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INTRODUCTION TO COMMUNICATION SYSTEMS

1.1. INTRODUCTION

The word communicate is illustrated in the English dictionary as an act of passing on news, information, feelings, heat, motion, an illness etc. In its electrical sense for an electrical engineer, the term *communications* refers to the sending, reception and processing of information by electrical means. This can be achieved through wire telegraphy, telephony or by radio.

Modern communication system equipment has been highly simplified and made compact with the use of transistors, integrated circuits etc. The new equipment based on solid state technology is highly reliable as compared to the old tube versions. Communication systems now-a-days have a large variety *e.g.* telegraphy, telephony, broadcasting, radar, radiotelemetry, radio navigation, computer communications and point to point communication systems such as microwave links, satellite communication etc. All the aspects of above-mentioned communication systems will be discussed in this book to the point a communication engineer should know.

1.2. COMMUNICATION SYSTEMS

There are some common basic terms and building blocks in a communication system which are important to be discussed before describing the individual systems.

1.2.1. Information. Information theory as a separate section is discussed in Chapter 2. Basically the function of a communication system is to pass on a message. The origin of the message is from some information source as shown in Fig. 1.1.

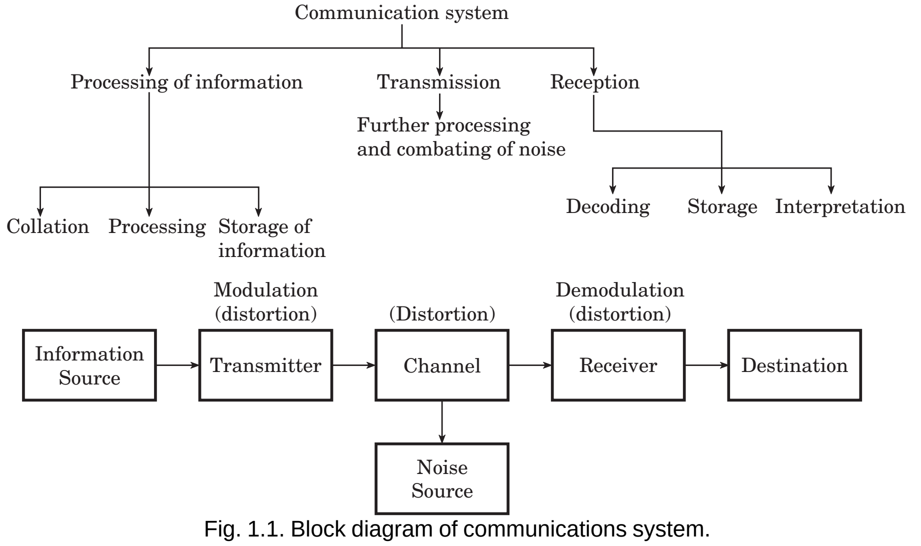

Fig. 1.1. Block diagram of communications system.

The message may be in form of words, code symbols, telephonic message, picture etc. The amount of information conveyed by a message depends upon the predictability of that message. More information is conveyed if the message is less predictable. No information is conveyed if the message is less predictable. No information is conveyed if the message is redundant. The amount of information conveyed in any message is measured in bits.

1.2.2. Transmitter. The message to be communicated has to be first converted to an electrical signal by the help of a suitable trans-ducer. The electrical signal so obtained has to be suitably processed and amplified before being sent on the channel. A basic transmitter block schematic is shown in Fig. 1.2.

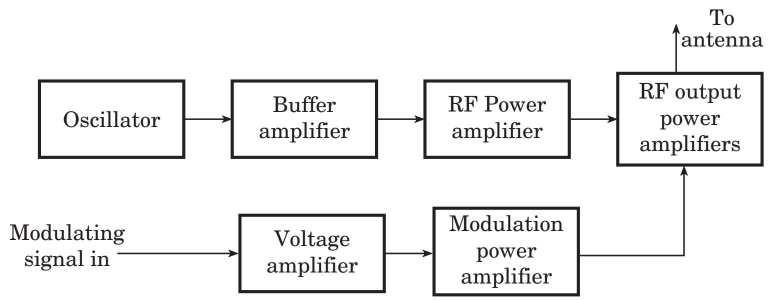

Fig. 1.2. Block schematic of a basic transmitter.

1.2.3. Channel. *Channel* here refers to the medium over which the transmitted signal travels. It can be wires, open space, coaxial cable or any other medium. Some unwanted energy termed *noise* is added to the signal at the transmitter, receiver as well as the channel stage. This energy is purely random and cannot be predicted. Since noise will be received along with the signal at the receiver, it obviously places limitation on the transmission system. Noise can interfere with the signal at any point in the communication system. To reduce the effect of noise, the signal strength should be kept high, otherwise noise will completely mark the signal and the signal will be rendered unintelligible.

1.2.4. Receiver. Fig. 1.3 shows a block schematic of an AM superhet receiver. There are a wide range of communications receivers available, each being for specific purpose governed by specific factors such as frequency of operation, bandwidth, Noise figure, output level, type of modulation etc. The destination of the received signal also governs the type of receiver. The output of the receiver may be used to drive a loudspeaker, a television picture tube, teletypewriter, magnetic recording discs, radar displays etc. Each separate use requires separate receiver design.

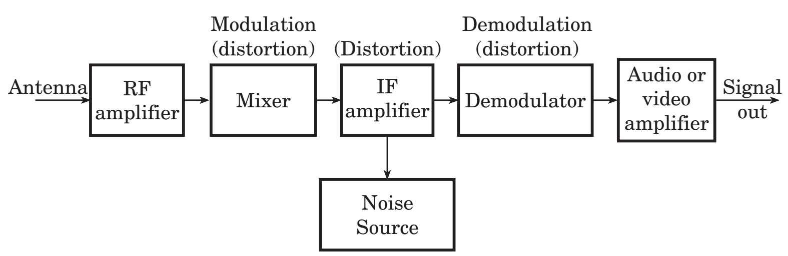

Fig. 1.3. Block schematic of AM superheterodyne receiver.

