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New Scheme Based On AICTE Flexible Curricula

Electronics & Communication Engineering IV-Semester

EC403 Analog Communication

Unit-1

Frequency domain representation of signal: Fourier transform and its properties, condition of existence, Fourier transform of impulse, step,signum , cosine, sine, gate pulse, constant, properties of impulse function. Convolution theorem (time & frequency), correlation(auto & cross), energy & power spectral density

Unit-2

Introduction: Overview of Communication system, Communication channels Need for modulation, Baseband and Pass band signals, Amplitude Modulation: Double side band with Carrier (DSB-C), Double side band without Carrier, Single Side Band Modulation, DSB-SC, DSB-C, SSB-SC,Generation of AM, DSB-SC, SSB-SC, VSB-SC & its detection,Vestigial Side Band (VSB).

Unit-3

Types of angle modulation, narrowband FM,wideband FM, its frequency spectrum, transmission BW, methods of generation (Direct & Indirect), detection of FM (discriminators: balanced, phase shift and PLL detector),pre emphasis and de-emphasis. FM transmitter & receiver: Block diagram ofFM transmitter& receiver, AGC, AVC, AFC,

Unit-4

AM transmitter& receiver: Tuned radio receiver &super heterodyne, limitation of TRF, IF frequency, image signal rejection, selectivity, sensitivity and fidelity ,Noise in AM, FM

Unit-5

Noise: Classification of noise, Sources of noise, Noise figure and Noise temperature, Noise bandwidth, Noise figure measurement, Noise in analog modulation, Figure of merit for various AM andFM, effect of noise on AM &FM receivers.

REFERENCES

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List of Experiments:

1. To analyze characteristics of AM modulator & Demodulators.
2. To analyze characteristics of FM modulators& Demodulators.
3. To analyze characteristics of super heterodyne receivers.
4. To analyze characteristics of FM receivers.
5. To construct and verify pre emphasis and de-emphasis and plot the wave forms.
6. To analyze characteristics of Automatic volume control and Automatic frequency control.
7. To construct frequency multiplier circuit and to observe the waveform.
8. To design and analyze characteristics of FM modulatorand AM Demodulator using PLL.

Unit -1

Frequency domain representation of signal: Fourier transform and its properties, condition of existence, Fourier transform of impulse, step,signum , cosine, sine, gate pulse, constant, properties of impulse function. Convolution theorem (time & frequency), correlation (auto & cross), energy & power spectral density

Time domain and frequency domain representation of signal –

Signal contains information about a variety of things and activities in the physical world. As a matter of fact, electrical signal may be represented in two equivalent forms, as a voltage signal or current signal. This means that an electrical signal may be represented either in the form of a voltage source or in the form of a current source.

Now, an electrical signal, either a voltage signal of current signal, may further be represented in two forms. These two types of representation are as –

1. Time domain representation.
2. Frequency domain representation.

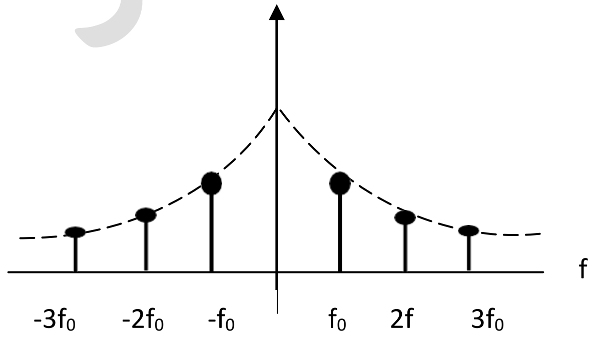
Frequency domain representation – In frequency domain, a signal is represented by its frequency spectrum.

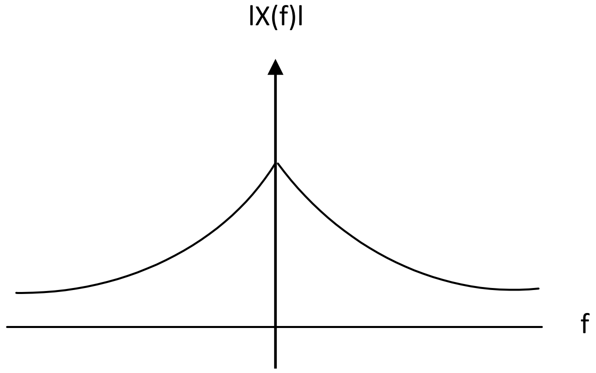
Time domain representation – In frequency domain, a signal is represented by its frequency spectrum. To obtain frequency spectrum of a signal, Fourier series and Fourier transformation are used.

Fourier series is used to get frequency spectrum of time-domain signal, when the signal is periodic function of time. With the help of frequency spectrum of series, a given periodic function of time may be expressed as the sum of an infinite number of sinusoids whose frequency is harmonically related.

Fourier Transformation –

As we know, how to represent periodic signals that are extended over the interval (-∞, ∞) using the Fourier series. Non-periodic time limited signals can also be represented by the Fourier series. However, the non-periodic signals which extended from -∞ to ∞ can be represented more conveniently using the Fourier transformation in the frequency domain.





(a) Line spectrum showing vertical lines at f_0, f_1…. (b) Continuous spectrum as f→ 0

It is possible to find the Fourier transformation of periodic signal as well. For the periodic signals, $T_0 \rightarrow \infty$. Hence the frequency $F_0 = \frac{1}{T_0} \rightarrow 0$. Therefore, the difference between the spectral components which is F_0 becomes very small and they come very close to each other. Due to this, the frequency spectrum appears to be continuous as shown in above figure (a) and (b).

Fourier transform may be expressed as

$$X(w)=F[x(t)]=\int_{-\infty}^{\infty} x(t)\, e^{-wt} dt$$

In the above equation X(w) is called the Fourier transform of x(t). In other words X(w) is the frequency domain representation of time domain function x(t). This means that we are converting a time domain signal into its frequency domain representation with the help of Fourier transform. Conversely if we want to convert frequency domain signal into corresponding time domain signal, we will have to take inverse Fourier transform of frequency domain signal. Mathematically, Inverse Fourier transforms.

$$F^{-1}[(w)] = x(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} (w) e^{wt} dw$$

Example

Q.1 Find the Fourier transform of a single-sided exponential function$e^{-at}u(t)$.

Solution: $e^{-at}u(t)$ is single sided function because her the main function e^{-at} is multiplied by unit step function u(t), then resulting signal will exist only for t>0.

$$u(t)= \{1 \text{ for } t>1$$

$$=\{0 \text{ for elsewhere}$$

Now, given that x(t)= $e^{-at}u(t)$

$$X(w)=F[x(t)]=\int_{-\infty}^{\infty} x(t)\, e^{-wt} dt$$

Or $X(w)=\int_{-\infty}^{\infty} e^{-at}\, u(t)\, e^{-wt} dt$

$$=\int_{0}^{\infty} e^{-at}\; e^{-wt} dt$$

$$=\int_{0}^{\infty} e^{-t(a+w)} dt$$

$$\frac{-1}{(a+w)}[e^{-\infty} - e\;]= \frac{-1}{(a+w)}[0-1]=\frac{1}{(a+w)}$$

To obtain the above expression in the proper form we write

$$X(w)=\frac{-1}{(a+w)} * \frac{(a-w)}{(a-w)}$$

$$X(w)=\frac{(a-w)}{(a^2+w^2)} = \frac{a}{(a^2+w^2)} - \frac{w}{(a^2+w^2)}$$

Obtaining the above expression in polar form

$$X(w)=\frac{1}{\sqrt{a^2+w^2}}\ e^{-tan^{-1}(\frac{w}{a})}$$

As we know that

$$X(w)=|(w)|e^{\varphi(w)}$$

On comparison amplitude spectrum

$$|(w)|=\frac{1}{\sqrt{a^2+w^2}}$$

$$\varphi(w)=-tan^{-1}(\frac{w}{a})$$

Properties of Continuous Time Fourier Transform (CTFS)

1. **Time Scaling Function**

   Time scaling property states that the time compression of a signal results in its spectrum expansion and time expansion of the signal results in its spectral compression. Mathematically,

   $$\text{If } x(t) \longleftrightarrow X(w)$$

   Then, for any real constant a,

   $$x(at) \longleftrightarrow \frac{1}{|a|}X(\frac{w}{a})$$

   Proof: The general expression for Fourier transform is

   $$X(w)=F[x(t)]=\int_{-\infty}^{\infty} x(t)\, e^{-wt}\,dt$$

   Now $\quad F[x(at)]=\int_{-\infty}^{\infty} x(at)\, e^{-wt}\,dt$

   Putting At=y

   We have $\quad dt=\frac{dy}{a}$

   Case (i): When a is positive real constant –

   $$F[x(at)]=\int_{-\infty}^{\infty} x(y)\, e^{-(\frac{w}{a})y}\frac{dy}{a}=\frac{1}{a}\int_{-\infty}^{\infty} x(y)\, e^{-(\frac{w}{a})y}=\frac{1}{a}X(\frac{w}{a})$$

   Case (ii): When a is negative real constant

   $$F[x(at)]=\frac{-1}{a}X(\frac{w}{a})$$

Combining two cases, we have

$$F[x(at)]=\frac{1}{|a|}X(\frac{w}{a}) \text{ Or } x(at) \longleftrightarrow \frac{1}{|a|}X(\frac{w}{a})$$

The function x(at) represents the function x(t) compressed in time domain by a factor a. Similarly, a function $X(\frac{w}{a})$ represents the function X(w) expanded in frequency domain by the same factor a.

2. **Linearity Property -** Linearity property states that Fourier transform is linear. This means that

   If $\quad x_1(t) \longleftrightarrow X_1(w)$

   And $x_2(t) \quad X_2(w) \longleftrightarrow$

Then $a_1 x_1(t) + a_2 x_2(t)$ $\longleftrightarrow$ $a_1X_1(w) + a_2X_2(w)$

3. Duality or Symmetry Property

If $x(t)$ $\longleftrightarrow$ $X(w)$

Then $X(t)$ $\longleftrightarrow$ $2\pi x(-w)$

Proof -The general expression for Fourier transform is

$$F^{-1}[(w)] = x(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty}(w)e^{wt}dw$$

Therefore,

$$x(-t)= \frac{1}{2\pi}\int_{-\infty}^{\infty}(w)e^{-wt}dw$$

$$2\pi x(-t) = \int_{-\infty}^{\infty}(w)e^{-wt}dw$$

Since w is a dummy variable, interchanging the variable t and w we have

$$2\pi x(-w) = \int_{-\infty}^{\infty}(t)e^{-wt}dw=F[X(t)]$$

Or $F[X(t)]= 2\pi x(-w)$

Or $X(t)$ $\longleftrightarrow$ $2\pi x(-w)$

For an even function $x(-w)=x(w)$

Therefore, $X(t)$ $\longleftrightarrow$ $2\pi x(w)$

4. Time Shifting property - Time Shifting property states that a shift in the time domain by an amount b is equivalent to multiplication by e^{-jwb} in the frequency domain. This means that magnitude spectrum $|X(w)|$ Remains unchanged but phase spectrum θ(w) is changed by -wb.

If $x(t)$ $\longleftrightarrow$ $X(w)$

Then $X(t-b)$ $\longleftrightarrow$ $X(w)e^{-jwb}$

Proof:

$$X(w)=F[x(t)]=\int_{-\infty}^{\infty}x(t)\, e^{-jwt}dt$$

And $F[x(t-b)]=\int_{-\infty}^{\infty}x(t-b)\, e^{-jwt}dt$

Putting (t – b) =y, so that dt=dy

$$F[x(t-b)]=\int_{-\infty}^{\infty}x(y)\, e^{-jw(b+y)}dy = \int_{-\infty}^{\infty}x(y)\, e^{-jwb}e^{-jwy}\, dy$$

Or $F[x(t-b)]=e^{-jwb}\int_{-\infty}^{\infty}x(y)\, e^{-jwy}\, dy$

Since y is a dummy variable, we have

$$F[x(t-b)]=e^{-jwb}X(w)=X(w)e^{-jwb}$$

Or $x(t-b)$ $\longleftrightarrow$ $X(w)e^{-jwb}$

5. Frequency Shifting Property

Frequency shifting property states that the multiplication of function x(t) by e^{jw_0t} is equivalent to shifting its fourier transform X(w) in the positive direction by an amount w_0. This means that the spectrum X(w) is translated by an amount c. hence this property is often called frequency translated theorem. Mathematically .

If x(t) $\longleftrightarrow$ X(w)

Then $e^{w_0 t}$ x(t) $\longleftrightarrow$ X(w-w_0)

Proof: General expression for fourier transform is

$$X(w)=F[x(t)]=\int_{-\infty}^{\infty} x(t)\, e^{-wt} dt$$

Now, $F[e^{w_0 t}x(t)] =\int_{-\infty}^{\infty} x(t)\, e^{w_0 t} e^{-wt} dt$

Or $F[e^{w_0 t}x(t)] =\int_{-\infty}^{\infty} x(t)\, e^{-(w-w_0)t} dt$

Or $F[e^{w_0 t}x(t)] =X(w - w_0)$

Or $e^{w_0 t}$ x(t) $\longleftrightarrow$ X(w-w_0)

6. Time Differentiation Property

The time differentiation property states that the differentiation of a function x(t) in the time domain is equivalent to multiplication of its fourier transform by a factor jw. Mathematically

If x(t) $\longleftrightarrow$ X(w)

Then $\frac{dx(t)}{dt}$ x(t) $\longleftrightarrow$ jw X(w)

Proof:The general expression for fourier transform is

$$F^{-1}[(w)] = x(t) = \frac{1}{2\pi}\int_{-\infty}^{\infty}(w)e^{wt}dw$$

Taking differentiation, we have

$$\frac{dx(t)}{dt} = \frac{1}{2\pi}\frac{d}{dt}\left[\int_{-\infty}^{\infty}(w)e^{wt}dw\right]$$

Interchanging the order of differentiation and integration, we have

$$\frac{dx(t)}{dt} = \frac{1}{2\pi}\int_{-\infty}^{\infty}\frac{d}{dt}[(w)e^{wt}]dw$$

$$= \frac{1}{2\pi}\int_{-\infty}^{\infty} jw(w)e^{wt}dw$$

$$\frac{dx(t)}{dt} = F^{-1}[jw(w)]$$

$$F\left[\frac{dx(t)}{dt}\right] = jw(w)$$

$$\frac{dx(t)}{dt} \longleftrightarrow jw(w)$$

Condition of existence of thefourier transformation –Some condition should be satisfied by a signal x(t), then only it is possible to obtain the fourier transformation of x(t). For the periodic signals, the integration is obtained over one period, however, for a periodic it will be obtain over a range - ∞ to ∞. The signal x(t) will have to satisfy the following conditions so that its fourier transformation can be obtained :

1. The function x(t) should be single valued in any finite time interval T.
2. It should have a finite number of discontinuities in any finite interval T.
3. The function x(t) should have a finite number of maxima and minima in any finite interval of time T.
4. The function x(t) should have should be absolutely integral function. This means that $\int_{-\infty}^{\infty}|x(t)|dt < \infty$.

The condition stated above is sufficient conditions, but they are not the necessary conditions.

Fourier transform of some signals-

1. **Transform of Gate -**A gate function is rectangular pulse. Figure 1.2 shows gate function. The function or rectangular pulse shown in figure 1.3 is written as rect $(\frac{t}{\tau})$.

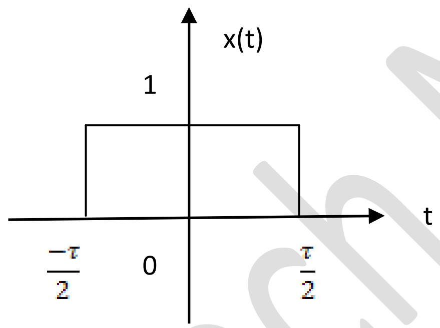


Figure No. 1.2 Gate Function

From the above figure it is clear that rect$(\frac{t}{c})$ represents a gate pulse of height or amplitude unity and width r.

$$x(t)= \quad \text{rect}\,(\frac{t}{c}) = \{1, \text{for} \frac{-c}{2} < t < \frac{c}{2}\}$$

$$\{0, otherwise\}$$

2. **Unit Impulse functions -** A unit impulse function was invented by P.A.M. Diarc and so it is also called as Delta function. It is denoted by(t).

