Principle of Communications

Chapter 1 : Introduction

Q. 1 Explain an electronic communication system with the help of a block diagram.

Dec. 10, May 12, Dec. 12, Dec. 14

Ans. : The block diagram of the simplest possible communication system is as shown in Fig. 1.1.

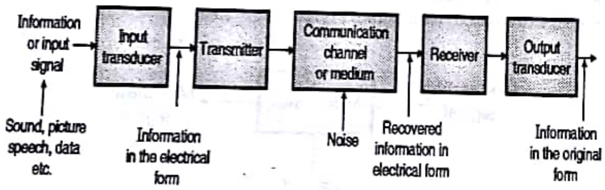

Fig. 1.1: Block diagram of the basic communication system

As seen from the Fig. 1.1, the elements of a basic communication system are transmitter, a communication medium (channel) and the receiver. When the transmitted signal is travelling from the transmitter to the receiver over a communication channel, noise gets added to it.

The elements of basic communication system are as follows :

1. Information or input signal
2. Input transducer
3. Transmitter
4. Communication channel or medium
5. Noise
6. Receiver
7. Output transducer

1. Information or Input signal

The communication systems have been developed for communicating useful information from one place to the other. This information can be in the form of a sound signal like speech or music, or it can be in the form of pictures (TV signals) or it can be data information coming from a computer.

2. Input transducer

The information in the form of sound, picture or data signals cannot be transmitted as it is. First it has to be converted into a suitable electrical signal. The input transducer block does this job. The input transducers commonly used in the communication systems are microphones, TV camera etc.

3. Transmitter

The function of the transmitter block is to convert the electrical equivalent of the information to a suitable form. In addition to that it increases the power level of the signal. The power level should be increased in order to increase the range of transmitted signal. The transmitter consists of the electronic circuits such as amplifier, mixer, oscillator and power amplifier.

4. Communication channel or medium

The communication channel is the path used for transmission of electronic signal from one place to the other. The communication medium can be conducting wires, cables, optical fibre or free space. Depending on the type of communication medium, two types of communication systems will exist. They are :

(i) Wired communication or line communication

(ii) Wireless communication or radio communication

1. Line communication

The line communication systems use the communication mediums like the simple wires or cables or optical fibers. The examples of such systems, are telegraph and telephone systems, cable T.V. etc. Due to physical connection from one point to the other, these systems cannot be used for the communication over long distances.

2. Radio communication

The radio communication systems use the free space as their communication medium. They do not need the wires for sending the information from one place to the other.

The radio or TV broadcasting, satellite communication are the examples of the wireless communication. These systems transmit the signal using a transmitting antenna in the free space.

The transmitted signal is in the form of electromagnetic waves. A receiving antenna will pick up this signal and feed it to the receiver. Radio communication can be used for the long distance communication such as from one country to the other or even from one planet to the other.

5. Noise

Noise is an unwanted electrical signal which gets added to the transmitted signal when it is travelling towards the receiver. Due to noise, the quality of the transmitted information will degrade. Once added, the noise cannot be separated out from the information. Hence noise is a big problem in the communication systems. (Specially analog communication systems). The noise can be either natural or manmade. The sources of natural noise are lightning or radiation from the sun and stars etc. The man made noise includes the noise produced by electrical ignition systems of the automobiles, welding machines, electric motors etc. Even though noise cannot be completely eliminated, its effect can be reduced by using various techniques.

6. Receiver

The process of reception is exactly the opposite process of transmission. The received signal is amplified, demodulated and converted into a suitable form. The receiver consists of electronic circuits like mixer, oscillator, detector, amplifier etc.

7. **Output transducers**

The output transducer converts the electrical signal at the output of the receiver back to the original form i.e. sound or TV pictures etc.

The typical examples of the output transducers are loud speakers, picture tubes, computer monitor etc.

Q. 2 What are the advantages and disadvantages of digital communication. Also draw block diagram of PCM and explain it. May 14

Ans. :

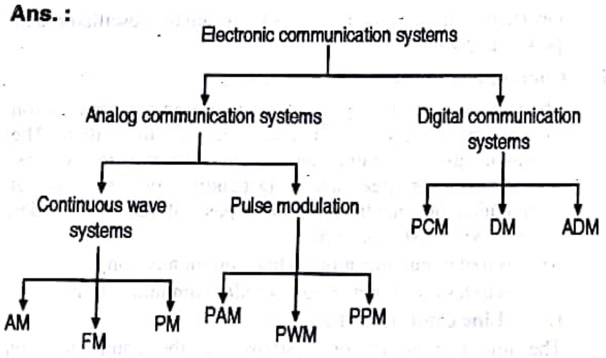

Fig. 1.2 : Classification based on analog or digital communication

Digital communication :

Definition : The modulation system or technique in which the transmitted signal is in the form of digital pulses of constant amplitude, constant frequency and phase is called as digital modulation system.

Examples : Pulse Code Modulation (PCM), Delta Modulation (DM), Differential PCM (DPCM) and Adaptive Delta Modulation (ADM) are the examples of digital modulation. In the PCM and DM, a train of digital pulses is transmitted by the transmitter.

All the pulses are of constant amplitude, width and position. The information is contained in the combination of the transmitted pulses.

Advantages of Digital Communication :

Some of the advantages of digital communication are as follows :

1. Due to the digital nature of the transmitted signal, the interference of additive noise does not introduce many errors. So digital communication has a better noise immunity.
2. Due to the channel coding techniques used in digital communication, it is possible to detect and correct the errors introduced during the data transmission.
3. Repeaters can be used between transmitter and receiver to regenerate the digital signal. This improves the noise immunity further.
4. Due to the digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.
5. TDM (Time Division Multiplexing) technique can be used to transmit many voice channels over a single common transmission channel.
6. Digital communication is useful in military applications where only a few permitted receivers can receive the transmitted signal.
7. Digital communication is becoming simpler and cheaper as compared to the analog communication due to the invention of high speed computers and Integrated Circuits (ICs).

Disadvantages of Digital Communication

Some of the important disadvantages of digital communication are :

1. The bit rates of digital systems are high. Therefore, they require a larger channel bandwidth as compared to analog systems.
2. Digital modulation needs synchronization in case of synchronous modulation.

Q. 3 Define the term : Modulation. May 16

Ans.:

In the **Modulation** process, two signals are used namely the **modulating signal** and the **carrier.** The modulating signal is nothing but the base and signal or information signal while carrier is a high frequency sinusoidal signal.

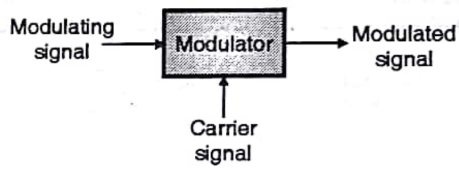

Fig. 1.3 : Modulation

Q. 4 Explain the need of modulation. Dec. 11

Ans. : A question may be asked as, when the baseband signals can be transmitted directly why to use the modulation ?

The answer is that the baseband transmission has many limitations which can be overcome using modulation. It is as explained below.

In the process of modulation, the baseband signal is "translated" i.e. shifted from low frequency side to high frequency side of the frequency spectrum.

This frequency shift is proportional to the frequency of carrier. The modulation process has the following advantages :

Q. 5 Write Short Notes on : International standards for communication systems and assignments. May 10, Dec. 11

Ans. :

1. Wireless communication systems use the atmosphere as the transmission channel.
2. The interference and propagation conditions are mainly dependent on the transmission frequency.
3. It is possible to use any type of modulation such as AM, FM, SSB, PSK, FSK etc.
4. However in order to regulate the things, and to minimize the interference, government regulations specify the following :
   1. Modulation type
   2. Bandwidth
   3. Power
   4. Type of information that can be transmitted over the designated frequency bands.

5. The frequency assignments and technical standards are set internationally by the International Telecommunications Union (ITU).
6. The ITU is a specialized agency of United Nations (U.N.) and its head quarters is in Geneva Switzerland.
7. The ITU is has been structred into three parts :
   1. The Radio communication sector (ITU – R)
   2. The Telecommunications standardization section (ITU-T)
   3. The Telecommunications development sector (ITU-D)
8. The ITU-R sector provides frequency assignments and ensures that the EM spectrum is utilized efficiently.
9. The ITU-T sector checks the technical, operating and traffic problems. It recommends worldwide standards for the Public Telecommunications Network (PTN) and related radio systems.
10. The ITU-D sector provides technical assistance specially for the developing countries.
11. In 1992 ITU was organized into two main sector :
    1. The International Telegraph and Telephone Consultive Committee (CCITT).
    2. The International Radio Consultative Committee (CCIR).
12. All the member nations of ITU are free to decide the spectral usage and standards to be adopted in their territory by abiding the overall frequency plan and standards adopted by ITU.
13. Generally each nation establishes an agency to look after the administration of radio frequency assignments in their country. For example, in U.S. they have FCC i.e. Federal Communication Commission (FCC).
14. The FCC has subdivided the international frequency standards so as to accommodate 70 different categories of services and 9 million transmitters.

Q. 6 Explain types of communication channels.

Dec. 14, May 10

Ans. :

1. A communication channel provides the connection between the transmitter and receiver.
2. Different types of channels are as follows :
   1. Wireline channels
   2. Fiber optic channels
   3. Wireless Electromagnetic channels.
   4. Underwater Acoustic channels
   5. Storage channels

1. Wireline Channels

(i) These channels use the pair of wires that carry the signal in the electrical form.

(ii) The telephone network is the best example of wireline channel. Fig. 1.4 shows the frequency spectrum allocated to various wireline channels.

(iii) The wireline channels are used for the transmission of voice as well as data information.

(iv) The examples of wireline channels are the twisted pair wire lines and coaxial cables. They are also called as guided electromagnetic channel.

2. Fiber Optic Channels

(i) The optical fiber can work as a channel. The information which is to be transmitted is converted into light and then passed over the optical fiber to the receiver.

(ii) At the receiver the light is converted back to signal by means of an optical detector.

(iii) Fig. 1.5 shows the basic optical fiber link.

(iv) The light source or modulator used in transmitter is either an LED or LASER. Information is transmitted by varying (modulating) the intensity of light source with the information signal.

(v) The propagating light is amplified periodically along the transmission path to compensate for the attenuation of signal.

(vi) At the receiver, the light intensity is detected by the optical detector such as a photodiode.

(vii) In future the fiber optic channels will replace all the wireline channels in the telephone network.

3. Wireless Electromagnetic Channels

(i) In radio communication systems, the transmitter radiates its output in the form of electromagnetic waves using the transmitting antenna.

(ii) These waves travel towards the receiver through free air which acts as a communication channel.

(iii) In the electromagnetic spectrum, various frequency bands are allotted for different applications.

(iv) The propagation of EM waves can take place by means of one of the following types :
   1. Ground wave propagation.
   2. Sky wave propagation.
   3. Space wave propagation (Line of Sight or LOS)

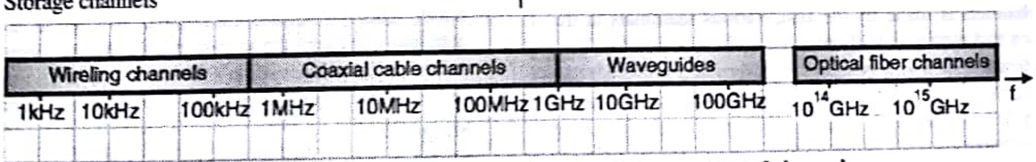

Fig. 1.4 : Frequency spectrum allocation for different types of channels

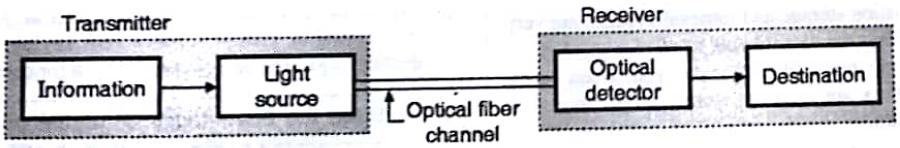

Fig. 1.5 : Fiber optic link

(v) Fig. 1.6 shows the relation between frequency and the type of propagation.

(vi) Ground wave propagation is dominant mode of propagation in the frequency band as 30 kHz to 3 MHz.

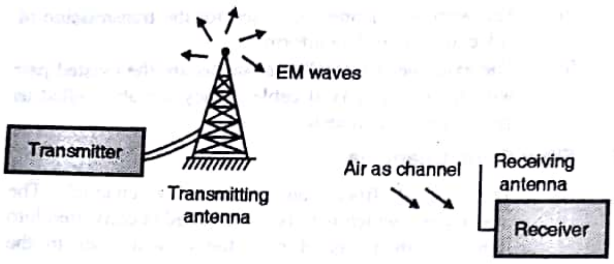

Fig. 1.6 : Wireless electromagnetic channel

4. Underwater Acoustic Channels

(i) The EM waves can not propagate over long distances underwater except at extremely low frequencies. But transmission at such low frequencies is very expensive because large and powerful transmitters are required to be used.

(ii) The underwater communication is required when oceans are being explored. The data collected by the sensors placed underwater should be transmitted first to the surface and then to the data collection center.

(iii) The attenuation of EM waves in water is expressed in terms of the skin depth which is defined as the distance over which a signal is attenuated by 1/e.

(iv) For sea waters the skin depth is given by

$$\delta = \frac{50}{\sqrt{f}}$$

where f is frequency in Hz and δ = Depth in metres.

(v) If f = 10 kHz then the skin depth δ = 2.5m. In contrast the acoustic (sound) signals can propagate over a distance of tens or hundreds of kilometres.

(vi) A shallow water acoustic channel is a multipath channel that means signal get propagated over multiple paths, due to signal reflections from bottom and surface of sea. These multipath signals undergo different delays. These delayed signals tend to partially cancel each other which results in the signal fading.

(vii) In addition to this there will be frequency dependent attenuation. This attenuation is proportional to square of the signal frequency. The noise on the acoustic channels is made by the fish, various mammals in the sea and men near harbours.

(viii) However it is still possible to design and implement an efficient and highly reliable acoustic communication system, for the transmission of digital signals over long distances.

5. Storage Channels

(i) The information storage and retrieval systems are very important when the data is being handled.

(ii) The magnetic tapes, digital audio and video tapes, magnetic disks are used for storing large amount of computer data.

(iii) Optical disks are used for storing the computer data, music and video information. All these data storing systems can be characterised as **storage channels.**

(iv) The process of storing data is equivalent to transmission whereas the process of retrieval is equivalent to reception.

(v) Noise is generated by the electronic components and interference from the adjacent tracks.

Q. 7 Explain digital communication system in details.
Dec. 06, Dec. 14, May 15

Ans.: Two of the most commonly used digital communication systems are PCM (Pulse Code Modulation) and DM (Delta Modulation).

Fig. 1.7 shows the block diagram of a digital communication system. In this diagram three basic signal processing operations have been included. They are :

1. Source coding
2. Channel coding and
3. Modulation.

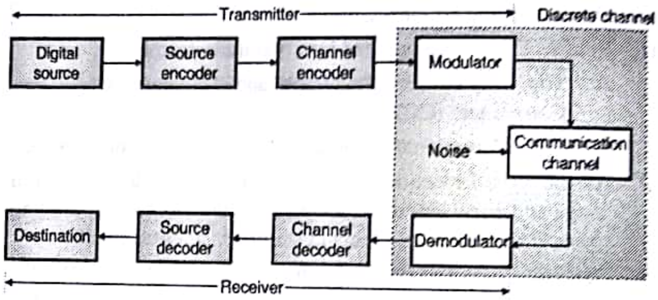

Fig. 1.7 : Digital communication system

The source of information is assumed to be digital. If it is analog then it must be converted first to digital.

1. Source coding

In source coding the source encoder converts the digital signal generated at the source output into another signal in digital form. Source encoding is used to reduce or eliminate redundancy for ensuring an efficient representation of the source output. Different source coding techniques are PCM, DM, ADM etc.

The conversion of signal from one form to the other is called as mapping. Such a mapping is usually one as to one. Due to elimination of redundancy the source coding provides an efficient representation of the source output.

2. Channel coding

Channel encoding is done to minimize the effect of channel noise. This will reduce the number of errors in the received data and will make the system more reliable. Channel coding technique introduces some redundancy. The channel encoder maps the incoming digital signal into a channel input.

3. Modulation

Modulation is used for providing an efficient transmission of the signal over the channel. The modulator can use any of the CW digital modulation techniques such as ASK (amplitude shift keying's), FSK (frequency shift keying) or PSK (phase shift keying). The demodulator is used for demodulation.

Q. 8 State the advantages of digital communication over analog communication. Justify each point.
Dec. 06, May 16

Ans. :

Advantages of digital Communication

1. Due to the digital nature of the transmitted signal, the interference of additive noise does not introduce many errors. So digital communication has a better noise immunity.
2. Due to the channel coding techniques used in digital communication, it is possible to detect and correct the errors introduced during the data transmission.
3. Repeaters can be used between transmitter and receiver to regenerate the digital signal. This improves the noise immunity further and also extends the range of communication.
4. Due to the digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.
5. TDM (Time Division Multiplexing) technique can be used to transmit many voice channels over a single common transmission channel. Thus digital telephony is possible to achieve.
6. Digital communication is suitable in military applications where only a few permitted receivers can receive the transmitted signal.
7. Digital communication is becoming simpler and cheaper as compared to the analog communication due to the invention of high speed computers and Integrated Circuits (ICs).

Q. 9 Compare analog and digital communication system. **May 09, Dec. 10, May 14**

Ans. :

Sr. No.	Analog modulation	Digital modulation
1.	Transmitted modulated signal is analog in nature.	Transmitted signal is digital i.e. train of digital pulses.
2.	Amplitude, frequency or phase variations in the transmitted signal represent the information or message.	Amplitude, width or position of the transmitted pulses is constant. The message is transmitted in the form of code words.
3.	Noise immunity is poor for AM, but improved for FM and PM.	Noise immunity is excellent.
4.	It is not possible to separate out noise and signal. Therefore repeaters cannot be used.	It is possible to separate signal from noise. So repeaters can be used.
5.	Coding is not possible.	Coding techniques can be used to detect and correct the errors.
6.	Bandwidth required is lower than that for the digital modulation methods.	Due to higher bit rates, higher channel bandwidths is needed.
7.	FDM is used for multiplexing.	TDM is used for multiplexing.
8.	Not suitable for transmission of secret information in military applications.	Due to coding techniques, it is suitable for military applications.
9.	Analog modulation systems are AM, FM, PM, PAM, PWM etc.	Digital modulation systems are PCM, DM, ADM, DPCM etc.

Chapter 2 : Fourier Transform

Q. 1 What are the energy signals and power signals ? **May 10, May 14**

Ans. :

Power signal :

A signal is called as a power signal if its "average normalized power" is non-zero and finite. It has been observed that almost all the periodic signals are power signals.

Energy signals :

A signal having a finite non-zero total normalized energy is called as an energy signal. It has been observed that almost all the non-periodic signals defined over a finite period, are energy signals. As these signals are defined over a finite period, they are called as time limited signals.

Q. 2 What is convolution of signals ? **May 15**

Ans. : Convolution is a mathematical operation which is used as a tool by communication engineers for system analysis, probability theory and transform calculations.

Convolution can be performed in time as well as frequency domain. The convolution of two functions x (t) and y (t) is defined as,

$$\text{Convolution : } x(t) * y(t) = \int_{-\infty}^{\infty} x(\tau) \cdot y(t-\tau)\, d\tau \quad \ldots(1)$$

Equation (1) is called as convolution integral. Note that the independent variable here is "t" which is same as the independent of the functions x (t) and y (t) which are being convolved.

The integration is always performed with respect to a dummy variable such as τ and t is treated as a constant as far as the integration is concerned.

The process of convolution involves following operations of y (τ) while the signal x (τ) remains unchanged :

1. Folding or time reversal to obtain y ($-\tau$).
2. Time shifting the folded signal y ($-\tau$) to obtain y ($t-\tau$).
3. Multiplication of x (τ) and y ($t-\tau$).
4. Integration of the product term x (τ) $\cdot$ y ($t-\tau$).

Q. 3 Write short notes on : Properties of Fourier transforms. May 11. Dec. 11

Ans. :

Some of the important properties of the Fourier transform are listed as follows :

Linearity or superposition	Area under X (f)
Time scaling	Differentiation in time domain
Duality or symmetry	Integration in time domain
Time shifting	Conjugate function
Frequency shifting	Multiplication in time domain (Multiplication theorem)
Area under x (t)	Convolution theorem.

Let us understand these properties one-by-one.

Property 1 : Linearity or Superposition :

If $x_1(t) \overset{F}{\longleftrightarrow} X_1(f)$ and $x_2(t) \overset{F}{\longleftrightarrow} X_2(f)$ represent the Fourier transform pairs and if a_1 and a_2 are constants then we can write,

$$[a_1 x_1(t) + a_2 x_2(t)] \overset{F}{\longleftrightarrow} [a_1 X_1(f) + a_2 X_2(f)] \quad ...(1)$$

That means the linear combination of inputs gets transformed into linear combination of their Fourier transforms.

This property can be used to obtain the Fourier transform of a complicated function say x(t) by decomposing it in the form of sum of simpler functions, say $x_1(t)$ and $x_2(t)$.

Property 2 : Time Scaling :

1. $x(\alpha t)$ represents a time scaled signal and $X(f/\alpha)$ represents the frequency scaled signal or scaled frequency spectrum.
2. For $\alpha < 1$, $x(\alpha t)$ represents a compressed signal but $X(f/\alpha)$ represents an expanded version of X (f).
3. And for $\alpha > 1$, $x(\alpha t)$ will be an expanded signal in the time domain. But its Fourier transform $X(f/\alpha)$ represents a compressed version of X (f).

"α" being a constant, can be positive or negative. i.e. $\alpha > 0$ or $\alpha < 0$. Let us find the F.T. considering both the possibilities.

Property 3 : Duality or Symmetry Property :

This property states that, if $x(t) \overset{F}{\longleftrightarrow} X(-f)$

Then $X(t) \overset{F}{\longleftrightarrow} x(-f)$ ...(2)

i.e. t and f can be interchanged.

Meaning

1. In the term x(t), "x" represents the shape of the signal and "t" shows that the variable is time.
2. And in the term X (– f), "X" represents the shape of the spectrum and "f" shows that the variable is frequency.
3. The Duality theorem tell us that if $x(t) \overset{F}{\longleftrightarrow} X(-f)$ then the shape of the signal in the time domain and the shape of the spectrum can be interchanged.

Property 4 : Time Shifting

The time shifting property states that if x (t) and X (f) form a Fourier transform pair then,

$$x(t - t_d) \overset{F}{\longleftrightarrow} e^{-j2\pi f t_d} X(f) \quad ...(3)$$

Here the signal $x(t - t_d)$ is a time shifted signal. It is the same signal x (t) only shifted in time.

Property 5 : Area under x(t)

This property states that the area under the curve x(t) equals the value of its Fourier transform at f = 0.

i.e. if $x(t) \overset{F}{\longleftrightarrow} X(f)$ then,

$$\text{Area under } x(t) = \int_{-\infty}^{\infty} x(t)\, dt = X(0) \quad ...(4)$$

$$\text{Area under } x(t) = \int_{-\infty}^{\infty} x(t)\, dt$$

$$= \int_{-\infty}^{\infty} x(t)\, e^{-j2\pi ft}\, dt \text{ at } f = 0.$$

$$= X(f) \text{ at } f = 0$$

$\therefore$ Area under x (t) = X (0). **...Proved.**

Property 6 : Area under X (f)

If $x(t) \overset{F}{\longleftrightarrow} X(f)$ then the area under X (f) is equal to the value of signal x(t) at t = 0.

That means if $x(t) \overset{F}{\longleftrightarrow} X(f)$ then,

$$\text{Area under } X(f) = \int_{-\infty}^{\infty} X(f) \cdot df = x(0) \quad ...(5)$$

Proof : By definition of inverse Fourier transform,

$$x(t) = \int_{-\infty}^{\infty} X(f) \cdot e^{j2\pi ft}\, df$$

Substitute t = 0 in this equation to get,

$$x(0) = \int_{-\infty}^{\infty} X(f) \cdot e^0\, df = \int_{-\infty}^{\infty} X(f)\, dt$$

The RHS of this equation is the area under X (f). Hence the property is proved

Property 7 : Frequency Shifting

The frequency shifting characteristics states that if x(t) and X(f) form a Fourier transform pair then,

$$e^{j2\pi f_c t}\, x(t) \overset{F}{\longleftrightarrow} X(f - f_c) \quad ...(6)$$

Here f_c is a real constant.

Proof :

$$F[e^{j2\pi f_c t}\, x(t)] = \int_{-\infty}^{\infty} e^{j2\pi f_c t}\, x(t)\, e^{-j2\pi f t}\, dt$$

$$= \int_{-\infty}^{\infty} x(t)\, e^{-j2\pi (f - f_c) t}\, dt$$

$$= X(f - f_c)$$ **...Proved.**

The term $X(f - f_c)$ represents a shifted frequency spectrum. The whole spectrum is thus shifted right by "f_c" in the frequency domain, when the signal x (t) is multiplied by $e^{j2\pi f_c t}$ in the time domain.

Property 8 : Differentiation in Time Domain

Some processing techniques involve differentiation and integration of the signal x(t). **This property is applicable if and only if the derivative of x (t) is Fourier transformable.**

Statement :

Let $x(t) \overset{F}{\longleftrightarrow} X(f)$ and let the derivative of x(t) be Fourier transformable. Then,

$$\frac{d}{dt} x(t) \overset{F}{\longleftrightarrow} j2\pi f X(f) \qquad ...(7)$$

Q. 4 Find the fourier transform of the decaying exponential pulse shown in Fig. 2.1. Dec. 12

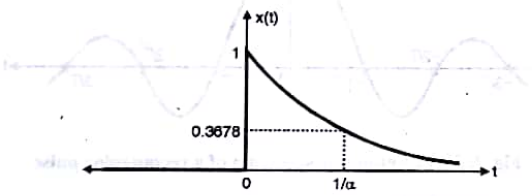

Fig. 2.1 : Decaying exponential pulse

Ans. : The exponential pulse shown in the Fig. 2.1 can be represented mathematically as follows :

$$x(t) = e^{-\alpha t} \quad \text{for } t \geq 0$$
$$= 0 \quad \text{for } t < 0 \qquad ...(1)$$

It can be represented in an alternate way as,

$$x(t) = e^{-\alpha t} u(t) \qquad ...(2)$$

The meaning of both the Equations (1) and (2) is the same. This is because u (t) = 1 for t ≥ 0. So multiplying by does not affect the original function.

To find the fourier transform :

$$F[x(t)] = \int_{-\infty}^{\infty} e^{-\alpha t} u(t) \cdot e^{-j2\pi f t} dt$$

...as per definition of FT.

$$= \int_{-\infty}^{\infty} e^{-\alpha t} e^{-j2\pi f t} dt = \int_{-\infty}^{\infty} e^{-(\alpha + j2\pi f)t} dt$$

$$= \frac{-1}{(\alpha + j2\pi f)} \left[e^{-(\alpha + j2\pi f)t}\right]_0^{\infty}$$

$$= \frac{-1}{(\alpha + j2\pi f)} [e^{-\infty} - e^0] = \frac{-1}{(\alpha + j2\pi f)} [0 - 1]$$

$$= \frac{1}{(\alpha + j2\pi f)}$$

This is the required result.

$$\therefore \quad e^{-\alpha t} u(t) \overset{F}{\longleftrightarrow} \frac{1}{(\alpha + j2\pi f)} \qquad ...(3)$$

Q. 5 Obtain the fourier transform of the delta function shown in Fig. 2.2(a). Dec. 09, Dec.12 , Dec. 14

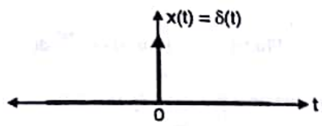

Fig. 2.2(a) : Delta function

Ans. :

1. By the definition of fourier transform.

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi f t} dt$$

$$= \int_{-\infty}^{\infty} \delta(t) e^{-j2\pi f t} dt \qquad ...(1)$$

We cannot substitute the value of δ (t) directly in the Equation (1) because it is infinitely large at t = 0. Therefore let us use the sifting property of the delta function.

2. Sifting property of delta function : The shifting property states that

$$\int_{-\infty}^{\infty} f(t)\, \delta(t - t_d)\, dt = f(t_d) \qquad ...(2)$$

Let us use this property in Equation (1) as follows :

3. In Equation (2) assume that $t_d = 0$ and $f(t) = e^{-j2\pi f t}$

$$\therefore \quad X(f) = \int_{-\infty}^{\infty} e^{-j2\pi f t} \cdot \delta(t - 0)$$

∴ by using Equation (2).

$$X(f) = e^{-j2\pi f t_d} \quad \text{but } t_d = 0$$

$$\therefore \quad X(f) = e^{-j2\pi \cdot 0} = 1 \qquad ...(3)$$

$$\text{Thus } \delta(t) \overset{F}{\longleftrightarrow} 1$$

The amplitude spectrum of the delta function is as shown in the Fig. 2.2(b). This shows that the delta function contains all the frequencies from – ∞ to ∞ with equal amplitudes. The fourier transform of a delta function is a dc signal.

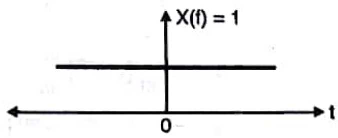

Fig. 2.2(b) : Amplitude spectrum of delta function

Q. 6 Obtain the fourier transform of a unit step function. Dec. 14

Ans. :

A unit step function is mathematically defined as,

$$u(t) = 1 \quad \text{for } t \geq 0$$
$$= 0 \quad \text{elsewhere}$$

Using the definition of the fourier transform we get,

$$F[u(t)] = \int_{-\infty}^{\infty} u(t) \cdot e^{-j2\pi ft}\, dt$$

But unit step function is present only for $t \geq 0$.

$$\therefore \quad F[u(t)] = \int_{0}^{\infty} 1 \cdot e^{-j2\pi ft}\, dt = \frac{1}{-j2\pi f}\left[e^{-j2\pi ft}\right]_0^{\infty}$$

$$= \frac{1}{-j2\pi f}\left[e^{-\infty} - e^{0}\right]$$

$$\therefore \quad F[u(t)] = \frac{1}{-j2\pi f}[0-1] = \frac{1}{j2\pi f}$$

$$\therefore \quad u(t) \overset{F}{\leftrightarrow} \frac{1}{j2\pi f} \qquad \text{...Ans.}$$

Q. 7 Obtain the fourier transform of a rectangular pulse of duration T and amplitude A as shown in Fig. 2.3(a). **Dec. 10**

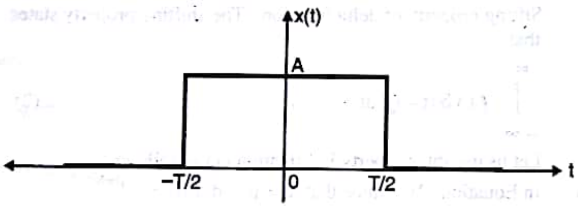

Fig. 2.3(a) : Rectangular pulse

Ans. : The rectangular pulse shown in Fig. 2.3(a) can be expressed mathematically as,

$$\text{rect}(t/T) = A \qquad \text{for } -T/2 \leq t \leq T/2$$
$$= 0 \qquad \text{elsewhere}$$

This is also known as the gate function. Therefore the fourier transform will be,

$$F[x(t)] = X(f) = \int_{-\infty}^{\infty} x(t) \cdot e^{-j2\pi ft}\, dt \qquad \text{...by definition of F. T.}$$

$$= \int_{-T/2}^{T/2} A\, e^{-j2\pi ft}\, dt$$

$$= \frac{A}{-j2\pi f}\left[e^{-j2\pi ft}\right]_{-T/2}^{T/2}$$

$$= \frac{A}{-j2\pi f}\left[e^{-j\pi fT} - e^{j\pi fT}\right]$$

$$= \frac{A}{j2\pi f}\left[e^{j\pi fT} - e^{-j\pi fT}\right]$$

$$= \frac{A}{\pi f}\left[\frac{e^{j\pi fT} - e^{-j\pi fT}}{2j}\right] \qquad \text{...(1)}$$

As per the Euler's theorem, $\sin\theta = \dfrac{e^{j\theta} - e^{-j\theta}}{2j}$

Applying this to Equation (1),

$$\text{we get, } F[x(t)] = \frac{A}{\pi f}[\sin(\pi fT)] \qquad \text{...(2)}$$

Multiply and divide the RHS of Equation (2) by T to get,

$$F[x(t)] = AT\,\frac{\sin(\pi fT)}{\pi fT} \qquad \text{...(3)}$$

In the above equation,

$$\frac{\sin(\pi fT)}{\pi fT} = \text{sinc}(fT) \qquad \text{...(4)}$$

$$\therefore \quad F[x(t)] = AT\,\text{sinc}(fT)$$

$$\therefore \quad A\,\text{rect}(t/T) \overset{F}{\leftrightarrow} AT\,\text{sinc}(fT)$$

Thus the rectangular pulse transforms into a sinc function.

Amplitude spectrum

The amplitude spectrum of the rectangular function is as shown in Fig. 2.3(b).

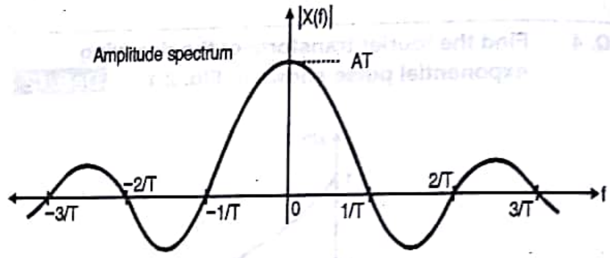

Fig. 2. 3(b) : Amplitude spectrum of a rectangular pulse

As we already know,

$$\text{sinc}(0) = 1 \qquad \therefore \quad AT\,\text{sinc}(0) = AT$$

The sinc function will have zero value for the following values of "fT":

$$\text{Sinc}(fT) = 0 \qquad \text{for } fT = \pm 1, \pm 2, \pm 3, \ldots\ldots$$

$$\text{i.e. for } f = \pm\frac{1}{T}, \pm\frac{2}{T}, \pm\frac{3}{T} \ldots\ldots$$

The phase spectrum has not been shown as it has zero value for all the values of f.

To absorb negative values of | X (f) | in the phase shift

The negative amplitude of the amplitude spectrum | X (f) | can be made positive by introducing a phase shift of ± 180° in the phase spectrum. This is as shown in Fig. 2.3(c). A negative phase shift for positive frequency and positive phase shift for the negative frequency is introduced in order to maintain symmetry of the phase spectrum.

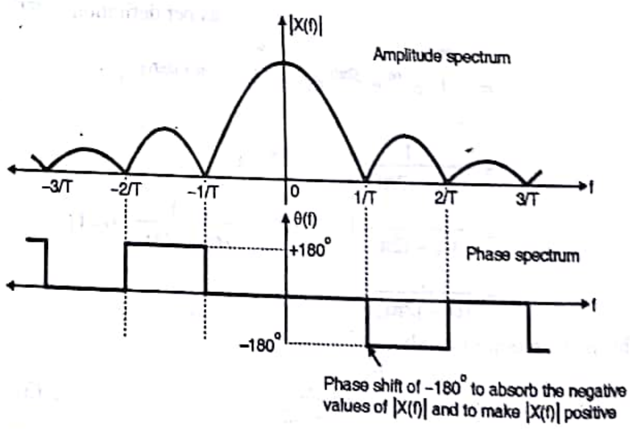

Fig. 2.3(c) : Amplitude and phase spectrums for a rectangular pulse. Negative values of | X (f) | have been absorbed in the additional phase shift of ± 180° in the phase spectrum

Q. 8 State and prove the properties of Fourier transform : Time shifting. May 09, May 10, Dec. 10, Dec. 13, Dec. 15, May 16

Ans. : The time shifting property states that if x (t) and X (f) form a Fourier transform pair then,

$$x(t-t_d) \overset{F}{\longleftrightarrow} e^{-j2\pi f t_d} X(f) \qquad ...(1)$$

Here the signal $x(t-t_d)$ is a time shifted signal. It is the same signal x (t) only shifted in time.

Proof :

$$F[x(t-t_d)] = \int_{-\infty}^{\infty} x(t-t_d) \cdot e^{-j2\pi f t} dt \qquad ...(2)$$

$$\text{Let } (t-t_d) = \tau,$$
$$\therefore \quad t = t_d + \tau$$
$$\therefore \quad dt = d\tau.$$

Substituting these values in Equation (1) we get,

$$F[x(t-t_d)] = \int_{-\infty}^{\infty} x(\tau) \cdot e^{-j2\pi f (t_d+\tau)} d\tau$$

$$= e^{-j2\pi f t_d} \int_{-\infty}^{\infty} x(\tau) e^{-j2\pi f \tau} d\tau$$

$$\therefore \quad F[x(t-t_d)] = e^{-j2\pi f t_d} X(f) \qquad \text{...Proved.}$$

This shows that the time shifting does not have any effect on the amplitude spectrum, but there is an additional phase shift of $-2\pi f t_d$, which is denoted by the term $e^{-j2\pi f t_d}$.

Significance of time shifting in communication systems

If signal x(t) is transmitted by a transmitter, then due to the distance travelled, this signal becomes a time delayed signal $x(t-t_d)$ when it reaches the receiver.

The time delay "t_d" is dependent on the distance between the transmitter and the receiver.

The time shifting property explains the effect of such time shifting on the spectrum of the signal. It tells us that there is no effect of time shifting on the amplitude spectrum but there is an additional phase shift of $-2\pi f t_d$.

Q. 9 State and prove the differentiation in time domain property of the Fourier Transform. Dec. 15, Dec. 16

Ans. : Some processing techniques involve differentiation and integration of the signal x(t). **This property is applicable if and only if the derivative of x (t) is Fourier transformable.**

Statement : Let $x(t) \overset{F}{\longleftrightarrow} X(f)$ and let the derivative of x(t) be Fourier transformable. Then,

$$\frac{d}{dt} x(t) \overset{F}{\longleftrightarrow} j2\pi f X(f) \qquad ...(1)$$

Proof : By the definition of inverse Fourier transform,

$$x(t) = \int_{-\infty}^{\infty} X(f) e^{+j2\pi f t} df$$

Therefore

$$\frac{d}{dt} x(t) = \frac{d}{dt}\left[\int_{-\infty}^{\infty} X(f) e^{+j2\pi f t} df\right]$$

$$= \int_{-\infty}^{\infty} X(f) \left(\frac{d}{dt} e^{+j2\pi f t}\right) df$$

$$\frac{d}{dt} x(t) = \int_{-\infty}^{\infty} [X(f) \cdot j2\pi f] e^{+j2\pi f t} df$$

As per the definition of the inverse Fourier transform the term inside the square bracket must be the Fourier transform of $\frac{d}{dt} x(t)$.

$$\therefore \quad F\left[\frac{d}{dt} x(t)\right] = j2\pi f X(f)$$

$$\text{OR} \quad \frac{d}{dt} x(t) \overset{F}{\longleftrightarrow} j2\pi f X(f). \qquad \text{...Proved.}$$

Meaning :

Differentiating the signal in time domain is equivalent to multiplying its Fourier transform by (j2πf) in the frequency domain. Thus differentiation will enhance the high frequency components since $|j2\pi f X(f)| > |X(f)|$.

Q. 10 Prove time convolution property of Fourier transform. May 09, May 10, Dec. 10, May 12, Dec. 13, May 16

Ans. : This property states that the convolution of signals in the time domain will be transformed into the multiplication of their Fourier transforms in the frequency domain.

$$\text{i.e. } [x_1(t) * x_2(t)] \overset{F}{\longleftrightarrow} X_1(f) X_2(f) \qquad ...(1)$$

Proof : The convolution of the two signals in the time domain is defined as,

$$x_1(t) * x_2(t) = \int_{-\infty}^{\infty} x_1(\lambda) \cdot x_2(t-\lambda) d\lambda \qquad ...(2)$$

Taking the Fourier transform of the convolution.

$$F[x_1(t) * x_2(t)] = \int_{-\infty}^{\infty}\left[\int_{-\infty}^{\infty} x_1(\lambda) \cdot x_2(t-\lambda) d\lambda\right] e^{-j2\pi f t} dt \qquad ...(3)$$

Multiply and divide the RHS of the Equation (2) by $e^{-j2\pi f \lambda}$ to get,

$$F[x_1(t) * x_2(t)] = \int_{-\infty}^{\infty} x_1(\lambda) \cdot e^{-j2\pi f \lambda} d\lambda \cdot \int_{-\infty}^{\infty} x_2(t-\lambda) \cdot e^{-j2\pi f t} \cdot e^{j2\pi f \lambda} dt$$

$$= \int_{-\infty}^{\infty} x_1(\lambda) e^{-j2\pi f \lambda} d\lambda \int_{-\infty}^{\infty} x_2(t-\lambda) e^{-j2\pi f (t-\lambda)} dt \qquad ...(4)$$

Let $(t-\lambda) = m$ in Equation (2)

$$\therefore \quad F[x_1(t) * x_2(t)] = \int_{-\infty}^{\infty} x_1(\lambda) \cdot e^{-j2\pi f \lambda} d\lambda \int_{-\infty}^{\infty} x_2(m) \cdot e^{-j2\pi f m} dm$$

Using the definition of the Fourier transform to the RHS we get,

$$F[x_1(t) * x_2(t)] = X_1(f)\, X_2(f) \quad ...(5)$$

This is the required result.

Significance of convolution in a communication system:

Consider a communication system with input x(t), output y(t) and impulse response h(t), as shown in Fig. 2.4

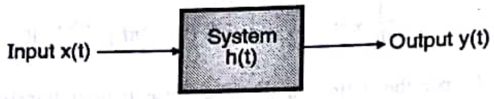

Fig. 2.4

The relation between x(t), h(t) and y(t) is as follows,

$$y(t) = x(t) * h(t) \quad ...(6)$$

That means y(t) is obtained by taking convolution of x(t) and h(t).

Method I :

Output of a system y(t) can be obtained by taking the convolution of input x(t) and impulse response of the system h (t).

$$y(t) = x(t) * h(t)$$

Method II :

The output y(t) can be obtained using the Fourier transform.

Let $y(t) \overset{F}{\longleftrightarrow} Y(f),\ x(t) \overset{F}{\longleftrightarrow} X(f)$ and $h(t) \overset{F}{\longleftrightarrow} H(f)$.

Then taking the F.T. of Equation (16) we get,

$$Y(f) = X(f) \cdot H(f)$$

Now take IFT to get y (t) = IFT [Y (f)]

Multiplication and taking IFT is simpler than obtaining the convolution. Hence in practice we can use this method to obtain y (t) i.e. output of a system.

$$F\{Im[x(t)]\} = \frac{1}{2j}[x(f) - X^*(-f)] \quad ...(7)$$

A signal cannot be band limited and time limited simultaneously

To justify this statement let us first consider a time limited signal and its Fourier transform. A rectangular pulse of duration T and amplitude A is an excellent example of a time limited signal.

We know that its Fourier transform is a sinc pulse which extends from $f = -\infty$ to $f = +\infty$ even though the amplitude of the sinc function goes on decreasing with increase in frequency.

Thus a time limited signal results into an unlimited spectrum in frequency domain. Now consider a rectangular pulse of width "f" in the frequency domain. This is a band limited signal. Due to the principle of duality, the inverse Fourier transform of this rectangular pulse is a sinc function in time domain which extends from $t = -\infty$ to $t = +\infty$.

Thus band limiting in the frequency domain results in a "time unlimited" signal in the time domain. Thus a signal cannot be band limited or time limited simultaneously.

Q. 11 Obtain the fourier transform of a cosine wave having a frequency f_o and peak amplitude of unity and plot its spectrum. Refer Fig. 2.5(a).

Dec. 09

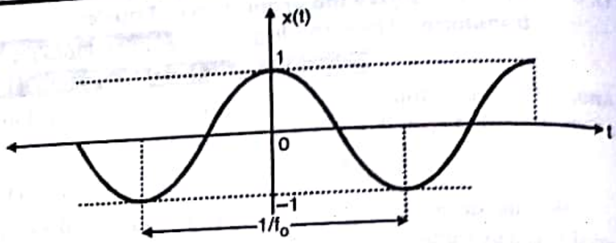

Fig. 2.5(a)

Ans. :

A cosine wave can be mathematically represented as,

$$x(t) = A\cos(2\pi f_o t)$$

But A = 1 $\quad \therefore x(t) = \cos(2\pi f_o t) \quad ...(1)$

By Euler's identity we can write,

$$\therefore x(t) = \cos(2\pi f_o t) = \frac{e^{j2\pi f_o t} + e^{-j2\pi f_o t}}{2} \quad ...(2)$$

The fourier transform of x (t) is given by,

$$X(f) = \int_{-\infty}^{\infty} x(t) \cdot e^{-j2\pi f t}\, dt$$

Substituting the value of x (t) we get,

$$X(f) = \int_{-\infty}^{\infty} \left[\frac{e^{j2\pi f_o t} + e^{-j2\pi f_o t}}{2}\right] e^{-j2\pi f t}\, dt$$

$$= \frac{1}{2}\int_{-\infty}^{\infty} [e^{-j2\pi(f-f_o)t} + e^{-j2\pi(f+f_o)t}]\, dt$$

$$\therefore X(f) = \frac{1}{2}\int_{-\infty}^{\infty} e^{-j2\pi(f-f_o)t}\, dt + \frac{1}{2}\int_{-\infty}^{\infty} e^{-j2\pi(f+f_o)t}\, dt \quad ...(3)$$

In which we have found the fourier transform of a dc signal. In that example we have proved that,

$$\int_{-\infty}^{\infty} e^{-j2\pi f t}\, dt = \delta(f) \quad ...(4)$$

Using this property for the RHS of Equation (3) we get,

$$X(f) = \frac{1}{2}\delta(f - f_o) + \frac{1}{2}\delta(f + f_o) \quad \textbf{...Ans.}$$

The frequency spectrum is as shown in the Fig. 2.5(b) which shows that two impulses are present one at f_o and the other at $-f_o$.

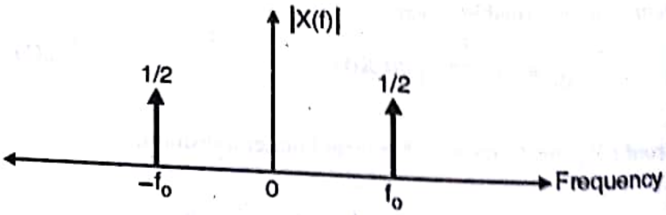

Fig. 2.5(b) : Spectrum of a cosine wave

Q. 12 Find the fourier transform of a sinewave having frequency of f_o and peak amplitude of unity. Also plot its frequency spectrum. May 15

Ans. :

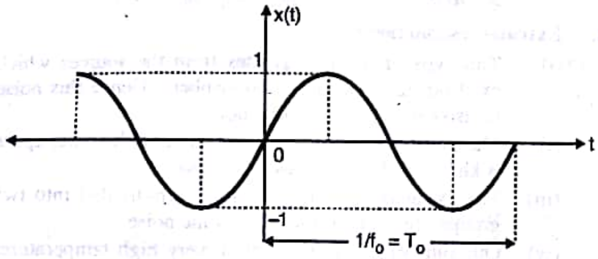

Fig. 2.6(a)

This example is identical. Express the signal x (t) as,

$$x(t) = \sin(2\pi f_o t) \quad ...(1)$$

Using Euler's identity express x (t) as follows :

$$x(t) = \sin(2\pi f_o t) = \frac{e^{j2\pi f_o t} - e^{-j2\pi f_o t}}{2j} \quad ...(2)$$

The fourier transform of x (t) is given by,

$$X(f) = \int_{-\infty}^{\infty} \sin(2\pi f_o t)\, e^{-j2\pi f t}\, dt$$

Substituting the value of sin $(2\pi f_o t)$ we get,

$$X(f) = \int_{-\infty}^{\infty} \frac{1}{2j}\left[e^{-j2\pi f_o t} - e^{-j2\pi f_o t}\right] e^{-j2\pi f t}\, dt$$

$$= \frac{1}{2j}\int_{-\infty}^{\infty} e^{-j2\pi(f-f_o)t}\, dt - \frac{1}{2j}\int_{-\infty}^{\infty} e^{-j2\pi(f+f_o)t}\, dt \quad ...(3)$$

We know that,

$$\int_{-\infty}^{\infty} e^{-j2\pi f t}\, dt = \delta(f)$$

Hence Equation (3) can be written as follows :

$$X(f) = \frac{1}{2j}\left[\delta(f-f_o) - \delta(f+f_o)\right] \quad ...(4)$$

The amplitude spectrum of sin $(2\pi f_o t)$ is represented as,

$$|X(f)| = \frac{1}{2}\left[\delta(f-f_o) - \delta(f+f_o)\right] \quad ...(5)$$

The term $\delta(f - f_0)$ is a delta function in the frequency domain which is shifted right by frequency "f_0". Its amplitude is + 1/2 and the term $\delta(f + f_o)$ represents a delta function which is shifted left by frequency "f_o". Its amplitude is – 1/2.