1.3. AMPLIFIERS

An amplifier is one of the major blocks of transmitters and receivers. Design of amplifier used is again governed by a number of factors which includes frequency, bandwidth, noise allowed, type of transmitter or receiver etc. Amplifier design is one of the main subsection of communication engineering. This subsection is not being discussed in detail in this book as it falls out of the scope of this book.

1.4. MODULATION

This is a very important operation for transmission over long distances. In this process of modulation, either of any three factors, *i.e.* amplitude, phase or frequency of a high frequency since wave is varied in accordance with the instantaneous value of the modulating signal. This gives rise to amplitude, phase or frequency modulation respectively.

To execute the process of modulation is quite complicated and needs a well designed electronic circuitry.

1.5. NEED FOR MODULATION

Why at all do we need modulation for transmitting signal over long distances? Can we transmit unmodulated carrier or the modulating signal as it is over long distances? These are some of the questions which a beginner in communication engineering comes cross.

The impossibility of transmitting the modulating signal is explained here. There are a number of difficulties in transmitting the modulating signal itself but only one problem is illustrated here. Consider an audio frequency signal of say fran 0.20 KHz. For efficient transmission and reception of electromagnetic signals, the transmitting and receiving antennas should atleast have a height comparable to a quarter wavelength of frequency used. Thus for 20 KHz, quarter wavelength approximates 3750 metres. A vertical antenna of this size is physically not possible to realise.

Another argument in favour of modulation is that if transmission is carried at audio frequencies, then all the signals will get mixed up in the air with one another and it will not be possible to separate them. Thus in order to simultaneously transmit a number of signals, these signals are transformed into different portions of the electromagnetic spectrum. Once signals are translated to a particular frequency, tuned circuits can be used at the receiver to receive a particular signal and reject the other signals. Tuning of receivers is made variable so that a number of frequencies can be tuned by the same receiver set.

An unmodulated carrier cannot be used to transmit intelligence or information as it has a fixed amplitude, frequency and phase and these three variables have no relation with the modulating signal here. For information transmission, any one of the three variables have to be deviated from its unmodulated value and the rate of deviation is the same as the modulating signal.

1.6. CONTAMINATIONS

The channel attenuates the signal and distorts its waveform. The length of the channel increases attenuation, varying from a few percent for short distances to much higher orders of magnitude for interplanetary communication. The distortion of the waveform occurs because of different amounts of attenuation and phase shift suffered by different frequency components of the signal. For example, a square pulse is rounded or "spread out" by the process. This type of distortion is called ***linear distortion.*** It can be partly corrected at the receiver by an equalizer with gain and phase characteristics complementary to those of the channel.

Non-linear distortion may also occur through attenuation that varies with the signal amplitude. Such distortion can also be partly corrected by a complementary equalizer at the receiver.

1.7. NOISE

The signal is also contaminated along the path by undesirable signals lumped under the broad term **noise** which are random and unpredictable signals. Its causes are either external or internal:

1. **External noise.** It includes interference from signals transmitted on nearby channels, man-made noise generated by faulty contact switches for electrical equipment, by automobile ignition radition, fluorescent lights, and natural noise from lightning, electrical storms, solar and intergalactic radiation. With proper care, external noise can be minimized or even eliminated.
2. **Internal noise.** It results from thermal motion of electrons in conductors, random emission, and diffusion or recombination of charged carriers in electronic devices. Proper

care can reduce the effect of internal noise but can never eliminate it. Noise is one of the basic factors that sets a limit on the rate of communication.

1.8. THE AUDIO SPECTRUM

The human ear can detect sounds over a range of frequencies; *i.e.*, it can hear sounds of different pitch. A sensitive ear can hear sounds of frequencies ranging from about 30 Hz up to 20,000 Hz, though most people have a range somewhat less than this.

Sound of a given frequency means that the air is vibrating with that number of oscillations per second. In order to transmit this sound, the microphone of a telephone converts the sound into an equivalent number of electrical oscillations per second. The telephone channels over which we wish to send data are, then, designed to transmit electrical oscillations of a range equivalent to the frequencies of the human voice, although these frequencies are often changed for transmission purposes.

In telephones, in fact, the whole range of the human voice is not transmitted. It is found that this was unnecessary for the understanding of the speech and the recognition of the speaker. Fig. 1.4 illustrates the characteristics of human speech.

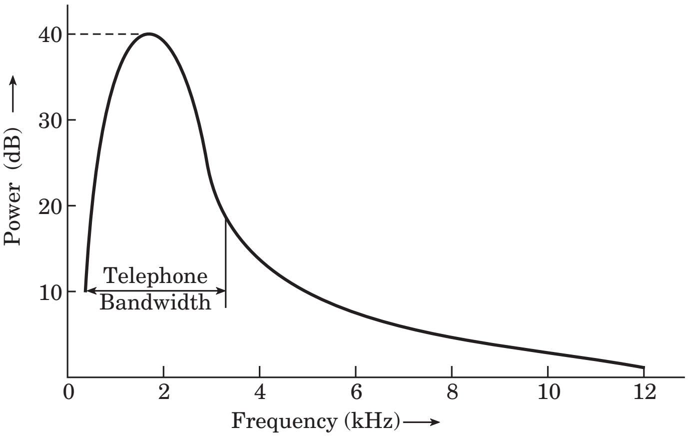

Fig. 1.4. Human speech spectrum.