Mathematically,

$$\delta(t)=0, t\neq 0$$

And, $\int_{-\infty}^{\infty} \delta(t)dt=1$

Figure 1.3 shows the graphical representation of a unit impulse function. The following points may be observed about a unit-impulse function:

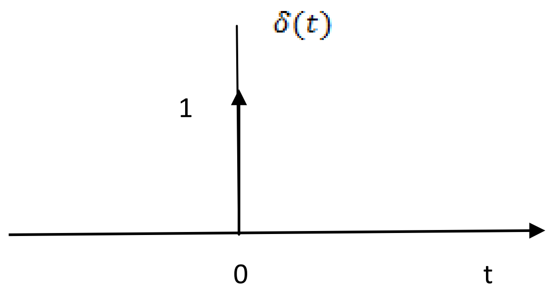


Fig.1.3 Unit Impulse function

3. Fourier Transform of Cosine wace -

e=cos +jsin

And e⁻ =cos –jsin

Hence 2cos =e+e⁻ Or cos = $\frac{e+e^-}{2}$

And2jsin=e-e⁻ Or

sin= $\frac{e+e^-}{2}$

Hence, $\cos w_0 t = \frac{e^{0} + e^{-0}}{2}$

We know that $e^{w_0 t} \longleftrightarrow 2\pi(w - w_0)$

And $e^{-w_0 t} \longleftrightarrow 2\pi(w + w_0)$

So that $\cos w_0 t \longleftrightarrow \frac{1}{2}[2\pi(w - w_0) + 2\pi(w + w_0)]$

Or $\cos w_0 t \longleftrightarrow [\pi(w - w_0) + \pi(w + w_0)]$

4. Fourier Transform of Constant–

Using relation , $(t) = 1$

Using duality property of fourier transform,

$$\longleftrightarrow x(t)(\mathbf{f})$$

Now, x(t) = (t) and X(f) = 1

$$(t) \qquad 1$$

$(\mathbf{f})$ is an even function, $(\mathbf{f}) = (-\mathbf{f})$

Therefore1 $\longleftrightarrow (\mathbf{f})$

Also, using relation $(t) \longleftrightarrow 1$

Using duality property of Fourier transform,

$$x(t) \longleftrightarrow (\omega)$$

$$(t) \longleftrightarrow 2\pi x(-\omega)$$

Now $x(t) = (t)$ and $(\mathbf{f}) = 1$

$$(t) \longleftrightarrow 1$$

$$1 \longleftrightarrow 2\pi(-\omega)$$

(ω) is an even function, $(\omega) = (-\omega)$

$1 \quad 2\pi.(\omega) \longleftrightarrow$

Convolution of signals may be done either in time domain or frequency domain. So there are following two theorems of convolution associated with Fourier transforms:

1. Time convolution theorem.
2. Frequency convolution theorem.

Time convolution theorem The time convolution theorem states that convolution in time domain is equivalent to multiplication of their spectra in frequency domain. Mathematically,

If, $x_1(t) \longleftrightarrow X_1(\omega)$ and $x_2(t) \longleftrightarrow X_2(\omega)$

Then, $x_1(t) * x_2(t) \longleftrightarrow X_1(\omega).X_2(\omega)$

Proof: $F[x1(t) * x2(t)] = \int_{-\infty}^{\infty} [\ _1(t) \ * \ _2(t)]e^{-\omega t}dt$

we have $x_1(t) * x_2(t) = \int_{-\infty}^{\infty} [\ _1(r) \ _2(t-r)]dr$

$F[x1(t) * x2(t)] = \int_{-\infty}^{\infty} \{\int_{-\infty}^{\infty} [\ _1(r) \ _2(t-r)dr]e^{-\omega t}dt\}$

Interchanging the order of integration, we have

$$F[x1(t) * x2(t)] = \int_{-\infty}^{\infty} {}_1(r)\{\int_{-\infty}^{\infty} [\ _2(t-r)dr]e^{-\omega t}dt\}$$

Letting t-r = p, in the second integration, we have, t=p+r and dt = dp

$$F[x1(t) * x2(t)] = \int_{-\infty}^{\infty} {}_1(r)\{\int_{-\infty}^{\infty} [\ _2(p)e^{-\omega(p+c)}dp]\,dr\}$$

$$= \int_{-\infty}^{\infty} {}_1(r)\int_{-\infty}^{\infty} [\ _2(p)e^{-\omega p}dp]\ e^{-\omega c)}dr$$

$$= \int_{-\infty}^{\infty} {}_1(r)\,{}_2(\omega)e^{-\omega t}dr$$

$$= {}_1(\omega)\,{}_2(\omega)$$

$x_1(t) * x_2(t) \longleftrightarrow {}_1(\omega)\,{}_2(\omega)$ This is time convolution theorem.

Frequency convolution theorem - The frequency convolution theorem states that the multiplication of two functions in time domain is equivalent to convolution of their spectra in frequency domain. Mathematically,

if $x_1(t) \longleftrightarrow X_1(\omega)$ and $x_2(t) \longleftrightarrow X_2(\omega)$

then, $x_1(t)x_2(t)\ \frac{1}{2\pi}[\ _1(\omega) * \ _2(\omega)]$

Proof: -$F[x_1(t)x_2(t)] = \int_{-\infty}^{\infty} [x_1(t)x_2(t)]\mathrm{e}^{-\omega t}dt$

$$\int_{-\infty}^{\infty} [\frac{1}{2\pi}\int_{-\infty}^{\infty} x_1(\lambda)\mathrm{e}^{t}d\lambda]\, x_2(t)\mathrm{e}^{-\omega t}dt$$

Interchanging the order of integration, we get

$$F[x_1(t)x_2(t)] = \frac{1}{2\pi}\int_{-\infty}^{\infty} x_1(\lambda)\, [\int_{-\infty}^{\infty} x_2(t)\mathrm{e}^{-\omega t}\mathrm{e}^{t}dt]\, d\lambda$$

$$F[x_1(t)x_2(t)] = \frac{1}{2\pi}\int_{-\infty}^{\infty} x_1(\lambda)\, [\int_{-\infty}^{\infty} x_2(t)\mathrm{e}^{-(\omega-)t}dt]\, d\lambda$$

$$F[x_1(t)x_2(t)] = \frac{1}{2\pi}\int_{-\infty}^{\infty} x_1(\lambda)x_2(\omega-\lambda)d\lambda$$

$$F[x_1(t)x_2(t)] = \frac{1}{2\pi}[\ _1(\omega) * \ _2(\omega)]$$

$$x_1(t)x_2(t) = \frac{1}{2\pi}[\ _1(\omega) * \ _2(\omega)]$$

$$2\pi\, x_1(t)\, x_2(t) = [\ _1(\omega) * \ _2(\omega)]$$

This is frequency convolution theorem in radian frequency in terms of frequency, we get

$$F[x_1(t)\, x_2(t)] = \ _1(\mathbf{f}) * \ _2(\mathbf{f})$$

Unit -2

Introduction: Overview of Communication system, Communication channels Need for modulation, Baseband and Pass band signals, Amplitude Modulation: Double side band with Carrier (DSB-C), Double side band suppressed carrier(DSB-SC), Single Side Band suppressed carrier(SSB-SC),Generation of AM, DSB-SC, SSB-SC, VSB-SC & its detection,Vestigial Side Band (VSB).

Overview of Communication system - Communication is the process of establishing connection or link between two points for information exchange. OR

Communication is simply the basic process of exchanging information. The electronics equipments which are used for communication purpose are called communication equipments. Different communication equipments when assembled together form a communication system. Typical example of communication system is line telephony and line telegraphy, radio telephony and radio telegraphy, radio broadcasting, point-to-point communication and mobile communication.

Figure shows the block diagram of a general communication system, in which the different functional elements are represented by blocks.

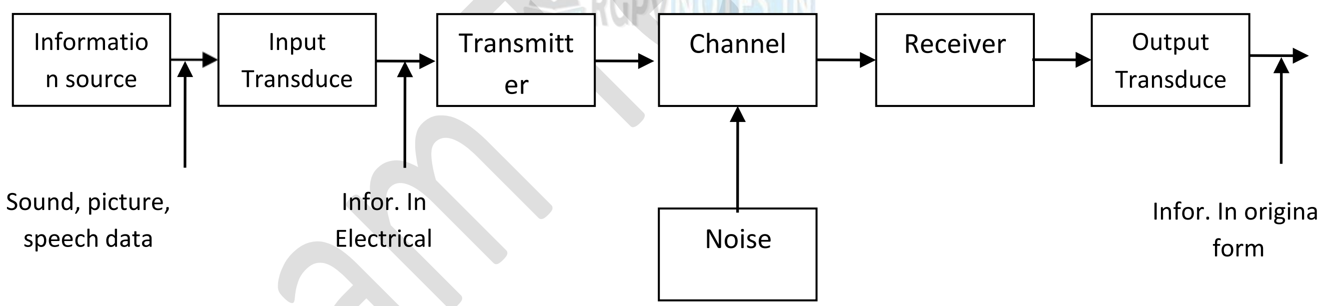


Figure No. 2.1: General communication system

Information Source

As we know, a communication system serves to communicate a message or information. This information originates in the information source. In general, there can be various messages in the form of words, group of words, code, symbols, sound signal etc. However, out of these messages, only the desired message is selected and communicated. Therefore, we can say that the function of information source is to produce required message which has to be transmitted.

Input Transducer

A transducer is a device which converts one form of energy into another form. The message from the information source may or may not be electrical in nature. In a case when the message produced by the information source is not electrical in nature, an input transducer is used to convert it into a time-varying electrical signal. For example, in case of radio-broadcasting, a microphone converts the information or massage which is in the form of sound waves into corresponding electrical signal.

Transmitter

The function of the transmitter is to process the electrical signal from different aspects. For example in radio broadcasting the electrical signal obtained from sound signal, is processed to restrict its range of audio frequencies (up to 5 kHz in amplitude modulation radio broadcast) and is often amplified. In wire telephony, no real processing is needed. However, in long-distance radio communication, signal amplification is necessary before modulation.

The Channel and the Noise

The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver. There are two types of channels, namely point-to-point channels and broadcast channels.

Example of point-to-point channels is wire lines, microwave links and optical fibers. Wire-lines operate by guided electromagnetic waves and they are used for local telephone transmission.

In case of microwave links, the transmitted signal is radiated as an electromagnetic wave in free space. Microwave links are used in long distance telephone transmission. An optical fiber is a low-loss, well-controlled, guided optical medium. Optical fibers are used in optical communications. Although these three channels operate differently, they all provide a physical medium for the transmission of signals from one point to another point. Therefore, for these channels, the term point-to-point is used.

On the other hand, the broadcast channel provides a capability where several receiving stations can be reached simultaneously from a single transmitter. An example of a broadcast channel is a satellite in geostationary orbit, which covers about one third of the earth's surface. During the process of transmission and reception the signal gets distorted due to noise introduced in the system.

Noise is an unwanted signal which tends to interfere with the required signal. Noise signal is always random in character. Noise may interfere with signal at any point in a communication system. However, the noise has its greatest effect on the signal in the channel.

Receiver

The main function of the receiver is to reproduce the message signal in electrical form from the distorted received signal. This reproduction of the original signal is accomplished by a process known as the demodulation or detection. Demodulation is the reverse process of modulation carried out in transmitter.

Destination

Destination is the final stage which is used to convert an electrical message signal into its original form. For example in radio broadcasting, the destination is a loudspeaker which works as a transducer i.e. converts the electrical signal in the form of original sound signal.

Need for Modulation - Modulation is extremely necessary in communication system because of the following reasons:

1. **Avoids mixing of signals -** One of the basic challenges facing by the communication engineering is transmitting individual messages simultaneously over a single communication channel. A method by which many signals or multiple signals can be combined into one signal and transmitted over a single communication channel is called multiplexing. We know that the sound frequency range is 20 Hz to 20 KHz. If the multiple baseband sound signals of same frequency range (I.e. 20 Hz to 20 KHz) are combined into one signal and transmitted over a single communication channel without doing modulation, then all the signals get mixed together and the receiver cannot separate them from each other. We can easily overcome this problem by using the modulation technique. By using modulation, the baseband sound signals of same frequency range (i.e. 20 Hz to 20 KHz) are shifted to different frequency ranges. Therefore, now each signal has its own frequency range within the total bandwidth. After modulation, the multiple signals having different frequency ranges can be easily transmitted over a single communication channel without any mixing and at the receiver side, they can be easily separated.
2. **Increase the range of communication -** The energy of a wave depends upon its frequency. The greater the frequency of the wave, the greater the energy possessed by it. The baseband audio signals frequency is very low so they cannot be transmitted over large distances. On the other hand, the carrier signal has a high frequency or high energy. Therefore, the carrier signal can travel large distances if radiated directly into space. The only practical solution to transmit the baseband signal to a large distance is by mixing the low energy baseband signal with the high energy carrier signal. When the low frequency or low energy baseband signal is mixed with the high frequency or high energy carrier signal, the resultant signal frequency will be shifted from low frequency to high frequency. Hence, it becomes possible to transmit information over large distances. Therefore, the range of communication is increased.

3. **Wireless communication -**In radio communication, the signal is radiated directly into space. The baseband signals have very low frequency range (I.e. 20 Hz to 20 KHz). So it is not possible to radiate baseband signals directly into space because of its poor signal strength. However, by using the modulation technique, the frequency of the baseband signal is shifted from low frequency to high frequency. Therefore, after modulation, the signal can be directly radiated into space.
4. **Reduces the effect of noise -**Noise is an unwanted signal that enters the communication system via the communication channel and interferes with the transmitted signal. A message signal cannot travel for a long distance because of its low signal strength. Addition of external noise will further reduce the signal strength of a message signal. So in order to send the message signal to a long distance, we need to increase the signal strength of the message signal. This can be achieved by using a technique called modulation.In modulation technique, a low energy or low frequency message signal is mixed with the high energy or high frequency carrier signal to produce a new high energy signal which carries information to a long distance without getting affected by the external noise.
5. **Practicability of antennas -**When the transmission of a signal occurs over free space, the transmitting antenna radiates the signal out and receiving antenna receives it. In order to effectively transmit and receive the signal, the antenna height should be approximately equal to the wavelength of the signal to be transmitted. Now,

$$Wavelength\ (\lambda) = \frac{Velocty\ \ (V)}{Frequency\ (f)} = \frac{3 * 10^8}{Frequency(Hz)}\ meters$$

Types of Modulation - The types of modulations are broadly classified into continuous-wave modulation and pulse modulation.

1. Continuous-wave modulation.
2. Pulse modulation.

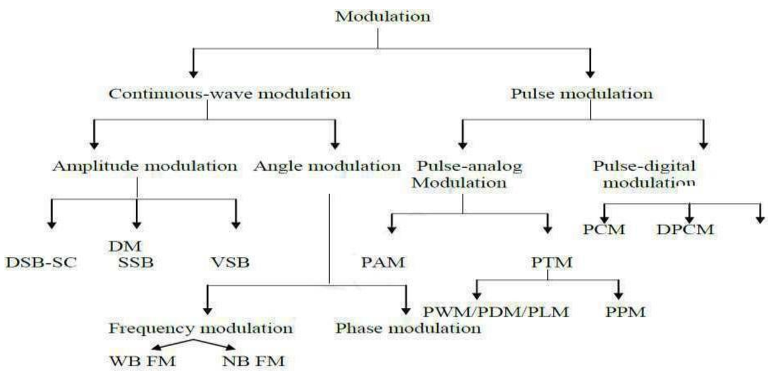


Figure No. 2.2: Types of Modulation

Continuous-wave Modulation or Analog Modulation -

In the continuous-wave modulation, a high frequency sine wave is used as a carrier wave. This is further divided into amplitude and angle modulation.

- If the amplitude of the high frequency carrier wave is varied in accordance with the instantaneous amplitude of the modulating signal, then such a technique is called as Amplitude Modulation.
- If the angle of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Angle Modulation.

The angle modulation is further divided into frequency and phase modulation.

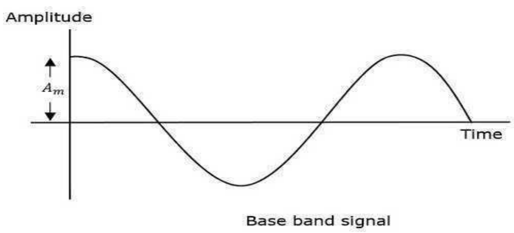
- If the frequency of the carrier wave is varied, in accordance with the instantaneous value of the modulating signal, then such a technique is called as Frequency Modulation.
- If the phase of the high frequency carrier wave is varied in accordance with the instantaneous value of the modulating signal, then such a technique is called as Phase Modulation.

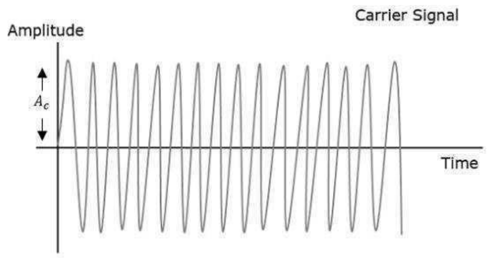
Pulse Modulation or Digital Modulation –

In Pulse modulation, a periodic sequence of rectangular pulses is used as a carrier wave. This is further divided into analog and digital modulation.

- In analog modulation technique, if the amplitude, duration or position of a pulse is varied in accordance with the instantaneous values of the baseband modulating signal, then such a technique is called as Pulse Amplitude Modulation (PAM) or Pulse Duration/Width Modulation (PDM/PWM), or Pulse Position Modulation (PPM).
- In digital modulation, the modulation technique used is Pulse Code Modulation (PCM) where the analog signal is converted into digital form of 1s and 0s. As the resultant is a coded pulse train, this is called as PCM.

Amplitude modulation - According to the standard definition, "The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal." Which means, the amplitude of the carrier signal which contains no information varies as per the amplitude of the signal, at each instant, which contains information? This can be well explained by the following figures 2.3.





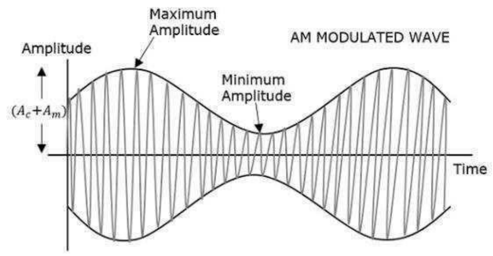


Figure No. 2.3: Types of Modulation

The modulating wave which is shown first is the message signal. The next one is the carrier wave, which is just a high frequency signal and contains no information, while the last one is the resultant modulated wave.It can be observed that the positive and negative peaks of the carrier wave are interconnected with an imaginary line. This line helps recreating the exact shape of the modulating signal. This imaginary line on the carrier wave is called as **Envelope**. It is the same as the message signal.