Therefore the spectrum is as shown in Fig.2.6(b).

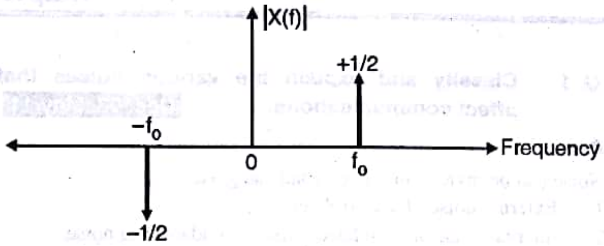

Fig. 2.6(b) : Spectrum of a sine wave

Q. 13 State and prove the convolution property of the Fourier transform and find the Fourier transform of the following signal : $y(t) = e^{-at}\, u(t) * u(t)$ Dec. 11

Ans. :

$$y(t) = e^{-at}\, u(t) * u(t)$$

Step 1 : Fourier transform of e^{-at} u(t)

Let $x_1(t) = e^{-at}\, u(t)$

$$\therefore X_1(f) = \int_0^{\infty} e^{-at}\, e^{-j2\pi f t}\, dt$$

$$= \int_0^{\infty} e^{-(a+j2\pi f)t}\, dt$$

$$= \frac{-1}{(a+j2\pi f)}\left[e^{-(a+j2\pi f)t}\right]_0^{\infty}$$

$$\therefore X_1(f) = \frac{-1}{(a+j2\pi f)}\left[e^{-\infty} - e^0\right]$$

$$\therefore X_1(f) = \frac{1}{(a+j2\pi f)} \quad ...(1)$$

Step 2 : F.T. of u (t)

Let $x_2(t) = u(t)$

$$\therefore X_2(f) = \int_0^{\infty} 1 e^{-j2\pi f t}\, dt$$

$$= \frac{-1}{j2\pi f}\left[e^{-j2\pi f t}\right]_0^{\infty}$$

$$\therefore X_2(f) = \frac{-1}{j2\pi f}\left[e^{-\infty} - e^0\right] = \frac{1}{(j2\pi f)} \quad ...(2)$$

Step 3 : FT of y (t)

$$Y(f) = F\left[e^{-at}\, u(t) * u(t)\right] = X_1(f)\, X_2(f)$$

$$Y(f) = \frac{1}{(a+j2\pi f)} \times \frac{1}{(j2\pi f)} \quad ...Ans.$$

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Chapter 3 : Noise

Q. 1 Classify and explain the various noises that affect communications. Dec. 03, May 11

Ans. :

Noise can be divided into two broad categories :

1. External noise of uncorrelated noise.
2. Internal noise or correlated noise or fundamental noise.

The classification of noise sources is shown in Fig. 3.1

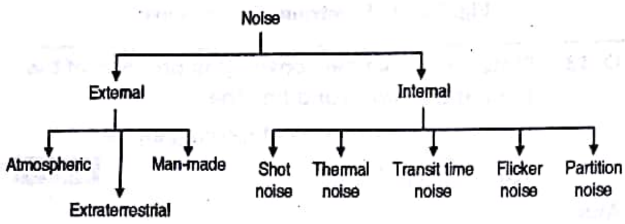


Fig. 3.1

External Noise (Uncorrelated Noise)

It is defined as the noise that is generated outside the device or circuit. As shown in Fig. 3.1., the external noise can be of three types :

1. Atmospheric noise
2. Extraterrestrial and
3. Man made noise

Fundamental or Internal Noise :

The fundamental sources of noise are within the electronic equipment. They are called fundamental sources because they are the integral part of the physical nature of the material used for making electronic components. This type of noise follows certain rules. Therefore it can be eliminated by properly designing the electronic circuits and equipments.

Q. 2 Classify and explain the various noises that affect communications. May 11

Ans. :

The fundamental sources of noise are within the electronic equipment. They are called fundamental sources because they are the integral part of the physical nature of the material used for making electronic components. This type of noise follows certain rules. Therefore, it can be eliminated by properly designing the electronic circuits and equipments.

It is defined as the noise that is generated outside the device or circuit. As shown in Fig. 3.1., the external noise can be of three types :

1. Atmospheric noise
2. Extraterrestrial and
3. Man made noise

1. Atmospheric noise

(i) This type of noise gets produced within the Earth's atmosphere. The common source of this type of noise is lightning. This type of noise is in the form of impulses or spikes which covers a wide frequency band typically upto 30 MHz.

(ii) The sputtering, cracking etc heard from the loud speakers of radio is due to atmospheric noise.

(iii) This type of noise becomes insignificant above 30 MHz.

2. Extraterrestrial noise

(i) This type of noise originates from the sources which exist outside the Earth's atmosphere. Hence this noise is also called as deep space noise.

(ii) The noise originating from the sun and the outer space is known as **Extraterrestrial Noise.**

(iii) The extraterrestrial noise can be sub-divided into two groups : (a) Solar noise (b) Cosmic noise.

(iv) Our sun being a large body at very high temperatures radiates a lot of noise. The noise radiation from sun varies with the temperature changes on its surface.

(v) The temperature changes follow a cycle of 11 years hence the cycle of great electrical disturbances (noise) also repeats after every 11 years.

(vi) The cosmic noise comes from the stars. This is identical to the noise radiated by sun because stars also are large hot bodies.

(vii) This noise is called as black body noise or thermal noise and it is distributed uniformly over the entire sky. The noise also gets originated from the center of our galaxy, other galaxies and special type of stars such as "Quasars" and "Pulsars".

3. Man made noise (Industrial noise)

The man made noise is generated due to the make and break process in a current carrying circuit. The examples are the electrical motors, welding machines, ignition system of the automobiles, thyristorised high current circuits, fluorescent lights, switching gears etc. This type of noise is also called as industrial noise.

Q. 3 What are the types of Internal Nosie ? May 15

Ans. : The fundamental noise sources produce different types of noise. They are as follows:

1. Thermal noise
2. Shot noise
3. Partition noise
4. Low frequency or flicker, noise.
5. High frequency or transit time, noise.
6. Avalanche noise.
7. Burst noise.

Let us know them one by one.

1. Shot Noise

(i) The shot noise is produced due to shot effect. Due to the shot effect, shot noise is produced in all the amplifying devices or for that matter in all the active devices.

(ii) The shot noise is produced due to the random variations in the arrival of electrons (or holes) at the output electrode of an amplifying device.

(iii) Therefore it appears as a randomly varying noise current superimposed on the output. The shot noise "sounds" like a shower of lead shots falling on a metal sheet if amplified and passed through a loud speaker.

(iv) The shot noise has a uniform spectral density like thermal noise. The exact formula for the shot noise can be obtained only for diodes.

(v) For all other devices an approximate equation is stated. The mean square shot noise current for a diode is given as :

$$I_n^2 = 2\, I_{dc}\, q\, B \text{ Amperes}^2 \quad ...(1)$$

where, I_{dc} = Direct current across the junction (in Amp)

q = Electron charge $= 1.6 \times 10^{-19}$ C.

B = Effective noise bandwidth in Hz.

For the amplifying devices the shot noise is :

1. Inversely proportional to the transconductance of the device.
2. Directly proportional to the output current.

2. Partition Noise

Partition noise is generated when the current gets divided between two or more paths. It is generated due to the random fluctuations in the division. Therefore the partition noise in a transistor will be higher than that in a diode. The devices like gallium arsenide FET draw almost zero gate bias current, hence keeping the partition noise to its minimum value.

3. Low Frequency or Flicker Noise :

The flicker noise will appear at frequencies below a few kilohertz. It is sometimes called as "1/f" noise. In the semiconductor devices, the flicker noise is generated due to the fluctuations in the carrier density (i.e. density of electrons and holes).

These fluctuations in the carrier density will cause the fluctuations in the conductivity of the material. This will produce a fluctuating voltage drop when a direct current flows through a device. This fluctuating voltage is called as flicker noise voltage.

The mean square value of flicker noise voltage is proportional to the square of direct current flowing through the device.

4. Thermal Noise or Johnson Noise

The free electrons within a conductor are always in random motion. This random motion is due to the thermal energy received by them. The distribution of these free electrons within a conductor at a given instant of time is not uniform.

It is possible that an excess number of electrons may appear at one end or the other of the conductor. The average voltage resulting from this non-uniform distribution is zero but the average power is not zero.

As this power has appeared as a result of the thermal energy, it is called as the "thermal noise power".

The average thermal noise power is given by,

$$P_n = k\, T B \text{ Watts} \quad ...(2)$$

where, k = Boltzmann's constant $= 1.38 \times 10^{-23}$ Joules/Kelvin.

B = Bandwidth of the noise spectrum (Hz).

T = Temperature of the conductor, °Kelvin

Equation (2) indicates that a conductor operated at a finite temperature can work as a generator of electrical energy. The thermal noise power P_n is proportional to the noise BW and conductor temperature.

5. High Frequency or Transit Time Noise

If the time taken by an electron to travel from the emitter to the collector of a transistor becomes comparable to the time period of the signal which is being amplified then the transit time effect will take place.

This effect is observed at very high frequencies typically in the VHF range. Due to the transit time effect some of the carriers may diffuse back to the emitter.

This gives rise to an input admittance, the conductance component of which increases with frequency.

The very small currents induced in the input of the device by means of the random fluctuations in the output current will create random noise at high frequencies. Once this noise appears, it goes on increasing with frequency at a rate of 6 dB per octave.

6. Correlated Noise

This is the form of internal noise. It is present in the circuit if and only if the signal is present. This is how it is correlated with the signal and hence called as the correlated noise.

In other words this type of noise will be absent if the signal is absent. Correlated noise is produced due to the nonlinear amplification process.

The types of correlated noise are :

1. Harmonic distortion 2. Intermodulation distortion

Both these are basically the nonlinear distortions.

All the electronic circuits exhibit a nonlinear behaviour and hence produce nonlinear distortions.

7. Impulse Noise

This type of noise is in the form of spikes of high amplitude and short duration, in the total noise spectrum. The shapes of these noise pulse are undefined as shown in Fig. 3.2

This type of noise is produced by electromechanical relays and solenoids, electric motors, fluorescent lights, ignition systems of automobiles and lightning.

The impulse noise produces a sharp popping or crackling sound in the audio circuits and produces errors in the data communication circuits.

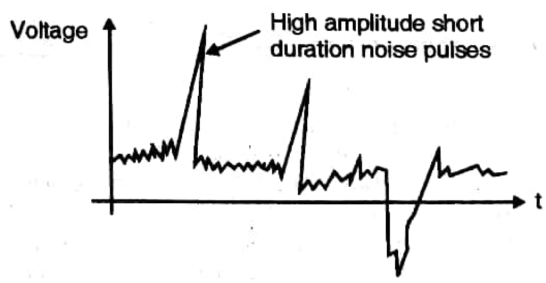


Fig. 3.2 : Impulse noise

8. Interference

This is a type of external noise. The meaning of the word "Interference" is "to disturb or detract from".

The interference is produced when information signal from one source produces frequencies that fall outside their allocated frequency band and interfere with the frequency band allocated to some other information signal as shown in Fig. 3.3. Interferences take place in the radio frequency spectrum when harmonics or cross-product frequencies from one source fall into the passband of the other source.

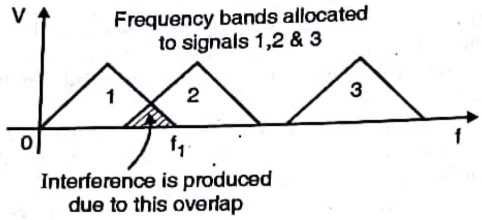


Fig. 3.3 : Interference

Q. 4 Explain the terms : Shot noise and Equivalent noise temperature.

Dec. 04, Dec. 13, May 15, Dec. 15

Ans.: The shot noise is produced due to shot effect. Due to the shot effect, shot noise is produced in all the amplifying devices or for that matter in all the active devices.

The shot noise is produced due to the random variations in the arrival of electrons (or holes) at the output electrode of an amplifying device.

Therefore it appears as a randomly varying noise current superimposed on the output. The shot noise "sounds" like a shower of lead shots falling on a metal sheet if amplified and passed through a loud speaker.

The shot noise has a uniform spectral density like thermal noise. The exact formula for the shot noise can be obtained only for diodes. For all other devices an approximate equation is stated. The mean square shot noise current for a diode is given as :

$$I_n^2 = 2 I_{dc} q B \text{ Amperes}^2 \quad ...(1)$$

where, I_{dc} = Direct current across the junction (in Amp)

q = Electron charge = 1.6×10^{-19} C.

B = Effective noise bandwidth in Hz.

For the amplifying devices the shot noise is :

1. Inversely proportional to the trans conductance of the device.
2. Directly proportional to the output current.

Q. 5 Write a short note on : Partition Noise. **May 15**

Ans. :

Partition noise is generated when the current gets divided between two or more paths. It is generated due to the random fluctuations in the division. Therefore, the partition noise in a transistor will be higher than that in a diode.

The devices like gallium arsenide FET draw almost zero gate bias current, hence keeping the partition noise to its minimum value.

Q. 6 Write a short note on : Low Frequency or Flicker Noise. **May 15**

Ans. : The flicker noise will appear at frequencies below a few kilohertz. It is sometimes called as "1/f" noise.

In the semiconductor devices, the flicker noise is generated due to the fluctuations in the carrier density (i.e. density of electrons and holes).

These fluctuations in the carrier density will cause the fluctuations in the conductivity of the material. This will produce a fluctuating voltage drop when a direct current flows through a device. This fluctuating voltage is called as flicker noise voltage.

The mean square value of flicker noise voltage is proportional to the square of direct current flowing through the device.

Q. 7 Explain the term thermal noise.

Dec. 03, May 04, Dec. 07, May 15, May 16

Ans. : The free electrons within a conductor are always in random motion. This random motion is due to the thermal energy received by them. The distribution of these free electrons within a conductor at a given instant of time is not uniform. It is possible that an excess number of electrons may appear at one end or the other of the conductor.

The average voltage resulting from this non-uniform distribution is zero but the average power is not zero.

As this power has appeared as a result of the thermal energy, it is called as the "thermal noise power".

The average thermal noise power is given by,

$$P_n = k\,TB \text{ Watts} \quad ...(1)$$

where, k = Boltzmann's constant

$= 1.38 \times 10^{-23}$ Joules/Kelvin.

B = Bandwidth of the noise spectrum (Hz).

T = Temperature of the conductor, °Kelvin

Equation (1) indicates that a conductor operated at a finite temperature can work as a generator of electrical energy.

The thermal noise power P_n is proportional to the noise BW and conductor temperature.

Q. 8 Write a short note on : High Frequency or Transit Time Noise **May 15**

Ans.: If the time taken by an electron to travel from the emitter to the collector of a transistor becomes comparable to the time period of the signal which is being amplified then the transit time effect will take place.

This effect is observed at very high frequencies typically in the VHF range. Due to the transit time effect some of the carriers may diffuse back to the emitter. This gives rise to an input admittance, the conductance component of which increases with frequency.

The very small currents induced in the input of the device by means of the random fluctuations in the output current will create random noise at high frequencies. Once this noise appears, it goes on increasing with frequency at a rate of 6 dB per octave.

Q. 9 An amplifier has a bandwidth of 4 MHz with 10 kΩ as the input resistor. Calculate the rms noise voltage at the input to this amplifier if the room temperature is 25ºC. **Dec.16**

Ans. : Given : $B = 4$ MHz, $R_i = 10\ k\Omega$, $T = 25°C = 298°$ K

Rms noise voltage $V_n = \sqrt{4\,kTBR}$

$= \sqrt{4 \times 1.38 \times 10^{-23} \times 298 \times 4 \times 10^6 \times 10 \times 10^3}$

$V_n = 25.65\ \mu V$

Q. 10 Write a short note on : Correlated Noise. **May 15**

Ans. : This is the form of internal noise. It is present in the circuit if and only if the signal is present. This is how it is correlated with the signal and hence called as the correlated noise. In other words this type of noise will be absent if the signal is absent. Correlated noise is produced due to the nonlinear amplification process.

The types of correlated noise are :

1. Harmonic distortion
2. Intermodulation distortion

Both these are basically the nonlinear distortions.

All the electronic circuits exhibit a nonlinear behaviour and hence produce nonlinear distortions.

1. Harmonic distortion

(i) The dynamic characteristics of a transistor as shown in Fig. 3.4 is non-linear in nature. Due to this non-linearity, the output waveform of an amplifier will be slightly different from the ac input waveform.

(ii) This type of distortion is known as non-linear distortion or harmonic distortion or amplitude distortion.

(iii) Harmonic distortion indicates the presence of those frequency components (harmonics) in the output of an amplifier which are not present on its input side.

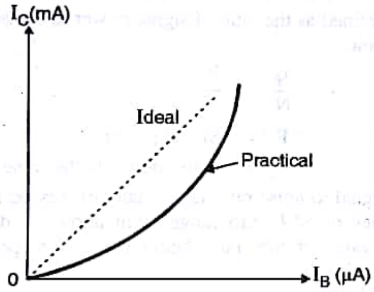


Fig. 3.4 : Non-linear dynamic characteristics of a BJT

(iv) The harmonic distortion suggests the presence of those frequency components in the amplifier output which were absent on the input side of the amplifier. The frequency component which has the same frequency of the input is known as the fundamental frequency component.

(v) The other frequency components in the output, which are integer multiples of fundamental frequency component are known as "Harmonics".

(vi) If the frequency of the fundamental component is say f_o Hz then $2\ f_o$, $3\ f_o$, $4\ f_o$ .... etc. are the harmonic components. $2\ f_o$ is called as second harmonic, $3\ f_o$ is called as third harmonic and so on. These harmonic components are as shown in Fig. 3.5.

(vii) If the input signal is a pure sine wave of frequency f_o then the output should consist of only the fundamental component i.e. f_o. All the other harmonic components are actually unwanted components.

(viii) When they get added to the fundamental component, the output waveform will get distorted as shown in Fig. 3.5. The shape of this distorted signal is decided by the amplitudes and relative phases of the harmonics that are added to the fundamental.

(ix) Usually the fundamental component has the highest amplitude. The amplitude of harmonic components decrease with increase in the order of harmonics. i.e. second harmonic has higher amplitude than all the other harmonics.

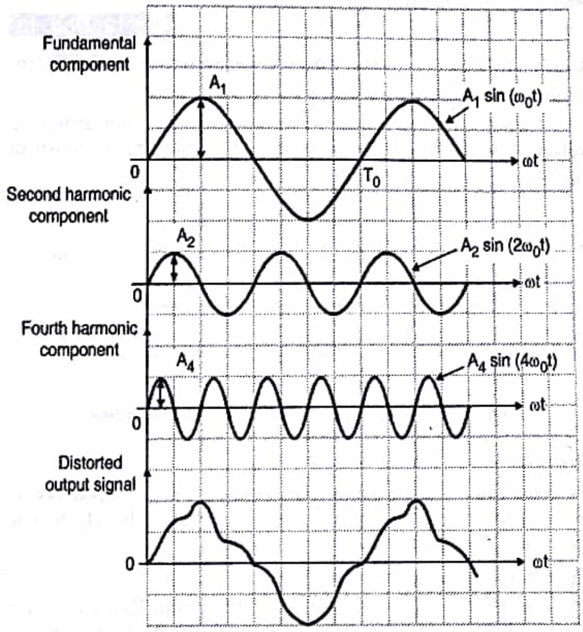


Fig. 3.5 : Different harmonic components

(x) Third harmonic amplitude is lower than that of the second harmonic component but higher than that of the rest of harmonic components and so on.

Let the amplitude of the fundamental component be A_1, the amplitude of second harmonic component be A_2... and the amplitude of n^{th} harmonic component be A_n. Then the percent harmonic distortion introduced by the n^{th} harmonic component is given by,

$$\%\ n^{th} \text{ harmonic distortion} = \%\ D_n = \frac{|A_n|}{|A_1|} \times 100 \quad ...(1)$$

Thus the second harmonic distortion is given by,

$$\%\ D_2 = \frac{|A_2|}{|A_1|} \times 100 \quad ...(2)$$

Similarly the third harmonic distortion is given by,

$$\%\ D_3 = \frac{|A_3|}{|A_1|} \times 100 \quad ...(3)$$

Total Harmonic Distortion (THD)

The output signal can get distorted due to presence various harmonic components simultaneously. The effective value of distortion due to various harmonic components is given by the Total Harmonic Distortion (THD) denoted by % D.

$$\%\ D = \sqrt{D_2^2 + D_3^2 + D_4^2 + ...} \times 100 \quad ...(4)$$

where, D = Total harmonic distortion

2. Intermodulation distortion

(i) When two or more signals of different frequencies are mixed in a nonlinear device then we get sum and difference frequencies of the original frequencies produced which are called as **cross products.**

(ii) The unwanted cross product frequencies can interfere with the message signal to degrade the system performance.

Q. 11 Write short notes on White noise.

May 12, Dec. 13

Ans. : The effect of "white" noise on the performance of different communication systems.

"White" noise is the noise whose power spectral density is uniform over the entire frequency range of interest, as shown in Fig. 3.6.

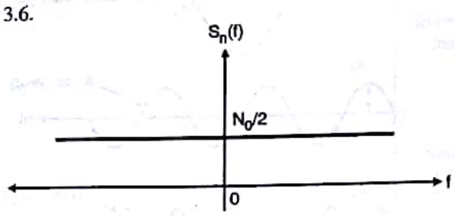


Fig. 3.6 : Power spectral density of white noise

Why is it called as white noise ?

The white noise contains all the frequency components in equal proportion. This is analogous with white light which is a superposition of all visible spectral components.

Why is it called as gaussian noise ?

The white noise has a Gaussian distribution. That means the PDF of white noise has the shape of Gaussian PDF. Therefore it is called as gaussian noise.

Power spectral density of white noise

As shown in Fig. 3.6 the power spectral density (psd) of a white noise is given by,

$$S_n(f) = \frac{N_0}{2} \qquad ...(1)$$

This equation shows that the power spectral density of white noise is independent of frequency. As N_0 is constant, the psd is uniform over the entire frequency range including the positive as well as the negative frequencies. N_0 in Equation (3.3.11) is defined as :

$$N_0 = KT_e \qquad ...(2)$$

where, K = Boltzmann's constant and

T_e = Equivalent noise temperature of the system.

Example of white noise

The best example of white noise is the thermal or Johnson noise.

Q. 12 Define the following term : Noise bandwidth.

Dec. 03, Dec. 12, May 15, May 16

Ans. : Assume that a white noise is present at the input of a receiver (filter). Let the filter have a transfer function H(f) as shown in Fig. 3.7.

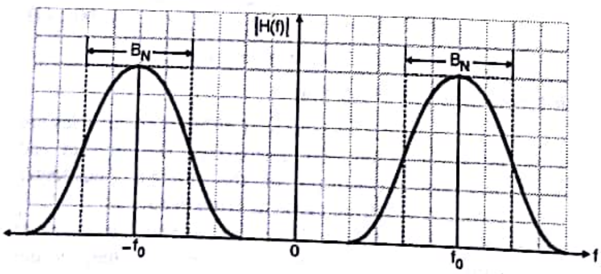


Fig. 3.7 : Noise bandwidth of a filter

This filter is being used to reduce the noise power actually passed on to the receiver. Now draw the frequency response of an ideal (rectangular) filter as shown by the dotted plot in Fig. 3.7. The center frequency of this ideal filter also is f_0.

Let the bandwidth "B_N" of the ideal filter be adjusted in such a way that the noise output power of the ideal filter is exactly equal to the noise output power of a real R-C filter.

Then B_N is called as the noise bandwidth of the real filter. Thus the noise bandwidth "B_N" is defined as the bandwidth of an ideal (rectangular) filter which passes the same noise power as passed by the real filter.

Q. 13 Define the following term : Signal to noise ratio.

Dec. 03, Dec. 04, May 06, May 09, Dec. 09, May 15, May 16

Ans. : In the communication systems the comparison of signal power with the noise power at the same point is important, to ensure that the noise at that point is not excessively large.

It is defined as the ratio of signal power to the noise power at the same point.

$$\therefore \frac{S}{N} = \frac{P_s}{P_n} \qquad ...(1)$$

Where, P_s = Signal power

P_n = Noise power at the same point.

The signal to noise ratio is normally expressed in dB and the typical values of S/N ratio range from about 10 dB to 90 dB. Higher the value of S/N ratio better the system performance in presence of noise.

$$S/N\ (dB) = 10 \log_{10} (P_s / P_n) \qquad ...(2)$$

The powers can be expressed in terms of signal and noise voltages as follows :

$$P_s = \frac{V_s^2}{R} \text{ and } P_n = \frac{V_n^2}{R}$$

where V_s = Signal voltage and V_n = Noise voltage.

$$\therefore \frac{S}{N} = \frac{V_s^2/R}{V_n^2/R} = \left(\frac{V_s}{V_n}\right)^2 \qquad ...(3)$$

The signal to noise ratio in dB is given by,

$$(S/N)_{dB} = 10 \log_{10}\left[\frac{V_s}{V_n}\right]^2 = 20 \log_{10}\left[\frac{V_s}{V_n}\right] \qquad ...(4)$$

All the possible efforts are made to keep the signal to noise ratio as high as possible, under all the operating conditions.

Although the signal to noise ratio is a fundamental characteristics of a communication system it is often difficult to measure. In practice instead of measuring S/N, another ratio called (S + N)/N is measured.

Q. 14 What is meant by signal to noise ratio ? Discuss the importance of SNR in radio receivers.

Dec. 04, May 06, May 15

Ans. : The given amplifier is shown in Fig. 3.8. Let us obtain the equivalent circuit for it.

The equivalent circuit is obtained as follows :

Remove the amplifier and calculate the open circuit voltage between points A and B. This is the Thevenin's voltage V_{oc} or V_{TH} and it is denoted by V_s. (See Fig. 3.9(a))

$$\therefore V_s = V_{TH} = \frac{R_i}{R_i + R_s} \times E_s \qquad ...(5)$$

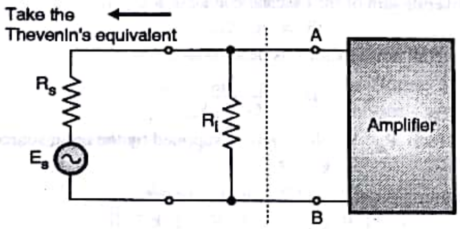
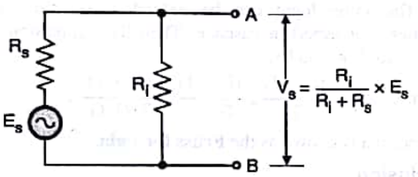
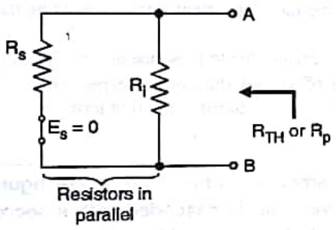


Fig. 3.8 : Given amplifier

Then calculate Thevenin's equivalent resistance R_{TH} or R_p, between A and B with E_s reduced to zero as shown in Fig. 3.9 (b).



(a) Calculation of Thevenin's voltage

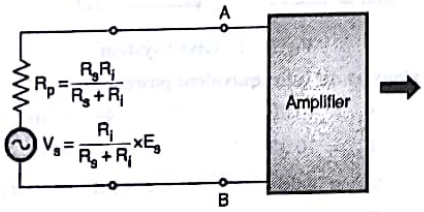


(b) Calculation of Thevenin's resistance

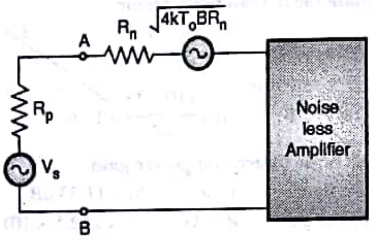
Fig. 3.9

From Fig. 3.9(b), $R_{TH} = R_p = R_s \parallel R_i = \frac{R_s R_i}{(R_s + R_i)}$ ...(6)

Hence the Thevenin's equivalent circuit of the amplifier is as shown in Fig. 3.9(c).



(c) Amplifier with Thevenin equivalent



(d) Final equivalent circuit

Fig. 3.9

Now complete the equivalent circuit as shown in Fig. 3.9 (d) by taking out the noise equivalent resistance R_n and the associated noise voltage, outside the amplifier.

Signal to noise ratio

The signal to noise ratio at the input is,

$$SNR_i = \frac{S_i}{N_i}$$

where S_i = Input signal power = V_s^2

and N_i = Input noise power = $V_n^2 = 4\,k\,T_o\,B\,(R_p + R_n)$

$$\therefore \; SNR_i = \frac{V_s^2}{4\,k\,T_o\,B\,(R_p + R_n)} \qquad ...(7)$$

Since the amplifier is noiseless, the SNR at its input and output will be the same.

Then the signal to noise ratio at the amplifier output is,

$$\frac{S}{N} = \frac{V_s^2}{4\,(R_p + R_n)\,k\,T_o\,B} \qquad ...(8)$$

Here $(R_p + R_n)$ represents the equivalent resistance producing noise at the input of the amplifier

Q. 15 Define the following term : Noise factor.

May 09, Dec. 10, May 12, Dec. 14, May 15

Ans. :

The noise factor (F) of an amplifier or any network is defined in terms of the signal to noise ratio at the input and the output of the system. It is defined as :

$$F = \frac{\text{S/N ratio at the input}}{\text{S/N ratio at the output}} \qquad ...(1)$$

$$= \frac{P_{si}}{P_{ni}} \times \frac{P_{no}}{P_{so}} \qquad ...(2)$$

where P_{si} and P_{ni} = Signal and noise power at the input

and P_{so} and P_{no} = Signal and noise power at the output.

The temperature to calculate the noise power is assumed to be the room temperature. The S/N at the input will always be greater than that at the output. This is due to the noise added by the amplifier. Therefore the noise factor is the means to measure the amount of noise added and it will always be greater than one. The ideal value of the noise factor is unity.

The noise factor F is sometimes frequency dependent. Then its value determined at one frequency is known as the spot noise factor and the frequency must be stated along with the spot noise factor.

Q. 16 Define the following term : Noise figure.

Dec. 03, May 09, May 10, Dec. 10, Dec. 14, May 16

Ans. :

Sometimes the noise factor is expressed in decibels. When noise factor is expressed in decibels it is called noise figure.

$$\text{Noise figure} = F_{dB} = 10 \log_{10} F \qquad ...(1)$$

Substituting the expression for the noise factor we get,

$$\text{Noise figure} = 10 \log_{10} \left[\frac{\text{S/N at the input}}{\text{S/N at the output}} \right]$$

$$\therefore \; F_{dB} = 10 \log_{10} (S/N)_i - 10 \log_{10} (S/N)_o$$

$$\therefore \; \text{Noise figure } F_{dB} = (S/N)_i \text{ dB} - (S/N)_o \text{ dB} \qquad ...(2)$$

The ideal value of noise figure is 0 dB.

Measure to improve the noise figure

To improve the noise figure of a receiver, the devices used for the amplifiers and mixer stages must produce low noise.

The diodes and FETs are therefore preferred. The receiver can be operated at low temperatures. This is done in satellite low noise receivers. Use high gain amplifiers. All this will collectively improve the noise figure of the receiver.

Q. 17 Define equivalent noise temperature.

May 03, Dec. 03, Dec. 04, May 08, May 09, May 12, Dec. 12, Dec. 13, Dec. 15, May 16

Ans. : The noise factor or noise figure can not always be used for measuring noise. Another way to represent the noise is by means of the equivalent noise temperature.

The equivalent noise temperature is used in dealing with the UHF and microwave low noise antennas, receivers or devices.

Definition : The equivalent noise temperature of a system is defined as the temperature at which the noisy resistor has to be maintained so that by connecting this resistor to the input of a noiseless version of the system, it will produce the same amount of noise power at the system output as that produced by the actual system.

Q. 18 Write short notes on : Friiss formula.

May 10, Dec. 10, May 12, Dec. 13, May 14, May 15, Dec. 15, Dec. 16

Ans. : In practice the filters or amplifiers are not used in isolated manner. They are used in the cascaded manner.

So in this section let us see the effect of cascading on the noise factor and noise temperature. The overall noise factor of such cascade connection can be determined as follows : Fig. 3.10 shows two amplifiers connected in cascade.

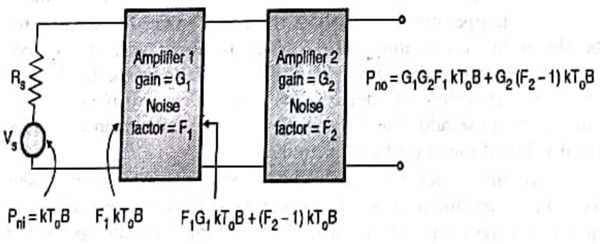


Fig. 3.10 : Two amplifiers connected in cascade

Let the power gains of the two amplifiers be G_1 and G_2 respectively and let their noise factors be F_1 and F_2 respectively.

The total noise power at the input of the first amplifier is given as :

$$P_{ni\,(total)} = F_1 k T_o B \quad ...(1)$$

The total noise power at the output of amplifier 1 will be the addition of two terms.

$$\therefore \text{ Noise input to amplifier 2} = G_1 F_1 k T_o B + (F_2 - 1) kT_o B \quad ...(2)$$

The first term represents the amplified noise power (by G_1) and the second term represents the noise contributed by the second amplifier. The noise power at the output of the second amplifier is G_2 times the input noise power to amplifier 2.

$$\therefore P_{no} = G_2 \times (\text{Noise input to amplifier 2})$$

$$\therefore P_{no} = G_1 G_2 F_1 k T_o B + G_2 (F_2 - 1) k T_o B \quad ...(3)$$

The overall gain of the cascade connection is given by,

$$G = G_1 G_2 \quad ...(4)$$

The overall noise factor F is defined as follows :

$$F = \frac{P_{no}}{G_1 G_2 P_{ni}} \quad ...(5)$$

Here P_{ni} = Noise power supplied by the input source $= k T_o B$

Substituting the values of P_{no} and P_{ni} we get,

$$F = \frac{G_1 G_2 F_1 k T_o B + G_2 (F_2 - 1) k T_o B}{G_1 G_2 k T_o B}$$

$$\therefore F = F_1 + \frac{(F_2 - 1)}{G_1} \quad ...(6)$$

The same logic can be extended for more number of amplifiers connected in cascade. Then the expression for overall noise factor F would be,

$$F = F_1 + \frac{(F_2 - 1)}{G_1} + \frac{(F_3 - 1)}{G_1 G_2} + \frac{(F_4 - 1)}{G_1 G_2 G_3} + \ldots \quad ...(7)$$

This formula is known as the **Friiss formula.**

Conclusion

This equation indicates that in the cascade configuration, the first stage is the most important stage in deciding the overall noise factor.

This is because due to presence of terms G_1, $G_1 G_2$ ...... in the denominators of second, third terms respectively ....., they become negligibly small as compared to the first term.

$$\therefore F \approx F_1 \quad ...(8)$$

Q.19 An amplifier with 10 dB noise figure and 4 dB power gain is cascaded with a second amplifier which has a 10 dB power gain. What is the overall noise figure and power gain ?

May 16

Ans. : The given system is as shown in Fig. 3.11.

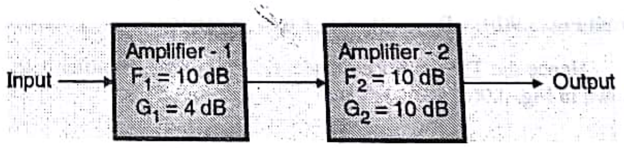


Fig. 3.11 : Given system

Step 1 : Convert dB into equivalent power ratios

F_1 = 10 dB $\quad \therefore F_1 = 10$

F_2 = 10 dB $\quad \therefore F_2 = 10$

G_1 = 4 dB $\quad \therefore$ 4 dB = 10 log G_1

$\therefore G_1 = 2.5$

Step 2 : Calculate the overall noise factor

$$F = F_1 + \frac{F_2 - 1}{G_1}$$

$$F = 10 + \frac{(10 - 1)}{2.5} = 13.6$$

Step 3 : Overall noise figure and power gain

Noise figure = 10 log (13.6) = 11.33 dB ...Ans.

Overall power gain = log (G_1 G_2) = log (2.5 × 10) = 1.39 dB ...Ans.

Q.20 A three stage amplifier has the following power gains and noise figure (as ratios, not in dB) for each stage :

Calculate the power gain, noise figure and the noise temperature for the entire amplifier, assuming matched conditions.

May 04, Dec. 07, Dec. 11

Stage	Power gain	Noise figure
1	10	2
2	20	4
3	30	5

Ans. :

Given : $G_1 = 10, G_2 = 20, G_3 = 30,$

$F_1 = 2, F_2 = 4, F_3 = 5.$

1. Draw the block diagram of the system

The block diagram of the cascaded amplifiers is as shown in Fig. 3.12.

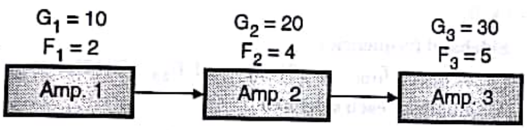


Fig. 3.12 : Cascaded amplifier system

2. Overall power gain

$$G = G_1 \times G_2 \times G_3$$

$$= 10 \times 20 \times 30 = 6000 \quad \text{...Ans.}$$

3. Overall noise figure

The overall noise factor is given by,

$$F = F_1 + \frac{(F_2 - 1)}{G_1} + \frac{(F_3 - 1)}{G_1 G_2}$$

$$\therefore F = 2 + \frac{(4-1)}{10} + \frac{(5-1)}{10 \times 20}$$

$$\therefore F = 2.32 \quad \text{...(1)}$$

$\therefore$ Overall noise figure F dB = 10 $\log_{10}$ (2.32)

$$= 3.654 \text{ dB} \quad \text{...Ans.}$$

4. Overall noise temperature

$$T_{eq} = (F-1)\,T_o$$

Let T_o = Room temperature = 300° K

$$\therefore T_{eq} = (2.32 - 1) \times 300$$

$$\therefore T_{eq} = 396°K \quad \text{...Ans.}$$

Q. 21 Explain the term thermal noise. Prove that noise voltage $V_N = \sqrt{4\,kTBR}$. For electronic device operating at a temperature of 17°C with a bandwidth of 10 kHz, **Dec 07, May 11**

Determine :

1. **Thermal noise power in dBm.**
2. **rms noise voltage for a 100 Ω internal resistance and a 100 Ω load resistance.**

Ans.: For thermal noise, refer Q.7. For the derivation of noise voltage refer Q.7.

Solution of example :

Given : T = 17° C = 290° K, B = 10 kHz

1. Thermal noise power :

$$P = kTB$$

$$= 1.28 \times 10^{-23} \times 290 \times 10 \times 10^3$$

$$= 4.002 \times 10^{-17}\ \text{W} \quad \text{...Ans.}$$

$$P_{(dBm)} = 10 \log_{10}\left(\frac{4.002 \times 10^{-17}}{1 \times 10^{-3}}\right)$$

$$= -133.98 \text{ dBm} \quad \text{...Ans.}$$

2. Rms noise voltage

Assume that the internal resistance of the device is in series with the load resistance.

$$\therefore R = 100 + 100$$

$$= 200\ \Omega$$

$$\therefore V_n = \sqrt{kTBR}$$

$$= \sqrt{4.002 \times 10^{-17} \times 200}$$

$$= 8.95 \times 10^{-8}\ \text{V} \quad \text{...Ans.}$$

Q. 22 For three cascaded amplifier stages, each with noise figure of 3dB and power gain of 10 dB, determine the total noise figure. **May 07**

Ans. :

$$F_1 = F_2 = F_3 = 3 \text{ dB},$$

$$G_1 = G_2 = G_3 = 10 \text{ dB}$$

Step 1 : Convert dB into equivalent power ratios

$F_1 = F_2 = F_3 = 3$ dB

$$\therefore 3 = 10 \log F_1$$

$$\therefore F_1 = 1.995 = F_2 = F_3$$

$G_1 = G_2 = G_3 = 10$ dB

$$\therefore 10 = 10 \log G_1$$

$$\therefore G_1 = 10$$

Step 2 : Calculate the overall noise factor

Using the Friiss formula,

$$F = F_1 + \frac{(F_2 - 1)}{G_1} + \frac{(F_3 - 1)}{G_1 G_2}$$

$$= 1.995 + \frac{(1.995 - 1)}{10} + \frac{(1.995 - 1)}{100}$$

$$= 1.995 + 0.0995 + 9.95 \times 10^{-3}$$

$$F = 2.0594$$

Step 3 : Calculate the overall noise figure

$$F_{dB} = 10 \log_{10} (2.0594)$$

$$\text{Noise figure} = 3.1374 \text{ dB} \quad \text{...Ans.}$$

Chapter 4 : Amplitude Modulation and Demodulation

Q. 1 A sinusoidal carrier has amplitude of 20 V and frequency 30 kHz. It is amplitude modulated by sinusoidal voltage of amplitude 3 V and frequency 2 kHz. Modulated voltage is developed across a 50 Ω resistance.

1. **Write the equation for modulated wave.**
2. **Plot the modulated wave showing maxima and minima of waveform.**
3. **Determine the modulation index.**
4. **Draw the spectrum of modulated wave.**

May 15

Ans. :

Given : $E_c = 20$ V, $E_m = 3$ V, $f_c = 30$ kHz, $f_m = 2$ kHz, $R_L = 50\ \Omega$

1. Modulation index,

$$m = \frac{E_m}{E_c} = \frac{3}{20} = 0.15 \qquad \text{...Ans.}$$

2. Equation for modulated wave :

$$\begin{aligned} s(t) &= E_c (1 + m \cdot \cos \omega_m t) \cos \omega_c t \\ &= 20 [1 + 0.15 \cos (2\pi \times 2 \times 10^3 t)] \\ &\quad \cos (2\pi \times 30 \times 10^3 t) \\ \therefore\ s(t) &= 20 [1 + 0.15 \cos (4\pi \times 10^3 t)] \cos (6\pi \times 10^4 t) \end{aligned}$$

...Ans.

3. The modulated waveform is as shown in Fig.4.1(a).
4. Spectrum of modulated wave :

The sideband frequencies are :

$$\begin{aligned} f_{USB} &= f_c + f_m \\ &= 30 + 2 = 32 \text{ kHz} \\ f_{LSB} &= f_c - f_m \\ &= 30 - 2 = 28 \text{ kHz} \end{aligned}$$

$$\text{Amplitude of each sideband} = \frac{m}{2} \times E_c$$

$$= \frac{0.15 \times 20}{2} = 1.5 \text{ V}$$

$$\begin{aligned} E_{max} &= E_c + E_m \\ &= 20 + 3 = 23 \text{ kHz} \\ E_{min} &= E_c - E_m \\ &= 20 - 3 = 17 \text{ kHz} \end{aligned}$$

The spectrum of AM wave is as shown in Fig. 4.1(b).

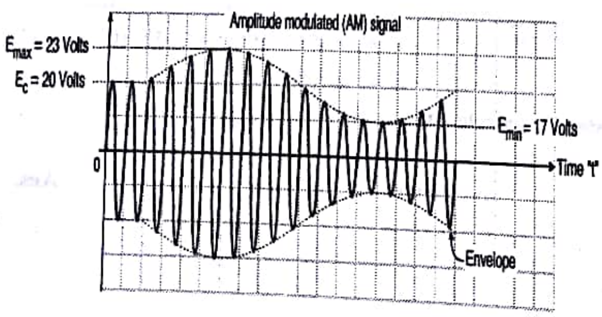


Fig. 4.1(a) : AM waveform

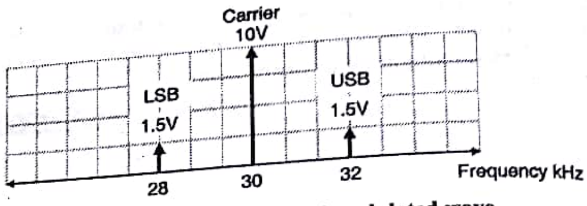


Fig. 4.1(b) : Spectrum of modulated wave

Q. 2 A sinusoidal carrier $V_c = 100 \cos (2\pi \cdot 10^5 t)$ is amplitude modulated by a sinusoidal voltage $V_m = 50 \cos (2\pi \cdot 10^3 t)$ upto a modulation depth of 50 %. Calculate the amplitude and frequency of each sideband and the rms voltage of the modulated carrier.

Dec. 11

Ans. :

Given : $E_c = 100$ V, $E_m = 50$ V, $m = 0.5$, $f_c = 100$ kHz, $f_m = 1$ kHz

1. **Sideband frequencies**

$$f_{USB} = 101 \text{ kHz and } f_{LSB} = 99 \text{ kHz}$$

2. **Amplitude of each sideband**

$$= \frac{m E_c}{2} = \frac{0.5 \times 100}{2} = 25 \text{ V}$$

3.

$$P_t = P_c \left(1 + \frac{m^2}{2}\right) \qquad \text{...(1)}$$

Let

$$P_t = \frac{(E_{AM})^2}{R}$$

where, E_{AM} = Rms value of modulated carrier

and

$$P_c = E_c^2 / 2R = \frac{(100)^2}{2R} = \frac{10^4}{2R}$$

Substituting these into Equation (1) we get,

$$\begin{aligned} \therefore\ \frac{E_{AM}^2}{R} &= \frac{10^4}{2R}\left(1 + \frac{0.5^2}{2}\right) \\ \therefore\ E_{AM}^2 &= 5625 \\ \therefore\ E_{AM} &= 75 \text{ Volts.} \end{aligned}$$

...Ans.

Q. 3 A sinusoidal carrier has an amplitude of 10 V and a frequency of 100 kHz. It is amplitude modulated by a sinusoidal voltage of amplitude 3 V and frequency 500 Hz. Modulated voltage is developed across a 75 Ω resistance.

1. **Write the equation for the modulated wave**
2. **Determine the modulation index.**
3. **Draw the spectrum of modulated wave.**
4. **Calculate the total average power.**
5. **Calculate the power carried by sidebands.**

Dec. 03, May 09, Dec. 14

Ans. :

Given : $E_c = 10$ V, $f_c = 100$ kHz, $E_m = 3$ V, $f_m = 500$ Hz, $R = 75\ \Omega$

1. Modulation index :

$$m = \frac{E_m}{E_c} = \frac{3}{10} = 0.3$$

2. Equation of modulated wave

$$e_{AM} = E_c\,(1 + m\cos\omega_m t)\cos 2\pi f_c t$$
$$= 10\,[1 + 0.3\cos(2\pi \times 500\,t)]\cos(2\pi \times 100 \times 10^3)$$
$$e_{AM} = 10\,[1 + 0.3\cos 1000\,\pi t]\cos(2\pi \times 10^3\,t) \quad \text{...Ans.}$$

3. Spectrum of AM wave

Various spectral components are as follows :

1. Carrier : $f_c = 100$ kHz
2. USB : $(f_c + f_m) = 100.05$ kHz
3. LSB : $(f_c - f_m) = 99.95$ kHz

Amplitudes of these spectral components are as follows :

1. Carrier :

$$E_c = 10\text{ V}$$

2. LSB and USB

$$mE_c / 2 = \frac{0.3 \times 10}{2} = 1.5\text{ V}$$

Spectrum is as shown in Fig. 4.2

4. Total average power

$$P_t = \left(1 + \frac{m^2}{2}\right) P_c$$
$$= \left(1 + \frac{m^2}{2}\right)\frac{E_c^2}{2R} = \left(1 + \frac{(0.3)^2}{2}\right)\frac{(10)^2}{2 \times 75}$$
$$P_t = 0.6967\text{ W} \quad \text{...Ans.}$$

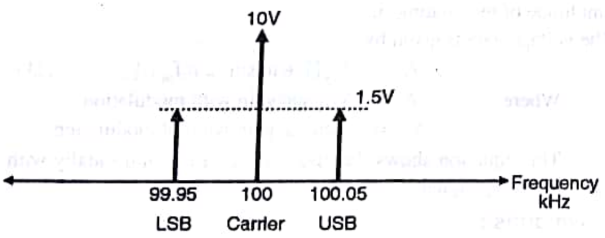


Fig. 4.2 : Spectrum of AM wave

5. Total sideband power

$$P_{SB} = \frac{m^2}{2} \times P_c$$
$$= \frac{(0.3)^2}{2} \times \frac{(10)^2}{2 \times 75} = 0.03\text{ W} \quad \text{...Ans.}$$

Q. 4 A sinusoidal carrier has an amplitude of 20 V and a frequency of 200 kHz. It is amplitude modulated by a sinusoidal voltage of amplitude 6 V and frequency 1 kHz. Modulated voltage is developed across a 80 Ω resistance.

