumbeats. Figure 1.3a shows a graphically represented digital signal.

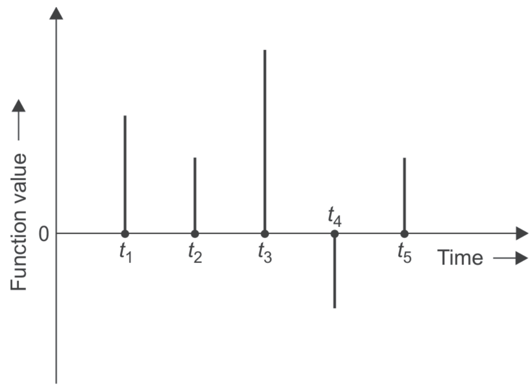

Fig. 1.3a: Digital signal

A digital signal does not have continuous function values on a time scale. It is discrete in nature, that is, it has some values at discrete timings. In between two consecutive values, the signal value is either zero or a different value. The sound signal produced by drumbeats is an example of a digital signal. This graphic actually does not represent a digital signal. Digital signals in their true sense correspond to binary digital signal, where the discrete amplitude of the signal is coded into binary digits represented by '0' and '1'. The analog signal which is continuous in time, is converted to discrete time, using the procedure called *sampling*. The continuous amplitude of the analog signal is converted to discrete amplitude using a process called *quantization*. Sampling and quantization are together termed *analog-to-digital conversion* (ADC) and the circuitry that performs this operation is called an analog-to-digital converter.

A digital signal, current or voltage have discrete values. A digital voltage has only two voltage levels, i.e. either high (some constant value of voltage) or low (zero). As stated above, the two levels in a digital signal are coded usually in a binary code. When voltage is low, it is represented by binary digit zero (=0) and when it is high it is represented by 1. Thus, the digital signals are in the form of pulses of equal levels. Figure 1.3b shows one such pulsating signal.

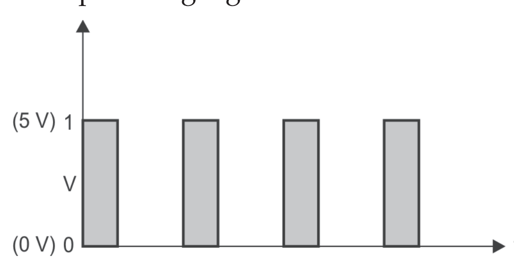

Fig. 1.3b: Digital signal

The digital signals can be used as effective commands. Digital signal zero refers to an open circuit, whereas 1 refers to closed circuit. Digits 0 and 1 are called bits. A group of bits is called *binary word* or a *byte.* A byte made of 3 bits can give 8 combinations (000, 001, 010, 100, 011, 101, 110 and 111). In general, if x is the number of bits in a byte, then the number of code communications will be

$$N = 2^x \tag{1.1}$$

In a commonly used 8-bit byte, the number of code combination is

$$N = 2^8 = 256$$

The advantage offered by digital transmission and processing of signals are numerous. Some of transmission and digital signal processing over analog transmission and analog signal processing are:

- The signal fidelity is controlled well in digital transmission.
- In the case of analog signals, the noise added in the channels gets amplified at each regeneration/amplifier along with the actual signal. In the case of digital transmission, regenerative noise (the noise which is amplified at the regenerator/amplifier is highly reduced.
- The transmitted analog signals usually contain redundant information, which is repetitive or similar to the main signal. Using digital transmission and efficient source coding techniques unique to digital transmission, such redundancies are removed providing better bandwidth utilization.
- The digital circuitry is relatively cheaper than the bulky analog transmission system. This is due to the tremendous growth achieved in the field of digital integrated chip design, very large-scale integration (VLSI), and the availability of advanced digital signal processing algorithms and processors.
- Analog circuitry is subject to component tolerances, which are inherent in the analog circuitry. The same component may respond slightly differently in different environmental conditions. With digital circuitry, such tolerances are removed and identical outputs are delivered irrespective of the external environmental conditions.
- One can easily store and recover digital signals from magnetic media compared to the analog signals.

1.3 TRANSMITTER

Transmitter is the next block in the communication system as shown in Fig. 1.1.

The basic components of a transmitter are:

(i) Message (ii) Signal generator, and (iii) Antenna

There are two basic modes of communication:

(i) *Point-to-point communication.* In this mode, communication takes place over a link between a signal transmitter and receiver, e.g. telephony.

(ii) *Broadcast mode.* In this mode, a large number of receivers are linked to a signal transmitter, e.g. radio and television.

The baseband signal, which is the output of an input transducer, is input to the transmitter. This baseband signal, $s(t)$, is suitable for transmission in the form in which it is generated by the transducer. The transmitter section processes the signal before transmission. However, the nature of processing depends on the type of communication system. The processing carried out for signal transmission in the analog form is different from signal transmission in the digital form.

There are following two options for processing signals before transmission:

(i) *The baseband signal*, which lies in the low frequency spectrum, is translated to a higher frequency spectrum.

(ii) *The baseband signal* is transmitted without translating it to a higher frequency spectrum.

In the former case, the *communication* system is known as a *carrier communication system*. In this system, the baseband signal is carried by a higher frequency signal, called the *carrier signal*. In the latter case, the system is known as a *baseband communication system*, because the baseband signal is transmitted without translating it to a higher frequency spectrum. However, some processing if the signal is required before its transmission, e.g. a train of pulses that are to be transmitted can be replaced by a series of two sine waves of different frequencies before transmission. One of these two frequencies represents a low and the other represents a high value of the digital pulse. Obviously, the baseband signal is converted into a corresponding series of sine waves of two different frequencies before transmission. Figure 1.4 illustrates this process.

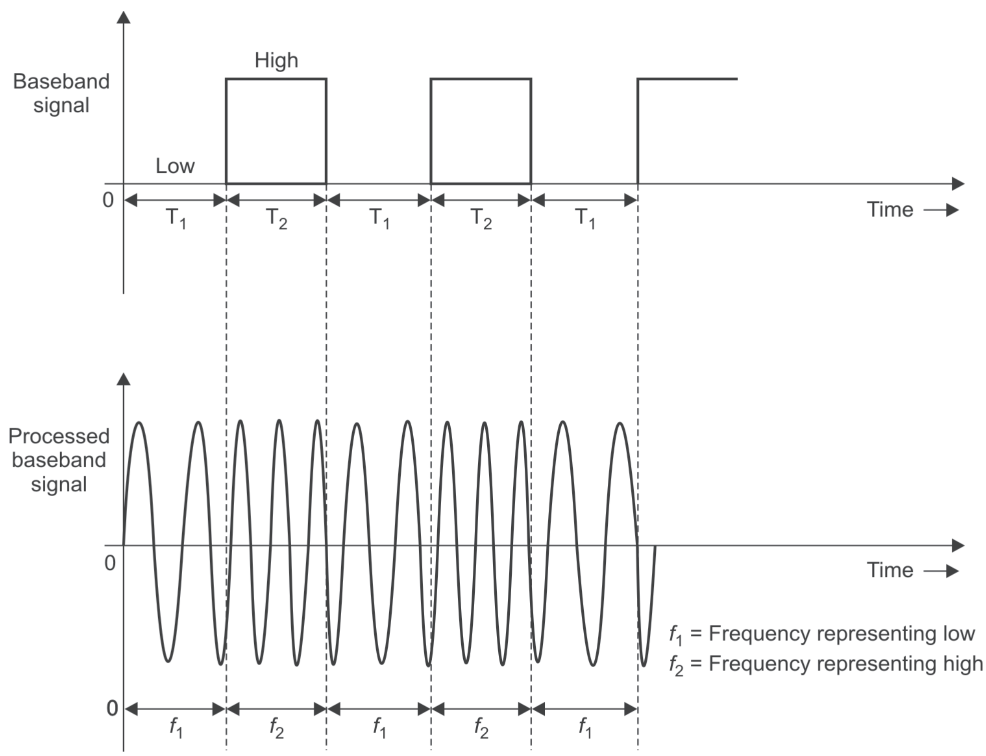

Fig. 1.4: Processing a baseband signal

The carrier communication system is based on the principle of translating a low frequency baseband signal to a higher frequency spectrum. This process is known as *modulation*.

If the baseband signal is a digital signal, the carrier communication system is known as a *digital communication system*. In this case, digital modulation technologies and methods are employed. If the baseband signal is an *analog signal*, the carrier communication system is called an *analog communication* system, and the analog modulation techniques are employed for processing. Figure 1.5a shows the block diagram of a typical transmitter.

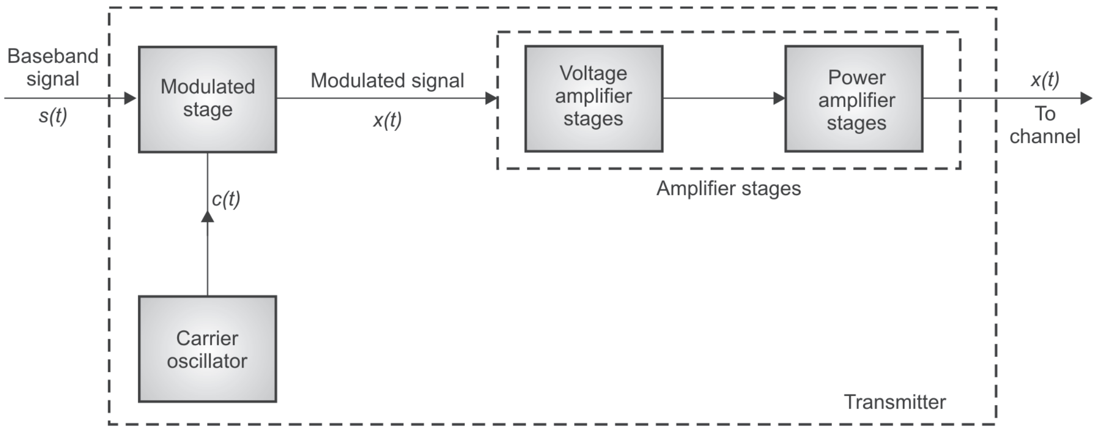

Fig. 1.5a: Schematic block representation of an analog transmitter section

Figure 1.5a shows that the baseband signal $s(t)$ is applied to the modulated stage. This stage translates the baseband signal from its low frequency to a high frequency spectrum. This stage also receives another input called the *carrier signal*, $c(t)$, which is generated by a high frequency carrier oscillator. Modulation takes place at this stage with the baseband and the carrier signals as two inputs. After modulation, the baseband signal is translated to a high frequency spectrum and the carrier signal is said to be modulation by the baseband signal. The output of the modulated stage is called the *modulated signal* and is designated as $x(t)$.

The vottage of the modulated signal is then amplified to drive the last stage of the transmitter, called the power amplier stage (Fig. 1.5). This stage amplifies the power of the modulated signal so that is carries enough power to reach the receiver stage of the communication system. Finally, the signal is passed to the transmission medium or channel. Figure 1.5b shows the block diagram of an electronic transmission system.

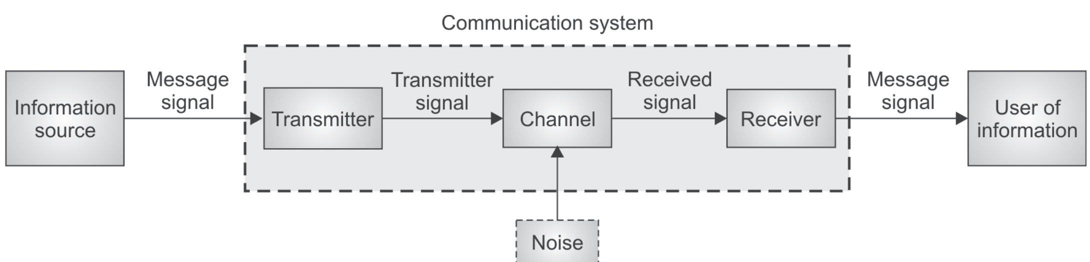

Fig. 1.5b: Electronic transmission system block diagram

1.4 RADIO FREQUENCY (RF) SPECTRUM

Radio signals are transmitted through electromagnetic waves. The radio waves have a wide frequency range starting from a few ten kilohertz (kHz) to several thousand megahertz (MHz). This wide range of frequencies is called the *radio spectrum* or RF spectrum.