1.9 SIGNAL POWER UNITS

1. **Decibel.** It is a unit of power ratio and, therefore, not an absolute unit. It is defined as ten times the logarithm of power ratio (to base 10) as

$$\text{Power in decibels} = 10 \log_{10} \frac{P_1}{P_2} \text{ dB}$$

The unit is also used to give amplitude ratios. Since power is proportional to square of the amplitude, the amplitude ratio in decibels is defined as

$$\text{Amplitude ratio} = 20 \log_{10} \frac{A_1}{A_2} \text{ dB}$$

where A_1 and A_2 are two amplitude levels.

Quantities like amplifier gain, noise levels, losses in transmission lines, signal-to-noise ratios and differences in sound intensity are expressed in decibels.

Sound levels are expressed in the logarithmic unit such as decibels because the response of the human ear is proportional to the logarithm of sound energy. For example, if to human ear a noise sounds twice as great as another, it is not twice the power, but it is nearly 2 dB greater than the other.

The advantage of decibels is that the power losses and gains can be arithmatically added or subtracted.

Example 1.1. *A signal is transmitted over a transmission line which reduces it in power in a ratio of 20:1. It then passes over another section of the line which reduces it in a ratio of 7:1. Calculate the net reduction in power.*

Solution.

For first line section, reduction $R_1 = 10 \log_{10} 20$

$= 13.01$ dB

For second line section, reduction $R_2 = 10 \log_{10} 7$

$= 8.45$ dB

Net reduction $R = R_1 + R_2 = 13.01 + 8.45$

$= 21.46$ dB

To obtain absolute value of the reduction R, we take antilogaritm as

$$10 \log_{10} R = 21.46$$

$$R = \text{antilog} \frac{21.46}{10} = 140$$

This value is correct because $\frac{20}{1} \times \frac{7}{1} = 140$

2. **Bel.** It is an earlier unit of power ratio and equal to a power of 10 to 1. Since it is too large for most purposes, the decibel which is one-tenth of the bell is mostly used.

3. **Neper.** This unit expresses power ratio as a logarithm to the base e. For amplitude ratios it is defined as

Amplitude ratio in nepers $= \log_e \frac{A_1}{A_2}$ nepers

Therefore power ratio is expressed as

Power ratio in nepers $= \frac{1}{2} \log_e \frac{P_1}{P_2}$ nepers.

Since $\log_e x = (\log_{10} x) (\log_e 10)$,

Power ratio in nepers
$$= \frac{1}{2} . \log_{10} . \frac{P_1}{P_2} . \log_e 10$$
$$= \frac{1}{20} \log_e 10 . \left(10 \log_{10} \frac{P_1}{P_2}\right)$$
$$= 8.686 \times \text{Power ratio in dB}$$

Therefore, **1 neper = 8.786 dB.**

4. **One Milliwatt Decibel (dBm).** This unit represents absolute power with reference to one milliwatt level and defined as

$$\text{dBm} = 10 \log \frac{\text{Power}}{1 \text{ mW}}$$

5. **One Watt Decibel (dBW).** This unit represents absolute power with reference to one watt level and defined as

$$\text{dBm} = 10 \log \frac{\text{Power}}{1 \text{ W}}$$

Two signals represented in dBm or dBW cannot be added/subtracted in dBm or dBW. They should be converted back to mW or W and then added/substracted.

Example 1.2. *Calculate: (a) dBm* for 1 *mW* signal and 2 *mW* signal. (*b*) dBw for 1 mW signal and 1 W signal. (*c*) Sum of 13 dBm and 10 dBm.

Solution.

(*a*) $$(1 \text{ mW}) \text{ in dBm} = 10 \log \frac{1 \text{ mW}}{1 \text{ mW}} = 0 \text{ dBm}$$

$$(2 \text{ mW}) \text{ in dBm} = 10 \log \frac{2}{1} = 3 \text{ dBm}$$

(*b*) $$(1 \text{ mW}) \text{ in dBW} = 10 \log \frac{1 \text{ mW}}{1 \text{ W}} = 10 \log 10^{-3} = -30 \text{ dBW}$$

$$(1 \text{ W}) \text{ in dBW} = 10 \log \frac{1W}{1W} = 0 \text{ dBW}$$

(*c*) $$13 \text{ dBm} \equiv 10 \log \frac{1 \text{ mW}}{1 \text{ mW}}$$

or $$\text{mW} = \text{antilog} \frac{13}{10} = 20 \text{ mW}$$

Similarly, $$10 \text{ dBm} \equiv 10 \text{ mW}$$

Therefore, sum $$= 20 + 10 = 30 \text{ mW.} = 10 \log 30 = 14.8 \text{ dBm}$$

6. **Zero Transmission Level Point (OTLP) and dBm$_0$.** In telephones, it is convenient to define some point as the **zero transmission level** and to measure the signal and noise levels at other points in the system relative to it. If in this instance a signal has experienced a 12 dB loss relative to the toll exchange, it will be at the – 12 dB level point or at a relative level of – 12 dB.

When referring signals back to the zero transmission level point, the units are expressed in **dBmo.** The dBmo unit indicates what the power would have at been at the zero transmission point.

When applying a test tone to a system, the tone level at the OTLP is set at dBm = 0.

If the signal power at a relative level point *R* is *dBr*, then:

dBm = dBm$_0$ + dBr

The term *dBmo* refers to noise power at OTLP.