1. Write the equation for the modulated wave.

2. Determine the modulation index.

3. Draw the spectrum of modulated wave.

4. Calculate the total average power. **Dec. 15**

Ans. :

Given : $E_c = 20$ V, $f_c = 200$ kHz, $E_m = 6$ V, $f_m = 1$ kHz, $R_L = 80\ \Omega$.

1. Modulation index,

$$m = \frac{E_m}{E_c} = \frac{6}{20} = 0.3 \quad \text{...Ans.}$$

2. Equation for modulated wave :

$$s(t) = E_c\,(1 + m \cdot \cos\omega_m t)\cos\omega_c t$$
$$= 20\,[1 + 0.3\cos(2\pi \times 1 \times 10^3\,t)]\cos(2\pi \times 200 \times 10^3\,t)$$
$$\therefore\ s(t) = 20\,[1 + 0.3\cos(2\pi \times 10^3\,t)]\cos(4\pi \times 10^5\,t) \quad \text{...Ans.}$$

3. The modulated waveform is as shown in Fig. 4.3(a).

4. Spectrum of modulated wave :

The sideband frequencies are :

$$f_{USB} = f_c + f_m = 200 + 1 = 201\text{ kHz}$$
$$f_{LSB} = f_c - f_m = 200 - 1 = 199\text{ kHz}$$

Amplitude of each sideband $= \frac{m}{2} \times E_c$

$$= \frac{0.3 \times 20}{2} = 3\text{ Volts}$$
$$E_{max} = E_c + E_m = 20 + 6 = 26\text{ V}$$
$$E_{min} = E_c - E_m = 20 - 6 = 14\text{ V}$$

The spectrum of AM wave is as shown in Fig. 4.3(b).

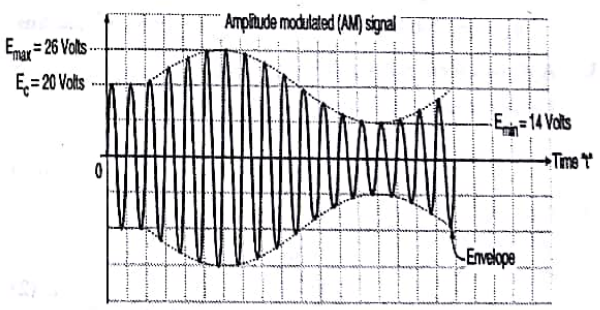


Fig. 4.3(a) : AM waveform

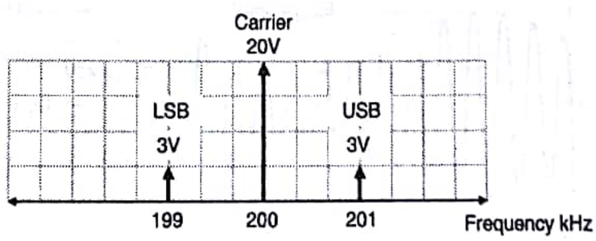


Fig. 4.3(b)

Total average power :

$$P_t = \left(1+\frac{m^2}{2}\right) P_c = \left(1+\frac{m^2}{2}\right)\cdot\frac{E_c^2}{2R}$$

$$\therefore \quad P_t = \left(1+\frac{0.3^2}{2}\right)\times\frac{20^2}{2\times 80}$$

$$\therefore \quad P_t = 2.6125 \text{ W} \quad \text{...Ans.}$$

Q. 5 The AM transmitter develops an unmodulated power output of 400 Watts across a 50 ohms resistive load. The carrier is modulated by a sinusoidal signal with a modulation index of 0.8. Assuming $f_m = 5$ kHz and $f_c = 1$ MHz :

1. **Obtain the value of carrier amplitude V_c and hence write the expression for AM signal.**
2. **Find the total sideband power.**
3. **Draw the AM wave for the given modulation index.** **Dec. 16**

Ans. : $P_c = 400$ W, $R_L = 50\ \Omega$, m = 0.8, $f_m = 5$ kHz, $f_c = 1$ MHz.

1. **Carrier amplitude V_c**

The carrier power $\quad P_c = \frac{V_c^2}{2R_L}$

$$\therefore \quad V_c = \sqrt{2R_L P_c} = \sqrt{2\times 50\times 400} = 200 \text{ Volts}$$

Thus peak carrier voltage is 200 V.

2. **Expression for AM wave**

$$e_{AM} = E_c\,[1 + m\cos(2\pi f_m t)]\cos(2\pi f_c t)$$

Substituting E_c = 200 V,

m = 0.8, f_m = 5 kHz, f_c = 1 MHz we get,

$$e_{AM} = 200\,[1 + 0.8\cos(2\pi\times 5\times 10^3 t)]\cos(2\pi\times 1\times 10^6 t)$$

$$\therefore e_{AM} = 200\,[1 + 0.8\cos(10^4 \pi t)]\cos(2\pi\times 10^6 t) \quad \text{...Ans.}$$

3. **AM waveform for m = 0.8**

Let us calculate E_m from E_c. Since $m = E_m / E_c$.

$$E_m = m\times E_c = 0.8\times 200 = 160 \text{ V} \quad \text{...(1)}$$

$$\therefore \quad E_{max} = E_c + E_m = 200 + 160 = 360 \text{ V}$$

$$\text{and} \quad E_{min} = E_c - E_m = 200 - 160 = 40 \text{ V} \quad \text{...(2)}$$

The AM wave for m = 0.8 is as shown in Fig.4.4.

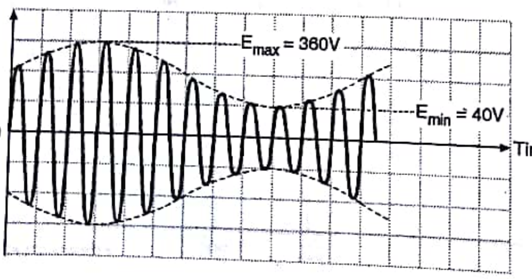


Fig. 4.4 : AM wave for m = 0.8

Q. 6 Explain in detail generation of any method used in AM. **May 15**

Ans. : Fig. 4.5 shows a circuit which is basically a small signal class A amplifier. It is called as an emitter modulator circuit and used as amplitude modulator.

The circuit has two inputs namely the RF carrier and the modulating signal. When the modulating signal is absent, only the carrier is applied, the circuit works only as a class A amplifier and we get amplified carrier at the output.

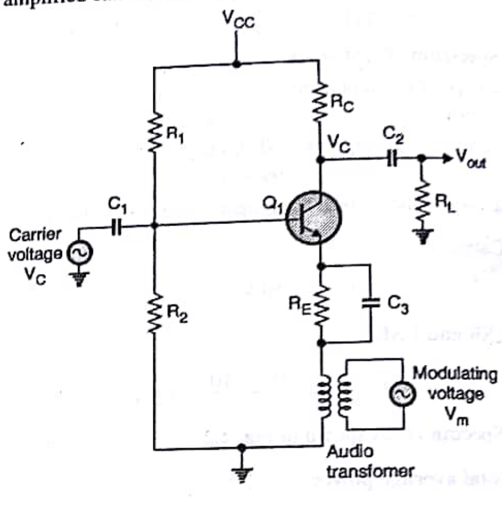


Fig. 4.5 : Single transistor emitter modulator

When the modulating signal is applied, the transistor operates as nonlinear device and multiplication of the carrier and modulating signal will take place. The gain of the amplifier is dependent on the modulating voltage. Hence the gain varies in a sinusoidal manner at the rate equal to the frequency of the modulating signal. The modulation index (m) is proportional to the amplitude of modulating signal.

The voltage gain is given by,

$$A = A_Q\,[1 + m\sin(2\pi f_m t)] \quad \text{...(1)}$$

Where $\quad A$ = Voltage gain with modulation

A_Q = Voltage gain without modulation

This equation shows that the gain A varies sinusoidally with the modulating signal.

Waveforms :

The waveforms for the emitter modulator are shown in Fig. 4.6.

The modulating signal applied at the emitter of transistor drives it into saturation and cut off. This ensures that the amplifier operation will be nonlinear as required for the modulation to take place.

If we analyze the voltage waveform at the collector, then it includes the following frequency components :

1. Carrier
2. Upper and lower sidebands
3. Modulating signal.

The unwanted modulating signal is removed by the coupling capacitor C_2 and we get the AM signal at the output (across R_L).

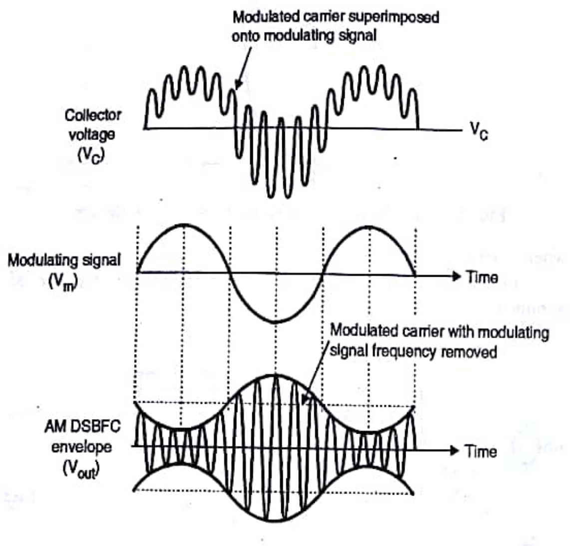


Fig. 4.6 : Waveforms of emitter modulator

Q. 7 Draw the block diagram of AM transmitter and explain each block in brief.

May 05, Dec. 06, Dec. 10, Dec. 13

Ans.: The block diagram of a low level AM transmitter is as shown in Fig. 4.7.

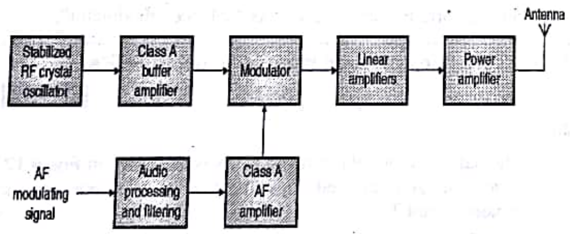


Fig. 4.7 : Low level AM transmitter

The RF oscillator produces the carrier signal. The RF oscillator is stabilized in order to maintain the carrier frequency deviation within a prescribed limit. The carrier frequency is equal to the transmitter frequency and it should remain very stable.

The amplified modulating signal is applied to the modulator along with the carrier. At the output of the modulator we get the AM wave. This AM signal is then amplified using a chain of linear amplifiers to raise its power level.

The linear amplifiers can be class A, AB or B type amplifiers. The linear amplifiers are used in order to avoid the waveform distortion in AM wave. However these amplifiers posses a low efficiency. The amplitude modulated signal is then transmitted using transmitting antenna.

The transistorized modulator circuits can be used for low level modulator due to the low power which is to be handled. The low level transmitter does not require a large AF modulator power so its design is simplified. However the overall efficiency is much lower compared to high level modulation. This is due to the use of less efficient linear amplifiers.

Q. 8 Write short note on : High level AM transmitter.

May 06, May 09

Ans. : Many of the AM transmitters use the high level modulation technique. The block diagram of high level AM transmitter is as shown in Fig. 4.8

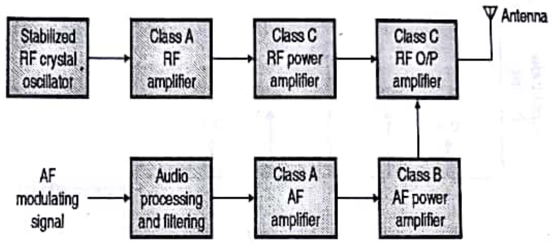


Fig. 4.8 : High level AM transmitter

Operation :

Here the carrier generated by the stabilized crystal oscillator is first amplified to the adequate power level using class C RF power amplifiers.

The modulating signal also is amplified to a high power level before modulation takes place. If we want 100 % modulation then the power of modulating signal must be 33 % of the total power. So if 1500 W total power is to be transmitted, the modulating power will be 500 W. This highlights the need to amplify the modulating signal to an adequate power level.

The modulation takes place in the last class C RF amplifier. The modulator output is AM wave which can be directly transmitted.

The collector modulated transistorized circuit or plate modulated vacuum tube modulator is used as modulator stage. The advantage of high level modulation is its high efficiency due to the use of highly efficient class C amplifiers. The disadvantage is that a large AF power amplifier is needed to raise the modulating signal to the adequate power level.

Q. 9 An amplitude modulated waveform has a form :

$X_C(t) = [10(1 + 0.6\cos 2000\pi t + 0.4\cos 4000\pi t)\cos 20{,}000\pi t]$;

1. **Sketch the amplitude spectrum of $X_C(t)$**
2. **Find the power content of each spectral component including carrier.**
3. **Find the total power and sideband power.**
4. **What is the modulation index ?**

May 10, May 14, May 16

Ans. :

1. Modulation index

$$m_t = \sqrt{m_1^2 + m_2^2} = \sqrt{(0.6)^2 + (0.4)^2}$$

$$= 0.721$$

2. Spectrum of AM wave

$$f_c = 1 \text{ kHz}$$

$$f_{m1} = \frac{2000\pi}{2\pi} = 1 \text{ kHz}$$

$$f_{m2} = \frac{4000\pi}{2\pi} = 2 \text{ kHz}$$

$$USB_1 = f_c + f_{m1}$$
$$= (1+1) = 2\text{ kHz},$$
$$USB_2 = f_c + f_{m2} = (1+2) = 3\text{ kHz}$$
$$LSB_2 = f_c - f_{m2} = (1-2) = -1\text{ kHz}$$
$$LSB_1 = f_c - f_{m1} = (1-1) = 0$$

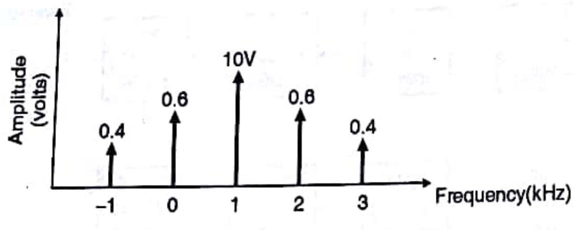


Fig. 4.9

3. **Total power :**

Total power in the sidebands

$$P_{SB} = (m_t^2/2)\,P_c$$

$$P_{SB} = \left[\frac{(0.721)^2}{2}\right] P_c = 0.2599\,P_c$$

$$\text{But}\quad P_c = \frac{E_c^2}{2R} = \frac{(10)^2}{2R} = \frac{100}{2R}$$

$$\therefore\ P_{SB} = \frac{0.259 \times 100}{2R} = \frac{25.9}{2R} = \frac{12.9}{R}\ W$$

Q. 10 Write short note on balanced modulator.

Dec. 09, Dec. 12, May 15

Ans. : The nonlinear modulators are also known as the balanced modulators. The balanced modulators are used to suppress the unwanted carrier in an AM wave.

The carrier and modulating signals are applied to the inputs of the balanced modulator and we get the DSB signal with suppressed carrier at the output of the balanced modulator. Thus the output consists of the upper and lower sidebands only.

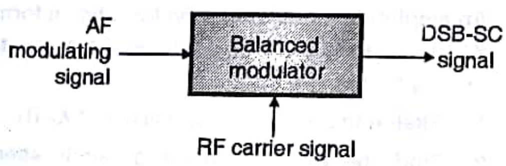


Fig. 4.10 : Block diagram of balanced modulator

Principle of operation :

The principle of operation of a balanced modulator states that if two signals at different frequencies are passed through a "nonlinear resistance" then at the output we get an AM signal with suppressed carrier. The device having a nonlinear resistance can be a diode or a JFET or even a bipolar transistor.

A semiconductor diode is a good example of a nonlinear device. The relation between the voltage across a device (v) and the device current (i) is nonlinear as shown in Fig. 4.11(a) and it can be mathematically expressed as :

$$i = a\,v + b\,v^2$$

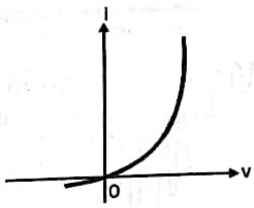


Fig. 4.11(a) : Nonlinear characteristics of a device

where a and b are constants.

Fig. 4.11(b) shows a possible arrangement for the use of nonlinear elements for DSB-SC modulator.

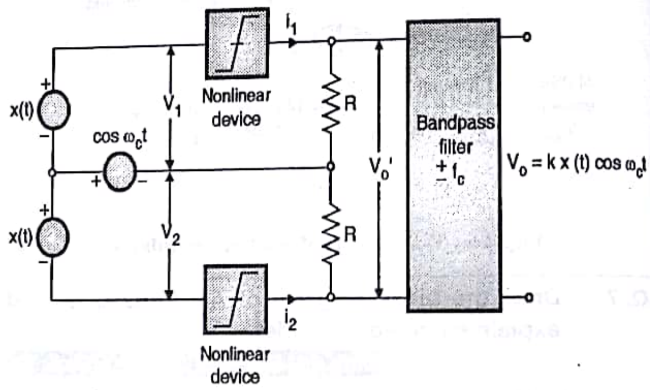


Fig. 4.11(b) : DSB-SC modulator using nonlinear device

Such an arrangement is called as "balanced modulator".

Q. 11 Explain balanced modulator using FET's.

Dec. 13

Ans. :

The balanced modulator using FETs is as shown in Fig. 4.12. The carrier voltage is applied "in phase" to the two gates via the transformers T_2 and T_1.

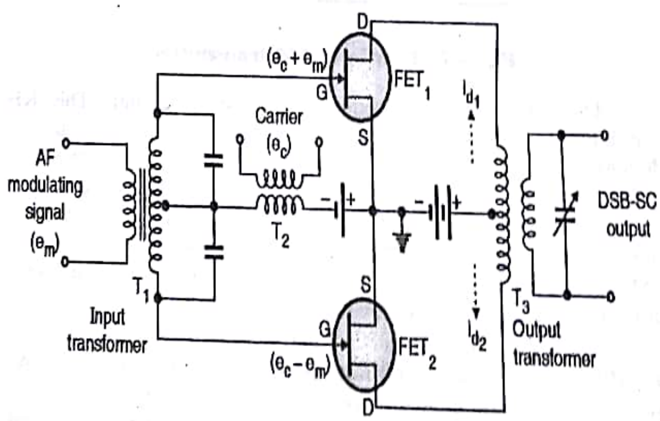


Fig. 4.12 : Balanced modulator using FETs

However the modulating signal appears 180° out of phase at the gates of the two FETs. This is because the input transformer T_1 is a centre tapped transformer.

Mode I (Operation in the absence of modulating signal) :

(i) In the absence of modulating signal, both the FETs conduct simultaneously due to the same carrier voltage applied to their gates. Both the FETs are identical in all respect to each other.

(ii) Their drain currents are equal in magnitude but opposite in direction through the primary of output transformer T_3 as shown in the Fig. 4.12.

(iii) Due to this their magnetic fields cancel each other, inducing a zero secondary voltage. Thus the output of transformer T_3 is zero. The carrier is thus suppressed.

Mode II (Carrier and modulating signal both present) :

(i) When the modulating signal is applied along with the carrier, the drain currents of the two FETs flow due to the combined effect of carrier and the modulating signal.

(ii) The FET currents due to carrier are equal and opposite and hence cancel each other.

(iii) However FET currents due to the modulating signal are equal but not opposite so they do not cancel out. This is because the modulating signal is applied 180° out of phase to the two FETs.

(iv) Hence at the output of the circuit we get a DSB-SC signal. For the 100 % suppression of the carrier, both the FETs must have the identical characteristics i.e. it should be a matched pair and the transformer centre taps must be exactly at the centre of the windings.

(v) Practically this is not possible hence carrier will be heavily suppressed but not completely removed.

Q. 12 Explain any one generation method of SSBSC AM. **Dec. 16**

Ans. : There are three practical methods of suppressing the unwanted sideband. They are as follows :

1. Filter method
2. The phase shift method
3. The third method.

In all these methods balanced modulators are used to suppress the carrier but each method uses a different technique to remove the unwanted sideband.

All the three systems can remove the upper or lower sideband as desired, to get a SSB signal from the DSB-SC signal. Let us understand the methods one by one.

Q. 13 Explain any one generation method of SSBSC AM. **Dec. 16**

Ans. : The block diagram for the filter method is as shown in Fig. 4.13(a).

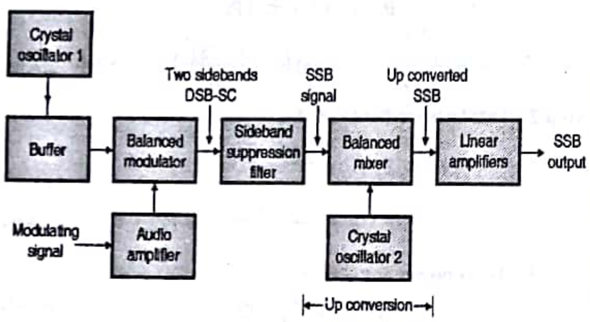


Fig. 4.13(a) : Filter method for sideband suppression

Operation :

1. The modulating signal is amplified using an audio amplifier and applied to the balanced modulator. The other input to the balanced modulator is the carrier signal generated by the crystal oscillator 1. This carrier frequency produced by this crystal oscillator is much less as compared to the carrier which is to be actually transmitted.
2. The balanced modulator will suppress the carrier and will produce upper and lower sidebands at its output. At the output of balanced modulator we get a DSB-SC signal.
3. Out of the two sidebands the unwanted sideband is heavily attenuated by the sideband suppression filter and the other sideband is passed through without much attenuation. Thus we get a SSB-SC signal at the output of the filter.
4. The frequency of this sideband at the output of the filter is very low. So the frequency is raised up to the transmitter frequency by the combination of the balanced mixer and second crystal oscillator. This is called as frequency up conversion.
5. This signal is then amplified using linear amplifiers (class B or AB). Linear amplifiers are used to avoid any waveform distortion.

Why to modulate at low frequency and use frequency up conversion ?

1. The filter used to suppress the unwanted sideband must have a flat passband and very high attenuation outside the passband. And the filter response must change from zero attenuation to full attenuation over a range of about 600 Hz.
2. To fulfill these requirements at very high operating frequency the Q of the tuned circuit used in the filter has to be extremely high obtaining such a high Q at very high frequency is practically not possible.
3. Hence the modulation is carried out at low frequency and then the frequency is raised by means of up conversion.

Frequency spectrums at various points

The frequency spectrums at various points for the filter method are as shown in Fig. 4.13(b).

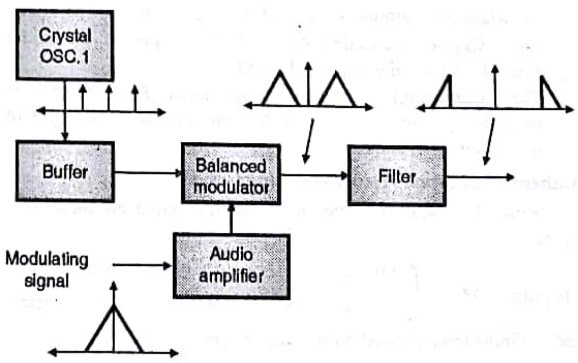


Fig. 4.13(b) : Frequency spectrums at various points for filter method

Q. 14 Explain the sideband generation in SSB using phase shift method.

Dec. 03, Dec. 05, May 06, Dec. 06, May 07, May 09, May 10, Dec. 11, Dec. 12

Ans. : The block diagram for the phase shift method of SSB generation is as shown in Fig. 4.14(a). This system is used for the suppression of lower sideband.

This system uses two balanced modulators M_1 and M_2 and two 90° phase shifting networks as shown in the Fig. 4.14(a).

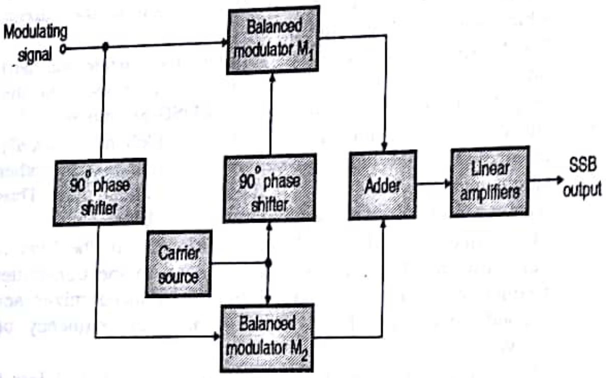


Fig. 4.14(a) : Phase shift method to suppress the LSB

Operation :

The operation of the phase shift method is as follows :

1. The balanced modulator M_1 has two inputs, the modulating signal without any phase shift and the RF carrier with a 90° phase shift.
2. The other balanced modulator M_2 receives the modulating signal with a 90° phase shift and carrier without any phase shift.
3. At the output of both the balanced modulators we DSB-SC signal consisting of both sidebands. The carrier is completely removed.
4. The Upper Sidebands (USBs) at the outputs of both the balanced modulators lead the carrier by 90°. But LSB at the output of M_1 leads the carrier by 90° and the LSB at the output of M_2 lags behind the carrier by 90°. Thus the LSBs are out of phase.
5. So when the outputs of M_1 and M_2 are applied to the adder, the LSBs are cancelled out and the output of the adder consists of only the upper sideband.
6. The linear amplifiers will follow the adder. They are class B or AB type amplifiers used to amplify the USB without introducing any distortion.

Mathematical proof of sideband suppression :

Refer Fig. 4.14(a). The inputs to the balanced modulator M_1 are

$$\text{Inputs to } M_1 \begin{cases} \cos \omega_m t & \text{...Modulating signal as it is} \\ \cos(\omega_c t + 90°) & \text{....90° phase shifted carrier.} \end{cases}$$

And the inputs to balanced modulator M_2 are

$$\text{Inputs to } M_2 \begin{cases} \cos(\omega_m t + 90°) & \text{....90° shifted modulating signal} \\ \cos \omega_c t & \text{....carrier as it is} \end{cases}$$

So the output of $M_1 = \cos(\omega_c t + 90°) \cdot \cos \omega_m t$ ...(1)

$$= \underbrace{\frac{1}{2}\cos[\omega_c t + \omega_m t + 90°]}_{\text{USB with 90° advance}} + \underbrace{\frac{1}{2}\cos[\omega_c t - \omega_m t + 90°]}_{\text{LSB with 90° delay}}$$

And output of $M_2 = \cos \omega_c t \cdot \cos(\omega_m t + 90°)$

$$= \underbrace{\frac{1}{2}\cos[\omega_c t + \omega_m t + 90°]}_{\text{USB with 90° advance}} + \underbrace{\frac{1}{2}\cos[\omega_c - \omega_m t - 90°]}_{\text{LSB with 90° delay}} \quad ...(3)$$

Output of the adder = Output of M_1 + output of M_2

$= \cos(\omega_c t + \omega_m t + 90°)$ ...(4)

This output is obtained by adding Equations (2) and (3). The LSBs in the outputs of M_1 and M_2 are 180° out of phase with respect to each other. Hence they are cancelled out when added. So the adder output contains only the upper sideband.

Suppression of the upper sideband :

We can suppress the USB and generate the SSB signal consisting of the LSB by arranging the blocks in the phase shift method as shown in Fig. 4.14(d).

Here the modulating and the carrier signals are applied to the upper balanced modulator M_1 directly (without any phase shift). Whereas both these signals are 90° phase shifted and then applied to the lower balanced modulator M_2.

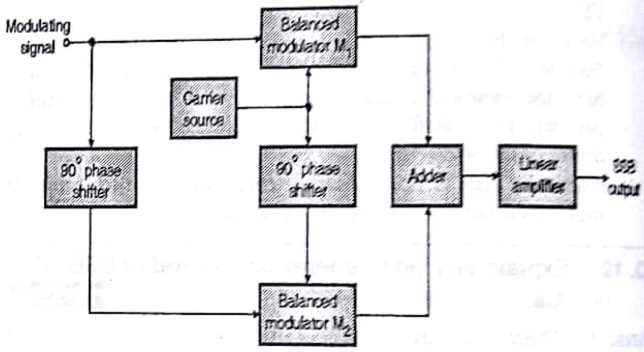


Fig. 4.14(d) : Phase shift method to suppress the USB

Q. 15 **A 400 W carrier is modulated to depth of 75%. Calculate total power in following forms of AM :**

1. **DSB – FC**
2. **DSB – SC**
3. **SSB – FC**
4. **SSB – SC.** **Dec. 12**

Ans. :

Given : $P_c = 400$ W, $m = 75\% = 0.75$

To find : Total Power

Step 1 : Total power for DSB-FC

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

$$= 400\left(1 + \frac{(0.75)^2}{2}\right) = 512.5 \text{ W} \quad \text{...Ans.}$$

Step 2 : Total power for DSB–SC

$$P_t = \frac{P_c m^2}{2}$$

$$= \frac{400 \times (0.75)^2}{2} = 112.5 \text{ W} \quad \text{...Ans.}$$

Step 3 : Total power for SSB-FC ...(2)

$$P_t = P_c\left(1 + \frac{m^2}{4}\right)$$

$$= 400\left(1 + \frac{(0.75)^2}{4}\right) = 456.25 \text{ W} \quad \text{...Ans.}$$

Step 4 : Total power for SSB-SC

$$P_t = \frac{P_c m^2}{4}$$

$$= \frac{400 \times (0.75)^2}{4} = 56.25 \text{ W} \quad \text{...Ans.}$$

Q. 16 What are the limitations of TRF receiver ? Explain how these limitations are avoided using superheterodyne receiver ?

Dec. 04, Dec. 07, Dec. 10, Dec. 13, May 16

Ans. :

Until shortly before World War II, almost all the radio receivers were of the TRF type. The virtues of this receiver are its simplicity and high sensitivity. The block diagram of the TRF receiver is as shown in Fig. 4.15

As seen from the Fig. 4.15, this receiver consists of two or three "tunable" RF amplifiers. All these amplifiers are tuned simultaneously to the desired signal frequency. A detector, AF amplifier and power amplifier follow the tuned RF amplifiers.

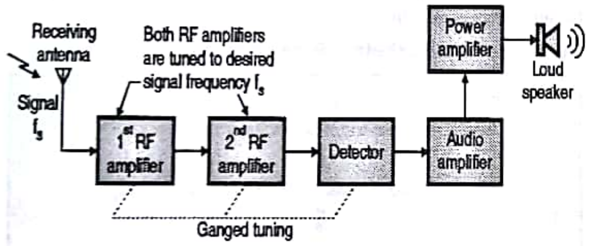


Fig. 4.15 : The TRF receiver

Operation :

1. The AM transmission takes place in the medium wave MW band or in the Short Wave (SW) band.
2. The frequency range for the MW band is from 530 kHz to 1640 kHz. Different radio stations operate at different frequencies in this range of frequency. e.g. the Pune station operates at a frequency of 800 kHz.
3. The operation of the TRF receiver is as follows :
   (i) Due to the electromagnetic waves passing over the receiving antenna, voltage is induced in it. This induced signal consists of signals from various transmitting stations.
   (ii) The RF amplifiers are tuned simultaneously to select and amplify the desired signal and reject all the other signals. Tuning means we adjust the resonant frequency of the tuned circuits equal to the desired station frequency. Ganged tuning means simultaneous tuning of tuned circuits in all the RF amplifier stages using a gang capacitor (a variable capacitor).
   (iii) The amplified signal is then demodulated (detected) by the detector. The carrier signal is bypassed and only the modulating signal is recovered in this process.
   (iv) The detected signal is amplified to the adequate voltage and power level using the audio amplifier and power amplifier and given to the loudspeaker for reproduction.

Problems in the TRF Receiver

Even though simple in operation, this receiver has some serious problems.

They are :

1. Instability
2. Variation in the bandwidth over the tuning range
3. Insufficient selectivity at high frequencies, and poor adjacent channel rejection. Let us discuss them one by one.

1. Instability

(i) The overall gain of the RF amplifier stages is very very high.

(ii) So a very small feedback signal from its output to input with correct phase (positive feedback) can initiate oscillations in the RF amplifier stage.

(iii) For example if the gain is 50,000 then to make the "Aβ" product equal to 1, the required feedback signal is only 1/50,000 of the output. Such a small feedback signal will start the oscillations and the amplifiers no more work as amplifiers.

(iv) This feedback takes place through the stray capacitances in the circuit. The reactance of stray capacitance decreases at higher frequencies which results in the increased feedback.

(v) Thus the possibility of oscillatory behaviour and therefore instability will increase with increased frequency. Once the oscillations begin, the RF amplifiers will stop to work as amplifier and operate as oscillators.

2. Variation in the bandwidth

(i) When the receiver is tuned, it is tuned to the carrier frequency (f_c) i.e. the station frequency and the tuned circuit is expected to select the carrier and the sidebands of the desired signal.

(ii) That means it must have adequate bandwidth (BW). For a tuned circuit,

$$BW = \frac{f_r}{Q} \quad \text{...(1)}$$

Where f_r is the resonant frequency which is f_c and Q is the quality factor.

(iii) Let us assume that the required BW = 10 kHz. This should remain constant at all the carrier frequencies.

Now at $f_r = f_c = 535$ kHz i.e. at the lowest frequency in the MW band the Q of the tuned circuit is,

$$Q = \frac{535}{10} = 53.5 \quad \text{...(2)}$$

And at $f_r = f_c = 1640$ kHz, i.e. at the highest frequency in the MW band, for the same bandwidth of 10 kHz the required value of Q will be,

$$Q = \frac{1640}{10} = 164 \quad \text{...(3)}$$

This value of Q is practically unobtainable due to various losses taking place at high frequency.

At the most we can obtain a Q of 120 at this frequency. Now the corresponding bandwidth will be,

$$BW = \frac{f_r}{Q} = \frac{1640 \text{ kHz}}{120} = 13.7 \text{ kHz} \quad \text{...(4)}$$

But the required BW is 10 kHz. Due to increased bandwidth the receiver will pick the adjacent channel along with the desired one.

3. **Insufficient selectivity**

Due to increased BW at higher frequencies, the ability of the TRF receiver to select the desired signal and reject all others is seriously affected. This is called loss of selectivity.

Due to these problems of instability and poor adjacent channel rejection, the TRF receivers are not used. They are replaced by the superheterodyne receivers.

Q. 17 What do you mean by superheterodyne receiver? Explain its working with neat waveform. May 05, May 07, Dec. 07, Dec. 09, Dec. 10, Dec. 13, May 16

Ans. : The problems in the TRF receiver are solved in this receiver by converting every selected RF signal (station) to a fixed lower frequency called as the "Intermediate Frequency (IF)".

This frequency contains the same modulation as the original carrier. The IF signal is then amplified and detected to get back the modulating signal. The intermediate frequency is lower than the lowest frequency that is to be received i.e. 530 kHz.

As the "IF" is lower than the lowest RF signal frequency, the possibility of oscillations and instability is minimized. Also the required value of Q for constant BW does not depend on the frequency of desired input signal, because the "IF" is constant and same for all the incoming RF signals.

Thus the superheterodyne receiver solves all the problems associated with the TRF receiver. The radio and TV receivers operate on the principle of **super heterodyning.** The block diagram of a superheterodyne radio receiver is shown in Fig. 4.19.

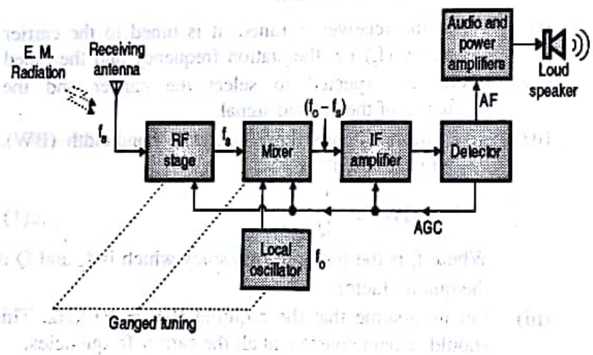


Fig. 4.16 : The superheterodyne receiver

Operation : The DSBFC or AM signal transmitted by the transmitter travels through the air and reaches the receiving antenna. This signal is in the form of electromagnetic waves. It induces a very small voltage (few μV) into the receiving antenna.

RF stage : The RF stage is an amplifier which is used to select the desired signal and reject other out of many, present at the antenna. It also reduces the effect of noise. At the output of the RF amplifier we get the desired signal at frequency "f_s".

Mixer : The mixer receives signals from the RF amplifier at frequency (f_s) and from the local oscillator at frequency (f_o) such that $f_o > f_s$.

Intermediate Frequency (IF)

The mixer will mix these signals to produce signals having frequencies f_s, f_o, $(f_o + f_s)$ and $(f_o - f_s)$. Out of these the difference of frequency component i.e. $(f_o - f_s)$ is selected and all others are rejected. This frequency is called as the Intermediate Frequency (IF).

$$\therefore \quad \text{I.F.} = (f_o - f_s) \qquad ...(1)$$

This frequency contains the same modulation as the original signal f_s. In order to maintain a constant difference between the local oscillator frequency and the incoming frequency, ganged tuning is used. This is simultaneous tuning of RF amplifier, mixer and local oscillator and it is achieved by using ganged tuning capacitors (Tuning control knob in radio set).

This intermediate frequency signal is then amplified by one or more IF amplifier stages. IF amplifiers satisfy most of the gain (and hence sensitivity) and the bandwidth requirements of the receiver. Therefore, the sensitivity and selectivity of this receiver do not change much with changes in the incoming frequency.

The amplified IF signal is detected by the detector to recover the original modulating signal. This is then amplified and applied to the loudspeaker.

AGC means automatic gain control. This circuit controls the gains of the RF and IF amplifiers automatically to maintain a constant output voltage level even when the signal level at the receiver input is fluctuating. This is done by feeding a controlling dc voltage to the RF and IF amplifiers. The amplitude of this dc voltage is proportional to the detector output.

The sequence of operation of a superheterodyne receiver can be summarized as follows :

Summary of superheterodyne principle

- Select the desired station at frequency f_s by tuning the RF amplifier and local oscillator.

⬇

- Local oscillator is tuned to frequency f_o with $f_o > f_s$.

⬇

- Mixer produces IF. Note that IF = $(f_o - f_s)$

⬇

- Output of mixer is an AM signal with two sidebands and carrier equal to IF. The IF amplifier amplifies this signal

⬇

- Detector will demodulate this signal to recover the modulating signal. (AF signal)

⬇

- The audio amplifier and power amplifier will amplify the AF signal and apply it to the loud speaker

Definition of heterodyning :

The process of mixing two signals having different frequencies to produce a new frequency is called as heterodyning.

Q. 18 Explain the following in relation of radio receiver : Sensitivity May 14

Ans. : Sensitivity of a radio receiver is defined as its ability to amplify weak signals. It is often defined in terms of the input voltage that must be applied at the input of the receiver to obtain a standard output power.

Sensitivity is measured in μV or decibels, below 1 V. Fig. 4.17 shows the typical sensitivity curve of a receiver. It shows the variation of sensitivity over the Medium Wave (MW) band of frequencies.

The sensitivity curve of Fig. 4.17 indicates that the receiver input (in μV) required to obtain the same standard output power changes with carrier frequency.

Required input voltage is minimum at 850 kHz and increases on both the sides of 850 kHz. If two receivers A and B are available with the required input voltages of 4 μV and 6 μV respectively to produce the standard output, then receiver A will be more sensitive. From the curve it is evident that the radio receiver is the most sensitive at about 850 kHz. The sensitivity of the receiver is decided by the gain of RF and IF amplifiers.

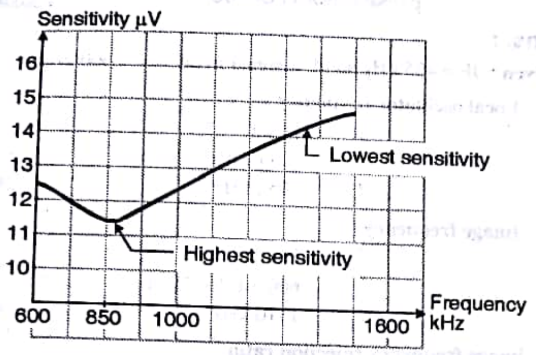


Fig. 4.17 : Sensitivity curve

How to improve the sensitivity ?

The sensitivity of a radio receiver is dependent on the RF and IF amplifier stages. By increasing the gains of these stages it is possible to increase the sensitivity of a receiver.

Q. 19 Explain the following in relation of radio receiver : Selectivity

May 09, Dec. 09, Dec. 11, May 14, May 15

Ans. : The selectivity of a receiver is defined as its ability to reject unwanted signals. It is expressed in the graphical form as shown in the Fig. 4.18

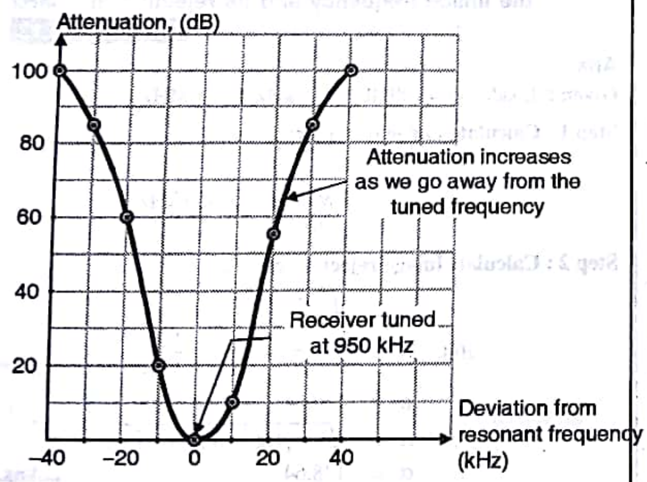


Fig. 4.18 : Typical selectivity curve

The radio receiver is tuned to the signal frequency,

$$f_s = 950 \text{ kHz.}$$

It shows that the receiver offers a minimum rejection at 950 kHz i.e. at the tuned frequency, but the rejection increases as the input signal frequency deviates on both the sides of 950 kHz.

The selectivity of a superheterodyne receiver is determined by the frequency response characteristics of the IF amplifier. The responses of the mixer and RF amplifier stages also play a small but significant role.

The selectivity decides the adjacent channel rejection of a receiver. Higher the selectivity better is the adjacent channel rejection and less is the adjacent channel interference.

How to increase the selectivity ?

The selectivity of a receiver depends on the IF amplifier. Higher the "Q" of the tuned circuit used in the IF amplifier, better is the selectivity.

Q. 20 Explain : Fidelity. **Dec. 09, Dec. 14**

Ans. :

The fidelity is the ability of a receiver to reproduce all the modulating frequencies equally. The fidelity basically depends on the frequency response of the AF amplifier. The typical fidelity curve is as shown in the Fig. 4.19.

High fidelity is essential in order to reproduce a good quality music faithfully i.e. without introducing any distortion. For this it is essential to have a flat frequency response over a wide range of audio frequencies. The fidelity curve for a receiver shown in Fig. 4.19 is basically the frequency response of the AF amplifier stage in the receiver.

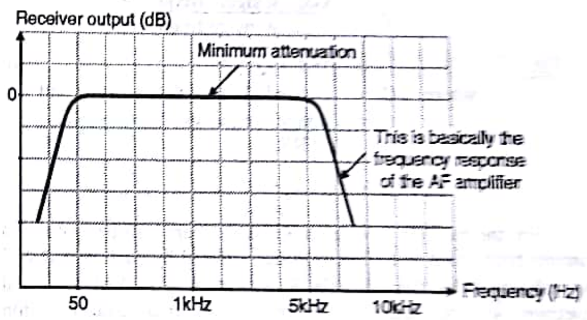


Fig. 4.19 : Fidelity curve

The procedure to plot the fidelity curve is as follows :

Q. 21 Write short note on : Image frequency and its rejection. **Dec. 04, May 06, May 08, Dec. 09, May 12, Dec. 12, Dec. 14, Dec. 15**

Ans. :

In the broadcast AM receivers the local oscillator frequency is higher than the incoming signal frequency by intermediate frequency i.e.

$$f_o = f_s + \text{I.F.} \quad \text{...(1)}$$

$$\text{or} \quad \text{I.F.} = (f_o - f_s) \quad \text{...(2)}$$

If two frequency components f_1 and f_2 are mixed in a mixer then mixer output consists of following components. If $(f_1 > f_2)$,

$$\text{Output of mixer : } f_1, f_2 (f_1 - f_2) \text{ and } (f_1 + f_2) \quad \text{...(3)}$$

Out of which the difference component i.e. $(f_1 - f_2)$ is selected using a tuned circuit after the mixer.

Now assume that the local oscillator frequency is set to "f_o" and an unwanted signal at frequency $f_{si} = (f_s + \text{IF})$ also somehow reaches at the input of the mixer along with the desired signal frequency f_s. Then the mixer output consists of the four frequency components corresponding to f_{si} as follows,

f_{si} , f_o, $(f_{si} + f_o)$ and $(f_{si} - f_o)$

But $\quad f_{si} = (f_o + f_{si})$

$\therefore \; f_o$, $(f_o + IF)$, $(2f_o + IF)$ and IF ...(4)

Where the last component at IF is the difference between f_{si} and f_o. This component will also be amplified by the IF amplifier along with the desired signal at frequency f_s.

This will create interference because both the stations corresponding to carrier frequencies f_s and f_{si} will be received at the same position on the frequency dial of a radio receiver.

This unwanted signal at frequency f_{si} is known as the "image frequency" and it is said to be the "image" of the signal frequency f_s. The relation between f_s and f_{si} is :

Image frequency : $\quad f_{si} = f_s + 2IF$ ...(5)

Remedy

The image frequency must be rejected by the receiver. The image rejection depends on the front end selectivity of the receiver i.e. the selectivity of the RF circuit. The image rejection must be achieved before the IF stage because once it reaches the IF stage it cannot be removed. The use of RF amplifier thus improves the image frequency rejection.

Image rejection using a single tuned circuit

The rejection of an image frequency by a single tuned circuit is given by,

$$\alpha = \frac{\text{Gain at signal frequency}}{\text{Gain at image frequency}}$$

$$= \sqrt{1 + Q^2 \rho^2} \qquad ...(6)$$

where $\quad Q$ = Loaded Q of the tuned circuit and

α = Image Frequency Rejection Ratio (IFRR)

$$\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} \qquad ...(7)$$

For the receiver having an RF stage there will be two tuned circuits both tuned to f_s. The rejection offered by each circuit will be calculated by the formula in Equation (6). And the total rejection will be equal to the product of individual rejection introduced by the individual tuned circuits as follows,

IFRR of two tuned circuits $= \alpha_1 \, \alpha_2$ ...(8)

For improving the capability of a receiver to reject image frequency, the value of Image Frequency Rejection Ratio (IFRR) should be as high as possible.

Q. 22 What do you mean by double spotting ?

May 07, May 09, Dec. 11, May 12, Dec. 12, Dec. 13, May 14

Ans. : Double spotting means the same stations gets picked up at two different nearby points, on the receiver dial. It happens due to poor front end selectivity i.e. inadequate image frequency rejection.

Double spotting is harmful because a weak station may be masked by the reception of a strong station at the same point, on the dial. Double spotting can be reduced by increasing the front end selectivity of the receiver. Inclusion of the RF amplifier stage will help in avoiding the double spotting.

In order to know more about the double spotting please refer to Q.18. Referring to this example you will understand an interesting thing. The frequency difference between the two points on the dial where the same station is tuned is exactly 2 IF. This is shown in Fig. 4.17.

The double spotting can be used to calculate the Intermediate Frequency (IF) practically.