We may note that the RF spectrum is classified according to the applications of the spectrum in different service areas. Table 1.1 summarizes the classification of the RF spectrum along with the associated applications in communication systems.

Table 1.1: Classification of the RF spectrum along with the associated applications in communication systems

Radio frequency range	Wavelenth, λ (meters)	Class	Applications
10–30 kHz	$3\times10^4 - 10^4$	Very low frequency (VLF)	Point-to-point communication (long distance)
30–300 kHz	10^4–10^3	Low frequency (LF)	Point-to-point communication (long distance) and navigation
300–3000 kHz	10^3–10^2	Medium frequency (MF)	Radio broadcasting
3–30 MHz	10^2–10	High frequency (HF)	Overseas radio broadcasting, point-to-point radio telegraphy, and telephony
30–300 MHz	10–1.0	Very high frequency (VHF)	FM broadcast, television, and radar
300–3000 MHz	1.0–0.1	Ultra high frequency (UHF)	Television and navigation
3000–30,000 MHz	0.1–0.01	Super high frequency (SHF)	Radar navigation and radio relays

1.5 CHANNEL OR MEDIUM

The signal from the transmitter section passes to the transmission medium after the required processing. The signal propagate through the transmission medium and is received at the other side by the receiver section. The transmission medium between the transmitter and receiver is called a *channel*.

The channel is very important part of the communication system. For example, most of the noise is added to the signal during its transmission through the channel. The transmitted signal should have adequate power to withstand the channel noise. However, the channel characteristics also impose constraints on the *bandwidth*. The bandwidth is the frequency range that can be transmitted by a communication system. Therefore, the channel characteristics are also taken into consideration as a design parameter while designing the transmitter and receiving equipment.

We may note that, in general, the transmitting power, signal bandwidth and cost of the communication system are affected by the channel characteristics.

1.5.1 Classification of Channels

Depending on the physical implementation, one can classify channels into the following two groups:

- Hardwire
- Softwire

(i) Hardwire Channels

These are man-made structures and widely used as the transmission medium. There are three possible implementations of a hardwire channel. These are:

- Transmission lines
- Waveguides
- Optical fiber cables (OFC)

Twisted pair cables are used in landline telephony and coaxial cables are used for cable TV transmission. These are examples of transmission lines.

Transmission lines are not suitable for ultra high frequency (UHF) transmission. Waveguides are employed as medium to transmit signals at UHF range. These are hollow, circular, or rectangular metallic structures. The signals enter the waveguide, are reflected at the metallic walls, and propagate towards the other end of the waveguide.

In general, there is always a physical link between the transmitter and receiver in hardware channels. A communication system that makes use of hardwire channel is known as a *line communication system*. Examples of such systems are landline telephony and cable TV network.

(ii) Softwire Channels

One can use certain nature resources as transmission medium for signals. Such transmission media are termed *softwire* channels. The possible natural resources that can be used as softwire channels are as follows:

- Air or open space
- Seawater

In communication systems that use softwire channels, there is no physical link between the transmitter and the receiver. The transmitter passes the signals in the required form to the softwire channel. The signals propagate through the natural resource and reach the receiver.

Air or open space is the most widel used softwire channel. The signals are transmitted in the form of *electromagnetic* (em) waves, also called *radio waves*. Radio waves possess the capability of travelling through open space at a speed equal to that of light, i.e. 3×10^8 m/s. The transmitter section converts the electrical signal into electromagnetic waves or radiation by using a transmitting antenna. These waves are radiated into the open space by the transmitting *antenna.*

At the receiver side, another antenna, called the receiving antenna, is used to pick up radio waves and convert them into the corresponding electrical signals. Systems that use radio waves to transmit signals through open space are called *radio communication systems*. Examples of such systems are radio broadcast, television transmission, satellite communication, and cellular communication.

A *SONAR* (sound navigation and ranging) system makes use of seawater as the channel of communication. This system transmits audible baseband signals that easily propagate through the seawater. SONAR systems are mainly used in submarines and ships to detect the presence of enemy submarines and ships.

1.6 NOISE

In electronic and communication engineering, noise is defined as *unwanted electrical energy of random* and unpredictable nature present in the system due to any cause.

In Fig. 1.1, the noise block represents the total noise present in the system, contributed by all the sources. The noise signal, $n(t)$, is applied to the block channel. However, this does not mean that the noise is intermingled with the signal only during its propagation through the channel. In fact, the channel contributes the major part of the noise. However, other noise sources along the communication can also add noise to the signal. The noise may also be mixed with the signal from within the transmitting and receiving equipment. As it is not possible to show all the individual sources of noise along the communication chain, we have illustrated, only one noise block in Fig. 1.1, beneath the channel block, as the channel is the main source of noise.

The noise introduced by the transmission medium is called *extraneous* or *external noise*, and the noise introduced by the transmission and reception equipment is called *internal noise*. The main cause of internal noise is the *thermal agitation* of atoms and electrons of the electronic components used in the equipment.

(i) External Noise

External noise is divided into following three groups:

1. Atmospheric or static noise.
2. Man-made or industrial noise.
3. Extraterrestrial or space noise.

Outer space is also a source of external noise that comes from the sun and stars. This noise is divided into:

- Solar noise
- Cosmic noise.

(ii) Internal Noise

The noise introduced within the receiver is known as internal noise. Internal noise can be grouped into:

- Thermal or Johnson's noise
- Shot or transistor noise

Thermal or Johnson's noise

This noise is also called *white noise*. It is generated with the resistors used in the circuit. This noise appears due to the random motion of electrons and molecules inside the resistor. The resistive part of the complex impedance in a circuit is also a source of thermal noise. Due to thermal energy, the electrons and molecules inside a resistor agitate and generate random electrical signals, which is noise. The maximum output noise power, P_n, of a resistor is given by

$$P_n = kTB \tag{1.2}$$

where, $k(= 1.38 \times 10^{-23}$ J/K) is Boltzmann constant, T is absolute temperature and B is bandwidth in Hz.

One can measure thermal noise by connecting a high sensitive AC voltmeter across the resistor. The random movement of electrons constitutes a current and develops a voltage across the resistor. This voltage is the noise voltage, and is measured by an AC voltmeter.

Shot or transistor noise

This noise is generated used in an amplifier. Shot noise is not only generated in an amplifier, it is also present in active devices, e.g. bipolar junction transistor (BJT). Shot noise appears due to the random movement of the electrons and holes inside the transistor and the presence of random electrons at the output terminal of the transistor in an amplifier. If there is an amplifier in the next stage this noise is amplified. If the amplifier noise is fed to a speaker, it produces an effect similar to lead shots falling on a metallic sheet, and hence the name shot noise.

The mathematical expression for shot noise shows that it is directly proportional to the output current, and inversely proportional to the transconductance of the transistor.

(iii) SNR and Noise Figure (F)

The SNR of a circuit is defined as the ratio of the signal power to the noise power at a point in the circuit. SNR is the measure of the signal power relative to the noise power at a particular point in a circuit.

If P_S is signal power and P_N is noise power, then SNR, expressed as $\frac{S}{N}$, is given as:

$$\frac{S}{N} = \frac{P_s}{P_n} \tag{1.3}$$

If $P_s = V_s^2 R$ and $P_n = V_n^2 R$, then

$$\frac{S}{N} = \frac{P_s}{P_n} = \frac{V_s^2 R}{V_n^2 R} = \frac{V_s^2}{V_n^2} \tag{1.4}$$

where V_s is signal voltage and V_n is noise voltage.

Further, it is assumed that both the signal and noise power are dissipated in the same resistor. Therefore, SNR is expressed in terms of decibels (dB) as:

$$\left(\frac{S}{N}\right)_{dB} = 10 \log_{10} \frac{V_s^2}{V_n^2}$$

$$\Rightarrow \left(\frac{S}{N}\right)_{dB} = 20 \log_{10} \frac{V_s}{V_n} \tag{1.5}$$

If, at a particular point in a circuit, the signal and noise voltage are given as 3.5 mV and 0.75 mV, respectively, SNR is obtained as:

$$\left(\frac{S}{N}\right)_{dB} = 20 \log_{10} \frac{3.5}{0.75}$$

$$\Rightarrow \text{SNR} = 20 \log_{10} (4.66)$$

$$\Rightarrow \text{SNR} = 13.38 \text{ dB}$$

Obviously, the SNR of the circuit at the point is 13.38 dB.

We may note that the noise figure is the measure of the noise introduced by the circuit. One can define it as the ratio of the signal-to-noise power at the input of the circuit and signal-to-noise power at the output of the circuit.

Noise figure (F) is expressed as:

$$F = \frac{\frac{S}{N}\text{Power at the input terminals of the circuit}}{\frac{S}{N}\text{Power at the output terminals of the circuit}}$$

One can see that if F is unity, the noise power introduced by the circuit is zero, as both the input and output $\frac{S}{N}$ powers are same.

1.7 RECEIVER

Receiver is a set up that receives the signal sent by transmitter through communication channel and process it and finally converts into or oiginal message. The most important function of receiver is *demodulation* and sometimes *decoding* as well which is reverse of modulation in transmitter. The output of the receiver may be fed to a loud speaker, video displaying, teleprinter, TV, picture tube or computer. The transmitter and receiving must be in synchronization with the modulation and coding methods used. Figure 1.6 shows schematic arrangement for transmission and reception.

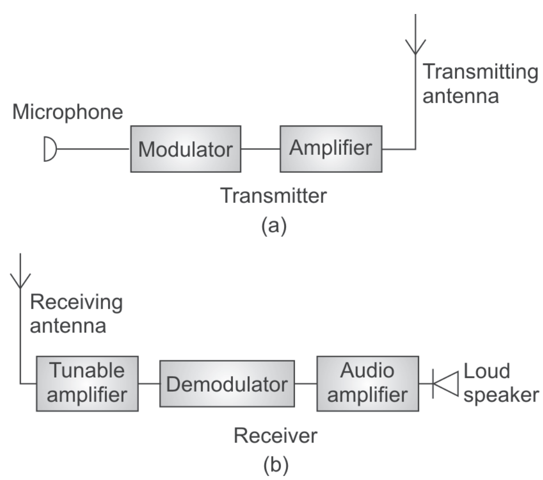

Fig. 1.6a and b: Schematic arrangement for transmission and reception of a message signal

The transmitter performs modulation on the baseband signal to translate it to a higher spectrum from its low frequency spectrum. The receiver in turn, performs an operation known as *demodulation*, which brings the baseband signal from the higher frequency spectrum to its original low frequency spectrum. The demodulation process removes the high frequency carrier from the received signal and retrieves the original baseband. This occurs in *carrier communication system*.

In baseband communication systems, assume that the transmitter replaces the digital baseband signal with a series of sinusoidal waveforms of different frequencies

(Fig. 1.4). When the receiver receives this signal, it recovers the original baseband pulse by replacing the two sinusoidal waveforms with the corresponding original levels.

The recovered baseband signal is then handed over to the final destination, which uses a transducer to convert this electrical signal to its original form. It is essential that enough signal power is given to the transducer so that is satisfactorily reproduces the message. Therefore, before handing over the recovered baseband signal to its final destination, the voltage and power are amplified by the amplifier stages.