Example 1.3. *Determine the signal power at the zero transmission level point if the power is 11 dBm at a point with a relative level of – 12 dBr.*

Solution. We have $$\text{dBm} = 11, \text{dBr} = -12$$

Therefore, $$\text{dBm}_0 = \text{dBm} - \text{dBr}$$

$$= 11 - (12) = 23.$$

Example 1.4. *The power of a signal at the OTLP is 6dBm$_0$. What is the power in dBm if its value at a relative level is 7 dBr? Also calculate power in mW.*

Solution.

$$\text{dBm} = \text{dBm}_0 + \text{dBr} = 21$$

Now $$\text{dBm} = 10 \log_{10} \frac{\text{Power}}{1 \text{ mW}}$$

$$21 = 10 \log_{10} \frac{\text{Power}}{1 \text{ mW}}$$

or $$\text{Power} = \text{antilog } 2.1 \text{ mW} = 125 \text{ mW}$$

1.10. VOLUME UNIT

It is a measure of the power level, or volume, of broadcast programmes and certain types of speech and music. If a simple decibel meter is used to monitor the programme power level, the indicating needle will attempt to follow every fluctuation of the power level and will be difficult to read. To overcome this difficulty, a dc millimeter with a slow response time is used to measure *VU's*. When a steady sine wave is suddenly applied to such a *VU* meter, the indicating needle will move to within 90% of the steady state value of 0.3 sec and overswing the steady state value by less than 1.5%.

A standard volume-indicator is calibrated to read *OVU* when connected to a 600-ohm circuit carrying 1 mW of sine wave power within the 35 to 10 KHz band. It will not, however, indicate the true power of complex waves, but read some value between the average and peak value.

The *VU* indicators are often used to monitor speech and music levels. Otherwise, excessive levels can cause distortion in loudspeakers, handsets, and tape recorders, and also overmodulation in ratio transmitters, or intermodulation in amplifiers.

1.11. SIGNAL-TO-NOISE RATIO

The signal-to-noise ratio is defined as the ratio of the signal power to the noise power. The channel distorts the signal, and noise accumulates along the path.

In addition, the signal strength decreases while the noise level increases with distance from the transmitter. Thus, the SNR decreases along the channel.

As typical noise variations are quite small, of the order of microvolts. The *S/N* ratios are large enough in ordinary systems under ordinary conditions and therefore, noise can go unnoticed. But in long-range or minimum power systems, the received signal may be as small as the noise or even less than the noise itself. For voice communication, as *S/N* ratio of 30 dB is considered satisfactory. For video, the minimum level should be 45 dB.

If signal strength is insufficient, increasing more stages of amplification at the receiver does not help because the noise also gets equally amplified along with noise, leaving the S / N ratio unimproved. Increasing the transmitted power can increase the ratio but cannot increased indefinitely because of technical limitations. Alternatively, we can exchange bandwidth with *S/N* ratio via modulator and coding techniques.

The second role of the signal power is not as obvious, although it is very important. We shall demonstrate in Section 1.14 that *the channel bandwidth and signal power are exchangeable.* That is, to maintain a given rate and accuracy of information transmission, we can trade signal power for B and vice versa. Thus, one may reduce bandwidth if one is willing to increase signal power or one may reduce signal power if one is willing to increase B.

1.12. ANALOG AND DIGITAL SIGNALS

A signal or a **baseband signal** is the electrical waveform representing the message which appears at the output of a transducer and becomes input to a communication system.

There are essentially two types of signals:

(*i*) *Continuous or Analog, and* (*ii*) *Discrete or Digital.*

Continuous or analog signal v(t) is obtained when the electrical waveform is a replica or analogous to the input waveform. It can be represented or specified in relation to two relatively single quantities:

1. The range of values within which $v(t)$ occurs, *i.e.* difference between its maximum and minimum values. The range can readily be changed using an amplifier or an alternator.
2. The time and frequency relation of the waveform, *i.e.*, how rapidly the signal changes with time. In fact, the behaviour of a communication system is specified in terms of its frequency response.

Analog messages are characterized by data whose value varies over a continuous range. For example, the temperature or the atmospheric pressure of a certain location can vary over a continuous range and can assume infinite possible values. Similarly, a speech waveform has amplitudes that vary over a continuous in contrast to only a finite number of possible digital messages.

The basis principle of transmission of digital message is transmitting a finite set of electrical waveforms by which the message in the form of letters or numbers each of which is defined by a signal level. For example, in the Morse code, a mark can be transmitted by an electrical pulse of amplitude A/2, and a space can be transmitted by a pulse of amplitude –A/2. In an μ-ary case, μ distinct electrical pulses (or waveforms) are used ; each of the pulses represents one of the μ possible symbols. The task of the receiver is to extract a message from a distorted and noisy signal at a channel output.

Message extraction is often easier from digital signals than from analog signals. Consider a binary case: Two symbols are encoded as rectangular pulses of amplitudes $A/2$ and $-A/2$. *The only decision at the receiver is the selection between two possible pulses received, not the details of the pulses shape ; the decision is readily made with reasonable certainity even if the pulses are distorted and noisy* (see Fig. 1.5). *Hence a digital communication system can transmit messages with greater accuracy than an analog system in the presence of distortion and noise.*