Q.23 For an AM broadcast-band superheterodyne receiver with RF and IF frequencies of 600 kHz and 455 kHz respectively. Determine :

1. Local oscillator frequency

2. Image frequency

3. Image frequency rejection ratio for a preselector Q of 100.

May 09

Ans. :

Given : IF = 455 kHz and f_s = signal frequency = 600 kHz

1. Local oscillator frequency

$$f_o = f_s + IF$$

$$= (600 + 455) \text{ kHz}$$

$$\therefore \; f_o = 1055 \text{ kHz} \qquad \text{...Ans.}$$

2. Image frequency

$$f_{si} = f_s + 2\,IF$$

$$= 600 + (2 \times 455)$$

$$f_{si} = 1510 \text{ kHz} \qquad \text{...Ans.}$$

3. Image frequency rejection ratio :

Given $\quad Q = 100$

$$\alpha_1 = \sqrt{1 + Q^2 \rho^2}$$

where $\quad \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = \frac{1510}{600} - \frac{600}{1510}$

$$\therefore \; \rho = 2.119$$

$$\alpha = \sqrt{1 + [(100)^2 \times (2.119)^2]}$$

$$\alpha = 211.90 \qquad \text{...Ans.}$$

Q. 24 In an AM radio receiver the loaded Q of the antenna circuit at the input to the mixer is 100. If the intermediate frequency is 455 kHz, calculate the image frequency and its rejection at 1 MHz.

Dec.11, Dec.13

Ans. :

Given : Loaded Q = 100, IF = 455 kHz, f_s = 1 MHz

Step 1 : Calculate image frequency :

$$f_{si} = f_s + 2\,IF$$

$$= 1 \text{ MHz} + (2 \times 455 \text{ kHz})$$

$$= 1910 \text{ kHz}$$

Step 2 : Calculate image rejection ratio :

$$\alpha = \sqrt{1 + Q^2 \rho^2}$$

But $\quad \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = \frac{1910}{1000} - \frac{1000}{1910}$

$$\rho = 1.3864 \quad \text{and} \quad Q = 100$$

$$\therefore \; \alpha = \sqrt{1 + [100^2 \times (1.3864)^2]}$$

$$\therefore \; \alpha = 138.64 \qquad \text{...Ans.}$$

Q. 25 Explain working of envelope detector. An audio signal of bandwidth 5 kHz is modulated on carrier frequency 1 MHz using conventional AM. Determine range of values of RC for successful demodulation of this signal using an envelop detector.

May 12

Ans. : The envelope detector is a simple and very efficient device which is suitable for the detection of a narrowband AM signal. A narrowband AM wave is the one in which the carrier frequency f_c is much higher as compared to the bandwidth of the modulating signal.

An envelope detector produces an output signal that approximately follows the envelope of the input AM signal. That means at the output of this detector we get the modulating signal back. The envelope detector is used in all the commercial AM radio receivers.

Circuit diagram :

The circuit diagram of the envelope detector is shown in Fig. 4.20. It consists of a diode and RC filter.

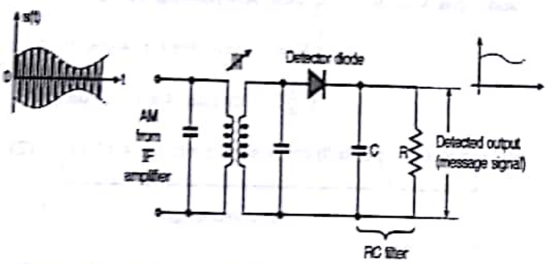


Fig. 4.20 : Envelope detector for detection of AM wave

Operation of Envelope Detector

The standard AM wave is applied at the input of the detector.

In every positive half cycle of the input the detector diode is forward biased. It will charge the filter capacitor C connected across the load resistance R to almost the peak value of the input voltage. C is a small capacitor and R is a large resistance.

As soon as the capacitor charges to the peak value, the diode stops conducting. The capacitor will discharge slightly through R between the positive peaks as shown in the Fig. 4.21.

The discharging process continues until the next positive half cycle. When the input signal becomes greater than the capacitor voltage, the diode conducts again and the process repeats itself.

Waveforms :

The input-output waveforms for the envelope detector are shown in Fig. 4.21. It shows the charging discharging of the filter capacitor and the approximate output voltage.

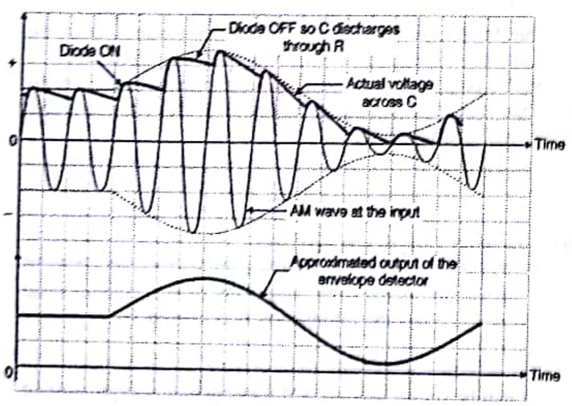


Fig. 4.21 : Input output waveforms for an envelope detector

It can be seen from these waveforms, that the envelope of the AM wave is being recovered successfully. We assume that the diode is ideal which presents a zero resistance when it is ON and infinite resistance when it is OFF. We also assume that the AM wave applied to the input of the detector is supplied by a source having internal resistance R_s.

Selection of the RC time constants :

The capacitor charges through D and R_s when the diode is on and it discharges through R when the diode is off. The charging time constant R_s C should be short as compared to the carrier period $1 / f_c$.

$$R_s C << 1 / f_c \qquad ...(1)$$

On the other hand the discharging time constant RC should be long enough so that the capacitor discharges slowly through the load resistance R.

But this time constant should not be too long which will not allow the capacitor voltage to discharge at the maximum rate of change of the envelope.

$$\therefore \frac{1}{f_c} << RC << \frac{1}{f_m} \qquad ...(2)$$

where f_m = Maximum modulating frequency.

Q. 26 What is peak clipping and diagonal clipping in diode detectors ?

May 05, May 06, May 11, Dec. 11, Dec. 16

Ans. : There are two types of distortions which can occur in the detector output. They are :

1. Diagonal clipping and
2. Negative peak clipping.

Diagonal clipping :

This type of distortion occurs when the RC time constant of the load circuit is too long. Due to this the RC circuit cannot follow the fast changes in the modulating envelope. The diagonal clipping is as shown in Fig. 4.22.

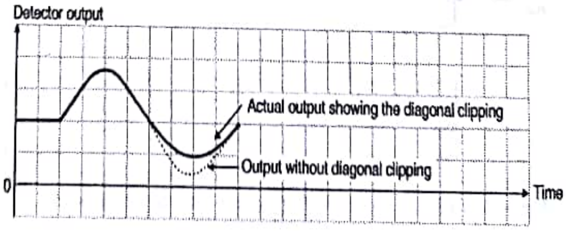


Fig. 4.22 : Diagonal clipping

We can avoid the diagonal clipping distortion by keeping the modulation index "m" less than 0.6.

Negative peak clipping :

This distortion occurs due to a fact that the modulation index on the output side of the detector is higher than that on its input side. So at higher depths of modulation of the transmitted signal, the over modulation (more than 100% modulation) may take place at the output of the detector.

The negative peak clipping will take place as a result of this over modulation as shown in Fig. 4.23.

Remedy : One way to reduce or eliminate the distortions is to choose the RC time constants properly as discussed. Another way is to limit the modulation index of the transmitted signal to about 70 percent.

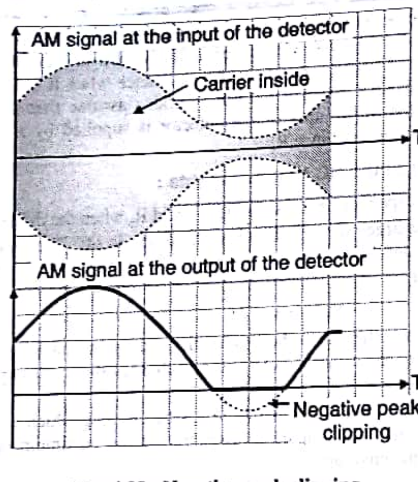


Fig. 4.23 : Negative peak clipping

Q. 27 Explain the generation and demodulation of SSBSC. **Dec. 15**

Ans. : The product modulator is a type of coherent SSB demodulator. To recover the modulating signal from the SSB-SC signal, we require a phase coherent or synchronous demodulator.

The block diagram of the coherent SSB-SC demodulator is as shown in Fig. 4.24. The received SSB signal is first multiplied with a locally generated carrier signal.

The locally generated carrier should have exactly the same frequency as that of the suppressed carrier.

The product modulator multiplies the two signals at its input and the product signal is passed through a low pass filter with a bandwidth equal to f_m.

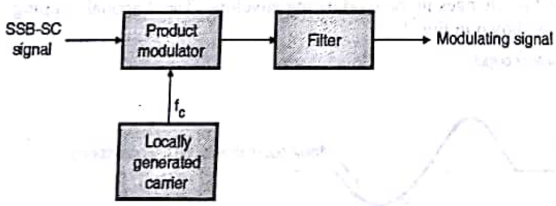


Fig. 4.24 : Block diagram of coherent SSB demodulator

At the output of the filter we get the modulating signal back. Mathematically this can be proved as follows :

Analysis of the coherent detector

Let the SSB wave at the input be given by,

$$s(t) = \frac{1}{2} E_c \left[x(t) \cos(2\pi f_c t) \pm \hat{x}(t) \sin(2\pi f_c t) \right]$$

The locally generated carrier is $\cos(2\pi f_c t)$.

$\therefore$ Output of the product modulator is given by,

$$v(t) = s(t) \cdot \cos(2\pi f_c t)$$

$$= \frac{1}{2} E_c \cos(2\pi f_c t) \cdot [x(t) \cos(2\pi f_c t) \pm \hat{x}(t) \sin(2\pi f_c t)]$$

$$v(t) = \frac{1}{2} E_c\, x(t) \cos(2\pi f_c t) \cos(2\pi f_c t) \pm \frac{E_c}{2} \hat{x}(t) \cos(2\pi f_c t) \sin(2\pi f_c t) \quad ...(1)$$

But $\cos A \cos B = \frac{1}{2} \{\cos(A+B) + \cos(A-B)\}$

and $\cos A \sin B = \frac{1}{2} \{\sin(A+B) - \sin(A-B)\}$

$$\therefore \; v(t) = \frac{1}{4} E_c\, x(t) [\cos(4\pi f_c t) + \cos(0)] \pm \frac{1}{4} E_c\, \hat{x}(t) [\sin(4\pi f_c t) - \sin(0)]$$

$$\therefore \; v(t) = \underbrace{\frac{1}{4} E_c\, x(t)}_{\text{Scaled message signal}} + \underbrace{\frac{1}{4} E_c [x(t) \cos(4\pi f_c t) \pm \hat{x}(t) \sin(4\pi f_c t)]}_{\text{Unwanted terms}} \quad ...(2)$$

When v(t) is passed through the filter, it will allow only the first term to pass through and will reject all other unwanted terms.

Thus at the output of the filter we get the scaled message signal and the coherent SSB demodulation is achieved.

$$\therefore \; v_o(t) = \frac{1}{4} E_c\, x(t) \quad ...(3)$$

Phase error and frequency error in coherent detection

The coherent detection explained in the preceding section, assumed the ideal operating conditions in which the locally generated carrier is in perfect synchronization with the carrier at the transmitter.

But in practice a phase error ϕ may arise in the locally generated carrier wave. The detector output will get modified due to phase error as follows :

$$v_o(t) = \frac{1}{4} E_c\, x(t) \cos\phi \pm \frac{1}{4} E_c\, \hat{x}(t) \sin\phi \quad ...(4)$$

In the above expression, the plus sign corresponds to the SSB input signal with only USB whereas the negative sign corresponds to SSB input with only LSB.

Due to the presence of the Hilbert transform $\hat{x}(t)$ in the output, the detector output will suffer from the phase distortion.

Such a phase distortion does not have serious effects with the voice communication. But in the transmission of music and video it will have untolerable effects.

Chapter 5 : Frequency Modulation and Demodulation

Q. 1 Compare frequency modulation. **Dec. 12**

Ans. : In sinusoidal Frequency Modulation (FM), the modulating signal $x(t) = E_m \cos(2\pi f_m t)$ is a pure sinusoidal signal. The carrier signal c (t) is also a sinewave at much higher frequency.

FM is a system of modulation in which the instantaneous frequency of the carrier is varied in proportion with the amplitude of the modulating signal. The amplitude of the carrier signal remain constant. Thus the information is conveyed via frequency changes.

FM was first practically tried in 1936 as an alternative to AM. As will be shown later on, FM transmission is more resistant to noise than A.M. The time domain display of FM wave is as shown in the Fig. 5.1.

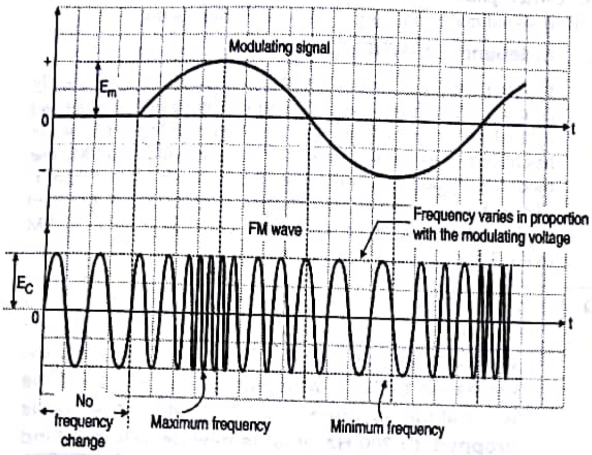


Fig. 5.1 : Time domain display of FM wave

The observation from the Fig. 5.1 are as follows :

1. The amount by which the carrier frequency deviates from its unmodulated value is called as **"deviation"**. The deviation (δ) is made proportional to the instantaneous value of modulating voltage.
2. The rate at which these frequency variations or oscillations takes place in the FM wave is equal to the modulating frequency (f_m).
3. The amplitude of the FM wave always remains constant. This is the biggest advantage of FM.

Q. 2 Find the mathematical expression of FM signal.

Dec. 06, Ma 07, Dec. 13, May 14, Dec. 16

Ans. : We know that the FM wave is a sinewave having a constant amplitude and a variable instantaneous frequency. As the instantaneous frequency is changing continuously, the angular velocity "ω" of an FM wave is the function of ω_c and ω_m.

Therefore the FM wave is represented by,

$$e_{FM} = s(t) = E_C \sin [F(\omega_c, \omega_m)] \quad ...(1)$$

$$= E_C \sin \theta(t) \quad ...(2)$$

$$\text{where} \quad \theta(t) = F(\omega_c, \omega_m) \quad ...(3)$$

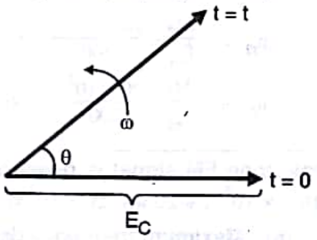


Fig. 5.2 : Frequency modulated vector

As shown in Fig. 5.2, $E_C \sin \theta(t)$ is a rotating vector. If "E_C" is rotating at a constant velocity "ω" then we could have written that $\theta(t) = \omega t$. But in FM this velocity is not constant. In fact it is changing continuously. The angular velocity of FM wave is given as,

$$\omega = \omega_c [1 + k_f E_m \cos \omega_m t] \quad ...(4)$$

Hence to find "θ (t)" we must integrate "ω" with respect to time.

$$\therefore \quad \theta(t) = \int \omega \, dt$$

$$= \int \omega_c [1 + k_f E_m \cos \omega_m t] \, dt \quad ...(5)$$

$$\therefore \quad \theta(t) = \omega_c \int [1 + k_f E_m \cos \omega_m t] \, dt$$

$$= \omega_c \left[t + \frac{k_f E_m \sin \omega_m t}{\omega_m} \right]$$

$$= \omega_c t + \frac{k_f E_m \omega_c \sin \omega_m t}{\omega_m}$$

$$\therefore \quad \theta(t) = \omega_c t + \frac{k_f E_m f_c \sin \omega_m t}{f_m} \quad ...(6)$$

As per the definition, $\delta = k_f E_m f_c$

$$\therefore \quad \theta(t) = \omega_c t + \frac{\delta \sin \omega_m t}{f_m} \quad ...(7)$$

Substitute this value of θ (t) in Equation (2) to get the equation for the FM wave as,

$$e_{FM} = s(t) = E_C \sin \left[\omega_c t + \frac{\delta}{f_m} \sin \omega_m t \right] \quad ...(8)$$

But $\frac{\delta}{f_m} = m_f$ i.e. the modulation index of FM wave. Hence the equation for FM wave is given as,

$$e_{FM} = E_C \sin [\omega_c t + m_f \sin \omega_m t] \quad ...(9)$$

This is the expression for a FM wave, where m_f represents the modulation index.

Meaning of mathematical representation

The mathematical expression for a FM wave is as follows :

$$e_{FM} = E_C \sin [\omega_c t + m_f \sin \omega_m t]$$

→ Frequency of FM wave varies according to the modulating signal

→ FM wave is a sine wave

→ Peak amplitude of FM wave is constant and equal to the peak amplitude of the carrier

1. The amplitude of FM wave is constant and equal to the amplitude of the carrier i.e. E_C.
2. FM wave is sinusoidal i.e. it has a shape of sine or cosine wave.
3. The frequency of FM wave is not constant. It varies continuously, above and below the carrier frequency f_c.

Q. 3 Explain the following : Narrow band and wideband FM. **May 04, Dec. 04, Dec. 05, Dec. 09**

Ans. :

The FM systems are basically classified into two types :

1. Narrowband FM
2. Broadband FM

1. Narrowband FM

(i) A narrowband FM is the FM wave with a small bandwidth. The modulation index m_f of narrowband FM is small as compared to one radian.

(ii) Hence the spectrum of narrowband FM consists of the carrier and upper sideband and a lower sideband.

For small values of m_f the values of the J coefficients are,

$$\left.\begin{aligned} J_0(m_f) &= 1, J_1(m_f) = m_f/2 \\ J_n(m_f) &= 0 \text{ for } n > 1 \end{aligned}\right\} \quad ...(5.3.21)$$

Therefore a narrowband FM wave can be expressed mathematically as follows.

$$e_{FM} = s(t) = \underbrace{E_C \sin \omega_c t}_{\text{Carrier}} + \underbrace{\frac{m_f E_C}{2} \sin(\omega_c + \omega_m) t}_{\text{USB}} - \underbrace{\frac{m_f E_C}{2} \sin(\omega_c - \omega_m) t}_{\text{LSB}} \quad ...(5.3.22)$$

The (–) sign associated with the LSB magnitude represents a phase shift of 180°. Practically the narrowband FM systems have m_f less than 1. The maximum permissible frequency deviation is restricted to about 5 kHz. This system is used in FM mobile communications such as police wireless, ambulances, taxicabs etc.

Generation of NBFM

Fig. 5.3 shows the generation of NBFM using balanced modulator.

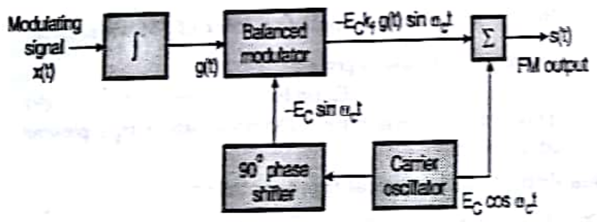


Fig. 5.3 : Generation of narrow band FM

The modulating signal is integrated and applied to a balanced modulator. The other input to the balanced modulator is a 90° phase shifted carrier signal. The output of balanced modulator contains only the sidebands because it suppresses the carrier. These sidebands are added with the carrier to obtain the NBFM signal.

Phasor diagram of NBFM

Fig. 5.4 shows the phasor diagram of NBFM.

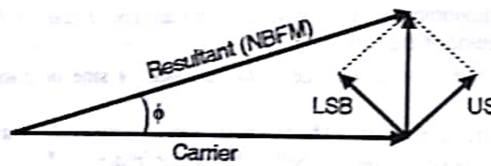


Fig. 5.4 : Phasor diagram of NBFM

This phasor diagram has been drawn by referring to Equation (11). The carrier phasor is represented by $E_C \sin \omega_c t$. It is drawn along the horizontal axis and remains fixed always.

The upper sideband is represented by $\frac{1}{2} m_f E_C \sin(\omega_c + \omega_m)$. It is drawn as a phasor rotating in counter clockwise direction at an angular velocity ω_m.

Similarly LSB is represented by $\frac{1}{2} m_f E_C \sin(\omega_c - \omega_m)$. This phasor rotates in the clockwise direction at an angular velocity of ω_m.

The resultant of the two sidebands is always perpendicular to the carrier phasor and the resultant phasor that represents the NBFM signal makes an angle ϕ with the carrier vector.

2. Wideband FM (WBFM)

For large values of modulation index m_f, the FM wave ideally contains the carrier and an infinite number of sideband located symmetrically around the carrier. Such a FM wave has infinite bandwidth and hence called as wideband FM. The modulation index of wideband FM is much higher than 1. The maximum permissible deviation is 75 kHz and it is used in the entertainment broadcasting applications such as FM radio, TV etc.

Q. 4 n an FM system, when the Audio Frequency (AF) is 500 Hz and the AF voltage is 2.4 V, the deviation is 4.8 kHz. If the AF voltage is now increased to 7.2 V, what is new deviation ? If the AF voltage is raised to 10 V while the AF is dropped to 200 Hz, what is new deviation ? Find modulation index in each case. **Dec. 12**

Ans. :

Given : f_{m1} = 500 Hz, E_{m1} = 2.4 V, Δf_1 = 4.8 kHz, f_{m2} = 200 Hz, E_{m2} = 7.2 V E_{m3} = 10 V

To find :

1. Deviation Δf_2
2. Modulation index
3. Deviation Δf_3

Step 1 : Calculate value of k

$$\text{Deviation } \Delta f = K E_{m1}$$

$$\therefore K = \frac{\Delta f_1}{E_{m1}} = \frac{4.8 \times 10^3}{2.4} = 2 \text{ kHz}$$

Step 2 : Calculate deviation Δf_2

$$\begin{aligned} \Delta f_2 \text{ for } E_{m2} &= 7.2 \text{ V} = K E_{m2} \\ &= 2000 \times 7.2 = 14.4 \text{ kHz} \quad \text{...Ans.} \end{aligned}$$

Step 3 : Calculate deviation Δf_3

$$\begin{aligned} \Delta f_3 \text{ for } E_{m3} &= 10 \text{ V and } f_{m2} \\ &= 200 \text{ Hz is } kE_{m3} \\ &= 2000 \times 10 = 20 \text{ kHz} \quad \text{...Ans.} \end{aligned}$$

Step 4 : Calculate modulation index in each case

$$m_{f1} = \frac{\Delta f_1}{f_{m1}} = \frac{4.8 \times 10^3}{500} = 9.6 \quad \text{...Ans.}$$

$$m_{f2} = \frac{\Delta f_2}{f_{m1}} = \frac{14.4 \times 10^3}{500} = 28.8 \quad \text{...Ans.}$$

$$m_{f3} = \frac{\Delta f_3}{f_{m2}} = \frac{20 \times 10^3}{200} = 100 \quad \text{...Ans.}$$

Q. 5 A single tone FM signal is given by $V_{FM}(t) = 10 \sin(16\pi \times 10^6 . t + 20 \sin 2\pi \times 10^3 t)$

Find : (a) Maximum frequency deviation.

(b) BW of FM by using Carson's rule. **Dec. 15**

Ans. :

Calculate bandwidth of FM using Carson's rule :

$$BW = 2[\Delta f + f_{m(max)}] = 2[20 + 1]$$

$$BW = 42 \text{ kHz} \quad \text{...Ans.}$$

Q. 6 List the different methods of FM generation.

May 03, May 04, Dec. 04, May 06, May 11

Ans. :

There are two basic methods of generating the FM waves as follows :

1. Direct methods and
2. Indirect methods

The classification of FM generation methods is shown in Fig. 5.5.

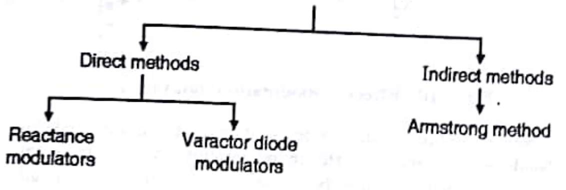


Fig. 5.5 : Classification of FM generation methods

Q. 7 Explain generation of FM by Armstrong method.

May 03, May 04, Dec. 04, May 05, May 06, May 08, May 09, May 11, Dec. 11, Dec. 14, May 15

Ans. :

This method can be understood by dividing it into four parts.

Part 1	:	How to obtain FM from phase modulator ?
Part 2	:	Implementation of phase modulator.
Part 3	:	Combining parts 1 and 2 to obtain the indirect method.
Part 4	:	Use of frequency multipliers and amplifiers.

Part 1 : How to generate FM from PM ?

In PM along with the phase variation, some frequency variation also takes place. Higher modulating voltages produce greater phase shift which in turn produces greater frequency deviation.

And higher modulating frequencies produce a faster rate of change of modulating voltage hence they also produce greater frequency deviation.

Thus in PM the carrier frequency deviation is proportional to modulating voltage as well as the modulating frequency. But in FM the frequency deviation is only proportional to the modulating voltage regardless of its frequency.

To correct this problem, the modulating signal is passed through a low pass RC filter as shown in Fig. 5.7. Due to this the high frequency modulating signals are attenuated but there is no change in the amplitudes of low frequency modulating signals.

The filter output is then applied to a phase modulator alongwith the carrier as shown in Fig. 5.7. Hence the extra deviation in the carrier f_c due to higher modulating frequency is compensated by reducing the amplitude of the high frequency modulating signals.

Hence the frequency deviation at the output of the phase modulator will be effectively proportional only to the modulating voltage and we obtain an FM wave at the output of phase modulator.

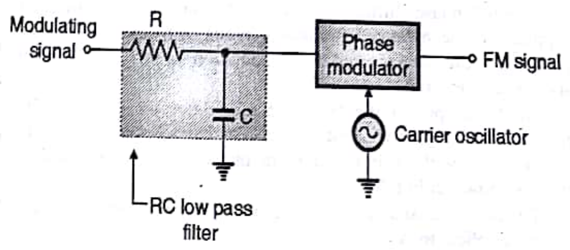


Fig. 5.7 : Generation of FM using phase modulation

Due to this arrangement the frequency deviation at the output of phase modulator, corresponding to higher modulating frequencies is reduced. The result is FM produced by a phase modulator.

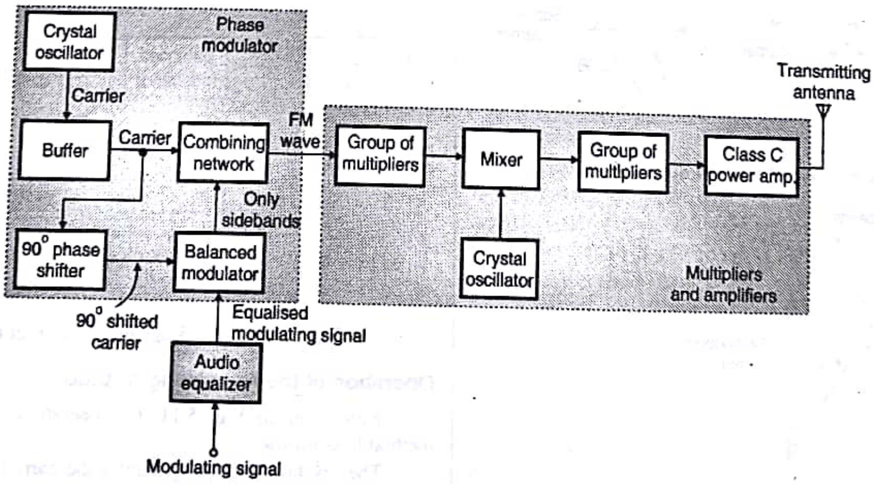


Fig. 5.6 : Indirect method [Armstrong method] of FM generation

Part 2 : Implementation of the phase modulator :

The block diagram of phase modulator circuit is shown in Fig. 5.8

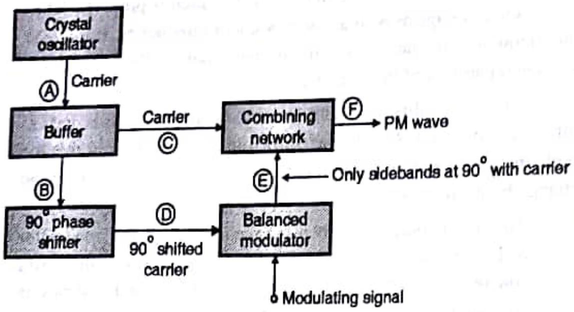


Fig. 5.8 : Phase modulator circuit

Operation :

The crystal oscillator produces a stable unmodulated carrier which is applied to the "90° phase shifter" as well as the "combining network" through a buffer.

The 90° phase shifter produces a 90° phase shifted carrier. It is applied to the balanced modulator along with the modulating signal. Thus the carrier used for modulation is 90° shifted with respect to the original carrier.

At the output of the balanced modulator we get DSB-SC signal i.e. A.M. signal without carrier. This signal consists of only two sidebands with their resultant in phase with the 90° shifted carrier as shown in Fig. 5.9.

The two sidebands and the original carrier without any phase shift are applied to a combining network. At the output of the combining network we get the resultant of vector addition of the carrier and two sidebands as shown in Fig. 5.9.

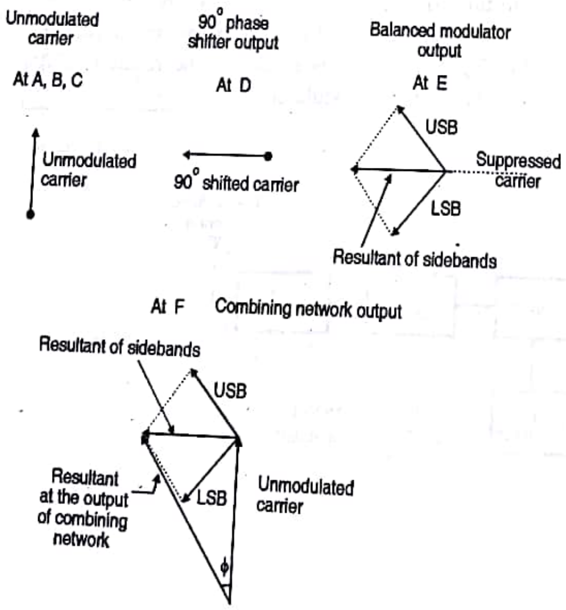


Fig. 5.9 : Phasors explaining the generation of P.M.

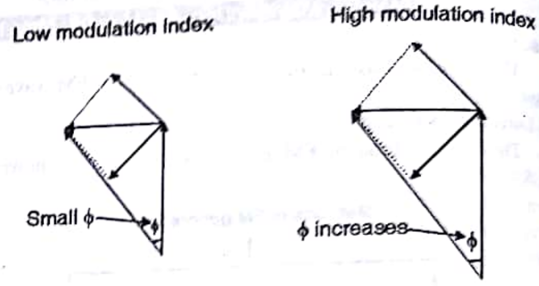


Fig. 5.10 : Effect of modulation index on ϕ

Now as the modulation index is increased, the amplitude of sidebands will also increase. Hence the amplitude of their resultant increases. This will increase the angle "ϕ" made by the resultant with unmodulated carrier. The angle "ϕ" decreases with reduction in modulation index as shown in Fig. 5.10. Thus the resultant at the output of the combining network is phase modulated. Hence the block diagram of Fig. 5.10 operates as a phase modulator.

Part 3 : Combine parts 1 and 2

Combining parts 1 and 2 we get the block diagram of the Armstrong method of FM generation as shown in Fig. 5.11

The audio equalizer block shown in Fig. 5.11 is nothing but an RC low pass filter. The role of RC filter has already been discussed in part - 1.

The modulating signal is passed through the audio equalizing circuit and applied to the phase modulator circuit.

We get the FM wave at the output of the combining network. Thus in the indirect method of FM generation we use phase modulation to obtain FM.

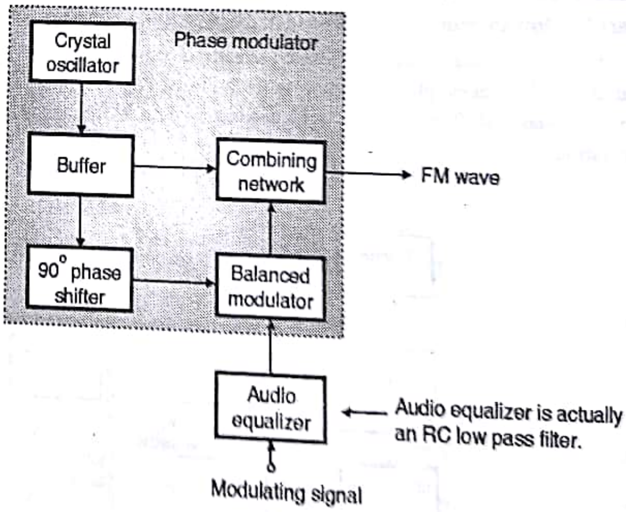


Fig. 5.11: Block diagram of indirect method

Operation of the Armstrong Method

Now refer to Fig. 5.11 The operation of the Armstrong method is as follows :

The crystal oscillator generates the carrier at low frequency typically at 1 MHz. This is applied to the combining network and a 90° phase shifter.

The modulating signal is passed through an audio equalizer to boost the low modulating frequencies, for the reason discussed. The modulating signal is then applied to a balanced modulator.

The balanced modulator produces two sidebands such that their resultant is 90° phase shifted with respect to the unmodulated carrier. The unmodulated carrier and 90° shifted sidebands are added in the combining network.

output of the combining network we get FM wave. This FM wave has a low carrier frequency f_c and low value of the modulation index m_f. The carrier frequency and the modulation index are then raised by passing the FM wave through the first group of multipliers. The carrier frequency is then raised by using a mixer and then the f_c and m_f both are raised to the required high values using the second group of multipliers. The effect of multiplication and mixing

The FM signal with high f_c and high m_f is then passed through a class C power amplifier to raise the power level of the FM signal.

Q. 8 Explain the FM noise triangle.

May 03, Dec. 04, May 05, May 06, Dec. 06, May 08, Dec. 13

Ans. : We know that the frequency modulation is more immune to the noise than AM or PM. However the noise immunity of FM is not the same for all the modulating frequencies. In fact it depends on the modulating frequency.

Consider a single noise voltage which has a frequency which falls in the passband of the receiver. This noise voltage will mix with the carrier to produce interference.

As shown in Fig. 5.12 the noise vector is superimposed on the carrier vector and it is rotating at a relative angular velocity $(\omega_n - \omega_c)$. Due to this the amplitude and phase of the carrier will keep changing continuously.

The amplitude and phase angle of the resultant shown in Fig. 5.12 will keep changing because the noise vector has a relative rotational motion with respect to the carrier

The maximum deviation in the amplitude = V_n ...(1)

and the maximum deviation in the phase = $\phi = \sin^{-1}(V_n/V_c)$ ...(2)

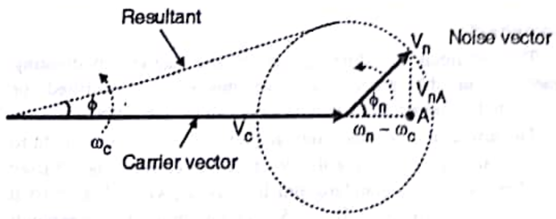


Fig. 5.12 : Effect of noise on the carrier

Thus amplitude and phase both of the carrier signal change due to noise.

Let us see the effect of these amplitude and phase change on the AM and FM receivers.

The AM receiver will not get affected due to the phase changes, it will only be affected due to the amplitude changes in carrier due to noise.

On the other hand FM receiver will not be affected by the amplitude changes but it will be affected by the phase changes introduced by the noise. Because these phase changes will be associated with frequency changes.

Effect of modulating frequency

In AM the change in noise frequency or change in modulating frequency does not affect the signal to noise ratio. Hence we get a rectangular distribution for AM as shown in Fig. 5.12.

In FM however the effect is entirely different. Referring Fig. 5.12 we conclude that the **noise phase modulates the carrier**. Now let us see the effect of modulating frequency.

From Fig. 5.12, the phase modulation of the carrier can be mathematically expressed as,

$$\phi = \tan^{-1}\left[\frac{V_{nA}}{V_c}\right] \quad ...(3)$$

Where V_{nA} = Perpendicular noise component

For the phase modulated signal the instantaneous phase angle $\theta(t)$ is expressed as,

$$\theta(t) = \phi_c + \phi(t) \quad ...(4)$$

where ϕ_c = Carrier phase angle

$\phi(t)$ = Instantaneous change in phase angle with time.

Differentiating both the sides with respect to t we get,

$$\frac{d\,\theta(t)}{dt} = \frac{d\,\phi_c}{dt} + \frac{d}{dt}[\phi(t)] \quad ...(5)$$

But the instantaneous angular frequency is given by,

$$\omega_i = \frac{d\,\theta(t)}{dt}$$

$$\therefore \quad \omega_i = \frac{d\,\phi_c}{dt} + \frac{d}{dt}[\phi(t)] \quad ...(6)$$

But $\frac{d\,\phi_c}{dt} = \omega_c$ and $\frac{d}{dt}[\phi(t)] = \omega_p(t)$

$$\therefore \quad \omega_i = \omega_c + \omega_p(t) \quad ...(7)$$

where $\omega_p(t)$ = Angular frequency deviation due to phase modulation.

$$\therefore \quad \omega_p(t) = \frac{d}{dt}[\phi(t)]$$

$$\therefore \quad 2\pi f_p(t) = \frac{d}{dt}[\phi(t)]$$

$$\therefore \quad f_p(t) = \frac{1}{2\pi}\frac{d}{dt}[\phi(t)] \quad ...(8)$$

$f_p(t)$ is the deviation in frequency due to phase modulation caused by noise. The FM receiver will produce an output voltage proportional to this frequency deviation.

This is an unwanted output voltage and hence should be treated as noise.

That means the noise in the receiver output is proportional to the rate of change of phase with respect to time. As this rate is proportional to frequency, the noise output goes on increasing with increase in the modulating frequency.

The noise phase modulates the carrier. Therefore if the ratio of noise voltage to carrier voltage remains constant then the value of modulation index due to noise (and therefore the, maximum phase deviation) remains constant. Thus as we reduce the noise sideband frequency, the modulation index due to noise does not change. (because it is PM)

But as we reduce the modulating signal frequency, the modulation index increases (because this is FM). Therefore the noise to signal ratio in FM goes on decreasing with reduction in the modulating signal frequency.

Assuming the noise frequencies are evenly spread across the entire frequency band of the receiver; we can see that noise output from the FM receiver decreases uniformly with noise sideband frequency in FM. In AM it remains constant. This is as shown in Fig. 5.13 The triangular noise distribution in FM is known as the "Noise Triangle".

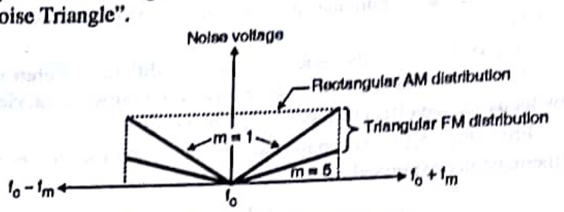


Fig. 5.13 : FM noise triangle

Conclusions :

1. The important conclusion from the discussion is that in FM the effect of noise is more prominent at higher modulating frequencies. Or we can say that in FM for a constant deviation (δ), the effect of noise increases with decrease in the modulation index m_f. At lower modulating frequencies, m_f is large and FM systems are less affected by the noise. (See Fig. 5.13)
2. In AM receiver the noise output will remain constant irrespective of the frequency hence we get a rectangular distribution as shown in Fig. 5.13

Q. 9 Explain in short pre-emphasis and de-emphasis.

May 03, May 04, May 05, Dec. 05, Dec. 06, May 09, Dec. 10, Dec. 11, May 12, Dec. 13, Dec. 14, May 15, Dec. 15, May 16

Ans. : It has been observed that in FM, the noise has a greater effect on the higher modulating frequencies. This effect can be reduced by increasing the value of modulation index (m_f) for higher modulating frequencies (f_m).

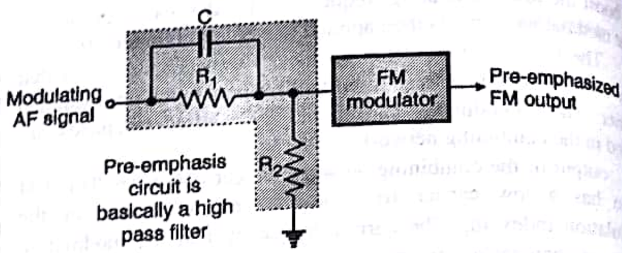
This can be done by increasing the deviation "δ" and δ can be increased by increasing the amplitude of modulating signal for the modulating signals of higher frequencies. Thus if we "boost" the amplitude of higher frequency modulating signals artificially then it will be possible to improve the noise immunity at higher modulating frequencies.

The artificial boosting of higher modulating frequencies is called as **pre-emphasis.** Boosting of higher frequency modulating signal is achieved by using the pre-emphasis circuit of Fig. 5.14(a). The modulating AF signal is passed through a **high pass RC filter,** before applying it to the FM modulator.

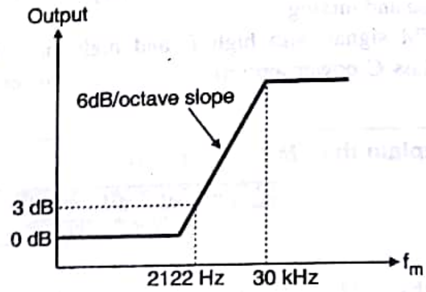
As f_m increases, reactance of C decreases and modulating voltage applied to FM modulator goes on increasing. The frequency response characteristics of the RC high pass network is shown in Fig. 5.14(b).

The boosting is done according to this prearranged curve. The amount of pre-emphasis in US FM transmission and sound transmission in TV has been standardized at 75 μsec. The pre-emphasis circuit is basically a high pass filter. The 75 μsec indicates the time constant of the RC circuit used for the pre-emphasis.

The pre-emphasis is carried out at the transmitter. The corner frequency for the RC high-pass network is 2122 Hz as shown in Fig. 5.14(b).



(a) Typical pre-emphasis circuit



(b) Pre-emphasis characteristics

Fig. 5.14

The pre-emphasis circuit is used at the transmitter as shown in the block diagram of Fig. 5.16.

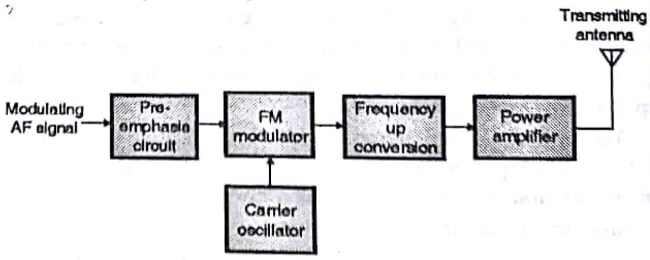


Fig. 5.15 : FM transmitter including the pre-emphasis

De-emphasis :

The artificial boosting given to the higher modulating frequencies in the process of pre-emphasis is nullified or compensated at the receiver by a process called "De-emphasis".

The artificially boosted high frequency signals are brought to their original amplitude using the de-emphasis circuit. The 75 μsec de-emphasis circuit is standard and it is as shown in Fig. 5.16 It shows that it is a low pass filter. 75 μsec de-emphasis corresponds to a frequency response curve that is 3 dB down at a frequency whose RC time constant is μsec.

$$\text{i.e.} \quad f = \frac{1}{2\pi RC} = \frac{1}{2\pi \times 75 \times 10^{-6}}$$

$$\therefore \quad f = 2122 \text{ Hz.}$$

The demodulated FM is applied to the De-emphasis circuit with increase in f_m the reactance of C goes on decreasing and the output of de-emphasis circuit will also reduce as shown in Fig. 5.16.

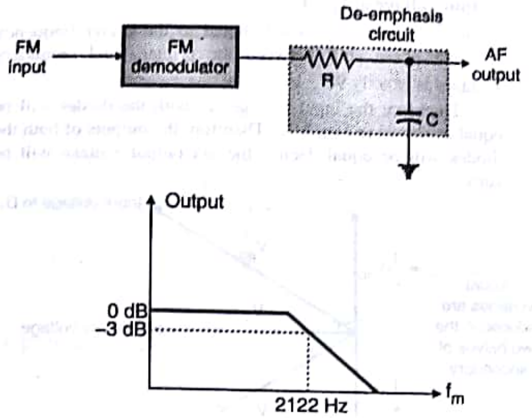
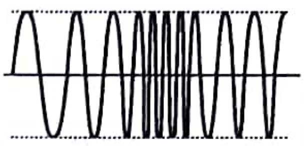
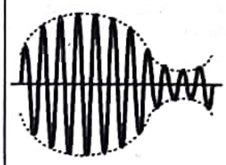


Fig. 5.16 : Typical de-emphasis circuit and its characteristics

Q. 10 Compare AM with FM.

May 03, Dec. 03, May 04, Dec. 05, May 09

Ans. :

Sr. No.	FM	AM
1.	Amplitude of FM wave is constant. It is independent of the modulation index.	Amplitude of AM wave will change with the modulating voltage.
2.	Hence transmitted power remains constant. It is independent of m_f.	Transmitted power is dependent on the modulation index.
3.	All the transmitted power is useful.	Carrier power and one sideband power are useless.
4.	FM receivers are immune to noise.	AM receivers are not immune to noise.
5.	It is possible to decrease noise further by increasing deviation.	This feature is absent in AM.
6.	Bandwidth = 2 [δ + f_m]. The bandwidth depends on modulation index.	BW = $2f_m$. It is not dependent on the modulation index.
7.	BW is large. Hence wide channel is required.	BW is much less than FM.
8.	Space wave is used for propagation. So radius of transmission is limited to line of sight.	Ground wave and sky wave propagation is used. Therefore larger area is covered than FM
9.	Hence it is possible to operate several transmitters on same frequency.	Not possible to operate more channels on the same frequency.
10.	FM transmission and reception equipment are more complex.	AM equipments are less complex.
11.	The number of sidebands having significant amplitudes depends on modulation index m_f.	Number of sidebands in AM will be constant and equal to 2.
12.	The information is contained in the frequency variation of the carrier.	The information is contained in the amplitude variation of the carrier.
13.	FM wave 	AM wave : 
14.	**Applications :** Radio, TV broadcasting, police wireless, point to point communications.	**Applications :** Radio and TV broadcasting.

Q. 11 Explain working of Foster Seeley discriminator with phasor diagram.

Dec. 03, May 05, May 06, Dec. 07, Dec. 09, May 10, Dec. 11, May 12, Dec. 12, May 14, Dec. 15, May 16, Dec. 16

Ans. : The phase discriminator or Foster Seeley discriminator is as shown in Fig. 5.17. If you compare this circuit with the balanced slope detector circuit then you will find that the diode and load arrangement is same in both the circuits.

But the method of applying the input voltage to the diodes which is proportional to the frequency deviation is entirely different. The Foster Seeley discriminator is thus derived from the balanced modulator.

Here the primary and the secondary windings both are tuned to the same center frequency "f_c" of the incoming signal. This simplifies the tuning process to a great extent and it will yield better linearity than the balanced slope detector.

Principle of operation :

Even though the primary and secondary tuned circuits are tuned to the same center frequency, the voltages applied to the two diodes D_1 and D_2 are not constant.

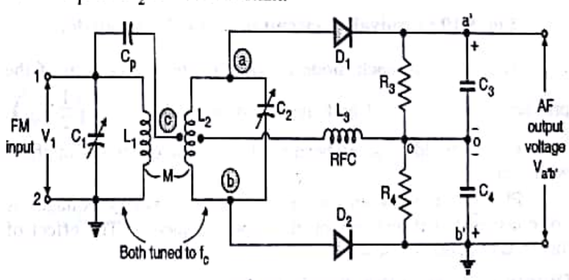


Fig. 5.17 : Phase discriminator

They vary depending on the frequency of the input signal. This is due to the change in phase shift between the primary and secondary windings depending on the input frequency.

Operation : The phase discriminator can be redrawn as shown in Fig. 5.18 to understand the concept of its operation.