Mathematical Expression

Let modulating signal be $-m(t) = A_m Sn(\omega_m t)$

Let carrier signal be $-c(t) = A_C\, Sn(\omega_C t)$

Where $\mathbf{A_m}$ = maximum amplitude of the modulating signal.

$\mathbf{A_c}$ = maximum amplitude of the carrier signal.

The standard form of an Amplitude Modulated wave is defined as –

$$S(t) = A_c[1 + K_a m(t)]Sn(\omega_c t)$$

$$S(t) = A_c[1 + K_a A_m Sn(\omega_m t)]Sn(\omega_c t)$$

$$S(t) = A_c[1 + \mu Sn(\omega_m t)]Sn(\omega_c t)$$

Where, $\mu = K_a A_m$, called Modulation Index or Modulation depth.

A carrier wave, after being modulated, if the modulated level is calculated, then such an attempt is called as **Modulation Index** or **Modulation Depth**. It states the level of modulation that a carrier wave undergoes.If the maximum and minimum values of the envelope of the modulated wave are represented by A_{max} and A_{min} respectively as shown in figure 2.4.

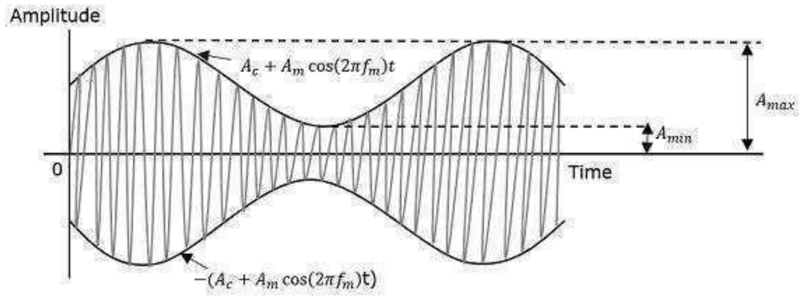


Figure No. 2.4: Types of Modulation

Then, the equation for the Modulation Index.

$$A_{max}=A_c(1+\mu)$$

Since, at A_{max} the value of sin θ is 1,

$$A_{min}=A_c(1-\mu)$$

Since, at A_{min} the value of sin θ is -1, then on further solving the above equation,

$$\frac{A_{max}}{A_{mn}} = \frac{(1+\mu)}{(1-\mu)}$$

$$A_{max} - \mu.\ A_{max} = A_{min} + \mu.\ A_{min}$$

$$\mu = \frac{A_{max} - A_{mn}}{A_{max} + A_{mn}}$$

Hence, the equation for Modulation Index is obtained. μ denotes the **modulation index** or **modulation depth**. This is often denoted in percentage called as **Percentage Modulation**. It is the extent of modulation denoted in percentage, and is denoted by m.

For a perfect modulation, the value of modulation index should be 1, which means the modulation depth should be 100%.

For instance, if this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as Under-modulation. Such a wave is called as an **under-modulated wave** as shown in figure 2.5.

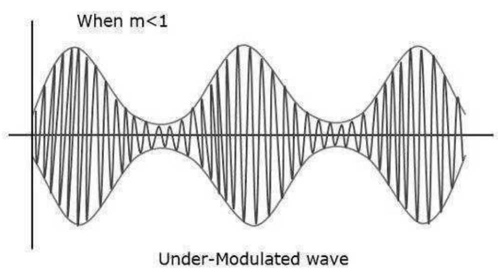


Figure No. 2.5: Under Modulation

If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an over-modulated wave. It would look like the following figure.

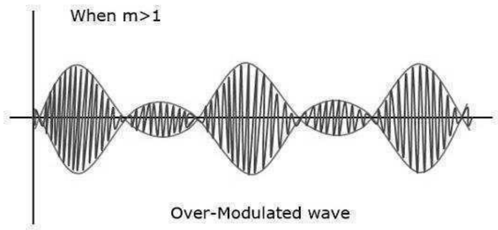


Figure No. 2.6: Over Modulation

As the value of modulation index increases, the carrier experiences a 180° phase reversal, which causes additional sidebands and hence, the wave gets distorted. Such over modulated wave causes interference, which cannot be eliminated.

Double side band with Carrier (DSB-C)

The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal i.e. the amplitude of the carrier signal containing no information varies as per the amplitude of the signal containing information, at each instant.

In the process of Amplitude Modulation, the modulated wave consists of the carrier wave and two sidebands. The modulated wave has the information only in the sidebands. **Sideband** is nothing but a band of frequencies, containing power, which are the lower and higher frequencies of the carrier frequency.

The transmission of a signal, which contains a carrier along with two sidebands, can be termed as **Double Sideband with Carrier** system or simply **DSB-C**. It is plotted as shown in the following figure.

Sideband - A Sideband is a band of frequencies, containing powers, which are the lower and higher frequencies of the carrier frequency. Both the sidebands contain the same information. The representation of amplitude modulated wave in the frequency domain is as shown in the following figure 2.7.

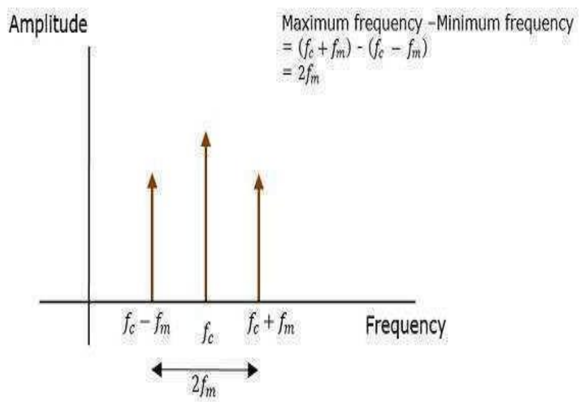


Figure No. 2.7: Representation of AM wave in frequency domain

Both the sidebands in the figure contain the same information. The transmission of such a signal which contains a carrier along with two sidebands can be termed as Double Sideband with Carrier system (DSB-C), or simply DSB-FC. It is plotted as shown in the following figure 2.8.

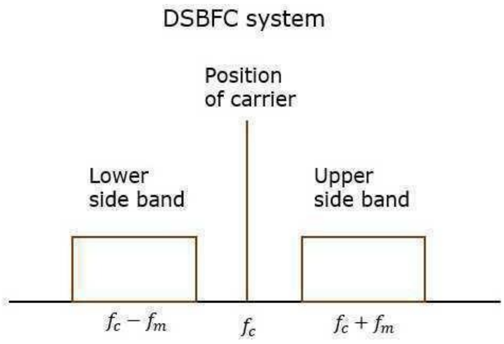


Figure No. 2.8: Spectrum of AM wave in frequency domain

If this carrier is suppressed and the saved power is distributed to the two sidebands, then such a process is called as **Double Sideband Suppressed Carrier** system or simply **DSBSC**. It is plotted as shown in the figure 2.9.

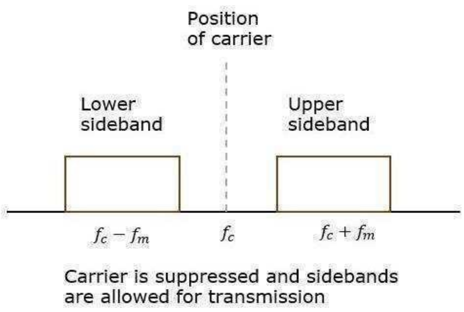


Figure No. 2.9: AMDSB-SCpectrum

Mathematical expression:

Let $m(t) = A_m Cos(\omega_m t)$ is the baseband message and c$(t) = A_C\ Cos\ (\omega_C t)$is called the carrier wave. The carrier frequency, f_C should be larger than the highest spectral component in $m(t)$.

Consider a sinusoidal carrier signal$C(t)$. is defined as

$$c(t) = A_C Cos\ (2\pi f_C t)$$

Where A_C= Amplitude of the carrier signal

Mathematically, we can represent the equation of DSB-C wave as the product of modulating and carrier signals.

$$s(t) = m(t).c(t)$$

$$s(t) = A_C\ Cos\ (\omega_c t). A_m \mathrm{Cos}(\omega_m t)$$

$$s(t) = A_C A_m \; Cos\;(\omega_c t).\mathrm{Cos}(\omega_m t)$$

Double side band suppressed carrier (DSB-SC) -

Let $m(t) = A_m sn(\omega_m t)$ is the baseband message and c$(t) = A_C\; sn(\omega_C t)$is called the carrier wave as shown in figure 2.10(a) and (b) respectively. The carrier frequency, f_C should be larger than the highest spectral component in $m(t)$.

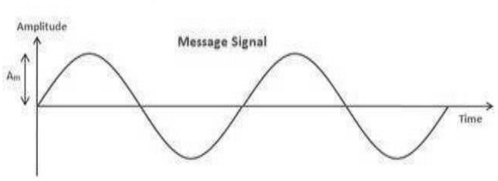


Figure No. 2.10(a): Message signal

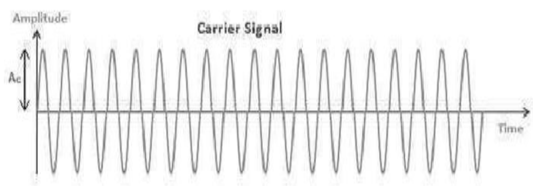


Figure No. 2.10(b): Carrier signal

Mathematically, we can represent the equation of DSB-C wave as the product of modulating and carrier signals.

$$s(t) = A_c[1 + K_a m(t)].sn(\omega_c t)$$

$$s(t) = A_c[1 + K_a A_m \sin(\omega_m t)].sn(\omega_c t)$$

$$s(t) = A_c[1 + \mu \sin(\omega_m t)].sn(\omega_c t)$$

μ is called modulation index or Modulation depth.

$$s(t) = A_c sn(\omega_c t) + \mu A_c sn(\omega_m t).sn(\omega_c t)$$

$$s(t) = A_c\, sn(2\pi f_c t) + \frac{\mu A_c}{2}[\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

$$s(t) = A_c\, sn(2\pi f_c t) + \frac{\mu A_c}{2}[\cos 2\pi(f_c - f_m)t + \cos 2\pi(f_c + f_m)t]$$

$$s(t) = A_c\, sn(2\pi f_c t) + \frac{\mu A_c}{2}\cos 2\pi(f_c - f_m)t + \frac{\mu A_c}{2}\cos 2\pi(f_c + f_m)t$$

$$\longleftrightarrow \qquad \longleftrightarrow \qquad \longleftrightarrow$$

Carrier wave **LSB** **USB**

Modulated wave is a combination of three waves moving together having frequencies f_c , $(f_c - f_m)$, $(f_c + f_m)$.

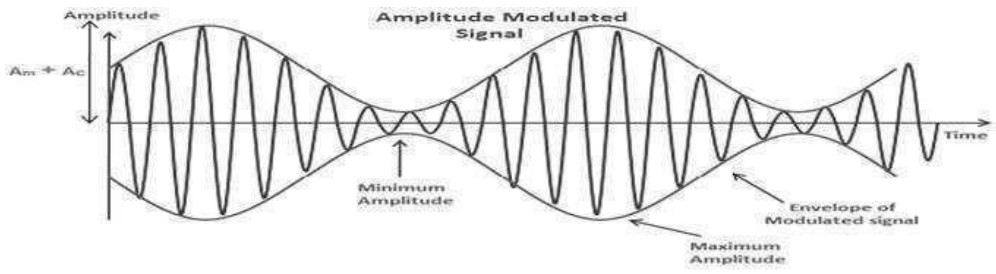


Figure No. 2.11: Amplitude Modulated signal

Spectrum of AM modulated signal –

On taking Fourier transform of message signal, carrier signal and amplitude modulated signal we can draw the spectrum of these signals as –

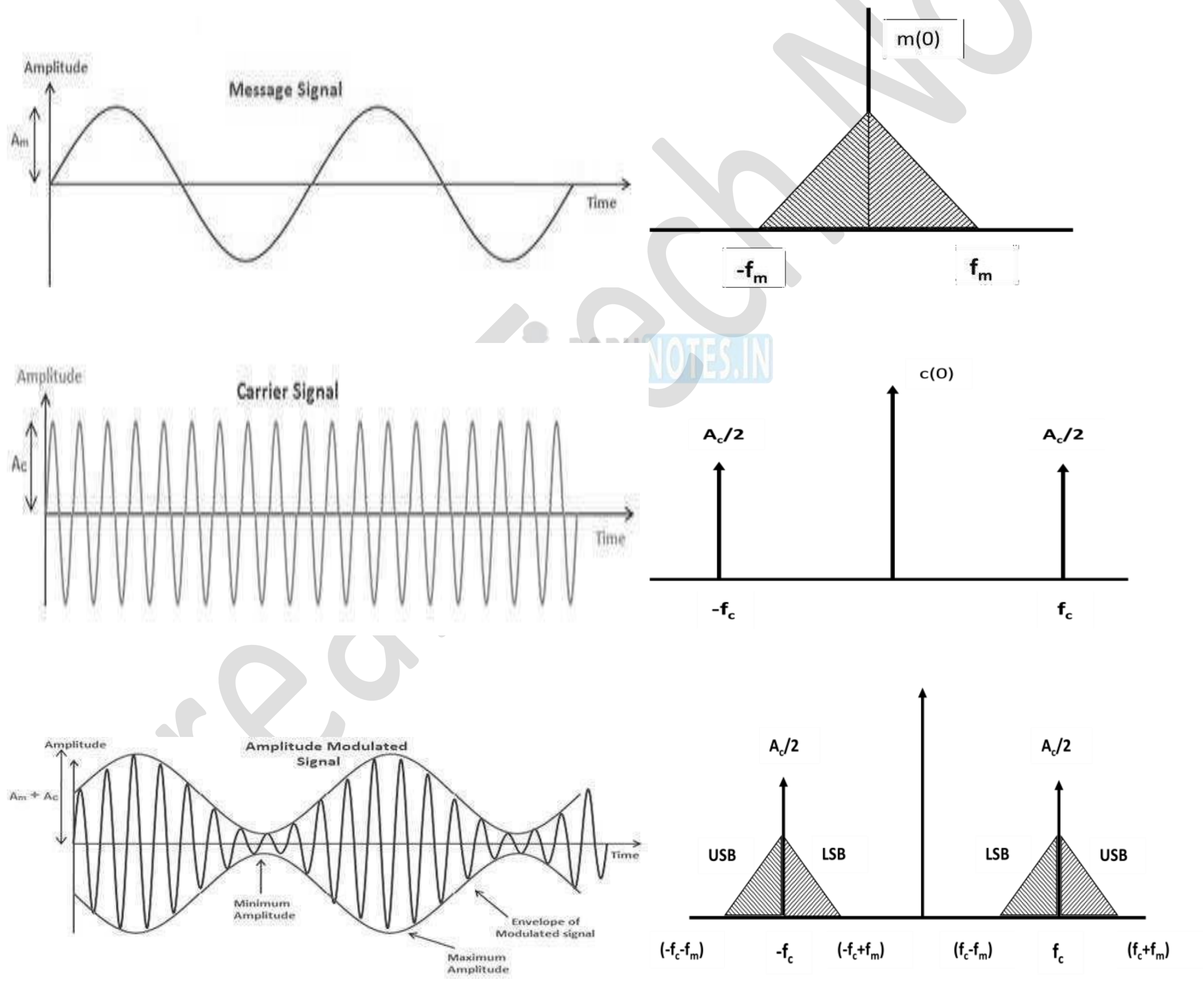


Figure No. 2.12: spectrum of Amplitude Modulated signal

The AM-SC signal exhibits phase-reversal at zero crossings, which is obvious from the waveform of figure 3.6. From the spectrum of figure 2.12 it is obvious that the impulses at $\pm \omega_c$ aremissing which means

(a) The carrier term $\omega_c$is suppressed in the spectrum. Hence, it is called suppressed carrier system.

(b) The base band is present twice in the modulated spectrum. The modulated signal consist of ($\pm \omega_c + \omega_m$) and ($\pm \omega_c - \omega_m$) frequency term. Positive and associated negative frequency terms are necessary for a real signal. The two terms mentioned above called sidebands. The term ($\pm \omega_c + \omega_m$) is called upper sideband and the term ($\pm \omega_c - \omega_m$) lower sideband. Thus this system produces two sidebands corresponding to each frequency component in modulating signal. This system is therefore called Double side band suppressed carrier (DSB-SC).

Bandwidth of DSB-SC AM wave:

We know bandwidth can be measured by subtracting lowest frequency of the signal from highest frequency of the signal in upper sideband.

For amplitude modulated wave it is given by

$$\text{Band Width} = f_{max(USB)} - f_{min(USB)}$$

$$\text{Band Width} = (f_c + f_m) - (f_c - f_m)$$

$$\text{Band Width} = f_m + f_m$$

$$\text{Band Width} = 2f_m$$

Therefore the bandwidth required for the amplitude modulation is twice the frequency of the modulating signal.