Figure 1.7 shows the block diagram of a typical receiver section.

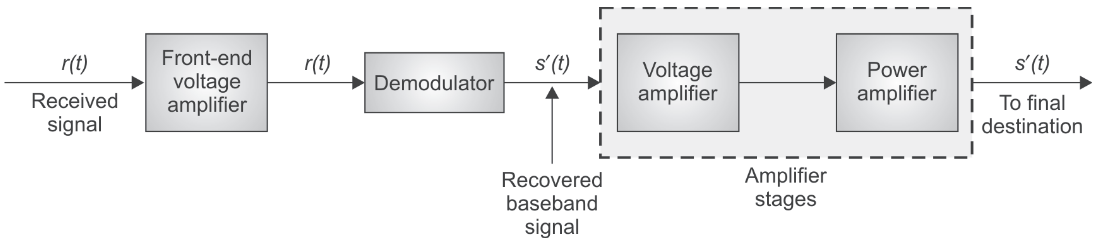

Fig. 1.7: Detailed block diagram of a typical receiver section

Figure 1.7 shows that the received signal, $r(t)$, is first amplified by the front-end voltage amplifier. This is done to strengthen the received signal, which is weak and to facilitate easy processing. Next this signal is given to the demodulator, which in turn, demodulates the received signal to recover the original baseband signal. The type of demodulation is based on the type of modulation employed at the transmitter. After recovering the original baseband signal, its voltage and power is amplified before applying it to the final destination block.

1.8 FINAL DESTINATION

The final destination is the last block of the communication chain. The destination is the user receives the information and interprets it for useful purposes.

The user can either be a human being or a machine that can make certain decisions based on the information received. For example, if a message is transmitted in the form of spoken words, then the user can be a human being who makes certain decisions on its reception. A live telecast of a football match is meant for a human being as a user. In some cases, a computer can be user that interprets the received information and takes action based on the information.

The destination block converts the baseband signal, which contains information in electrical from back into its original nonelectrical form, to be delivered to the user. The constituents of the destination block are an output transducer, which converts the baseband signals from its electrical form to its original nonelectrical form and the user. The output transducer works opposite to the input transducer.

If the transmitter transmits a voice signal, then the output transducer is the loudspeaker. This function is reversed in a microphone. The speaker delivers the message in the form of sound energy, which contains some information, to a human being who is the user of this information. The user interprets this information and accordingly takes some decisions. This completes the communication process. Figure 1.8 shows the detailed block diagram of the final destination.

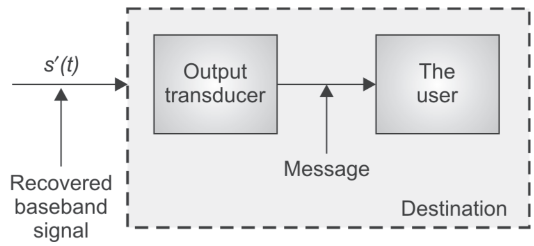

Fig. 1.8: Detailed block diagram of the final destination

A detailed schematic diagram of a complete communication chain of a basic communication system is shown in Fig. 1.9. The diagram shows all the blocks of Fig. 1.1 with their detailed processes.

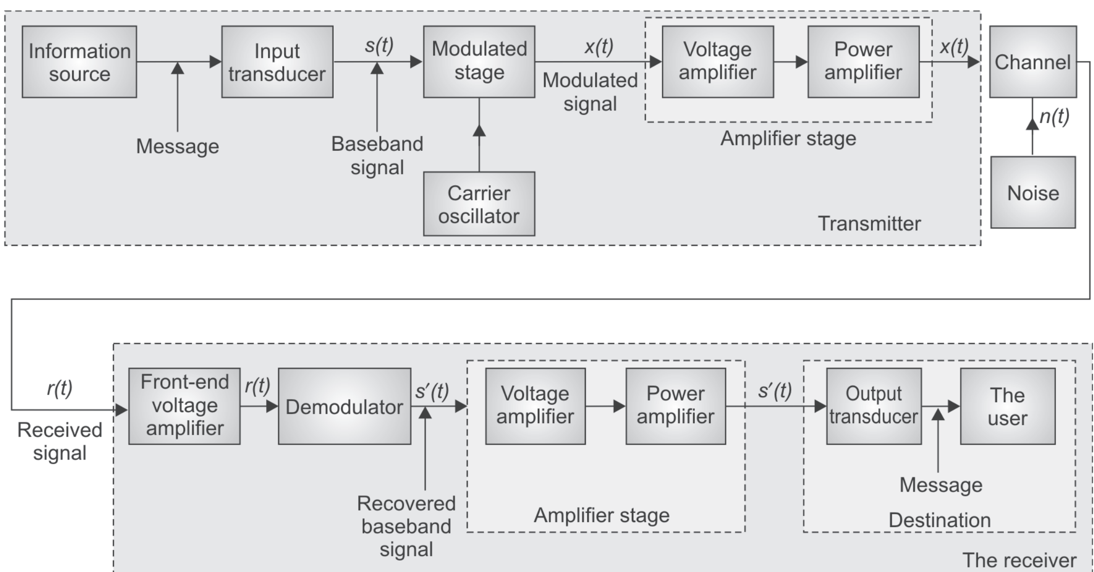

Fig. 1.9: Detailed schematic diagram of a basic communication system

1.9 INFORMATION AND BANDWIDTH

The basic aim of a communication system is to transmit information from a source to destination. The information is transmitted from a transmitter to a receiver through the channel. However, the amount of information that can be transmitted on a channel depends on the channel bandwidth, which is also the bandwidth of the communication system. As the RF spectrum is limited, the channel bandwidth and the bandwidth of the specific communication system are allocated by the Federal Communication Commission (FCC), which works in cooperation with the International Telecommunications Union (ITU), an agency of the United Nations. Obviously, the channel bandwidth is an important parameter that decides the amount of information transmitted in a particular communication system.

Hartley's law specifies the relation between the information and bandwidth of a communication system. Another factor that is involved with the Hartley's law is the transmission time, which is the time taken by the transmitted signal to reach the receiver. According to Hartley's law, the bandwidth of the system is directly proportional to the product of bandwidth of the system and time of transmission, i.e.

$$\text{Information} \propto (\text{bandwidth} \times \text{transmission time}) \tag{1.6}$$

According to Hartley's law, a communication system will accommodate more information if the bandwidth of the system is large. For instance, the bandwidth of a television transmission system is 6 MHz, While it is 200 kHz in case of FM radio broadcast. Therefore, the information transmitted in a television transmission system is more than that transmitted in an FM radio broadcast.

1.10 HIGH FREQUENCY EFFECTS

1.10.1 Frequency Ranges in use

Theoretically all radio frequency signals above DC will radiate electromagnetic energy and thus propagate through space. However, for those frequencies below about 15 kHz the radiation is insignificant. This figure should not be confused with the higher limit of the audibility of sound waves by the human ear. Radio and sound waves propagate via completely different phenomena. Typically then, the part of the electromagnetic spectrum that is suitable for radio communications lies in the range 15 kHz to 300 GHz, although current applications above about 30 GHz are very limited. The ranges and nomenclature of the various sub-bands in use are shown in Table 1.2.

Table 1.2: Sub-band ranges

Frequency	Name	Wavelength
Up to 30 kHz	Very low frequency (VLF)	>10 km
30–300 kHz	Low frequency (LF)	10 km to 1 km
300 kHz to 3 MHz	Medium frequencies (MF)	1 km to 100 m
3–30 MHz	High frequencies (HF)	100 m to 10 m
30–300 MHz	Very high frequencies (VHF)	10 m to 1 m
300 MHz to 3 GHz	Ultra high frequencies (UHF)	1 m to 10 cm
3–30 GHz	Super high frequencies (SHF)	10 cm to 1 cm
30–300 GHz	Extra high frequencies (EHF)	1 cm to 1 mm

Table 1.3: Microwave frequency bands

	Frequency ranges (GHz)	
Band	American	European
P	0.2–1.0	0.2–0.375
L	1–2	0.375–1.5
S	2–4	1.5–3.75
C	4–8	3.75–6
X	8–12.5	6–11.5
j	–	11.5–18
Ku	12.5–18	–
K	18–26.5	18–30
Ka	26.5–40	–
Q	–	30–47

The frequency ranges used for microwave communications are commonly known by the American Radar Engineering Standard (Table 1.3). The alternative classification used in Europe is also shown in Table 1.3.

Communication systems are operated at high frequencies due to several reasons, as explained in Section 1.10. The circuits at high frequencies are confronted with some unwanted effects. These are termed high frequency effects in communication systems and are related to the wires used in the circuits.

When an AC current is passed through a wire, it exhibits the presence of inductance due to the magnetic field produced around the wire. At low frequencies, the inductive reactance in the wire is very small and can be neglected. Therefore, one can consider the inductor as a short because the inductive reactance is very small. However, at higher frequencies, the inductive reactance is large and one cannot ignore it.

The high inductive reactance results in a high impedance offered by the wire to the high frequency current. The high impedance attenuates the signal and results in loss of power. Further, it also introduces a phase shift in the signal, which may alter the signal. The amount of inductance in a wire is a function of its length. The longer the wire, the higher the inductance. Therefore, to minimise the higher frequency effects in wire, the lengths of the wires should be as small as possible.

Another effects the appears in circuits at higher frequencies are the *stray capacitance*. The stray capacitance appears between two closely placed wires in the circuit. This capacitance also appears between the two ends of resistor and across a coil used in the circuit. The capacitive reactance at low frequencies is very high and does not affect the performance of the circuit. However, at higher frequencies, the capacitive reactance is very small and it becomes a short circuit. This short circuit effects provides a path for the signal to bypass the resistor or coil. This affects the functioning of the circuit, because an unwanted signal is made available at the other end of the resistor or coil through the stray capacitance. Cutting the component leads to the shortest possible length minimizes the effect of stray capacitance.

The stray capacitance that appears between two wires in the circuit provides a path for the signal at higher frequencies. The signal leaks from one wire to another and results in the loss of signal power. The unwanted signal that passes through the stray capacitance between two wires alters the signal that propagates through the wires. This is also not desired as it affects the performance of the systems. Placing the wires at larger distances apart can minimize the effects of stray capacitance appearing between two wires.

1.11 MODULATION

As stated in the previous section, in a carrier communication system, the baseband signal of a low-frequency spectrum is translated to high frequency spectrum. This is achieved through modulation. *Modulation is the process by virtue of which some characteristic of a high frequency sinusoidal wave is varied in accordance with the instantaneous amplitude of the baseband signal.* A grossly understand definition of the process modulation is an "up shifting" of the message frequencies (intact) to a range more useful for transmission. A message that contains all of the frequencies between 1 Hz and 5000 Hz, a bandwidth of 5 kHz, could be shifted upward in frequency to the range of 100,001 to 105,000 Hz and retain the 500 Hz bandwidth. If this is done correctly the message will not lose its meaning; the receiver will simply downshift the signal to its original span of 1 Hz to 5 kHz. This, of course, is an over simplification, because the conversion requires several intermediate steps.

Need for Modulation: Two signals are involved in the modulation process, the *baseband signal* and the *carrier signal*. The baseband signal is to be transmitted to the receiver. The frequency of this signal is generally low. In the modulation process, this baseband signal is called the *modulating signal*. The waveform of this signal is unpredictable, e.g. the waveform of *speech signal* is random in nature and cannot be predicted. In this case the speech signal is the modulating signal.

The other signal involved with the modulation is a high frequency *sinusoidal wave*. This signal is called the carrier signal or carrier. The frequency of the carrier signal is always much higher than that of the baseband signal. After modulation, the baseband signal of low frequency is transferred to the high frequency carrier, which carries the information in the form of some variations. After the completion of the modulation process, some characteristic of the carrier is varied such that the resultant variations carry the information.

