DEPARTMENT OF

ELECTRONICS AND COMMUNICATION ENGINEERING

EC208

ANALOG COMMUNICATION ENGINEERING

COURSE MATERIAL

JAWAHARLAL COLLEGE OF ENGINEERING & TECHNOLOGY

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MODULE I

INTRODUCTION TO COMMUNICATION SYSTEM

Communication System

Communication is the process of establishing connection or link between two points for information exchange.
It involves

- Generation of Message signal
- Encoding of the signal
- Transmission of encoded signal to destination through the channel
- Decoding & Reproduction of the original signal

Elements of a Communication System

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

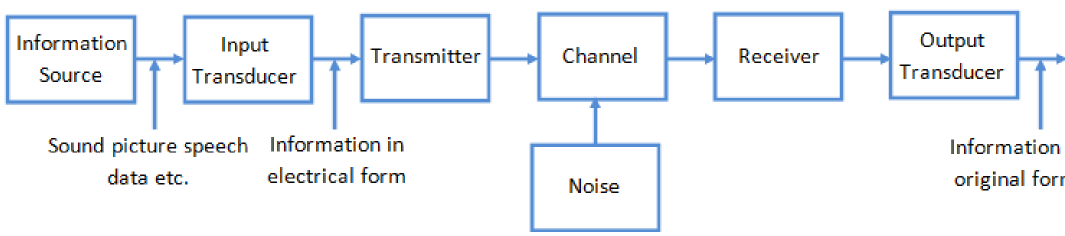


Fig 1: Block Diagram of Communication System

The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.

(i) 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.

(ii) 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.

(iii) 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.

Modulation is the main function of the transmitter. In modulation, the message signal is superimposed upon the high-frequency carrier signal.

In short, we can say that inside the transmitter, signal processing such as restriction of range of audio frequencies, amplification and modulation of are achieved.

All these processing of the message signal are done just to ease the transmission of the signal through the channel.

(iv) 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 are wire lines, microwave links and optical fibres. 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 fibre is a low-loss, well-controlled, guided optical medium. Optical fibres 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 tend 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.

(v) 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.

(vi) Destination

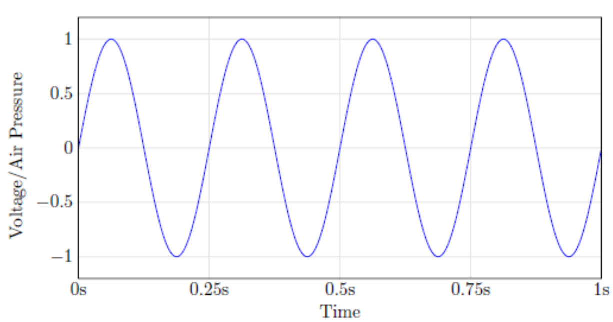
Destination is the final stage which is used to convert an electrical message signal into its original form.

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.

TIME & FREQUENCY DOMAINS

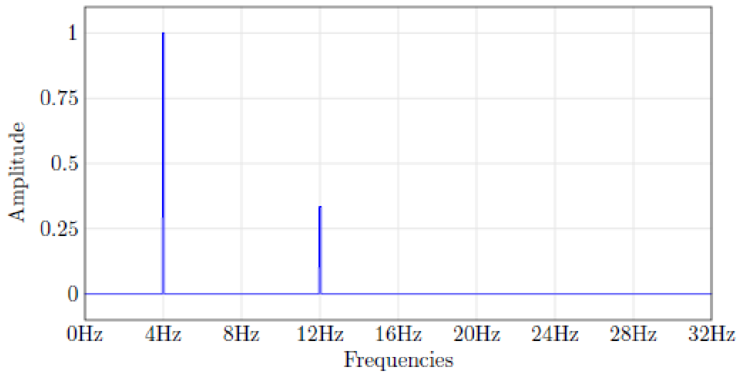
Time Domain

Representation of a physical quantity over time is called Time domain representation .Here Horizondal axis will be time.



Frequency Domain

Representation of a physical quantity over Frequency is called Frequency domain representation .Here Horizondal axis will be Frequency



NEED OF MODULATION/ADVANTAGES OF MODULATION

1. Reduction in the height of antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Multiplexing is possible
5. Improves quality of reception

We will discuss each of these advantages in detail below.

1. *Reduction in the height of antenna*

For the transmission of radio signals, the antenna height must be multiple of $\lambda/4$,where λ is the wavelength.

$\lambda = c/f$

Where c : is the velocity of light

f: is the frequency of the signal to be transmitted

The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows:

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10\times10^3} = 7500\ meters\ i.e.\ 7.5\ km$$

The antenna of this height is practically impossible to install.

Now, let us consider a modulated signal at f = 1 MHz . The minimum antenna height is given by,

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10\times10^6} = 75\ meters$$

This antenna can be easily installed practically . Thus, modulation reduces the height of the antenna .

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver can not separate them from each other .

Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals .

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted . They get heavily attenuated .

The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance .

The modulation process increases the frequency of the signal to be transmitted . Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously .

This is possible only with modulation.

The multiplexing allows the same channel to be used by many signals . Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time .

5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent . This improves quality of reception

NOISE IN COMMUNCATION SYSTEM

Noise is often described as the limiting factor in communication systems: indeed if there as no noise there would be virtually no problem in communications.

Noise is a general term which is used to describe an unwanted signal which affects a wanted signal. These unwanted signals arise from a variety of sources which may be considered in one of two main categories:-

a) Interference, usually from a human source (man-made)
b) Naturally occurring random noise.

Interference arises for example, from other communication systems (cross talk), 50 Hz supplies (hum) and harmonics, switched mode power supplies, thyristor circuits, ignition (car spark plugs) motors … etc. Interference can in principle be reduced or completely eliminated by careful engineering (i.e. good design, suppression, shielding etc). Interference is essentially deterministic (i.e. random, predictable), however observe.

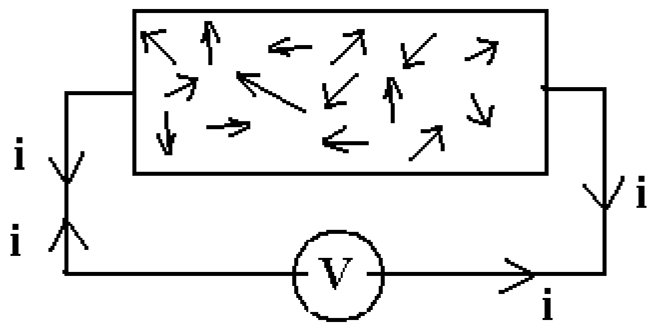
When the interference is removed, there remains naturally occurring noise which is essentially random (non-deterministic),. Naturally occurring noise is inherently present in electronic communication systems from either 'external' sources or 'internal' sources.

Naturally occurring external noise sources include atmosphere disturbance (e.g. electric storms, lighting, ionospheric effect etc), so called 'Sky Noise' or Cosmic noise which includes noise from galaxy, solar noise and 'hot spot' due to oxygen and water vapour resonance in the earth's atmosphere. These sources can seriously affect all forms of radio transmission and the design of a

2. THERMAL NOISE (JOHNSON NOISE)

This type of noise is generated by all resistances (e.g. a resistor, semiconductor, the resistance of a resonant circuit, i.e. the real part of the impedance, cable etc).

Free electrons are in contact random motion for any temperature above absolute zero (0 degree K, ~ -273 degree C). As the temperature increases, the random motion increases, hence thermal noise, and since moving electron constitute a current, although there is no net current flow, the motion can be measured as a mean square noise value across the resistance.



Experimental results (by Johnson) and theoretical studies (by Nyquist) give the mean square noise voltage as $\bar{V}^2 = 4kTBR \ (volt^2)$

Where k = Boltzmann's constant = 1.38 x 10^{-23} Joules per K

T = absolute temperature

B = bandwidth noise measured in (Hz)

R = resistance (ohms)

The law relating noise power, N, to the temperature and bandwidth is

N = k TB watts

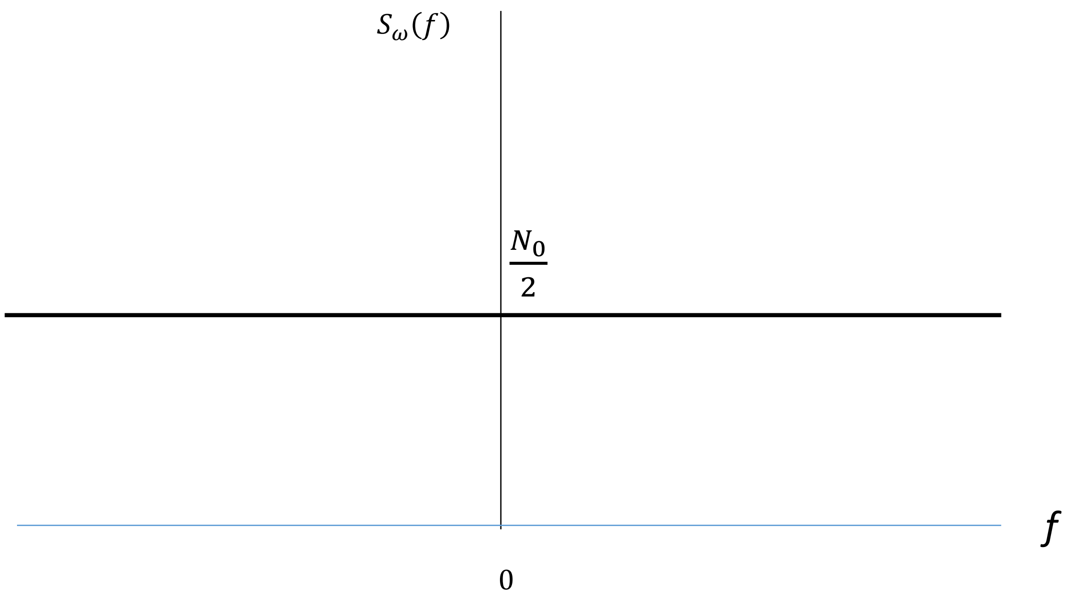
WHITE NOISE

'white noise' is the one whose power spectral density is independent of the operating frequency. ie.it has a uniform 'spectral density'.

Note – <u>noise power spectral density</u> is the noise power measured in a 1 Hz bandwidth i.e. watts per Hz. A uniform spectral density means that if we measured the thermal noise in any 1 Hz bandwidth from ~ 0Hz → 1 MHz → 1GHz …….. 10,000 GHz etc we would measure the same amount of noise.

We can express the power spectral density of white noise ,with sample function ω(t) as

$$S_\omega(f) = \frac{N_0}{2}$$



Characteristics of white Noise

3. SHOT NOISE

Shot noise was originally used to describe noise due to random fluctuations in electron emission from cathodes in vacuum tubes (called shot noise by analogy with lead shot). Shot noise also occurs in semiconductors due to the liberation of charge carriers, which have discrete amount of charge, in to potential barrier region such as occur in *pn* junctions. The discrete amounts of charge give rise to a current which is effectively a series of current pulses.

For *pn* junctions the mean square shot noise current is

$$I_n^2 = 2\left(I_{DC} + 2I_o\right)q_e B \qquad (amps)^2$$

Where

I_{DC} is the direct current as the *pn* junction (amps)

I_0 is the reverse saturation current (amps)

q_e is the electron charge = 1.6 x 10^{-19} coulombs

B is the effective noise bandwidth (Hz)

Shot noise is found to have a uniform spectral density as for thermal noise.

4. LOW FREQUENCY OR FLICKER NOISE

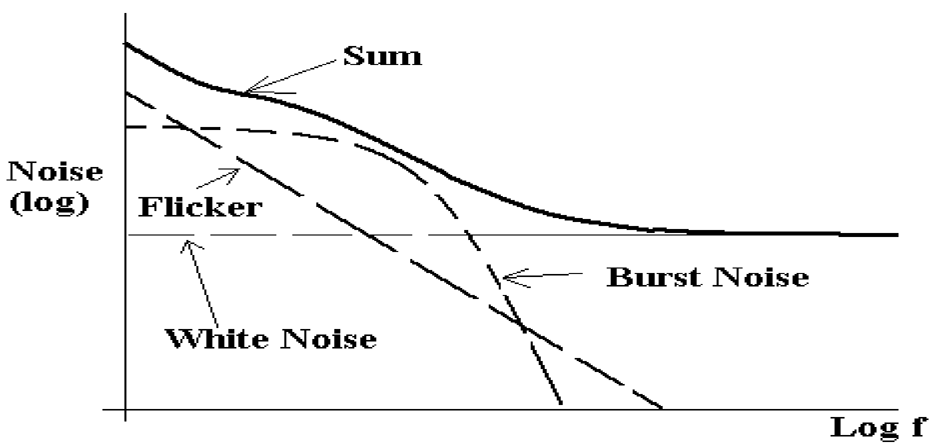
Active devices, integrated circuit, diodes, transistors etc also exhibits a low frequency noise, which is frequency dependent (i.e. non uniform) known as flicker noise or 'one – over – f' noise. The mean square value is found to be proportional to $\left(\frac{1}{f}\right)^n$ where f is the frequency and n= 1. Thus the noise at higher frequencies is less than at lower frequencies. Flicker noise is due to impurities in the material which in turn cause charge carrier fluctuations.

6. BURST NOISE OR POPCORN NOISE

Some semiconductors also produce burst or popcorn noise with a spectral density which is proportional to $\left(\frac{1}{f}\right)^2$.

GENERAL COMMENTS

The diagram below illustrates the variation of noise with frequency.



For frequencies below a few KHz (low frequency systems), flicker and popcorn noise are the most significant, but these may be ignored at higher frequencies where 'white' noise predominates.

Thermal noise is always presents in electronic systems. Shot noise is more or less significant depending upon the specific devices used for example as FET with an insulated gate avoids junction shot noise. As noted in the preceding discussion, all transistors generate other types of 'non-white' noise which may or may not be significant depending on the specific device and application. Of all these types of noise source, white noise is generally assumed to be the most significant and system analysis is based on the assumption of thermal noise. This assumption is reasonably valid for radio systems which operates at frequencies where non-white noise is greatly reduced and which have low noise 'front ends' which, as shall be discussed, contribute most of the internal (circuit) noise in a receiver system. At radio frequencies the sky noise contribution is significant and is also (usually) taken into account.

Obviously, analysis and calculations only gives an indication of system performance. Measurements of the noise or signal-to-noise ratio in a system include all the noise, from whatever source, present at the time of measurement and within the constraints of the measurements or system bandwidth

NOISE TEMPERATURE

The equivalent noise temperature of a system is defined as the temperature at which a noisy resistor has to be maintained such that , by connecting the resistor to the input of noiseless version of the system, it produces the same available noise power at the output of the system as that produced by all the sources of noise in the actual system.

SIGNAL POWER TO NOISE POWER RATIO

Let S= signal power (mW) & N= noise power (mW) Then

$$\left(\frac{S}{N}\right)_{ratio} = \frac{S}{N}$$

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

Also recall that

$$S_{dBm} = 10\log_{10}\left(\frac{S(mW)}{1mW}\right)$$

$$and\ N_{dBm} = 10\log_{10}\left(\frac{N(mW)}{1mW}\right)$$

$$i.e.\left(\frac{S}{N}\right)_{dB} = 10\log_{10} S - 10\log_{10} N$$

$$\left(\frac{S}{N}\right)_{dB} = S_{dBm} - N_{dBm}$$

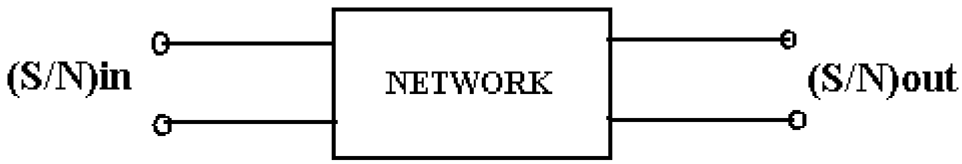
Powers are usually measured in dBm (or dBw) in communications systems. The equation $\left(\frac{S}{N}\right)_{dB} = S_{dBm} - N_{dBm}$ is often the most useful.

The $\left(\frac{S}{N}\right)$ at various stages in a communication system gives an indication of system quality and performance in terms of error rate in digital data communication systems and 'fidelity' in case of analogue communication systems. (Obviously, the larger the $\left(\frac{S}{N}\right)$, the better the system will be).

Noise, which accompanies the signal is usually considered to be additive (in terms of powers) and its often described as Additive White Gaussian Noise, AWGN, noise. Noise and signals may also be multiplicative and in some systems at some levels of $\left(\frac{S}{N}\right)$, this may be more significant then AWGN.

NOISE FACTOR – NOISE FIGURE

Consider the network shown below, in which $\left(\frac{S}{N}\right)_{IN}$ represents the $\left(\frac{S}{N}\right)$ at the input and $\left(\frac{S}{N}\right)_{OUT}$ represents the $\left(\frac{S}{N}\right)$ at the output.



In general $\left(\frac{S}{N}\right)_{IN} \geq \left(\frac{S}{N}\right)_{OUT}$, i.e. the network 'adds' noise (thermal noise tc from the network devices) so that the output (S/N) is generally worst than the input.

The amount of noise added by the network is embodied in the Noise Factor F, which is defined by

$$\text{Noise factor } F = \frac{\left(S/N\right)_{IN}}{\left(S/N\right)_{OUT}}$$

F equals to 1 for noiseless network and in general F > 1.

The noise figure in the noise factor quoted in dB

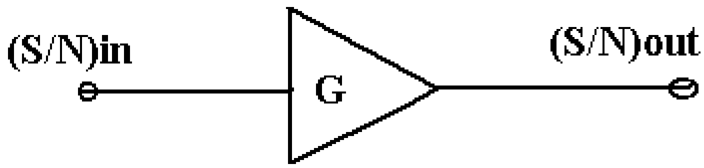
i.e. Noise Figure F dB = 10 $\log_{10}$ F F ≥ 0 dB

The noise figure / factor is the measure of how much a network degrades the $(S/N)_{IN}$, the lower the value of F, the better the network.

The network may be active elements, e.g. amplifiers, active mixers etc, i.e. elements with gain > 1 or passive elements, e.g. passive mixers, feeders cables, attenuators i.e. elements with gain <1.

Noise Figure – Noise Factor for Active Elements

For active elements with power gain G>1, we have



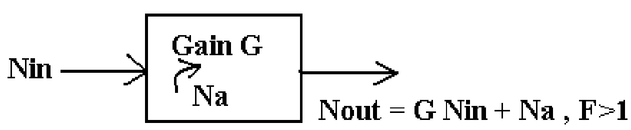
$$F = \frac{\left(S/N\right)_{IN}}{\left(S/N\right)_{OUT}} = \frac{S_{IN}}{N_{IN}} \frac{N_{OUT}}{S_{OUT}}$$

But S_{OUT} = G S_{IN}

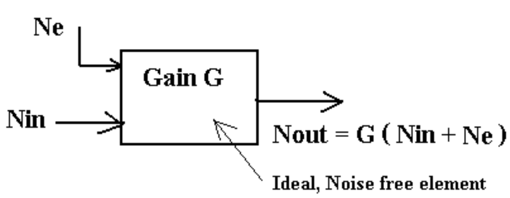
Therefore $$F = \frac{S_{IN}}{N_{IN}} \frac{N_{OUT}}{G\,S_{IN}}$$

$$F = \frac{N_{OUT}}{G\,N_{IN}}$$

If the N_{OUT} was due only to G times N_{IN} the F would be 1 i.e. the active element would be noise free. Since in general F v> 1 , then N_{OUT} is increased by noise due to the active element i.e.