The capacitor C_p is a coupling capacitor which passes all the frequencies at the input to the center tap of the transformer secondary.

C_4 will bypass resistance R_4 as $R_4 >> X_{C4}$. Therefore voltage across RFC will be equal to the input voltage V_1.

$$V_{RFC} = V_1 \quad ...(1)$$

It can be proved that the secondary voltage V_{ab} gets divided equally across the upper and lower halves of the secondary as shown in Figs. 5.19 and 5.20.

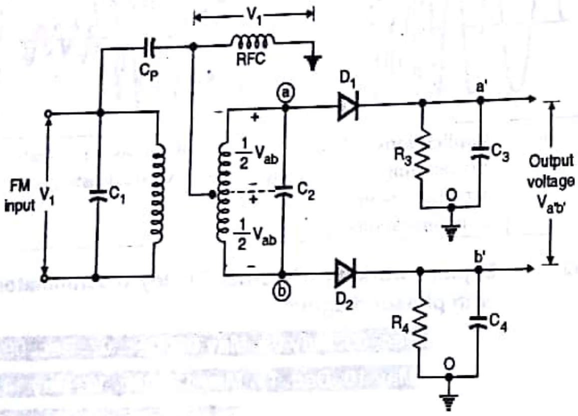


Fig. 5.18 : Phase discriminator redrawn

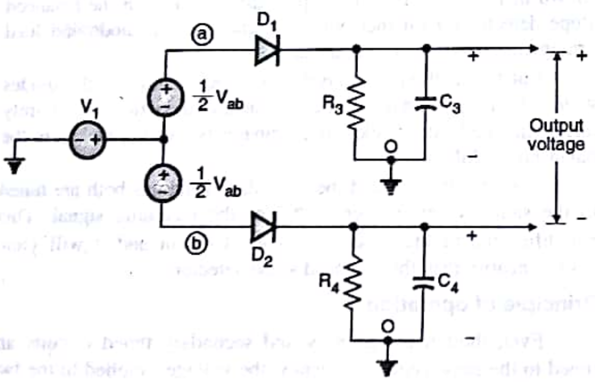


Fig. 5.19 : Equivalent circuit of phase discriminator

Thus input to each diode is equal to the vector sum of the primary voltage V_1 and half the secondary voltage i.e. $\left(\frac{1}{2} V_{ab}\right)$. Vector sum should be taken because these voltages are not in phase with each other.

Phase shift between the primary and secondary voltages is not constant but it depends on the input frequency. The effect of this is as explained below.

Output voltage of the discriminator

Output voltage of the phase discriminator is equal to the difference between the outputs of the two diode rectifiers.

$$\therefore V_o = V_{a'b'} = V_{a'o} - V_{b'o} \quad ...(2)$$

As the diode drops are not known, we cannot calculate the output exactly. But it is sure that the output will be proportional to the voltage applied at the inputs of diodes D_1 and D_2.

$$\therefore V_o \propto V_{ao} - V_{bo} \quad ...(3)$$

1. **Output voltage at $f_{in} = f_c$:**

When the input frequency is equal to the center frequency (f_c), the phase shift between the primary and secondary voltages is exactly 90°.

Therefore the input voltages to both the diodes will be equal as shown in Fig. 5.20. Therefore the outputs of both the diodes will be equal. Hence the net output voltage will be zero.

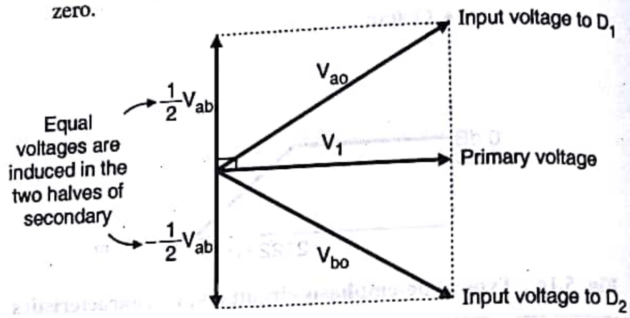


Fig. 5.20 : Phasor diagram for $f_{in} = f_c$

2. **Output voltage for $f_{in} > f_c$**

At input frequencies above the center frequency f_c, secondary voltage V_{ab} leads the primary voltage V_1 by less than 90°. Hence input voltage to D_1 i.e. V_{ao} is higher than input to D_2 i.e. V_{bo}. The output voltage will therefore be positive for $f_{in} > f_c$.

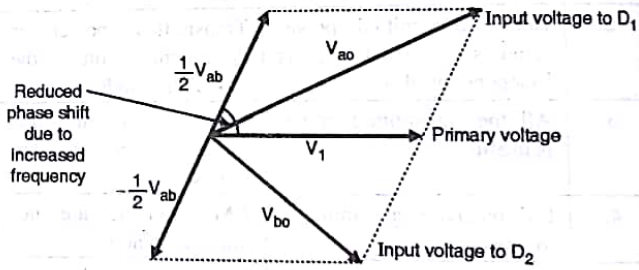


Fig. 5.21 : Phasor diagram for $f_{in} > f_c$

3. **Output voltage for $f_{in} < f_c$**

For input frequencies below the center frequency f_c, the secondary voltage V_{ab} leads the primary voltage V_1 by more than 90° as shown in Fig. 5.21. Hence input voltage to D_1 is less than input voltage to D_2. Therefore the output voltage will be negative for $f_{in} < f_c$.

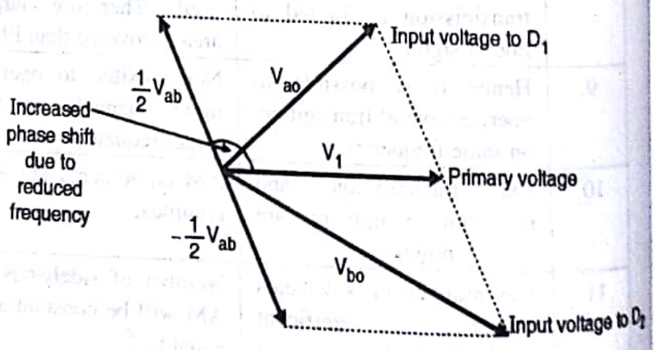


Fig. 5.22: Phasor diagram for $f_{in} < f_c$

Q. 12 Write short note on ratio detector.

May 07, May 08, Dec. 11, May 15

Ans. :

1. Ratio detector is another frequency demodulator circuit.
2. The circuit diagram of a basic ratio detector is as shown in Fig. 5.23.
3. If you compare this circuit with the Foster Seeley discriminator discussed you will notice that these two circuits are identical except for the following changes :
   1. The direction of diode D_2 is reversed.
   2. A large value capacitor C_5 has been included in the circuit.
   3. The output is taken somewhere else.

Operation and phasor diagrams :

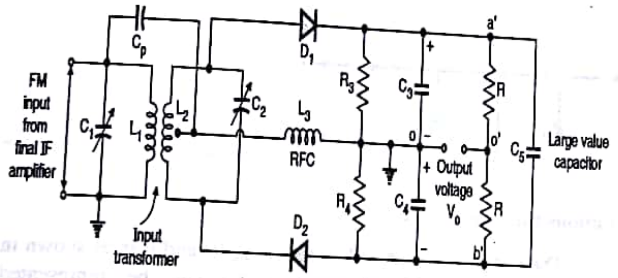


Fig. 5.23 : Ratio detector circuit

It can be shown that the ratio detector output voltage is equal to half of the difference between the output voltages from the individual diodes.

$$\therefore \quad V_o = \frac{V'_{ao} - V'_{bo}}{2}$$

Hence similar to the phase discriminator the output voltage is proportional to the difference between the individual output voltages. Due to this reason, the operation of the ratio detector is identical to the phase discriminator.

The S type frequency output curve can be derived in an identical way. The phasor diagrams are also identical.

The additional feature of the ratio detector is the amplitude limiting action which is incorporated due to the large capacitor C_5. Due to this the amplitude limiter is not required prior to the ratio detector.

Q. 13 Explain Foster Seeley discriminator with neat block diagram and compare the performance with ratio detector.

Dec. 15, Dec. 16

Ans. :

Sr. No.	Parameter	Phase discriminator	Ratio detector
1.	Alignment/tuning	Not critical	Not critical
2.	Output characteristics depends on	Primary and secondary phase relation	Primary and secondary phase relation.
3.	Linearity of output characteristics	Very good	Good
4.	Amplitude limiting	Not provided inherently	Provided by the ratio detector
5.	Applications	FM radio, satellite station receiver etc.	TV receiver sound section, narrow band FM receivers.

Chapter 6 : Pulse Analog Modulatio

Q. 1 Write a short note on : Sampling Theorem.

Dec. 03, May 04, May 06, Dec. 07, May 09, May 10, Dec. 10, May 11, Dec. 11, May 12, Dec. 12, May 14, Dec. 15, May 16, Dec. 16

Ans. : In order to represent the original message signal "faithfully" (without loss of information), it is necessary to take as many samples of the original signal as possible. Higher the number of samples, closer is the representation.

The number of samples depends on the "sampling rate" and the maximum frequency of the signal to be sampled. Sampling theorem was introduced to the communication theory in 1949 by Shannon. Therefore this theorem is also called as **"Shannon's sampling theorem"**.

The statement of sampling theorem in time domain, for the bandlimited signals of finite energy is as follows :

Statement :

1. If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2 W) seconds apart.
2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (1/2 W) seconds apart.

Combined statement of sampling theorem : A continuous time signal x (t) can be completely represented in its sampled form and recovered back from the sampled form if the sampling frequency $f_s \geq 2$ W where "W" is the maximum frequency of the continuous time signal x (t).

Q. 2 State and prove sampling theorem.

May 10, Dec. 10, Dec. 13, Dec. 15, May 16, Dec. 16

Ans. :

The assumptions made for this proof are as follows :

Assumptions : Let x (t) be a continuous time analog signal as shown in Fig. 6.1.

Let x (t) be a signal with finite energy and infinite duration. Let x (t) be a strictly bandlimited signal. That means it does not contain any frequency components above "W" Hz.

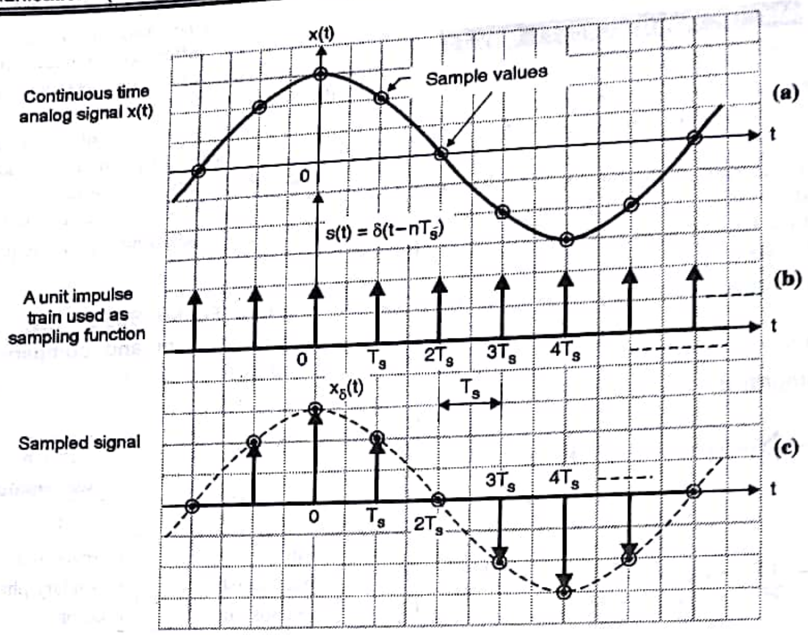


Fig. 6.1 : Sampling of a continuous time signal x (t)

Let s (t) be the sampling function as shown in Fig. 6.1. It is a train of unit impulses, spaced by a period of T_s seconds. This sampling function samples the original signal at a rate of "f_s" samples per second. Therefore "T_s" represents the sampling period such that,

$$T_s = \frac{1}{f_s} = \text{Sampling period} \quad ...(1)$$

$$\text{and} \quad f_s = \frac{1}{T_s} = \text{Sampling rate.}$$

Procedure to be followed :

We are going to follow the steps given below to prove the sampling theorem:

Step 1	:	Represent the sampling function s (t) mathematically.
Step 2	:	Represent the sampled signal x_δ (t) mathematically.
Step 3	:	Obtain the Fourier transform of the sampled signal.
Step 4	:	Prove that the sampled signal x_δ (t) completely represents x (t).
Step 5	:	Represent x (t) as summation of sinc functions (interpolation).
Step 6	:	Graphical representation of the interpolation process.
Step 7	:	Actual recovery of x (t) using an ideal low pass filter.

Part 1 : Sampling theorem :

Spectrum of the sampled signal

Step 1 : Represent the sampling function s (t) mathematically

Fig. 6.1 shows the sampling function s (t) which is a train of unit impulses. The spacing between the adjacent unit impulses is T_s seconds, therefore the frequency of the sampling function is equal to the sampling frequency f_s.

The sampled signal is denoted by x_δ (t) and it is as shown in Fig. 6.1. The sample function s (t) can be represented mathematically as follows :

$$s(t) = ...\delta(t+2T_s) + \delta(t+T_s) + \delta(t) + \delta(t-T_s) + \delta(t-2T_s) + ...$$

$$\therefore s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad ...(2)$$

Step 2 : Represent the sampled signal x_δ (t) mathematically :

Fig. 6.1 shows the sampled signal x_δ (t) graphically. It is present only at the sampling instants i.e. T_s, 2 T_s etc. and its instantaneous amplitude is equal to the amplitude of original signal x (t) at the sampling instants.

This is shown by the encircled points in Fig. 6.1. Let us represent the instantaneous amplitude of x (t) at the various sampling points $t = n\,T_s$ as x (n T_s). This is the amplitude of the encircled points of Fig. 6.1.

Looking at the sampled signal x_δ (t) we can say that the sampled signal is obtained by multiplying x (t) and s (t).

$$\therefore x_\delta(t) = x(t) \times s(t) = x(nT_s) \times s(t) \quad ...(3)$$

Substituting the expression for s (t) from Equation (2) we get the mathematical expression for the sampled signal x_δ (t) as,

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s) \quad ...(4)$$

Step 3 : Obtain the Fourier transform of the sampled signal :

The fourier transform of a train of impulses (dirac delta function) is given by,

$$X(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$$

Here we have the similar pulse train as sampling function s (t). Therefore the Fourier transform of the sampling function is given by,

$$S(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \qquad \text{...(5)}$$

Note that f_0 has been replaced by f_s in the above equation.

The sampled signal in the time domain is represented as product of x (t) and s (t).

$$\text{i.e.} \quad x_\delta(t) = x(t) \times s(t) \qquad \text{...(6)}$$

Taking the Fourier transform of both the sides we get,

$$\text{i.e.} \quad X_\delta(f) = X(f) * S(f) \qquad \text{...(7)}$$

This is because the Fourier transform of the product of two signals in the time domain is the convolution of their Fourier transforms. Substituting the value of S(f) from Equation (5) we get,

$$X_\delta(f) = X(f) * \left[f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \right] \qquad \text{...(8)}$$

where $*$ denotes convolution. Interchanging the orders of convolution and summation results in :

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f) * \delta(f - nf_s) \qquad \text{...(9)}$$

From the properties of delta function, we find that the convolution of X (f) and $\delta(f - nf_s)$ is equal to $X(f - nf_s)$. Hence the above equation can be simplified as follows :

F.T. of the sampled signal,

$$X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s) \qquad \text{...(10)}$$

where X (f) = Fourier transform of the original signal x (t).

4. Prove that sampled signal $x_\delta(t)$ completely represents x (t)

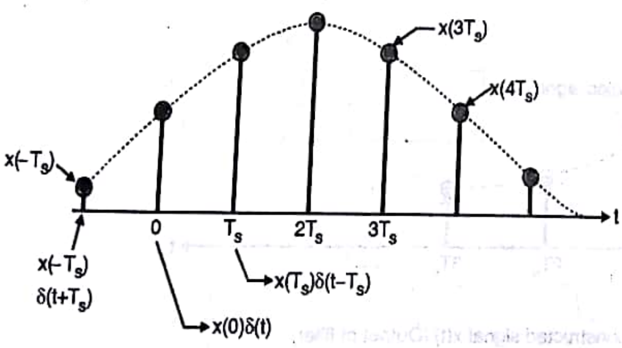


Fig. 6.2 : Sampled signal $x_\delta(t)$

$x_\delta(t)$ can be represented in the summation form as follows (Refer Fig. 6.2).

$$x_\delta(t) = \dots x(-T_s)\,\delta(t + T_s) + x(0)\,\delta(t) + x(T_s)\,\delta(t - T_s) + \dots \qquad \text{...(11)}$$

$$x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

We can obtain another useful expression for the fourier transform $X_\delta(f)$ by taking the fourier transform of both the sides of the equation stated above as,

$$X_\delta(f) = \sum_{n=-\infty}^{\infty} x(nT_s)\, e^{-j2\pi nfT_s} \qquad \text{...(12)}$$

This equation is the fourier transform of a discrete time signal x_δ (t). Therefore it is called as the Discrete Fourier Transform (DFT). Compare it with the definition of fourier transform of a continuous time signal. i.e.

$$X(f) = \int_{-\infty}^{\infty} x(t)\, e^{-j2\pi ft}\, dt$$

As the signal is discrete, the integration sign has been replaced by the summation sign and "t" has been replaced by "nT_s".

Now consider

$$X_\delta(f) = f_s X(f) + \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} f_s X(f - nf_s)$$

$$\therefore \quad X(f) = \frac{1}{f_s} X_\delta(f) - \sum X(f - nf_s)$$

But in the range $-W \le f \le W$ the second term of the above expression will not be present

$$\therefore \quad X(f) = \frac{1}{f_s} X_\delta(f) \qquad \text{...(13)}$$

Substitute $f_s = 2W$ and $X_\delta(f)$ from Equation (12) to get,

$$X(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(nT_s) \cdot e^{-j2\pi nfT_s} \qquad \text{...(14)}$$

This is the frequency spectrum of x (t) in terms of $x(nT_s)$ i.e. the sampled signal.

Substitute $T_s = 1/2W$ to get,

$$X(f) = \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W) \cdot e^{-j2\pi n f/2W} \quad \dots -W \le f \le W \qquad \text{...(15)}$$

Part 2 of the sampling theorem :

5. Reconstruction of signal from samples

This is the second part of the sampling theorem. From Equation (15) we can obtain x (t) by taking the Inverse Fourier Transform (IFT).

$$x(t) = \text{IFT}\{X(f)\}$$

$$= \text{IFT}\left\{ \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W).\, e^{-j\pi fn/W} \right\}$$

Using the definition of inverse Fourier transform,

$$x(t) = \int_{-W}^{W} \frac{1}{2W} \sum_{n=-\infty}^{\infty} x(n/2W) \cdot e^{-j\pi fn/W}\, e^{j2\pi ft}\, df$$

Interchanging the order of summation and integration we get,

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W) \frac{1}{2W} \int_{-W}^{W} e^{j2\pi f\left(t - \frac{n}{2W}\right)}\, df$$

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W) \cdot \frac{1}{2W} \times \frac{1}{j2\pi\left[t-\frac{n}{2W}\right]} \cdot \left[e^{j2\pi f(t-n/2W)}\right]_{-W}^{W}$$

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W) \frac{1}{j4\pi W\left[t-\frac{n}{2W}\right]} \cdot \left[e^{j2\pi W(t-n/2W)} - e^{-j2\pi W(t-n/2W)}\right]$$

$$= \sum_{n=-\infty}^{\infty} x(n/2W) \cdot \left[\frac{e^{j2\pi W(t-n/2W)} - e^{-j2\pi W(t-n/2W)}}{j4\pi W\left[t-\frac{n}{2W}\right]}\right]$$

The term inside the square bracket is a "sine" function.

$$\therefore\ x(t) = \sum_{n=-\infty}^{\infty} x(n/2W) \frac{\sin(2\pi Wt - n\pi)}{(2\pi Wt - n\pi)} \qquad \text{...(16)}$$

We can simplify the equation above by using the definition of the "sinc function". The sinc function is defined as :

$$\text{sinc}\, x = \frac{\sin(\pi x)}{\pi x} \qquad \text{...(17)}$$

Therefore Equation (16) can be written as :

$$x(t) = \sum_{n=-\infty}^{\infty} x(n/2W)\,\text{sinc}\,(2Wt - n) \qquad \text{...(18)}$$

Equation (18) provides an interpolation formula for reconstructing the original signal x (t) from the sequence of sample values {x (n /2 W)}. The "sinc" function plays the role of an interpolation function. Each sample x (n / 2 W) is multiplied by a delayed version of the interpolation function i.e. sinc function. Then all these resulting waveforms are added to obtain x (t).

6. Graphical representation of the interpolation process

Let us re-arrange Equation (18) as follows :

$$x(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\text{sinc}\,2W\left(t - \frac{n}{2W}\right)$$

This is because $\frac{1}{2W} = T_s$

$$\therefore\ x(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\,\text{sinc}\,2W(t - nT_s) \qquad \text{...(19)}$$

Let us expand this equation to write,

$$x(t) = x(0)\,\text{sinc}\,2Wt + x(\pm T_s)\,\text{sinc}\,2W(t \pm T_s) + x(\pm 2T_s)\,\text{sinc}\,2W(t \pm 2T_s) + \dots \qquad \text{...(20)}$$

(a) First term : x (0) sinc 2 Wt

This will have a maximum amplitude at t = 0. The maximum amplitude is equal to the sample value x (0) at t = 0. This sinc function will pass through zeros at t = ± 1/2 W, ± 1/4 W ...etc. This is as shown in Fig. 6.3.

(b) Second term : $x(\pm T_s)$ sinc 2 W $(t \pm T_s)$:

This sinc function will have maximum amplitude at $t = \pm T_s$. The maximum amplitude is equal to the sample value $x(\pm T_s)$ at $t = \pm T_s$ respectively. Thus sinc 2 W $(t \pm T_s)$ represents shifted sinc function i.e. "sinc 2 Wt" by a period $\pm T_s$. This is as shown in Fig. 6.3.

Similarly the third term, $x(\pm 2T_s)$ sinc 2 W $(t \pm 2T_s)$ represents shifted sinc function "sinc 2 Wt" by a period of $\pm 2T_s$ and so on. We can plot all these sinc functions along with the sampled signal $x_\delta(t)$ as shown in Fig. 6.3. Note that the peak amplitude of any sinc function is equal to the corresponding sample value $x(nT_s)$.

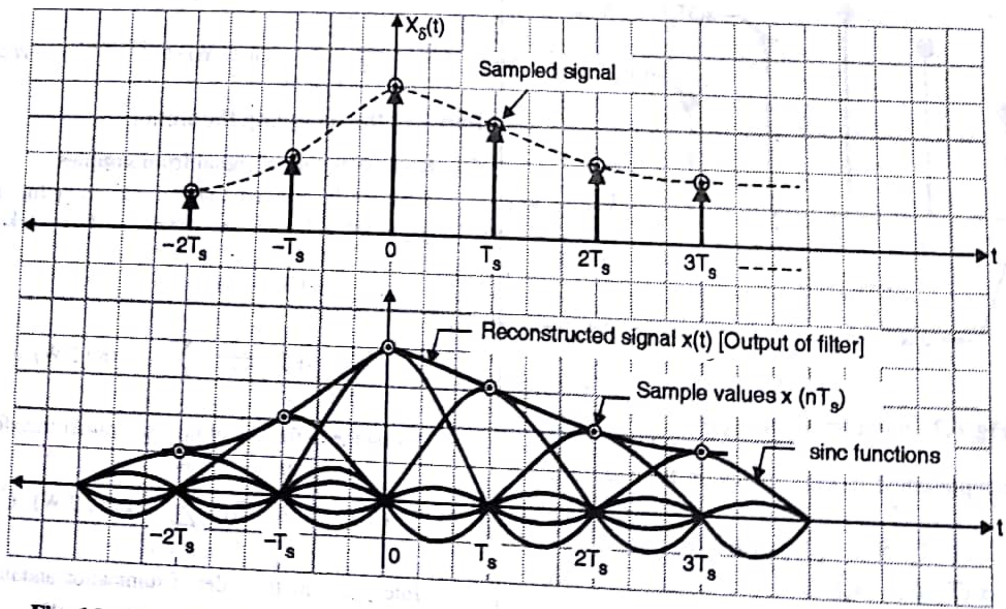


Fig. 6.3 : Reconstruction of the original signal x (t) from its samples using the interpolation

7. **Actual reconstruction of the original signal by using a low pass filter :**

   Thus the peaks of the sinc pulses represent the amplitudes of the samples.

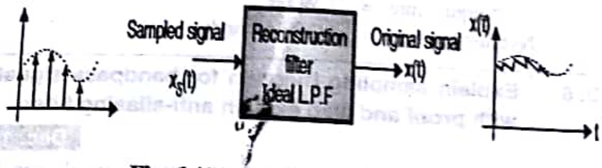


Fig. 6.4(a) : Reconstruction filter

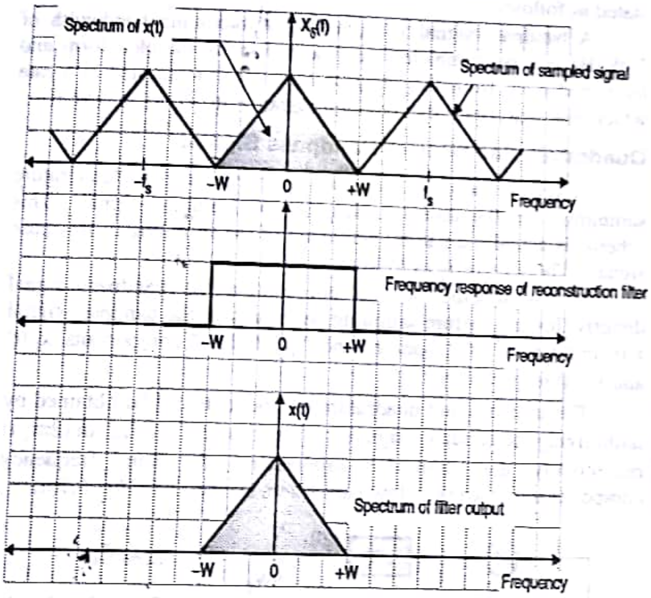


Fig. 6.4(b) : Operation of reconstruction

The signal x (t) expressed in Equation (18) is then passed through an ideal low pass filter to recover the original signal x (t). This low pass filter is therefore called as the reconstruction filter. This is shown in the graphical representation of Fig. 6.4(a).

Assume that the cut-off frequency of the ideal low pass filter is adjusted precisely to W Hz. The frequency response of the reconstruction filter is shown in Fig. 6.4(b).

When the sampled signal x_δ (t) is applied at the input, this filter will allow only the shaded portion in the spectrum of x_δ (t) to pass through to the output and will block all other frequency components. Thus the frequency components only corresponding to x (t) will be passed through to the output and the original signal x (t) is recovered.

Q. 3 State and prove sampling theorem and explain the aliasing error. **Dec. 04, Dec. 07, May 11, Dec. 11, May 14, Dec. 16**

Ans. :

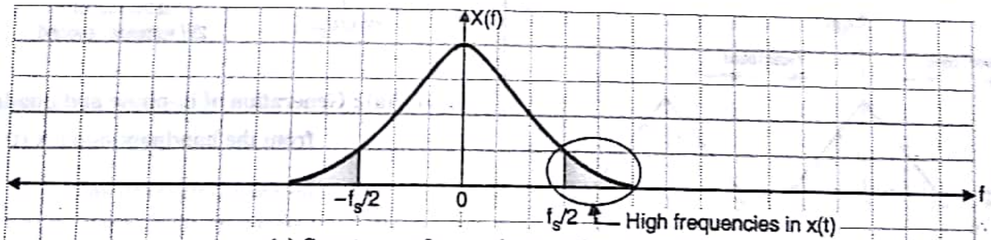
If the signal x (t) is not strictly bandlimited and / or if the sampling frequency f_s is less than 2 W, then an error called **aliasing** or **fold over error** is observed. The adjacent spectrums overlap if $f_s < 2$ W. This is shown in Fig. 6.5(b).

The signal x (t) is not strictly bandlimited. The spectrum of signal x (t) is shown in Fig. 6.5(b).

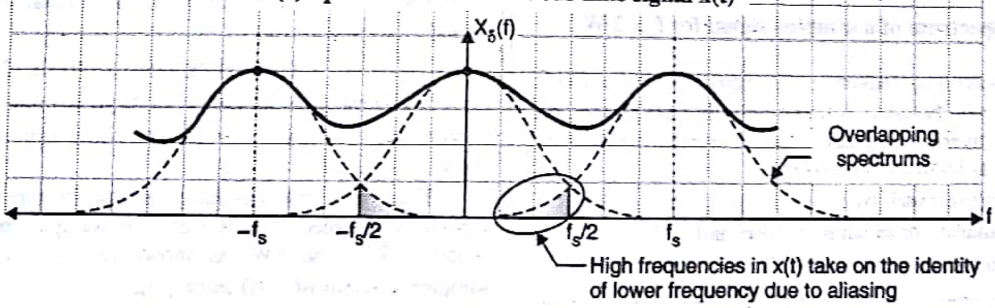
The spectrum X_δ (f) of the discrete time signal x_δ (t) is shown in Fig. 6.5(b) which is nothing but the sum of X (f) and infinite number of frequency shifted replicas of it as explained.

Consider the two replicas of X (f) which are centered about the frequencies f_s and $-f_s$. If we use a reconstruction filter with its pass-band extending from $-f_s / 2$ to $+ f_s / 2$ then its output will not be an undistorted version of the original signal x (t). Some distortion will be present in the filter output.

The distortion occurs due to the overlapping of the adjacent spectrums as shown in Fig. 6.5(b). Due to this overlapping, it is seen that the portions of the frequency shifted replicas are "folded over" inside the desired spectrum.



(a) Spectrum of a continuous time signal x(t)



(b) Spectrum of the sampled version of x(t) with $f_s < 2W$

Fig. 6.5

Due to this "fold over", high frequencies in X (f) are reflected into low frequencies in X_δ (f). This can be understood by comparing the shaded portions of the spectra shown in Fig. 6.5(a) and (b).

Aliasing : This phenomenon of a high frequency in the spectrum of the original signal x(t), taking on the identity of lower frequency in the spectrum of the sampled signal x_δ (t) is called as aliasing or fold over error.

Effect of aliasing

Due to aliasing some of the information contained in the original signal x (t) is lost in the process of sampling.

Q. 4 Prove sampling theorem for low pass signals. What is the use of antialiasing filter ? May 12, Dec. 14

Ans. : Aliasing can be completely eliminated if we take the following action : Use a bandlimiting low pass filter and pass the signal x (t) through it before sampling as shown in Fig. 6.6(a).

This filter has a cutoff frequency at f_c = W, therefore it will strictly bandlimit the signal x (t) before sampling takes place. This filter is also called as antialiasing filter or prealias filter.

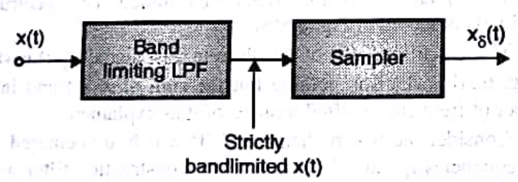


Fig. 6.6(a) : Use of a bandlimiting filter to eliminate aliasing

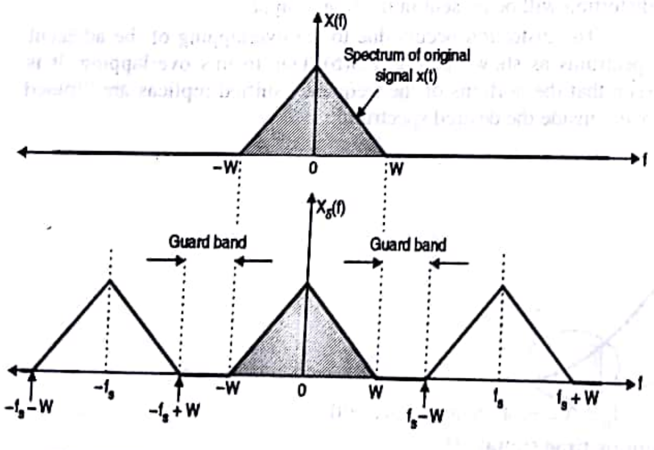


Fig 6.6(b) : Spectrum of a sampled signal for $f_s > 2$ W

Increase the sampling frequency f_s to a great extent i.e. $f_s >>>$ 2 W. Due to this, even though x (t) is not strictly bandlimited, the spectrums will not overlap. A guard band is created between the adjacent spectrums as shown in Fig. 6.6(b).

Thus aliasing can be prevented by :

1. Using an antialiasing or prealiasing filter and
2. Using the sampling frequency $f_s > 2$ W.

Q. 5 State sampling theorem for low pass signals. What is Nyquist rate ? Dec. 12

Ans. :

The minimum sampling rate of "2 W" samples per second for a signal x (t) having maximum frequency of "W" Hz is called as "Nyquist rate". The reciprocal of Nyquist rate i.e. 1/2 W is called as the Nyquist interval.

Nyquist rate = 2 W Hz

Nyquist interval = 1/2 W seconds

Q. 6 Explain sampling theorem for bandpass signals with proof and also explain anti-aliasing filter. Dec. 14

Ans. : The sampling theorem for the bandpass signals can be stated as follows :

A bandpass signal x (t), having a maximum bandwidth of 2 W Hz can be completely represented in its sampled form and recovered back from the sampled form if it is sampled at a rate which is at least twice the maximum bandwidth. (i.e. $f_s \geq 4$ W.)

Quadrature Sampling of Bandpass Signals

In this section, we consider a scheme called "quadrature sampling" for the uniform sampling of bandpass signals. This scheme is actually a natural extension of the sampling of low pass signals. The scheme is as follows :

In this scheme, we do not sample the bandpass signal directly. Instead, before sampling we represent the bandpass signal x (t) in terms of its "in-phase" and "quadrature" components, x_I (t) and x_Q (t) respectively.

The in-phase and quadrature components can be obtained by multiplying the bandpass signal x (t) by cos ($2\pi f_c$ t) and sin ($2\pi f_c$ t) respectively and then by suppressing the sum frequency components by means of low pass filters as shown in Fig. 6.7(a).

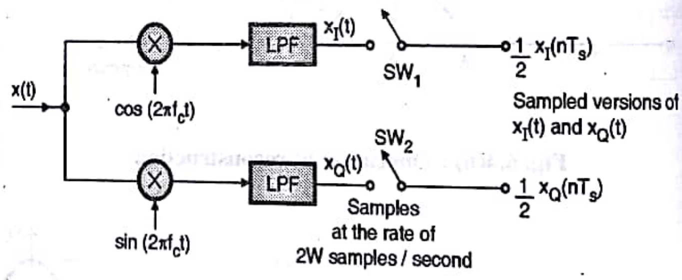


Fig. 6.7(a) : Generation of in-phase and quadrature samples from the bandpass signal x (t)

If x_I (t) = In-phase component and

x_Q (t) = Quadrature component.

Then we can express the bandpass signal x (t) in terms of x_I (t) and x_Q (t) as follows :

$$x(t) = x_I(t)\cos(2\pi f_c t) - x_Q(t)\sin(2\pi f_c t) \quad ...(21)$$

Under the assumption of $f_c > W$, it is found that x_I (t) and x_Q (t) both are "low pass signals" extending from – W to + W as shown in Fig. 6.7(b).

Then both the in-phase and quadrature components are separately sampled at a rate of 2 W samples per second by the switches SW_1 and SW_2 as shown in Fig. 6.7(a) to obtain the sampled versions of x_I (t) and x_Q (t).

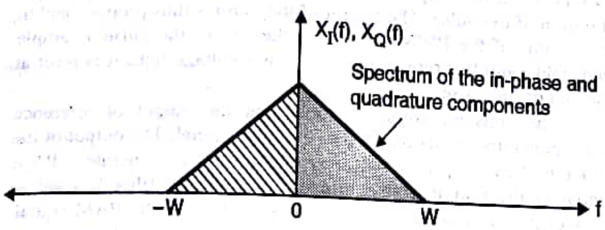


Fig. 6.7(b) : Spectrum of the in-phase and quadrature components of x (t)

In order to reconstruct the original bandpass signal from its quadrature sampled version, we first reconstruct the in-phase component $x_I(t)$ and quadrature component $x_Q(t)$ from their respective sampled versions $x_I(nT_s)$ and $x_Q(nT_s)$ by means of reconstruction filters. Then multiply $x_I(t)$ and $x_Q(t)$ by $\cos(2\pi f_c t)$ and $\sin(2\pi f_c t)$ respectively and add the result. The reconstruction process of x (t) is shown in Fig. 6.7(c).

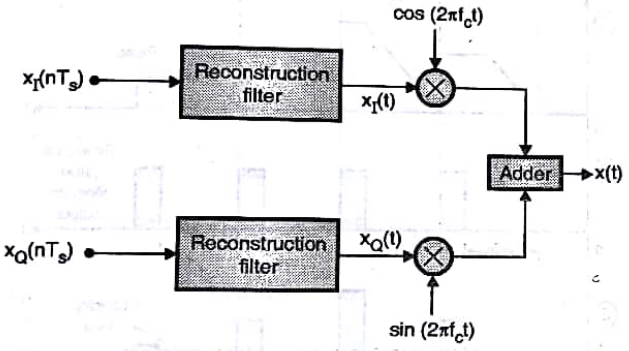


Fig. 6.7(c) : Reconstruction of the bandpass signal x(t)

Q. 7 A bandpass signal has a spectral range that extends from 30 kHz to 75 kHz. Find the sampling frequency.

Ans. :

Given : $f_1 = 30$ kHz and $f_2 = 75$ kHz.

Since $f_1 = 30$ kHz and $f_2 = 75$ kHz.

$\therefore$ Bandwidth $B = f_2 - f_1 = 75 - 30 = 45$ kHz ...(1)

Let us assume that $f_s = 2B = 2 \times 45 = 90$ kHz ...(2)

Note that here $f_s = 3 f_1$ i.e. f_s is the third harmonic of f_1 or f_1 and f_s are harmonically related with each other. Therefore we have to use the sampling theorem stated below to find out the value of f_s.

Sampling theorem for bandpass signals

The sampling theorem for the bandpass signals can be stated as follows :

A bandpass signal x (t), having a maximum bandwidth of 2 W Hz can be completely represented in its sampled form and recovered back from the sampled form if it is sampled at a rate which is at least twice the maximum bandwidth. (i.e. $f_s \geq 4$ W.)

Hence the sampling frequency for the given bandpass signal is given by,

$$f_s = 2B = 90 \text{ kHz} \quad \text{...Ans.}$$

Q. 8 With proper waveforms explain principles of qPWM, PAM, PPM systems of modulat

Dec. 03, May 16

Ans. : The other type of a pulse analog modulation is the Pulse Width Modulation (PWM). In PWM, the width of the carrier pulses varies in proportion with the amplitude of modulating signal. The waveforms of PWM are as shown in Fig. 6.8

As seen from the waveforms, the amplitude and the frequency of the PWM wave remains constant. Only the width changes. That is why the "information" is contained in the width variation. This is similar to FM. As the noise is normally "additive" noise, it changes the amplitude of the PWM signal.

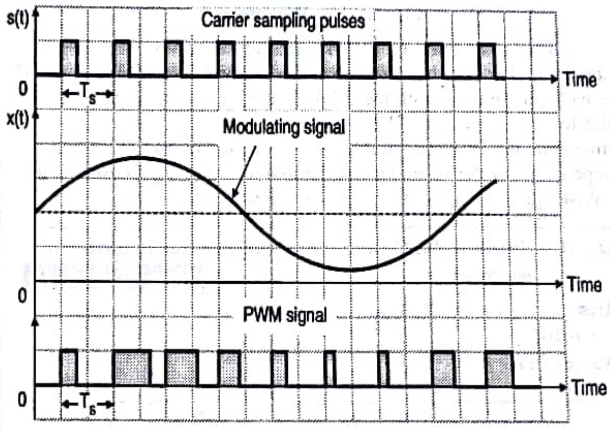


Fig. 6.8 : PWM signal [Trail edge modulated signal]

At the receiver, it is possible to remove these unwanted amplitude variations very easily by means of a limiter circuit.

As the information is contained in the width variation, it is unaffected by the amplitude variations introduced by the noise. Thus the PWM system is more immune to noise than the PAM signal.

Q. 9 Draw the block diagram of PWM generator. Explain the working giving waveforms at the output of each block.

Dec. 15, May 16

Ans. : The block diagram of Fig. 6.9(a) can be used for the generation of PWM as well as PPM.

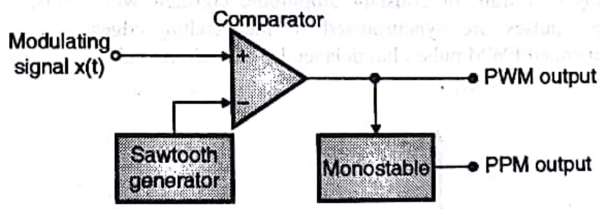


Fig. 6.9(a) : PWM and PPM generator

A sawtooth generates a sawtooth signal of frequency f_s, therefore the sawtooth signal in this case is a sampling signal. It is applied to the inverting terminal of a comparator. The modulating signal x (t) is applied to the non-inverting terminal of the same comparator. The comparator output will remain high as long as the instantaneous amplitude of x (t) is higher than that of the ramp signal.

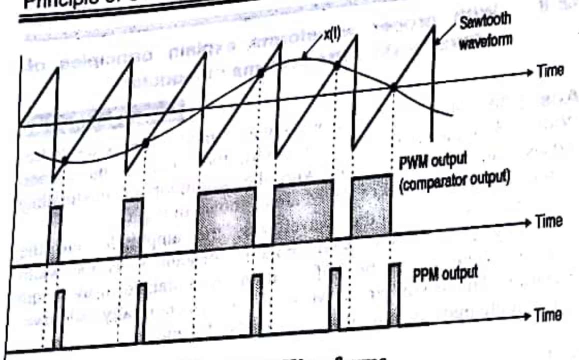


Fig. 6.9(b) : Waveforms

This gives rise to a PWM signal at the comparator output as shown in Fig. 6.9(b). Note that the leading edges of the PWM waveform coincide with the falling edges of the ramp signal. Thus the leading edges of PWM signal are always generated at fixed time instants. However the occurrence of its trailing edges will be dependent on the instantaneous amplitude of x (t). Therefore this PWM signal is said to be trail edge modulated PWM.

Q. 10 Explain PWM generation and degeneration method. **May 03, Dec. 15**

Ans.: The block diagram of PWM detector is as shown in Fig. 6.10.

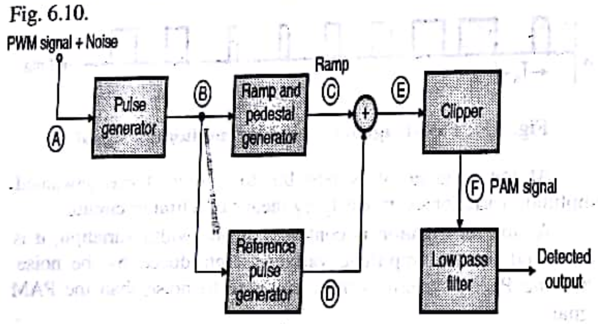


Fig. 6.10 : PWM detection circuit

Operation: The PWM signal received at the input of the detection circuit is contaminated with noise. This signal is applied to pulse generator circuit which regenerates the PWM signal. Thus some of the noise is removed and the pulses are squared up. The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude, constant width pulses. These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval.

The regenerated PWM pulses are also applied to a ramp generator. At the output of it we get a constant slope ramp for the duration of the pulse. The height of the ramp is thus proportional to the widths of the PWM pulses. At the end of the pulse a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.

The constant amplitude pulses at the output of reference pulse generator are then added to the ramp signal. The output of the adder is then clipped off at a threshold level to generate a PAM signal at the output of the clipper. A low pass filter is used to recover the original modulating signal back from the PAM signal. The waveforms for this circuit are as shown in Fig. 6.11.

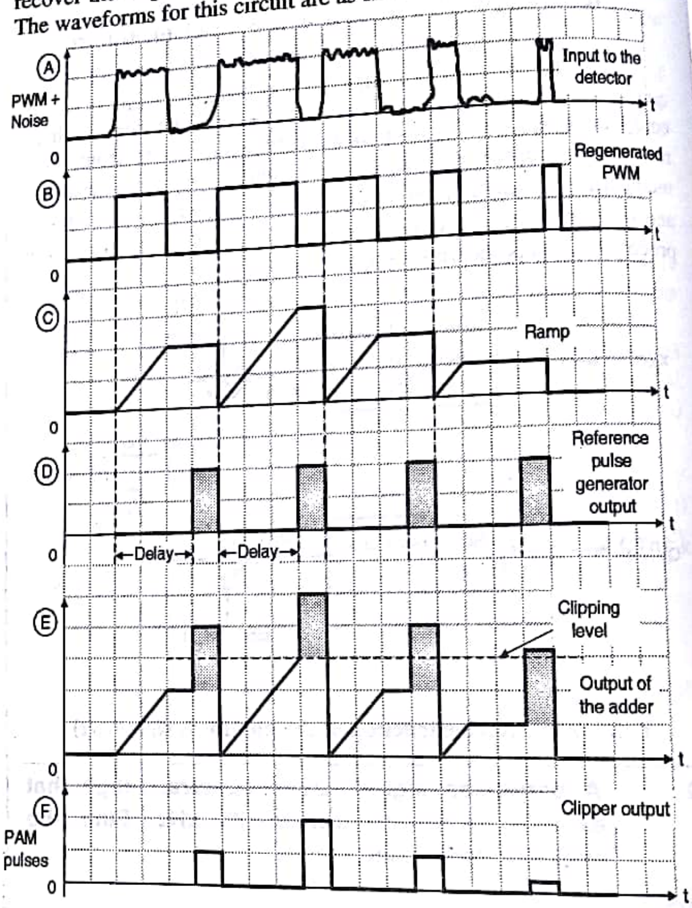


Fig. 6.11 : Waveforms for PWM detection circuit

Frequency Spectrum of PWM Wave : Fig. 6.12 shows the spectrum of a PWM signal for a sinusoidal modulating signal with a frequency f_m. The spectrum shows that the modulating frequency f_m and many of its sidebands are present.

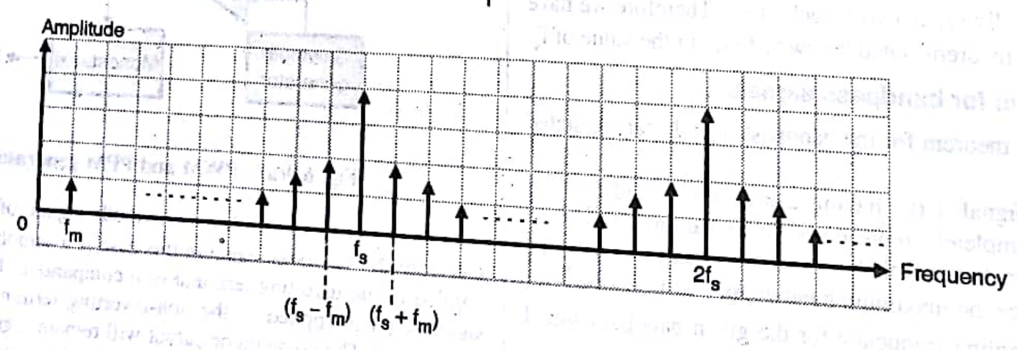


Fig. 6.12 : Spectrum of PWM signal

Q. 11 Explain generation and demodulation of PPM.

Dec. 03, May 15, Dec. 16

Ans. : In PPM the amplitude and width of the pulsed carrier remains constant but the position of each pulse is varied in proportion with the amplitudes of the sampled values of the modulating signal. The position of the pulses is changed with respect to the position of reference pulses.

The PPM pulses can be derived from the PWM pulses as shown in Fig. 6.13. Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.

The vertical dotted lines drawn in Fig. 6.13 are treated as reference lines to measure the shift in position of PPM pulses. The leading edge of each PPM pulse coincides with the trailing pulse of a PWM pulse.