A low-frequency baseband signal is thus translated to a high frequency carrier such that the information is coded in the variations in one of the parameters of the carrier. At the receiver end, these variations are detected through the demodulation process to recover the original baseband signal.

Clearly, audio signals having bandwidth of about 20 kHz, i.e. such low frequency signals cannot be transmitted directly to long distance and hence require to be modulated. There can be problems if modulation is not used.

The baseband signal will be transmitted as it is, if modulation is not employed. However, the system designer could confront the following problem:

(i) Antenna height: In order to transmit a signal effectively, e.g. in radio communications, the height of the antenna should be comparable to the wavelength of the signal. If h is the height of antenna, λ is the wavelength of the signal, f is the frequency of the signal and c is the velocity of light (3×10^8 ms^{-1}), then the required height of the antenna, according to rule is:

$$h = \lambda/4 \tag{1.7}$$

where

$$\lambda = \frac{c}{f} \tag{1.8}$$

The quarter wavelength may be considered as the required height of antenna as given by Eq. (1.7). Substituting Eq. (1.8) in Eq. (1.7), the height of antenna is:

$$\lambda = \frac{c}{4f} \tag{1.9}$$

From Eq. (1.9), it is clear that the *height of an antenna is inversely proportional to the signal frequency*. Consider the example of radio broadcast system that transmits programs in the audio range. The audio frequency range is 20 Hz to 20 kHz. It is now required to calculate the height of the antenna required to transmit the two extreme frequencies, assuming that the audio frequency signal is converted into radio waves. On substituting f = 20 Hz in Eq. (1.9), one obtains

$$h = \frac{3 \times 10^8}{4 \times 20} = 3750 \times 10^3 \text{ m}$$

$$\therefore \qquad h = 3750 \text{ km}$$

Now, substituting, f = 20 kHz in Eq. (1.9), one obtains

$$h = \frac{3 \times 10^8}{4 \times 20 \times 10^3} = 3.750 \text{ km}$$

Therefore, to transmit audio signals, the height of the required antenna lies between 3.7g km and 3750 km. To transmit the entire range of audio signals, the height of the antenna should be 3750 km, which is greater of the two heights. It is obvious that this height is not *feasible*.

The solution of this problem lies in Eq. (1.10), which signifies that the height of an antenna is inversely proportional to the frequency of the signals. *The higher the frequency of the signal,* the lower, *the required* height of the antenna. Clearly, if the frequency of the transmitting signal is increased, the height of the antenna can be reduced and brought within practical limits. This is done using modulation. The modulation process translates the low frequency baseband signal to a higher frequency spectrum. Therefore, if a modulated signal is transmitted instead of the baseband signal, then the height of antenna is reduced.

Suppose the audio signal, ranging from 20 Hz to 20 kHz, modulate a carrier signal of 1.5 MHz. After modulation, the baseband signal is translated to the central frequency of 1.5 MHz. The antenna height required to transmit a radio signal of 1.5 MHz is obtained using Eq. (1.10) as:

$$h = \frac{3 \times 10^8}{4 \times 20 \times 10^6} = 50 \text{ m}$$

This height is clearly within practical limits, as it is possible to erect a 50 m high antenna. Therefore, the height of antenna makes it mandatory to use modulation.

(ii) Narrow banding: In almost all communication systems, the baseband signal covers a wide band of frequencies, e.g. audio signals lie in the range, 20 Hz to 20 kHz, while video signal lie in the range, 0 to 5 MHz. In radio communication systems, the frequency of the transmitted signal decides the height of antenna. If an audio signal is to be transmitted, then each frequency component within the audio range suggests a particular height of antenna. If the baseband signal is directly transmitted, then the communication system requires as many antenna as the number of frequency components. As this is not practical, the only solution is to devise a method by which a smaller number of antennas, preferably a single antenna, can be used to satisfactorily transmit the entire range.

The solution to this problem can be found by analyzing the band-edge ratio which is an important parameter for antennas.

In an audio baseband signal ranging from 20 Hz and 20 kHz, the band edge ratio is

$$\frac{20}{20 \times 10^3} = \frac{1}{1000} = 1:1000$$

Therefore, the band-edge ratio of the audio range is 1: 1000. The practical significance of this ratio is that if the height of the antenna required to transmit 20 KHz signal is 1 unit, then the height of the antenna required to 20 Hz signal will be 1000 units. Since the band edge ratio is very large, it is not possible to transmit the entire audio range using a signal antenna. lf, by some means, this ratio is lowered, then it may be possible to solve this problem.

The band-edge ratio is lowered by modulation. Let a carrier signal of 1.5 MHz be modulated by an audio baseband single. After modulation, the lowest and highest frequency components will be

Lowest frequency component

$$= (1.5 \times 10^6 + 20) \text{ Hz} = 1500020 \text{ Hz}$$

Highest frequency component

$$= (1.5 \times 10^6 + 20 \times 10^3) \text{ Hz} = 1520000 \text{ Hz}$$

The band-edge ratio after modulation is

$$\frac{1500020}{1520000} = 0.9868 \cong 1$$

Therefore, the band-edge ratio is at l:1

After modulation, the band-edge ratio is reduced to 1:1 from 1:1000. Thus, a single antenna can be used to transmit the entire range of baseband signals. With this arrangement, a wide frequency band, ranging from 20 Hz to 20 kHz, is practically narrowed so that the ratio of the lowest to the highest frequency components is 1:1. This is called *narrow banding and becomes possible only because of the modulation process*.

(iii) Poor radiation and penetration: For linear antenna of length *l*, it is observed that

$$\text{Power radiated} \propto \left(\frac{1}{\lambda}\right)^2 \tag{1.10}$$

Clearly, for the same antenna length, the power radiated by short wavelength or high frequency signals would be large. If the audio signals of longer wavelength are directly radiated into space, they die out after covering some distance due to low power radiated by the antenna. This reveals that to transmit them over large distances they are superimposed on high frequency carrier waves. In other words, to get sufficient radiated power from the antenna, the frequency of the transmitted signal have to be increased. This is achieved by using the modulation process before transmitting the signals.

There are modes of propagation of radio waves in free space. One of the modes of propagation of radio signals through free space makes use of the ionospheric layers of atmosphere. For satisfactory reception of transmitted signals, it is required that the radio waves efficiently penetrate the ionospheric layers.

According to the propagation theory radio waves with higher frequencies penetrate more into the layers, as compared to those with lower frequencies. If the low frequency baseband signals are directly transmitted, then the radio waves so produced will not be able to effectively penetrate the ionospheric layers. Therefore, it is necessary to increase the frequencies of these signals before transmission. This is another reason for using modulation.

(**iv) Diffraction angle:** In some radio communication systems, such as microwave links and radar systems, radio waves are radiated in the form of narrow beams. A narrow beam is specifically required in these systems to ensure that radio waves of high power intensity are received at the receiver. This high power intensity is required so that the signal can cope with the noise and disturbance present in the space. Further, the concentration of the radiated energy is more at the specific target, i.e. the receiver, if the beam is narrow.

When the narrow beam travels through the channel, it may strike some obstacles resulting in diffraction. After diffraction, the beam spreads and its narrowness reduces. This result is reduced power intensity at the receiver, which is undesirable. It is established that the *spreading of a beam is proportional to the diffraction angle.* The diffraction angle (α) is proportional to the ratio of the wavelength (α) of the signal to the diameter (D) of the antenna, i.e.

$$\alpha \propto \frac{\lambda}{D} \tag{1.11}$$

But $$\lambda = c/f$$

Hence $$\alpha \propto \frac{c}{fD} \tag{1.12}$$

Eq. (1.7) clearly reveals that the diffraction angle is inversely proportional to the frequency *f*, of the signal. Therefore, to reduce the diffraction angle, the frequency of the signal should be increased. Due to this fact, the baseband signal is not transmitted at its own frequency. The modulation process increases the frequency of the transmitted signal. As a result, the diffraction angle reduces, which in turn helps to avoid extra spreading of the beam.

(v) Multiplexing: This is the technique that is most widely used in nearly all types of communication systems. Multiplexing facilitates the simultaneous transmission of multiple messages over a single transmission channel.

Multiplexing allows the maximum possible utilization of the available bandwidth of the system. Bandwidth is an important entity in any communication system. The use of multiplexing also makes the cummunication system economical because more than one signal can be transmitted through a single channel. Multiplexing is possible in communication systems only through modulation.

In order to transmit different audio signals (which fall in the same spectral range) simultaneously through one antenna range so that it is easily recoverable and distinguishable from other signals at the receiving station. At the receiver, the different signals can be easily separated because they are at different frequencies, and these can be delivered to the next stages of the receiver for further processing, clearly multiplexing becomes possible only because of modulation.

1.11.1 Continuous Wave Modulation

We can represent a sinusoidal carrier wave

$$c(t) = A_c \cos(\omega_c t + \phi_0) \tag{1.13}$$

where $c(t)$ is voltage or current signal strength, A_c is the amplitude, $\omega_c(= 2\pi f_c)$ is the angular frequency and is the initial phase of the carrier wave. Let $\omega_c t + \phi_0 = \phi$, then ϕ is called phase of the wave. Eq. (1.13) reveals that a earner wave can be modulated in the following, two ways:

(i) Amplitude Modulation (AM): This is the oldest technique of modulation, in which an analog baseband signal is translated to a high frequency spectrum. *AM is defined as the modulation process that varies the instantaneous amplitude of the carrier signal in accordance with the instantaneous amplitude of the modulated signal*.

The modulating signal is the baseband signal that is to be transmitted to the destination. This signal is low in frequency random and unpredictable.

The carrier signal carries the information in the form of variations in one of its parameters. This signal is always a high frequency sinusoidal wave. In amplitude modulation, the information is coded as the variations in the amplitude of the carrier signal. These variations are proportional to the instantaneous amplitude of the modulating signal.

AM is used extensively in many applications including television broadcasting, radio, broad casting, satellite communication, and cable television transmission.

(ii) Angle or Angular Modulation: Frequency and Phase Modulation

The total phase angle $\phi = (\omega_c t + \phi_0)$ (Eq. 1.13) comprises the frequency, f_c and initial phase angle ϕ_0. According to definition of modulation, the frequency and phase of a carrier can vary in accordance with the modulation signal. In other words, the modulating signal can vary the total phase angle of the carrier signal to get a particular form of modulation. The type of modulation is known as *angle modulation*. Angle modulation can be performed in two ways:

- Frequency modulation
- Phase modulation

The total phase angle (ϕ) of a carrier signal has a frequency, f_c and an initial phase, ϕ_0 as its constituents. A modulating signal may vary the frequency of the carrier keeping the amplitude and phase constant. This type of modulation is called *frequency modulation* (FM).

A modulating signal may also vary the phase of the carrier signal, and maintain constant values of amplitude and frequency. The technique gives rise to another form of modulation, called phase modulation.

1.11.2 Amplitude Modulation

Amplitude modulation can be further explained by using the physical appearance of the amplitude modulate (AM) wave and other signals involved. To explore the physical appearance of the AM signal, the following signals are considered:

Modulating signal: A pure sine wave whose amplitude is A_m and instantaneous amplitude is a_m.

Carrier signal: A pure sine wave whose amplitude is A_c and instantaneous amplitude is a_c. The frequency of this signal is much higher than that of the modulating signal. Further, the amplitude of the carrier signal, A_c is also greater than that of the modulating signal ($A_c > A_m$).