Na represents 'added' noise measured at the output. This added noise may be referred to the input as extra noise, i.e. as equivalent diagram is



Ne is extra noise due to active elements referred to the input; the element is thus effectively noiseless.

Hence $$F = \frac{N_{OUT}}{G\,N_{IN}} = F = \frac{G(N_{IN} + N_e)}{G\,N_{IN}}$$

Rearranging gives,

$$N_e = (F - 1)\, N_{IN}$$

Noise Figure – Noise Factor for Passive Elements

The theoretical argument for passive networks (e.g. feeders, passive mixers, attenuators) that is networks with a gain < 1 is fairly abstract, and in essence shows that the noise at the input, N_{IN} is attenuated by network, but the added noise Na contributes to the noise at the output such that N_{OUT} = N_{IN} .

Thus, since F = $\frac{S_{IN}}{N_{IN}} \frac{N_{OUT}}{S_{OUT}}$ and N_{OUT} = N_{IN} .

$$F = \frac{S_{IN}}{G\, S_{IN}} = \frac{1}{G}$$

If we let L denote the insertion loss (ratio) of the network i.e. insertion loss

$$L_{dB} = 10 \log L$$

Then

$$L = \frac{1}{G}$$ and hence for passive network

$$F = L$$

Also, since T_e = (F-1) T_S

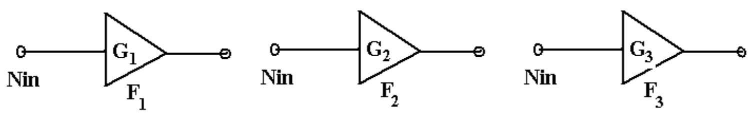
Then for passive network

$$T_e = (L-1)\, T_S$$

Where T_e is the equivalent noise temperature of a passive device referred to its input.

System Noise Figure

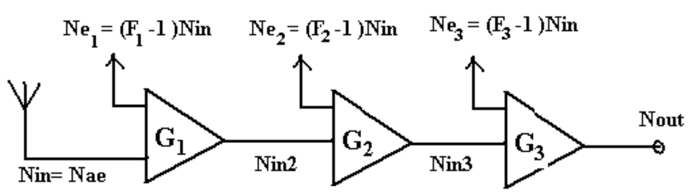
Assume that a system comprises the elements shown below, each element defined and specified separately.



The gains may be greater or less than 1, symbols F denote noise factor (not noise figure, i.e. not in dB).

Assume that these are now cascaded and connected to an aerial at the input, with

$N_{IN} = N_{ae}$ from the aerial.



Note: - N_{IN} for each stage is equivalent to a source at a temperature of 290 K since this is how each element is specified. That is, for each device/ element is specified at 290 K.

Now , $N_{OUT} = G_3\left(N_{IN3} + N_{e3}\right)$

$$= G_3\left(N_{IN3} + (F_3 - 1)N_{IN}\right)$$

Since $N_{IN3} = G_2\left(N_{IN2} + N_{e2}\right) = G_2\left(N_{IN2} + (F_2 - 1)N_{IN}\right)$

and similarly $N_{IN2} = G_1\left(N_{ae} + (F_1 - 1)N_{IN}\right)$

then

$$N_{OUT} = G_3\left[G_2\left[G_1 N_{ae} + G_1(F_1 - 1)N_{IN}\right] + G_2\left(F_2 - 1\right)N_{IN}\right] + G_3\left(F_3 - 1\right)N_{IN}$$

The overall system Noise Factor is

$$F_{sys} = \frac{N_{OUT}}{GN_{IN}} = \frac{N_{OUT}}{G_1 G_2 G_3 N_{ae}}$$

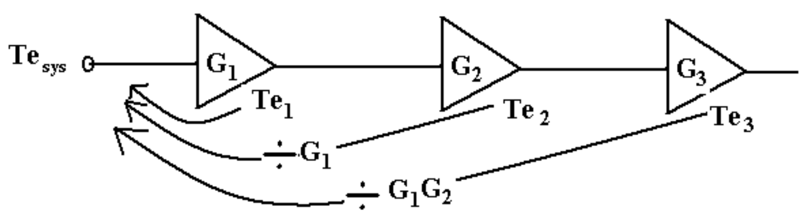
$$= 1 + (F_1 - 1)\frac{N_{IN}}{N_{ae}} + \frac{(F_2 - 1)}{G_1}\frac{N_{IN}}{N_{ae}} + \frac{(F_3 - 1)}{G_1 G_2}\frac{N_{IN}}{N_{ae}}$$

If we assume N_{ae} is ≈ N_{IN} , i.e. we would measure and specify F_{sys} under similar conditions as F_1, F_2 etc (i.e. at 290 K), then for *n* elements in cascade.

$$F_{sys} = F_1 + \frac{(F_2 - 1)}{G_1} + \frac{(F_3 - 1)}{G_1 G_2} + \frac{(F_4 - 1)}{G_1 G_2 G_3} + \ldots\ldots\ldots. + \frac{(F_n - 1)}{G_1 G_2 \ldots\ldots\ldots G_{n-1}}$$

The equation is called FRIIS Formula.

This equation indicates that the system noise factor depends largely on the noise factor of the first stage if the gain of the first stage is reasonably large. This explains the desire for "low noise front ends" or low noise most head preamplifiers for domestic TV reception. There is a danger however; if the gain of the first stage is too large, large and unwanted signals are applied to the mixer which may produce intermodulation distortion. Some receivers apply signals from the aerial directly to the mixer to avoid this problem. Generally a first stage amplifier is designed to have a good noise factor and some gain to give an acceptable overall noise figure.



T_{e1} is already referred to input of 1st stage.

T_{e2} is referred to input of the 2nd stage – to refer this to the input of the 1st stage we must divide T_{e2} by G_1.

T_{e3} is referred to input of third stage, ($\div G_1G_2$) to refer to input of 1st stage, etc.

It is often more convenient to work with noise temperature rather than noise factor.

Given a noise factor we find T_e from T_e = (F-1)290.

Note also that the gains (G_1G_2 G_3 etc) may be gains > 1 or gains <1, i.e. losses L where L = $\frac{1}{G}$.

NARROW BAND NOISE

The receiver of communication system usually includes some provision of preprocessing of received signal. This may be in the form of narrowband filter whose bandwidth just large enough to pass the received signals The noise appears at the output of such filter is called ***Narrowband noise.***

There are two specific representations for Narrowband noise

1. The narrowband defined in terms of a pair of components called ***inphase*** & ***Quadrature components***.
2. The narrowband defined in terms of two other components called envelope & Phase.

REPRESENTATION OF NARROWBAND NOISE IN TEMS OF CALLED *INPHASE* & *QUADRATURE COMPONENTS.*

Narrowband Noise n(t) of bandwidth 2B centered on frequency fc can be represented in standard form as

$$n(t) = n_I(t)\cos 2\pi f_c t - n_Q(t)\cos 2\pi f_c t$$

Where $n_I(t)$ *is called* ***inphase*** component ***and*** $n_Q(t)$ ***Quadrature*** component of $n_I(t)$.

The inphase & ***Quadrature components*** have the properties

1. The inphase component $n_I(t)$ *and Quadrature* component $n_Q(t)$ of narrowband noise n(t) have zero mean
2. If the narrowband noise n(t) is Gaussian then its inphase component $n_I(t)$ *and Quadrature* component $n_Q(t)$ are jointly Gaussian

3. If the narrowband noise n(t) is stationary then its inphase component $n_I(t)$ *and Quadrature* component $n_Q(t)$ are jointly stationary.
4. Both inphase component $n_I(t)$ *and Quadrature* component $n_Q(t)$ have the same power spectral density,which is related to the power spectral density $S_N(f)$ of narrowband noise n(t) as

$$S_{N1}(f) = S_{NQ}(f) = \begin{cases} S_N(f-f_c) + S_N(f+f_c) \ ; & -B \le f \le B \\ 0 & ; \ Otherwise \end{cases}$$

5. The inphase component $n_I(t)$ *and Quadrature* component $n_Q(t)$ have the same varience as narrowband signal n(t)
6. The cross – spectral density of inphase *and Quadrature* component***s*** *of narrowband signal n(t) is purely imaginary.*

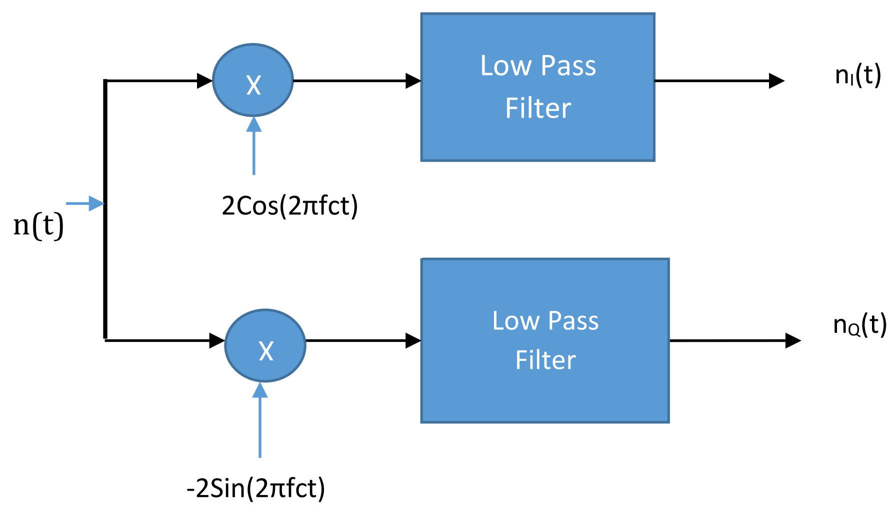
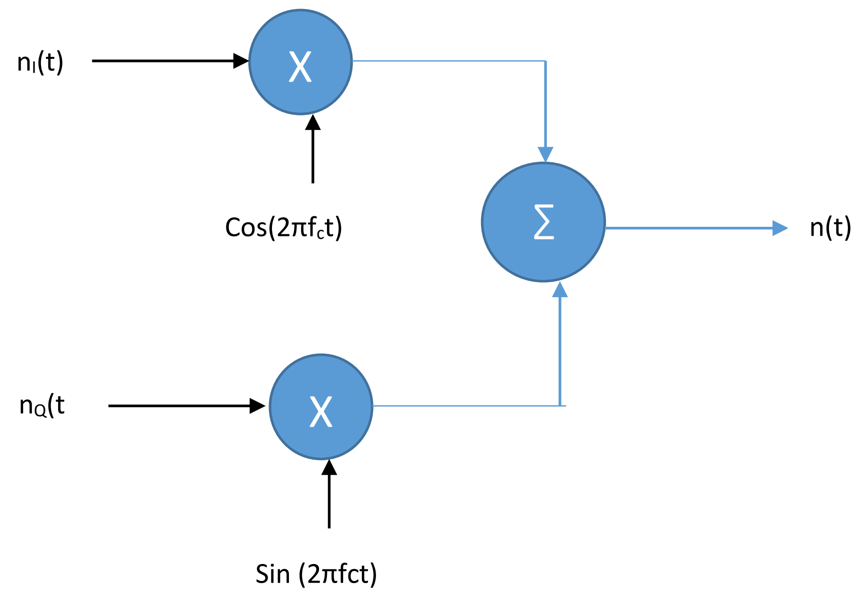


Figure (a) : Extraction of inphase *and Quadrature* component



Figure(b) : *Generation of Narrowband noise from inphase and Quadrature components*

REPRESENTATION OF NARROWBAND NOISE IN TERMS OF ENVELOP & PHASE COMPONENTS

We can represent n(t) in terms of its envelop & phase components as follows

$n(t) = r(t)Cos[\,2\pi f_c t + \psi(t)]$

Where $r(t) = \;[[n_I^2(t) + n_Q^2(t)]^{\frac{1}{2}}$

$$\psi(t) = \; tan^{-1}\left[\frac{n_Q(t)}{n_I(t)}\right]$$

The function r(t) is called the envelop of n(t) and the function $\psi(t)$ *is called Phase of* $n(t)$

The envelope r(t) and phase $\psi(t)$ are both sample functions of low pass random process.

The time interval between two successive peaks of envelope r(t) is approximately $\frac{1}{B}$ where 2B is the bandwidth of narrow band noise n(t).

The probability distributions of r(t) and $\psi(t)$ may be obtained from those of $n_Q(t)$ and $n_I(t)$as follows. Let N_I and N_Q are independent Gaussian Random variables obtained by observing the random processes represented by sample functions $n_I(t)$ and $n_Q(t)$ .The joint probability density function may be

$f_{\text{NI,NQ}}(n_I, n_Q) = \frac{1}{2\pi\sigma^2} exp\left(-\frac{n_I^2 + + n_Q^2}{2\sigma^2}\right)$

Accordingly the probability of joint event that N_I lies between n_I and $n_I + dn_I$ and that N_Q lies between n_Q and $n_Q + dn_Q$ is given by

$$f_{\text{NI,NQ}}(n_I, n_Q) dnI\ dnQ = \frac{1}{2\pi\sigma^2} exp\left(-\frac{n_I^2 + + n_Q^2}{2\sigma^2}\right) dnI\ dnQ$$

Define the transformation

$$nI = r\ cos\psi$$

$$nQ = r\ sin\psi$$

In limiting case ,we may equate the two incremental areas and we can write

$$dnI\ dnQ = r\ dr\ d\psi$$

Now let R and ψ denote the random variables obtained by observing the random processes represented by r(t) and $\psi(t)$ respectively. Then the joint probability can be written as

$$f_{\text{R},\psi}(r, \psi) = \frac{r}{2\pi\sigma^2} exp\left(-\frac{r^2}{2\sigma^2}\right)$$

The PDF is independent of the angle ψ ,which means that the random variables R and ψ *are* statistically independent.

SINE WAVE PLUS NARROWBAND NOISE

If we add the sinusoidal wave Acos(2πfct) to the narrowband noise n(t),where A and fc are constants. We are assuming that the frequency of the sinusoidal wave is the same as the nominal carrier frequency of the noise. A sample function of sinusoidal wave plus noise can be expressed as

$$x(t) = A\cos(\ 2\pi fct) + n(t)$$

Representing the narrowband noise n(t) of its inphase and quadrature phase components

$$x(t) = n_I'(t)\cos(2\pi fct) - n_Q(t)\sin(2\pi fct)$$

Where

$$n_I'(t) = A + n_I(t)$$

We assume that n(t) is Gaussian with zero mean and variance σ^2 ,so

1. Both $n_I'(t)$ *and* $n_Q(t)$ *are Gaussian and statistically independant*

2. The mean of $n'_I(t)$ $is\ A\ and\ that\ of\ n_Q(t)\ is\ zero$
3. $the\ variance\ of$ Both $n'_I(t)\ and\ n_Q(t)\ are$ σ²

   We may therefore express the joint probability density function of Random variables N'$_I$ and N$_Q$, corresponding to $n'_I(t)\ and\ n_Q(t)$as follows

$$f_{\mathrm{N'I,NQ}}(n'_I, n_Q) = \frac{1}{2\pi\sigma^2} exp\left(-\frac{(n'_I - A)^2 + n_Q^2}{2\sigma^2}\right)$$

Let r(t) denote envelope of x(t) and ψ(t) the phase then

$$r(t) = \left\{[n'_I(t)]^2 + n_Q^2(t)\right\}^{\frac{1}{2}}$$

$$\psi(t) = tan^{-1}\left(\frac{n_Q(t)}{n'_I(t)}\right)$$

The joint PDF of Random variables R and ψ corresponding to r(t) and $\psi(t)$

$$f_{\mathrm{R},\psi}(r, \psi) = \frac{r}{2\pi\sigma^2} exp\left(-\frac{r^2 + A^2 - 2Ar\cos\psi}{2\sigma^2}\right)$$

Inorder to get the Probability density function of R we integrate the above equation over all possible values of ψ obtaining the marginal density

$$f_R(r) = \int_0^{2\pi} f_{\mathrm{R},\psi}(r, \psi)\, d\,\psi$$

$$= \frac{r}{2\pi\sigma^2} exp\left(-\frac{r^2+A^2}{2\sigma^2}\right)\int_0^{2\pi} exp\left(\frac{Ar}{\sigma^2}\cos\psi\right) d\psi$$

The RHS of above equation can be identified in terms of defining integral for the modified Bessel function of first order

Io(x) = $\frac{1}{2\pi}\int_0^{2\pi}$ exp($xcos\psi$) $d\psi$

Thus letting x = $\frac{Ar}{\sigma^2}$ $, we\ ay\ write$

$f_R(r)$=$\frac{r}{\sigma^2}\, exp\left(-\frac{r^2+A^2}{2\sigma^2}\right) Io\left(\frac{A_r}{\sigma^2}\right)$

This relation is called *Racian Distribution*

MODULE II
AMPLITUDE MODULATION

2.1: SINUSOIDAL MODULATION

The process of changing some characteristic (e.g. amplitude, frequency or phase) of a carrier wave in accordance with the intensity of the signal is known as **modulation**.

Modulation means to "change". In modulation, some characteristic of carrier wave is changed inaccordance with the intensity (*i.e*. amplitude) of the signal. The resultant wave is called modulatedwave or radio wave and contains the audio signal. Therefore, modulation permits the transmission tooccur at high frequency while it simultaneously allows the carrying of the audio signal.
When a sinusoidal carrier signal is modulated by a sinusoidal modulating signal, the process is called **sinusoidal modulation**

Types of Modulation

As you will recall, modulation is the process of changing amplitude or frequency or phase of a carrier
wave in accordance with the intensity of the signal. Accordingly, there are three basic types of modulation,
namely ;

(i) amplitude modulation ***(ii)*** frequency modulation ***(iii)*** phase modulation

2.2 AMPLITUDE MODULATION

When the amplitude of high frequency carrier wave is changed in accordance with the intensity of the signal, it is called **amplitude modulation**.
In amplitude modulation, only the amplitude of the carrier wave is changed in accordance with the intensity of the signal. However, the frequency of the modulated wave remains the same *i.e*. carrier frequency

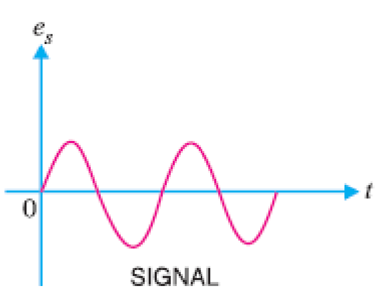


Fig: 2.1 Signal

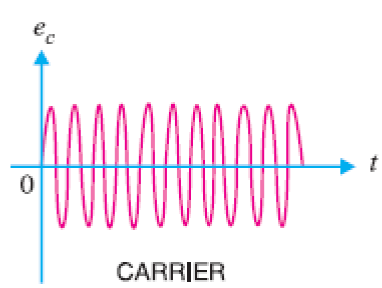


Fig: 2.2 Carrier

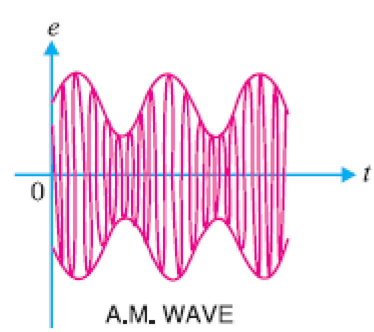


Fig:2.3 A M Wave

Note that the amplitudes of both positive and negative half-cycles of carrier wave are changed in accordance with the signal. For instance, when the signal is increasing in the positive sense, the amplitude of carrier wave also increases. On the other hand, during negative half-cycle of the signal, the amplitude of carrier wave decreases. Amplitude modulation is done by an electronic circuit called *modulator.*

The following points are worth noting in amplitude modulation :

(*i*) The amplitude of the carrier wave changes according to the intensity of the signal.