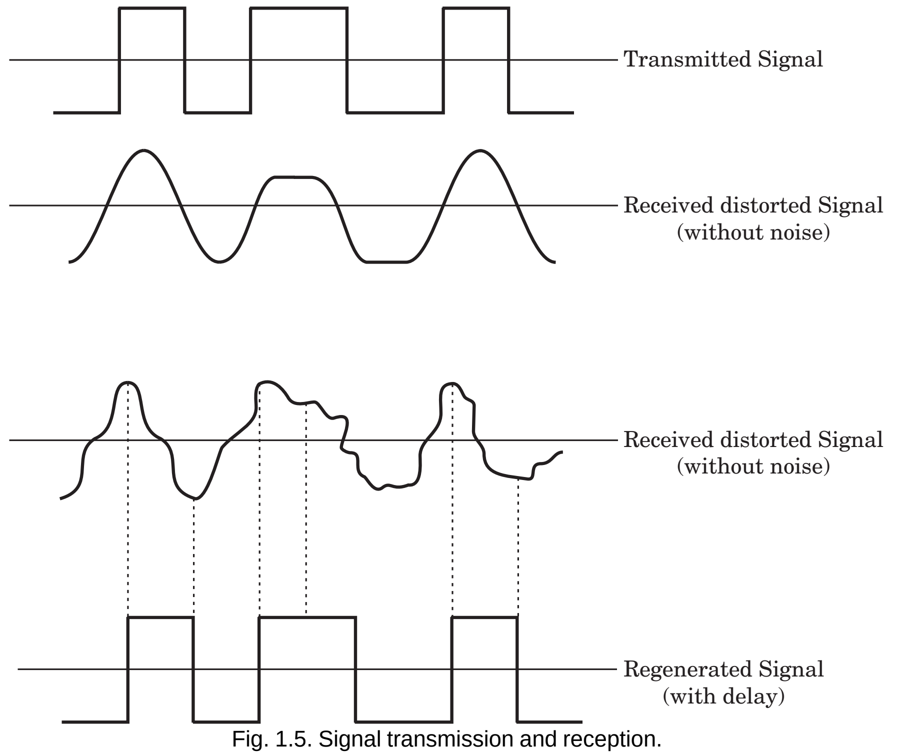

Fig. 1.5. Signal transmission and reception.

Another significant advantage of digital communication is the possibility of using ***regenerative repeaters*** in which a repeater station detects pulses and transmits new clean pulses, thus combating accumulation of distortion and noise and enabling information transmission over longer distances with greater accuracy.

The analog message, however, *demands accurate reproduction of waveform.* Even a light distortion or interference in the waveform will cause an error in the received signal. A further difficulty as we shall see later that *a regenerative repeater is not viable for analog signals because the noise and distortion, no matter how small, cannot be cleaned up from a signal.*

Therefore, in analog communications, *the distortion and the noise interference are cumulative over the entire transmission path.* In addition, the signal is attenuated continuously over the transmission path; thus with increasing distance, the signal becomes weaker, whereas the distortion and noise become stronger. Ultimately the signal, overwhelmed by the distortion and the noise, is destroyed. Any amplication of the signal is of no avail as the noise is also amplified in the same proportion. Consequently, *the distance over which an analog message can be transmitted is limited by the transmitter power.* The analog systems have now been replaced with digital systems as the latter has become more economical because of a dramatic cost reduction achieved in the fabrication of digital circuitry.

1.13. BASEBAND AND CARRIER COMMUNICATION

Baseband communication corresponds to transmission of the band of frequencies of the signal delivered by the source or the input transducer. In telephony, the baseband in the audio band (band to voice signals) of 0 to 3.4 kHz. In television, the baseband is the video band occupying 0 to 4.3 MHz. For digital data, using bipolar signaling at a rate of f_0 pulses/second, the baseband is 0 to f_0 Hz.

In baseband communication, baseband signals are transmitted without modulation, that is, without any shift in the range of frequencies of the signal. Because the baseband signals have sizeable power at low frequencied, they cannot be transmitted over a radio link but are suitable for transmission over a pair of wires or coaxial cables. Local telephone communication and short-haul pulse modulation systems (between two exchanges) use baseband communication. Because baseband communication uses only baseband frequencies, its uses are rather restricted.

Since the transmission of signals at lower frequencies is in general more difficult, it is desirable to shift the signal spectrum to a higher-frequency range by modulation. It has many advantages:

1. The vast spectrum of frequencies available because of technological advances cannot be utilized by a baseband scheme. By modulation several baseband signals and shifting their spectra to non-overlapping bands, once can use all the available bandwidth more efficiently.
2. Long-haul communication over a radio link also requires modulation to shift the signal spectrum to higher frequencies to enable efficient power radiation using antennas of reasonable dimensions.
3. It facilitates exchange of transmission bandwidth for the signal-to-noise ratio.

Communication that uses modulation to shift the frequency spectrum of a signal is known as **carrier communication.** In this mode, one of the basic parameters (amplitude, frequency, or phase) of a sinusoidal carrier of high frequency w_c is varied so that any instant its deviation from unmodulated value is proportion to the baseband or modulating signal $v_m(t)$. This results is **amplitude modulation (AM), frequency modulation (FM) or phase modulation (PM),** respectively. The latter two types of modulation are similar, in essence, and are grouped under the name **angle modulation.** Modulation is used to transmit analog as well as digital baseband signals.

1.13.1. Types of Modulation. The performance of communication system is largely governed by the type of modulation and its uses. In a system's design, many different modulation techniques can be employed to meet various communication requirements.

The two basic types of modulation are according to the kind of carrier wave used:

1. Continuous wave (C-W) modulation. In this case, the carrier is a sinusoidal wave. It is a continuous process, and, therefore, suited to signals that are continuously varying with time. The sinusoidal carrier is at a frequency much higher than any of the frequency components contained in the message or the modulating signal. The modulation process is characterized by ***frequency translation,*** which means the message spectrum, *i.e.*, its frequency content is shifted upward to a new and higher band of frequencies.

2. Pulse Modulation. Here, the carrier is a periodic train of pulses. It is a discontinuous, discrete process. That is, the pulses are present only at certain distinct intervals of time. Hence, it is most suited to messages that are discrete in nature. However, with the help of ***sampling techniques,*** continuously varying signals can be transmitted on pulsed carrier. Generally, pulse modulation and coding go hand in hand as in several modern communication systems.

In many complex systems, both CW as well as pulse modulation may be used together. Therefore, the type of modulation can be based on being analog or coded (digital):

1. Analog Modulation. Here, the modulation parameter varies in direct proportion to the modulating signal discussed already.

2. Coded or Digital Modulation. Here, a digital transformation takes place whereby the message is converted from one symbolic language to another. If the message is originally a continuous time function, it must be sampled and quantized prior to encoding.

It must be noted that both types –CW or pulse, analog or digital–must be *reversible process.* That is message can be recovered at the receiver by the process of demodulation.

Here it is important to note that pulse-modulated signals (PAM, PWM, PPM, PCM and DM) are, despite the term modulation, are baseband signals. The baseband coding schemes are actually coding schemes for baseband transmission. These signals must still modulate a carrier in order to shift their spectra.