Single Side Band suppressed carrier (SSB-SC) –

Double sideband suppressed carrier system (DSB-SC) doubles the bandwidth of the modulated signal as compared to the baseband signal. This is because the sideband appears twice in the modulated signal as shown in figure 2.13 (b). The baseband ranges between 0 to $\omega_m$figure 2.13 (a) This bandwidth becomes $2\omega_m$ after modulation as shown in figure 2.13 (b)

The message signal appears twice in the DSB-SC signal, and it unnecessarily increases the bandwidth. Lower the bandwidth of the modulated signal, more is the number of channels that can be accommodate in a given frequency space. It is therefore desirable to transmit only one sideband, as this contains the entire information content in the message signal and at the same time it reduces the bandwidth by half. This means we can accommodate twice the number of channels in a given frequency space by using a single sideband in place of both the sidebands.

Modulation of this type provides a single side band with suppressed carrier is known as single sideband suppressed carrier system (SSB-SC). The spectrum of SSB-SC with LSB and USB is shown in figure 2.13 (c).

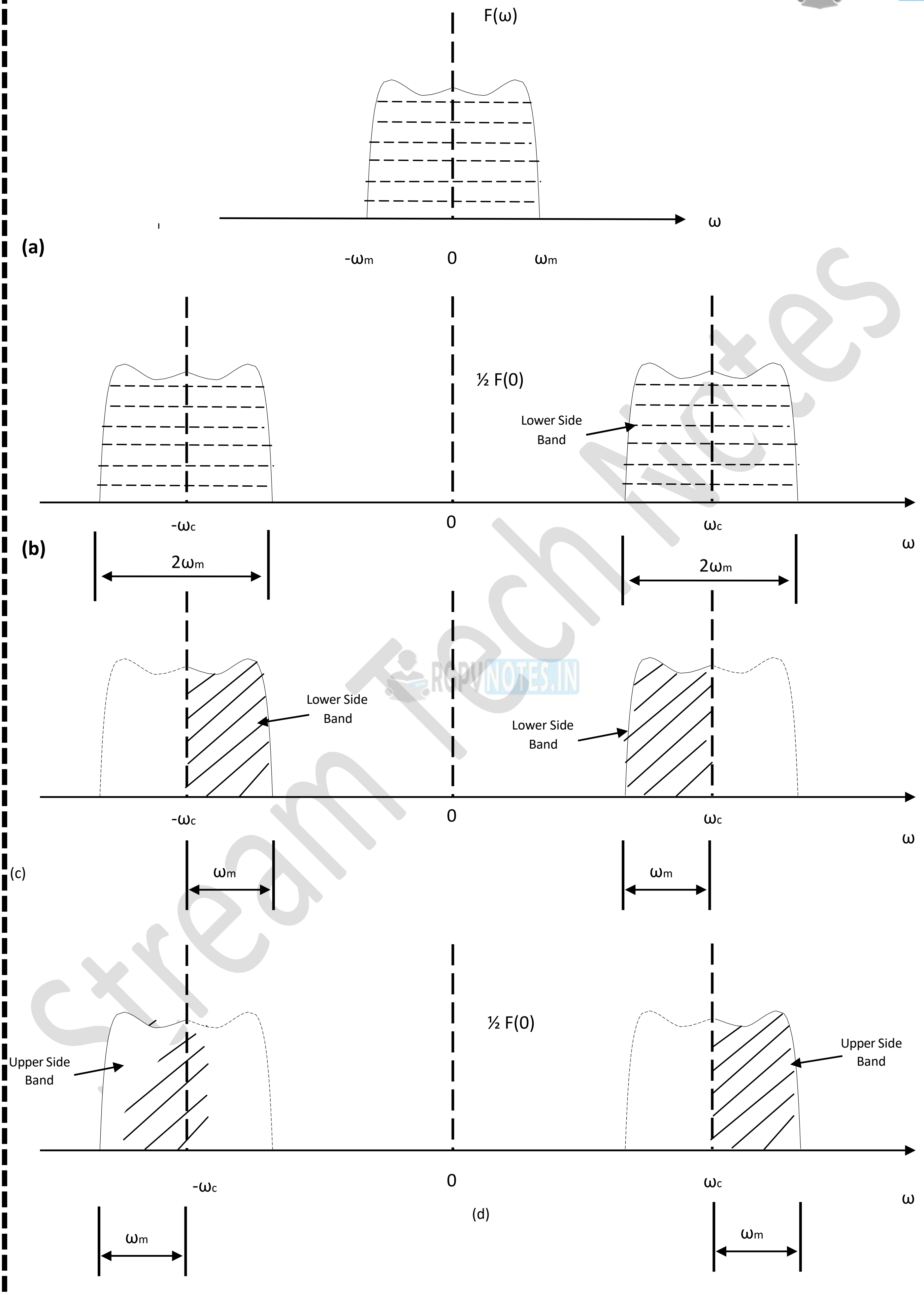


Figure No. 2.13: (a) Message Signal (b) DSB-SC Signal (c) SSB-SC LSB spectrum (d) SSB-SC USB spectrum

Both the spectra of SSB-SC signals are symmetrical about the vertical axis, so that they represent real signal. The bandwidth of SSB-SC signal is $\omega_m$same as the bandwidth of the baseband signal.

Generation of AM:

Generation of DSB-SC AM signal:

The generation of a DSB-SC modulated wave consists simply of the product of the message signal m(t) and the carrier wave A_c Cos ($2\pi f_c t$). Devices for achieving this requirement are called a product modulator. There are two methods to generate DSB-SC waves. They are:

- Balanced modulator.
- Ring modulator.

Balanced Modulator:

1. Balanced modulator consists of two identical AM modulators which are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as balanced modulator as shown in figure 2.14.
2. Assume that two AM modulators are identical, except for the sign reversal of the modulating signal applied to the input of one of the modulators.
3. The same carrier signal C (t) = Ac Cos (2πfct) is applied as one of the inputs to these two AM modulators.
4. The modulating signal m(t) is applied as another input to the upper AM modulator. Whereas, the modulating signal with opposite polarity, −m(t) is applied as another input to the lower AM modulator.

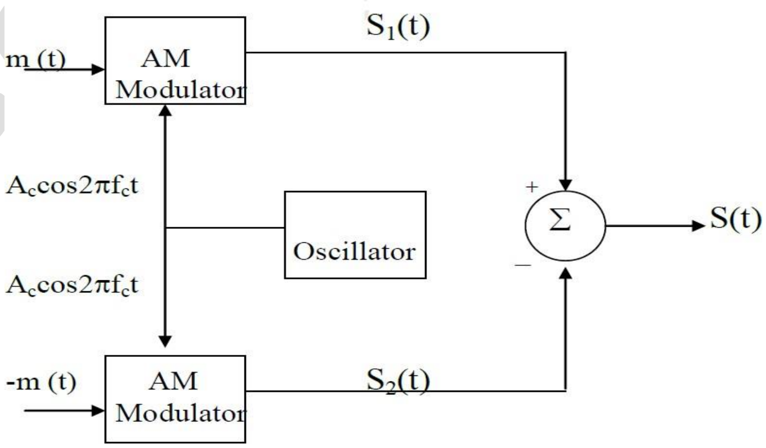


Figure No. 2.14: Balanced modulator

Mathematical analysis:

The outputs of the two AM modulators can be expressed as follows:

$$S_1(t) = A_c\,[1+k_a m(t)]\,\mathrm{Cos}\,2\pi f_c t$$

$$S_2(t) = A_c[1 - k_a m(t)] \cos 2\pi f_c t$$

Subtracting $S_2(t)$ from $S_1(t)$, we obtain

$$S(t) = S_1(t) - S_2(t)$$

$$S(t) = 2A_c k_a m(t) \cos(2\pi f_c t)$$

Hence, except for the scaling factor $2k_a$ the balanced modulator output is equal to product of the modulating signal and the carrier signal. The Fourier transform of S (t) is

$$S(f) = k_a A_c [M(f-f_c) + M(f+f_c)]$$

Assume that the message signal is band-limited to the interval $-W \le f \le W$ as shown in figure 2.15 and its DSB-SC modulated spectrum is shown in figure 2.16.

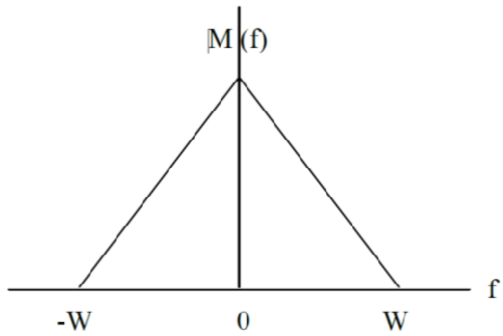


Figure No. 2.15: Spectrum of Baseband signal

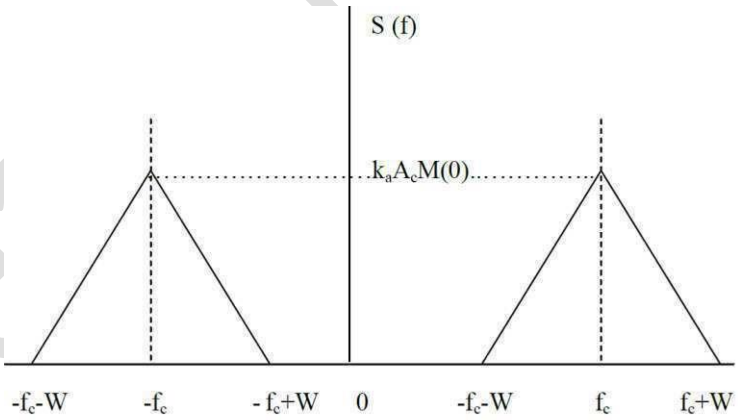


Figure No. 2.16: Spectrum of DSBSC wave

Ring modulator:

One of the most useful product modulator, for generating a DSBSC wave, is the ring modulator shown in figure 2.17.

1. In this diagram, the four diodes D1, D2, D3 and D4 are connected in the ring structure. Hence, this modulator is called as the ring modulator.

2. The diodes are controlled by a square-wave carrier C (t) of frequency fc, which applied longitudinally by means of to center-tapped transformers. If the transformers are perfectly balanced and the diodes are identical, there is no leakage of the modulation frequency into the modulator output.
3. The message signal m(t) is applied to the input transformer. Whereas, the carrier signals C (t) is applied between the two centre-tapped transformers.
4. For positive half cycle of the carrier signal, the diodes D1 and D3 are switched ON and the other two diodes D2 and D4 are switched OFF. In this case, the message signal is multiplied by +1.
5. For negative half cycle of the carrier signal, the diodes D2 and D4 are switched ON and the other two diodes D1 and D3 are switched OFF. In this case, the message signal is multiplied by -1. This results in 1800 phase shift in the resulting DSBSC wave.

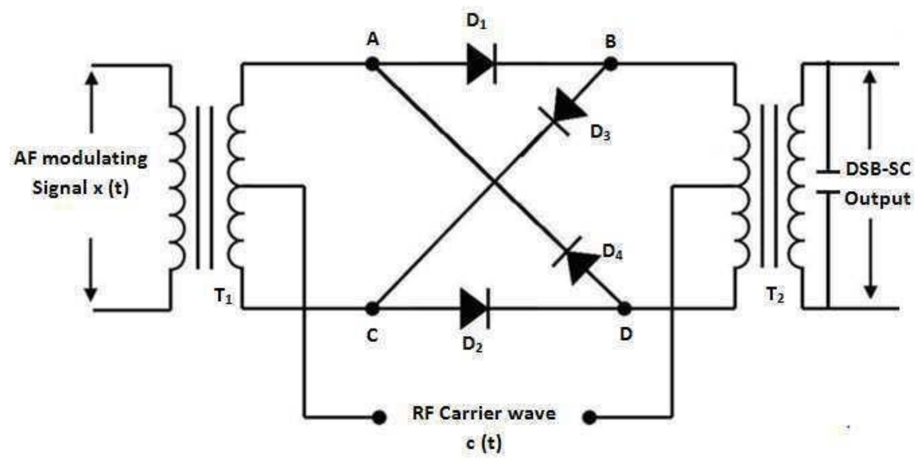


Figure No. 2.17: Ring modulator

Mathematical Analysis:

The square wave carrier c (t) can be represented by a Fourier series as follows:

$$C(t) = \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos 2\pi f_c t(2n-1)$$

$$= 4/\pi \,\mathrm{Cos}(2\pi fct) \ + \text{higher order harmonics}(n=1)$$

Now, the Ring modulator output is the product of both message signal m (t) and carrier signal c (t).

$$S\,(t) = c\,(t)\; m\,(t)$$

$$S\,(t) == \frac{4}{\pi}\sum_{n=1}^{\infty}\frac{(-1)^{n-1}}{2n-1}\cos 2\pi f_c t(2n-1)\; m\,(t) \qquad \text{For } n=1$$

$$S\,(t) = 4/\pi \,\mathrm{Cos}\,(2\pi f_c t)\; m\,(t)$$

There is no output from the modulator at the carrier frequency i.e the modulator output consists of modulation products. The ring modulator is also called as a double-balanced modulator, because it is balanced with respect to both the message signal and the square wave carrier signal.

The Fourier transform of S (t) is

$$S(f) = 2/\pi\, [M(f-f_c) + M(f+f_c)]$$

Assume that the message signal is band-limited to the interval –W ≤f≤ W as shown in figure 2.27 and its DSB-SC modulated spectrum in figure 2.28.

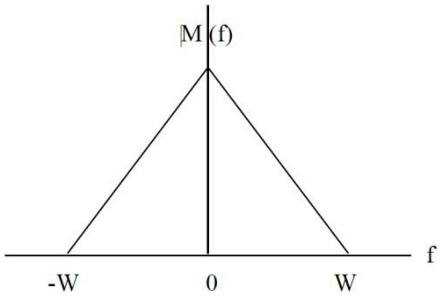


Figure No. 2.18: Spectrum of Baseband signal

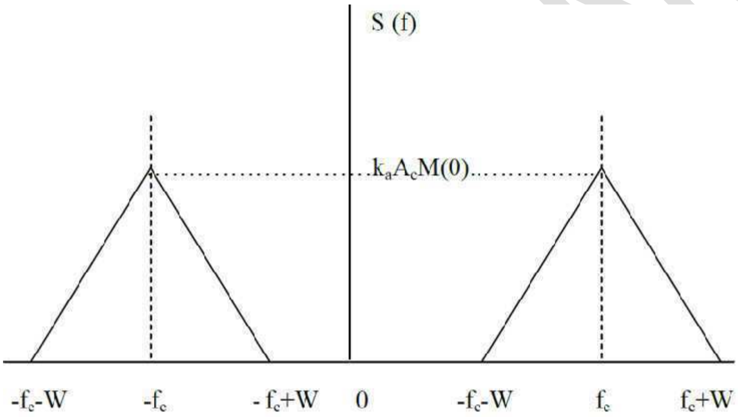


Figure No. 2.19: Spectrum of DSBSC wave

Coherent Detection of DSB-SC AM Waves:

The base band signal can be recovered from a DSB-SC signal by multiplying DSB-SC wave S (t) with a locally generated sinusoidal signal and then low pass filtering the product. It is assumed that local oscillator signal is coherent or synchronized, in both frequency and phase, with the carrier signal C (t) used in the product modulator to generate S (t). This method of demodulation is known as coherent detection or synchronous demodulation.

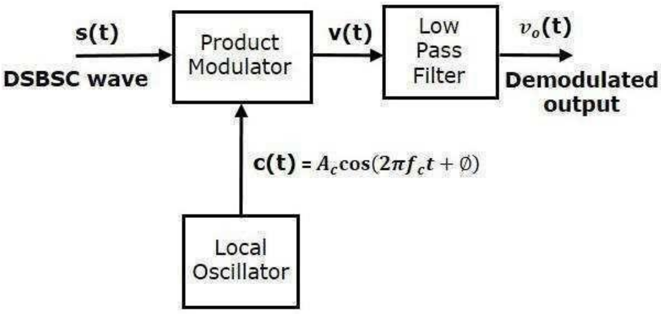


Figure No. 2.19: Coherent detection of DSB-SC signal

Mathematical Analysis of coherent detection:

The product modulator produces the product of both input signal s(t) and local oscillator signal and the output of the product modulator is v (t).

$$S(t) = A_c \cos(2\pi fct) \cdot m(t)$$

$$C(t) = A_c \cos(2\pi fct + \emptyset)$$

$$V(t) = C(t)\, S(t)$$

$$V(t) = A_c \cos(2\pi f_c t + \emptyset)\, S(t)$$

$$V(t) = A_c \cos(2\pi f_c t + \emptyset)\, A_c \cos(2\pi f_c t)\, m(t)$$

$$V(t) = A_c^2 \cos(2\pi f_c t + \emptyset) \cos(2\pi f_c t)\, m(t)$$

$$V(t) = \frac{{}^{2}{}_{c}}{2} \cos\emptyset\ m(t) + \frac{{}^{2}{}_{c}}{2} \cos(4\pi f_c t + \emptyset)\, m(t)$$

In the above equation, the first term is the scaled version of the message signal. It can be extracted by passing the above signal through a low pass filter. Therefore, the output of low pass filter is

$$V_o(t) = \frac{{}^{2}{}_{c}}{2} \cos\emptyset\ m(t)$$

The Fourier transform of V_o (t) is

$$V_O(f) = \frac{{}^{2}{}_{c}}{2} \cos\emptyset\ M(f)$$

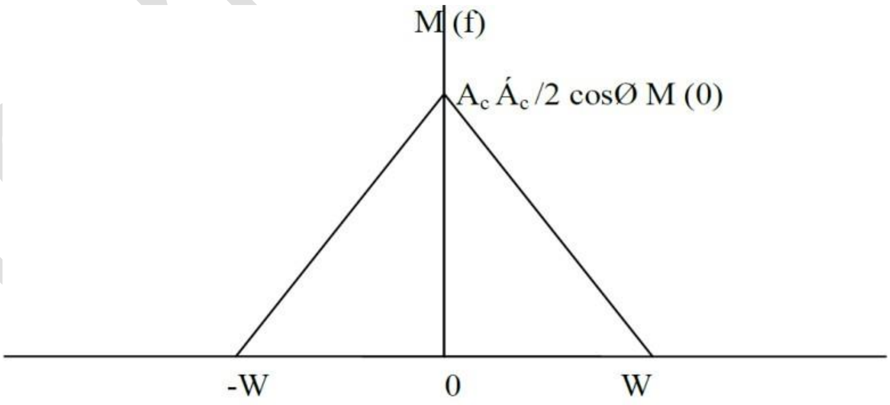


Figure No. 2.19: DSB-SC demodulated output

The demodulated signal is proportional to the message signal m (t) when the phase error is constant. The amplitude of this demodulated signal is maximum when Ø=0, the local oscillator signal and the carrier signal should be in phase, i.e., there should not be any phase difference between these two signals. The demodulated signal amplitude will be zero, when Ø=±π/2. This effect is called as quadrature null effect.