(*ii*). The amplitude variations of the carrier wave is at the signal frequency *fs*.

(iii). The frequency of the amplitude modulated wave remains the same *i.e*. carrier frequency *fc*.

2.3 MODULATION INDEX

An important consideration in amplitude modulation is to describe the depth of modulation *i.e*. the extent to which the amplitude of carrier wave is changed by the signal. This is described by a factor called modulation factor which may be defined as under :

The ratio of change of amplitude of carrier wave to the amplitude of normal carrier wave is called the **modulation factor** *m i.e*.

Modulation factor, $m = \frac{\text{Amplitude change of carrier wave}}{\text{Normal carrier amplitude(unmodulated)}}$

The value of modulation factor depends upon the amplitudes of carrier and signal. Figure shows amplitude modulation for different values of modulation factor *m*.

(i). When signal amplitude is zero, the carrier wave is not modulated as shown in Fig. 2.4 (*i*). The amplitude of carrier wave remains unchanged.

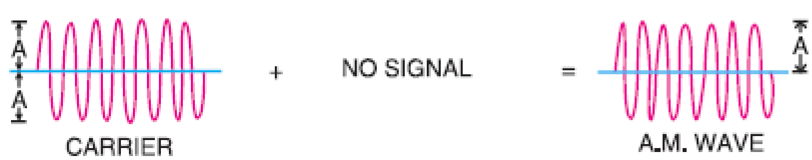


Fig: 2.4 signal with 0% Modulation

Amplitude change of carrier = 0

Amplitude of normal carrier = A

Modulation factor, *m* = 0/A = 0 or 0%

(ii) When signal amplitude is equal to the carrier amplitude as shown the amplitude of carrier varies between 2A and zero.

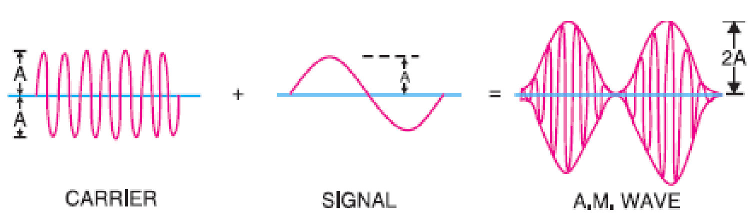


Fig: 2.5 signal with 100% Modulation

Amplitude change of carrier = 2 A -A = A

$m = \frac{\text{Amplitude change of carrier wave}}{\text{Normal carrier amplitude}}$ = A /A = 1 or 100 % In this case, the carrier is said to be 100% modulated.

(*iii*) When the signal amplitude is one-half the carrier amplitude as shown fig the amplitude of carrier wave varies between 1.5 A and 0.5 A.
Amplitude change of carrier = 1.5 A - A = 0.5 A
Modulation factor, m = 0.5 A/A = 0.5 or 50 %
In this case, the carrier is said to be 50% modulated.

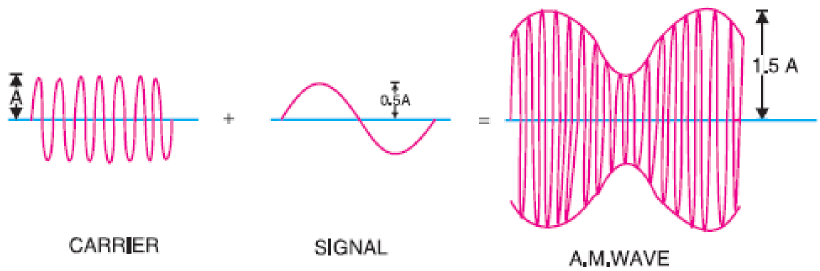


Fig: 2.6 signal with 50% Modulation

(*iv*) When the signal amplitude is 1.5 times the carrier amplitude as shown in Fi , the maximum value of carrier wave becomes 2.5 A.
Amplitude change of carrier wave = 2.5 A -A = 1.5 A
Modulation factor, m = 1.5 A/A =*1.5* or 150 %
In this case, the carrier is said to be 150% modulated *i.e*. over-modulated.

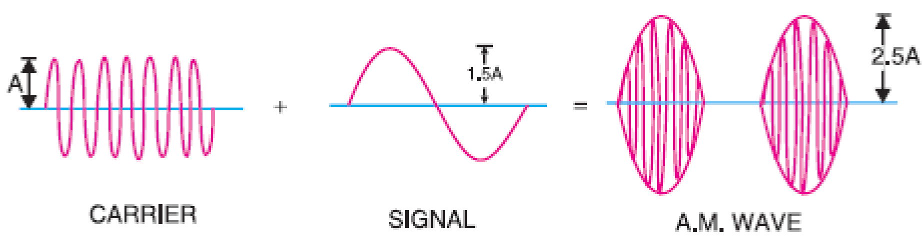


Fig: 2.7 signal with 150% Modulation(over-modulated.)

Importance of modulation Index/factor. Modulation factor is very important since it determines the strength and quality of the transmitted signal. In an AM wave, the signal is contained in the variations of the carrier amplitude. When the carrier is modulated to a small degree (*i.e*. small m), the amount of carrier amplitude variation is small. Consequently, the audio signal being transmitted will not be very strong. The greater the degree of modulation (*i.e. m*), the stronger and clearer will be the audio signal. It may be emphasised here that if the carrier is overmodulated (*i.e.* $m > 1$), distortion will occur during reception. This condition is shown in Fig. (*2.7*). The AM waveform is clipped and the envelope is discontinuous. Therefore, degree of modulation should never exceed 100%.

Example 1 *If the maximum and minimum voltage of an AM wave are Vmax and Vmin respectively, then show that modulation factor m is given by :*

$$m = \frac{V_{max} - V_{min}}{V_{Max} + V_{min}}$$

Soln:

Fig 2.8 shows the waveform of amplitude modulated wave.

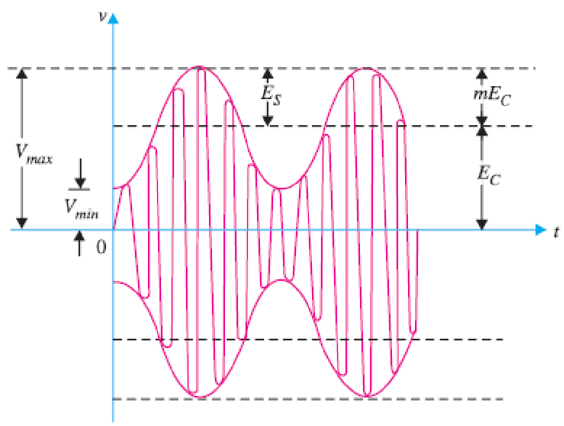


Fig: 2.8 Amplitude Modulated Wave

Let the amplitude of the normal carrier wave be *EC*.Then

$$Ec = \frac{V_{max} + V_{min}}{2}$$

If *ES* is the signal amplitude, then it is clear from Fig that :

$$Es = \frac{V_{max} - V_{min}}{2}$$

$$ES = m\,EC$$

$$\frac{V_{max} - V_{min}}{2} = m\,\frac{V_{max} + V_{min}}{2}$$

Or

$$m = \frac{V_{max} - V_{min}}{V_{Max} + V_{min}}$$

***Example 2**: The maximum peak-to-peak voltage of an AM wave is 16 mV and the minimum peak-to-peak voltage is 4 mV. Calculate the modulation factor.*

Solution. Fig.2.9 shows the conditions of the problem.
Maximum voltage of AM wave is

Vmax = 16mV/2 = 8mV

Minimum voltage of AM wave is

Vmin = 4mV/2 = 2mV

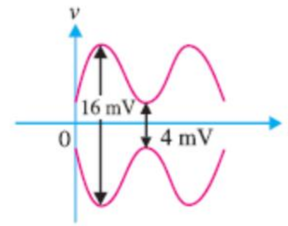


Fig: 2.9

$m = \frac{V_{max}-V_{min}}{V_{Max}+V_{min}}$= 6mV/10mV = 0.6 =60%

***Example 3** : A carrier of 100V and 1200 kHz is modulated by a 50 V, 1000 Hz sine wave signal. Find the modulation factor.*

Solution.
Modulation factor, *m* = 50V/100V = 0.5

2.4 ANALYSIS OF AMPLITUDE MODULATED WAVE/ EFFECTIVE VOLTAGES IN AM WAVE

A carrier wave may be represented by :

$ec = EC \cos \omega c\, t$

where *ec* = instantaneous voltage of carrier
EC = amplitude of carrier
ω*c* = 2π*fc*
= angular velocity at carrier frequency *fc*

In amplitude modulation, the amplitude *EC* of the carrier wave is varied in accordance with the intensity of the signal as shown in Fig. 16.6. Suppose the modulation factor is *m*. It means that signal
produces a maximum change of *m EC* in the carrier amplitude. Obviously, the amplitude of signal is
m EC. Therefore, the signal can be represented by :

$es = m\, EC \cos \omega s\, t$

where *es* = instantaneous voltage of signal
m EC = amplitude of signal
ω *s* = 2π *fs* = angular velocity at signal frequency *fs*

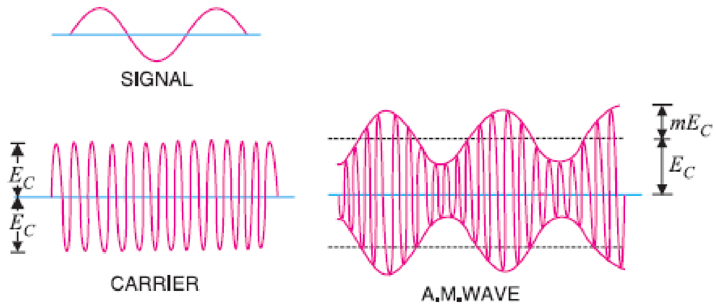


Fig: 2.10

The amplitude of the carrier wave varies at signal frequency *fs*. Therefore, the amplitude of AM wave is given by :
Amplitude of AM wave = $EC + m\,EC \cos \omega s\, t = EC\,(1 + m \cos \omega s\, t)$

The instantaneous voltage of AM wave is :

$$e = \text{Amplitude} \times \cos \omega c\, t$$

$$= EC\,(1 + m \cos \omega s\, t) \cos \omega c\, t$$

$$= EC \cos \omega c\, t + m\,EC \cos \omega s\, t \cos \omega c\, t$$

$$= Ec \cos \omega ct + \frac{mEc}{2}[2 \cos \omega st \cos \omega ct]$$

$$= Ec \cos \omega ct + \frac{mEc}{2}[\cos(\omega c + \omega s)t + \cos(\omega c - \omega s)t]$$

$$= Ec \cos \omega ct + \frac{mEc}{2}\cos(\omega c + \omega s)t + \frac{mEc}{2}\cos(\omega c - \omega s)t$$

The following points may be noted from the above equation of amplitude modulated wave:
(*i*) The AM wave is equivalent to the summation of three sinusoidal waves; one having amplitude *EC* and frequency ***fc*, the second having amplitude $mEC/2$ and frequency (*fc* + *fs*) and the third having amplitude $mEC/2$ and frequency *fc* - *fs*.
(*ii*) The AM wave contains three frequencies *viz fc*, *fc* + *fs* and *fc* -*fs*. The first frequency is the carrier frequency. Thus, the process of modulation does not change the original carrier frequency but produces two new frequencies (*fc* + *fs*) and (*fc* - *fs*) which are called sideband frequencies.
(*iii*) The sum of carrier frequency and signal frequency *i.e.* (*fc* + *fs*) is called *upper sideband frequency*. The *lower sideband frequency* is *fc* - *fs i.e.* the difference between carrier and signal frequencies.

2.4.1. SIDEBAND FREQUENCIES IN AM WAVE

In an amplitude modulated wave, the sideband frequencies are of our interest. It is because the signal frequency *fs* is contained in the sideband frequencies. Fig. shows the frequency spectrum of an amplitude modulated wave. The frequency components in the AM wave are shown by vertical lines.

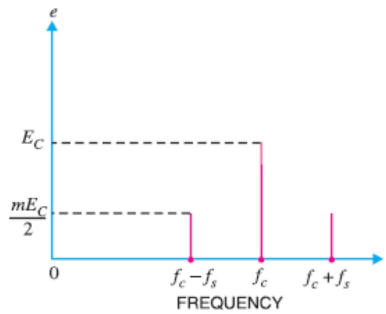


Fig: 2.11

The height of each vertical line is equal to the amplitude of the components present. It may be added here that in practical radio transmission, carrier frequency *fc* is many times greater than signal frequency *fs*. Hence, the sideband frequencies are generally close to the carrier frequency. It may be seen that a carrier modulated by a single frequency is equivalent to three simultaneous signals; the carrier itself and two other steady frequencies *i.e. fc* + *fs* and *fc* -*fs*.

Let us illustrate sideband frequencies with an example. Suppose the carrier frequency is 400 kHz and the signal frequency is 1 kHz. The AM wave will contain three frequencies *viz* 400 kHz, 401 kHz and 399 kHz. It is clear that upper sideband frequency (401 kHz) and lower sideband frequency (399 kHz) are very close to the carrier frequency (400 kHz).

Bandwidth. In an AM wave, the bandwidth is from (*fc* − *fs*) to (*fc* + *fs*) *i.e.*, 2 *fs*. Thus in the above example, bandwidth is from 399 to 401 kHz or 2 kHz which is twice the signal frequency. Therefore, we arrive at a very important conclusion that *in amplitude modulation, bandwidth is twice the signal frequency*. The tuned amplifier which is called upon to amplify the modulated wave must have the required bandwidth to include the sideband frequencies. If the tuned amplifier has insufficient bandwidth, the upper sideband frequencies may not be reproduced by the radio receiver.

Example 4. *A 2500 kHz carrier is modulated by audio signal with frequency span of 50 − 15000 Hz. What are the frequencies of lower and upper sidebands ? What bandwidth of RF amplifier is required to handle the output ?*

Solution. The modulating signal (*e.g.* music) has a range of 0.05 to 15 kHz. The sideband frequencies produced range from *fc* ± 0.05 kHz to *fc* ± 15 kHz. Therefore, upper sideband ranges from 2500.05 to 2515 kHz and lower sideband ranges from 2499.95 to 2485 kHz.

The sideband frequencies produced can be approximately expressed as 2500 ± 15 kHz. Therefore, bandwidth requirement = 2515 − 2485 = 30 kHz. Note that bandwidth of *RF* amplifier required is twice the frequency of highest modulating signal frequency.

2.5 POWER IN AM WAVE

The power dissipated in any circuit is a function of the square of voltage across the circuit and the effective resistance of the circuit. Equation of AM wave reveals that it has three components of amplitude EC , $m\ EC\ /2$ and $m\ EC\ /2$. Clearly, power output must be distributed among these components

carrier power $Pc = \frac{\left(\frac{Ec}{\sqrt{2}}\right)^2}{R} = \frac{(Ec)^2}{2R}$

Total power of side bands $Ps = \frac{\left(\frac{mEc}{2\sqrt{2}}\right)^2}{R} + \frac{\left(\frac{mEc}{2\sqrt{2}}\right)^2}{R} = \frac{m^2Ec^2}{8R} + \frac{m^2Ec^2}{8R}$

$$= \frac{m^2Ec^2}{4R}$$

Total power Pt = Pc + Ps

$$= \frac{(Ec)^2}{2R} + \frac{m^2Ec^2}{4R}$$

$$= \frac{(Ec)^2}{2R}\left\{1 + \frac{m^2}{2}\right\}$$

$$P_T = Pc\left\{1 + \frac{m^2}{2}\right\}$$

$I_T{}^2 = I_C{}^2\left\{1 + \frac{m^2}{2}\right\}$ ....since $P=I^2R$

2.6 AMPLITUDE MODULATORS

2.6.1 Transistor Am *Modulator

Fig. 16.8 shows the circuit of a simple AM modulator. It is essentially a *CE* amplifier having a voltage gain of A. The carrier signal is the input to the amplifier. The modulating signal is applied in the emitter resistance circuit.

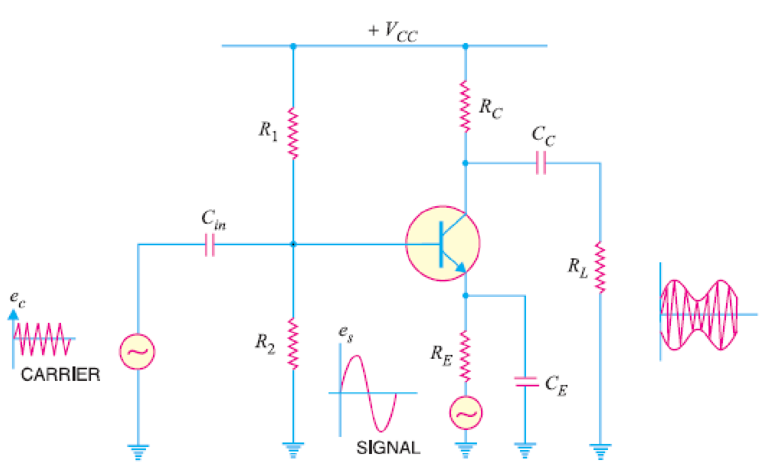


Fig: Transistorised Amplitude Modulator

Working. The carrier *ec* is applied at the input of the amplifier and the modulating signal *es* is applied in the emitter resistance circuit. The amplifier circuit amplifies the carrier by a factor "*A*" so that the output is *Aec*. Since the modulating signal is a part of the biasing circuit, it produces lowfrequency variations in the emitter circuit. This in turn causes *variations in "*A*". The result is that amplitude of the carrier varies in accordance with the strength of the signal. Consequently, amplitude modulated output is obtained across *RL*. It may be noted that carrier should not influence the voltage gain A; only the modulating signal should do this. To achieve this objective, carrier should have a small magnitude and signal should have a large magnitude.

2.6.2 Square law modulator

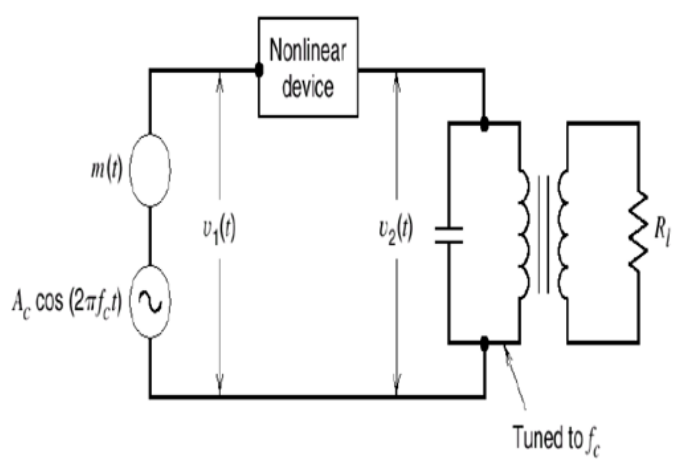
When the output of a device is not directly proportional to input throughout the operation, the device is said to be non-linear. The Input-Output relation of a non-linear device can be expressed as

$$Vo = ao + a_1 Vin + a_2 V_{in}^2 + a_3 V_{in}^3 + a_4 V_{in}^4 + \cdots$$

When the input is very small ,the higher power terms can be neglected.Hence the output is approximately given by

$$Vo = ao + a_1 Vin + a_2 V_{in}^2$$

When the output is considered up to square of the in put, the device is called a square law device and the square law modulator is as shown in the figure.