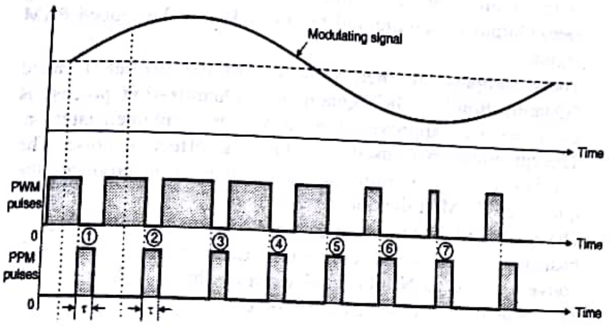


Fig. 6.13 : PPM pulses generated from PWM signal

The PPM pulses marked 1, 2 and 3 etc. in Fig. 6.13 go away from their respective reference lines. This is corresponding to increase in the modulating signal amplitude. Then as the modulating voltage decreases the PPM pulses 4, 5, 6, 7 come progressively closer to their respective reference lines.

Q. 12 Explain generation and demodulation of PPM.

Dec. 04, Dec. 14, May 15, Dec. 16

Ans. : The PPM signal can be generated from PWM signal as shown in Fig. 6.9(a). The same block diagram has been repeated in Fig. 6.14 as shown.

The PWM pulses obtained at the comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered. Hence corresponding to each trailing edge of PWM signal, the monostable output goes high. It remains high for a fixed time decided by its own RC components. Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting as shown in Fig. 6.9(b).

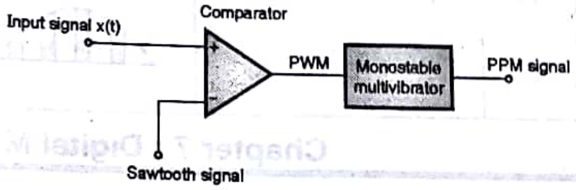


Fig. 6.14 : Generation of PPM signal

Demodulation of PPM : The PPM demodulator block diagram is as shown in Fig. 6.15.

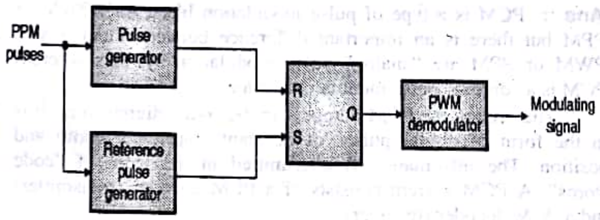


Fig. 6.15 : PPM demodulator circuit

The operation of the demodulator circuit is explained as follows :

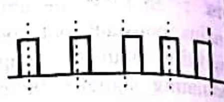
1. The noise corrupted PPM waveform is received by the PPM demodulator circuit.
2. The pulse generator develops a pulsed waveform at its output of fixed duration and apply these pulses to the reset pin (R) of a SR flip-flop.
3. A fixed period reference pulse is generated from the incoming PPM waveform and the SR flip-flop is set by the reference pulses.
4. Due to the set and reset signals applied to the flip-flop, we get a PWM signal at its output. The PWM signal can be demodulated using the PWM demodulator.

Q. 13 Write a short note : Comparison of PWM, PAM and PPM.

May 06, Dec. 07

Ans. :

Sr. No.	Parameter	PAM	PWM	PPM
1.	Type of carrier	Train of pulses	Train of pulses	Train of pulses
2.	Variable characteristic of the pulsed carrier	Amplitude	Width	Position
3.	Bandwidth requirement	Low	High	High
4.	Noise immunity	Low	High	High
5.	Information is contained in	Amplitude variations	Width variation	Position variation
6.	Transmitted power	Varies with amplitude of pulses	Varies with variation in width	Remains constant
7.	Need to transmit synchronizing pulses	Not needed	Not needed	Necessary
8.	Complexity of generation and detection	Complex	Easy	Complex

Sr. No.	Parameter	PAM	PWM	PPM
9.	Similarity with other modulation systems	Similar to AM	Similar to FM	Similar to PM
10.	Output waveforms		(D-463) 	

Chapter 7 : Digital Modulation Techniques

Q. 1 Describe PCM and also explain the PCM encoder and decoder with block diagram.

Dec. 07. Dec. 15

Ans. : PCM is a type of pulse modulation like PAM, PWM or PPM but there is an important difference between them. PAM, PWM or PPM are "analog" pulse modulation systems whereas PCM is a "digital" pulse modulation system.

That means the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position. The information is transmitted in the form of "code words". A PCM system consists of a PCM encoder (transmitter) and a PCM decoder (receiver).

The essential operations in the PCM transmitter are sampling, quantizing and encoding. All these operations are usually performed in the same circuit called as Analog-to-Digital (A to D) converter. It should be understood that the PCM is not modulation in the conventional sense. Because in modulation, one of the characteristics of the carrier is varied in proportion with the amplitude of the modulating signal. Nothing of that sort happen in PCM.

PCM Transmitter (Encoder) : Block diagram of the PCM transmitter is as shown in **Fig. 7.1.**

Operation of PCM transmitter :

Operation of the PCM transmitter is as follows :

1. The analog signal x (t) is passed through a bandlimiting low pass filter, which has a cut-off frequency $f_c = W$ Hz. This will ensure that x (t) will not have any frequency component higher than "W". This will eliminate the possibility of aliasing.
2. The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal.
3. These samples are then subjected to the operation called "Quantization" in the "Quantizer". Quantization process is the process of approximation as will be explained later on. The quantization is used to reduce the effect of noise. The combined effect of sampling and quantization produces the quantized PAM at the quantizer output.
4. The quantized PAM pulses are applied to an encoder which is basically an A to D converter. Each quantized level is converted into an N bit digital word by the A to D converter. The value of N can be 8, 16, 32, 64 etc.
5. The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses.
6. A pulse generator produces a train of rectangular pulses with each pulse of duration "τ" seconds. The frequency of this signal is "f_s" Hz. This signal acts as a sampling signal for the sample and hold block. The same signal acts as "clock" signal for the parallel to serial converter. The frequency "f_s" is adjusted to satisfy the Nyquist criteria.

PCM Receiver (Decoder)

Fig. 7.2 shows the block diagram of a PCM receiver.

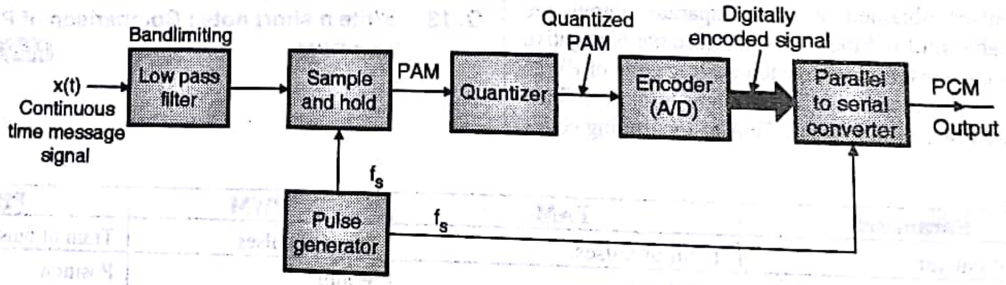


Fig. 7.1 : PCM transmitter (Encoder)

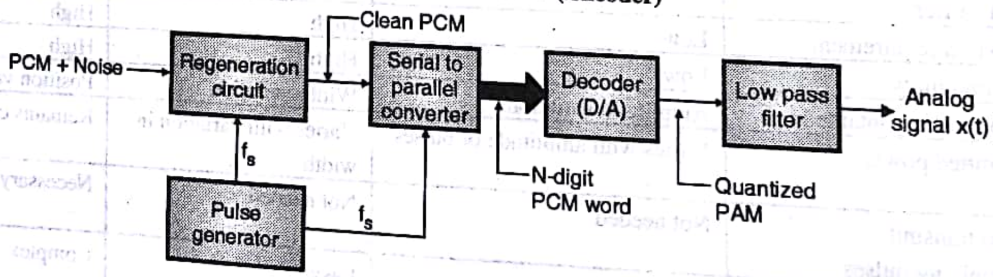


Fig. 7.2 : PCM receiver (Decoder)

Operation of PCM receiver :

A PCM signal contaminated with noise is available at the receiver input. The regeneration circuit at the receiver will separate the PCM pulses from noise and will reconstruct the original PCM signal. The pulse generator has to operate in synchronization with that at the transmitter. Thus at the regeneration circuit output we get a "clean" PCM signal.

The reconstruction of PCM signal is possible due to the digital nature of PCM signal. The reconstructed PCM signal is then passed through a serial to parallel converter. Output of this block is then applied to a decoder. The decoder is a D to A converter which performs exactly the opposite operation of the encoder. The decoder output is the sequence of a quantized multilevel pulses. The quantized PAM signal is thus obtained, at the output of the decoder. This quantized PAM signal is passed through a low pass filter to recover the analog signal, x (t). The low pass filter is called as the reconstruction filter and its cut off frequency is equal to the message bandwidth W.

Q. 2 Draw the block diagram of a PCM system and explain the function of each block.

Dec. 04, Dec. 07, May 09, May 14, Dec. 15

Ans. : Block diagram of the PCM transmitter is as shown in Fig. 7.3.

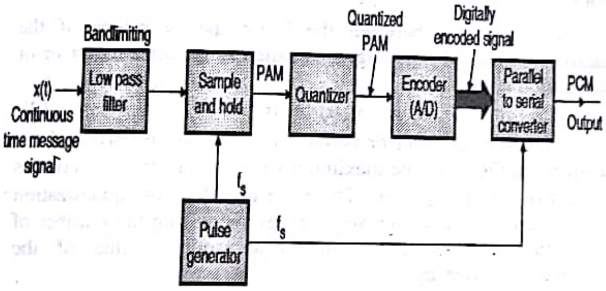


Fig. 7.3 : PCM transmitter (Encoder)

Operation of PCM transmitter :

Operation of the PCM transmitter is as follows :

The analog signal x (t) is passed through a bandlimiting low pass filter, which has a cut-off frequency f_c = W Hz. This will ensure that x (t) will not have any frequency component higher than "W". This will eliminate the possibility of aliasing.

The band limited analog signal is then applied to a sample and hold circuit where it is sampled at adequately high sampling rate. Output of sample and hold block is a flat topped PAM signal.

These samples are then subjected to the operation called "Quantization" in the "Quantizer". Quantization process is the process of approximation as will be explained later on. The quantization is used to reduce the effect of noise. The combined effect of sampling and quantization produces the quantized PAM at the quantizer output.

The quantized PAM pulses are applied to an encoder which is basically an A to D converter. Each quantized level is converted into an N bit digital word by the A to D converter. The value of N can be 8, 16, 32, 64 etc. The encoder output is converted into a stream of pulses by the parallel to serial converter block. Thus at the PCM transmitter output we get a train of digital pulses.

A pulse generator produces a train of rectangular pulses with each pulse of duration "τ" seconds. The frequency of this signal is "f_s" Hz. This signal acts as a sampling signal for the sample and hold block.

The same signal acts as "clock" signal for the parallel to serial converter. The frequency "f_s" is adjusted to satisfy the Nyquist criteria.

Waveforms :

The waveforms at various points in the PCM transmitter are as shown in Fig. 7.4

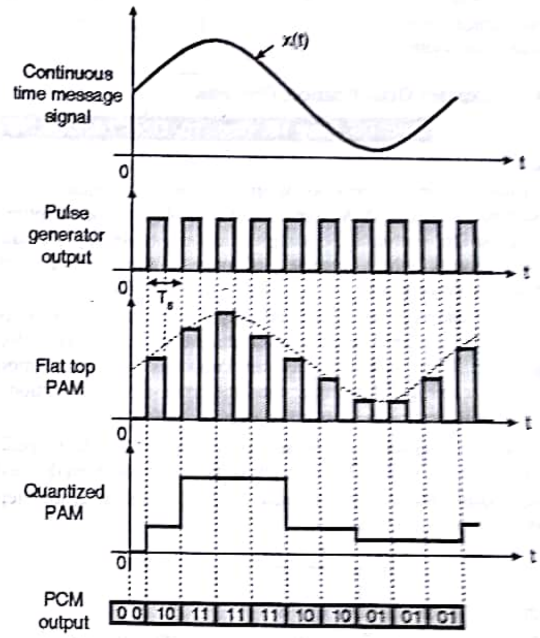


Fig. 7.4 : Waveforms at different points in PCM transmitter

Q. 3 Describe PCM and also explain decoder with block diagram.

Dec. 07, May 09, May 14, Dec. 15

Ans. : Fig. 7.5 shows the block diagram of a PCM receiver.

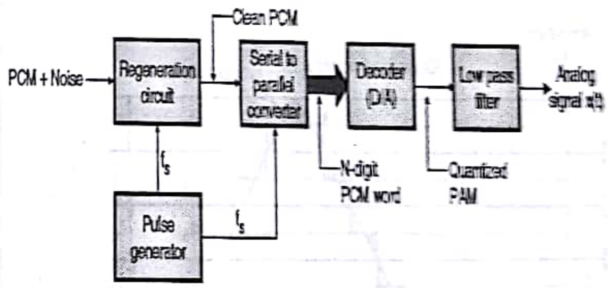


Fig. 7.5 : PCM receiver (Decoder)

Operation of PCM receiver :

A PCM signal contaminated with noise is available at the receiver input. The regeneration circuit at the receiver will separate the PCM pulses from noise and will reconstruct the original PCM signal. The pulse generator has to operate in synchronization with that at the transmitter. Thus at the regeneration circuit output we get a "clean" PCM signal.

The reconstruction of PCM signal is possible due to the digital nature of PCM signal. The reconstructed PCM signal is then passed through a serial to parallel converter. Output of this block is then applied to a decoder.

The decoder is a D to A converter which performs exactly the opposite operation of the encoder. The decoder output is the sequence of a quantized multilevel pulses. The quantized PAM signal is thus obtained, at the output of the decoder. This quantized PAM signal is passed through a low pass filter to recover the analog signal, x (t). The low pass filter is called as the reconstruction filter and its cut off frequency is equal to the message bandwidth W.

Q. 4 Explain Quantization Process

Dec. 06, May 10, Dec. 10, Dec. 14, Dec. 16

Ans. :

Quantization is a process of approximation or rounding off. The sampled signal in PCM transmitted is applied to the quantizer block. Quantizer converts the sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels.

Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level. These standard levels are known as the "quantization levels". Refer to Fig. 7.6 to understand the process of quantization.

The quantization process takes place as follows :

The input signal x (t) is assumed to have a peak to peak swing of V_L to V_H Volts. This entire voltage range has been divided into "Q" equal intervals each of size "s". "s" is called as the step ize and its value is given as,

$$s = \frac{V_H - V_L}{Q} \qquad ...(1)$$

In Fig. 7.6, the value of Q = 8

At the center of these ranges, the quantization levels q_0, q_1,...q_7 are placed. Thus the number of quantization levels is Q = 8. The quantization levels are also called as decision thresholds.

x_q (t) represents the quantized version of x (t). We obtain x_q (t) at the output of the quantizer. When x (t) is in the range Δ_0, then corresponding to any value of x (t), the quantizer output will be equal to "q_0". Similarly for all the values of x (t) in the range Δ_1, the quantizer output is constant equal to "q_1".

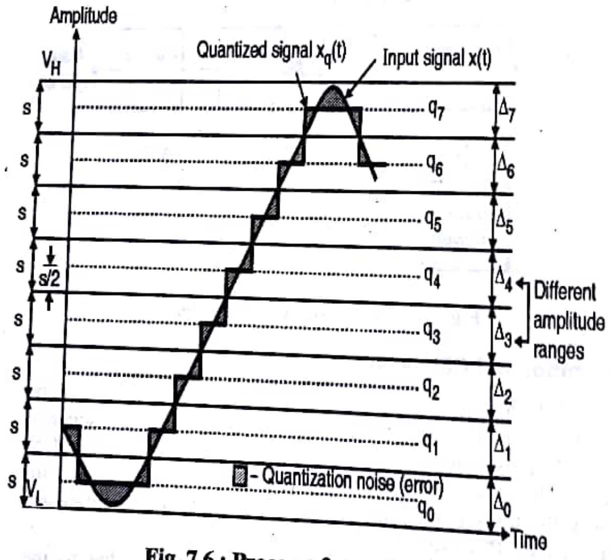


Fig. 7.6 : Process of quantization

Thus in each range from Δ_0 to Δ_7, the signal x (t) is rounded off to the nearest quantization level and the quantized signal is produced. The quantized signal x_q (t) is thus an approximation of x (t). The difference between them is called as **quantization error or quantization noise.**

This error should be as small as possible. To minimize the quantization error we need to reduce the step size "s" by increasing the number of quantization levels Q.

Why is quantization required ?

If we do not use the quantizer block in the PCM transmitter, then we will have to convert each and every sampled value into a unique digital word. This will need a large number of bits per word (N). This will increase the bit rate and hence the bandwidth requirement of the channel.

To avoid this, if we use a quantizer with only 256 quantization levels then all the sampled values will be finally approximated into only 256 distinct voltage levels.

So we need only 8 bits per word to represent each quantized sampled value. Thus the number of bits per word can be reduced. This will eventually reduce the bit rate and bandwidth requirement.

Q. 5 Explain what is meant by quantization noise ? Explain in detail.

Dec. 13

Ans.:

The difference between the instantaneous values of the quantized signal and input signal is called as quantization error or quantization noise.

$$\epsilon = x_q(t) - x(t) \qquad ...(2)$$

The quantization error is shown by shaded portions of the waveform in Fig. 7.6. The maximum value of quantization error is ± s / 2 where s is step size. Therefore to reduce the quantization error we have to reduce the step size by increasing the number of quantization levels i.e. Q. The mean square value of the quantization is given by,

$$\text{Mean square value of quantization error} = \frac{s^2}{12} \qquad ...(3)$$

The relation between the number of quantization levels Q and the number of bits per word (N) in the transmitted signal can be found as follows :

1. Because each quantized level is to be converted into a unique N bit digital word, assuming a binary coded output signal.
2. The number of quantization levels Q = Number of combinations of bits/word.

$$\therefore \quad Q = 2N \qquad ...(4)$$

Thus if N = 4 i.e. 4 bits per word then the number of quantization levels will be 2^4 i.e.16.

Signal to Quantization Noise Ratio (SNR_q)

1. This ratio is the figure of merit for the PCM systems. The signal to quantization noise ratio with a sinusoidal input signal to the PCM system is expressed as,

$$\frac{S_i}{N_q} = [1.8 + 6N] \text{ dB}$$

...For a sinusoidal signal ...(5)

2. This equation shows that the signal to quantization noise ratio is solely dependent on the number of bits per word i.e. N.
3. This ratio should be as high as possible, which can be achieved by increasing N. But this increases the bit rate and hence bandwidth of the PCM system.
4. Therefore the number of bits per word is a compromise between high SNR_q and bandwidth requirements.

Q. 6 What is companding ? **Dec. 03, May 06, Dec. 06, May 08, May 11**

Ans. : Companding is non-uniform quantization. It is required to be implemented to improve the signal to quantization noise ratio of weak signals. The quantization noise is given by,

$$N_q = s^2 / 12$$

This shows that in the uniform quantization once the step size is fixed, the quantization noise power remains constant.

But the signal power is not constant. It is proportional to the square of signal amplitude. Hence signal power will be small for weak signals, but quantization noise power is constant.

Therefore the signal to quantization noise ratio for the weak signals is very poor. This will affect the quality of signal. The remedy is to use **companding.**

Companding is a term derived from two words, compression and expansion.

Companding = Compressing + Expanding

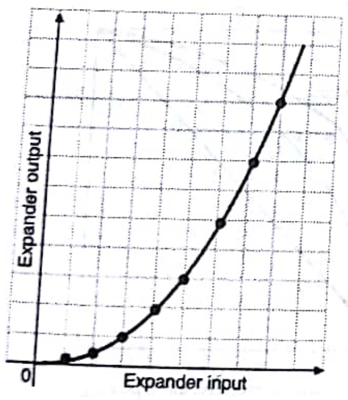
Practically it is difficult to implement the non-uniform quantization because it is not known in advance about the changes in the signal level. Therefore a trick is used. The weak signals are amplified and strong signals are attenuated before applying them to a uniform quantizer.

This process is called as "compression" and the block that provides it is called as a "compressor". At the receiver exactly opposite process is followed which is called expansion. The circuit used for providing expansion is called as an "expander". The compression of signal at the transmitter and expansion at the receiver is combined to be called as "companding". The process of companding is shown in the block diagram form in Fig. 7.7.

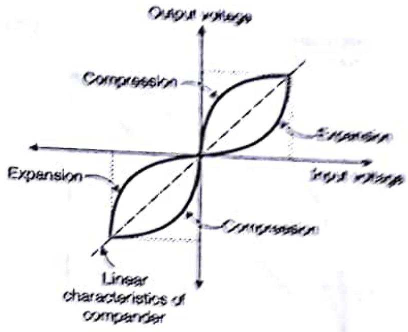
Fig. 7.7 : Model of companding

Expander Characteristics :

The expander characteristics is shown in Fig. 7.8. This characteristics is exactly the inverse of the compressor characteristics. It provides small gain of weak input signals and large gains for strong input signals. This ensures that all the artificially boosted signals by the compressor are brought back to their original amplitudes at the receiver.



(a) Expander characteristics



(b) Companding curves for PCM

Fig. 7.8

Q. 7 Show how companding reduces the quantization error ? Give compander characteristic. **May 11**

Ans. : Fig. 7.8(b) shows the compander characteristics which is the combination of the compressor and expander characteristics.

Due to the inverse nature of compressor and expander charactristics, the overall characteristics of the compander is a straight line (dotted line in Fig. 7.8(b)).

This indicates that all the boosted signals are brought back to their original amplitudes.

Companding Laws

Ideally we need a linear compressor characteristic for small amplitudes of the input signal and a logarithmic characteristic elsewhere. Practically this is achieved by using two methods :

1. μ - law companding
2. A - law companding.

1. μ - Law Companding

In the μ-law companding, the compressor characteristic is continuous. It is approximately linear for smaller values of input levels and logarithmic for high levels of input signal.

The μ -law compressor characteristic is mathematically expressed as,

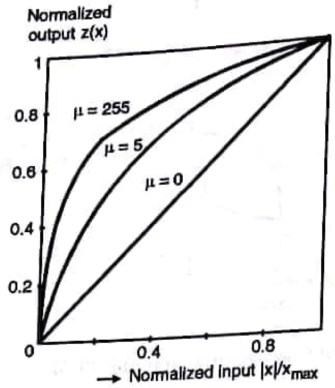
$$z(x) = (\text{sgn } x) \frac{\ln(1 + \mu |x| / x_{max})}{\ln(1 + \mu)} \quad ...(1)$$

where $0 \le |x| / x_{max} \le 1$.

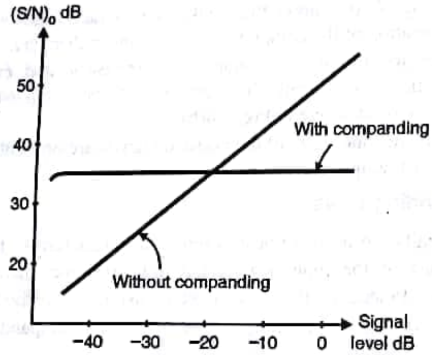
Here z (x) represents the output and x is the input to the compressor. $|x| / x_{max}$ represents the normalized value of input with respect to the maximum value x_{max}. The (sgn x) term represents ± 1 i.e. positive and negative values of input and output.

The μ -law compressor characteristics for different values of μ are as shown in Fig. 7.9(a). The practically used value of μ is 255. The characteristic corresponding to μ = 0 corresponds to the uniform quantization. It is a straight line.

The μ-law companding is used for speech and music signals. It is used for PCM telephone systems in United States, Canada and Japan. Fig. 7.9(b) shows the variation of signal to quantization noise ratio with respect to signal level, with and without companding. It is clearly seen that SNR is almost constant at all the signal levels when companding is used.



(a) Compressor characteristic of a μ-law compressor



(b) PCM performance with μ-law companding

Fig. 7.9

2. A - Law Companding

In the A-law companding, the compressor characteristic is of piecewise nature, made up of a linear segment for low level inputs and a logarithmic segment for high level inputs.

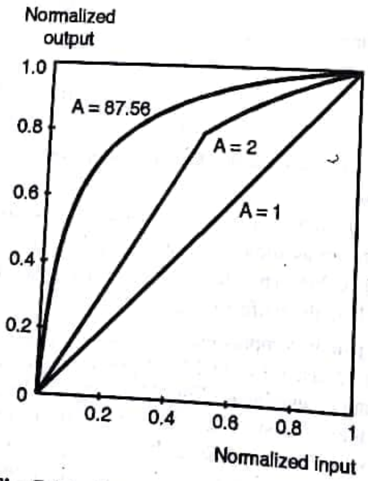


Fig. 7.10 : Compressor characteristics of A-law compressor

Fig.7.10 shows the A-law compressor characteristics for different values of A. Corresponding to A = 1 we observe that the characteristic is linear which corresponds to a uniform quantization.

The practically used value of "A" is 87.56. The A-law companding is used for PCM telephone systems in Europe. The linear segment of the characteristics is for low level inputs whereas the logarithmic segment is for high level input. It is mathematically expressed as,

$$\frac{z(x)}{x_{max}} = \begin{cases} \dfrac{A\,|x|/x_{max}}{1+\log_e A} & 0 \le \dfrac{|x|}{x_{max}} \le 1 \\[2ex] \dfrac{1+\log_e [A\,|x|/x_{max}]}{1+\log_e A} & \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases} \quad ...(2)$$

Q. 8 Explain A-law and μ-law companding. **Dec. 12**

Ans. :

In the μ-law companding, the compressor characteristic is continuous. It is approximately linear for smaller values of input levels and logarithmic for high levels of input signal.

The μ-law compressor characteristic is mathematically expressed as,

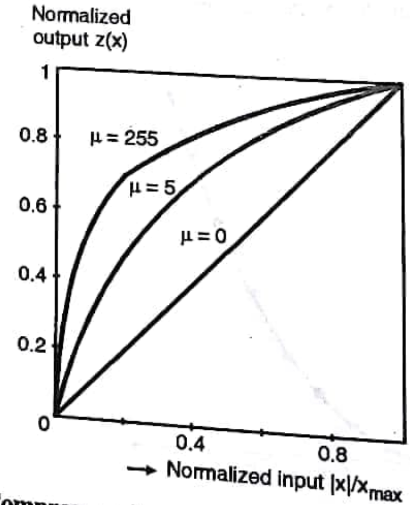
$$z(x) = (\text{sgn } x)\frac{\ln (1+\mu\,|x|/x_{max})}{\ln (1+\mu)} \quad ...(1)$$

where $0 \le |x|/x_{max} \le 1$.

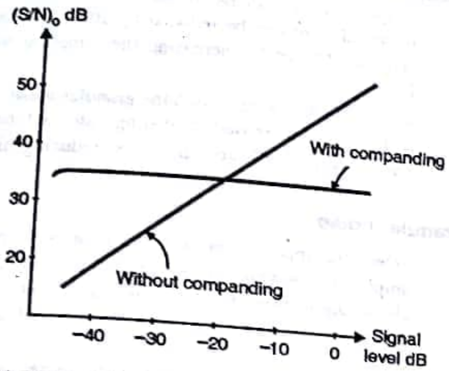
Here z (x) represents the output and x is the input to the compressor. $|x|/x_{max}$ represents the normalized value of input with respect to the maximum value x_{max}. The (sgn x) term represents ± 1 i.e. positive and negative values of input and output.

The μ -law compressor characteristics for different values of μ are as shown in Fig. 7.11(a). The practically used value of μ is 255. The characteristic corresponding to μ = 0 corresponds to the uniform quantization. It is a straight line.

The μ-law companding is used for speech and music signals. It is used for PCM telephone systems in United States, Canada and Japan. Fig. 7.11(b) shows the variation of signal to quantization noise ratio with respect to signal level, with and without companding. It is clearly seen that SNR is almost constant at all the signal levels when companding is used.



(a) Compressor characteristic of a μ-law compressor



(b) PCM performance with μ-law companding

Fig. 7.11

Q. 9 Explain A-law and μ-law companding. **Dec. 12**

Ans. :

In the A-law companding, the compressor characteristic is of piecewise nature, made up of a linear segment for low level inputs and a logarithmic segment for high level inputs.

Fig. 7.12 shows the A-law compressor characteristics for different values of A. Corresponding to A = 1 we observe that the characteristic is linear which corresponds to a uniform quantization.

The practically used value of "A" is 87.56. The A-law companding is used for PCM telephone systems in Europe.

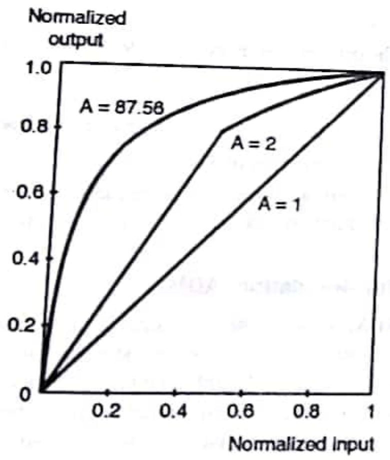


Fig. 7.12 : Compressor characteristics of A-law compressor

The linear segment of the characteristics is for low level inputs whereas the logarithmic segment is for high level input. It is mathematically expressed as,

$$\frac{z(x)}{x_{max}} = \begin{cases} \dfrac{A\,|x|/x_{max}}{1+\log_e A} & 0 \le \dfrac{|x|}{x_{max}} \le 1 \\[2ex] \dfrac{1+\log_e [A\,|x|/x_{max}]}{1+\log_e A} & \dfrac{1}{A} \le \dfrac{|x|}{x_{max}} \le 1 \end{cases} \qquad ...(2)$$

Q. 10 Explain the delta modulator transmitter and receiver with neat block diagrams.

May 03, May 04, May 07, May 09, Dec. 13, Dec. 16

Ans. : The block diagram of a delta modulator transmitter is as shown in the Fig. 7.13.

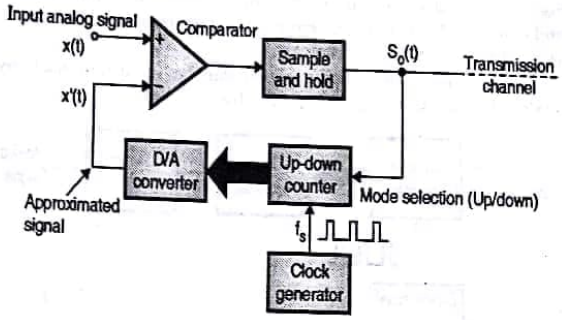


Fig. 7.13 : D.M. transmitter

x (t) is the analog input signal and x′ (t) is the quantized (approximated) version of x (t). Both these signals are applied to a comparator.

The comparator output goes high if x (t) > x′ (t) and it goes low if x (t) < x′ (t). Thus the comparator output is either 1 or 0. The sample and hold circuit will hold this level (0 or 1) for the entire clock cycle period.

The output of the sample and hold circuit is transmitted as the output of the DM system. Thus in DM, the information which is transmitted is only whether x (t) > x′ (t) or vice versa. Also note that one bit per clock cycle is being sent. This will reduce the bit rate and hence the BW.

The transmitted signal is also used to decide the mode of operation of an up/down counter. The counter output increments by 1 if $S_o(t) = 1$ and it decrements by 1 if $S_o(t) = 0$, at the falling edge of each clock pulse. This is as shown in the waveform in the Fig. 7.14.

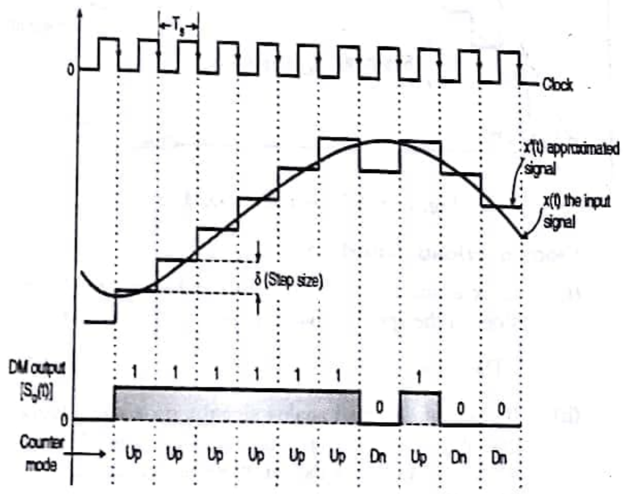


Fig. 7.14 : D.M. waveforms

The counter output is converted into analog signal by a D to A converter. Thus we get the approximated signal x′ (t) at the output of the D to A converter.

Q. 11 Explain the delta modulator transmitter and receiver with neat block diagrams.

May 03, May 04, May 09, Dec. 10, Dec. 16

Ans. : The block diagram of the D.M. receiver is as shown in Fig. 7.15. Compare it with the transmitter block diagram, you will find that it is identical to the chain of blocks producing the signal $x'(t)$ i.e. the approximated signal.

The original modulating signal can be recovered back by passing this signal through a low pass filter.

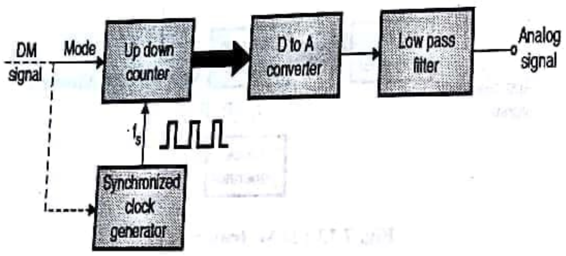


Fig. 7.15 : D.M. receiver

Q. 12 What are the drawbacks of delta modulation ? How can they be minimized.

May 03, Dec. 03, Dec. 06, May 07, Dec. 07, May 08, May 11, May 12, May 16

Ans. : The DM system is subjected to two types of quantization error or distortions :

1. Slope overload distortion and
2. Granular noise.

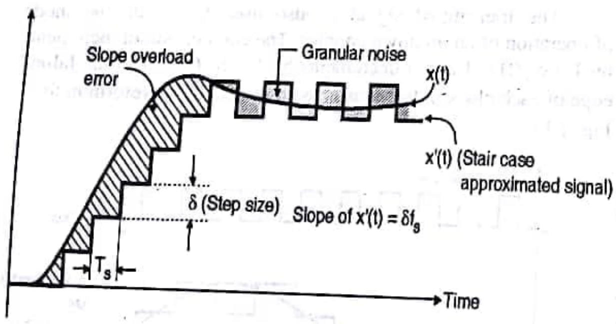


Fig. 7.16: Distortions in D.M.

1. Slope overload distortion

(i) Look at the Fig. 7.16. Due to small step size (δ), the slope of the approximated signal $x'(t)$ will be small.

$$\text{The slope of } x'(t) = \frac{\delta}{T_s} = \delta f_s \quad ...(1)$$

(ii) If slope of the input analog signal $x(t)$ is much higher than that of $x'(t)$ over a long duration then $x'(t)$ will not be able to follow the variations in $x(t)$, at all.

(iii) The difference between $x(t)$ and $x'(t)$ is called as the slope overload distortion. Thus the slope overload error occurs when slope of $x(t)$ is much larger than slope of $x'(t)$.

(iv) The slope overload error can be reduced by increasing slope of the approximated signal $x'(t)$.

(v) Slope of $x'(t)$ can be increased and hence the slope overload error can be reduced by either increasing the step size "δ" or by increasing the sampling frequency f_s.

(vi) However with increase in δ the granular noise increases and if f_s is increased, signaling rate and bandwidth requirements will go up. Thus reducing the slope overload error is not easy.

2. Granular noise

(i) When the input signal $x(t)$ is relatively constant in amplitude, the approximated signal $x'(t)$ will fluctuate above and below $x(t)$ as shown in Fig. 7.16. The difference between $x(t)$ and $x'(t)$ is called as granular noise.

(ii) The granular noise is similar to the quantization noise in the PCM system. It increases with increase in the step size δ. To reduce the granular noise, the step size should be as small as possible.

(iii) However this will increase the slope overload distortion. In the linear delta modulator the step size δ is not variable. If it is made variable then the slope overload distortion and granular noise both can be controlled. A system with a variable step size is known as the Adaptive Delta Modulator (ADM).

Q. 13 What is the disadvantage of delta modulation ? Explain with a neat diagram, how it is removed in Adaptive Delta Modulation ?

Dec. 03, May 04, Dec. 10, Dec. 13, Dec. 15

Ans:

1. The two distortions discussed i.e. slope overload error and granular noise are present.
2. Practically the signaling rate with no slope overload error will be much higher than that of PCM.

The slope overload error can be reduced by using another type of delta modulation, called as Adaptive Delta Modulation (ADM).

Adaptive Delta Modulation (ADM)

In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore $x'(t)$ will catch up with $x(t)$ more rapidly.

Whenever the slope of input signal is large, the step size of the staircase approximated signal $x'(t)$ is increased. On the other hand when the input signal is varying slowly the step size is reduced. Thus the step size is adapted as per the level of input signal.

Adaptive Delta Modulation :

In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore $x'(t)$ will catch up with $x(t)$ more rapidly. The ADM transmitter is as shown in Fig. 7.17.

If you compare this block diagram with that of the linear delta modulator, then you will find that except for the counter being replaced by the digital processor, the remaining blocks are identical. Let us understand the operation of the digital processor. For that carefully see the waveforms of Fig. 7.17.

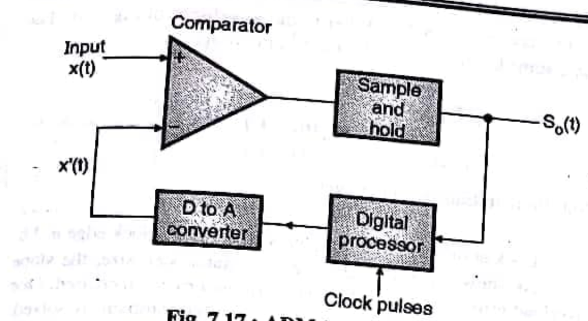


Fig. 7.17 : ADM transmitter

Operation :

In response to the k^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response i.e. $(k-1)^{th}$ clock edge.

If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "δ". If the directions are opposite then the processor will decrease the magnitude of the present step by "δ". $S_o(t)$ in the Fig. 7.17, i.e. the output of the ADM system is given as,

$S_o(t) = +1$ if $x(t) > x'(t)$ just before the k^{th} clock edge.

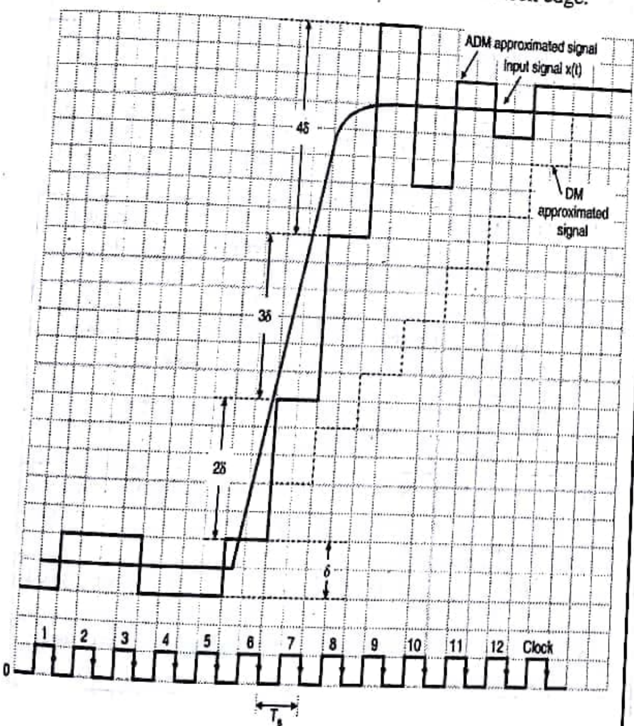


Fig. 7.18 : Waveforms of ADM

and $S_o(t) = -1$ if $x(t) < x'(t)$ just before the k^{th} clock edge.

Then the step size at the sampling instant k is given by,

$$\underbrace{\delta(k)}_{\text{Step size at } k^{th} \text{ clock edge}} = \underbrace{[\delta(k-1)]}_{\text{Step size at } (k-1)^{th} \text{ clock edge}} \; \underbrace{S_o(k)}_{\text{Output at } k^{th} \text{ edge}} + \underbrace{\delta}_{\text{Basic step size}} \; \underbrace{S_o(k-1)}_{\text{Output at } (k-1)^{th} \text{ clock edge}} \quad ...(1)$$

Let us take an example :

Refer to the waveforms of Fig. 7.18. Let us assume k = 6, i.e. consider the 6^{th} clock edge.

$$\therefore \quad k-1 = 5$$

$$\therefore \quad \delta(k-1) = \delta(5) = \delta$$

$$S_o(k) = S_o(6) = +1$$

$$S_o(k-1) = S_o(5) = +1$$

Substitute in Equation (1) to get,

$$\delta(6) = \delta + \delta = 2\,\delta \quad ...(2)$$

Look at the Fig. 7.18, the step size at the 6^{th} clock edge is 2 δ.

As shown in Fig. 7.18, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rate than the PCM system. Therefore the BW required is also less than a comparable PCM system.

Q. 14 How adaptive DM is improvement of linear DM ? Draw block diagram of adaptive DM and explain its working.

Dec. 09, May 10, Dec. 11, Dec. 12, Dec. 13

Ans. : In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore x′ (t) will catch up with x (t) more rapidly.

Whenever the slope of input signal is large, the step size of the staircase approximated signal x′ (t) is increased.

On the other hand when the input signal is varying slowly the step size is reduced. Thus the step size is adapted as per the level of input signal.

Adaptive Delta Modulation

In the ADM system, the step size is not constant. Rather when the slope overload occurs the step size becomes progressive larger and therefore x′ (t) will catch up with x(t) more rapidly.

The ADM transmitter is as shown in Fig. 7.19.

If you compare this block diagram with that of the linear delta modulator, then you will find that except for the counter being replaced by the digital processor, the remaining blocks are identical. Let us understand the operation of the digital processor. For that carefully see the waveforms of Fig. 7.19.

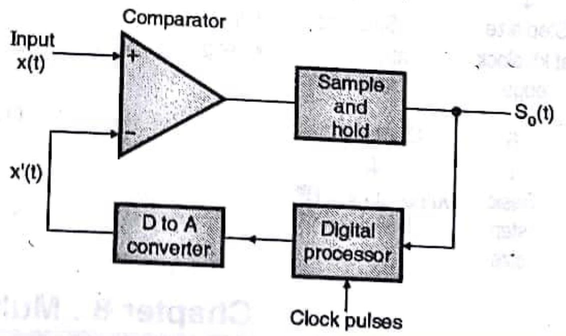


Fig. 7.19 : ADM transmitter

Operation :

In response to the k^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the i.e. $(k-1)^{th}$ clock edge.

If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "δ". If the directions are opposite then the processor will decrease the magnitude of the present step by "δ".

$S_o(t)$ in the Fig. 7.19, i.e. the output of the ADM system is given as,

$S_o(t) = +1$ if $x(t) > x'(t)$ just before the k^{th} clock edge.

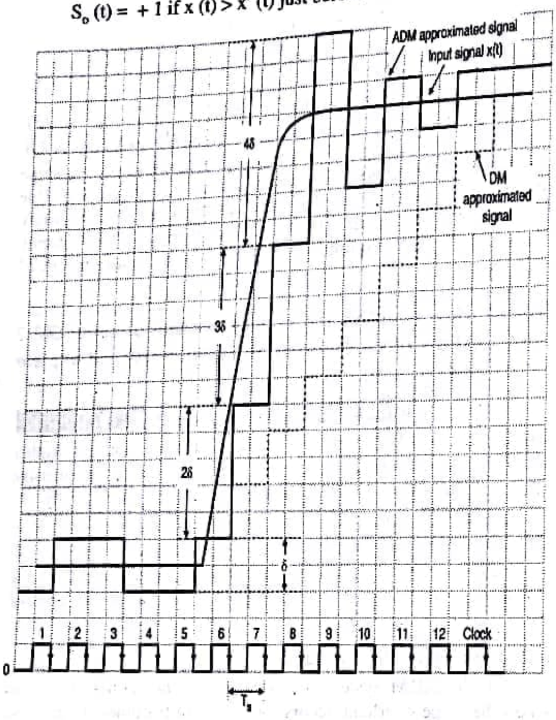


Fig. 7.20 : Waveforms of ADM

and $S_o(t) = -1$ if $x(t) < x'(t)$ just before the k^{th} clock edge.

Then the step size at the sampling instant k is given by,

$$\underbrace{\delta(k)}_{\text{Step size at } k^{th} \text{ clock edge}} = \underbrace{[\delta(k-1)]}_{\text{Step size at } (k-1)^{th} \text{ clock edge}} \; \underbrace{S_o(k)}_{\text{Output at } k^{th} \text{ edge}} + \underbrace{\delta}_{\text{Basic step size}} \; \underbrace{S_o(k-1)}_{\text{Output at } (k-1)^{th} \text{ clock edge}} \quad ...(1)$$

Let us take an example : Refer to the waveforms of Fig. 7.20. Let us assume k = 6, i.e. consider the 6^{th} clock edge.

$$\therefore \quad k-1 = 5$$
$$\therefore \quad \delta(k-1) = \delta(5) = \delta$$
$$S_o(k) = S_o(6) = +1$$
$$S_o(k-1) = S_o(5) = +1$$

Substitute in Equation (1) to get,

$$\delta(6) = \delta + \delta = 2\delta \quad ...(2)$$

Look at the Fig. 7.20, the step size at the 6^{th} clock edge is 2δ.

As shown in Fig. 7.20, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rate than the PCM system. Therefore the BW required is also less than a comparable PCM system.

Q. 15 Compare PCM and DM.

May 04, Dec. 09, May 14, May 16

Ans. : The PCM, DM and ADM all are digital pulse modulation systems. Table 7.1 shows the comparison of these systems.

Table 7.1 : Comparison of PCM, DM and ADM

Sr. No.	Parameter	PCM	DM	ADM
1.	Number of bits per sample	N can be 4, 8, 16, 32, 64 etc.	N = 1	N = 1
2.	Step size	Depends on the number of Q levels.	Step size is fixed	Step size is variable
3.	Distortions / errors	Quantization error	Slope overload and granular noise	Granular noise
4.	Signaling rate and bandwidth	Highest	Low, if the input is slow varying	Lowest
5.	System complexity	Complex	Simple	Simple
6.	Feedback from output	No feedback	Feedback is present	Feedback is present
7.	Noise immunity	Very good	Very good	Very good
8.	Use of repeaters	Possible	Possible	Possible

Chapter 8 : Multiplexing Techniques

Q. 1 Write short notes on Multiplexing techniques

May 03, May 04, May 05, Dec. 05, May 06, Dec. 06, May 10, Dec. 10, May 11, Dec. 11, May 12, Dec. 14, May 16

Ans.: Multiplexing : Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel.

Due to multiplexing it is possible to increase the number of communication channels so that more information can be transmitted. The typical applications of multiplexing are in telemetry and telephony or in the satellite communication. The concept of a simple multiplexer is illustrated in Fig. 8.1.

The multiplexer receives a large number of different input signals. Multiplexer has only one output which is connected to the single communication channel.

The multiplexer combines all input signals into a single composite signal and transmits it over the communication medium.

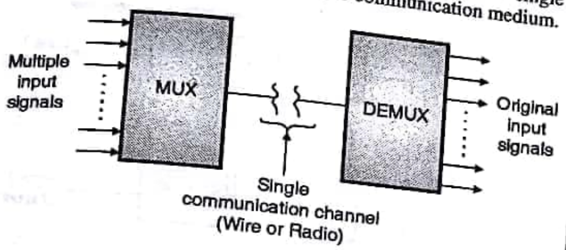


Fig. 8.1 : Concept of multiplexing

Sometimes the composite signal is used for modulating a carrier before transmission.

At the receiving end, of communication link, a demultiplexer is used to separate out the signals into their original form.

The operation of demultiplexer is exactly opposite to that of a multiplexer. Demultiplexing is the process which is exactly opposite to that of multiplexing.

Types of Multiplexing :

Different types of multiplexing are :

1. Frequency Division Multiplexing (FDM).
2. Time Division Multiplexing (TDM).
3. Wavelength Division Multiplexing (WDM).
4. Code Division Multiplexing (CDM).
5. Space Division Multiplexing (SDM).