Generation of SSB waves:

Frequency Discrimination Method or Filter method:

The frequency discrimination or filter method of SSB generation consists of a product modulator, which produces DSBSC signal and a band-pass filter to extract the desired side band and reject the other and is shown in the figure 2.20. Application of this method requires that the message signal satisfies two conditions.

1. The message signal m(t) has low or no low-frequency content. M(ω) has a “hole” at zero-frequency Example: - speech, audio, music.
2. The highest frequency component W of the message signal m(t) is much less than the carrier frequency.

Then, under these conditions, the desired side band will appear in a non-overlapping interval in the spectrum in such a way that it may be selected by an appropriate filter.

In designing the band pass filter, the following requirements should be satisfied:

1. The pass band of the filter occupies the same frequency range as the spectrum of the desired SSB modulated wave.
2. The width of the guard band of the filter, separating the pass band from the stop band, where the unwanted sideband of the filter input lies, is twice the lowest frequency component of the message signal.

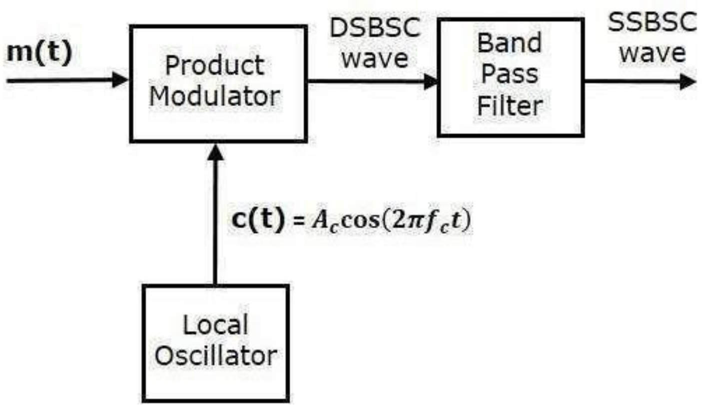


Figure No. 2.20: Frequency Discrimination Method block diagram

Phase discrimination method

1. The phase discriminator consists of two product modulators I and Q, supplied with carrier waves in-phase quadrature to each other as shown in figure 2.21.

2. The incoming base band signal m(t) is applied to product modulator I, producing a DSBSC modulated wave that contains reference phase sidebands symmetrically spaced about carrier frequency fc.

3. The Hilbert transform m^(t) of m(t) is applied to product modulator Q, producing a DSBSC modulated that contains side bands having identical amplitude spectra to those of modulator I, but with phase spectra such that vector addition or subtraction of the two modulator outputs results in cancellation of one set of side bands and reinforcement of the other set.

4. The use of a plus sign at the summing junction yields an SSB wave with only the lower side band, whereas the use of a minus sign yields an SSB wave with only the upper side band. This modulator circuit is called Hartley modulator.

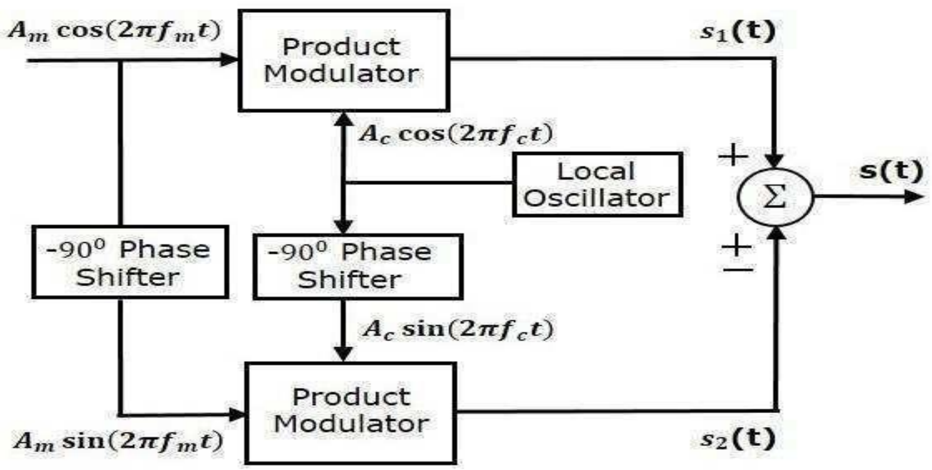


Figure No. 2.21: Phase discrimination method block diagram

Demodulation of SSB waves:

Coherent detection: It assumes perfect synchronization between the local carrier and that used in the transmitter both in frequency and phase. The carrier signal which is used for generating SSBSC wave is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. Following is the block diagram of coherent detector.

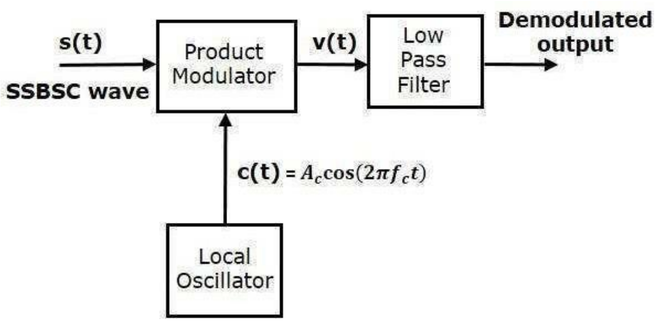


Figure No. 2.21: Coherent detection method block diagram

In this process, the message signal can be extracted from SSBSC wave by multiplying it with a coherent carrier and then the resulting signal is passed through a Low Pass Filter. The output of this filter is the desired message signal.

Mathematical Analysis:

$$S(t) = A_m Ac/2 \, Cos[2\pi(f_c - f_m)t]$$

The output of the local oscillator is

$$c(t) = A_c \, Cos(2\pi f_c t)$$

From the figure, we can write the output of product modulator as

$$v(t) = s(t)c(t)$$

Substitute s(t) and c(t) values in the above equation

$$V(t) = \frac{mc}{2}\cos[2\pi(f_c + f_m)t] \quad A_c \cos(2\pi f_c t)$$

$$V(t) = \frac{m\,c^2}{4}\cos(2\pi f_m t) + \frac{m\,c^2}{4}\cos[2\pi(2f_c - f_m)t]$$

In the above equation, the first term is the scaled version of the message signal the scaling factor is $\frac{c^2}{4}$. It can be extracted by passing the above signal through a low pass filter.Therefore, the output of low pass filter is

$$V_0(t) = \frac{m\,c^2}{4}\cos(2\pi f_m t)$$

Vestigial side band Modulation:

Vestigial sideband is a type of Amplitude modulation in which one side band is completely passed along with trace or tail or vestige of the other side band. VSB is a compromise between SSB and DSBSC modulation. In SSB, we send only one side band, the bandwidth required to send SSB wave is w. SSB is not appropriate way of modulation when the message signal contains significant components at extremely low frequencies. To overcome this VSB is used.The word “vestige” means “a part” from which, the name is derived.

VSBSC Modulation is the process, where a part of the signal called as vestige is modulated along with one sideband. The frequency spectrum of VSBSC wave is shown in the figure 2.37.Along with the upper sideband, a part of the lower sideband is also being transmitted in this technique. Similarly, we can transmit the lower sideband along with a part of the upper sideband.

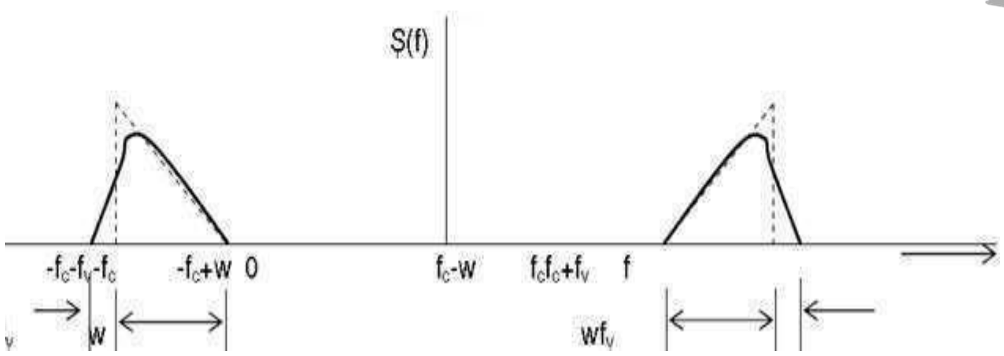


Figure 2.22 Spectrum of VSB containing vestige of USB

The vestige of the Upper sideband compensates for the amount removed from the Lower sideband. The bandwidth required to send VSB wave is

$$B = w + f_v$$

Where f_v is the width of the vestigial side band.

Therefore, VSB has the virtue of conserving bandwidth almost as efficiently as SSB modulation, while retaining the excellent low-frequency base band characteristics of DSBSC and it is standard for the transmission of TV signals.

Generation of VSB Modulated wave:

To generate a VSB modulated wave, we pass a DSBSC modulated wave through a sideband-shaping filter. The modulating signal m(t) is applied to a product modulator. The output of the local oscillator is also applied to the other input of the product modulator.

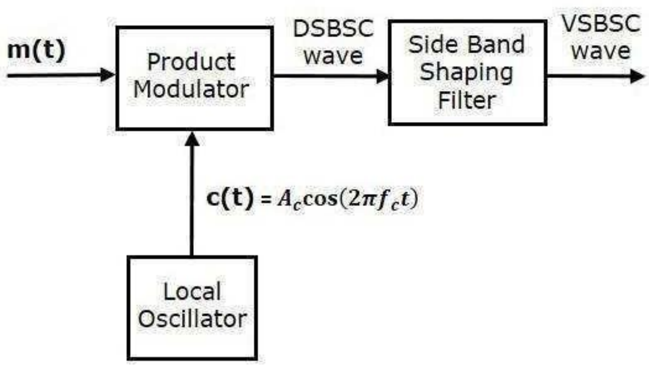


Figure 2.23 VSB modulator

Mathematical Analysis:

The output of the product modulator is then given by:

$$P(t) = A_c \cos(2\pi f_c t)\, m(t)$$

Apply Fourier transform on both sides

$$P(f) = \frac{A_c}{2}[M(f-f_c)+M(f+f_c)]$$

The above equation represents the equation of DSBSC frequency spectrum.

Let the transfer function of the sideband shaping filter be H(f). This filter has the input p(t) and the output is VSBSC modulated wave S(t).The Fourier transforms of p(t) and S(t) are P(f) and S(f) respectively.

$$S(f)=P(f)H(f)$$

Substitute P(f) in the above equation.

$$S(f)=\frac{A_c}{2}[M(f-f_c)+M(f+f_c)]H(f)$$

The above equation represents the equation of VSBSC frequency spectrum.

Demodulation of VSBSC

Demodulation of VSBSC wave is similar to the demodulation of SSBSC wave. Here, the same carrier signal which is used for generating VSBSC wave is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. The VSBSC demodulator is shown in the figure 2.24.

In this process, the message signal can be extracted from VSBSC wave by multiplying it with a carrier, which is having the same frequency and the phase of the carrier used in VSBSC modulation. The resulting signal is then passed through a Low Pass Filter. The output of this filter is the desired message signal.

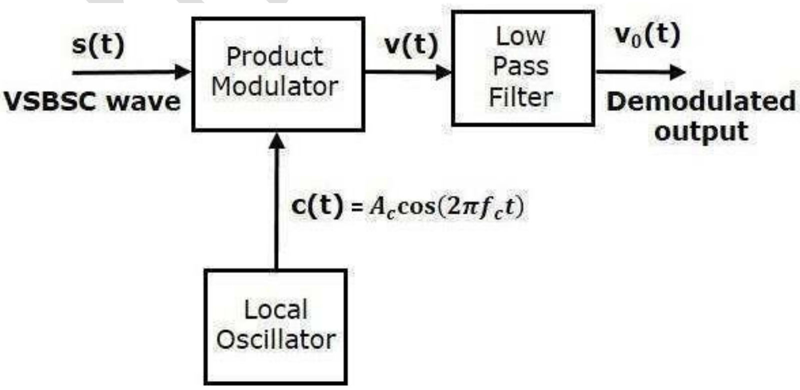


Figure 2.24 Demodulation of VSB-SC signal

Advantages of VSB

1. The main advantage of VSB modulation is the reduction in bandwidth. It is almost as efficient as the SSB.
2. Due to allowance of transmitting a part of lower sideband, the constraint on the filter has been relaxed. So practically, easy to design filters can be used.

3. It possesses good phase characteristics and makes the transmission of low frequency components possible.

Application of VSB

VSB modulation has become standard for the transmission of television signal. Because the video signal need a large transmission bandwidth if transmitted using DSB-FC or DSB-SC techniques.

Comparison of amplitude modulation techniques:

1. In commercial AM radio broadcast systems standard AM is used in preference to DSBSC or SSB modulation.
2. Suppressed carrier modulation systems require the minimum transmitter power and minimum transmission bandwidth. Suppressed carrier systems are well suited for point –to-point communications.
3. SSB is the preferred method of modulation for long-distance transmission of voice signals over metallic circuits, because it permits longer spacing between the repeaters.
4. VSB modulation requires a transmission bandwidth that is intermediate between that required for SSB or DSBSC.
5. DSBSC, SSB, and VSB are examples of linear modulation. In Commercial TV broadcasting; the VSB occupies a width of about 1.25MHz, or about one-quarter of a full sideband.
6. In standard AM systems the sidebands are transmitted in full, accompanied by the carrier. Accordingly, demodulation is accomplished by using an envelope detector or square law detector. On the other hand in a suppressed carrier system the receiver is more complex because additional circuitry must be provided for purpose of carrier recovery.
7. Suppressed carrier systems require less power to transmit as compared to AM systems thus making them less expensive.
8. SSB modulation requires minimum transmitter power and maximum transmission band with for conveying a signal from one point to other thus SSB modulation is preferred.
9. VSB modulation requires a transmission band width that is intermediate of SSB or DSBSC.
10. In SSB and VSB modulation schemes the quadrature component is only to interfere with the in phase component so that power can be eliminated in one of the sidebands.

Unit -3

Types of angle modulation, narrowband FM, wideband FM, its frequency spectrum, transmission BW, methods of generation (Direct & Indirect), detection of FM (discriminators: balanced, phase shift and PLL detector),pre emphasis and de-emphasis. FM transmitter & receiver: Block diagram of FM transmitter& receiver, AGC, AVC, AFC.

Introduction - Consider a sinusoid, $A_c \cos(2\pi f_c t+\phi_0)$, where Ac is the (constant) amplitude, f_c is the (constant) frequency in Hz and ϕ_0 is the initial phase angle. Let the sinusoid be written $A_c \cos[\theta(t)]$ where $\theta(t) = (2\pi f_c t+\phi_0)$. Relaxing the condition that Ac be a constant and making it a function of the message signal m (t) , gives rise to amplitude modulation. We shall now examine the case where Ac is a constant but θ(t), instead of being equal to $(2\pi f_c t+\phi_0)$, is a function of m(t) . This leads to what is known as the angle modulated signal. Two important cases of angle modulation are Frequency Modulation (FM) and Phase modulation (PM).

An important feature of FM and PM is that they can provide much better protection to the message against the channel noise as compared to the linear (amplitude) modulation schemes. Also, because of their constant amplitude nature, they can withstand nonlinear distortion and amplitude fading. The price paid to achieve these benefits is the increased bandwidth requirement; that is, the transmission bandwidth of the FM or PM signal with constant amplitude and which can provide noise immunity is much larger than 2W , where W is the highest frequency component present in the message spectrum.

Concept of Frequency Modulation:

Frequency modulation: It is the form of angle modulation in which instantaneous frequency $f_I(t)$ is varied linearly with the information signal m(t)

$$f_I(t)=f_c+ k_f\, m(t) \qquad \text{..(3.1)}$$

Where f_c–un-modulated carrier, k_f–Frequency sensitivity of the modulator, m(t)-Information signal.