Concider a non linear device to which a carrier $C(t) = Ac\ Cos\ (2\pi f ct)$ and an information signal m(t) are fed simultaneously as shown in figure 2.7. The total input to the device at any instant is

$Vin = C(t) + m(t) = Ac\ Cos\ (2\pi fct) + m(t)$

As the level of input is very small the output can be considered up to a square of input

ie.

$$Vo = ao + a_1 Vin + a_2 V_{in}^2$$

$$Vo = ao + a_1\{Ac\ Cos\ (2\pi fct) + m(t)\} + a_2\{Ac \cos 2\pi fct + m(t)\}^2$$

$= ao + a_1 Ac\ Cos\ (2\pi fct) + a_1 m(t) + \frac{a_2 A_c^2}{2}(1 + \cos 4\pi fct) + a_2\{m(t)\}^2 + 2a_2 m(t) Ac\ Cos\ (2\pi fct)$

$= ao + a_1 Ac\ Cos\ (2\pi fct) + a_1 m(t) + \frac{a_2 A_c^2}{2}\cos 4\pi fct) + a_2 m^2\ (t) + 2a_2 m(t) Ac\ Cos\ (2\pi fct)$

Taking fourier Transform on both sides, we get

$$Vo(f) = \left(ao + \frac{a_2 A_c^2}{2}\right)\delta(t) + \frac{a_1 A_C}{2}[\delta(f - fc) + \delta(f + fc)] + a_1 M(f) + \frac{a_2 A_c^2}{2}[\delta(f - 2fc) + \delta(f + 2fc)] + a_2 M(f) + a_2 A_C[M(f - fc) + M(f + fc)]$$

Therefore the square law device output of 0V consists of
DC component at f = 0
Information signal ranging from 0 to W Hz and its second harmonics

Signal at fc and 2fc.

Frequency band centered at fc with a deviation of ±W, Hz

The required AM signal with a carrier frequency fc can be separated using a BPF ot the output of square law device .The filter should have l lower cutoff frequency between (fo+W) and 2fc

Therefore the filter output is

$$S(t) = a_1 Ac\ Cos\ (2\pi fct) + 2a_2 Ac\ m(t)\ Cos\ (2\pi fct)$$

$$= a_1 Ac\ \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos 2\pi fct$$

If m(t) = $Am\ Cos\ (2\pi fmt)$,we get

$$S(t) = a_1 Ac\ \left[1 + \frac{2a_2}{a_1} Am\ Cos\ (2\pi fmt)\right] \cos 2\pi fct$$

Comparing this with the standard representation of AM signal

$$S(t) = a_1 Ac\,[1 + Ka\,m(t)]\cos 2\pi fct$$

Therefore the modulation index

$$m = \frac{2a_2}{a_1} Am$$

The output AM signal is free from distortion and attenuation only when (fc-W) >2W or fc >3W

2.6.3 SWITCHING MODULATOR

Consider a semiconductor diode used as an ideal switch to which the carrier signal

$C(t) = \ Ac\ Cos\ (2\pi fct)$ and an information signal m(t) are applied simultaneously.

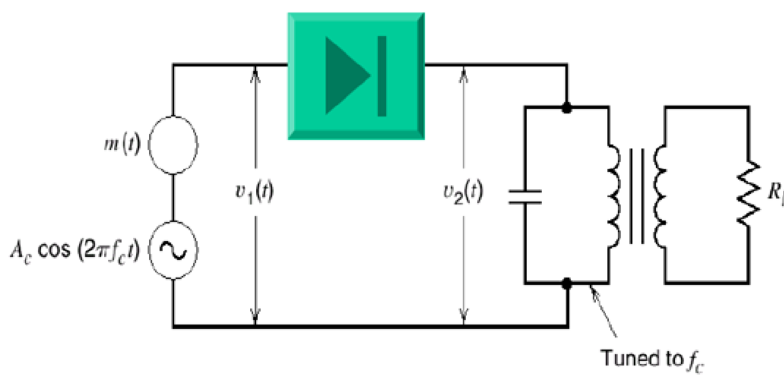


Fig : Switching modulator

The output for the diode at any instant is given by

$$V1 = C(t) + m(t)$$

$$V1 = Ac\ Cos\ (2\pi fct) + m(t)$$

When the peak amplitude of c(t) is maintained more than that of information signal, the operation is assumed to be dependent on only c(t) irrespective of m(t). When c(t) is positive, v2=v1since the diode is forward biased. Similarly, when c(t) is negative, v2=0 since diode is reverse biased. Based upon above operation, switching response of the diode is periodic rectangular wave with an amplitude unity and is given by

$$p(t) = \frac{1}{2} + \frac{1}{\pi}\sum_{n=\infty}^{\infty}\frac{(-1)^{n-1}}{2n-1}\ Cos\ [2\pi fct\{2n-1\}]$$

$$p(t) = \frac{1}{2} + \frac{2}{\pi}\ Cos\ (2\pi fct) - \frac{2}{3\pi}\cos(6\pi fct) + \cdots \dots\dots\dots\dots\dots..$$

Therefore the diodes response Vo is a product of switching response p(t) and input V1

$$V_2 = V_1 \times p(t)$$

$$V_2 = \left[Ac\ Cos\ (2\pi fct) + m(t)\left[\frac{1}{2} + \frac{2}{\pi}\ Cos\ (2\pi fct) - \frac{2}{3\pi}\cos(6\pi fct) + \cdots \dots\dots\dots\dots\dots..\right]\right]$$

Applying the Fourier Transform, we get

$$V_2(f) = \frac{Ac}{4}[\delta(f - f_c) + \delta(f + f_c)] + \frac{M(f)}{2} + \frac{Ac}{\pi}\delta(f) + \frac{Ac}{2\pi}[\delta(f - 2f_c) + \delta(f + 2f_c)]$$

$$+\frac{1}{\pi}M(f - fc) + M(f + fc) - \frac{Ac}{6\pi}[\delta(f - 4f_c) + \delta(f + 4f_c)] - \frac{Ac}{3\pi}[\delta(f - 2f_c) + \delta(f + 2f_c)]$$

$$-\frac{1}{3\pi}M(f - 3fc) + M(f + 3fc)$$

The diodes output V2 consists of

A DC component at f=0

Information signal ranging from 0 to W Hz and infinite number of frequency bands centred at *f, 2fc, 3fc, 4fc,……………*

The required AM signal centred at fc can be separated using BPF . The lower cutoff frequency for BPF should be between W and (fc –W) and upper cutoff frequency between(fc+ W) and 2fc. The filter output is given by the equation

$$S(t) = \frac{Ac}{2}\left[1 + \frac{4}{\pi}\frac{m(t)}{Ac}\right]\cos 2\pi fct$$

Therefore modulation index, $m = \frac{4}{\pi}\frac{Am}{Ac}$

The output AM signal if free from distortions and attenuations only when *fc-W > W*

Or *fc>2W*

Limitations Of Amplitude Modulation

Although theoretically highly effective, amplitude modulation suffers from the following drawbacks:

(*i*) *Noisy reception.* In an AM wave, the signal is in the amplitude variations of the carrier. Practically all the natural and man made noises consist of electrical amplitude disturbances. As a radio receiver cannot distinguish between amplitude variations that represent noise and those that contain the desired signal, therefore, reception is generally noisy.

(*ii*) *Low efficiency.* In amplitude modulation, useful power is in the sidebands as they contain the signal. As discussed before, an AM wave has low sideband power. For example, if modulation is 100%, the sideband power is only one-third of the total power of AM wave. Hence the efficiency of this type of modulation is low.

(*iii*) *Small operating range.* Due to low efficiency of amplitude modulation, transmitters employing this method have a small operating range *i.e*. messages cannot be transmitted over larger distances.

(*iv*) *Lack of audio quality.* This is a distinct disadvantage of amplitude modulation. In order to attain high-fidelity reception, all audio frequencies up to 15 kHz must be reproduced. This necessitates bandwidth of 30 kHz since both sidebands must be reproduced. But AM broadcasting stations are assigned bandwidth of only 10 kHz to minimise the interference from adjacent broadcasting stations. This means that the highest modulating frequency can be 5 kHz which is hardly sufficient to reproduce the music properly.

Example 5 . *A carrier wave of 500 watts is subjected to 100% amplitude modulation. Determine : (i) power in sidebands (ii) power of modulated wave*

Solution.

(i) Sideband power, $PS = \frac{1}{2}m^2P_c$ = ½ x 500 = 250W

Thus there are 125 W in upper sideband and 125 W in lower sideband.

(ii) Power of AM wave, $PT = PC + PS$ = 500 + 250 = **750 W**.

Example 6. *A 50 kW carrier is to be modulated to a level of (i) 80% (ii) 10%. What is the total sideband power in each case ?*

Solution

(i) $PS = \frac{1}{2}m^2P_c = \frac{1}{2}(0.8)^2 \times 50 = 16\ kW$

(ii) $PS = \frac{1}{2}m^2P_c = \frac{1}{2}(0.1)^2 \times 50 = 0.25\ kW$

Note the effect of modulation factor on the magnitude of sideband power. In the first case (*m* = 80%), we generated and transmitted 50 kW carrier in order to send 16 kW of intelligence. In the second case (*m* = 10%), the same carrier level — 50 kW — is used to send merely 250 W of intelligence. Clearly, the efficiency of operation decreases rapidly as modulation factor decreases. For this reason, in amplitude modulation, the value of *m* is kept as close to unity as possible.

Example 7. *A 40kW carrier is to be modulated to a level of 100%.*

(i) What is the carrier power after modulation ?

(ii) How much audio power is required if the efficiency of the modulated RF amplifier is 72% ?

Solution. Fig. shows the block diagram indicating the power relations.

(*i*) Since the carrier itself is unaffected by the modulating signal, there is no change in the carrier power level.

PC = **40 kW**

$PS = \frac{1}{2}m^2P_c = \frac{1}{2}\ (1)^2 \times 40 = 20\text{kW}$

Paudio = $P_{aodio} = \frac{P_S}{0.72}$ = 27.8kW

Example 8 . *An audio signal of 1 kHz is used to modulate a carrier of 500 kHz. Determine (i) sideband frequencies (ii) bandwidth required.*

Solution. Carrier frequency, *fc* = 500 kHz

Signal frequency, *fs* = 1 kHz

(*i*) As discussed in Art. 16.6, the *AM* wave has sideband frequencies of (*fc* + *fs*) and (*fc* -*fs*).

Sideband frequencies = (500 + 1) kHz and (500 - 1) kHz

= **501 kHz** and **499 kHz**

(iii) Bandwidth required = 499 kHz to 501 kHz = **2 kHz**

Example 9. *The load current in the transmitting antenna of an unmodulated AM transmitter is 8A. What will be the antenna current when modulation is 40% ?*

$I_T = I_C\sqrt{1+\frac{m^2}{2}} = 8\sqrt{1+\frac{0.4^2}{2}} = 8.31\text{ A}$

Example 10 . *The antenna current of an AM transmitter is 8A when only carrier is sent but it increases to 8.93A when the carrier is sinusoidally modulated. Find the % age modulation.*

Ans: m= 70.1%

Example 11. *The r.m.s. value of carrier voltage is 100 V. After amplitude modulation by a sinusoidal a.f. voltage, the r.m.s. value becomes 110 V. Calculate the modulation index.*

$$\frac{P_T}{Pc} = \left\{1+\frac{m^2}{2}\right\}$$

$$\left\{\frac{V_T}{Vc}\right\}^2 = \left\{1+\frac{m^2}{2}\right\}$$

$$\left\{\frac{110}{100}\right\}^2 = \left\{1+\frac{m^2}{2}\right\}$$

$$1.21 = \left\{1+\frac{m^2}{2}\right\}$$

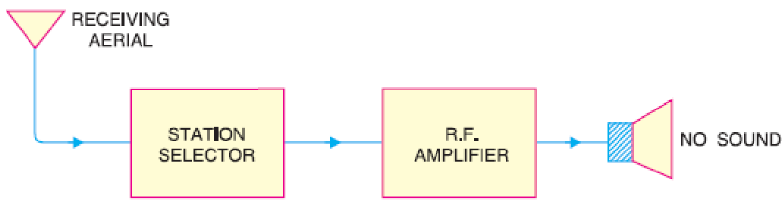
$$\frac{m^2}{2} = 0.21$$

∴ m=0.648

2.7 DEMODULATION

The process of recovering the audio signal from the modulated wave is known as **demodulation** *or* **detection.** At the broadcasting station, modulation is done to transmit the audio signal over larger distances to a receiver. When the modulated wave is picked up by the radio receiver, it is necessary to recover

the audio signal from it. This process is accomplished in the radio receiver and is called demodulation.
Necessity of demodulation. It was noted previously that amplitude modulated wave consists of carrier and sideband frequencies. The audio signal is contained in the sideband frequencies which are radio frequencies. If the modulated wave after amplification is directly fed to the speaker as shown in Fig, no sound will be heard. It is because diaphragm of the speaker is not at all able to respond to such high frequencies. Before the diaphragm is able to move in one direction, the rapid reversal of current tends to move it in the opposite direction i.e. diaphragm will not move at all. Consequently, no sound will be heard.

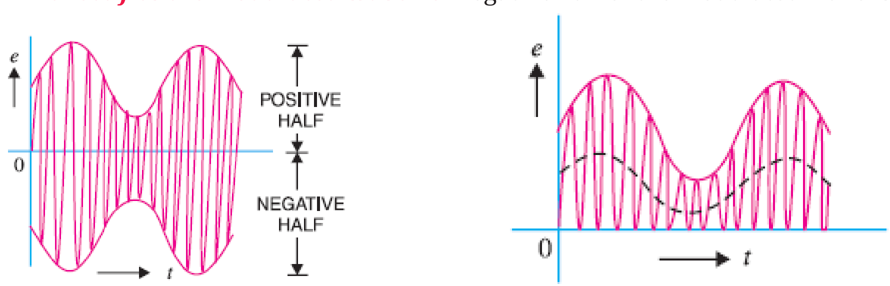


From the above discussion, it follows that audio signal must be separated from the carrier at a suitable stage in the receiver. The recovered audio signal is then amplified and fed to the speaker for conversion into sound.

2.7.1 Essentials In Demodulation

In order that a modulated wave is audible, it is necessary to change the nature of modulated wave. This is accomplished by a circuit called *detector.* A detector circuit performs the following two functions :

(i) ***It rectifies the modulated wave i.e.*** negative half of the modulated wave is eliminated.



(*i*) (*ii*)

As shown in Fig. (*i*), a modulated wave has positive and negative halves exactly equal. Therefore, average current is zero and speaker cannot respond. If the negative half of this modulated wave is eliminated as shown in Fig. (*ii*), the average value of this wave will not be zero since the resultant pulses are now all in one direction. The average value is shown by the dotted line in Fig (*ii*). Therefore, the diaphragm will have definite displacement corresponding to the average value of the wave. It may be seen that shape of the average wave is similar to that of the modulation envelope. As the signal is of the same shape as the envelope, therefore, average wave shape is of the same form as the signal.

(ii) It separates the audio signal from the carrier.

The rectified modulated wave contains the audio signal and the carrier. It is desired to recover the audio signal. This is achieved by a filter circuit which removes the carrier frequency and allows the audio signal to reach theload *i.e.*speaker.

2.7.2 A.M. Diode Detector
2.7.2 .1 Envelop detector

It is a simple and highly effective system. This method is used in most of the commercial AM radio receivers. An envelop detector is as shown below.

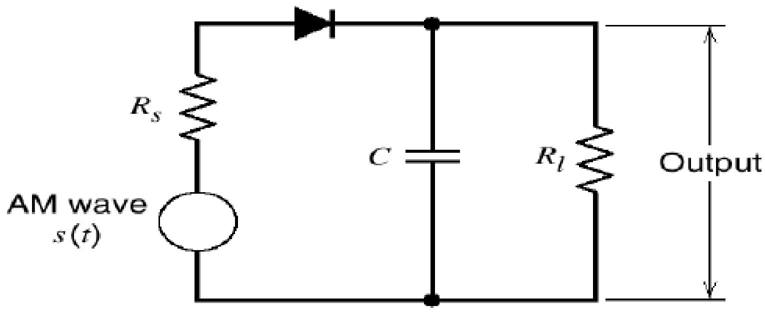


Figure: Envelope detector

During the positive half cycles of the input signals, the diode D is forward biased and the capacitor C charges up rapidly to the peak of the input signal. When the input signal falls below this value, the diode becomes reverse biased and the capacitor C discharges through the load resistor RL.
The discharge process continues until the next positive half cycle. When the input signal becomes greater than the voltage across the capacitor, the diode conducts again and the process is repeated.
The charge time constant (rf+Rs)C must be short compared with the carrier period, the capacitor charges rapidly and there by follows the applied voltage up to the positive peak when the diode is conducting.
That is the charging time constant shall satisfy the condition,

$$\left(r_f + R_s\right) \ll \frac{1}{f}$$

On the other hand the discharging time constant $R_L C$ must be long enough to ensure that the capacitor discharges slowly through the load resistor R_L between positive peaks of the carrier wave ,but not so long that the capacitor voltage will not discharge the maximum rate of change of modulating wave.

That discharge time constant shall satisfy the condition $\frac{1}{fc} \ll R_L C \ll \frac{1}{W}$

where 'W' is band width of the message signal.
The result is that the capacitor voltage or detector output is nearly the same as the envelope of AM wave.

Advantages of AM: Generation and demodulation of AM wave are easy. AM systems are cost effective and easy to build.
Disadvantages: AM contains unwanted carrier component, hence it requires more transmission power. The transmission bandwidth is equal to twice the message bandwidth.

2.8 AM TRANSMITTER

Transmitters that transmit AM signals are known as AM transmitters. These transmitters are used in medium wave (MW) and short wave (SW) frequency bands for AM broadcast. The MW band has frequencies between 550 KHz and 1650 KHz, and the SW band has frequencies ranging from 3 MHz to 30 MHz. The two types of AM transmitters that are used based on their transmitting powers are:

- High Level
- Low Level

High level transmitters use high level modulation, and low level transmitters use low level modulation. The choice between the two modulation schemes depends on the transmitting power of the AM transmitter. In broadcast transmitters, where the transmitting power may be of the order of kilowatts, high level modulation is employed. In low power transmitters, where only a few watts of transmitting power are required , low level modulation is used.

2.8.1High-Level and Low-Level Transmitters

Below figures show the block diagram of high-level and low-level transmitters. The basic difference between the two transmitters is the power amplification of the carrier and modulating signals.