1.14. THE BANDWIDTH CONSTRAINT

Bandwidth and S/N Ratio Relationship. In most of information transmission, a message must be sent in real time, the output signal keeping pace with the input as it comes from the source. Clearly, the rate at which the source is producing the message and hence the ***signaling speed*** governs the design of the system. Even if the source rate is adjustable, it is necessary to ensure efficient system utilization, *i.e.,* to minimize the time during which the information is sent.

It is obvious that rapid information transmission can takes place by using signals which change rapidly with time. In order to decrease transmission time, signals are to be speeded up.

However, any electrical system always include energy storage, and it is a well-known physical law that for a but lossless system, a change in stored energy requires definite amount of time. Therefore, we cannot arbitrarily increase signaling speed because finally the system would cease to respond to rapid signal changes.

Channel Bandwidth. *It is the range of frequencies that a system can transmit with reasonable fidelity.* For example, if a channel can transmit with reasonable fidelity a signal whose frequency components occupy a range from dc up to a maximum of 5000 Hz (5 KHz), the channel bandwidth B is 5 KHz.

Again consider the possibility of increasing the speed of information transmission by time compression of the signal. If a signal is compressed in time by a factor of two, it can be transmitted in half the time, and the speed of transmission is doubled. To transmit this compressed signal without distortion, the channel bandwidth must also be doubled. Thus, the rate of information transmission is directly proportional to B. More generally, if a channel of bandwidth B can transmit N pulses per second, then to transmit KN pulses per second we need a channel of bandwidth KB. We observe:

1. *The number of pulses/second that can be transmitted over a channel is directly proportional to its bandwidth B.*

2. The signal speed is conceived in terms of bandwidth, that is, the width of the signal spectrum in frequency domain. In other words, the rate at which a system can change stored energy is reflected by its usable frequency response, measured in terms of the ***system bandwidth.*** It means that if large amount of information is to be transmitted in a small amount of time, it requires wideband signals to contain the information and, in turn, wideband system would be necessary to accomodate these signals. Therefore, bandwidth is a fundamental limitation. *If it is not possible to have sufficient bandwidth, it may become necessary to decrease signaling speed and as a result increase transmission time.*

3. The task of equipment designer is governed largely not by absolute bandwidth but fractional bandwidth, which means absolute bandwidth divided by the centre frequency. A wideband signal when modulated on high-frequency carrier, leads to reduced fractional simplification of equipment

design. This is one of the reason why TV signals of bandwidth about 5 MHz are modulated on much higher carriers than *AM* radio where signal bandwidth is about 10 kHz.

Similarly, if fractional bandwidth is given, the absolute bandwidth can be increased almost indefinitely by going for higher carrier frequencies. For example a 5-GHz microwave system can carry information 10,000 times in a given period at a 300-kHz radio-frequency carrer. A laser beam of frequency 5×10^{14} Hz can carry 10^5 times information as compared to the microwave system equivalent to 10 million TV channels.

1.14.1 The Bandwidth S/N Ratio Trade-off. From above discussion, since the *S/N* ratio (*SNR*) is proportional to the signal power *S*, we can say that *SNR and bandwidth are exchangeable.* It will be shown later that the relationship between the bandwidth expansion factor and the *SNR* is exponential. Thus, if a given rate of information transmission requires a channel bandwidth B_1 and a signal-to-noise ratio $(S/N)_1$, then it is possible to transmit the same information over a channel bandwidth, B_2 and a signal-to-noise ratio $(S/N)_2$, where

$$(S/N)_2 = (S/N)_1^{B_1/B_2} \qquad ...(1.1)$$

It means that if the channel bandwidth is doubled, the required *SNR* is only a square root of the former *SNR* and tripling the channel bandwidth reduces the corresponding *SNR* to only a cube root of the former *SNR* ratio. Thus, *a relatively small increase in channel bandwidth buys a large advantage in terms of reduced transmission power. But a large increase in transmitted power buys a meager advantage in bandwidth reduction.* Hence, in practice, the exchange between *B* and *SNR* is usually in the sense of increasing *B* to reduce transmitted power and is rarely the other way around.

Note that Eq. 1.1 provides the upper bound on the exchange between the *S/N* ratio and bandwidth. Not all systems are capable of achieving this bound. For example, frequency modulation (*FM*) is one scheme that is commonly used in ratio broadcasting for improving the signal quantity at the receiver by increasing the transmission bandwidth. As shall be seen later, an *FM* system does not make efficient use of bandwidth in reducing the required *S/N* ratio, and its performance falls far short of that in Eq.1.1. *PCM* (pulse code modulation), on the other hand, comes closer (within 10 dB) in realizing the performance in Eq. 1.1. Generally speaking, transmission of signal in digital form comes much closer to the realization of the limit in Eq. 1.1 than does transmission of signals in analog form.

1.15 NUMBER SYSTEMS

As many communication systems involve conversion from analog to digital and *vice versa*, we here brush up our background in number systems which would be helpful in the study of pulse and data communication.

The ***decimal counting system*** is an example of a positional notation. The name decimal implies ten unique symbols : 0, 1, 2, 3, 4, 5, 6, 7, 8 and 9. Thus, the radix or base of the system is 10. The value of the number represented is given by the sum of the digit values, each weighted by a power of 10 determined by its position. For example $(6278.35)_{10}$ is

$$6 \times 10^3 + 2 \times 10^2 + 7 \times 10^1 + 8 \times 10^0 + 3 \times 10^{-1} + 5 \times 10^{-2}.$$

Similarly, in ***binary counting,*** the radix is two, and the value can be determined by the digit values weighted by a power of 2. For example, $(10010.11)_2$ is equivalent to

$$1 \times 2^4 + 0 \times 2^3 + 0 \times 2^2 + 1 \times 2^1 + 0 \times 2^{-1} + 1 \times 2^{-2}.$$

The maximum counting states that *n* binary digits can provide in total is 2^n. That is, 3-digit binary system can have $2^3 = 8$ states *i.e.*, it can go from 0 to 7. Therefore, the highest count by *n* bits is $2^n - 1$.