Integrating above equation with respect to time limit 0 to t and multiplying with 2π

$$2\pi \int (t)dt = 2\pi_c \int dt + 2\pi K \int m(t)\, dt$$

$$(t) = 2\pi_c \int dt + 2\pi K \int m(t)\, dt$$

$$S(t) = A_c Cs((t))$$

$$S(t) = A_c Cs(2\pi_c \int dt + 2\pi K \int m(t)\, dt)$$

$$S(t) = A_c Cs\,(2\pi_c t + 2\pi K \int m(t)\,dt) \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots.3.2$$

Phase modulation:

It is that form of Angle modulation in which angle $\phi_i(t)$ is varied linearly with the base band signal m(t) as as shown by

$$\phi_i(t) = K_p m(t)$$

$$S(t)=A_c \cos\ (\omega_i\ (t)+ \phi_i(t)\ )$$

$$S(t) = A_c Cs(2\pi_c t + K_p m(t)) \qquad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots.3.3$$

Relationship between PM and FM

PM and FM are closely related in the sense that the net effect of both is variation in total phase angle. In PM, phase angle varies linearly with m(t) where in FM phase angle varies linearly with the integral of m(t). In other words, we can get FM by using PM, provided that at first, the modulating signal is integrated, and then applied to the phase modulator. The converse is also true, i.e. we can generate a PM wave using frequency modulator provided that m(t) is first differentiated and then applied to the frequency modulator.

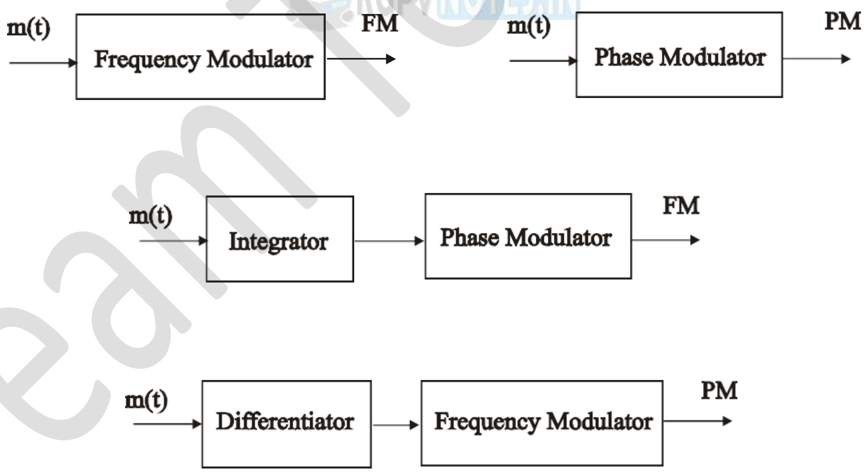


Figure 3.1 Relations between Fm and PM

Recall that a general sinusoid is of the form: $e_c = \sin(\omega_c t + \theta)$

Frequency modulation involves deviating a carrier frequency by some amount. If a sine wave was used to frequency modulate a carrier, the mathematical expression would be:

$$\omega_i = \omega_c + \Delta\omega \sin \omega_m t$$

Where
ω_i = instantane ous frequency
ω_c = carrier frequency
$\Delta\omega$ = carrier deviation
ω_m = modulation frequency

This expression shows a signal varying sinusoidal about some average frequency. However, we cannot simply substitute expression in the general equation for a sinusoid. This is because the sine operator acts upon angles, not frequency. Therefore, we must define the instantaneous frequency in terms of angles. It should be noted that the amplitude of the modulation signal governs the amount of carrier deviation, while the modulation frequency governs the rate of carrier deviation.

The term $\frac{d\theta}{dt}$ is an angular velocity and it is related to frequency and angle by the following relationship:

$$\omega = 2\pi f = \frac{d\theta}{dt}.$$

To find the angle, we must integrate ω with respect to time, we obtain:

$$\int \omega dt = \theta$$

We can now find the instantaneous angle associated with an instantaneous frequency:

$$\theta = \int \omega_i \, dt = \int (\omega_c + \Delta\omega \sin \omega_m t) dt$$
$$= \omega_c t - \frac{\Delta\omega}{\omega_m} \cos \omega_m t = \omega_c t - \frac{\Delta f}{f_m} \cos \omega_m t$$

This angle can now be substituted into the general carrier signal to define FM:

$$e_{fm} = \sin\left(\omega_c t - \frac{\Delta f}{f_m} \cos\omega_m t \right) \qquad \text{............................ (3.4)}$$

All FM transmissions are governed by a modulation index, β, which controls the dynamic range of the information being carried in the transmission.

$$\beta = \frac{\Delta f_c}{f_i}$$

Tone modulation:

Tone modulation is special case when message is sinusoidal as m(t) = Am cosωmt

For Phase Modulation equation become

$$s_{PM}(t) = A \cos[\omega_c t + \theta_0 + k_{PM} m(t)]$$

$$= A \cos[\omega_c t + \theta_0 + k_{PM} A_m \cos\omega_m t]$$

$$= A \cos[\omega_c t + \theta_0 + m_p \cos\omega_m t]$$

where mp = kPM Am is the phase modulation index, representing the maximum phase deviation $\Delta\theta$.

Frequency Modulation

$$s_{FM}(t) = A\cos[\omega_c t + \theta_0 + k_{FM}\int m(t)dt] = A\cos[\omega_c t + \theta_0 + k_{FM}\int A_m \cos\omega_m t dt]$$
$$= A\cos[\omega_c t + \theta_0 + \frac{k_{FM} A_m}{\omega_m}\sin\omega_m t] = A\cos[\omega_c t + \theta_0 + m_f \sin\omega_m t]$$

Where β = mf = kFMAm / ωm = Δω / ωm, i.e. the ratio of frequency deviation to the modulating frequency, is called the **frequency modulation index**.

$$\beta = \Delta\omega / \omega_m \quad \text{......} (3.5)$$

The relationship between phase deviation and frequency deviation in FM is given by

$$\Delta\theta = \beta = \Delta\omega / \omega_m \quad \text{......} (3.6)$$

Types of frequency modulation

The bandwidth of an FM signal depends on the frequency deviation. When the deviation is high, the bandwidth will be large, and vice-versa. According to the equation Δω = kFM|m(t)|max, for a given m(t), the frequency deviation, and hence the bandwidth, will depend on frequency sensitivity kFM. Thus, depending on the value of kFM (or Δω) we can divide FM into two categories: narrowband FM and wideband FM.

Narrowband FM (NBFM (β<<1))

When k_{FM} is small, the bandwidth of FM is narrow this type of FM is called narrowband FM.

Since when $x << 1$, $\cos x \approx 1$, $\sin x \approx x$, we have

$$s_{NBFM}(t) = A\cos[\omega_c t + \theta_0 + k_{FM}\int m(t)dt]$$
$$= A\cos(\omega_c t + \theta_0)\cos[k_{FM}\int m(t)dt] - A\sin(\omega_c t + \theta_0)\sin[k_{FM}\int m(t)dt]$$
$$\approx A\cos[\omega_c t + \theta_0] - Ak_{FM}\int m(t)dt \sin[\omega_c t + \theta_0]$$

Narrowband modulation methods:

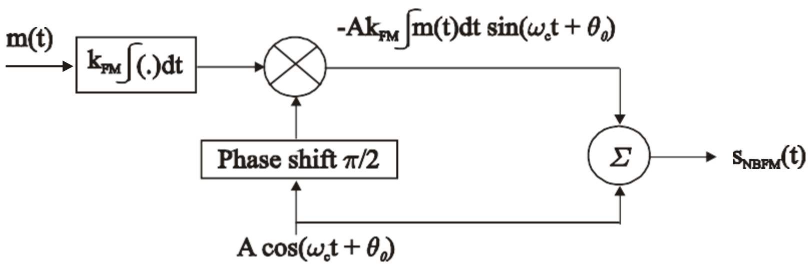


Figure 3.2.NBFM generation

Equation of narrowband frequency modulation (Tone modulation)

The message signal $m(t) = A_m \cos\omega_m t$

Signal waveform (assume $\theta_0 = 0$ for simplicity)

$$s_{NBFM}(t) = A\cos\omega_c t - Ak_{FM}\int m(t)dt \sin\omega_c t = A\cos\omega_c t - Ak_{FM}\int A_m \cos\omega_m t dt \sin\omega_c t$$
$$= A\cos\omega_c t - Ak_{FM}\frac{A_m}{\omega_m}\sin\omega_m t \sin\omega_c t = A\cos\omega_c t - Am_f \sin\omega_m t \sin\omega_c t$$
$$= A\cos\omega_c t + \frac{1}{2}Am_f \cos(\omega_c + \omega_m)t - \frac{1}{2}Am_f \cos(\omega_c - \omega_m)t$$

where $\beta = k_{FM}A_m / \omega_m$ is the FM modulation index.

Signal spectrum $S_{NBFM}(\omega) = \pi A[\delta(\omega - \omega_c) + \delta(\omega + \omega_c)] + (1/2)\pi Am_f[\delta(\omega - \omega_c - \omega_m) + \delta(\omega + \omega_c + \omega_m) - \delta(\omega - \omega_c + \omega_m) - \delta(\omega + \omega_c - \omega_m)]$

Narrowband FM demodulation method

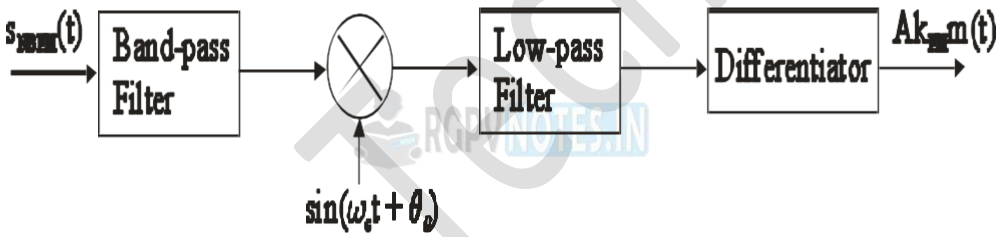


Figure 3.3.NBFM demodulation

Wideband FM (WBFM (β>>1))

When kFM is large, the bandwidth of FM is wide, this type of FM is called wideband FM.

It is usually very difficult to analyze a general FM signal, we will restrict our analysis to the wideband FM with sinusoidal signal.

The message signal $m(t) = Am \cos\omega mt$

Signal waveform (assume $\theta 0 = 0$ for simplicity)

$$s_{FM}(t) = A\cos[\omega_c t + Ak_{FM}\int m(t)dt] = A\cos[\omega_c t + Ak_{FM}\int A_m \cos\omega_m t dt]$$
$$= A\cos[\omega_c t + m_f \sin\omega_m t] = A\cos\omega_c t \cos(m_f \sin\omega_m t) - A\sin\omega_c t \sin(m_f \sin\omega_m t)$$

cos(mf sinωmt) and sin(mf sinωmt) can be expressed in Fourier series

$$\cos(m_f \sin\omega_m t) = J_0(m_f) + 2\sum_{n=1}^{\infty} J_{2n}(m_f)\cos 2n\omega_m t$$

$$\sin(m_f \sin\omega_m t) = 2\sum_{n=1}^{\infty} J_{2n-1}(m_f)\cos(2n-1)\omega_m t$$

Where $$J_n(m_f) = \sum_{m=0}^{\infty} \frac{(-1)^m (\frac{m_f}{2})^{2m+n}}{m!(m+n)!}$$

is the **Bessel function** of the first kind. Thus,

$$s_{FM}(t) = A\cos\omega_c t[J_0(m_f) + 2\sum_{n=1}^{\infty} J_{2n}(m_f)\cos 2n\omega_m t] - A\sin\omega_c t[2\sum_{n=1}^{\infty} J_{2n-1}(m_f)\cos(2n-1)\omega_m t]$$

by using $\cos\alpha\cos\beta = (1/2)[\cos(\alpha+\beta) + \cos(\alpha-\beta)]$,

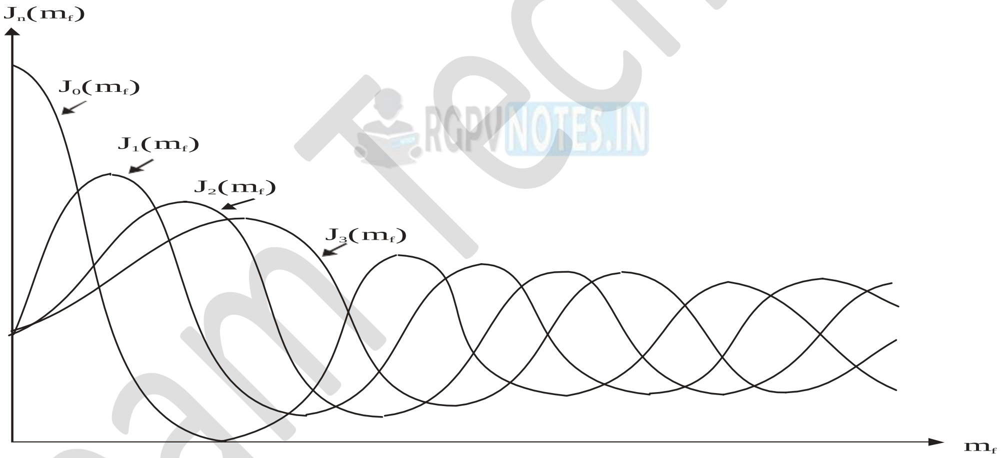
$\sin\alpha\sin\beta = (1/2)[\cos(\alpha-\beta) - \cos(\alpha+\beta)]$

and the property of Bessel function $$J_{-n}(m_f) = (-1)^n J_n(\beta_{FM})$$

sFM(t) can be written in the Bessel function form

$$s_{FM}(t) = A\sum_{n=-\infty}^{\infty} J_n(m_f)\cos[(\omega_c + n\omega_m)t]$$

The spectrum of $S_{FM}(t)$ $$S_{FM}(\omega) = A\pi \sum_{n=-\infty}^{\infty} J_n(m_f)[\delta(\omega - \omega_c - n\omega_m) + \delta(\omega + \omega_c + n\omega_m)]$$



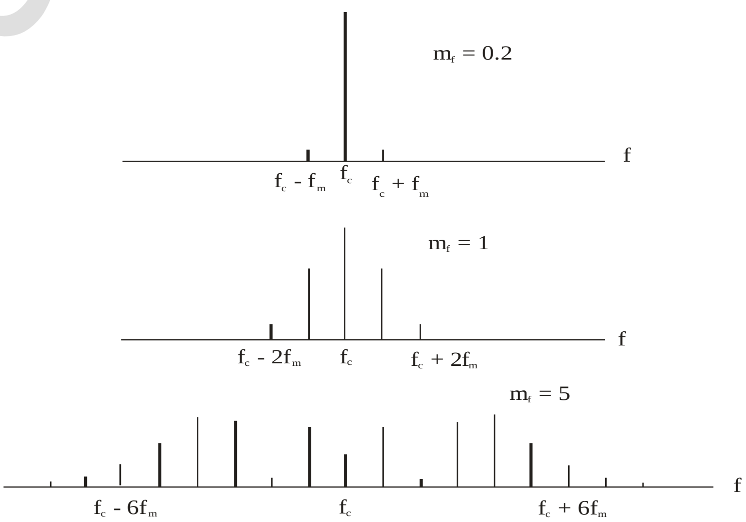


Figure 3.4.Typical plots of $S_{FM}(\omega)$ for different β.

The following observation can be made

- The carrier term cosωct has a magnitude of J0(mf). The maximum value of J0(mf) is 1 when mf = 0, which is equivalent to no modulation.
- Theoretically infinitely number of sidebands are produced, and the amplitude of each sideband is decided by the corresponding Bessel function Jn(mf). The presence of infinite number of sidebands makes the ideal bandwidth of the FM signal infinite.
- When mf is small, there are few sideband frequencies of large amplitude and, when mf is large, there are many sideband frequencies but with smaller amplitudes. Hence, in practice, to determine the bandwidth, it is only necessary to consider a finite number of significant sideband components.
- Thus, the sidebands with small amplitudes can be ignored. The sidebands having amplitudes more than or equal to 1% of the carrier amplitude are known as significant sidebands. They are finite in number.

Generation of frequency Modulation (FM)

Direct method for generation of FM

Reactance Modulator - The reactance modulator is a voltage controlled capacitor and is used to vary an oscillator's frequency or phase. A simplified circuit resembles:

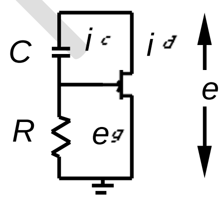


Figure 3.5.Reactance modulator

Since the gate does not draw an appreciable amount of current, applying Ohm's law in the RC branch results in:

$$e_g = i_C$$

$$i_C = \frac{e}{R - jX_C}$$

$$\therefore e_g = \frac{e}{R - jX_C} R$$

The JFET drain current is given by:

$$i_d = g_m e_g = g_m \frac{e}{R - jX_C} R$$

where gm is the trans-conductance.

The impedance as seen from the drain to ground is given by:

$$Z = \frac{e}{i_d} = e\frac{1}{g_m}\frac{R - jX_C}{e}\frac{1}{R} = \frac{1}{g_m} - j\frac{X_C}{g_m R}$$

Since trans-conductance is normally very large, the impedance reduces to:

$$Z \approx -j\frac{X_C}{g_m R} = \frac{-j}{2\pi f C g_m R}$$

The term in the denominator can be thought of as an equivalent capacitance:

$$C_{eq} = C g_m R$$

Then

$$Z = \frac{-j}{2\pi f C_{eq}}$$

Since the equivalent capacitance is larger than the original capacitor, we have created a capacitance amplifier. Because the value of this capacitance is a function of applied voltage, we actually have a voltage controlled capacitor. This device can be used to control an oscillator frequency, thus producing FM.