The carrier and modulated signals are shown in Fig. 1.10a and b, respectively. Fig. 1.10c shows the resulting AM wave, the carrier signal after modulation. Obviously, the amplitude of the carrier wave changes in accordance with the modulating wave. The modulating (Fig. 1.10b, amplitude modulates this carrier signal. According to the definition of AM, the instantaneous amplitude of the carrier signal varies in proportion to the instantaneous amplitude of the modulating signal, which means that after modulation, the amplitude of the carrier varies. The frequency and phase of the carrier signal, however, remain unaltered.

A careful observation of Fig. 1.10c reveals that

(i) Amplitude of the carrier wave changes according to the modulating signal.

(ii) Amplitude of the carrier wave changes at the frequency f_m of the modulating signal.

(iii) Frequency of the AM wave is equal to the unmodulated carrier frequency 'f_c'.

Clearly, the amplitudemodulated wave is the carrier signal with variations in its instantaneous amplitude. It can also be seen from Fig. 1.10c that the peak values of the carrier signal at different points in time vary according to the instantaneous amplitudes of the modulating signale. From 1.10c, it is also clear that the instantaneous amplitude of the carrier signal increases or decreases in accordance with the variations in the amplitude of modulating signal.

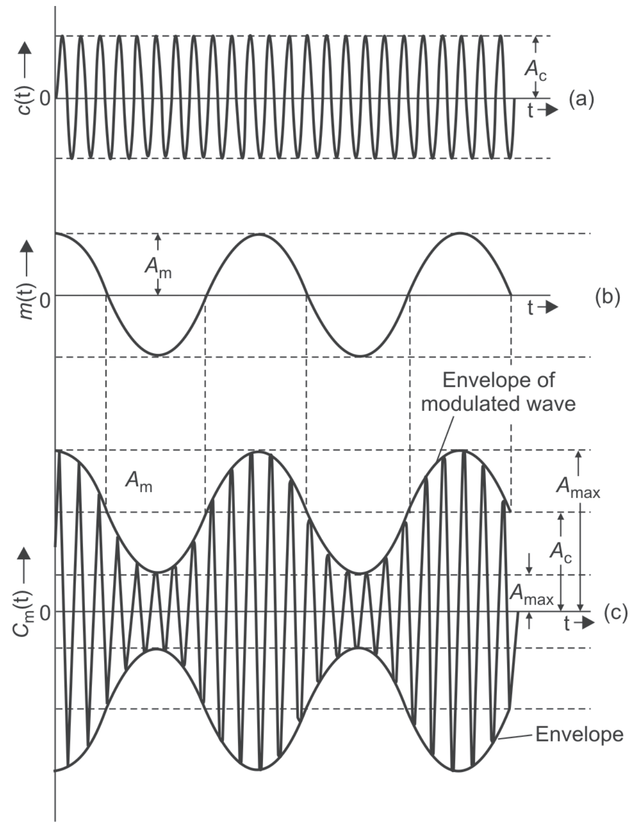

Fig. 1.10: Waveform related to amplitude modulation. (a) Carrier signal, (b) modulating signal, (c) amplitude-modulated wave

From Fig. 1.10c, it is clear that the instantaneous amplitude of the carrier signal increases or decreases in accordance with the variations in the amplitude of modulating signal. In addition, at the zero crossing of the modulating signal, there is no change in the amplitude of the carrier signal, and it remains at A_c because, at these instants, the amplitude of the modulating signal is zero. This way, the low frequency baseband signal is transmitted in the form of amplitude variations of the high frequency carrier signal. The information is therefore, carried by the envelope of the modulated wave.

1.11.2 Modulation Index or Modulation Factor or Depth of Modulation

It is defined as the ratio of the change in the amplitude of the carrier wave to the amplitude of the original carrier wave or as the ratio of amplitudes of the modulating signal and the carrier signal. The modulating index is designated as m_a or m, where a signifies modulation. This means.

$$m_a \text{ or } m \frac{A_m}{A_c} \quad (1.14)$$

$$= \frac{\text{Change in amplitude of carrier wave}}{\text{Amplitude of original carrier wave}}$$

Let A_{max} and A_{min} be the maximum or minimum voltages of AM wave. We have from Fig. 10c

$$A_m = \frac{A_{max} - A_{min}}{2} \quad (1.15)$$

and

$$A_c = A_{max} - A_m$$

$$= A_m - \frac{A_{max} - A_{min}}{2} \quad (1.16)$$

Hence, modulation factor (m) in terms of A_{max} and A_{min} is:

$$m = \frac{A_m}{A_c} = \frac{(A_{max} - A_{min})/2}{(A_{max} + A_{min})/2}$$

$$= \frac{A_{max} - A_{min}}{A_{max} + A_{min}} \quad (1.17)$$

The modulation index is a very important parameter that decides the physical appearance of an AM wave and subsequently put constraints on other related parameters of AM wave. Modulation factor m generally lies between 0 and 1.

The modulation index is also sometimes called the *modulation factor*, modulation coefficient, depth of modulation or degree of modulation. Generally, the value of misinfractions. When this value is multiplied by 100, m is expressed in percentage modulation as:

$$m = \frac{A_m}{A_c} \times 100\% \quad (1.18)$$

or

$$m = \frac{A_{max} - A_{min}}{A_{max} + A_{min}} \quad (1.19)$$

It is customary to express m as an percentage rather than a fractional value.

Analysis of Eq. (1.14) reveals that there may be three possible values of m depending on the values of A_m and A_c. The three possible values of m are:

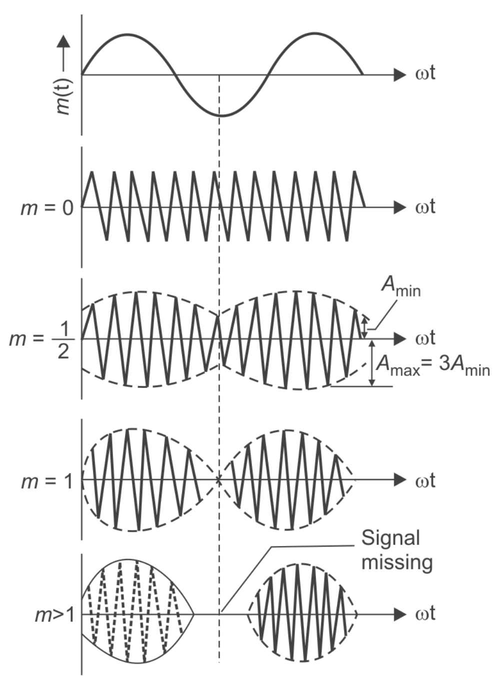

Fig. 1.11

- $m = 1$, when $A_m = A_c$, called 100% modulation
- $m > 1$, when $A_m > A_c$, called over modulation
- $m < 1$, when $A_m < A_c$, called under modulation

Thus, the value of m should lie between 0 and l, including 1 but excluding 0 , i.e.

$$0 < m \leq 1 \tag{1.20}$$

For m = 0% (carrier wave with unchanged amplitude), m = 50% (under modulated signal), m = 100% (full modulated signal) and m = 150% (over modulated signal), i.e. m = 0, 0.5, l, 1.5 values plotted in Fig. 1.11.

1.11.3 Equation of Amplitude Modulated Wave

Let us consider that modulating signal ($m(t)$) and carrier signal ($c(t)$) are represented by

$$m(t) = A_m \sin \omega_m t \tag{1.20(a)}$$

and

$$c(t) = A_c \sin \omega_c t \tag{1.20(b)}$$

where, $m(t)$ is instantaneous amplitude of the modulating signal,

A_m is amplitude (peak value) of the modulating signal,

$\omega_m = 2\pi f_m$, where ω_m is the angular frequency of modulating wave signal,

$\omega_c = 2\pi f_c$, is the angular frequency of the carrier signal and f_c is the frequency of carrier signal.

A_c is amplitude (peak value) of the carrier. During the AM process, the modulating signal, A_m, modulates the carrier signal. A_c to result in the AM wave, A to write this equation, we consider the following:

- The frequency of the carrier signal remains the same after modulation.
- The phase of the carrier signal remains the same after modulation.
- The instantaneous amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal after modulation as shown in Fig. 1.10c.

On the basis of above considerations, we can write the equation for the amplitude (A) of the modulated wave as:

$$A = A_c + m(t) = A_c + A_m \sin \omega_m t$$

$$= A_c = mA_c \sin \omega_m t$$

$$= A_c (1 + m \sin \omega_m t) \tag{1.21}$$

where $m = \dfrac{A_m}{A_c}$ is the modulation index.

The instantaneous amplitude of the AM is given by

$$c_m(t) = A \sin \omega_m t = A_c (1 + m \sin \omega_m t) \sin \omega_m t$$

$$= A \sin \omega_m t + \frac{mA_c}{2} \times 2 \sin \omega_m t \sin \omega_m t$$

$$= A \sin \omega_c t + \frac{mA_c}{2} [\cos (\omega_c + \omega_m)t - \cos(\omega_c - \omega_m)t]$$

or

$$c_m (t) = A_c \sin \omega_c t - \frac{mA_c}{2} \cos (\omega_c + \omega_m)t + \frac{mA_c}{2} \cos (\omega_c + \omega_m)t \tag{1.22}$$

Eq. (1.22) is the final mathematical expression of an amplitude modulated wave. The first term in Eq. (1.22) carrier itself in its unmodulated form and the other two are new components. The frequency of second and third components are $(f_c + f_m)$ and $(f_c - f_m)$ respectively. Second and third components in Eq. (1.22) are called *upper sideband* (USB) and *lower sideband* (LSB). We have

- The frequency of the original carrier wave of amplitude
  A_c is $f_c = \omega_c/2\pi$
- The frequency of upper sideband (USB)
  $= f_c + f_m = (\omega_c + \omega_m)/2\pi$ (1.23)
- The frequency of lower sideband (LSB)
  $= f_c - f_m = (\omega_c + \omega_m)/2\pi$ (1.24)

We may note that the unmodulated carrier does not contain any information, as it is totally predictable. The information is carried only by the two sides; LSB and USB.

The carrier frequency is many times greater than the modulating frequency and therefore sideband frequencies lie close to the carrier frequency (Fig. 1.12).

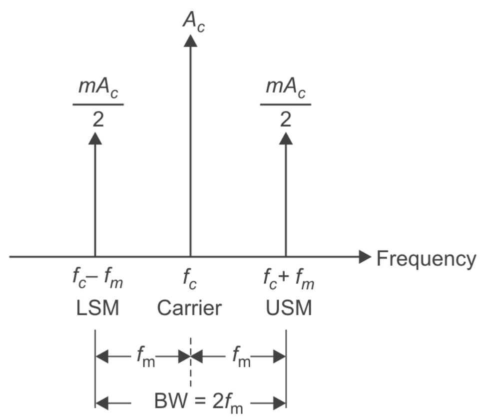

Fig. 1.12: Frequency spectrum of a sinusoidally AM signal

The difference of the highest and the lowest frequencies present in the AM signal is called the bandwidth (BW).

Thus, BW = $(f_c + f_m) - (f_c - f_m) = 2f_m$ (1.25)

Clearly, the bandwidth of an AM signal is twice that of the frequency of the modulating signal. The following facts can be mentioned from this frequency spectrum:

- The frequency of the carrier signal, f_c, is much higher than that of the modulating signal, f_m.
- The amplitude of the carrier signal, A_c, is greater than that of the modulating signal.
- The amplitude of LSB and USB are equal.
- The LSB and USB are equidistant from f_c and therefore, f_e, is also sometimes called the central frequency.
- To satisfactorily accommodate both the sidebands in the transmitted signal, the required bandwidth is as follows:

$$BW = (f_c + f_m) - (f_c - f_m) = 2f_m$$

From the above facts, it is clear that the low-frequency modulating signal is now translated to a high-frequency spectrum in the form of two side bands. Therefore, the information is carried by the two high-frequency sidebands.