There can also be other forms of system like the octal system, where the radix or base is 8 or hexa-system where the radix or base is 16.

In counting successively, the digits are used in turn, increasing the next higher power by one as shown on next page:

2^3	2^2	2^1	2^0	*Decimal*
0	0	0	0	0
0	0	0	1	1
0	0	1	0	2
0	0	1	1	3
0	1	0	0	4
1	0	0	0	8
1	0	0	1	9
1	1	0	0	12
1	1	0	1	13
1	1	1	0	14
1	1	1	1	15

1.15.1 Conversion of a Decimal Number of Binary Number. The successive remainders obtained by division (by 2) yield the bits of the binary number.

Example 1.4. *Convert* $(19)_{10}$ *to binary number.*

Solution.

	Remainder	
19/2 = 9	1	Least significant digit
9/2 = 4	1	
4/2 = 2	0	
2/2 = 1	0	
1/2 = 0	1	Most significant digit

Therefore, $(19)_{10} = (10011)_2$

Example 1.5. *Convert* $(119)_{10}$ *to octal number.*

Solution.

		Remainder
119/8	14	7
14/8	1	6
1/8	0	1

$(119)_{10} = (167)_8$

Example 1.6. *Convert* $(10011)_2$ *to decimal number.*

Solution.

$$\begin{aligned}(10011)_2 &= 1 \times 2^4 + 0 \times 2^3 + 0 \times 2^2 + 1 \times 2^1 + 1 \times 2^0 \\ &= 16 + 0 + 0 + 2 + 1 = (19)_{10}\end{aligned}$$

Example 1.7. *Convert* $(167)_8$ *to decimal number.*

Solution.

$$(167)_8 = 1 \times 8^2 + 6 \times 8^1 + 7 \times 8^0 = 64 + 48 + 7 = (119)_{10}.$$

1.15.2 Conversion of Decimal Fraction to Binary. The fractional part is successively multiplied by 2 to obtain the next significant digit and then the fractional part of the product is again taken for the next digits, till either a recurring digits or zero product is attained.

Example 1.8. *Convert* $(0.3)_{10}$ *to binary.*

Solution.

Decimal	Product	Binary
0.3×2	0.6	0
0.6×2	1.2	1
0.2×2	0.4	0
0.4×2	0.8	0
0.8×2	1.6	1
0.6×2	1.2	1 recurring

Therefore, $(0.3)_{10} = (0.01001)_2$.

Example 1.9. *Convert* $(0.6)_{10}$ *to octal number.*

Solution. Similar procedure in octal system can also be used. Here, one has to multiply by 8 in place of 2.

Decimal	Product	Octal
0.6×8	4.8	4
0.8×8	6.4	6
0.4×8	3.2	3
0.2×8	1.6	1
0.6×8	4.8	

Therefore, $(0.6)_{10} = (0.4631)_8$. 4 (recurring)

1.15.3. Sign Bit. Negative sign is represented by '1' and positive sign by '0'. Sign bit is placed first along with a comma before the binary code of that number. As +5 is represented by 0,101 but -5 is represented by 1,101 (or 0,0101 and 1,0101 respectively in *BCD*).

Negative numbers are also represented in the complement way. Any negative number can be conveniently represented by either in one's complement form or in two's complement form, **One's complement** representation is obtained by replacing every '1' by '0' and never '0' by '1' of that binary number. Thus -5 is represented by 1,010 (or 1,0010 in *BCD*). The bit appearing in front of comma is sign bit. Thus one's complement of n bit (binary bit) number X is given by $[(2^n - 1) - X]$.

Example 1.10. *Find one's complement of - 14.*

Solution. $[(2^4 - 1) - X] = [(16 - 1) - 14] = (15 - 14) = (1111 - 1110) = 0001$ excluding the sign bit. With the sign bit, one's complements of $-14 = 1,0001$.

Two's complement representation of a negative number is obtained by adding a '1' to the one's complement. Thus, the 2's complement of – 4 is 0001 + 1 = 0010 and with sign bit it is 1,0010. With the help of 1's complement of 2's complement, the addition/subtraction of number can be done more easily, but correctness is obtained only to those computations, where the answer does not exceed the limit of binary numbers.

Example 1.11. *Perform the following addition by 1's complement.*

Solution.

$$\begin{array}{r} +5 = 0,0101 \\ \underline{-3 = 1,1100} \\ 10,0001 \\ \hookrightarrow 1 \text{ overflow} \\ \underline{+2 = 0,0010} \end{array} \qquad \begin{array}{r} +3 = 0,0011 \\ \underline{+7 = 0,0111} \\ +10 = 0,1010 \end{array}$$

In case of 1's complement, the overflow is to be added to the least significant digit, as shown above.

Example 1.12. *Perform the following additions by 2's complement.*

Solution.

$$\begin{array}{rcl} +5 = 0{,}0101 & \quad & -5 = 0{,}1011 \\ \underline{-3 = 1{,}1101} & & \underline{+3 = 0{,}0011} \\ +2 = 10{,}0010 & & -2 = 1{,}1110 \\ \uparrow & & \\ \text{ignore} & & \end{array}$$

In case of 2's complement, the overflow is to be ignored, as shown above.

1.15.4. Binary Codes. The 8421 Code. The 8421 code expresses each decimal digit by its four bit binary equivalents. For example, 719 is represented by

$$\begin{array}{ccc} 7 & 1 & 9 \\ \downarrow & \downarrow & \downarrow \\ 0111 & 0001 & 1001 \end{array}$$

Therefore, 719 is equivalent to (0111 0001 1001) in 8421 code or Binary Coded Decimal (*BCD*).