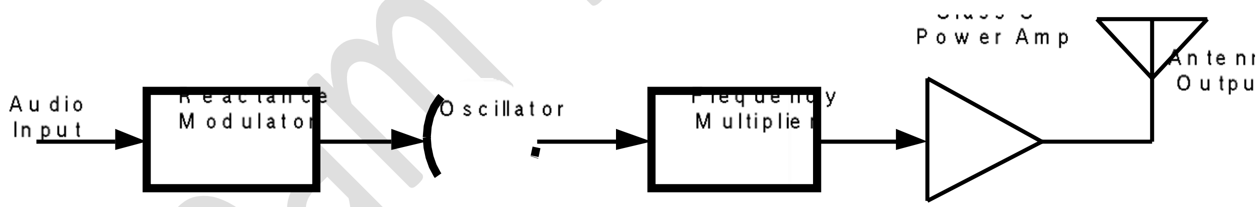


Figure 3.6.FM transmitter

(a) Indirect Method:

Because crystal oscillators are so stable, it is desirable to use them in modulator circuits. However, their extreme stability makes it difficult to modulate their frequency.

Fortunately, it is possible to vary the phase of a crystal oscillator. However, in order to use this as an FM source, the relationship between frequency and phase needs to be reexamined.

Frequency is the rate of change of angle, its first derivative:

$$\omega = \frac{d}{dt}\phi$$

The instantaneous phase angle is comprised of two components, the number of times the signal has gone through its cycle, and its starting point or offset:

$$\phi(t) = \underset{\text{rotating angle}}{\omega_c t} + \underset{\text{offset angle}}{\theta}$$

The instantaneous frequency is therefore:

$$\omega_i = \frac{d}{dt}\phi(t) = \frac{d}{dt}[\omega_c t + \theta] = \omega_c + \frac{d}{dt}\theta$$

From this we observe that the instantaneous frequency of a signal is its un-modulated frequency plus a change. This is equivalent to frequency modulation. Therefore we may write:

$$\omega_c + \frac{d}{dt}\theta = \omega_c + \omega_{eq}$$

$$\omega_{eq} = \frac{d}{dt}\theta$$

$$f_{eq} = \frac{1}{2\pi}\frac{d}{dt}\theta$$

This means that the output of a phase modulator is proportional to the equivalent frequency modulation.

If the angle is proportional to the amplitude of a modulation signal $\theta = k e_m$ Then:

$$f_{eq} = \frac{1}{2\pi}\frac{d}{dt}ke_m$$

and by integrating the modulation signal prior to modulation, we obtain:

$$f_{eq} = \frac{1}{2\pi}\frac{d}{dt}\int ke_m\,dt = \frac{k}{2\pi}e_m$$

This means that the equivalent frequency modulation is directly proportional to the amplitude of a phase modulation signal if the modulation signal is integrated first.

This indirect modulation scheme is the heart of the Armstrong modulator.

Detection of FM

Frequency Discrimination Method

The following figure shows the block diagram of FM demodulator using frequency discrimination method.

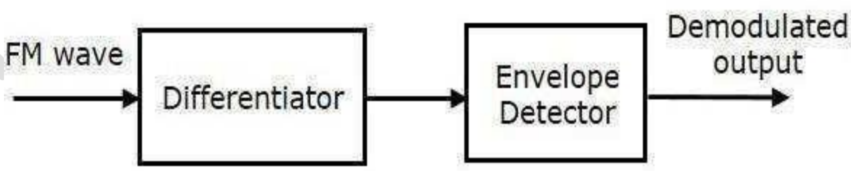


Fig 3.7 Frequency Discrimination Method

This block diagram consists of the differentiator and the envelope detector. Differentiator is used to convert the FM wave into a combination of AM wave and FM wave. This means, it converts the frequency variations of FM wave into the corresponding voltage (amplitude) variations of AM wave. We know the operation of the envelope detector. It produces the demodulated output of AM wave, which is nothing but the modulating signal.

Phase Discrimination Method

The following figure shows the block diagram of FM demodulator using phase discrimination method.

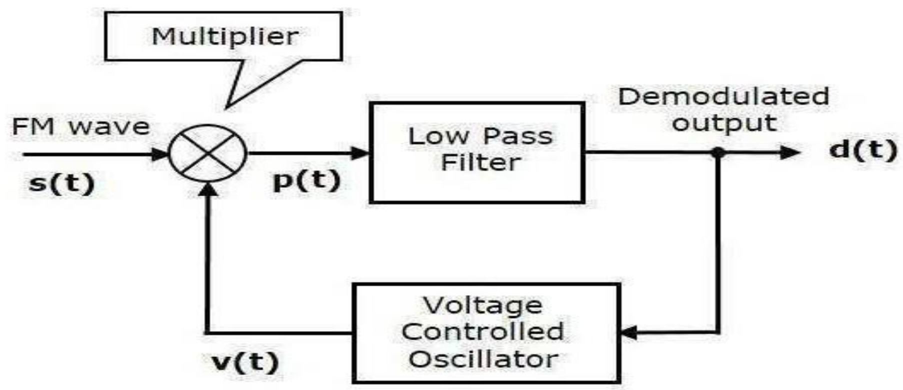


Fig 3.8 Phase Discrimination Method

This block diagram consists of the multiplier, the low pass filter, and the Voltage Controlled Oscillator (VCO). VCO produces an output signal v(t)v(t), whose frequency is proportional to the input signal voltage d(t)d(t). Initially, when the signal d(t)d(t) is zero, adjust the VCO to produce an output signal v(t)v(t), having a carrier frequency and −900−900 phase shift with respect to the carrier signal.

FM wave s(t)s(t) and the VCO output v(t)v(t) are applied as inputs of the multiplier. The multiplier produces an output, having a high frequency component and a low frequency component. Low pass filter eliminates the high frequency component and produces only the low frequency component as its output.

This low frequency component contains only the term-related phase difference. Hence, we get the modulating signal m(t)m(t) from this output of the low pass filter.

Pre-emphasis and De-emphasis

- **Pre-emphasis:** The noise suppression ability of FM decreases with the increase in the frequencies. Thus increasing the relative strength or amplitude of the high frequency components of the message signal before modulation is termed as Pre-emphasis. The Fig3 below shows the circuit of pre-emphasis.

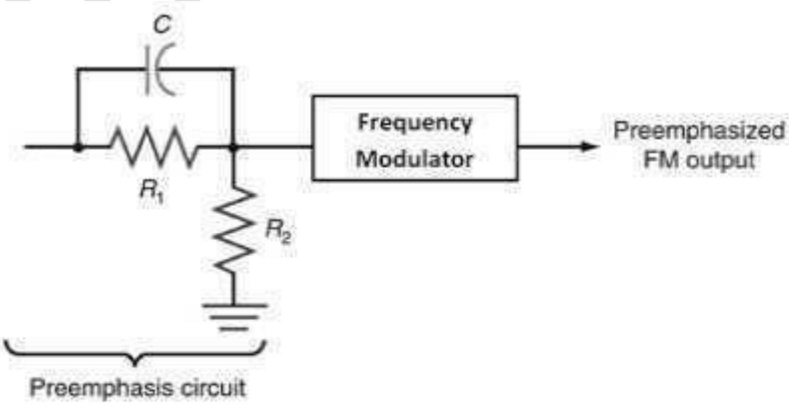


Fig3.9 Pre-emphasis circuit

- **De-emphasis:** In the de-emphasis circuit, by reducing the amplitude level of the received high frequency signal by the same amount as the increase in pre-emphasis is termed as De-emphasis. The Fig4. below shows the circuit of de-emphasis.

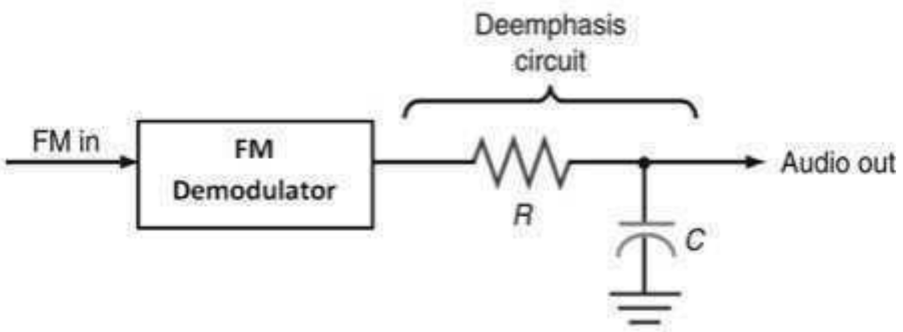


Fig3.10 De-emphasis circuit

- The pre-emphasis process is done at the transmitter side, while the de-emphasis process is done at the receiver side.
- Thus a high frequency modulating signal is emphasized or boosted in amplitude in transmitter before modulation. To compensate for this boost, the high frequencies are attenuated or de-emphasized in the receiver after the demodulation has been performed. Due to pre-emphasis and de-emphasis, the S/N ratio at the output of receiver is maintained constant.
- The de-emphasis process ensures that the high frequencies are returned to their original relative level before amplification.
- Pre-emphasis circuit is a high pass filter or differentiator which allows high frequencies to pass, whereas de-emphasis circuit is a low pass filter or integrator which allows only low frequencies to pass.

FM Transmitter and Receiver Block Diagram

FM Transmitter

FM transmitter is the whole unit, which takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted. The block diagram of FM transmitter is shown in the following figure.

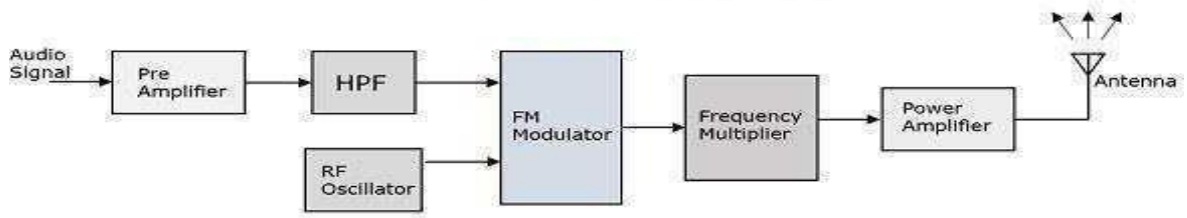


Fig3.11 FM Transmitter

The working of FM transmitter can be explained as follows.

- The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
- This signal is then passed to high pass filter, which acts as a pre-emphasis network to filter out the noise and improve the signal to noise ratio.
- This signal is further passed to the FM modulator circuit.
- The oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal.
- Several stages of frequency multiplier are used to increase the operating frequency. Even then, the power of the signal is not enough to transmit. Hence, a RF power amplifier is used at the end to increase the power of the modulated signal. This FM modulated output is finally passed to the antenna to be transmitted.

<u>**FM Receiver**</u>

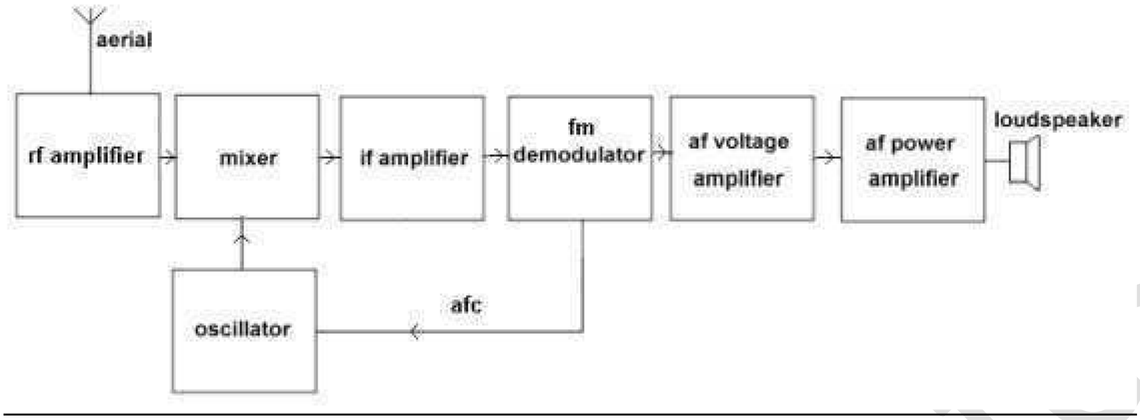


Fig3.12 FM Receiver

<u>**RF section**</u>

- Consists of a pre-selector and an amplifier
- Pre-selector is a broad-tuned band pass filter with an adjustable center frequency used to reject unwanted radio frequency and to reduce the noise bandwidth.
- RF amplifier determines the sensitivity of the receiver and a predominant factor in determining the noise figure for the receiver.

Mixer/converter section

- Consists of a radio-frequency oscillator and a mixer.
- Choice of oscillator depends on the stability and accuracy desired.
- Mixer is a nonlinear device to convert radio frequency to intermediate frequencies (i.e. heterodyning process).
- The shape of the envelope, the bandwidth and the original information contained in the envelope remains unchanged although the carrier and sideband frequencies are translated from RF to IF.

IF section

- Consists of a series of IF amplifiers and band pass filters to achieve most of the receiver gain and selectivity.
- The IF is always lower than the RF because it is easier and less expensive to construct high-gain, stable amplifiers for low frequency signals.
- IF amplifiers are also less likely to oscillate than their RF counterparts.

Detector section

- To convert the IF signals back to the original source information (demodulation).
- Can be as simple as a single diode or as complex as a PLL or balanced demodulator.

Audio amplifier section

- Comprises several cascaded audio amplifiers and one or more speakers

AGC (Automatic Gain Control)

- Adjust the IF amplifier gain according to signal level (to the average amplitude signal almost constant).
- AGC is a system by means of which the overall gain of radio receiver is varied automatically with the variations in the strength of received signals, to maintain the output constant.
- AGC circuit is used to adjust and stabilize the frequency of local oscillator.

Unit -4

AM transmitter & receiver : Tuned radio receiver & super heterodyne, limitation of TRF, IF frequency, image signal rejection, selectivity , sensitivity and fidelity , Noise in AM,FM.

Tuned Radio receiver (TRF)

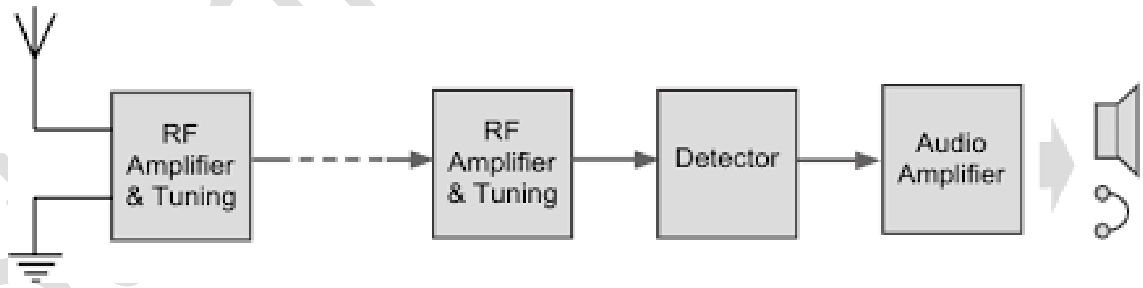
A tuned radio frequency receiver (or TRF receiver) is a type of radio receiver that is composed of one or more tuned radio frequency (RF) amplifier stages followed by a detector (demodulator) circuit to extract the audio signal and usually an audio frequency amplifier. This type of receiver was popular in the 1920s.The TRF receiver was patented in 1916 by Ernst Alexanderson. His concept was that each stage would amplify the desired signal while reducing the interfering ones. Multiple stages of RF amplification would make the radio more sensitive to weak stations, and the multiple tuned circuits would give it a narrower bandwidth and more selectivity than the single stage receiver's common at that time. All tuned stages of the radio must track and tune to the desired reception frequency.

This is in contrast to the modern superheterodyne receiver that must only tune the receiver's RF front end and local oscillator to the desired frequencies; all the following stages work at a fixed frequency and do not depend on the desired reception frequency.

The definition of the tuned radio frequency, TRF receiver is a receiver where the tuning, i.e. selectivity is provided by the radio frequency stage .In essence the simplest tuned radio frequency receiver is a simple crystal set. Tuning is provided by a tuned coil / capacitor combination, and then the signal is presented to a simple crystal or diode detector where the amplitude modulated signal, in this case, is recovered. This is then passed straight to the headphones. As vacuum tube / thermionic vale technology developed, these devices were added to provide more gain.

Typically a TRF receiver would consist of three main sections:

- ***Tuned radio frequency stages:*** This consisted of one of more amplifying and tuning stages. Early sets often had several stages, each proving some gain and selectivity.
- ***Signal detector:*** The detector enabled the audio from the amplitude modulation signal to be extracted. It used a form of detection called envelope detection and used a diode to rectify the signal.
- ***Audio amplifier:*** Audio stages to provide audio amplification were normally, but not always included.