Figure (a) shows the block diagram of high-level AM transmitter.

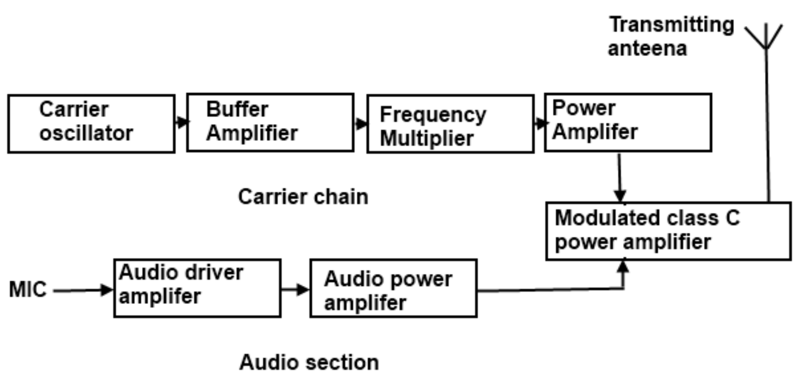


Figure (a) Block diagram of high level AM transmitter

In high-level transmission, the powers of the carrier and modulating signals are amplified before applying them to the modulator stage, as shown in figure (a). In low-level modulation, the powers of the two input signals of the modulator stage are not amplified. The required transmitting power is obtained from the last stage of the transmitter, the class C power amplifier.

The various sections of the figure (a) are:

- Carrier oscillator
- Buffer amplifier
- Frequency multiplier
- Power amplifier
- Audio chain
- Modulated class C power amplifier

Carrier oscillator

The carrier oscillator generates the carrier signal, which lies in the RF range. The frequency of the carrier is always very high. Because it is very difficult to generate high frequencies with good frequency stability, the carrier oscillator generates a sub multiple with the required carrier frequency. This sub multiple frequency is multiplied by the frequency multiplier stage to get the required carrier frequency. Further, a crystal oscillator can be used in this stage to generate a low frequency carrier with the best frequency stability. The frequency multiplier stage then increases the frequency of the carrier to its required value.

Buffer Amplifier

The purpose of the buffer amplifier is twofold. It first matches the output impedance of the carrier oscillator with the input impedance of the frequency multiplier, the next stage of the carrier oscillator. It then isolates the carrier oscillator and frequency multiplier.

This is required so that the multiplier does not draw a large current from the carrier oscillator. If this occurs, the frequency of the carrier oscillator will not remain stable.

Frequency Multiplier

The sub-multiple frequency of the carrier signal, generated by the carrier oscillator , is now applied to the frequency multiplier through the buffer amplifier. This stage is also known as harmonic generator. The frequency multiplier generates higher harmonics of carrier oscillator frequency. The frequency multiplier is a tuned circuit that can be tuned to the requisite carrier frequency that is to be transmitted

Power Amplifier

The power of the carrier signal is then amplified in the power amplifier stage. This is the basic requirement of a high-level transmitter. A class C power amplifier gives high power current pulses of the carrier signal at its output.

Audio Chain

The audio signal to be transmitted is obtained from the microphone, as shown in figure (a). The audio driver amplifier amplifies the voltage of this signal. This amplification is necessary to drive the audio power amplifier. Next, a class A or a class B power amplifier amplifies the power of the audio signal.

Modulated Class C Amplifier

This is the output stage of the transmitter. The modulating audio signal and the carrier signal, after power amplification, are applied to this modulating stage. The modulation takes place at this stage. The class C amplifier also amplifies the power of the AM signal to the reacquired transmitting power. This signal is finally passed to the antenna., which radiates the signal into space of transmission.

2.8.2 Low-Level Am Transmitter.

Figure shows the block diagram of a low-level AM transmitter.

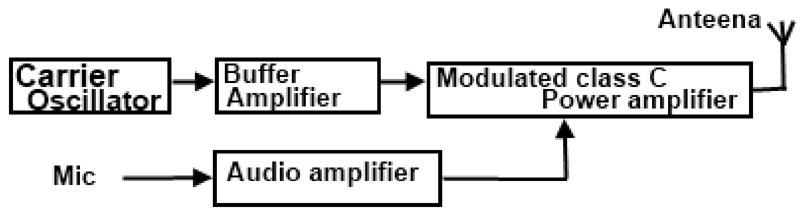


Figure (b) Block diagram of Low-level AM transmitter

The low-level AM transmitter shown in the figure (b) is similar to a high-level transmitter, except that the powers of the carrier and audio signals are not amplified. These two signals are directly applied to the modulated class C power amplifier.

Modulation takes place at the stage, and the power of the modulated signal is amplified to the required transmitting power level. The transmitting antenna then transmits the signal.

2.9 NONSINUSOIDAL MODULATION

When a sinusoidal carrier signal is modulated by a non-sinusoidal modulating signal, the process is called Non-sinusoidal modulation. Assume that a carrier is amplitude-modulated by a square wave which is made up of a fundamental sine wave and all odd harmonics. A modulating square wave will produce sidebands at frequencies based upon the fundamental sine wave as well as at the third, fifth, seventh, etc., harmonics, resulting in a frequency-domain plot like that shown in Fig.

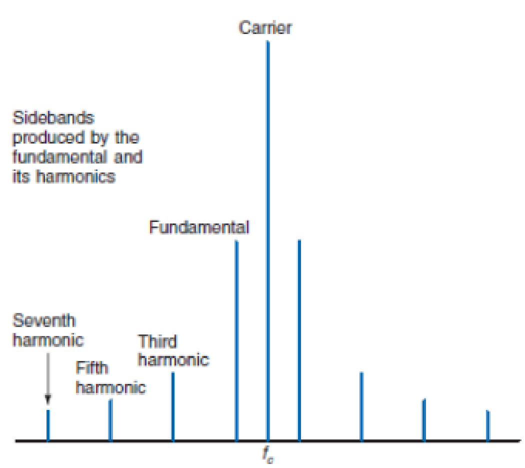


Fig: Frequency Spectrum of AM signal modulated by a square wave

Amplitude modulation by square waves or rectangular binary pulses is referred to as *amplitude-shift keying (ASK).* ASK is used in some types of data communication when binary information is to be transmitted.

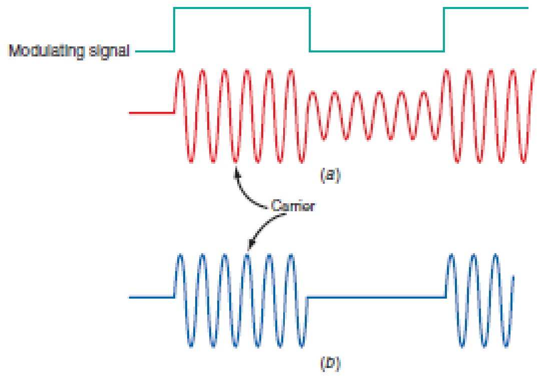
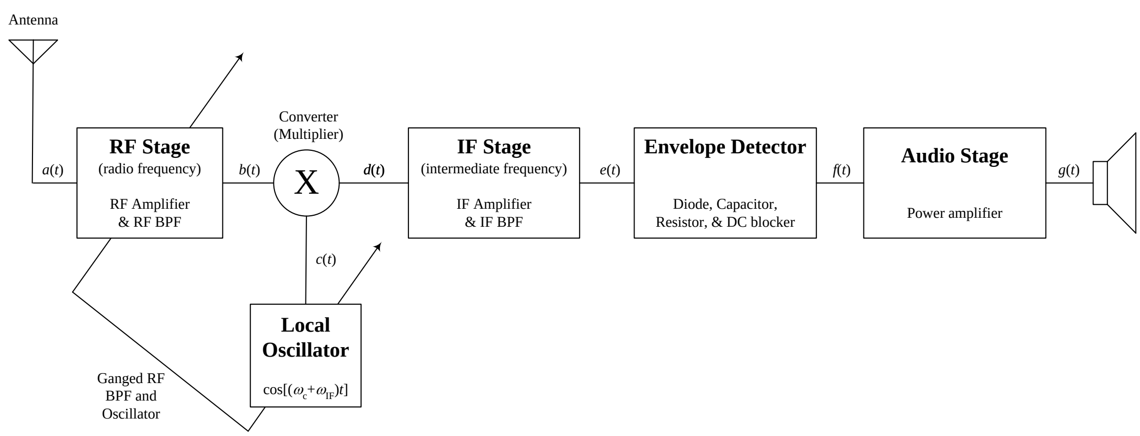


Fig : ASK (*a*) Fifty percent modulation. (*b*) One hundred percent modulation.

MODULE III
AM RECEIVERS

2.1: SUPERHETERODYNE AM RADIO RECEIVER

Since the inception of the AM radio, it spread widely due to its ease of use and more importantly, it low cost. The low cost of most AM radios sold in the market is due to the use of the full amplitude modulation, which is extremely inefficient in terms of power as we have seen previously. The use of full AM permits the use of the simple and cheap envelope detector in the AM radio demodulator. In fact, the AM demodulator available in the market is slightly more complicated than a simple envelope detector. The block diagram below shows the construction of a typical AM receiver and the plots below show the signals in frequency–domain at the different parts of the radio.



Description of the AM Superheterodyne Radio Receiver

Signal a(t) at the output of the Antenna: The antenna of the AM radio receiver receives the whole band of interest. So it receives signals ranging in frequency from around 530 kHz to 1650 kHz as shown by a(t) in the figure. Each channel in this band occupies around 10 kHz of bandwidth and the different channels have center frequencies of 540, 550, 560, …. , 1640 kHz.

Signal b(t) at the output of the RF (Radio Frequency) Stage: The signal at the output of the antenna is extremely week in terms of amplitude. The radio cannot process this signal as it is, so it must be amplified. The amplification does not amplify the whole spectrum of the AM band and it does not amplify a single channel, but a range of channels is amplified around the desired channel that we would like to receive. The reason for using a BPF in this stage although the desired channel is not completely separated from adjacent channels is to avoid possible interference of some channels later in the demodulation process if the whole band was allowed to pass (assume the absence of this BPF and try demodulating the two channels at the two edges of the AM band, you will see that one of these cannot be demodulated). Also, the reason for not extracting the desired channel alone is that extracting only that channel represents a big challenge since the filter that would have to extract it must have a constant bandwidth of 10 kHz and a center frequency in the range of 530 kHz to 1650 kHz. Such a filter is extremely difficult to design since it has a high Q–factor (center frequency/bandwidth) let alone the fact that its center frequency is variable. Therefore, the process of extracting only one channel is left for the following stages where a filter with constant center frequency may be used. Note in the block diagram above that the center frequency of the BPF in the RF stage is controlled by a variable capacitor with a value that is modified using a knob in the radio (the tuning knob).

Signal c(t) at the output of the Local Oscillator: This is simply a sinusoid with a variable frequency that is a function of the carrier frequency of the desired channel. The purpose of multiplying the signal b(t) by this sinusoid is to shift the center frequency of b(t) to a constant frequency that is called IF (intermediate frequency). Therefore, assuming that the desired channel (the channel you would like to listen to) has a frequency of fRF and the IF frequency that we would like to move that channel to is fIF, one choice for the frequency of the local oscillator is to be fRF + fIF. The frequency of the local oscillator is modified in the radio using a variable capacitor that is also controlled using the same tuning knob as the variable capacitor that controls the center frequency of the BPF filter in the RF stage. The process of controlling the values of two elements such as two variable capacitors using the same knob by placing them on the same shaft is known as GANGING.

Signal d(t) at the output of the Multiplier (Usually called frequency converter or mixer): The signal here should contain the desired channel at the constant frequency fIF regardless of the original frequency of the desired channel. Remember that this signal does not only contain the desired channel but it contains also several adjacent channels and also contain images of these channels at the much higher frequency 2fRF + fIF (since multiplying by a cosine shifts the frequency of the signal to the left and to the right). When this type of radios was first invented, a standard was set for the value for the IF frequency to be 455 kHz. There is nothing special about this value. A range of other values can be used.

Signal e(t) at the output of the IF Stage: Now that the desired channel is located at the IF frequency, a relatively simple to create BP filter with BW of 10 kHz and center frequency of fIF can be used to extract only the desired channel and reject all adjacent channels. This filter has a constant Q factor of about 455/10 = 45.5 (which is not that difficult to create), but more importantly has a constant center frequency. Therefore the output of this stage is the desired channel alone located at the IF frequency. This stage also contains a filter that amplifies the signal to a level that is sufficient for an envelope detector to operate on.

Signal f(t) at the output of the Envelope Detector: The signal above is input to an envelope detector that extracts the original unmodulated signal from the modulated signal and also rejects any DC that is present in that signal. The output of that stage becomes the original signal with relatively low power.

Signal g(t) at the output of the Audio Stage (Power Amplifier): Since the output of the envelope detector is generally weak and is not sufficient to drive a large speaker, the use of an amplifier that increases the power in the signal is necessary. Therefore, the output of that stage is the original audio signal with relatively high power that can directly be input to a speaker.

SECOND CHANNEL OR IMAGE FREQUENCY

One problem, which has to be contended with in the superheterodyne receiver, is its ability to pick up a second or imago frequency removed from the signal frequency by a value equal to twice the intermediate frequency.

To illustrate the point, refer Figure . In this example, we have a signal frequency of 1 MHz which mix to produce an IFof 455kHz. A second or image signal, with a frequency equal to 1 MHz plus (2 x 455) kHz or 1.910 MHz, can also mix with the 1.455 MHz to produce the 455 kHz.

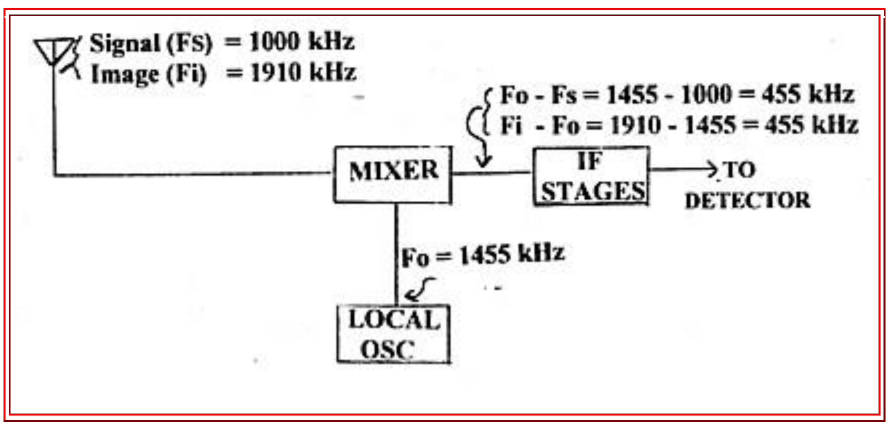


Figure : An illustration of how image frequency provides a second mixing product.

Reception of an image signal is obviously undesirable and a function of the RF tuned circuits (ahead of the mixer), is to provide sufficient selectivity to reduce the image sensitivity of the receiver to tolerable levels.

SINGLE SIDE BAND SUPPRESSED CARRIER MODULATION

Standard AM and DSBSC require transmission bandwidth equal to twice the message bandwidth. In both the cases spectrum contains two side bands of width W Hz, each. But the upper and lower sides are uniquely related to each other by the virtue of their symmetry about the carrier frequency. That is, given the amplitude and phase spectra of either side band, the other can be uniquely determined. Thus if only one side band is transmitted, and if both the carrier and the other side band are suppressed at the transmitter, no information is lost. This kind of modulation is called SSBSC and spectral comparison between DSBSC and SSBSC is shown in the figures (a) and (b)

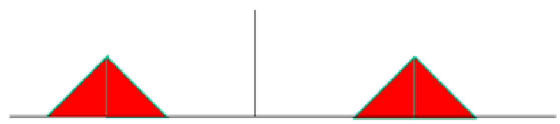


Fig (a): Spectrum of DSBSC

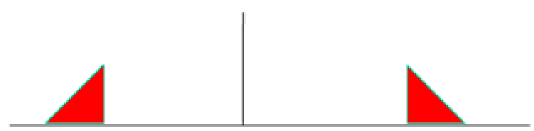


Fig (b) : Spectrum of SSBSC

Time-domain description

The time domain description of an SSB wave s(t) in the canonical form is given by the equation

$$\boldsymbol{S(t) = S_I(t)cos\,(2\pi fct) - S_Q(t)Sin(2\pi fct)}$$

Where
$S_I(t)$ *is the inphase component of SSB wave and* $S_Q(t)$ *is the Quadrature component of* SSB wave .
The inphase component $S_I(t)$ except for a scaling factor ,May be derived from S(t) by first multiplying S(t) by Cos(2πfct) and then passing the product through LPF.Similerly the Quadrature component S_Q(t) except for a scaling factor ,May be derived from S(t) by first multiplying S(t) by Sin(2πfct) and then passing the product through identical filter.
The fourier transform of $S_I(t)$ & S_Q(t) are related by

$$\boldsymbol{S_I(f) = \begin{cases} (f - f_c) + S((f + f_c), & -w \le f \le w \\ 0 & ;Otherwise \end{cases}}$$

$$\boldsymbol{S_Q(f) = \begin{cases} j(f - f_c) - S((f + f_c), & -w \le f \le w \\ 0 & ;Otherwise \end{cases}}$$

Where $-w \le f \le w$ defines the frequency band occupied by the message signal m(t)

Concider the SSB wave that is obtained by transmitting only the Upper side bands as shown in figure(a)
Two frequency shifted spectras S(f-fo) and S(f+fo) are obtained as shown in fig (b) & (c)
Corresponding to equations for inphase & quadrature phase components we can obtain figures which is (d) & (e)

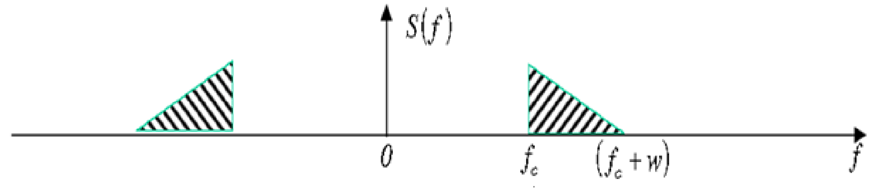


Fig (a) : Spectrum of SSB-SC USB

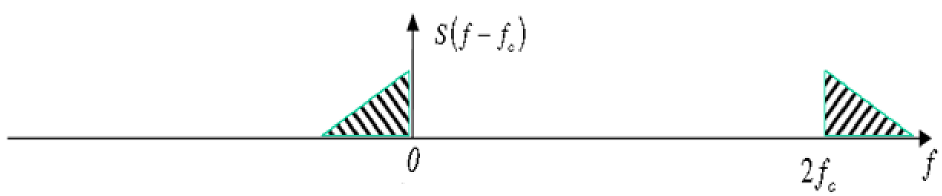
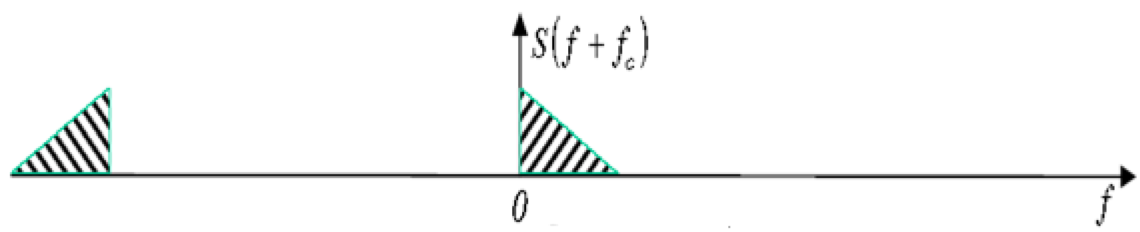
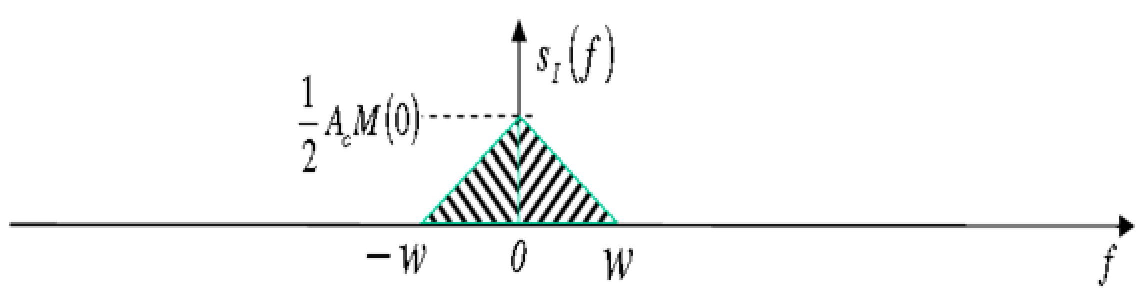