1.11.4 Advantages and Disadvantages of AM

Advantages of AM

(i) AM is an easier method for transmitting and receiving voice signals.
(ii) AM requires quite simple and cheaper transmitters and receivers.
(iii) Transmission of AM signal requires low carrier frequencies of 0.5–20 MHz.
(iv) Area in which AM transmission signal can be received is much larger compared to the case of FM transmission signal.

Disadvantages of AM

(i) AM gets easily disturbed (both atmospheric machinery from noise).
(ii) In AM, quality of audio signal is poor.
(iii) Efficiency of AM transmission is low.

1.11.5 Phase Modulation (PM)

Phase modulation is defined as the process in which the instantaneous phase of the carrier signal is varied in accordance with the instantaneous amplitude of the modulating signal. In this type of modulation, the amplitude and frequency of the carrier signal remains unaltered after PM. The modulating signal is mapped to the carrier signal in the form of variations in the instantaneous phase of carrier signal.

PM is another form of *angle modulation*. PM and frequency modulation (FM) are closely related to each other. In both the cases, the total phase angle of the modulated signal varies. However, PM is not as extensively used as FM.

1.11.6 Frequency Modulation (FM)

FM is defined as the process by which the instantaneous frequency of a carrier signal varies according to the instantaneous amplitude of the modulating signal. It is clear from the definition that the modulating signal is concealed in the variation in the frequency of the carrier signal. Therefore, after FM, the frequency of the carrier signal f_c, will not remain constant, but vary according to the amplitude of the modulating signal.

If the amplitude of the modulating signal increases, then there will be a corresponding increase in the frequency of the carrier signal from its original frequency, f_c. Similarly, the carrier frequency at all instants of time correspondingly decreases if there is a decrease in the amplitude of the modulating signal. The amplitude and phase of the carrier signal remain constant after FM. Salient features of FM signal are as follows:

(i) The amplitude of the modulated signal remains same as that of the carrier wave but its frequency changes in accordance with the modulating signal amplitude.
(ii) The amount of frequency deviation δ, in FM is proportional to the instantaneous value of the modulating signal, i.e. $\delta \propto e_m$. Clearly, louder are audio signal, greater is the frequency change in the modulated carrier.
(iii) The total change in the carrier frequency after modulation is from $+\delta$ to $-\delta$ and in 2δ. This is called the *carrier swing.*
(iv) The rate of frequency deviation is equal to the modulated frequency.

Frequency deviation (δ) is defined as the maximum amount by which the frequency of the modulated signal changes or shifts above or below the carrier frequency (f_c), i.e.

$$\delta = f_{max} - f_c = f_c - f_{min} \tag{1.26}$$

Obviously, the total change in the carrier frequency after modulation is from +δ to –δ and is 2δ.

Modulation Index (m_f) for an FM wave is defined as the ratio of the maximum frequency deviation to the modulating frequency. m_f plays a very important role in deciding the bandwidth of the FM system.

Deviation ratio (D_f): In practical FM systems, the modulation signal is non-sinusoidal signal that contains a band of frequencies. The FM signal produced by such a modulating signal is called a *multi-tone FM signal*. The frequency deviation, in this case, will vary with time and in proportion to the instantaneous modulating signal. In the case of non-sinusoidal FM, the deviation ratio (D_f) is defined as:

$$D_f = \frac{\text{Maximum permissible frequency deviation}}{\text{Highest modulating frequency}} = \frac{\Delta f}{W} \tag{1.27}$$

1.12 TYPES OF COMMUNICATION SYSTEMS

Communication systems may be categorized according to *nature of information or mode of transmission* or type of *transmission channel* used. Thus, one can classify communication system as follows.

(a) Based on the nature of information source

(i) *Speech transmission,* e.g. in a radio
(ii) *Picture as well speech transmission,* e.g. in televisions
(iii) FAX (*facsimile transmission*): FAX is used for exact reproduction documents, maps, still pictures, etc. at a distance place.
(iv) *Data transmission,* e.g. in computers

(b) Based on mode of transmission

(i) Analog communication: In this system, one makes use of analog signal and analog electronic circuit in such a manner that the output voltage varies continuously in accordance with input voltage. We may remember that an analog signal is a function of time, and has a continuous range of values. There is a definite function value of the analog signal at each point of time. Examples of analog communication systems are: Telex telegraphy, TV network, etc. A pure sine wave is an example of an analog signal or analog waveform.
An analog operation has many valued output and therefore it is less reliable. This is why analog communication system is now rarely used in modern communication system.

(ii) Digital communication: This system of communication makes use of an electronic circuit that can handle only digital signals. We may remember that a digital signal does not have continuous function values on a time scale. It is discrete in nature, i.e. it has some fixed values at discrete timings. In between two consecutive values the signal value is either zero or a different value, e.g. sound signal produced by drumbeats. FAX, e-mail.

We can see that in order to reproduce the rectangular/square wave shape, we have to superimpose all the harmonics: f_0, $2f_0$, $3f_0$, $4f_0$, $5f_0$, ...

This clearly implies an infinite bandwidth. However, the contributions from the higher harmonics are negligible, which limits the bandwidth. This means that the received wave is the distorted version of the transmitted wave. We may note that if the bandwidth is large enough to contain a few harmonics, the information is not lost and rectangular signal is exactly reproduced. This is due to the fact that higher the harmonics, the less is its contribution to the waveform, etc. Modern communication systems are truly based on digital electronic systems due to following advantages:

(i) The two-valued digital operation is more reliable as compared to many valued analog operation because in digital operation all the signals are easily identified as low and high.

(ii) The information to be transmitted in digital operation is already in pulse form and hence its transmission require simple technique.

(iii) A large number of digital signals can be transmitted through a single channel only.

(iv) The signal fidelity is controlled well in digital transmission.

(v) The digital circuitry is relatively cheaper than the bulky analog transmission system. This is due to the tremendous growth achieved in the fields of digital integrated *chip* design, very large scale integration (VLSI), and the availability of advanced digital signal processing algorithms and processors.

(vi) Analog circuitry is subject to component tolerance, which are inherent in the analog circuitry.

(vii) One can easily store and recover digital signal from magnetic-media compared to the analog signals.

As most real life signals are continuous and therefore, analog in nature, they should be converted efficiently into digital signals before they can take advantage of the digital signal communication and processing technique. The two basic processes of converting an analog signal to a digital signal are *sampling* and *quantization*.

(c) Based on the method of transmission channel: Transmission channeI implies the path over which the signal is transmitted. There are two ways:

(i) Line communication: Two wireline communication, optical fiber communication, and coaxial cable.

(ii) Space communication: In space communication, the channel is simply the physical space between the transmitting and receiving antennas, and the behaviour of signals in that medium. ·

(d) Based on the type of modulation: Usually, the message or information or speech signals have low frequencies and hence cannot be directly transmitted to long distances. These signals are superimposed or loaded on a high frequency wave which serves as the *carrier* of information.

This process is called *modulation*. Modulation is a process by virtue of which, some characteristic of a certain frequency sinusoidal wave is varied in accordance with the instantaneous amplitude of the baseband signal. Two signals are involved in the modulation process, *the baseband signal* and *the carrier signal*. The baseband signal is known as modulating signal and it is a low frequency signal. The carrier signal is always a high frequency sinusoidal wave. During the modulation process, the modulating signal varies the frequency amplitude or phase of the carrier in accordance with its instantaneous amplitude. After modulation, the carrier is said to be modulated by modulating signal. The output of the modulator is called the *modulated signal*.

The types of modulation for *sinusoidal continuous carrier wave are*:

(i) Amplitude modulation (AM)
(ii) Frequency modulation (FM)
(iii) Phase modulation (PM)

The types of modulation for *pulse carrier wave* are:

(i) Pulse amplitude modulation (PAM)
(ii) Pulse time modulation (PTM)
 (a) Pulse position modulation
 (b) Pulse with modulation (PWM) or pulse duration modulation (PDM)
(iii) Pulse code modulation (PCM)

However, PCM method is preferred for digital communication while other methods are suitable for analog communication.

Communication systems may also be *categorized* based on their *physical infrastructure* and the *specifications of the signals they transmit*. The physical infrastructure pertains to the type of the channel used and the hardware design of the transmitting and receiving equipment. The signal specifications signify the nature and type of the transmitted signal. The types of communication systems based on their infrastructure and signal specifications are discussed below.

Based on physical infrastructure: Based on the *physical structure*, there are two types of communication systems:

- *Line communication systems*
- *Radio communication systems*

There is a physical link, called the hardwire channel, between the transmitter and the receiver in line communication systems. In a radio communication system, there is no such link and natural resources, such as space and water are used as software channels. A particular communication system can be one of these two types. For example, radio broadcast is a purely radio communication system and cannot be categorized as a line communication system. On the other hand, landline telephony is purely a line communication system and cannot be typed as a radio communication system.

Consider a *TV system* in which a user can only receive the signals and view available channels. A television receiver cannot transmit the signals. Now consider telephony as another example. In this case, you can simultaneously send and receive signals. TV transmission is a one-way transmission, while telephony is a two-way transmission. In technical terms, one-way transmission is called simplex, while a two-way transmission is called duplex.

The one-way or two-way transmission feature of a communication system depends on the design of the equipment used on the two sides of the communication system, and is therefore included in the physical structure specifications of the system. As a rule, a communication system can be a *simplex* or a *duplex* system, but not both.

Therefore, based on the physical structure of a communication system, one can define two groups, and only one specification from each group is required to decide the type of communication system. These groups are:

- Line/radio communication
- Simplex/duplex communication

For instance, a TV communication system is a combination of the radio and simplex communication systems and landline telephony is a combination of the line and duplex communication systems. A particular communication system can be implemented as both line and radio communication system. For example, landline telephony is a line communication system. However, overseas or long-distance telephony is carried out through satellites and the system is called radio telephony as it makes use of radio waves for transmission and reception. This system is then categorized as a radio communication system.

Based on signal specifications: The signal specifications used to decide the type of communication system include:

- Nature of the baseband or information signal
- Nature of the transmitted signal

Based on the nature of the baseband signal, there are two types of communication systems:

- Analog communication systems
- Digital communication systems

Based on the nature of the transmitted signal, the baseband signal can either be transmitted as it is, without modulation, or through a carrier signal with modulation. The two systems can then be categorized as:

- *Baseband communication system*
- *Carrier communication system.*

Therefore, the four types of communication system categories based on signal specifications are:

- *Analog communication systems*
- *Digital communication systems*
- *Baseband communication systems*
- *Carrier communication systems*

Of the four, at least two types are required to specify a particular communication system. Thus, two groups are formed consisting each of two types such that at least one of the types from each group is necessarily required to specify a communication system. These groups can be formed as:

- Analog/digital communication systems
- Baseband/carrier communication systems

A particular communication system is either an analog communication system or a digital communication system at a time. For instance, *TV transmission* is an analog communication system while high definition television (HDTV) is a digital communication system. The internet is another example of a digital communication system.

Similarly, a particular communication system is either a baseband communication system or a carrier communication system. Examples of baseband communication systems are *landline telephony* and *fax*. Examples of carrier communication systems are TV transmission, radio broadcast, and cable TV.

Therefore, to completely describe a particular communication system, four of the eight types of communication systems are required. If any one type is missing, then the description of the communication system will not be complete. To make things simple and clear, *four groups containing these eight types are formed*, such that only one choice is possible from each group and the four choices together describe the communication system.