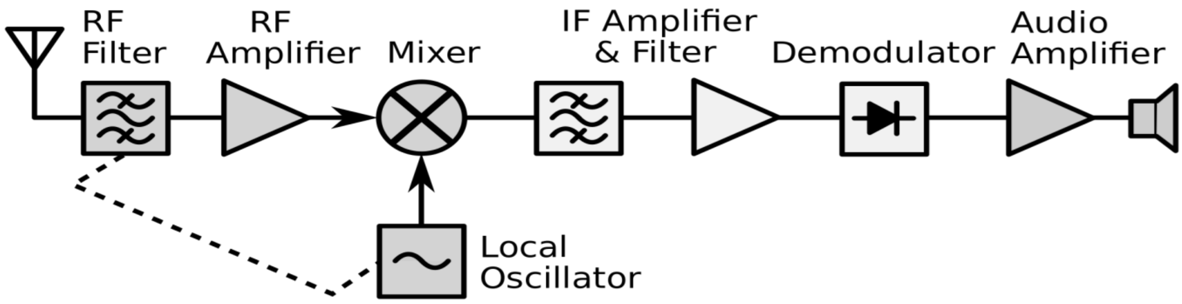


Tuned radio frequency receiver, TRF, block diagram

Superhetrodyne receiver

A superheterodyne receiver, often shortened to superhet, is a type of radio receiver that uses frequency mixing to convert a received signal to a fixed intermediate frequency (IF) which can be more conveniently processed than the original carrier frequency. It was invented by US engineer Edwin Armstrong in 1918 during World War I. Virtually all modern radio receivers use the super heterodyne principle.

The diagram shows the block diagram of a typical single-conversion super heterodyne receiver. The diagram has blocks that are common to super heterodyne receivers. The antenna collects the radio signal. The tuned RF stage with optional RF amplifier provides some initial selectivity; it is necessary to suppress the *image frequency* (see below), and may also serve to prevent strong out-of-pass band signals from saturating the initial amplifier. A local oscillator provides the mixing frequency; it is usually a variable frequency oscillator which is used to tune the receiver to different stations. The frequency mixer does the actual heterodyning that gives the super heterodyne its name; it changes the incoming radio frequency signal to a higher or lower, fixed, intermediate frequency (IF). The IF band-pass filter and amplifier supply most of the gain and the narrowband filtering for the radio. The demodulator extracts the audio or other modulation from the IF radio frequency; the extracted signal is then amplified by the audio amplifier.

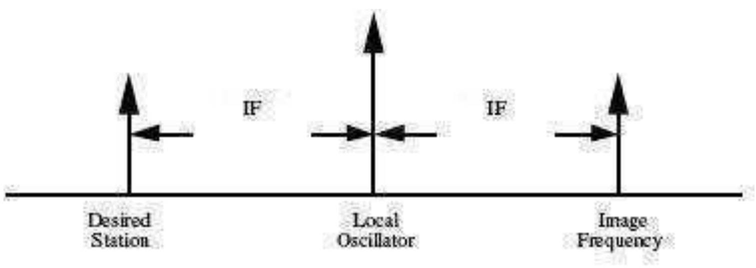


Limitation of TRF

The TRF's disadvantages as "poor selectivity and low sensitivity in proportion to the number of tubes employed. They are accordingly practically obsolete." Selectivity requires narrow bandwidth, and narrow bandwidth at a high radio frequency implies high Q or many filter sections. In contrast a super heterodyne receiver can translate the incoming high radio frequency to a lower intermediate frequency where selectivity is easier to achieve. An additional problem for the TRF receiver is tuning different frequencies. All the tuned circuits need to track to keep the narrow bandwidth tuning. Keeping several tuned circuits aligned is difficult. A super heterodyne receiver only needs to track the RF and LO stages; the onerous selectivity requirements are confined to the IF amplifier which is fixed-tuned.

IF Frequency

When the receiver demodulates the incoming desired signal at fRF, fRF, unfortunately it demodulates down to IF also an unwanted signal at fRF+2fIF.This frequency is called image frequency



To reduce the design complexity of the receivers the IF frequency is chosen in such a way that the signal at fRF+2fIF can be rejected by a simple tunable RF band pass filter such as a tank circuit with a variable capacitor.

Image Signal Rejection

Image rejection is the principal technical challenge in low-IF receivers. The choice of the IF, at low frequency, prevents any image rejection filtering from taking place at RF. In most cases, the polyphase filter is designed to minimize adjacent and alternate channel interference, thus making the filter design more complex and inadvertently more power consuming. Proper choice of the IF frequency, however, can place the image in the adjacent channel. Moreover, in order to discriminate between the IQ signals, the I and Q outputs have to be processed as a complex pair. Having said that, the utility of the polyphase filter is limited by the balance accuracy between the IQ signals. Unlike direct conversion, the ADCs in low-IF architecture have to operate at IF, thus implying stricter requirements on the converters. Finally, second order distortion can result in serious in-band channel interference. In most practical implementations, the low-IF architecture has been limited to somewhat narrowband applications for the reasons cited above.

Selectivity

Selectivity is a measure of the performance of a radio receiver to respond only to the radio signal it is tuned to (such as a radio station) and reject other signals nearby in frequency, such as another broadcast on an adjacent channel. Selectivity is usually measured as a ratio in decibels (dBs), comparing the signal strength received against that of a similar signal on another frequency. LC circuits are often used as filters; the Q ("Quality" factor) determines the bandwidth of each LC tuned circuit in the radio. The L/C ratio, in turn, determines their Q and so their selectivity,

There are practical limits to the increase in selectivity with changing L/C ratio:

- tuning capacitors of large values can be difficult to construct
- stray capacitance, and capacitance within the transistors or valves of associated circuitry, may become significant (and vary with time)
- the series resistance internal to the wire in the coil, may be significant (for parallel tuned circuits especially)
- large inductances imply physically large (and expensive coils) and/or thinner wire (hence worse internal resistance).

Sensitivity and fidelity

Sensitivity of a receiver is defined as the ability of the receiver to amplify weak signals received by the receiver. It is the voltage that must be applied at the input terminals of the receiver to achieve a minimum standard output at the output of the receiver. The factors that determine the sensitivity of super heterodyne receiver are gain of the IF amplifier, Noise figure of the receiver and gain of RF amplifier

The fidelity of a receiver is its ability to accurately reproduce, in its output, the signal that appears at its input. The broader the band passed by frequency selection circuits, the greater your fidelity. Good selectivity requires that a receiver pass a narrow frequency band. Good fidelity requires that the receiver pass a broader band to amplify the outermost frequencies of the sidebands. Receivers you find in general use are a compromise between good selectivity and high fidelity.

Noise in FM and AM

The signal to noise ratio, SNR or S/N ratio is one of the most straightforward methods of measuring radio receiver sensitivity. It defines the difference in level between the signal and the noise for a given signal level. The lower the noise generated by the receiver, the better the signal to noise ratio.

As with any sensitivity measurement, the performance of the overall radio receiver is determined by the performance of the front end RF amplifier stage. Any noise introduced by the first RF amplifier will be added to the signal and amplified by subsequent amplifiers in the receiver. As the noise introduced by the first RF amplifier will be amplified the most, this RF amplifier becomes the most critical in terms of radio receiver sensitivity performance. Thus the first amplifier of any radio receiver should be a low noise amplifier.

Signal to noise ratio formula

The signal to noise ratio is the ratio between the wanted signal and the unwanted background noise. It can be expressed in its most basic form using the S/N ratio formula below:

$$SNR = P_{signal}/P_{noise}$$

It is more usual to see a signal to noise ratio expressed in a logarithmic basis using decibels with the formula below:

$$SNR(dB) = 10\log_{10}(P_{signal}/P_{noise})$$

If all levels are expressed in decibels, then the formula can be simplified to the equation below:

$$SNR(dB) = P_{signal}(dB) - P_{noise}(dB)$$

The power levels may be expressed in levels such as dBm (decibels relative to a milliwatt, or to some other standard by which the levels can be compared.

Unit -5

Noise: Classification of noise , sources of noise, Noise figure and Noise temperature , Noise bandwidth, Noise figure measurement , Noise in analog modulation , Figure of merit for various AM and FM , effect of noise on AM and FM receivers.

Classification of noise and its sources

Noise may be put into following two categories: **External noises** i.e. noise whose sources are external and **Internal noise** i.e. whose noise sources are generated internally by the circuit or the communication system. **External noises** i.e. noise whose sources are external. Internal noise on the other hand can be easily evaluated mathematically and can be reduced to a great extent by proper design

External noise may be classified into the following three types:

1. Atmospheric noises :

Atmospheric Noise Atmospheric noise or static is caused by lighting discharges in thunderstorms and other natural electrical disturbances occurring in the atmosphere. These electrical impulses are random in nature. Hence the energy is spread over the complete frequency spectrum used for radio communication. Atmospheric noise accordingly consists of spurious radio signals with components spread over a wide frequency range. These spurious radio waves constituting the noise get propagated over the earth in the same fashion as the desired radio waves of the same frequency. Accordingly at a given receiving point, the receiving antenna picks up not only the signal but also the static from all the thunderstorms, local or remote. The field strength of atmospheric noise varies approximately inversely with the frequency. Thus large atmospheric noise is generated in low and medium frequency (broadcast) bands while very little noise is generated in the VHF and UHF bands. Further VHF and UHF components of noise are limited to the line-of sight (less than about 80 Km) propagation. For these two-reasons, the atmospheric noise becomes less severe at Frequencies exceeding about 30 MHz.

2. Extraterrestrial noises:

There are numerous types of extraterrestrial noise or space noises depending on their sources. However, these may be put into following two subgroups.

(a) Solar noise : This is the electrical noise emanating from the sun. Under quite conditions, there is a steady radiation of noise from the sun. This results because sun is a large body at a very high temperature (exceeding 6000°C on the surface), and radiates electrical energy in the form of noise over a very wide frequency spectrum including the spectrum used for radio communication. The intensity produced by the sun varies with time. In fact, the sun has a repeating 11-Year noise cycle. During the peak of the cycle, the sun produces some amount of noise that causes tremendous radio signal interference, making many frequencies unusable for communications. During other years. the noise is at a minimum level

(b)Cosmic noise :Distant stars are also suns and have high temperatures. These stars, therefore, radiate noise in the same way as our sun. The noise received from these distant stars is thermal noise (or black body noise) and is distributing almost uniformly over the entire sky. We also receive noise from the center of our own galaxy (The Milky Way) from other distant galaxies and from other virtual point sources such as quasars and pulsars.

3. Man-made noises or industrial noises :

By man-made noise or industrial- noise is meant the electrical noise produced by such sources as automobiles and aircraft ignition, electrical motors and switch gears, leakage from high voltage lines, fluorescent lights, and numerous other heavy electrical machines. Such noises are produced by the arc discharge taking place during operation of these machines. Such man-made noise is most intensive in industrial and densely populated areas. Man-made noise in such areas far exceeds all other sources of noise in the frequency range extending from about 1 MHz to 600 MHz

Internal noise may be put into the following five categories.

1. Thermal noise or white noise or Johnson noise :

Conductors contain a large number of 'free" electrons and "ions" strongly bound by molecular forces. The ions vibrate randomly about their normal (average) positions, however, this vibration being a function of the temperature. Continuous collisions between the electrons and the vibrating ions take place. Thus there is a continuous transfer of energy between the ions and electrons. This is the source of resistance in a conductor. The movement of free electrons constitutes a current which is purely random in nature and over a long time averages zero. There is a random motion of the electrons which give rise to noise voltage called thermal noise.

2. Shot noise :

Intermediation noise is produced when there is some non linearity in the transmitter, receiver, or intervening transmission system. Normally, these components behave as linear systems; that is, the output is equal to the input, times a constant. In a nonlinear system, the output is a more complex function of the input. Such non linearity can be caused by component malfunction or the use of excessive signal strength. It is under these circumstances that the sum and difference terms occur.

3. Flicker noise:

Flicker noise is a type of electronic noise with a 1/f, or pink power density spectrum. It is therefore often referred to as 1/f noise or pink noise, though these terms have wider definitions. It occurs in almost all electronic devices, and can show up with a variety of other effects, such as impurities in a conductive channel, generation and recombination noise in a transistor due to base current, and so on. 1/f noise in current or voltage is always related to a direct current because it is a resistance fluctuation, which is transformed to voltage or current fluctuations via Ohm's law.

4. Transit Time noise :

Transit time is the duration of time that it takes for a current carrier such as a hole or current to move from the input to the output. The devices themselves are very tiny, so the distances involved are minimal. Yet the time it takes for the current carriers to move even a short distance is finite. At low frequencies this time is negligible. But when the frequency of operation is high and the signal being processed is the magnitude as the transit time, then problem can occur. The transit time shows up as a kind of random noise within the device, and this is directly proportional to the frequency of operation.

5. Avalanche noise:

Avalanche noise is the noise produced when a junction diode is operated at the onset of avalanche breakdown, a semiconductor junction phenomenon in which carriers in a high voltage gradient develop sufficient energy to dislodge additional carriers through physical impact, creating ragged current flows.

<u>Noise Figure and noise Factor</u>

Noise figure (NF) and noise factor (*F*) are measures of degradation of the signal-to-noise ratio (SNR), caused by components in a signal chain. It is a number by which the performance of an amplifier or a radio receiver can be specified, with lower values indicating better performance.

The noise factor is defined as the ratio of the output noise power of a device to the portion thereof attributable to thermal noise in the input termination at standard noise temperature T_0 (usually 290 K). The noise factor is thus the ratio of actual output noise to that which would remain if the device itself did not introduce noise, or the ratio of input SNR to output SNR.

The noise *figure* is simply the noise *factor* expressed in decibels (dB). The noise figure is the difference in decibels (dB) between the noise output of the actual receiver to the noise output of an "ideal" receiver with the same overall gain and bandwidth when the receivers are connected to matched sources at the standard noise temperature T_0 (usually 290 K).

The noise factor *F* of a system is defined as :

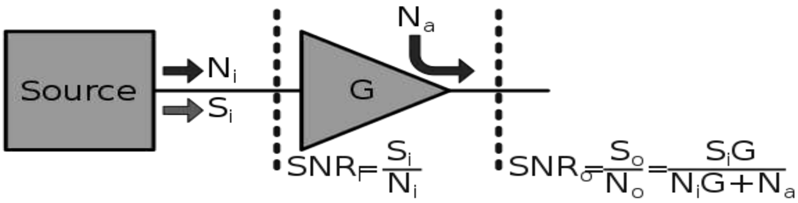
$F= SNR_i / SNR_o,$

Where SNR_i and SNR_o are the input and output signal-to-noise ratios respectively. The SNR quantities are power ratios.

The noise figure NF is defined as the noise factor in dB:

Noise Figure (NF)= $10 \, Log_{10} (F)=10\log_{10}$ **(**SNR_i / SNR_o**)** = $SNR_{i,dB} - SNR_{o,dB}$

This makes the noise figure a useful figure of merit for terrestrial systems, where the antenna effective temperature is usually near the standard 290 K. In this case, one receiver with a noise figure, say 2 dB better than another, will have an output signal to noise ratio that is about 2 dB better than the other. However, in the case of satellite communications systems, where the receiver antenna is pointed out into cold space, the antenna effective temperature is often colder than 290 K. In these cases a 2 dB improvement in receiver noise figure will result in more than a 2 dB improvement in the output signal to noise ratio. For this reason, the related figure of *effective noise temperature* is therefore often used instead of the noise figure for characterizing satellite-communication receivers and low-noise amplifiers. In heterodyne systems, output noise power includes spurious contributions from image-frequency transformation, but the portion attributable to thermal noise in the input termination at standard noise temperature includes only that which appears in the output via the principal frequency transformation of the system and excludes that which appears via the image frequency transformation.



Noise Temperature

Noise temperature is one way of expressing the level of available noise power introduced by a component or source. The power spectral density of the noise is expressed in terms of the temperature (in kelvins) :

$$P_N/B=K_bT$$

where:

- P_N is the noise power (in W, watts)
- B is the total bandwidth (Hz, hertz) over which that noise power is measured
- K_b is the Boltzmann constant (1.381×10^{-23} J/K, joules per kelvin)
- T is the noise temperature (K, kelvin)

Thus the noise temperature is proportional to the power spectral density of the noise .That is the power that would be absorbed from the component or source by a matched load. Noise temperature is generally a function of frequency, unlike that of an ideal resistor which is simply equal to the actual temperature of the resistor at all frequencies.

Noise Figure measurement

Noise temperature is one way of expressing the level of available noise power introduced by a component or source. The power spectral density of the noise is expressed in terms of the temperature (in kelvins) :

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where:

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Figure of Meritfor AM andFMSystem

Signal-to-Noise Ratio (SNR) is the ratio of the signal power to noise power. The higher the value of SNR, the greater will be the quality of the received output. Signal-to-Noise Ratio at different points can be calculated using the following formulas.

Input SNR = $(SNR)_I$=(Average power of modulating signal) / (Average power of noise at input)

Output SNR= $(SNR)_O$=(Average power of demodulated signal / (Average power of noise at output)

Channel SNR = $(SNR)_C$=(Average power of modulated signal)/ Average power of noise in message bandwidth)

The ratio of output SNR and input SNR can be termed as **Figure of Merit**. It is denoted by **Y**. It describes the performance of a device.

$$Y=(SNR)_O/(SNR)_I$$

Figure of merit of AM or FM receiver is

$$Y=(SNR)_O/(SNR)_C$$

It is so because for a receiver, the channel is the input.