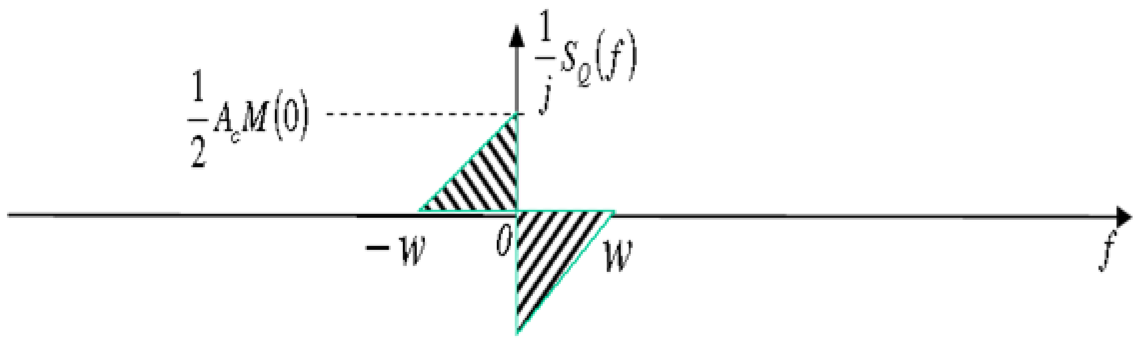
Fig (b): Spectrum of SSB-SC USB shifted right by fc



Fig(C) : Spectrum of SSB-SC USB shifted left by fc



Fig(d) : Spectrum of inphase component SSB-SC USB



Fig(d) : Spectrum of Quadrature component SSB-SC USB

From fig (e) it is found that

$S_I(f) = \frac{1}{2}Ac\ M(f)$

Where M(f) is the fourier transform of the message signal m(t). accordingly inphase component defined by equation is

$S_I(t) = \frac{1}{2}Ac\ m(t)$

From figure (e) it is found that

$$S_Q(f) = \begin{cases} \frac{-j}{2}\ Ac\ M(f), & f > 0 \\ 0 & ,f = 0 \\ \frac{j}{2}\ Ac\ M(f), & f < 0 \end{cases}$$

$$S_Q(f) = \frac{-j}{2}\ Ac\ sgn(f)\ M(f)$$

Where $sgn(f)$ is the Signum function

But from Hilbert transform ; $-j\ sgn(f)M(f) = \widehat{M}(f)$

Where $\widehat{M}(f)$ is the Hilbert transform of m(t)

Therefore

$$S_Q(t) = \frac{1}{2}Ac\ \hat{m}(t)$$

∴ The canonical form of SSB-SC USB can be represented by equation

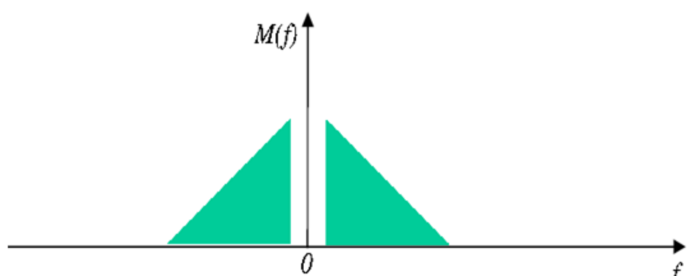
$$\boldsymbol{S_U(t) = \frac{1}{2}Ac\ m(t)cos\ (2\pi fct) - \frac{1}{2}Ac\ \hat{m}(t)Sin(2\pi fct)}$$

&

$$\boldsymbol{S_L(t) = \frac{1}{2}Ac\ m(t)cos\ (2\pi fct) + \frac{1}{2}Ac\ \hat{m}(t)Sin(2\pi fct)}$$

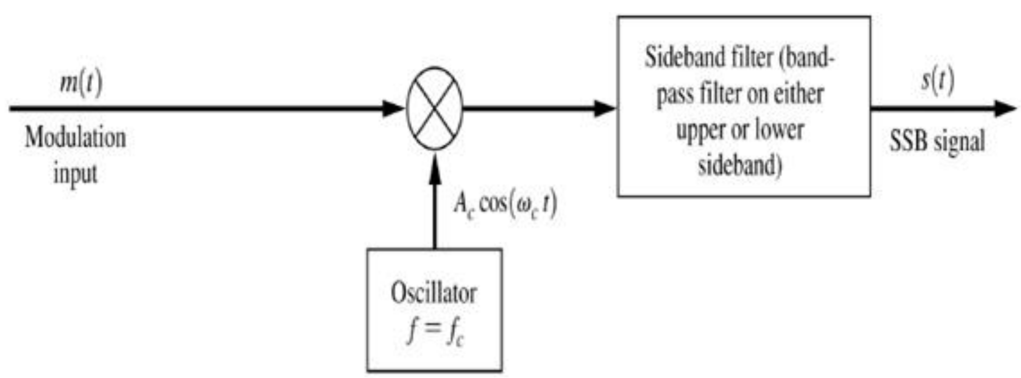
FREQUENCY DISCRIMINATION METHOD FOR GENERATING AN SSBSC MODULATED WAVE

Consider the generation of SSB modulated signal containing the upper side band only. From a practical point of view, the most severe requirement of SSB generation arises from the unwanted sideband, the nearest component of which is separated from the desired side band by twice the lowest frequency component of the message signal. It implies that, for the generation of an SSB wave to be possible, the message spectrum must have an energy gap centered at the origin as shown in figure. This requirement is naturally satisfied by voice signals, whose energy gap is about 600Hz wide.



Message spectrum with energy gap at the origin

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.



Filter Method

Application of this method requires that the message signal satisfies two conditions:

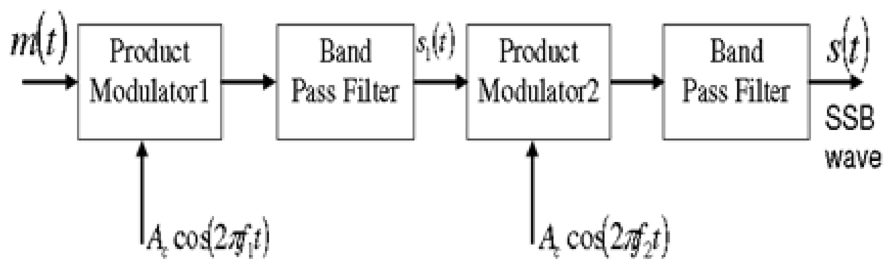
1. The message signal m(t) has no low-frequency content.
   Example: - speech, audio, music.
2. The highest frequency component W of the message signal m(t) is much less than the carrier frequency fc.

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.

When it is necessary to generate an SSB modulated wave occupying a frequency band that is much higher than that of the message signal, it becomes very difficult to design an appropriate filter that will pass the desired side band and reject the other. In such a situation it is necessary to resort to a multiple-modulation process so as to ease the filtering requirement. This approach is illustrated in the following figure involving two stages of modulation.



The SSB modulated wave at the first filter output is used as the modulating wave for the second product modulator, which produces a DSBSC modulated wave with a spectrum that is symmetrically spaced about the second carrier frequency f_2. The frequency separation between the side bands of this DSBSC modulated wave is effectively twice the first carrier frequency f_1, there by permitting the second filter to remove the unwanted side band.

PHASE DISCRIMINATION METHOD OF SSB GENERATION

Fig. 1 shows the block diagram for the phase shift method of SSB generation This system is used for the suppression of lower sideband . This system uses two balanced modulators M_1 and M_2 and two 90^o phase shifting networks as shown.

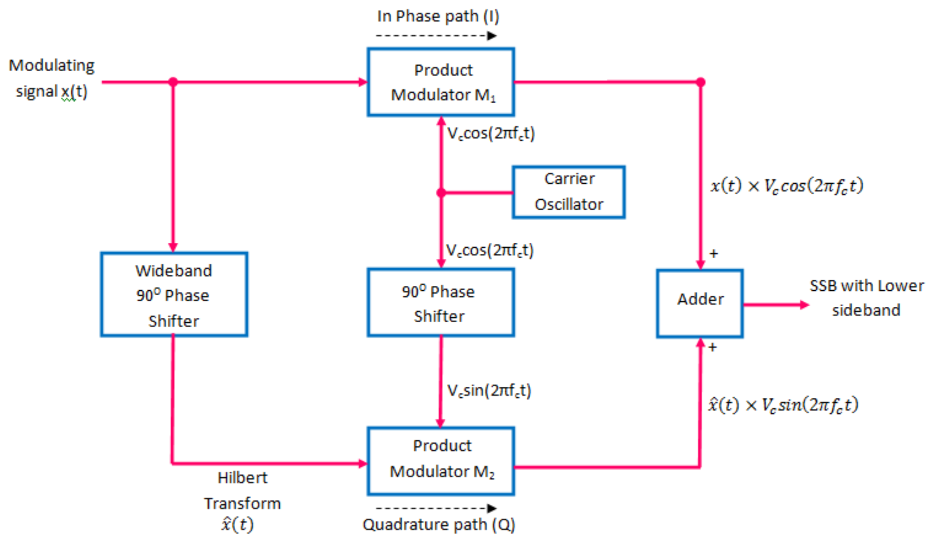


Fig 1: Phase shift method for generating SSB signal

WORKING OPERATION

The message signal x(t) is applied to the product modulator M_1 and through a 90^o phase shifter to the product modulator M_2. Hence, we get the Hilbert transform

$\hat{x}(t)$

at the output of the wideband 90^o phase shifter .

The output of carrier oscillator is applied as it is to modulator M_1 whereas it s passed through a 90^o phase shifter and applied to the modulator M_2.

$$Output\ of\ M_1 = x(t) \times V_c cos(2\pi f_c t)$$

$$and\quad Output\ of\ M_2 = \hat{x}(t) \times V_c sin(2\pi f_c t)$$

The outputs of M_1 and M_2 are applied to an adder .

$$Adder\ \ Output\ = x(t) \times V_c cos(2\pi f_c t) + \hat{x}(t) \times V_c sin(2\pi f_c t)$$

$$Or\quad Adder\ Output = V_c[x(t) \times cos(2\pi f_c t) + \hat{x}(t) \times sin(2\pi f_c t)]$$

Therefore,

This expression represents the SSB signal with only LSB i.e. it rejects the USB .

THE WEAVER METHOD OF SSB GENERATION

The Weaver architecture for the SSB-SC generation is shown in the Figure . It consists of four balanced modulators, two carrier signal generators, two audio low-pass filters, and two 90 phase-shift networks. In practice, the phase-shift networks can be replaced by using a pair of quadrature oscillators.

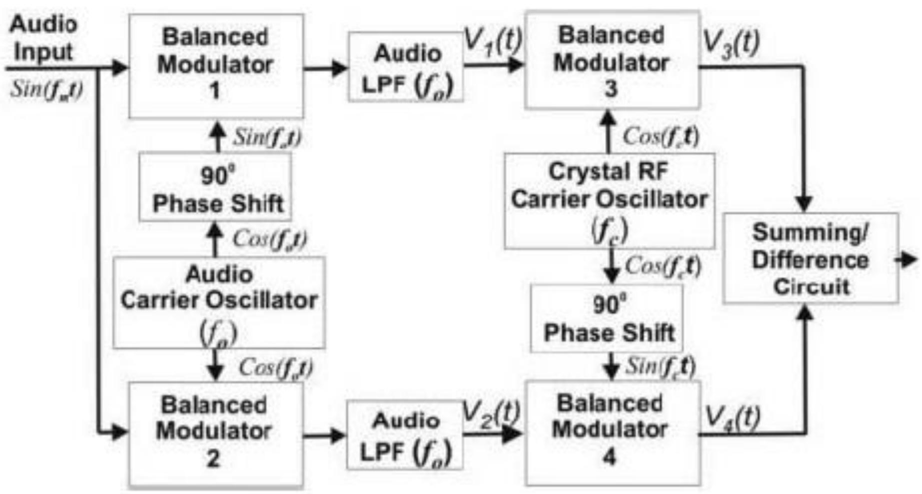


Figure : *The Weaver Method Block Diagram*

In the first stage, it makes an IQ modulation with fo at the centre of the audio band. The filtered output of the mixers are given by the signals

$V_1(t) = Low\ Pass\ (\ Sin\ 2\pi f_m t. Sin\ 2\pi f_o t) = \frac{1}{2}Cos[2\pi(f_o - f_m)t)$ ……………………(1)

$V_2(t) = Low\ Pass\ (\ Sin\ 2\pi f_m t. Cos\ 2\pi f_o t) = \frac{1}{2}Sin[2\pi(f_o - f_m)t)$……………………(2)

In the second stage, the two mixer outputs given in Eqs. (1) and (2) are IQ modulated with the RF carrier and is followed by a summing/difference circuit which generates the desired sideband signal,

$V_3(t) = \frac{1}{2}Cos[2\pi(f_o - f_m)]Cos\ 2\pi f_c t) = \frac{1}{4}\ (Cos\ [2\pi(f_c + (f_o - f_m))t] + \frac{1}{4}\ (Cos\ [2\pi(f_c - (f_o - f_m))t]$...(3)

$V_4(t) = \frac{1}{2}Sin[2\pi(f_o - f_m)]. Sin\ 2\pi f_c t) = \frac{1}{4}\ (Cos\ [2\pi(f_c + (f_o - f_m))t] - \frac{1}{4}\ (Cos\ [2\pi(f_c - (f_o - f_m))t]$...(4)

From (3) and (4), it is evident that an addition operation will yield the upper sideband, while a subtraction will yield the lower sideband .

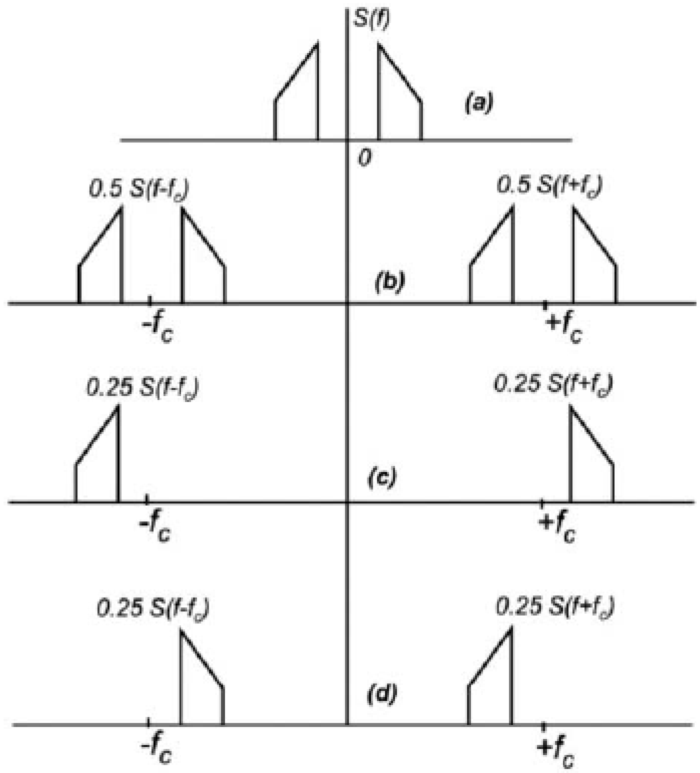


Figure 2: Amplitude spectrum of (a) Baseband signal. (b) Double sideband (DSB) signal. (c) Upper single sideband (USB) signal. (d) Lower single sideband (LSB) signal.

MODULE IV
ANGLE MODULATION:

3.1 INTRODUCTION

Angle modulation is one in which the angle of carrier wave is varied in accordance with the baseband signal. Let $\theta_i(t)$ denote the angle of a modulated sinusoidal carrier at time t; it is assumed to be a function of the information–bearing signal or message signal. We express the resulting angle-modulated wave as

$$S(t) = Ac\ Cos\ [\theta_i(t)] \quad \text{...(1)}$$

where Ac is the carrier amplitude

The *average frequency* in Hertz over an interval from t to t+Δt is given by

$$f_{\nabla t}(t) = \frac{\theta_i(t+\Delta t)-\theta_i(t)}{2\pi\Delta t} \quad \text{... (2)}$$

The *instantaneous frequency* the angle-modulated signal $s(t)$:

$$f_i(t) = \lim_{\Delta t\to 0} f_{\Delta t}(t) = \lim_{\Delta t\to 0}\left[\frac{\theta_i(t+\Delta t)-\theta_i(t)}{2\pi\Delta t}\right] = \frac{1}{2\pi}\frac{d\theta_i(t)}{dt} \quad \text{..................................(3)}$$

For an *unmodulated carrier*, the angle $\theta i(t)$ is given by

$$\theta_i(t) = 2\pi f c t + \phi_c \text{...(4)}$$

and corresponding phasor rotates with a constant angular velocity equal to $2\pi f$c. The constant ϕ_c is the value of $\theta i(t)$ at t=0.

There are an infinite number of ways in which the angle $\theta i(t)$ may be varied in some manner with the message (baseband) signal. We shall consider only two commonly used methods, *phase-modulation* and *frequency modulation*.