These four groups are:

- **Group I:** Line/radio communication system
- **Group II:** Simplex/duplex communication system
- **Group III:** Analog/digital communication system
- **Group IV:** Baseband/carrier communication system

Consider the television communication system. It is described as a radio communication system from group 1, simplex communication system from group II, analog communication system from group III, and carrier communication system from group IV. That is, *a television communication system is a radio-simplex-analog-carrier communication system.* This is the description of a television communication system in its totality.

Another example is landline telephony. This is described as line communication system from group I, duplex communication system from group II, analog communication system from group III, and baseband communication system from group IV. Therefore, telephony through landlines is a line-duplex-analog baseband communication system. Similarly, you can completely describe any communication system with these four groups.

ILLUSTRATIVE EXAMPLES

Example 1

A message signal of 12 kHz and peak voltage 20 V is used to modulate a carrier wave of frequency 12 MHz and peak voltage 30 V. Calculate the (i) modulation index, (ii) side band frequencies.

Solution

Here A_m = 20 V, A_c = 30 V, f_m = 12 kHz.

f_c = 12 MHz = 12000 kHz.

We have (i) modulation index

$$m = \frac{A_m}{A_c} = \frac{20}{30} = 0.67$$

(ii) $\text{USB} = f_c + f_m$
$= 12000 + 12$
$= 12012 \text{ kHz} = 12.012 \text{ MHz}$

$\text{LSB} = f_c - f_m$
$= 12000 - 12 = 11988 \text{ kHz}$
$= 11.099 \text{ MHz}$

Example 2

A 1.0 MHz carrier signal with amplitude 5 V is amplitude modulated by a 400 Hz sinusoidal modulating signal. The depth of modulation is 60%. Write the equation of this AM wave.

Solution

Here f_c = 1.0 MHz, f_m 400 Hz, A_c = 5 V,

$$m_a = 60\% \text{ or } 0.6$$

The equation of an AM wave is given as

$$C_m(t) = A_c\,(1 + m_a \sin \omega_m t) \sin \omega_c t$$

Where the carrier and modulating signals are given, respectively, as

$$c(t) = A_c \sin \omega_c t$$

$$m(t) = A_m \sin \omega_c t$$

The equation of the AM wave can be written in terms of f_c and f_m as:

$$A = A_c\,(1 + m_a \sin 2\pi \mathrm{f}_m t) \sin 2\pi \mathrm{f}_c t$$

Substituting the given values in the equation, one obtains:

$$c_m(t) = 5\,(1 + 0.6 \sin 2\pi \times 400\mathrm{t}) \sin 2\pi \times 10^6\, t$$

$$c_m(t) = 5\,(1 + 0.6 \sin 2513\mathrm{t}) \sin 6.283 \times 10^6\, t$$

This is required equation of the AM wave generated.

Example 3

A pure sinusoidal modulating signal of 15 kHz modulates a 100 MHz carrier signal with the maximum frequency deviation of 75 kHz. Calculate the maximum and minimum instantaneous frequencies of the FM signal.

Solution

Here f_m = 15 kHz, f_c = 100 MHz, and $f_{d(\max)}$ = ±75 kHz

The instantaneous frequency, f_i of the modulated carrier is given as:

$$f_i = f_c \pm f_d$$

In the above equation, the plus sign indicates the frequency deviation corresponding to the positive half of the modulating signal and the minus sign is used for the frequency deviation corresponding to the negative half of the modulating signal. The instantaneous frequency will be maximum when the frequency deviation is maximum, at positive peak amplitudes of the modulating signal. The minimum instantaneous frequency will occur when the frequency deviation is maximum, at the negative peak amplitudes of the modulating signal.

Thus:

$$f_{i(\max)} = f_c + f_{d(\max)}$$

$$f_{i(\max)} = (100 \times 10^6 + 75 \times 10^3)\ \text{Hz}$$

$$f_{i(\max)} = 100.075\ \text{MHz}$$

In addition,

$$f_{i(\min)} = f_c + f_{d(\max)}$$

$$f_{i(\min)} = (100 \times 10^6 + 75 \times 10^3)\ \text{Hz}$$

$$f_{i(\min)} = 99.925\ \text{MHz}$$

Therefore, the maximum instantaneous frequency of the given FM signal is 100.075 MHz and the minimum instantaneous frequency is 99.925 MHz. The instantaneous frequency of the FM signal lies in the range $fc \pm f_{i(\max)}$.

In this example, the instantaneous frequency lies between 99.925 MHz and 100.075 MHz.

Example 4

A sinusoidal carrier voltage of amplitude 80 V and frequency 800 kHz is amplitude modulated by a sinusoidal voltage of frequency 15 kHz resulting in minimum carrier amplitude of 72 V. Calculate (i) the sideband frequencies and (ii) modulation index.

Solution

Here: f_c = 800 kHz, f_m = 15 kHz

$$\therefore \quad \text{LSB} = f_c - f_m = 800 - 15 = 785 \text{ kHz}$$

$$\text{USB} = f_c + f_m = 800 + 15 = 815 \text{ kHz}$$

Example 5

A sinusoidal carrier voltage is amplitude modulated by a sinusoidal voltage of 10 kHz to a depth of 30%.

Calculate the frequency and amplitude of the two-sideband if the carrier frequency is 10 MHz and amplitude is 40 V.

Solution

Here f_m = 10 kHz = 0.010 MHz

$$\mu = 30\% = 0.30, f_c = 10 \text{ MHz}, A_c = 40 \text{ V}$$

Frequency of USB

$$= f_c + f_m = 10 + 0.010 = 10.01 \text{ MHz}$$

Frequency of LSB

$$= f_c - f_m = 10 - 0.010 = 9.99 \text{ MHz}$$

Amplitude of each sideband

$$= \frac{mA_c}{2} = \frac{0.30 \times 40}{2} = 6.0 \text{ V}$$

SUMMARY

1. **Transducer.** Transducer is a device which converts energy from one form to another. Transducer converts variations in physical quantity such as pressure, displacement, force, temperature, etc. into corresponding variations in the electrical signal at its output, e.g. a microphone converts a sound signal into an electrical signal.
2. **Signal.** Signal is the electrical analog of the information produced by the source. A signal is defined as any detectable transmitted energy that can be used to carry

information, i.e. a signal is a single valued function of time (that conveys the information) and which at every instant of time, takes a unique value. The communication signals based on the continuous nature of time and amplitude are divided into two types: *Analog signal* and *digital signal.*

3. **Noise.** Noise causes development of unwanted electrical signals which interfere with the information signal during its propagation through the transmission medium. As a result, a corrupted or distorted version of the transmitted signal reaches the receiver.
4. **Transmitter.** The transmitter is a set up that transmits the message to the receiving end through a communication channel. A transmitter makes the signal suitable for transmission. The communication channel introduces noise and degrades the signals. Signal attenuation, interference and other channel effects may cause this degradation. The receiver receives this degraded signal and recovers the original signal.
5. **Receiver.** Receiver is a device which receives the message signal sent by transmitter through communication channel and process it and finally converts into original message. The most important function of receiver is *demodulation* and sometimes *decoding* as well which is reverse of modulation in transmitter. The output of the receiver may be fed to loudspeaker, video displaying,teleprinter, TV picture-tube or computer. The transmitter and receiver must be in synchronization with the modulation and coding methods used. Figure 1.6b shows block diagram of receiver.
6. **Attenuation.** Attenuation is the loss of strength of a signal during its propagation through the transmission medium.
7. **Amplification.** Amplification is the process of increasing the amplitude and hence the strength of an electrical signal with the help of an electric circuit (consisting of atleast one transistor). Amplification is desired to compensate for the loss of strength of signal during the process of transmission. The additional energy required for the purpose is obtained from the DC source. Amplification is done where the strength of the signal gets weaker than the required level.
8. **Range.** Range is the largest distance between the source and the destination up to which a signal can be received with sufficient strength.
9. **Bandwidth.** The frequency range over which an equipment operates or the range over which the frequencies in a signal way is referred as bandwidth.
10. **Baseband.** The band of frequencies of the original signal (a signal not changed by modulation) as is produced by the source of information is referred baseband.
11. **Modulation.** It is defined as the process by which characteristic of a carrier is varied in accordance with a modulating wave. In this process some characteristic, usually an amplitude, frequency or phase angle of a high frequency carrier wave is varied in accordance with the instantaneous value of the low frequency information signal known as the modulated signal.
12. **Demodulation.** It is just the opposite of modulation. It recovers the original signal from the modulated signal at the receiving end. Obviously demodulation is the reverse process of modulation process that occurs at the transmitting end. Coherent demodulation requires the exact phase of the modulated waveform for demodulation.

13. **Repeater.** When a signal passes through the transmission medium may get attenuated due to the various energy losses along its path. This means a signal booster or amplifying repeater is required to be placed at suitable distance along its path. A *repeater* is combination of a transmitter, an amplifier and a receiver. This unit in the form of a repeater picks up a signal from the transmitter, amplifies and transmits it to the receiver sometimes with a change of carrier frequency.
    In order to extend the range of a communication system, multiple repeaters are placed suitable distances along the transmission path (Fig. 1.13). We may note a *communication satellite* is essentially a repeater station in a space.

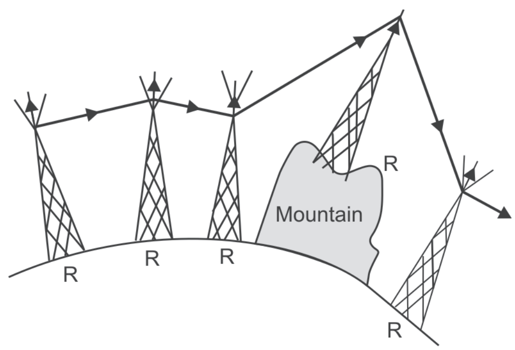

Fig. 1.13: Use of multiple amplifier repeaters extends the range of communication

REVIEW QUESTIONS

1. Draw a basic schematic diagram of a basic communication system and write the names of main constituents of a basic communication system.
2. Explain briefly the information source and input transducer of a communication system.
3. What are signals? How many types of signals are there? Briefly describe each of them.
4. Describe the working of transmitter in a communication system.
5. What is the role of channel or medium in a communication system? How many types of channels are there? Briefly exaplain them.
6. Identify the noise sources in a communication system. Define signal-to-noise ratio (SNR) and the noise figure of a system.
7. Explain the operation of the receiver in a communication system. Draw detailed block diagram of receiver section.
8. Identify the final destination in a communication system.
9. Describe the relationship between the information and bandwidth of a communication system.
10. Identify the high frequency effects on the components of a communication system.
11. What is modulation? What is the need of modulation?
12. Identify the use of modulation with respect to the antenna height in radio communication.

13. Identify the need for narrow banding in radio communication. Describe how narrow banding is achieved through modulation.
14. Identify how radiation and penetration in communication systems are improved with modulation.
15. Identify the need for modulation to provide a narrow diffraction angle of a radio beam in a communication system.
16. Explain the need for multiplexing in communication systems. Describe how multiplexing is possible with modulation.
17. Classify the communication systems based on their physical infrastructure.
18. Identify real-life communication systems based on their physical infrastructure.
19. Classify communication systems based on their signal specifications.
20. Identify real-life communication systems based on their signal specifications.

PROBLEMS

1. An AM signal is represented by the equation:

$$c_m(t) = 6.5\ (1 + 0.5 \sin 5027t) \cos 10053 \times 10^3\ t$$

Where t is expressed in seconds.