The Excess-3 Code. The excess-3 code is also a form of *BCD* (Binary Coded Decimal). To encode in this code, *number 3 is added to every decimal digit before converting it to binary.* For example, 1 in Excess-3 code will be equivalent to 1 + 3 = 4 of decimal in binary manner.

Therefore, 1 in Excess-3 code = 4 in decimal code
= 0100 of binary.

Similarly, 5 = 1000 (5 + 3 = 8 in decimal)

and, 9 = 1100 (9 + 3 = 8 in decimal)

Decimal			*Equivalent Excess 3 code*
0			0011
1			0100
2			0101
3			0110
4			0111
9			1100
10		0100	0011
11		0100	0100
12		0100	0101
134	0100	0110	0111

Other Weighted Codes. There could be other forms of weighted codes like 4221 or 6311, etc. If '1' is placed in any of the four bit positions, it will carry the weight of that position described by the code. For example:

1010 in 4221 code carries a weight of 6. But 1010 in 6311 code carries a weight of 7.

The Gray Code. The Gray code is a *cyclic code*. In this code, *each gray number differs from the preceding number by a single bit*. To convert binary to Gray code, the rules followed are illustrated by the following example.

Example 1.13. *Consider a binary number 1101.*

1. The first Gray digit is the same as the first binary digit.

1101	binary
1	Gray

2. Add the first two bits of binary number. The sum is the next gray digit disregarding the carry.

$$\begin{array}{ll} \overset{\wedge}{1}101 & \text{binary} \\ 10 & \text{Gray} \end{array}$$

3. Repeat step 2 to get third and fourth digit

$$\begin{array}{llll} 1\overset{\wedge}{1}01 & \text{binary} & 11\overset{\wedge}{0}1 & \text{binary} \\ 101 & \text{Gray} & 1011 & \text{Gray} \end{array}$$

Decimal	*Binary*	*Gray*
0	0000	0000
1	0001	0001
2	0010	0011
3	0011	0010
11	1010	1110
12	1100	1010
15	1111	1000

Self-Checking/Correcting Codes. *In these codes a parity bit is introduced in the one side of a code to detect the error.* It can follow either odd parity code or even parity code. This is an extra 1 bit depending on whether the total number of ones in the code is even or odd. In case of even parity code, '0' is put when the total number of ones are even and '1' is put when the total number or ones are odd, so to get the total number of ones always even, when the parity bit in included. Similarly, '1' or '0' is put in the odd parity code to make the total number of ones odd, when the parity bit is included. For example:

Decimal Number	8421 code	
	odd parity bit	*even parity bit*
0	0000 1	0000 1
1	0001 0	0001 1
5	0101 1	0101 0
8	1000 0	1000 1

A self-correcting code is one in which a parity check bit is given for both columns and rows. In this system, if there is an error, it will be corrected by itself provided the error is of singular bit. For example, in the even parity code, a self-correcting system can be given as

		Row parity bit	
		1	
Code	00001	1	
	01000	0	
	01100	1	
	01110	0	
	11001	1	*Column parity bit*

OBJECTIVE TYPE QUESTIONS AND ANSWERS

1. In a communication system, noise is likely toaffect the signal
(*a*) at the transmitter (*b*) in the information source
(*c*) in the channel (*d*) at the receiver

2. Indicate the *false* statement. Modulation is used to
(*a*) separte differing transmissions (*b*) reduce the use of practicable antennas
(*c*) allow the use of practicable antennas
(*d*) ensure that intelligence may be transmitted over long distances

3. For efficient transmission and reception of electromagnetic signals, the transmitting and receiving antennas should *atleast* have a height of

(*a*) half wavelength (*b*) one wavelength
(*c*) quarter wavelength (*d*) one-third wavelength

4. Apart from the transmitter, at which stage signal is modified and reshaped?

(*a*) Channel (*b*) Receiver
(*c*) Both of these

5. As the length of the channel increases, distortion of the transmitted signal

(*a*) increases (*b*) remains unaffected
(*c*) decreases

6. Linear distortion in electrical communication occurs because of different amount of

(*a*) Attenuation suffered by different frequency components of the signal
(*b*) Different amounts of phase shift suffered by difficult frequency components of the signal
(*c*) Both of these.

7. Linear distortion can partly be corrected at the receiver by an euqaliser with

(*a*) Gain characteristics complementary to those of the channel.
(*b*) Phase characteristics complementary to those of the channel.
(*c*) Both of these.

8. The distortion cuased by attentuation that varies with the signal amplitude is called

(*a*) Linear distortion (*b*) Non-linear distortion
(*c*) Both of these.

9. Which of the following distortions can be partly corrected by a complementary equaliser at the receiver?

(*a*) Linear distortion (*b*) Non-linear distortion
(*c*) Both of these

10. The one milliwatt decibel (dBm) is defined as

(*a*) $\text{dBm} = 10 \log \frac{\text{Power}}{1 \text{ mW}}$ (*b*) $\text{dBm} = \log \frac{\text{Power}}{1 \text{ mW}}$
(*c*) $\text{dBm} = \log \frac{1 \text{ mw}}{\text{Power}}$

11. One milliwatt in dBm is

(*a*) Zero dBm (*b*) 1 dBm
(*c*) 10 dBm

12.12 dBm is equivalent to

(*a*) 1 mW (*b*) 20 mW
(*c*) 100 mW

ANSWERS

1. (*c*)	**2.** (*b*)	**3.** (*c*)	**4.** (*b*)	**5.** (*a*)	**6.** (*c*)
7. (*c*)	**8.** (*b*)	**9.** (*c*)	**10.** (*a*)	**11.** (*a*)	**12.** (*b*)