Phase modulation (PM) is that form of angle modulation in which the instantaneous angle $\theta i(t)$ is varied linearly with the message signal as shown by

$$\theta_i(t) = 2\pi f ct + kp\, m(t) \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (5)$$

The term $2\pi fct$ represents the angle of the unmodulated carrier; kp represents the *phase sensitivity* of the modulator, expressed in *radians per volt* on the assumption that $m(t)$ is a voltage waveform. For convenience, we have assumed in Eq. (5) that the angle of the unmodulated carrier is zero at t=0. The phase-modulated signal $s(t)$ is thus described in the time domain by

$$S(t) = Ac\ Cos\ [2\pi f ct + kp\, m(t)] \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (6)$$

Frequency modulation (FM) is that form of angle modulation in which the instantaneous frequency $fi(t)$ is varied linearly with the message signal $m(t)$, as shown by

$$f_i(t) = f_c + k_f m(t) \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (7)$$

fc : The frequency of the unmodulated carrier

kf : The *frequency sensitivity* of the modulator (Hertz per volt)

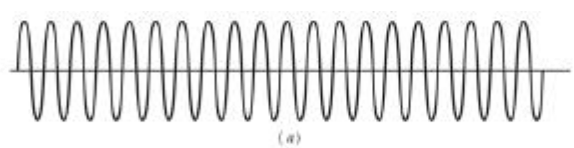
Integrating Eq. (7) with respect to time and multiplying the result by 2π, we get

$$\theta_i(t) = 2\pi f_c t + 2\pi k_f \int_0^t m(\tau)d\tau \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (8)$$

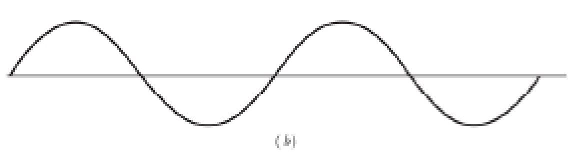
where, for convenience, we have assumed that the angle of theunmodulated carrier wave is zero at t=0. The frequency-modulated signal is therefore described in the time domain by

$$S(t) = Ac\ Cos\ \left[2\pi f_c t + 2\pi k_f \int_0^t m(\tau)d\tau\right] \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (9)$$

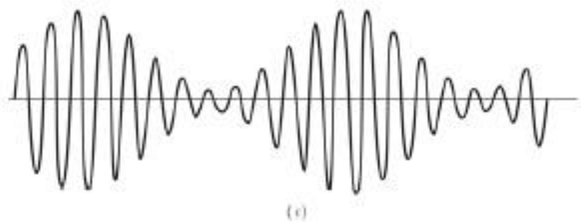
a) Carrier wave



b) Sinusoidal modulating signal

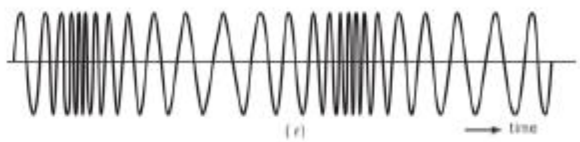


c) Amplitude-modulated signal



d) Phase-modulated signal

e) Frequency-modulated signal



PROPERTIES OF ANGLE-MODULATED WAVES

Property 1: Constancy of Transmitted Power:

◊ From both Eqs. (6) and (9), we readily see that the amplitude of PM and FM waves is maintained at a constant value equal to the carrier amplitude *Ac* for all time *t*, irrespective of the sensitivity factors *kp* and *kf* .

◊ Consequently, the average transmitted power of angle modulated waves is a constant, as shown by

$P_{av} = \frac{1}{2}A_c^2$

where it is assumed that the load resistance is 1 ohm & $P = V^2/R$

Property 2: Nonlinearity of the Modulation Process

◊ Both PM and FM waves violate the principle of superposition.

◊ For example, the message signal $m(t)$ is made up of two different components, $m_1(t)$ and $m_2(t)$

m(t)= m_1(t)+m_2(t)

◊ Let $s(t)$, $s1(t)$, and $s2(t)$ denote the PM waves produced by $m(t)$, m_1 (t) andm_2 (t) in Eq (5) respectively .We may express these PM waves as follows:

$$S(t) = Ac\ Cos\ [2\pi fct + kp\ (m_1(t) + m_2(t)]$$

$$S_1(t) = Ac\ Cos\ [2\pi fct + kp\ (m_1(t)]$$

$$m(t) = m_1(t) + m_2(t)$$

$S_2(t) = Ac\ Cos\ [2\pi fct + kp\ (m_2(t)]$ $\qquad$ $S(t) \neq S_1(t) + S_2(t)$

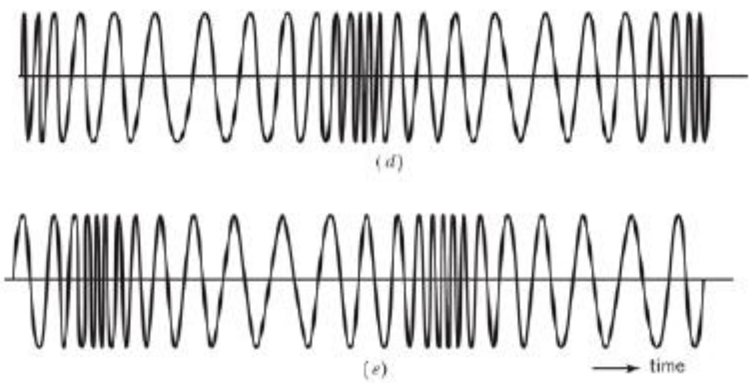
Frequency modulation offers superior noise performance compare to amplitude modulation,

Property 3: Irregularity of Zero-Crossings

◊ *Zero-crossing* are defined as the instants of time at which a waveform changes its amplitude from positive to negative value or the other way around.

◊ The zero-crossings of a PM or FM wave no longer have a perfect regularity in their spacing across the time-scale.

◊ The irregularity of zero-crossings in angle-modulated waves is attributed to the nonlinear character of the modulation process.



Property 4: Visualization Difficulty of Message Waveform

◊ In AM, we see the message waveform as the envelope of the modulated wave, provided the percentage modulation is less than 100 percent.

(AM: The percentage modulation over 100 percent→ phase reversal→ distortion)

◊ This is not so in angle modulation, as illustrated by the corresponding waveform of Figures *d* and *e* for PM and FM, respectively.

Property 5-Trade-OFF of Increased Transmission Bandwidth for Improved Noise Performance

◊ An important advantage of angle modulation over amplitude modulation is the realization of improved noise performance.

◊ This advantage is attributed to the fact that the transmission of a message signal by modulating the angle of a sinusoidal carrier wave is less sensitive to the presence of additive noise than transmission by modulating the amplitude of the carrier.

◊ The improvement in noise performance is achieved at the expense of a corresponding increase in the transmission bandwidth of angle modulation requirement modulation.

◊The use of angle modulation offers the possibility of exchanging an increase in the transmission bandwidth for an improvement in noise performance.

◊ Such a trade-off is not possible with amplitude modulation since the transmission bandwidth of an amplitude-modulated wave is fixed somewhere between the message bandwidth W and $2W$, depending on the type of modulation employed.

EQUIVALENCE BETWEEN PM AND FM

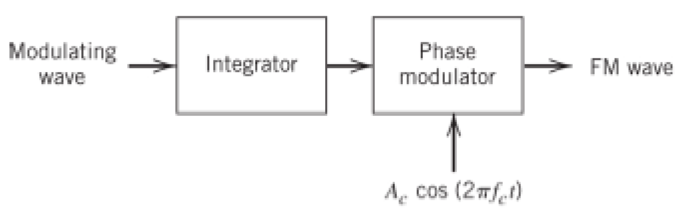
Comparing Eq. (6) with (9) reveals that an FM signal may be regarded as a PM signal in which the modulating wave is $\int_0^t m(\tau)d\tau$ in place of $m(t)$.

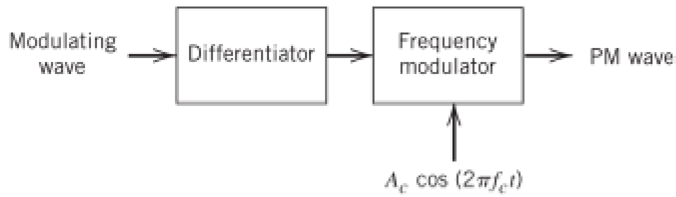
$$S(t) = Ac\ Cos\ [2\pi f ct + kp\ m(t)]$$
$$S(t) = Ac\ Cos\ \left[2\pi f_c t + 2\pi k_f \int_0^t m(\tau)d\tau\right]$$

The FM signal can be generated by first integrating $m(t)$ and then using the result as the input to a phase modulator.

◊ Conversely, a PM signal can be generated by first differentiating $m(t)$ and then using the result as the input to a frequency modulator.

◊ We may thus deduce all the properties of PM signals from those of FM signals versa. Henceforth, we concentrate attention on FM signals.





	$\theta_i(t)$	$f_i(t)$
Unmodulated signal	$2\pi f_c t$	f_c
PM signal	$2\pi f_c t + k_p m(t)$	$f_c + \frac{k_p}{2\pi}\frac{dm(t)}{dt}$
FM signal	$2\pi f_c t + 2\pi k_f \int_0^t m(\tau) d\tau$	$f_c + k_f m(t)$

3.2 FREQUENCY MODULATION

The FM signal $s(t)$ define by Eq. (6) is a nonlinear function of the modulating signal $m(t)$, which makes frequency modulation a nonlinear modulation process.

Consider then a sinusoidal modulating signal define by

$m(t) = Am\ Cos\ (2\pi f_m t)$ ....(10)

The instantaneous frequency of the resulting FM signal is

$f_i(t) = f_c + k_f A_m\ Cos\ (2\pi f_m t) = f_c + \Delta f\ Cos\ (2\pi f_m t)$ ....(11)

$\Delta f = k_f A_m$

The quantity Δf is called the *frequency deviation*, representing the maximum departure of the instantaneous frequency of the FM signal form the carrier frequency fc.

A fundamental characteristic of an FM signal is that the frequency deviation Δf is proportional to the amplitude of the modulating signal and is independent of the modulating frequency.

Using Eq. (11), the angle $\theta i(t)$ of the FM signal is obtained as

$$\theta_i(t) = 2\pi \int_0^t f_i(t)dt = 2\pi f_c t + \frac{\Delta f}{fm}\sin(2\pi f_m t)$$

The ratio of the frequency deviation Δf to the modulationfrequency fm is commonly called the *modulation index* of the FM signal.
The modulation index is denoted by β:

$$\beta = \frac{\Delta f}{fm}$$

$$\theta_i(t) = 2\pi f_c t + \beta \sin(2\pi f_m t)$$

The parameter β represents the phase deviation of the FM signal, i.e. the maximum departure of the angle $\theta i(t)$ from the angle $2\pi fct$ of the unmodulated carrier. β is measured in radians.
◊ The FM signal itself is given by

$$S(t) = Ac \cos[\,2\pi f_c t + \beta \sin(2\pi f_m t)]$$

Depending on the value of the modulation index β, we may distinguish two cases of frequency modulation:

◊ *Narrow-band FM*, for which β is small compared to one radian.

◊ *Wide-band FM*, for which β is large compared to one radian.

AVERAGE POWER OF FM SIGNAL

1. The spectrum of an FM signal contains a carrier component (n=0) and an infinite set of side frequencies located symmetrically on either side of the carrier at frequency separations of fm, $2fm$, $3fm$, ...
   (An AM system gives rise to only one pair of side frequencies.)
2. For the special case of β small compared with unity, only the Bessel coefficients $J_0(\beta)$ and $J_1(\beta)$ have significant values , so that the FM signal is effectively composed of a carrier and a single pair of side frequencies at $fc \pm fm$. (This situation corresponds to the special case of narrowband FM that was considered previously)
3. The amplitude of the carrier component of an FM signal is dependent on the modulation index β. The physical explanation for this property is that the envelope of an FM signal is constant, so that the average power of such a signal developed across a 1–ohm resistor is also constant, as shown by $P_{av} = \frac{1}{2}A_c^2$

SPECTRA OF FM SIGNALS

Here we are investigating the ways in which variations in the amplitude and frequency of a sinusoidal modulating signal affect the spectrum of the FM signal.

We know the equation for FM wave as

$$S(t) = Ac\ Cos\ \left[2\pi f_c t + 2\pi k_f \int_0^t m(\tau)d\tau\right]$$

To discover the frequency spectrum of a typical FM signal we can take the example of simple sine wave modulation at a modulation frequency, *fm*

$$m(t) = Am\cos(2\pi f_m t)$$

which produces an instantaneous FM signal frequency of

$$f_i(t) = f_c + k_f A_m \cos(2\pi f_m t)$$

This expression tells us that fi swings up and down either side of fcover a range of $\pm k_f A_m$.This range is usually described in terms of the modulated signal's peak frequency deviation value, defined as

$$\Delta f = k_f A_m$$

since it indicates the largest swing or deviation in frequency either side of fc. Note that its value depends upon the magnitude of the modulation, Am, but not upon the modulating frequency, fm

So;

$$\theta_i(t) = 2\pi f_c t + \frac{\Delta f}{fm}\sin(2\pi f_m t)$$

It is conventional to define a quantity called the modulation index,

$$\beta = \frac{\Delta f}{fm}$$

we can then write the FM wave in the form

$$S(t) = Ac\cos[\ 2\pi f_c t + \beta\sin(2\pi f_m t)]$$

This provides us with information on how the modulated signal varies with time. However we often need to know the frequency spectrum of the modulated wave – for example, in order to be able to determine the bandwidths of any filters, amplifiers, etc that we require.

So using Bessel function we can rewrite the above equation as

$$S(t) = AJ_0\{\beta\}cos\ (2\pi f_C t) + A\sum_{k=1}^{\infty} J_{2k}\{\beta\}[\sin\{2\pi(f_c + 2kf_m)\ t\} + \sin\{2\pi(f_c - 2kf_m)t\}] + A\sum_{k=1}^{\infty} J_{2k+1}\{\beta\}[\mathrm{Cos}\{2\pi(f_c + (2k+1)f_m)\ t\} - \mathrm{Cos}\{2\pi(f_c - (2k+1)f_m)t\}]$$

where $J_n\{\beta\}$ is the Bessel Function (first kind, integer order, n) for the value, β.

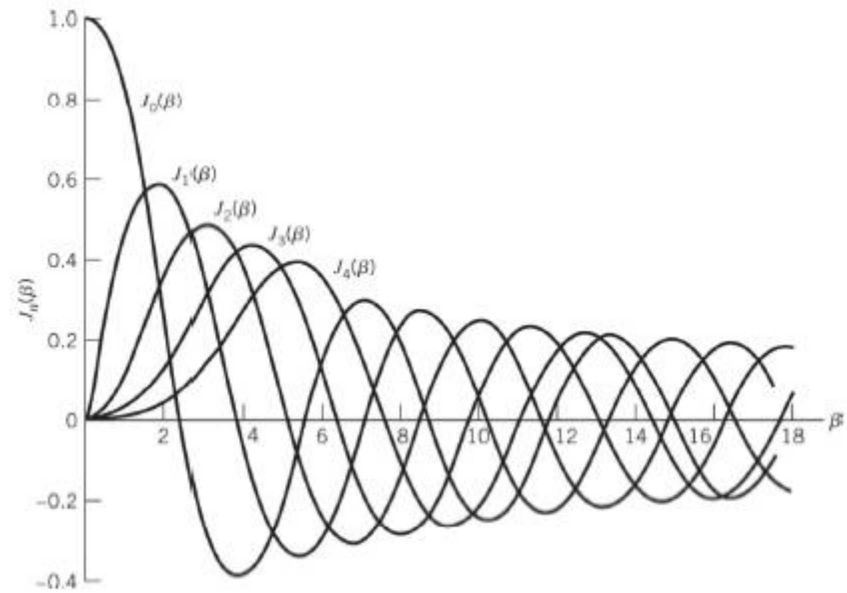


FIGURE: ***Plots of Bessel functions of the first kind***

Clearly, above expression is much more complicated than the equivalent for an AM wave modulated with a single sinewave. The sinewave modulated AM signal has only three spectral components, at the frequencies, fc, (fc +fm), and (fc -fm), . The FM wave has spectral components at all the frequencies, (fc +nfm) , where n can be any integer from -∞ to +∞. This result is rather startling since it means that, strictly speaking, a system has to provide an infinite bandwidth to carry an accurate FM (or PM) signal! Fortunately, there is a general tendency for $|J_n\{\beta\}| \to 0 \ as \ \ n \ \to \infty$

and the higher order Bessel function values fall quickly with n when the modulation index is small. In many practical situations we can arrange that $|\beta|<<\ 1$ and when this is true we find that

$$J_0\{\beta\} \approx 1 \qquad J_1\{\beta\} \ \approx \frac{\beta}{2} \quad ; \ \ J_n \approx 0 \ \ for\, n\ >1$$

An FM signal produced with a low modulation index (i.e. where Δf<< fm) is called a *narrowband FM signal*. For most purposes we can ignore the high-order Bessel function contributions and represent its spectrum with the approximation

$$S(t) = AJ_0\{\beta\}cos\ (2\pi f_C t) + A\, J_1\{\beta\}[cos\ \{2\pi(f_C + f_m)t\} - cos\ \{2\pi(f_C - f_m)t\}]$$

This narrowband FM (or PM) signal is similar to AM in that it has sideband components at $(f_C \pm f_m)$, hence it only requires a transmission bandwidth of $2f_m$. Its spectrum differs from AM in two ways. Firstly, the total amplitude of the modulated wave remains almost constant. Secondly, the two sideband components are “180 degrees out of phase” (their signs differ).

A ‘high-β’ FM wave can be thought of as a carrier whose frequency is varied over a relatively wide range. It will therefore require a transmission bandwidth of at least $2\Delta f$. Combining this result with that for a narrowband FM wave leads to Carson's Rule, that the minimum practical bandwidth required to transmit an FM/PM signal will be

$$B = 2(f_m + \Delta f)$$

This rule is a useful guide when we have to choose a system to carry an FM signal. It should be remembered, however, that in theory FM signals require an infinite bandwidth if we want to avoid any signal distortion during transmission.

AM VERSUS FM COMPARISON CHART		
	AM	**FM**
Stands for	AM stands for Amplitude Modulation	FM stands for Frequency Modulation
Modulating differences	In AM, a radio wave known as the "carrier" or "carrier wave" is modulated in amplitude by the signal that is to be transmitted. The frequency and phase remain the same.	In FM, a radio wave known as the "carrier" or "carrier wave" is modulated in frequency by the signal that is to be transmitted. The amplitude and phase remain the same.
Pros and cons	AM has poorer sound quality compared with FM, but is cheaper and can be transmitted over long distances. It has a lower bandwidth so it can have more stations available in any frequency range.	FM is less prone to interference than AM. However, FM signals are impacted by physical barriers. FM has better sound quality due to higher bandwidth.
Frequency Range	AM radio ranges from 535 to 1705 KHz (OR) Up to 1200 bits per second.	FM radio ranges in a higher spectrum from 88 to 108 MHz. (OR) 1200 to 2400 bits per second.
Bandwidth Requirements	Twice the highest modulating frequency. In AM radio broadcasting, the modulating signal has bandwidth of 15kHz, and hence the bandwidth of an amplitude-modulated signal is 30kHz.	Twice the sum of the modulating signal frequency and the frequency deviation. If the frequency deviation is 75kHz and the modulating signal frequency is 15kHz, the bandwidth required is 180kHz.