What information can be gathered from this equation:

2. Draw the frequency spectrum of the AM wave obtained in Q1 and evaluate the bandwidth of this AM wave.

Hint. The equation of the AM wave in Q1 is obtained as:

$C_m(t) = 5\ (1 + 0.6 \sin 2513t) \sin 6.283 \times 10^6 t$

The following amplitudes and frequencies of LSB and USB are obtained as:

- LSB frequency = $10^6 - 400$ = 999.6 kHz
- Amplitude = $\dfrac{m_a A_c}{2}$

$$= \frac{0.6 \times 5}{2} = 1.5 \text{ V}$$

- USB frequency

$$= 10^6 + 400$$

$$= 1000.4 \text{ kHz}$$

Amplitude = $\dfrac{0.6 \times 6}{2} = 1.5$ V

- The frequency of the carrier signal = 10^6 Hz = 1 MHz
- The amplitude of the carrier signal = 5 V

The bandwidth of the AM = $2f_m$ = 2 × 400 Hz = 800 Hz

Figure 1.14 shows the frequency spectrum of the AM wave obtained in Example 4

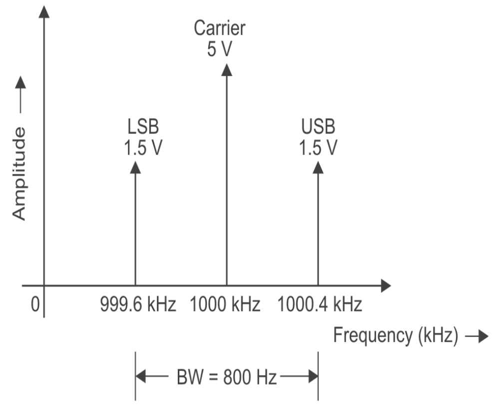

Fig. 1.14: Frequency spectrum of AM wave obtained in Q1

3. An audio signal of 20 kHz causes the frequency of a 10 MHz carrier signal to deviate by 4.2 kHz when the instantaneous amplitude is 1.4 V. At some instant, the audio voltage is increased to 15 V. Estimate the frequency deviation at this instant.

   Hint. Here f_m = 10 kHz, f_c = 10 MHz,

   A_m = 1.4 V, and f_d = 4.2 kHz

   In FM system, the frequency deviation is directly proportional to the instantaneous amplitude (voltage) of the modulating signal. The frequency deviation will be almost always linear because of the direct proportionality. Thus, if the frequency deviation for one volt of modulating signal is estimated, then the deviation for any modulating voltage can be calculated

$$\frac{f_d}{A_m} = \frac{4.2\text{ kHz}}{1.4\text{ V}} = \frac{3\text{ kHz}}{\text{V}}$$

   In the given FM system, the frequency deviates by 3 kHz per volt of the modulating signal. Thus, the frequency deviation at 15 V is estimated as:

$$(f_d)_{15\text{ V}} = \left(\frac{3\text{ kHz}}{\text{V}}\right) \times 15\text{ V}$$

$$(f_d)_{15\text{ V}} = 45\text{ kHz}$$

   Therefore, the carrier frequency deviates from its initial value, f_c by 45 kHz when the modulating signal voltage is 15 V.

4. The maximum peak-to-peak voltage of an AM wave is 16 mV and minimum peak-to-peak voltage is 8 mV. Calculate the modulating factor.

   Hint. Figure 1.15 depicts the situation

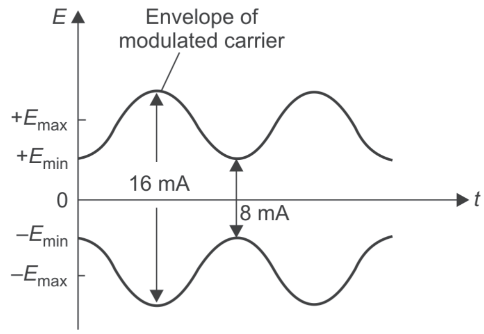

Fig. 1.15

From Fig. 1.15, we have

$$E_{max} = \frac{16}{2} = 8 \text{ mV}$$

$$E_{max} = \frac{8}{2} = 4 \text{ mV}$$

$$\text{Now } E_{max} = m = \frac{E_{max} - E_{min}}{E_{max} + E_{min}} = \frac{8-4}{8+4} = \frac{4}{12} = \frac{1}{3} = 0.33$$

$$\therefore \text{ Modulation index } m_f = \frac{A_m}{A_c} = \frac{A_c - A_{min}}{A_c} = \frac{80-72}{80} = 0.1$$

5. A sinusoidal carrier voltage of frequency 1200 kHz is amplitude modulated by a sinusoidal voltage of frequency 20 kHz resulting in maximum and minimum modulated carrier amplitudes of 11 OV and 90 V respectively. Calculate (i) the frequency of lower and upper sidebands, (ii) the unmodulated carrier amplitude, (iii) the modulation, and (iv) the amplitude of each sideband.

Hint. We have (i) Lower sideband frequency

$$= f_c - f_m = 1200 - 20 = 1180 \text{ kHz}$$

Upper sideband frequency

$$= f_c + f_m = 1200 + 20 = 1220 \text{ kHz}$$

(ii) Unmodulated carrier amplitude

$$= \frac{A_{max} + A_{min}}{2} = \frac{110 + 90}{2} = 100 \text{ V}$$

(iii) Modulation index

$$m = \frac{A_{max} - A_{min}}{A_{max} + A_{min}} = \frac{110 - 90}{110 + 90} = 0.1$$

(iv) Amplitude of each sideband

$$= \frac{mA_c}{2} = \frac{0.1 \times 100}{2} = 5\text{ V}$$

SHORT ANSWER QUESTIONS

1. What are the three basic units of a communication system?
   Ans. (i) Transmitter, (ii) communication channel, and (iii) receiver
2. What is communication?
   Ans. Communication is the process by which the information is transferred faithfully by using a suitable communication system and the transmission medium.
3. What is a digital signal?
   Ans. A digital signal is a signal in which current or voltage can take only one of two discrete values at a time.
4. What is an analog signal?
   Ans. Analog signal is a signal in which current or voltage varies continuously with time.
5. What do you understand by digital communication system?
   Ans. This communication system makes use of digital signals, e.g. fax
6. What do you understand by analog communication system?
   Ans. This communication system makes use of analog signals.
7. What is a byte?
   Ans. A byte is string of 8 bits.
8. What is a nibble? What is the relation between nibble and byte?
   Ans. A group of 4 bits is called a nibble. 1 nibble = ½ byte
9. What do you understand by a modulated wave?
   Ans. A modulated wave is the resultant wave obtained by the superposition of the modulating wave and the carrier wave.
10. What do you understand by a modulating signal?
    Ans. Modulating signal is a low frequency information signal.
11. Write the range of audio frequency signals.
    Ans. 20 to 20,000 Hz
12. What do you understand by amplitude modulation?
    Ans. The process in which the amplitude of the high frequency carrier wave changes in accordance with the instantaneous value of the modulating signal is called amplitude modulation.
13. What percentage of AM wave power is carried by sidebands for $m = 1$
    Ans. 33.3%
14. Name the type of modulation scheme preferred for digital communication.
    Ans. Pulse code modulation.
15. Name the device which can represent digital data by analog signals and vice versa.
    Ans. Modem.
16. What is a transducer ?
    Ans. This is a device which converts energy from one form to another, e.g. a microphone converts a sound signal into an electrical signal.

17. What do you understand by pulse modulation?
    Ans. Pulse modulation is the process in which some parameter of a pulse train is varied in accordance with the message signal or the modulated signal.
18. How one can decrease noise in FM receivers?
    Ans. By decreasing the frequency deviation, noise in FM receivers can be decreased.
19. Name the process by which exact reproduction of a document at a distant place can be received?
    Ans. Facsimile telegraphy (FAX).
20. Name an appropriate communication channel needed to send a signal of band width 100 kHz over a distance of 8 km.
    Ans. Twisted pair wire.
21. What do you understand by (channel) noise?
    Ans. In a transmitted wave, the undesired electrical signals present, constitute the noise.
22. How does the effective power radiated by an antenna vary with wavelength?
    Ans. Power radiated by antenna, $P \alpha (1/\lambda)^2$, i.e. power radiated from antenna is inversely proportional to the square of wavelength.
23. What do you understand by sampling of an analog signal?
    Ans. The process of generating narrow pulses having amplitudes proportional to the instantaneous amplitude of the analog signal is called sampling.
24. What do you understand by quantization of an analog signal?
    Ans. This is the process of dividing the maximum value of the analog voltage signal into a fixed number of levels in order to convert the pulse amplitude modulated signal into a binary code.
25. What is the main function of a modem?
    Ans. At the transmitting end a modem modulates the signal and demodulates it at the receiving end.
26. What type of modulation is required for television broadcast?
    Ans. We require for video signal to AM and for voice signal to FM.
27. What do you understand by data retrieval?
    Ans. This is the process of recovering the original data from the modulated data.
28. What type of modulation is required for commercial broadcast of voice signals?
    Ans. Frequency modulation is preferred.
29. What do you understand by bandwidth?
    Ans. The frequency range in which a transmitting system makes transmission is called the bandwidth.

MULTIPLE CHOICE QUESTIONS

1. A 2 kW wave is to be modulated to a 90% level. The total transmitted power would be
   (a) 3.82 kW (b) 2.81 kW
   (c) 1.81 kW (d) 4.81 kW

   Hint. $p_t = p_c \left(1 + \frac{ma^2}{2}\right) = 2000\left(1 + \frac{0.9^2}{2}\right) = 2810 \text{ W} = 2.810 \text{ kW}$

2. If the modulation index of an AM wave is changed from 0 to 0.5, the transmitted power increases by
(a) 25% (b) 50%
(c) 12.5% (d) 75%

Hint. When modulation index is 0, then

$$P_1 = \frac{A_c^2}{2}\left(1 + \frac{0^2}{2}\right) = \frac{A_c^2}{2} = 0.5A_c^2$$

When modulation index is 0.5, then

$$P_2 = \frac{A_c^2}{2}\left(1 + \frac{0.5^2}{2}\right) = \frac{1.25}{2} = A_c^2$$

$$\frac{P_2}{P_1} = \frac{0.5625\, A_c^2}{0.5A_c^2}$$

$\Rightarrow P_2 = 1.125\ P_1$. Thus, P_2 increases by 12.5%

3. If the modulation index of an AM wave is changed from 0 to 1, the transmitting power increases by
(a) 75% (b) 60%
(c) 90% (d) 50%

Hint. $P_t = P_c\left(1 + \frac{m_a^2}{2}\right)$

When modulation index is 0, then

$$P_1 = \frac{A_c^2}{2}\left(1 + \frac{0^2}{2}\right) = A_c^2/2$$

When modulation index is 1, then

$$P_2 = \frac{A_c^2}{2}\left(1 + \frac{1^2}{2}\right) = \frac{3}{4} - A_c^2$$

$$\Rightarrow \frac{P_{t2}}{P_{t1}} = \frac{3}{4}$$

Thus, $P_2 = 1.5\ P$, and P_2 increases by 50%

4. For an AM signal, the bandwidth is 10 kWz and the highest frequency component present is 705 kHz. The carrier frequency used for this AM signal is
   (a) 750 kHz (b) 705 kHz
   (c) 700 kHz (d) 695 kHz

   Hint. Bandwidth of an AM signal

$$BW = 2 f_m = 10 \text{ kHz}$$
$$\Rightarrow f_m = 5 \text{ kHz}$$
$$f_c + f_m = 705 \text{ kHz}$$
$$\Rightarrow f_c = 705 - 5 = 700 \text{ kHz}$$

5. A 60 kHz carrier is amplitude modulated by the speech band of 300 to 3000 Hz. The range of upper side band will be
   (a) 57 to 59.7 kHz (b) 60 to 59.7 kHz
   (c) 60.3 to 63 kHz (d) 58 to 59.3 kHz

   Hint. Upper side band = 60 + 0.3 to 60 + 3
   = 60.3 to 63 kHz

ANSWERS

1. (b) 2. (c) 3. (d) 4. (c) 5. (c)