Fig. 8.2 shows the classification of multiplexing techniques.

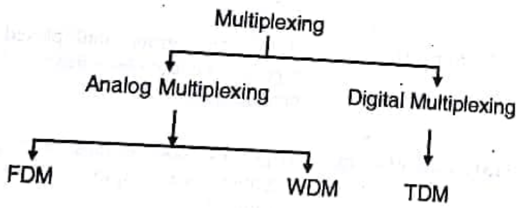


Fig. 8.2 : Classification of multiplexing techniques

Generally the FDM and WDM systems are used to deal with the analog information whereas the TDM systems are used to handle the digital information. In FDM many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth. In TDM the signals are not transmitted at a time, instead they are transmitted in different time slots. CDM and SDM are used in wireless mobile communication systems.

Q. 2 Explain TDM and FDM.

Dec. 03, May 08, May 09, May 12

Ans. :

Time Division Multiplexing (TDM) :

Most communication systems require the sharing of channels. The examples of such systems are a cable TV system with hundreds of channels sharing the same cable or one coaxial cable carrying hundreds of telephone signals on it.

Radio transmission is another example of sharing where number of radio stations share radio spectrum.

The other method is to allocate the full spectrum to each user for a short time. So all the users can use the whole spectrum one by one on the time sharing basis. This is the principle of **Time Division Multiplexing (TDM).**

Frequency Division Multiplexing (FDM)

The operation of FDM is based on sharing the available bandwidth of a communication channel among the signals to be transmitted. That means many signals are transmitted simultaneously with each signal occupying a different frequency slot within the total available bandwidth.

Each signal to be transmitted modulates a different carrier. The modulation can be AM, SSB, FM or PM. The modulated signals are then added together to form a composite signal which is transmitted over a single channel. The spectrum of composite FDM signal is shown in Fig. 8.3. Generally the FDM systems are used for multiplexing the analog signals.

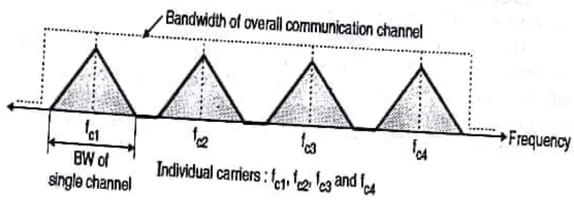


Fig. 8.3 : Spectrum of FDM signal

Q. 3 What is multiplexing in communication system ? Draw and explain in brief the transmitter and receiver of FDM.

May 09, May 12, May 16

Ans. : Fig. 8.4(a) shows the block diagram of an FDM transmitter. The signals which are to be multiplexed will each modulate a separate carrier. The type of modulation can be AM, SSB, FM or PM. The modulated signals are then added together to form a complex signal which is transmitted over a single channel.

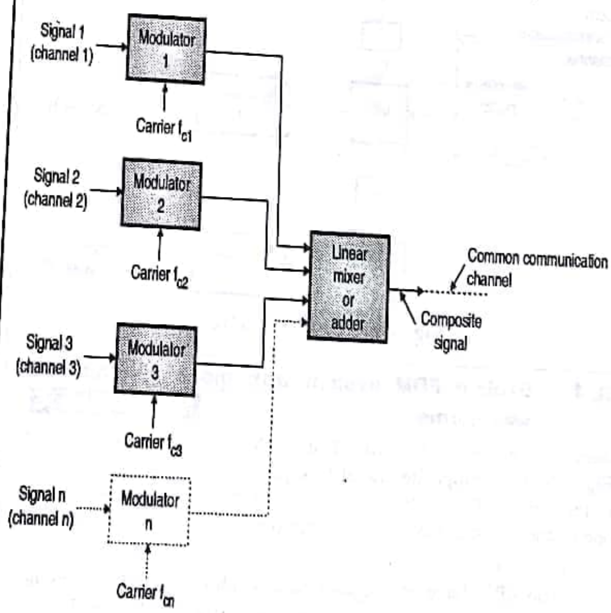


Fig. 8.4(a) : The FDM transmitter

Operation :

Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the corresponding signals.

The modulator outputs are added together in a linear mixer or adder. The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place. Different signals are thus added together in the time domain but they have their own separate identity in the frequency domain. This is as shown in the Fig. 8.4(a). The composite signal at the output of mixer is transmitted over the single communication channel as shown in Fig. 8.4(a). This signal can be used to modulate a radio transmitter if the FDM signal is to be transmitted through air. The block diagram of an FDM receiver is as shown in Fig. 8.4(b). The composite signal is applied to a group of Band Pass Filters (BPF).

Each BPF has a center frequency corresponding to one of the carriers used in the transmitter. i.e. f_{c1}, f_{c2}, ...f_{cn} etc.

The BPFs have an adequate bandwidth to pass all the channel information without any distortion. Each filter will pass through only its channel and reject all the other channels. Thus all the multiplexed channels are separated out. The channel demodulator then removes the carrier and recovers the original signal back.

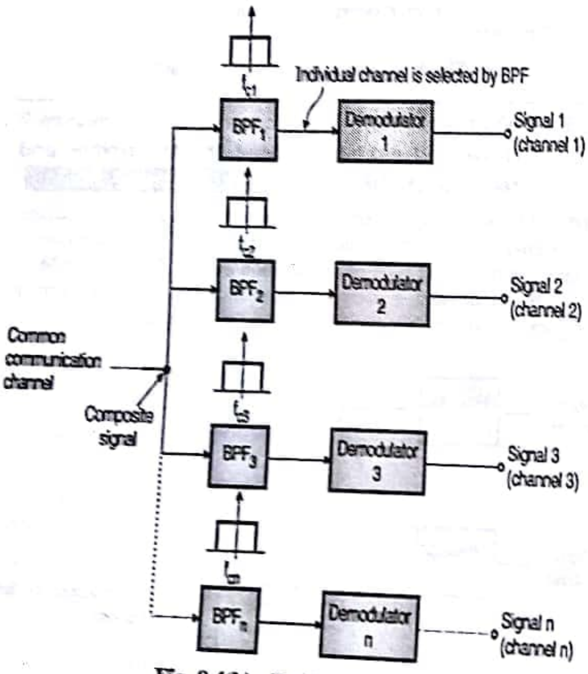


Fig. 8.4(b) : FDM receiver

Q. 4 Explain FDM system with block diagram and waveforms. **May 09, May 16**

Ans. : The block diagram of an FDM receiver is as shown in Fig. 8.5. The composite signal is applied to a group of Band Pass Filters (BPF). Each BPF has a center frequency corresponding to one of the carriers used in the transmitter

i.e. f_{c1}, f_{c2}, ...f_{cn} etc.

The BPFs have an adequate bandwidth to pass all the channel information without any distortion. Each filter will pass through only its channel and reject all the other channels. Thus all the multiplexed channels are separated out. The channel demodulator then removes the carrier and recovers the original signal back.

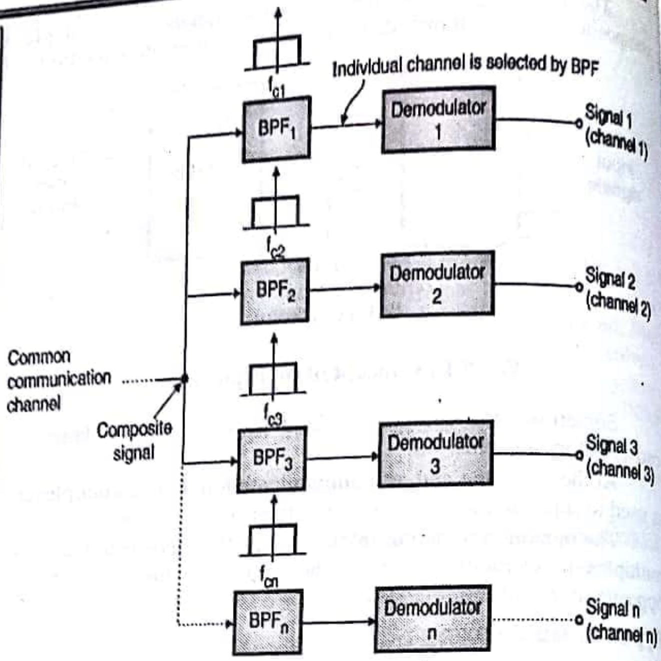


Fig. 8.5 : FDM receiver

Multiplexing Hierarchy in FDM

The different levels of multiplexing which is also called multiplexing hierarchy is as follows :

Level (1) : Basic Group. [12 voice channels multiplexed together].

↓

Level (2) : Super Group. [Upto 5 basic groups multiplexed together i.e. upto 60 voice channels].

↓

Level (3) : Master Group. [Upto 10 super groups multiplexed together i.e. upto 600 voice channels].

↓

Level (4) : Jumbo Group. [Upto 6 master groups multiplexed together i.e upto 3600 voice channels].

This hierarchy is used by AT and T and shown in Fig. 8.6

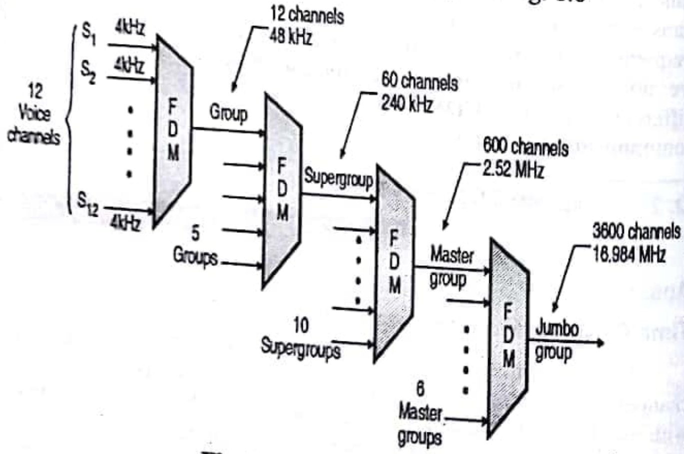


Fig. 8.6 : FDM hierarchy

The levels of multiplexing is also called as multiplexing hierarchy.

Basic group [12 voice channels]

The frequency plan for the typical basic group is as shown in Fig. 8.7. Here the 12 voice channels such as telephone channels modulate the carrier frequencies in the range of 60 to 108 kHz range. The carrier frequencies are spaced at 4 kHz from each other.

SSB modulation technique is used to save the bandwidth. Each voice channel is applied to a balanced modulator along with a carrier. The output of a balanced modulator consists of the upper and lower sidebands. Frequency plans of groups of FDM are nothing but the frequency spectrums. The frequency plan for the basic group of FDM is shown in Fig. 8.7.

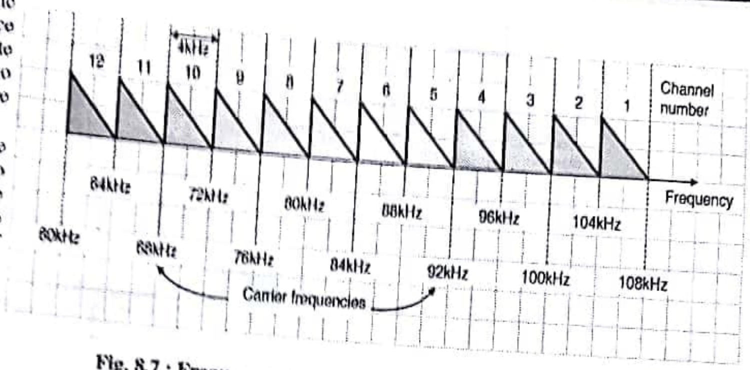


Fig. 8.7 : Frequency plan for the basic group of FDM

Super group : The frequency plan for a super group is as shown in Fig. 8.8. A super group consists of at the most 60 voice channels.

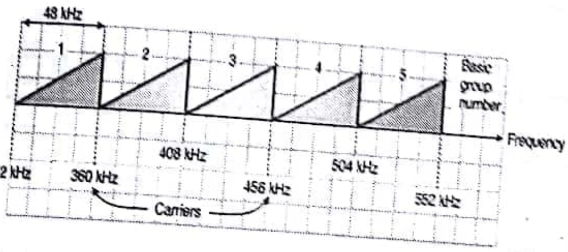


Fig. 8.8 : Frequency plan for a super group of FDM

Applications of FDM

Some of the important applications of FDM are :

1. Telephone systems.
2. AM (amplitude modulation) and FM (frequency modulation) radio broadcasting.
3. TV broadcasting
4. First generation of cellular phones used FDM.

Q. 7 Compare FDM and TDM.

Dec. 06, May 07, Dec. 09, May 10, Dec. 10, May 12, Dec. 13, Dec. 16

Ans. :

Sr. No.	FDM	TDM
1.	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signals which are to be multiplexed can occupy the entire bandwidth but they are isolated in the time domain.
2.	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signals.
3.	Synchronization is not required.	Synchronization is required.
4.	The FDM requires a complex circuitry at the transmitter and receiver.	TDM circuitry is not very complex.
5.	FDM suffers from the problem of crosstalk due to imperfect band pass filters.	In TDM the problem of crosstalk is not severe.
6.	Due to wideband fading in the transmission medium, all the FDM channels are affected.	Due to fading only a few TDM channels will be affected.
7.	Due to slow narrowband fading taking place in the transmission channel only a single channel may be affected in FDM.	Due to slow narrowband fading all the TDM channels may get wiped out.

Q. 8 What is multiplexing in communication system ? Draw the block diagram of TDM-PCM system and explain the same.

May 03, May 04, May 05, Dec. 05, May 06, Dec. 06, Dec. 07, May 10, Dec. 10, May 11, Dec. 11, Dec. 13, Dec. 15

Ans. : When a large number of PCM signals are to be transmitted over a common channel, multiplexing of these PCM signals is required. Fig. 8.9 shows the basic time division multiplexing scheme for PCM voice channels called as the **T_1 digital system.**

This system is used to convey a number of voice signals over telephone lines using wideband coaxial cable. Thus the communication medium used is a coaxial cable.

Operation of the T_1 system : The operation of the PCM-TDM system shown in Fig. 8.9 is as follows :

1. This system has been designed to multiplex 24 voice channels marked as S_1 to S_{24}. Each signal is bandlimited to 3.3 kHz, and the sampling is done at a standard rate of 8 kHz. This sampling rate is higher than the Nyquist rate. The sampling is done by the commutator switch SW_1.
2. These voice signals are selected one by one and connected to a PCM transmitter by the commutator switch SW_1, as it completes its rotation. The commutator switch remains in contact with each voice channel for a short time. Thus it samples each of the 24 channels.

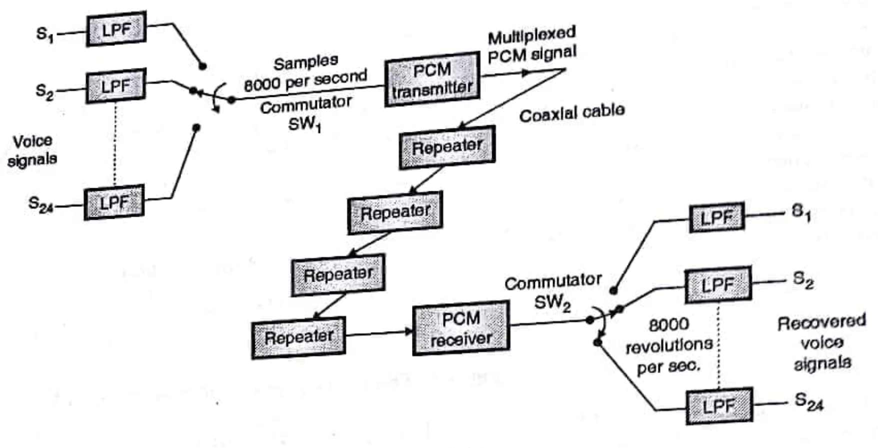


Fig. 8.9 : Block diagram of a basic PCM-TDM system

3. Each sampled signal is then applied to the PCM transmitter which converts it into a digital signal by the process of A to D conversion and companding. Each sampled voice signal is converted into an 8-bit PCM word.
4. The resulting digital waveform is transmitted over a co-axial cable. This waveform is called as the PCM-TDM signal.
5. Periodically, after every 6000 ft., the PCM-TDM signal is regenerated by amplifiers called "Repeaters". They eliminate the distortion introduced by the channel and remove the superimposed noise and regenerate a clean noise free PCM-TDM signal at their output. This ensures that the received signal is free from the distortions and noise.
6. At the destination the signal is companded, decoded and demultiplexed, using a PCM receiver. The PCM receiver output is connected to different low pass filters via the commutator switch SW_2. The LPF outputs are applied to the destination receivers (subscribers).
7. Synchronization between the transmitter and receiver commutators SW_1 and SW_2 is essential in order to ensure proper communication.

Bits/Frame

The commutators sweep continuously from S_1 to S_{24} and back to S_1 at the rate of 8000 revolutions per second (Sampling rate = 8000 samples/sec.).

This will generate 8000 samples per second of each signal (S_1 to S_{24}). Each sample is then encoded (converted) into an eight bit digital word. One complete revolution of commutator switches corresponds to generation of one frame which consists of all 24 voice channels.

Thus the digital signal generated during one complete sweep (revolution) of the commutator is given by :

$$1 \text{ Frame} \equiv 1 \text{ Revolution} = 24 \text{ Channels}$$
$$= 24 \times 8 \text{ bits} = 192 \text{ bits}$$

One frame of PAM-TDM is shown in Fig. 8.10. Each voice signal from S_1 to S_{24} is encoded into eight bits.

One frame corresponds to one revolution which is the time taken to transmit each signal once. Hence 1-frame corresponds to one-revolution of the commutator.

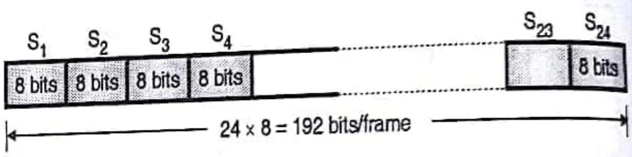


Fig. 8.10 : One frame and bits per frame

Frame Synchronization

As we have already seen, the synchronization between the transmitter and receiver commutators is essential.

Without such synchronization the receiver cannot know which received bits were generated by whom at the transmitter and are meant for which subscriber on the receiving side.

To provide such synchronization, an extra bit is transmitted preceding the 192 data bits carrying the information in each frame, as shown in Fig. 8.11.

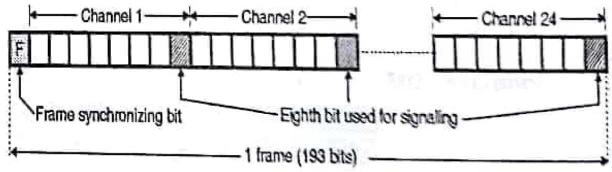


Fig. 8.11 : The PCM T_1 frame using frame synchronization and channel associated signaling

This bit is called as the frame synchronizing bit "F". Thus one frame synchronizing bit is transmitted per frame. This makes the total number of bits per frame to be 193. The time slots for the 24 signals and the extra frame synchronizing bit is as shown in Fig. 8.11.

Twelve successive F slots are used to transmit a 12 bit code. The code is 1101 1100 1000. This code is transmitted repeatedly once every 12 frames and it is used at the receiver to achieve synchronization between the transmitter and receiver commutators.

Bit rate

Bit rate means number of bits transmitted by a system per second. In the T_1 system; as each signal is sampled 8000 times per second :

1 frame (1 revolution of commutator) ≡ 1/8000 = 125 μsec.

But 1 frame consists of 193 bits.

∴ 193 bits are transmitted in 125 μsec.

$$\therefore \text{ Number of bits in 1 sec.} = \frac{193}{125 \times 10^{-6}} = 1.554 \times 10^{6}$$

$$\therefore \text{ Bit rate of } T_1 \text{ system} = 1.544 \text{ Mbits/sec.}$$

Bandwidth of T_1 system

$$\text{Minimum bandwidth } B_T = \frac{1}{2} \text{bit rate}$$

$$= \frac{1}{2} \times 1.544 \times 10^{6} = 772 \text{ kHz}$$

Duration of each bit :

$$193 \text{ bits} = 125 \ \mu S$$

$$1 \text{ bit} = (125/193) \ \mu S = 0.6476 \ \mu S$$

Chapter 9 : Line Codes

Q. 1 Write short notes on Line codes **Dec. 14**

Ans. :

In this type of encoding, the digital data which is normally binary in nature is converted into a sequence of discrete, discontinuous voltage pulses (digital signal).

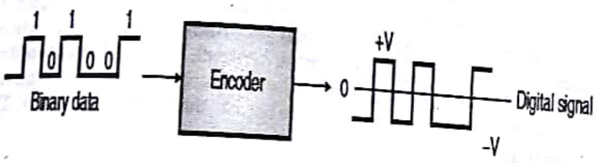


Fig. 9.1 : Digital to digital conversion

The digital data at the input of the encoder may not be suitable for transmission over a longer distance. Hence it is converted into the digital signal which is more suitable for long distance communication. The digital signals at the output of the encoder are known as the line codes.

Definition of Line Coding

The line coding is defined as the process of converting binary data, a sequence of bits to a digital signal. The digital data such as text, numbers, graphical images, audio and video are stored in computer memory in the form of sequences of bits. Line coding converts these sequences into digital signals as shown in Fig. 9.2.

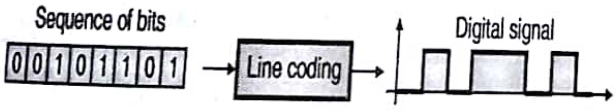


Fig. 9.2

Some Important Characteristics of Line Coding

Some of the important characteristics of line coding are :

1. Signal level and data level
2. Pulse rate and bit rate
3. DC component
4. Self synchronization

Q. 2 Write short notes on Properties of line codes. **Dec. 13**

Ans. :

Following are some of the important properties of line codes :

1. All the cable systems and other communication systems, do not allow transmission of a dc signal. Therefore the line signal must have a zero average (dc) value. NRZ bipolar formats usually satisfy this requirement. For this reason, long strings of element sequences having same polarity should not be transmitted.

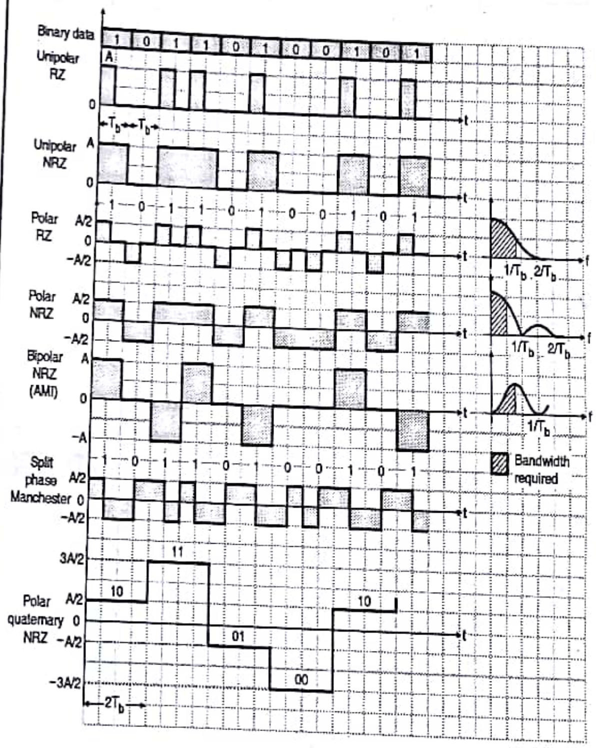


Fig. 9.3 : Various line codes

2. As the code adds redundancy, the code efficiency should be as high as possible.
3. To ensure synchronization at the receiver, the line signal should undergo a sufficient number of zero crossings that means the transmitted signal should always undergo transitions.
4. The crosstalk between channels should be minimized. To do so the amount of energy in the signal at low frequencies should be small.

Q.3 Draw the data formats (linecodes) of any five for the given binary signal 10101101. **Dec. 15**

Ans. :

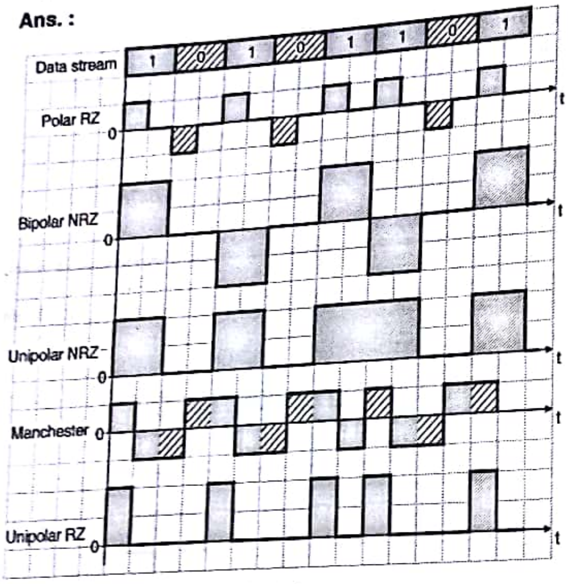


Fig. 9.4

Q. 4 The binary data 11010101 is transmitted over a baseband channel. Draw the waveform for transmitted data using the following data formats :

1. **Unipolar NRZ**
2. **Unipolar RZ**
3. **Bipolar RZ**
4. **Split phase Manchester**
5. **Polar quaternary NRZ.** **Dec. 16**

Ans. :

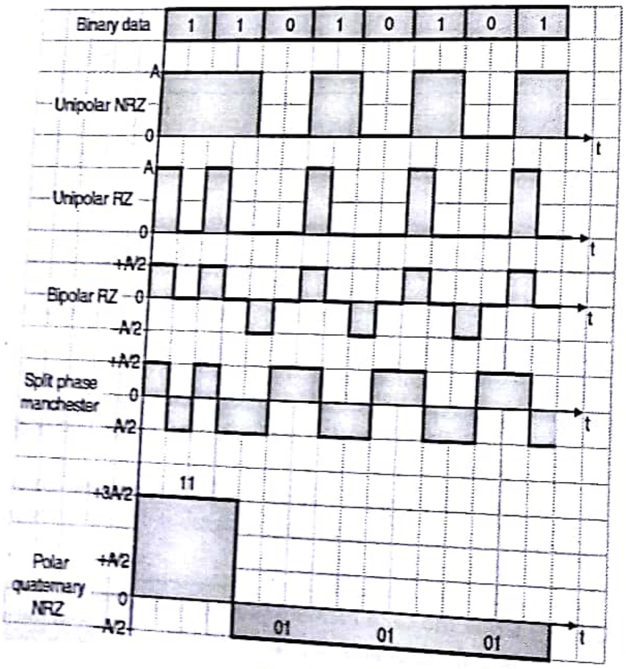


Fig. 9.5

Q. 5 What is Inter-Symbol Interference ? **May 15**

Ans. :

In a communication system when the data is being transmitted in the form of pulses (bits), the output produced at the receiver due to the other bits or symbols interferes with the output produced by the desired bit. This is called as Inter-Symbol Interference (ISI). The intersymbol interference will introduce errors in the detected signal at the receiver.

Cause of intersymbol interference

The ISI results because the overall frequency response of the system is never perfect and pulse spreading is bound to take place.

When a short pulse of duration T_b seconds is transmitted through a bandlimited transmission system, then various frequency components present in the input pulse are **differentially attenuated** and more importantly **differentially delayed** by the system.

Due to this the pulse appearing at the output of the system will be "dispersed" over an interval which is longer than "T_b" seconds as shown in Fig. 9.6. Due to this dispersion, the adjacent symbols will interfere with each in time domain other when transmitted over the communication channel. This will result in the Inter-Symbol Interference (ISI). The transmitted pulse of duration T_b seconds and the dispersed pulse of duration more than T_b seconds are shown in Fig. 9.6.

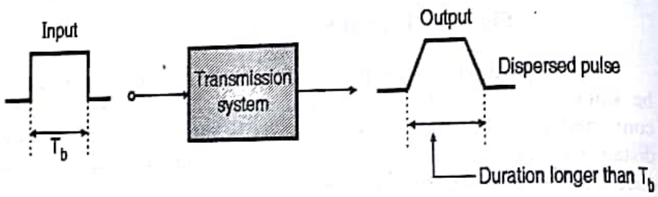


Fig. 9.6 : Cause of ISI

Effect of ISI : If the ISI and noise are absent totally then, the transmitted bit can be decoded correctly at the receiver. However errors will be introduced due to presence of ISI at the receiver output.

Due to this the receiver can make an error in deciding whether it has received a logic 1 or a logic 0. Another effect of ISI is the cross talk which may take place due to overlapping of the adjacent pulses due to spreading. It is necessary to use the special filters called equalizers in order to reduce ISI and its effect.

Remedy to Reduce the ISI

It has been proved that the function which produces a zero intersymbol interference is a "sinc function". Thus instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero. Using the since pulse for transmission is known as "Nyquist Pulse Shaping". The sinc pulse transmitted to have a zero ISI is shown in Fig. 9.7(a).

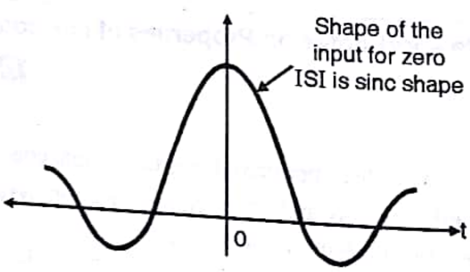
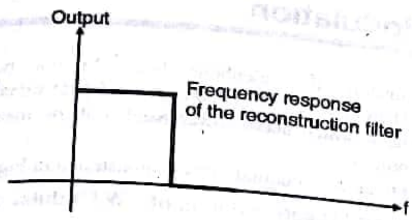


Fig. 9.7(a) Ideal pulse shape for zero ISI



(b) Frequency response of the filter

Fig. 9.7

We know that Fourier transform of a sinc pulse is a rectangular function. Therefore to preserve all the frequency components, the frequency response of the filter must be exactly flat in the pass band and zero in the attenuation band as shown in Fig. 9.7(b).

This type of filter is practically not available. Therefore practically the frequency response of the filter is modified as shown in Fig. 9.8 with different roll off factors "α" to obtain the practically achievable filter response curves.

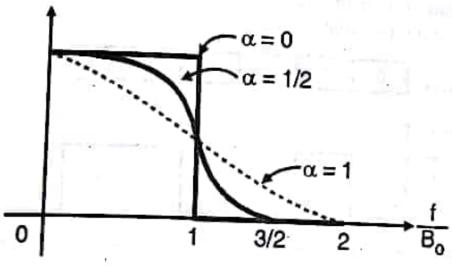
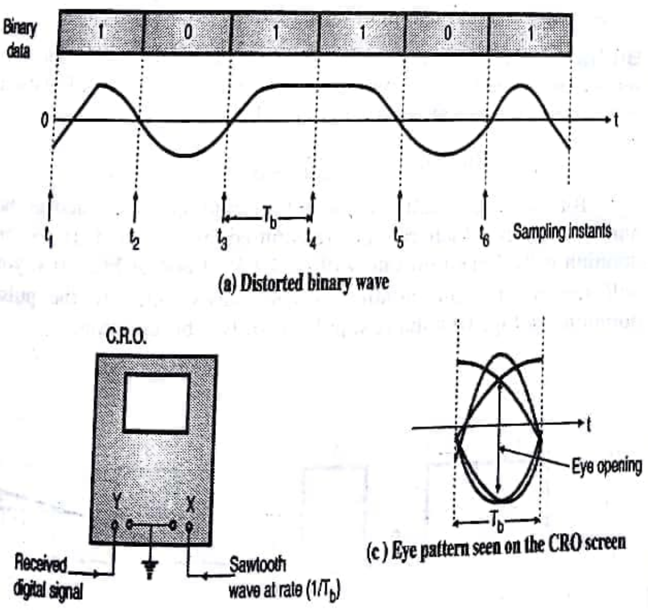


Fig. 9.8 : Practical filter characteristics

Q. 6 Explain Eye pattern with neat diagram.

Dec. 13, Dec. 14, Dec. 16

Ans. :



(a) Distorted binary wave

(b) Oscilloscope connections

(c) Eye pattern seen on the CRO screen

Fig. 9.9 : Obtaining eye pattern

Eye pattern is a pattern displayed on the screen of a Cathode Ray Oscilloscope (C.R.O.). The shape of this pattern is very similar to the shape of human eye. Therefore it is called as **eye pattern.** Eye pattern is used for studying the Inter-Symbol Interference (ISI) and its effects on various communication systems.

The eye pattern is obtained on the C.R.O. by applying the received signal to vertical deflection plates (Y-plates) of the C.R.O. and a sawtooth wave at the transmission symbol rate i.e. $(1 / T_b)$ to the horizontal deflection plates (X-plates) as shown in Fig. 9.9(c).

The received digital signal and the corresponding oscilloscope display are as shown in Figs. 9.9(a) and (c) respectively. The resulting oscilloscope display shown in Fig. 9.9(c) is called as the "eye pattern". This is due to its resemblance to the human eye.

The region inside the eye pattern is called as the eye opening. The eye pattern provides very important information about the performance of the system. The information obtainable is as follows (See Fig. 9.10)..

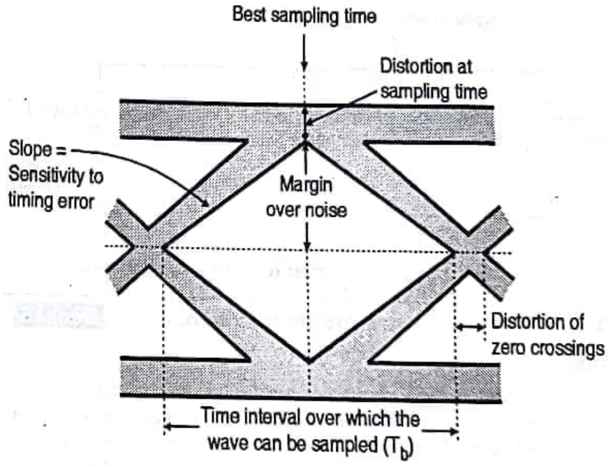


Fig. 9.10 : Interpretation of eye pattern

1. The width of the eye opening defines the time interval over which the received wave can be sampled, without an error due to ISI. The best instant of sampling is when the eye opening is maximum.
2. The sensitivity of the system to the timing error is determined by observing the rate at which the eye is closing as the sampling rate is varied.
3. The height of eye opening at a specified sampling time defines the margin over noise.
4. When the effect of ISI is severe, the eye is completely closed and it is impossible to avoid errors due to the combined effect of ISI and noise in the system.

Chapter 10 : Bandpass Modulation

Q. 1 Write short notes on : Basic digital transmission methods. May 11

Ans.: There are three basic types of modulation techniques for the transmission of digital signals.

These methods are based on the three characteristics of a sinusoidal signal; amplitude, frequency and phase. The corresponding modulation methods are then called as :

1. Amplitude Shift Keying (ASK)
2. Frequency Shift Keying (FSK)
3. Phase Shift Keying (PSK)
4. Quadrature Phase Shift Keying (QPSK) or 4-PSK.
5. Quadrature Amplitude Modulation (QAM).

QPSK is a multilevel modulation in which four phase shifts are used for representing four different symbols. At high bit rates, a combination of ASK and PSK is employed in order to minimize the errors in the received data.

This method is known as "Quadrature Amplitude Modulation (QAM)". Let us discuss these methods one by one. Fig. 10.1 shows the classification of digital to analog modulation systems.

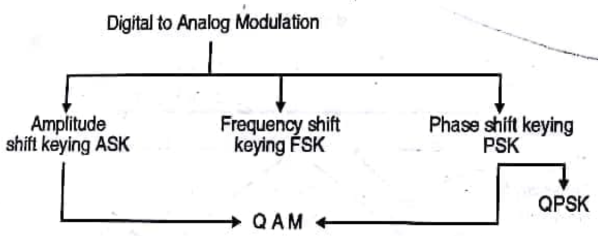


Fig. 10.1 : Types of digital to analog modulation

Q. 2 Explain the need for modulation. Dec. 15

Ans. :

The modem will modulate the digital data signal from the DTE (computer) into an analog signal. This analog signal is then transmitted on the telephone lines. The question is why can't we send the digital signal as it is on the telephone lines ? Why should we modulate it ?

Here is the answer for it. The digital data consists of binary 0s and 1s, therefore the waveform changes its value abruptly from high to low or low to high. In order to carry such a signal without any distortion being introduced, the communication medium needs to have a large bandwidth.

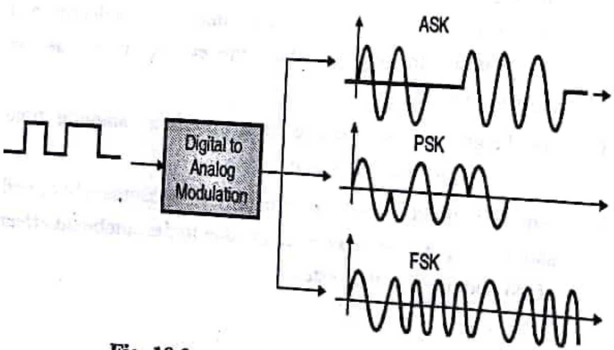


Fig. 10.2 : Digital to analog modulation

Unfortunately the telephone lines do not have high bandwidth. Therefore we have to convert the digital signal first into an analog signal which needs lower bandwidth by means of the modulation process.

Digital to analog modulation is demonstrated in Fig. 10.2.

Advantages and Disadvantages of CW Modulation :

1. The advantage of CW modulation techniques such as ASK, PSK, FSK etc. used for transmission of data is that we can use the telephone lines for transmission of high speed data. Due to the use of CW modulation the BW requirement is reduced.
2. The disadvantage of CW modulation is we need to use a MODEM alongwith every computer. This makes the system costly and complex.

Q. 3 What is Bit rate and Baud rate ? May 15

Ans. : The input data which is either analog or digital can also be represented by a digital signal. A digital signal is a discrete time signal having finite number of amplitudes. For example see the digital signal shown in Fig. 10.3. A 0 is represented by zero volt and a 1 by some positive voltage.

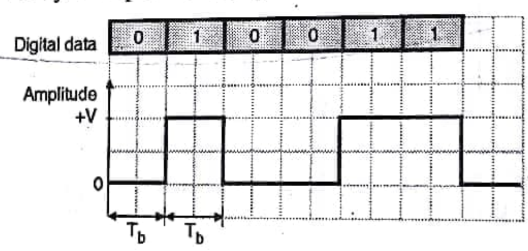


Fig. 10.3 : Digital signal

Bit Interval (T_b)

The bit interval is the time corresponding to one single bit (0 or 1). As shown in Fig. 10.4, time corresponding to a 0 or a 1 is T_b hence it is the bit interval or bit length.

Bit Rate : Bit rate is defined as the number of bits transmitted or sent in one second. It is expressed in bits per second (bps). Relation between bit rate and bit interval is as follows :

$$\text{Bit rate} = \frac{1}{\text{Bit interval}}$$

Bit rate is also called as signalling rate and is defined as the number of bits which can be transmitted in a second. If the bit duration is "T_b" then bit rate will be $1 / T_b$. Look at Fig. 10.4, you will see that the bit duration is necessarily equal to the pulse duration. In Fig. 10.4 the first pulse is of two bit duration.

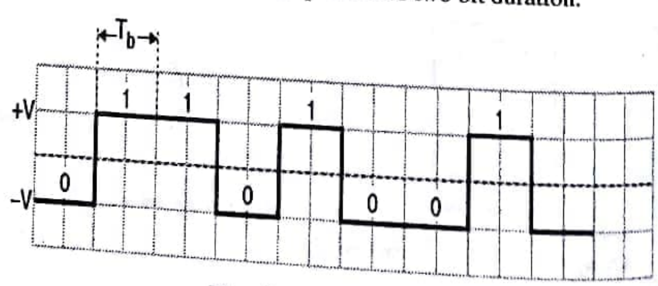


Fig. 10.4 : A bit stream

Bit rate is also called as signalling rate and it should be as high as possible. However with increase in bit rate the bandwidth of transmission medium (channel bandwidth) must be increased, in order to ensure that the signal is received without any distortion.

Bauds (or Baud Rate)

Baud is the unit of signalling speed or modulation rate or the rate of symbol transmission. It indicates the rate at which a signal level changes over a given period of time. When binary bits are transmitted as an electrical signal with two levels "0" and "1" the bit rate and the modulation rate i.e. baud rate are same. This is as shown in Fig. 10.5(a).

Note that for a two level signal (binary signal) the bit rate and bauds are equal. Now consider Fig. 10.5(b) where four different levels are used to represent the data.

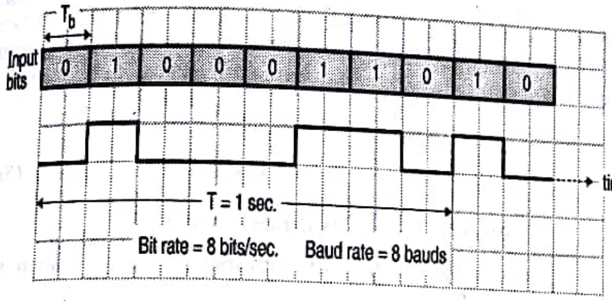


Fig. 10.5(a) : Baud rate for two level modulation

Each level is being represented by a combination of two bits i.e. 00 or 01 etc. The bit rate is therefore not equal to the baud rate. The bit rate is 8 bits/sec. but baud rate is only 4 bauds as there are 4-levels per second.

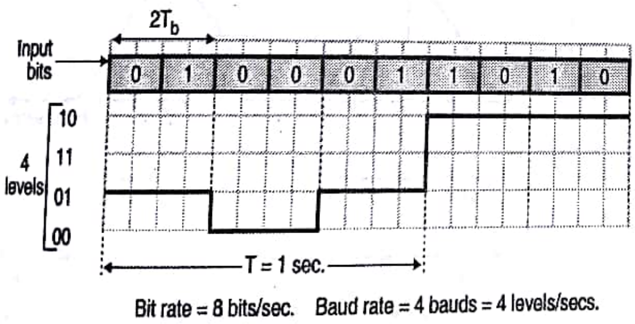


Fig. 10.5(b) : Baud rate for a four level modulation

Q. 4 What is ASK ? Explain with the help of suitable waveform. **Dec. 10, Dec. 11, May 14**

Ans.:

Amplitude Shift Keying (ASK) is the simplest type of digital CW modulation. Here the carrier is a sinewave of frequency f_c. We can represent the carrier signal mathematically as follows :

$$e_c = \sin (2\pi f_c t) \quad ...(1)$$

The digital signal from the computer is a unipolar NRZ signal which acts as the modulating signal. The ASK modulator is nothing but a multiplier followed by a band pass filter as shown in Fig. 10.6(a).

Due to the multiplication, the ASK output will be present only when a binary "1" is to be transmitted.

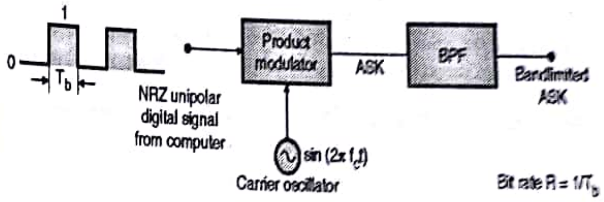


Fig. 10.6(a) : ASK generator

The ASK output corresponding to a binary "0" is zero as shown in Fig. 10.6(b).

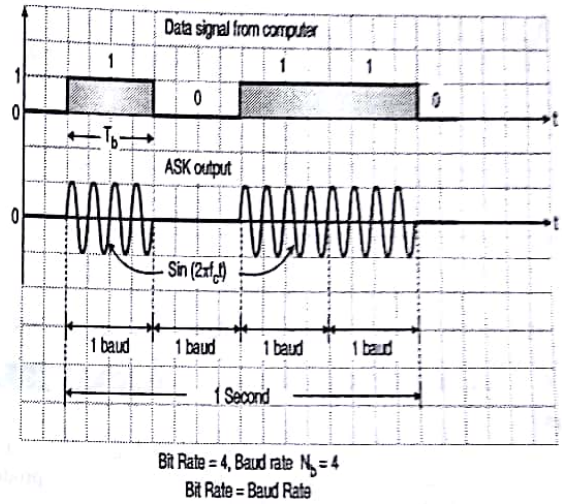


Fig. 10.6(b) : ASK waveforms

From the waveforms of Fig. 10.6(b) we can conclude that the carrier is transmitted when a binary 1 is to be sent and no carrier is transmitted when a binary 0 is to be sent.

The ASK signal can be mathematically expressed as follows :

$$V_{ASK}(t) = d \sin (2\pi f_c t) \quad ...(2)$$

where d = Data bit which can take values 1 or 0.

$$\left.\begin{aligned} \therefore\ V_{ASK}(t) &= \sin (2\pi f_c t) && \text{when } d = 1 \\ \text{and}\ V_{ASK}(t) &= 0 && \text{when } d = 0 \end{aligned}\right\} \quad ...(3)$$

Q. 5 Write short notes on : Frequency Shift keying. **Dec. 10, May 16, Dec. 16**

Ans. :

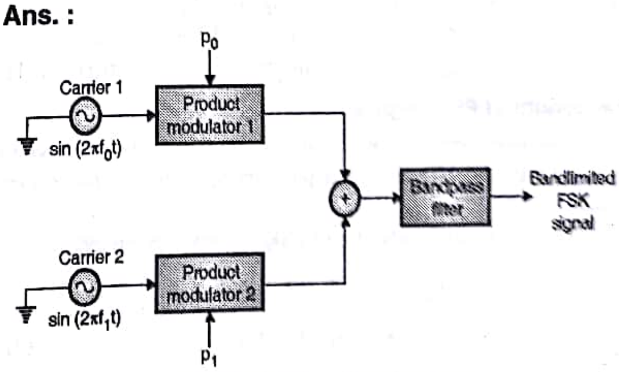


Fig. 10.7(a) : FSK generation

In "Frequency Shift Keying (FSK)", the frequency of a sinusoidal carrier is shifted between two discrete values, in response to the value (0 or 1) of the digital input signal.

One of these frequencies (f_1) represents a binary "1" and the other value (f_0) represents a binary "0". The representation of digital data using FSK is as shown in Fig. 10.7(b). Note that there is no change in the amplitude of the carrier.

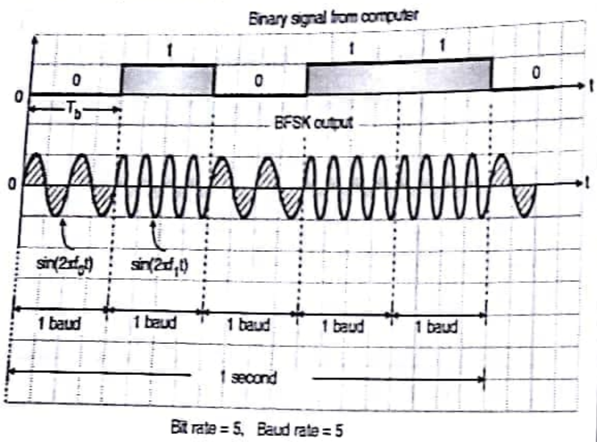


Fig. 10.7(b) : Representation of digital signal using FSK

Q. 6 Explain BFSK transmitter. May 16, Dec. 16

Ans. :

Refer to the FSK generator shown in Fig. 10.7(a). It is basically a Voltage Controlled Oscillator (VCO) which produce sinewaves at frequencies f_1 and f_0, respectively.

The oscillator outputs are applied to the inputs of two multipliers (product modulators).The other input to the two multipliers are the signals p_1 and p_0. The relation between p_1 p_0, and the data bit d (t) is as follows :

Data bit to be transmitted	Value of d (t)	Value of P_0	Value of P_1
binary 0	– 1	1	0
binary 1	+ 1	0	1

When a binary "0" is to be transmitted, $p_0 = 1$ and $p_1 = 0$ therefore the output of the first modulator only is present and the frequency of the transmitted signal is "f_0".

Similarly when a binary "1" is to be transmitted, $p_0 = 0$ and $p_1 = 1$ therefore the output of the other multiplier only is present and the frequency of the transmitted signal is "f_1".

The binary FSK signal is mathematically represented as :

$$V_{BFSK}(t) = p_0 \cdot \sin(2\pi f_0 t) + p_1 \sin(2\pi f_1 t) \quad ...(1)$$

Bandwidth of FSK Signal

The bandwidth of FSK signal is dependent on the pulse width T_b or bit rate $f_b = 1/T_b$ and the separation between the frequencies f_0 and f_1.

The maximum bandwidth of FSK system is given by,

$$B_{max} = \left(f_1 + \frac{f_b}{2}\right) - \left(f_0 - \frac{f_b}{2}\right)$$

$$= (f_1 - f_0 + f_b) \quad ...(2)$$

The bandwidth can be restricted by using a bandpass filter after the VCO in the FSK generator. The restricted bandwidth is given as :

$$B = |f_1 - f_0| + (1 + r)\frac{f_b}{2} \quad ...(3)$$

Where "r" is the factor related to the filter characteristics and its value lies between 0 and 1.

The separation between f_1 and f_0 is kept at least 2 $f_b/3$. Substitute this value in Equation (3) to get,

$$B_{max} = \frac{2}{3} f_b + f_b = \frac{5f_b}{3} \quad ...(4)$$

This shows that FSK requires larger bandwidth than ASK and PSK. Bandwidth for FSK in terms of Baud Rate :

For FSK also bit rate is equal to baud rate. This is due to the fact that each data bit at the input is treated as a separate symbol. We can imagine the FSK spectrum to be a combination of two ASK spectrums cantered at frequencies f_1 and f_0 as shown in Fig. 10.8. From Fig. 10.8 the expression for bandwidth is given by,

$$BW = \frac{N_b}{2} + (f_1 - f_0) + \frac{N_b}{2}$$

$$BW = (f_1 - f_0) + N_b \quad ...(5)$$

Where N_b = Baud rate = Bit rate = f_b

Minimum bandwidth will correspond to the situation in which $(f_1 - f_0) = N_b$

$$\therefore \; BW(min) = N_b + N_b = 2N_b = 2f_b \quad ...(6)$$

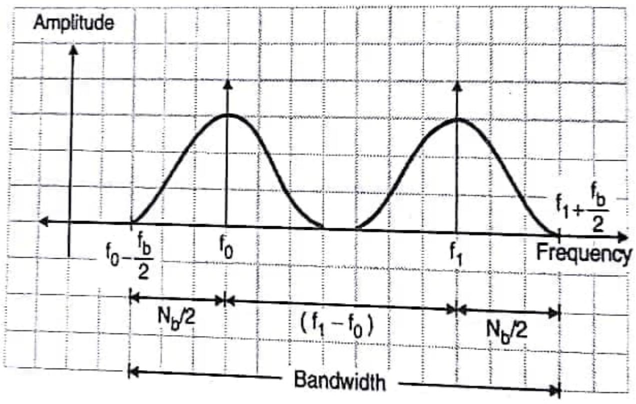


Fig. 10.8 : Spectrum of FSK

Q. 7 Explain the generation and detection of FSK signal. May 16

Ans. : The FSK receiver block diagram is as shown in Fig. 10.9(a). It is supposed to regenerate the original digital data signal from the FSK signal at its input.

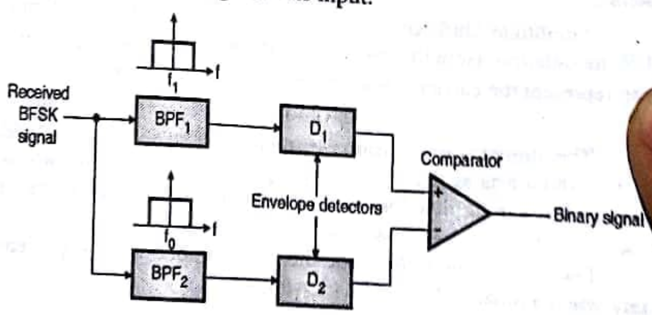


Fig. 10.9(a) : FSK receiver

The receiver consists of two band pass filters one with center frequency "f_0" and the other with a center frequency of "f_1". The envelope detectors are simple diode detectors which rectify and filter their inputs, to generate a dc voltage proportional to the ac input. Suppose a binary "1" is received. That means the received signal will be,

$$V_{BFSK}(t) = \sin(2\pi f_1 t) \quad ...(7)$$

Thus the BPF_1 will pass this signal to D_1. The output of BPF_2 will be 0, hence the output of D_2 is zero. Therefore the comparator output will be positive representing a logic "1".

Similarly if a binary "0" is received, the received FSK signal will have a frequency "f_0". The output of BPF_1 will be zero. The BPF_2 will pass this signal to D_2 to produce a proportional dc voltage. Output of D_1 is zero. Therefore comparator output will be zero which represents a logic "0". Thus the original data is recovered by the receiver.

Coherent FSK Demodulator : Fig. 10.9(b) shows the block diagram of a coherent FSK receiver.

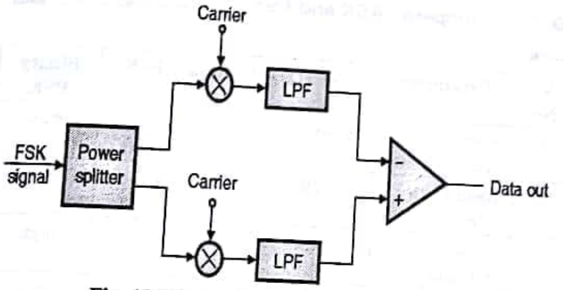


Fig. 10.9(b) : Coherent FSK demodulator

Operation : The incoming FSK signal is multiplied with the recovered (locally generated) carrier signals. The locally generated carriers have exactly the same frequency and phase of the reference carrier at the transmitter. But the two transmitted frequencies (f_H and f_L) are not generally continuous.

The multiplier outputs are passed through low pass filters and the filter outputs are applied to a comparator. The comparators output is a digital signal with 0 or 1 values. Thus the original message signal is recovered.

Q. 8 What is BPSK signal ? **May 14, May 16**

Ans. :

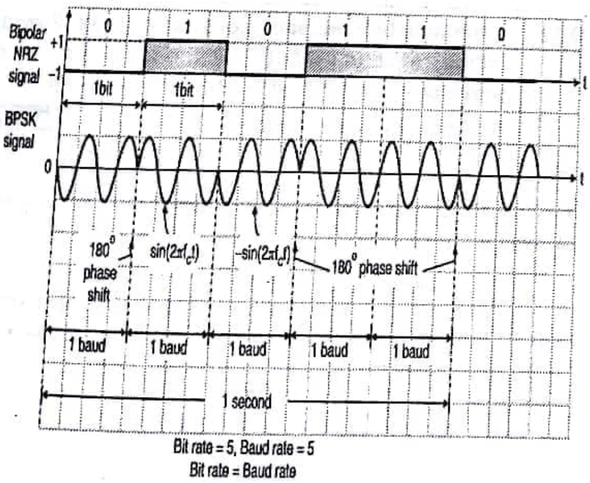


Fig. 10.10 : Binary Phase Shift Keying (BPSK)

The BPSK signal can be represented mathematically as :

$$V_{BPSK}(t) = \sin(2\pi f_c t)$$

when binary "0" is to be represented

$$\text{and } V_{BPSK}(t) = -\sin(2\pi f_c t)$$

$$= \sin(2\pi f_c t + \pi) \text{ when binary "1" is to be represented.}$$

Combining the two conditions we can write

$$V_{BPSK}(t) = d\sin(2\pi f_c t) \quad ...(1)$$

Where $d = \pm 1$

Q. 9 Draw block diagram of BPSK generation with waveform. **Dec. 13**

Ans. : The BPSK generation takes place as shown in Fig. 10.11

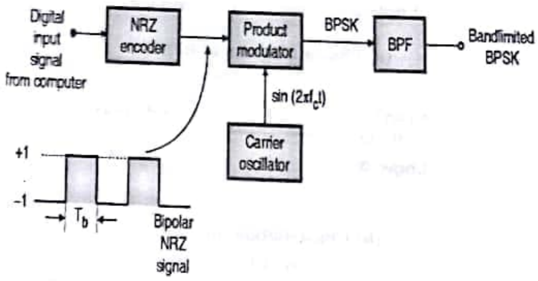


Fig. 10.11 : BPSK generation

The binary data signal (0s and 1s) is converted into a NRZ bipolar signal by an NRZ encoder, which is then applied to a multiplier (balanced modulator). The other input to the multiplier is the carrier signal ($2\pi f_c t$). The data bits 0s and 1s are converted into a bipolar NRZ signal "d" as shown in the following table.

Digital signal	Bipolar NRZ signal	BPSK output
Binary 0	d = 1	$V_{BPSK}(t) = \sin(2\pi f_c t)$
Binary 1	d = − 1	$V_{BPSK}(t) = -\sin(2\pi f_c t)$

Waveforms : Fig. 10.12 shows the input output waveforms for the diode ring modulator circuit.

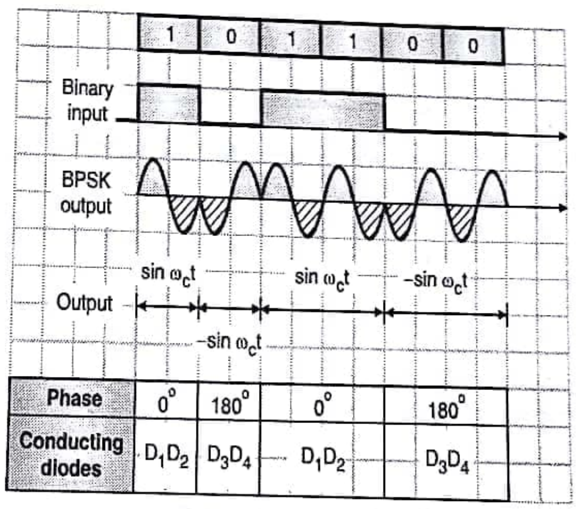


Fig. 10.12 : Waveforms of diode ring modulator

Truth table :

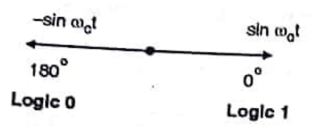
Table 10.1 gives the truth table of the diode ring modulator.

Table 10.1 : Truth table

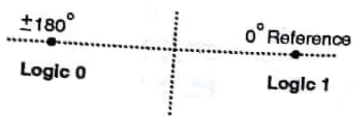
Binary Input	Output Phase
Logic 0	0°
Logic 1	180°

Phasor diagram :

Fig. 10.13(a) shows the phasor diagram of BPSK signal.



(a) Phasor diagram of BPSK



(b) Constellation diagram

Fig. 10.13

Constellation diagram : The constellation diagram of BPSK is as shown in Fig. 10.13(b).

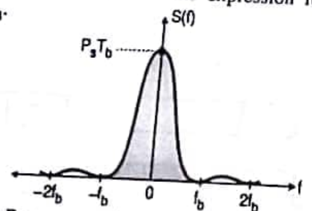
Q. 10 What is probability of error and bandwidth requirement for BPSK ? **Dec. 14**

Ans. : Probability of Error of BPSK

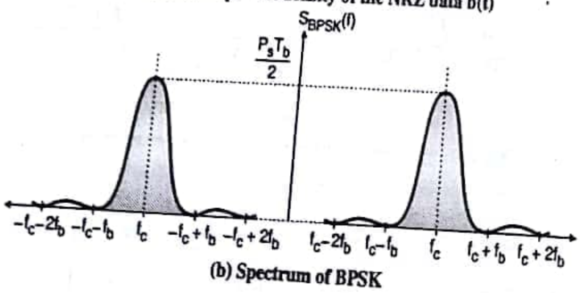
The expression for error probability is as follows :

$$\therefore \quad P_e = \frac{1}{2}\,\text{erfc}\left[\sqrt{\frac{E}{N_0}}\right] \qquad \ldots(2)$$

This is the expression for error probability of BPSK with matched filter receiver. This is the expression for the bit error probability P_B.



(a) Power spectral density of the NRZ data b(t)



(b) Spectrum of BPSK

Fig. 10.14 : Spectrum of BPSK

It indicates that the probability of error depends on the energy contents of the signal "E". It does not depend on the shape of the signal. As the energy increases, the value of erfc function will decrease and the probability of error will also reduce.

The spectrum of BPSK is as shown in Fig. 10.14.

Bandwidth of BPSK

From the frequency spectrum of BPSK signal, shown in Fig. 10.14(b), we can come to a conclusion that the bandwidth of a BPSK signal is given by,

$$BW = \text{Highest frequency} - \text{Lowest frequency in main lobe}$$

$$= (f_c + f_b) - (f_c - f_b)$$

$$\therefore \quad BW = 2 f_b \qquad \ldots(3)$$

$$\text{where} \quad f_b = 1/T_b$$

Thus the minimum bandwidth of BPSK signal is equal to twice the highest frequency contained in the baseband signal.

Q. 11 Compare : ASK and FSK **Dec. 09, Dec. 14**

Ans. :

Sr. No.	Parameter	Binary ASK	Binary FSK	Binary PSK
1.	Variable characteristic.	Amplitude	Frequency	Phase
2.	Bandwidth (Hz)	2R	$\lvert f_1 - f_0 \rvert + (1 + r)$ R	(1+ r) R
3.	Noise immunity.	low	high	high
4.	Error probability	high	low	low
5.	Performance in presence of noise.	Poor	Better than ASK	Better than FSK
6.	Complexity	Simple	Moderately complex	Very complex
7.	Bit rate	Suitable upto 100 bits/sec.	Suitable upto about 1200 bits/sec.	Suitable for high bit rates.
8.	Detection method.	Envelope	Envelope	Coherent

Q.12 If the data bit sequence consists of the following string of bits, what will be the nature of waveform transmitted by BPSK transmitter ? The data bit sequence is 1 0 1 1 1 0 1 0. **May 16**

Ans. :

The BPSK signal can also be expressed in terms of cosine wave as :

$$V_{BPSK}(t) = \sqrt{2 P_s}\, b(t) \cos \omega_c t$$

where $b(t) = \pm 1$ depending on the digital input signal.

Table 10.2 lists the values of b (t) and the transmitted signal V_{BPSK} for different bit intervals.

Table 10.2

Binary signal	1	0	1	1	1	0	1	0
b (t)	+1	−1	+1	+1	+1	−1	+1	−1
V_{BPSK} (t)	cos $\omega_c t$	− cos $\omega_c t$	cos $\omega_c t$	cos $\omega_c t$	cos $\omega_c t$	− cos $\omega_c t$	cos $\omega_c t$	− cos $\omega_c t$

The transmitted BPSK signal is as shown in Fig. 10.15

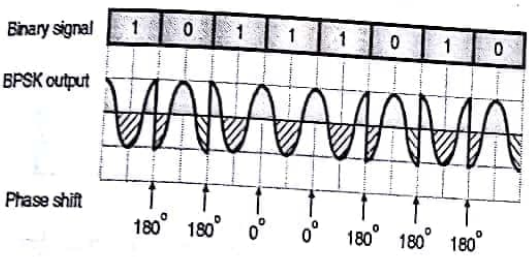


Fig. 10.15

Q.13 Draw the ASK, PSK and FSK waveforms for digital data 10100110. **May 12**

Ans. :

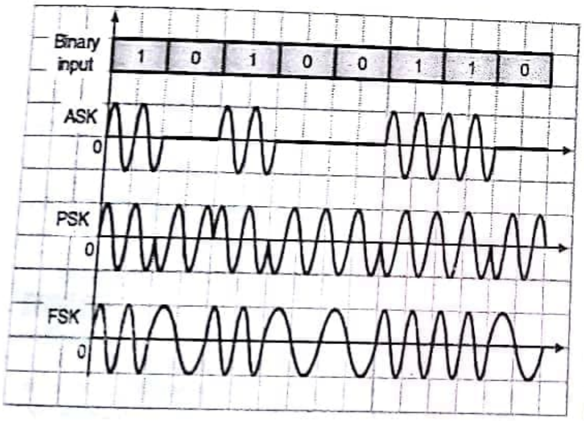


Fig. 10.16

Probability of Error of BPSK

The expression for error probability is as follows :

$$\therefore \quad P_e = \frac{1}{2}\operatorname{erfc}\left[\sqrt{\frac{E}{N_0}}\right] \qquad \ldots(2)$$

This is the expression for error probability of BPSK with matched filter receiver. This is the expression for the bit error probability P_B.

It indicates that the probability of error depends on the energy contents of the signal "E". It does not depend on the shape of the signal. As the energy increases, the value of erfc function will decrease and the probability of error will also reduce.

Q. 14 Explain the operating principle, working of differentially encoded phase shift keying modulator and demodulator. **Dec. 14**

Ans. :

The transmitter of a DEPSK system is identical to the DPSK transmitter shown in Fig. 10.17, but the receiver is completely different. The block diagram of DEPSK receiver is shown in Fig. 10.17. It shows that the signal b (t) is recovered from the received signal, using the synchronous demodulation technique.

This is same as the BPSK detection. Once the signal b (t) is recovered, it is applied to one input of an EX-OR gate.

The signal b (t) is also applied to a time delay circuit and the delayed signal $b(t - T_b)$ is applied to the other input of the EX-OR gate as shown in Fig. 10.17

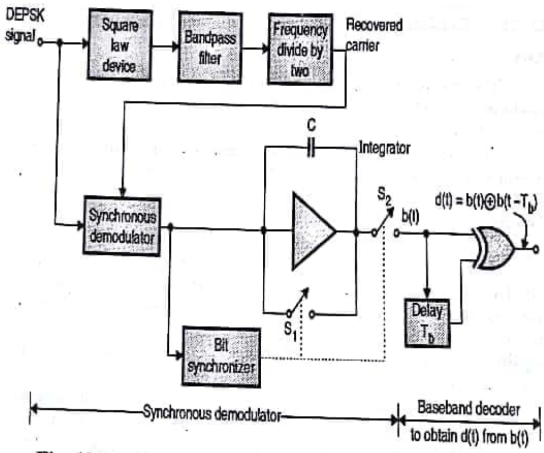


Fig. 10.17 : Receiver block diagram of DEPSK system

If $\quad b(t) = b(t - T_b)$

then output of the EX-OR gate will be 0.

$\therefore \quad d(t) = 0 \qquad \ldots \text{if } b(t) = b(t - T_b).$

and $\quad$ if $b(t) = \overline{b(t - T_b)}$ then output of the EX-OR gate will be 1.

$\therefore \quad d(t) = 1 \qquad \ldots \text{if } b(t) = \overline{b(t - T_b)}$

Errors in DEPSK system

We have seen that in DPSK there is a tendency for bit errors to occur in pairs but the single bit errors are also possible. However in DEPSK the errors will always occur in pairs. This is shown in Fig. 10.18.

In Fig. 10.18 the signals b (k), b (k − 1) and d (k) = b (k) ⊕ b (k − 1) are error free signals, whereas signal b′(k) is the same signal b(k) with one error.

Therefore its delayed version b′(k − 1) will also have one error. When b′(k) and b′ (k − 1) are added together (modulo-2 addition) in an EX-OR gate the resultant signal d′(k) has two errors as compared to the original error free signal d (k).

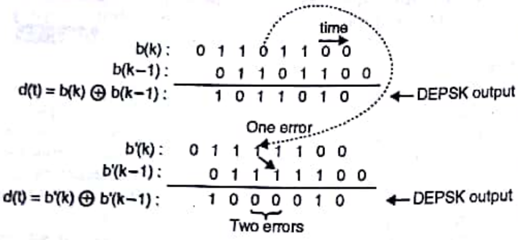


Fig. 10.18 : Errors in differentially encoded PSK occur in pairs

Q. 15 Explain QPSK. Dec. 15, Dec. 16

Ans. :

The modulation schemes discussed so far are all two level modulation. (ASK and BPSK), because they can represent only two states of the digital data (0 or 1).

Therefore the bit rate and the baud rate are same for these systems. The maximum bit rate which can be achieved using ASK, BFSK or BPSK systems does not meet the requirements of data communication systems.

This happens due to the limited bandwidth of the telephone voice channel. We can keep the baud rate same and increase the bit rate by using multilevel modulation techniques. In this type of systems, the data groups are divided into groups of two or more bits and each group of bits is represented by a specific value of amplitude, frequency or phase the carrier. QPSK (Quadrature PSK) is an example of such multilevel phase modulation.

In QPSK system two successive bits in a bit stream are grouped together to form a message and each message is represented by a distinct value of phase shift of the carrier.

The symbols and corresponding phase shifts are shown in Fig. 10.19

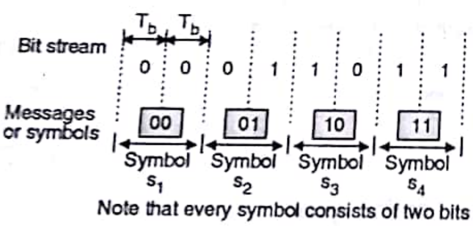


Fig. 10.19 : Grouping of bits in QPSK

Table 10.3 : Phase shift in QPSK

Symbol	Phase
00 (s_1)	0
01 (s_2)	90
10 (s_3)	180
11 (s_4)	270

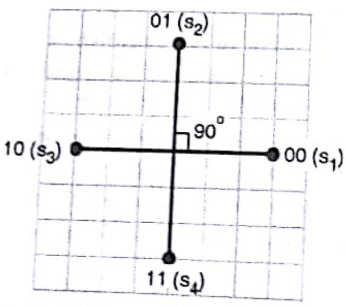


Fig. 10.20: Constellation diagram of QPSK

Each symbol or message contains two bits. So the symbol duration $T_s = 2\ T_b$. These symbols are transmitted by transmitting the same carrier frequency at four different phase shifts as shown in Table 10.3 and Fig. 10.20.

Since there are four phase shifts involved, this system is called as quadrature PSK or 4-PSK system. If the symbol 00 is to be transmitted then we have to transmit a carrier at 0° phase shift. If 01 is to be transmitted, then the same carrier is transmitted with a phase shift of 90°.

Similarly the message 10 and 11 are transmitted by transmitting the carrier at 180° and 270° respectively. This concept will be clear after referring to the QPSK waveform of Fig. 10.21

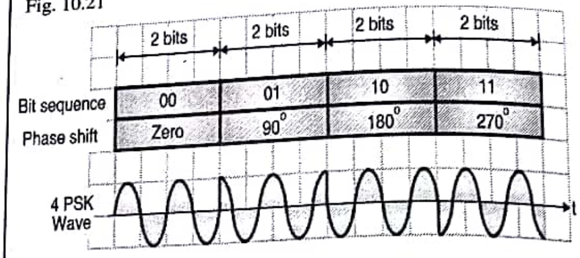


Fig. 10.21 : Waveforms of QPSK

Q. 16 What are the advantages of QPSK system ? May 14

Ans. :

Advantages of QPSK

1. Very good noise immunity.
2. Baud rate is half the bit rate therefore more effective utilization of the available bandwidth of the transmission channel.
3. Low error probability.

Due to these advantages the QPSK is used for very high bit rate data transmission.

Q. 17 Write short notes on M-ary phase shift keying Dec. 14

Ans. : In QPSK we grouped together two consecutive bits to form messages. Depending on the two bit message (00, 01, 10 or 11), a sinusoidal signal of duration equal to $2\ T_b$ which has a particular phase shift is transmitted. The QPSK signals are displaced by 90° or $\pi/2$ radians in phase with respect to each other.

The same principle can be further extended to obtain the M-ary PSK system. The M-ary PSK signals are obtained as follows :

1. Group "N" successive bits together to form N-bit symbols.
2. Each such symbol will extend over a period of N T_b, where T_b is the duration of one bit.
3. Due to the grouping of N bits per symbol, we can have $2^N = M$ possible symbols.
4. These M symbols are represented by sinusoidal signals of duration $T_s = NT_b$ which differ from one another by a phase $2\pi / M$ radians. Thus the M-ary PSK waveform can be mathematically represented as,

$$V_{M-ary\ PSK} = \sqrt{2 P_s} \cos(\omega_c t + \phi_m) \qquad ...(1)$$

where $m = 0, 1, ... (M-1)$

8 PSK System :

If three consecutive bits are grouped together to form a message or symbol, then there will be $2^3 = 8$ messages, in a PSK system.

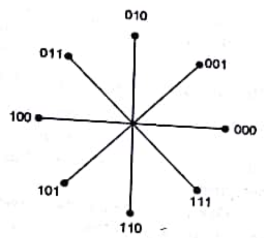
A PSK system that uses eight different phase shifts to transmit 8 symbols is known as 8 PSK system.

The messages, corresponding phases and the constellation diagram of 8-PSK is shown in Fig. 10.22

Baud rate is one third of bit rate for 8-PSK system. We can transmit thrice the number of symbols as compared to BPSK with the same bandwidth.

Symbol	Phase
000	0
001	45
010	90
011	135
100	180
101	225
110	270
111	315

(a)



(b) Constellation diagram

Fig. 10.22 : 8-PSK system

Spectrum of M-ary PSK

We can plot the PSK of the baseband M-ary signal as shown in Fig. 10.22(c).

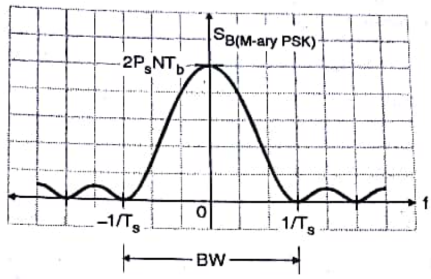


Fig. 10.22(c) : PSD of baseband M-ary PSK signal

Bandwidth of M-ary PSK

From the plot of PSD in Fig. 10.22(c).

$$BW = \frac{1}{T_s} - \left(-\frac{1}{T_s}\right) = \frac{2}{T_s}$$

But $\quad T_s = N\,T_b$

$$\therefore \quad BW = \frac{2}{NT_b}$$

But $\quad \frac{1}{T_b} = f_b$

$$\therefore \quad BW = \frac{2\,f_b}{N} \quad \text{...(2)}$$

We know that the bandwidth of a BPSK system is $2\,f_b$. The above expression tells us that with increase in number of bits per message, the bandwidth reduces.

Chapter 11 : Radiation & Propagation of Waves

Q. 1 Define the terms radiation and propagation

Ans. : In the first chapter of this book, we have shown the block diagram of a basic communication system. In that block diagram a block called "channel" is shown, between the transmitter and the receiver.

The signals transmitted by a transmitter will travel over this channel to the receiver. In the wireless radio communication, this channel is nothing but the free space.

The process of signal travel from the transmitter to receiver can be divided into two parts :

1. Radiation of the signal
2. Propagation

1. Radiation

Whenever a high frequency current flows through a conductor, the power measured on both the sides of the conductor is not same. A part of the power is dissipated in the resistance of the conductor and a part of it "escapes" into the free space. This escape of power is known as "radiation".

2. Propagation

This "radiated" power then propagates in space in the form of electromagnetic (EM) waves. The radiation and propagation of the radio waves cannot be seen. The theory of electromagnetic radiation was propounded by the British physicist J.C. Maxwell in 1857. His theory and mathematical expressions explaining the behaviour of the electromagnetic waves is universally accepted and used.

Q. 2 Explain, electromagnetic waves are transverse waves.

Ans.: The electromagnetic waves are oscillations, which propagate through free space. They travel through free space at the speed of light. These waves are known as electromagnetic waves because the electric and magnetic fields are simultaneously present.

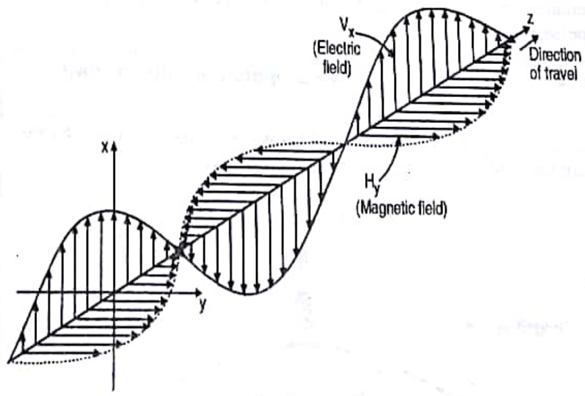


Fig. 11.1 : Transverse electromagnetic wave

The directions of these fields are perpendicular to each other and to the direction of propagation of the wave.

The electromagnetic waves are as shown in the Fig. 11.1. The propagation is somewhat similar to the outward travel of the waves in a water pond when a stone is thrown into it. However there is a fundamental difference between the two. The ripples on the surface of the water surface are **"Longitudinal"**. That means the oscillations are in the direction of propagation. The EM waves in contrast are **"transverse"** i.e. the oscillations are perpendicular to the direction of propagation. (Refer Fig. 11.1).

Electromagnetic Waves :

In this chapter we are going to discuss only about the EM waves travelling through the space. These waves are characterized by frequency and wavelength.

The EM waves are different from the other waves found in nature such as water waves or sound waves. Because EM waves consist only of time varying electric and magnetic fields and there is absolutely no need of any physical medium for their propagation.

Q. 3 Define plane of polarization.

Ans. :

The polarization of a plane EM wave is simply the orientation of the electric field vector with respect to the earth surface (i.e. looking at the horizon).

If the polarization remains constant then it is called as the **linear polarization.** The linear polarization can be of four types :

1. Horizontal polarization
2. Vertical polarization
3. Circular polarization
4. Elliptical polarization

Horizontal polarization :

If the electric field propagates in parallel with the earth surface then the EM wave is said to be horizontally polarized.

Vertical polarization :

If the electric field propagates perpendicular to the surface of earth, then the EM wave is said to be vertically polarized.

Circular polarization :

If the polarization vector rotates 360° as the EM wave travels wavelength through the space and the field strength is equal at all angles of polarization then the EM wave is said to have a circular polarization.

Elliptical polarization :

In the circular polarization, if the field strength varies with change in polarization, the wave is said to have an elliptical polarization.

Q.4 Write note on wave propagation with ground waves.

Ans. : Refer to the Fig. 11.2(a) showing the path followed by the radiated EM waves in ground wave propagation.

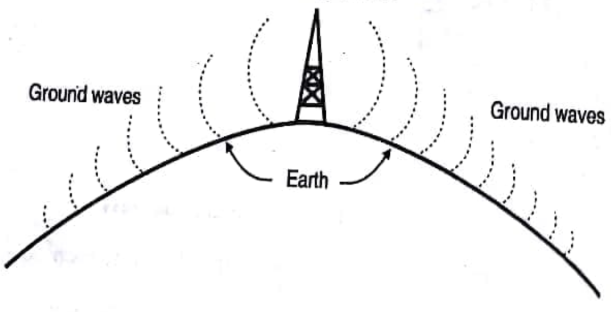


Fig. 11.2(a) : Ground wave propagation

The important points about the ground wave propagation are as follows :

1. The ground or surface wave leaves the antenna and remains closed to the earth. The ground wave will actually follow the curvature of the earth and therefore can travel a distance beyond the horizon.
2. The ground wave propagation is the strongest at the low and medium frequency ranges. The ground waves is the path chosen by the signal when the frequency is between 30 kHz and 3 MHz.
3. The ground waves must be vertically polarized to prevent short circuiting of the electric field component. The EM waves are said to be vertically polarized if all its electric intensity vectors are vertical. The EM wave shown in Fig. 11.1 is vertically polarized.

Attenuation of the Ground Waves

The ground waves get attenuated due to the following reasons :

While passing over the earth surface, the ground waves induce some current into it. Thus they loose some energy due to absorption.

Due to diffraction the wavefronts will gradually tilt over as shown in the Fig. 11.2(b). The angle of tilt (θ) goes on increasing as the ground waves progress over the surface of the earth. Eventually the wave "lies down" and "dies".

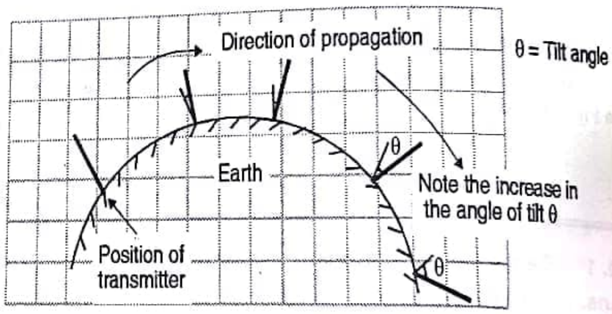


Fig. 11.2(b)

This distance depends on the type of surface, frequency of operation and the transmitted power.

The tilt angle (θ) increases with increase in frequency, hence puts a limitation on the range of transmission if the transmission takes place near the top of the medium frequency range (near 3 MHz).

Q. 5 What are ground waves and sky waves ?

Ans. :

Ground Waves

The ground waves get attenuated due to the following reasons :

1. While passing over the earth surface, the ground waves induce some current into it. Thus they loose some energy due to absorption.
2. Due to diffraction the wavefronts will gradually tilt over as shown in the Fig. 11.3 The angle of tilt (θ) goes on increasing as the ground waves progress over the surface of the earth. Eventually the wave "lies down" and "dies".
3. This distance depends on the type of surface, frequency of operation and the transmitted power.

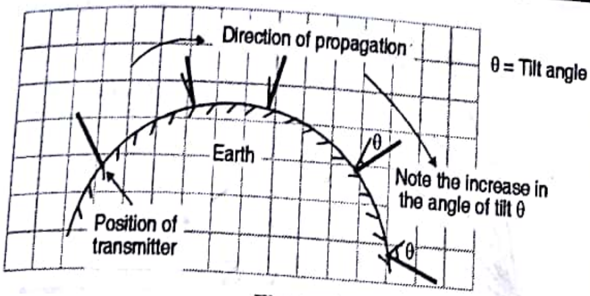


Fig. 11.3

4. The tilt angle (θ) increases with increase in frequency, hence puts a limitation on the range of transmission if the transmission takes place near the top of the medium frequency range (near 3 MHz).

Sky Waves

In sky wave propagation, the transmitted signal travels into the upper atmosphere, then it is bent or reflected back from there to earth. This bending or reflection of the signal takes place due to the presence of a layer called as ionosphere in the upper atmosphere. The layers of ionosphere and the principle of sky wave propagation is as illustrated in the Fig. 11.4

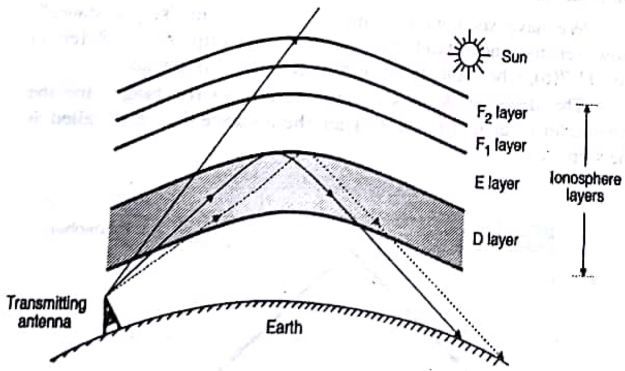


Fig. 11.4 : Principle of sky wave propagation

Q. 6 In respect to sky wave propagation explain the following terms : Virtual height And Critical Frequency (f_c)

Ans. :

Virtual Heigh

The concept of virtual height can be understood by looking at Fig. 11.5. The incident wave returns back to earth due to refraction.

In this process it bends down gradually and not sharply. But it is interesting to see that the incident and reflected rays follow exactly the same paths as though the signal would have been reflected from a surface located at greater height.

This height is called as the **virtual height**. If the virtual height of a layer is known then it is possible to find the angle of incidence required to return the wave to the ground at a selected point.

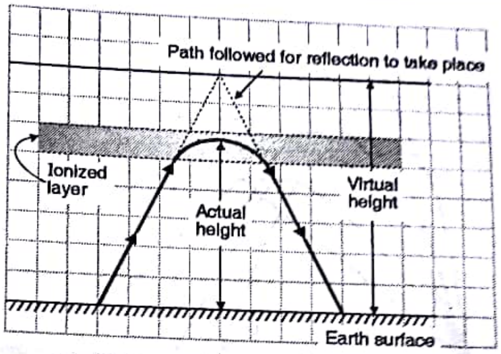


Fig. 11.5 : Virtual height of an ionized layer

Critical Frequency (f_c)

The critical frequency of a layer is defined as the maximum frequency that is returned back to the earth by that layer, when the wave is incident at an angle 90° (normal) to it. The critical frequency for the F_2 layer is between 5 to 12 MHz.

Q. 7 In respect to sky wave propagation explain the following terms : MUF

Ans.

The maximum usable frequency (MUF) is defined for a certain value of the angle of incidence θ rather than defining it at normal (θ = 90°) as in case of critical frequency.

This definition tells us that if angle θ is increased, then it is possible to operate at higher frequency than the critical frequency f_c. The MUF is given as,

$$MUF = \frac{\text{Critical frequency}}{\cos \theta} \quad ...(1)$$

$$= f_c \sec \theta \quad ...(2)$$

This equation is also known as the "Secant law". However the MUF is not defined in terms of the angle in practice. Rather it is defined as, MUF is the highest frequency that can be used for the sky wave communication between two given points on the earth.

Normally the values of MUF are in the range of 8 to 35 MHz. The highest operating frequency between two points is selected to be slightly less than MUF but it is not too less. Fig. 11.6 illustrates the effect of varying the angle of incidence keeping the frequency constant.

Skip Distance

The **skip distance** is defined as the shortest distance from a transmitter, measured along the surface of the earth at which a sky wave of fixed frequency returns back to the earth.

This frequency should be greater than the critical frequency f_c. Now refer Fig. 11.6 which shows the effect of variation in the angle of incidence θ keeping the frequency constant.

The angle of incidence θ is quite large for ray 1 and it is progressively reduced, as represented by the rays 2 and 3. Due to the reduction in angle θ, the rays return at points which are more and more close to the transmitter. In other words with decrease in angle θ, the skip distance decreases.

For the angle of incidence much less than that of ray 3, the rays 4 and 5 cannot return back to the earth surface and escape as shown in the Fig. 11.6.

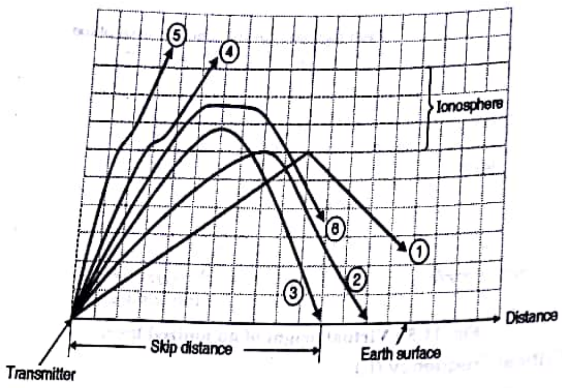


Fig. 11.6 : Effect of variation in θ at constant frequency

Finally if the angle of incidence is just slightly smaller than that corresponding to ray 3, then the wave will return but at a farther point on the earth (see ray 6). This is called as the upper ray and as shown in Fig. 11.6, it bends back very gradually as the ion density is changing very gradually at this angle. Due to its longer journey, through the ionosphere ray 6 is a weak signal.

Thus at a given frequency the angle corresponding to ray 3 will result in the shortest distance upto the point of return. Therefore this distance is the "skip distance".

Now if a higher frequency is beamed up at the angle of incidence of ray 3 then it will not return back to the earth. Thus MUF for given two points between which the communication takes place is the frequency which makes a given distance to be equal to the skip distance.

Q. 8 Explain the following terms : Skip zone and skip distance.

Ans. :

Skip Distance :

The **skip distance** is defined as the shortest distance from a transmitter, measured along the surface of the earth at which a sky wave of fixed frequency returns back to the earth. This frequency should be greater than the critical frequency f_c.

Now refer Fig. 11.7(a) which shows the effect of variation in the angle of incidence θ keeping the frequency constant.

The angle of incidence θ is quite large for ray 1 and it is progressively reduced, as represented by the rays 2 and 3. Due to the reduction in angle θ, the rays return at points which are more and more close to the transmitter. In other words with decrease in angle θ, the skip distance decreases.

For the angle of incidence much less than that of ray 3, the rays 4 and 5 cannot return back to the earth surface and escape as shown in the Fig. 11.7(a).

Finally if the angle of incidence is just slightly smaller than that corresponding to ray 3, then the wave will return but at a farther point on the earth (see ray 6). This is called as the upper ray and as shown in Fig. 11.7(a), it bends back very gradually as the ion density is changing very gradually at this angle. Due to its longer journey, through the ionosphere ray 6 is a weak signal.

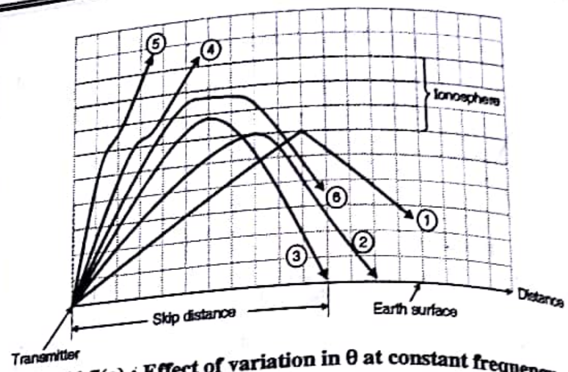


Fig. 11.7(a) : Effect of variation in θ at constant frequency

Thus at a given frequency the angle corresponding to ray 3 will result in the shortest distance upto the point of return. Therefore this distance is the "skip distance".

Now if a higher frequency is beamed up at the angle of incidence of ray 3 then it will not return back to the earth. Thus MUF for given two points between which the communication takes place is the frequency which makes a given distance to be equal to the skip distance.

Skip Zone :

We have seen what is meant by the term "skip distance". Now let us understand the meaning of skip zone. Refer to Fig. 11.7(b), where the distance A to B is the skip distance.

The distance A to C is the ground wave range, for the transmitter located at point A. Then the distance C to B is called as the skip zone.

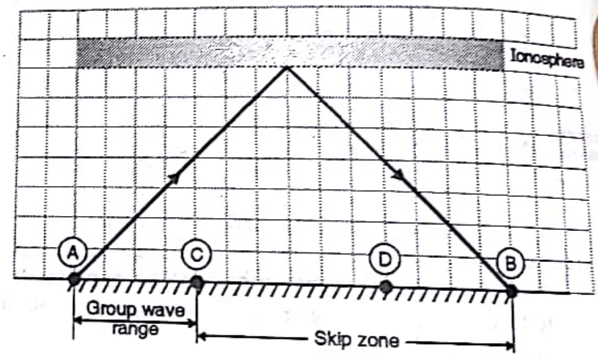


Fig. 11.7(b) : Concept of skip zone

Q. 9 Write short notes on : Diversity reception.

Ans. :

Diversity Reception :

Even though the AGC helps to a great extent to minimize the effect of fading, it is not helpful when the signal fades so much that it enters into the noise level.

The principle of diversity reception is based on the fact that the signal at different points on the earth or different frequency signals do not fade simultaneously.

There are two types of diversity reception systems :

1. Space diversity system
2. Frequency diversity system

1. **Space diversity**

   (i) In this system two or more receiving antennas are used. They are placed at points which are separated by about nine or more wavelengths.

   (ii) Receivers equal to the number of antennas are employed. The output stage of all the receivers is made common.

   (iii) As all the receivers receive the signal, the AGC from the receiver with the strongest signal at that moment is used to cut off all the other receivers. Thus only the signal from the strongest receiver is passed to the common output stage.

2. **Frequency diversity**

   (i) This system works on the similar principle of the space diversity. The signal is transmitted simultaneously at two or three different frequencies.

   (ii) Out of the signals received by different receivers which are tuned to different frequencies, only the strongest signal at a particular frequency is selected.

   (iii) Due to the use of two or three frequencies for transmitting the same signal more bandwidth is required and the frequency spectrum is wasted.]

   (iv) Therefore frequency diversity system is used only when it is not possible to use the space diversity.

Q. 10 Describe line of sight propagation for Electromagnetic waves.

Ans.: The sky wave propagation cannot take place above the frequencies of 30 MHz because the ionosphere cannot reflect back such high frequencies and the ground wave dies out near the transmitting antenna itself, due to the wavefront tilting.

Hence at frequencies above 30 MHz the space wave propagation is used.

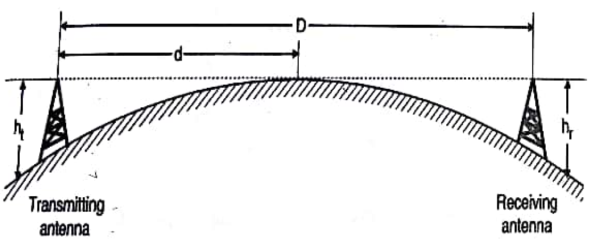


Fig. 11.8 : Space wave communication by space waves

The space wave propagation takes place by the space waves or direct waves as shown in Fig. 11.8. These waves travel in a straight line directly from the transmitting antenna to the receiving antenna. The direct or space waves are not refracted like sky waves nor do they follow the curvature of the earth like the ground waves.

Due to the straight line nature of the space waves they will at some point be blocked due to curvature of earth. If the signal is to be received beyond the horizon then the receiving antenna must be tall enough as shown in Fig. 11.8. The sky wave propagation is also called known as the line of sight propagation.

Q. 11 Write short notes on : Tropospheric scatter propagation.

Ans. : This type of propagation is also known as troposcatter or forward scatter propagation. The troposcatter propagation is used to obtain propagation of UHF signals beyond the horizon.

It uses the properties of the "troposphere" which is the nearest portion of the atmosphere. It is about 15 km above the earth surface.

The tropospheric scatter propagation can be explained as follows :

1. As shown in Fig. 11.9, two directional antennas are placed at points T and R, so that their beams will intersect each other midway, above the horizon.
2. The transmitting UHF antenna at T beams up the energy. The energy will be scattered by the troposphere in different directions as shown in Fig. 11.9 Sufficient radio energy is guided towards the receiving UHF antenna R. This happens due to the forward scatter as shown in Fig. 11.9. The receiving antenna will receive this radiation. Thus tropospheric scatter propagation can become a useful communication system.
3. The reason for the scattering is not fully known. There are two theories suggested. One of them suggests that scattering takes place due to the reflections from the "blobs" in the atmosphere. This is similar to scattering of a search light beam by the dust particles. The other theory suggests that scattering is due to the reflections from the atmospheric layers.
4. This phenomenon is a permanent and not a sporadic one. The frequencies most commonly used are 900 MHz, 2 GHz and 5 GHz.
5. The energy contents of the forward scatter which is received by the receiver is a very small percentage of the incident power. Hence a very high transmitting power is needed.

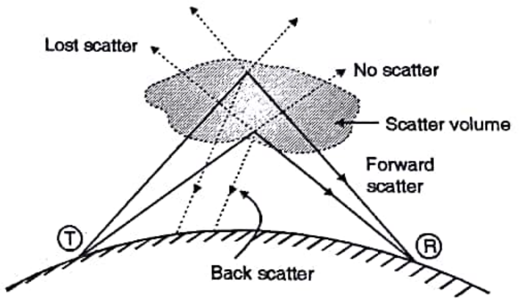


Fig. 11.9 : Tropospheric scatter propagation

Q.12 Comparison of Ground, Sky and Space Wave Propagation.

Ans. :

Sr. No.	Ground Wave Propagation	Sky Wave Propagation	Space Wave Propagation
1.	It exists in the frequency range of 30 kHz to 3 MHz.	Exists in the range of 3 MHz to 30 MHz.	Used for frequencies above 30 MHz.
2.	Used for radio broadcasting. (MW range).	Used for radio broadcasting. (SW range).	Used for TV and FM broadcasting.
3.	Ground waves are vertically polarized.	Vertically polarized.	Horizontally polarized.

Sr. No.	Ground Wave Propagation	Sky Wave Propagation	Space Wave Propagation
4.	Ground waves tilt progressively and eventually die. This limits the range of communication.	The transmission path is limited by the skip distance and curvature of earth.	The transmission path is limited by the line of sight and radio horizon.
5.	Ground waves are surface waves which travel along the surface of the earth.	Sky waves are reflected from the ionosphere. This is how communication takes place.	Space waves travel in a straight line from transmitter to receiver through space.
6.	The service range is a few hundred km.	Service range can be few thousand km.	Service range is not more than 100 km.
7.	Power loss takes place due to absorption by ground and due to tilting of waves.	Power loss due to absorption of energy by the layers of ionosphere.	Power loss due to the power absorption and scattering by the tall and massive objects.
8.	Problem of fading is not very severe.	Problem of fading is severe. Diversity reception is used.	Fading is not severe but shadow zones due to tall objects and ghost interference are serious problems.
9.	Application in MW band radio.	Short wave (SW) band radio.	TV transmission, FM transmission, Satellite communication.
10.	Limitations : Limited range, tall antennas required, high transmission power required.	Skip distance, power loss due to absorption of energy by the layers.	Distance (range) is limited, fading takes place due to rain and fog.

❑❑❑