AM VERSUS FM COMPARISON CHART		
	AM	**FM**
Zero crossing in modulated signal	Equidistant	Not equidistant
Complexity	Transmitter and receiver are simple but syncronization is needed in case of SSBSC AM carrier.	Tranmitter and reciver are more complex as variation of modulating signal has to be converted and detected from corresponding variation in frequencies.(i.e. voltage to frequency and frequency to voltage conversion has to be done).
Noise	AM is more susceptible to noise because noise affects amplitude, which is where information is "stored" in an AM signal.	FM is less susceptible to noise because information in an FM signal is transmitted through varying the frequency, and not the amplitude

BLOCK DIAGRAM OF SUPER HETERODYNE RECEIVER

In superheterodyne radio receivers, the incoming radio signals are intercepted by the antenna and converted into the corresponding currents and voltages. In the receiver, the incoming signal frequency is mixed with a locally generated frequency. The output of the mixer consists of the sum and difference of the two frequencies. The mixing of the two frequencies is termed **heterodyning**. Out of the two resultant components of the mixer, the sum component is rejected and the difference component is selected. The value of the difference frequency component varies with the incoming frequencies, if the frequency of the local oscillator is kept constant. It is possible to keep the frequency of the difference components constant by varying the frequency of the local oscillator according to the incoming signal frequency. In this case, the process is called ***Superheterodyne*** and the receiver is known as a ***superheterodyne radio receiver.***

SUPERHETERODYNE AM RECEIVER

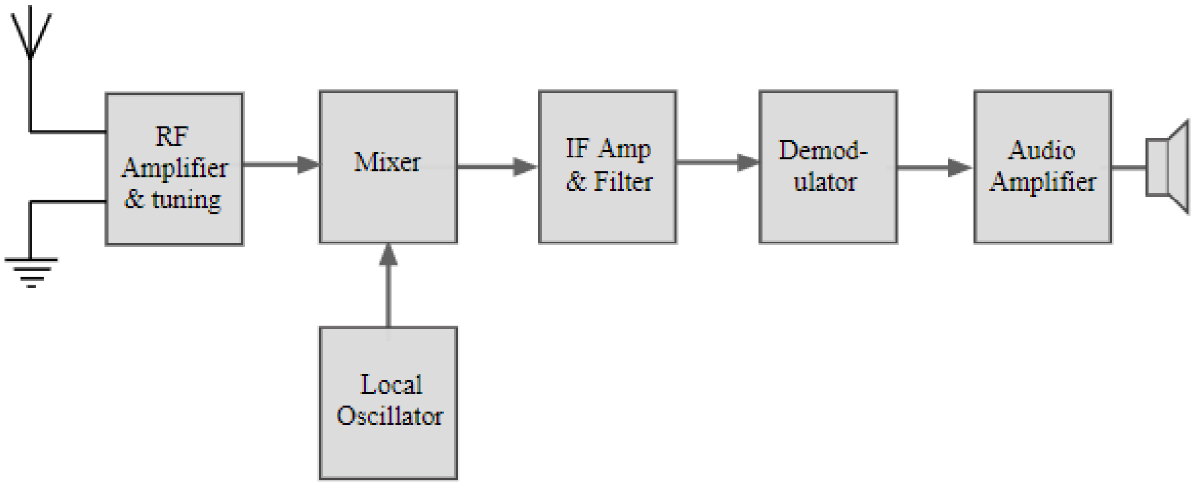
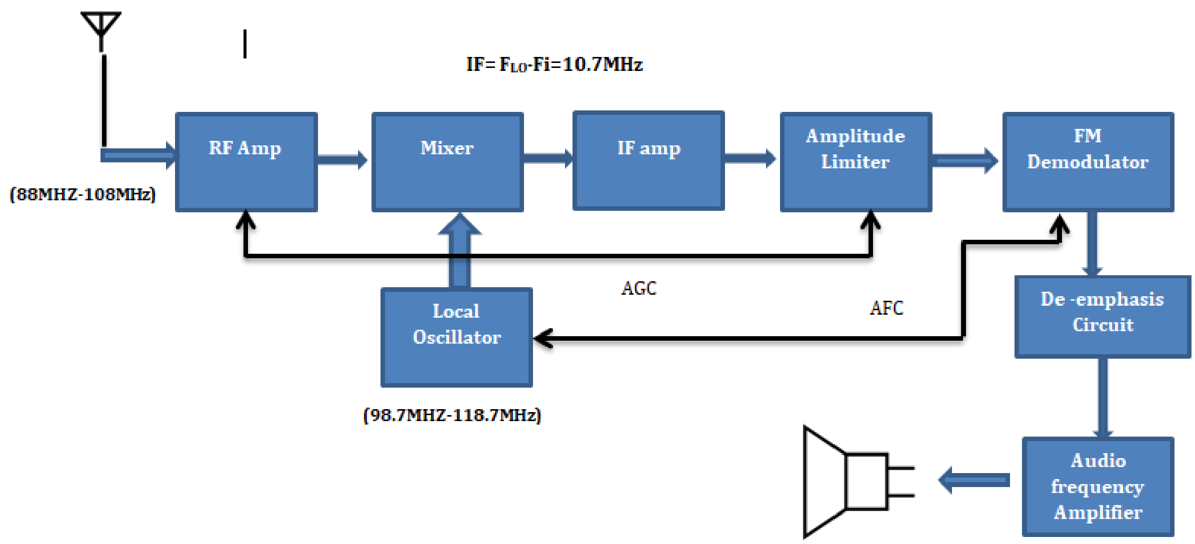


Fig 6.14: Block diagram of a basic superheterodyne AM receiver

- The RF Amplifier: The receiving antenna intercepts the radio signals and feeds the RF amplifier, The RF amplifier selects the desired signal frequency and amplifies its voltage, The RF amplifier is a small-signal voltage amplifier that operates in the RF range. This amplifier is tuned to the desired signal frequency by using capacitive tuning.
- **The mixer**: After suitable amplification of the RF signal it is fed to the mixer. The mixer takes another input from a local oscillator, which generates a frequency according to the frequency of the selected signal so that the difference equals a predetermined value. The mixer consists of a non-linear device, such as a transistor. Due to the non-linearity, the mixer output consists of sum and difference frequency components along with their higher harmonics. A tuned circuit at the output of the mixer selects only the difference component while rejecting all other components. The difference component is called the intermediate frequency or IF the value of IF frequency is always constant and is equal to 455 KHz.
- **IF amplifier** : The IF signal is fed to an IF amplifier with two amplifier stages. This provides enough signal amplification so that the signal is properly detected.
  All the three stages (RF Amplifier, Mixer & IF amplifier) are tuned at the same time to the required frequency through the ganged Capacitors, which consists of the three tuning capacitors.
- **Detector**: The amplified IF signal is fed to the linear diode detector, which demodulates the received AM signal. The output of the detector stage is the original modulating signal. This signal is given to the audio driver stage, which amplifies its voltage to drive the power amplifier, which is the last stage of the receiver.
- **The Loud speaker**: The power of the modulating signal and finally is passed to the power amplifier amplifies the speaker. The speaker converts the audio currents into sound energy.

SUPER HETERODYNE FM RECEIVER

FM Receiver operates in VHF Band of 88-108Mhz and have intermediate frequency of 10.7MHz and band width 150KHZ. A super heterodyne FM Receiver has a Receiving antenna ,RF Amplifier, Mixer & Local oscillator, IF Amplifier, Amplitude limiter, FM demodulator, De-emphasis circuit, AF Amplifier & Loud speaker.



WORKING

The difference between SH AM & FM receiver is that in FM the decoder block of AM receiver is replaced with 3 blocks- Amplitude limiter, FM demodulator, De-Emphasis circuit.

- **RF Amplifier**:-The receiving antenna receives many signals from many broadcasting stations. RF amplifier selects a particular signal and amplifies that because while transmitting the signal travelled a long distance and hence its energy should have been lost. RF Amplifier uses a tuned circuit (parallel LC Circuit)for selecting the signal. The RF amplifier of FM Receiver should be designed to handle a large bandwidth of 150KHz.
- **Mixer & Local Oscillator**:- The RF amplifier, Mixer & Local oscillator are ganged tuned to produce an IF of 10.7MHz.The process of mixing two different frequencies to produce a new frequency is called Heterodying. The IF is the difference between Local oscillator frequency & frequency from RF amplifier. The L.O frequency is 98.7MHz to 118MHz.It is always kept at higher value than RF amplifier. Mixer does the Heterodyne action. The output of mixer is $IF=F_{LO}-Fi=10.7MHz$.
- **IF amplifier:-**It is a multistage tuned class A amplifier which amplifies IF signals and there by provides high selectivity, gain, better adjacent channel rejection ratio. The bandwidth =150KHz.

- **Amplitude limiter:-**In FM amplitude should be constant. Limiter is used to eliminate the amplitude variations. Amplitude variations may occur due to noise or interferences introduced in the channel.
- **FM Demodulator/Discriminator:-**Discriminator converts frequency variations into amplitude variations ie.FM to AM and then Detector circuit extract the original message signal by removing HF signals.
- **De-Emphasis circuit**:-It is used to remove the extra boost given to the AF signal in the pre emphasis circuit of transmitter. It is a simple low pass filter.
- **AF amplifier**:- The output of de-emphasis circuit is the applied to AF amplifier to improve the strength of AF signal and fed to loud speaker. In addition to there is AGC (automatic gain control) and AFC(automatic frequency control) is used in FM Receiver.

ADVANTAGES OF SUPER HETERODYNE RECEIVER

The super heterodyne receiver has a number of advantages over TRF receiver:

1. Improved selectivity.

2. More uniform selectivity

3. Improved receiver stability

4. Higher gain per stage.

5. Uniform bandwidth because of fixed IF .

APPLICATIONS OF SUPERHETRODYNE RECEIVER

1. AM receiver .
2. FM receiver
3. SSB receiver
4. Communication receivers
5. Television receivers
6. Radar receivers

TUNING RANGE

Many radio receivers are fixed-tuned to a specific signal frequency, while others are designed to ,be continuously adjustable over a range (or band) of frequencies. Tuning of RF amplifiers and the oscillator is accomplished by varying the capacitance (or sometimes the inductance) in resonant circuits that act as band pass filters. The turning range is usually limited by the range over which the capacitance can vary, typically a maximum of about 10:1

The resonant frequency of a high Q tuned circuit is given by $f_0 = \frac{1}{2\pi\sqrt{LC}}$

The circuit ***frequency tuning range*** ratio R_f is defined as the ratio of its maximum frequency to its minimum frequency and the corresponding ***capacitance tuning range ratio*** Rc is the ratio of maximum capacity to minimum capacity.

Applying these ratios to the resonant equation gives

$$R_c = \frac{C_{max}}{C_{min}}$$

$$R_f = \frac{F_{max}}{F_{min}} = \sqrt{R_c}$$

If the oscillator frequency is chosen to be above the receive signal frequency. then the tuning range of" the oscillator tuned circuit will be smaller than that of the RF amplifier tuned circuits. If the oscillator frequency is below the signal, then its tuning range will be larger than that of the RF circuits and also its harmonics may fall within the signal range to cause interference.

TRACKING

The Receiver has a number of tunable circuits such as the antenna or mixer or a local oscillator tuned circuits. In a super heterodyne receiver mainly there are three tuned circuits. Antenna or RF tuned circuits, mixer tuned circuit, and local oscillator tuned circuit. All these circuits must be ; tuned to get proper RF input to get IF frequency at the output of mixer .For this reason the capacitors in the various stages are ganged (mechanically coupled to each other). Due to this arrangement, it is possible to use one tuning control to vary the tuning capacitors simultaneously.

The local oscillator frequency *Fo* must be accurately adjusted to a value which is above the signal frequency Fs by IF .That is *Fo =Fs + IF*. If this tuning is not done accurately, then the frequency difference Fo-Fs is not correct. This type of errors are knows as tracking errors.

Tracking is a process in which the local oscillator follows or tracks the signal frequency to have a correct frequency difference. Due to the tracking errors, the stations will appear away from their correct position on the frequency dial of the receiver. Practically it is not possible to keep a constant difference between Fo and Fs always. Hence some tracking error is always present. The tracking can be two point tracking or three point tracking. In two point tracking, the tracking error is zero at two points. But in three point tracking, the tracking error is zero at three points. One is i at high frequency end, the second is at the low frequency end and the third one will be in the middle of the tuning range.

Methods for tracking:

There are three methods used for tracking:

1. Padder tracking
2. Trimmer tracking
3. Three point tracking

Padder Tracking

In padder tracking the oscillator tunes below the frequency (it should be in midband). So the IF created is higher than it should be and a positive error is created. The figure shows the connection of tuned circuit for padder tracking and tracking error in padder tracking.

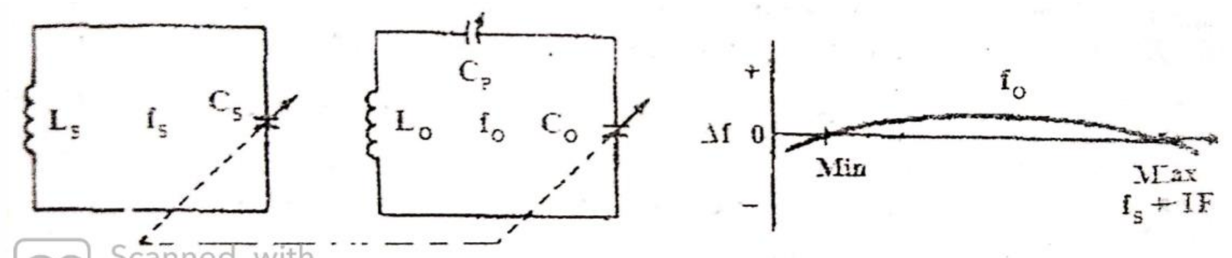


Figure 5.6 (a) Padder tracking circuit (b) Tracking error

Trimmer tracking

In trimmer tracking, the oscillator tunes higher the frequency (it should be in midband). So the *IF* created is less than it should be, and a negative error is created. The connection of tuned circuit for trimmer tracking, and the tracking error in trimmer tracking is shown in figure 5.7

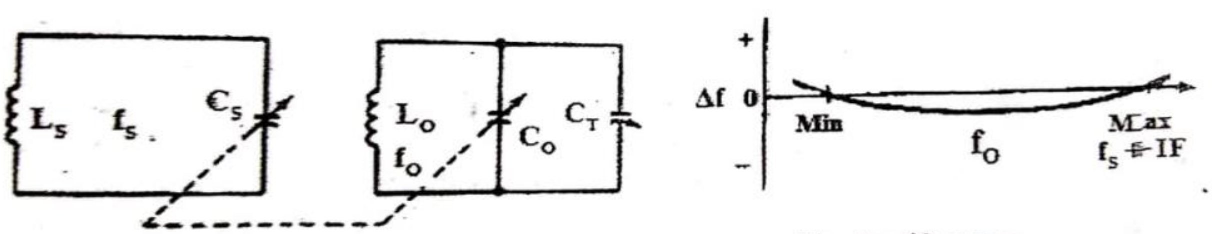


Figure 5.7 (a) Trimmer tracking circuit (b) Tracking error

DIGITAL PHASE MODULATION

Binary phase shift keying (BPSK) is the simplest form of digital phase modulation. For BPSK, each symbol consists of a single bit.

DIGITAL MODULATION

If the information signal is digital and the amplitude of the carrier is varied proportional to the information signal, a digitally modulated signal called *amplitude shift keying* (ASK) is produced.

If the frequency (f) is varied proportional to the information signal, *frequency shift keying* (FSK) is produced, and if the phase of the carrier (0) is varied proportional to the information signal, *phase shift keying (PSK) is* produced.

If both the amplitude and the phase are varied proportional to the information signal, *quadrature amplitude modulation* (QAM) results. ASK, FSK, PSK, and QAM are all forms of digital modulation:

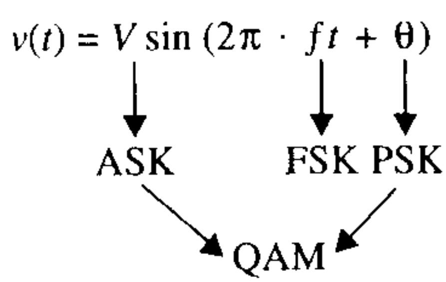


Figure : shows a simplified block diagram for a digital modulation system.

PHASE-SHIFT KEYING

Phase-shift keying (PSK) is a form of *angle-modulated, constant-amplitude* digital modulation.

BINARY PHASE-SHIFT KEYING

The simplest form of PSK is *binary phase-shift keying* (BPSK), where N = 1 and *M = 2. (Where* M = number of discrete signal or voltage levels ; *N* = number of bits necessary)

Therefore, with BPSK, two phases ($2^1 = 2$) are possible for the carrier.

One phase represents a logic 1, and the other phase represents a logic 0. As the input digital signal changes state (i.e., from a 1 to a 0 or from a 0 to a 1), the phase of the output carrier shifts between two angles that are separated by 180°.

Hence, other names for BPSK are *phase reversal keying* (PRK) and *biphase modulation.* BPSK is a form of square-wave modulation of a *continuous wave (CW)* signal.

DEFINITION

In BPSK ,the phase of the sinusoidal carrier is changed according to the data bit to be transmitted. Also a bipolar NRZ signal is used to represent the digital data coming from the digital source.

EXPRESSION FOR BPSK

In a BPSK , the binary symbols 0 & 1 modulate the phase of the carrier .Let us assume that the carrier as

$$S(t) = A\cos(2\pi f_c t)$$

Here A represents peak value of sinusoidal carrier.For the standard 1Ω load resistor ,the power dissipated would be

$$P = \frac{1}{2} A^2$$

$$A = \sqrt{2P}$$

Now when the symbol is changed , then the phase of the carrier will also be changed by an amount of 180 degrees(ie. Π radians).

Let us consider

For symbol 1 we have

$$s_1(t) = \sqrt{2P}\cos(2\pi f_c t)$$

If next symbol is 0 , we have

$$s_2(t) = \sqrt{2P}\cos(2\pi f_c t + \pi)$$

Now because $\cos(\theta+\pi) = -\cos\theta$, the last equation can be written as

$$s_2(t) = -\sqrt{2P}\cos(2\pi f_c t)$$

With above equation , we can define BPSK signal completely as

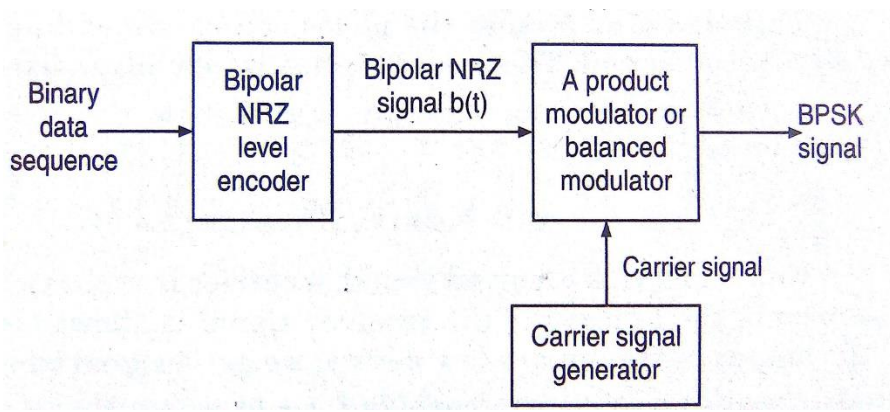
$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t)$$

$$Where\ \ b(t) = +1\ when\ binary\ 1\ is\ to\ be\ transmitted$$

-1 when binary 0 is to be transmitted.

GENERATION OF BPSK SIGNAL

BPSK signal can be generated by applying carrier signal to a balanced modulator. The binary data signal (0s and 1s) is converted into a NRZ bipolar signal by an NRZ encoder. Here the bipolar signal b(t) is applied as modulating signal to the balanced modulator.



Generation of BPSK

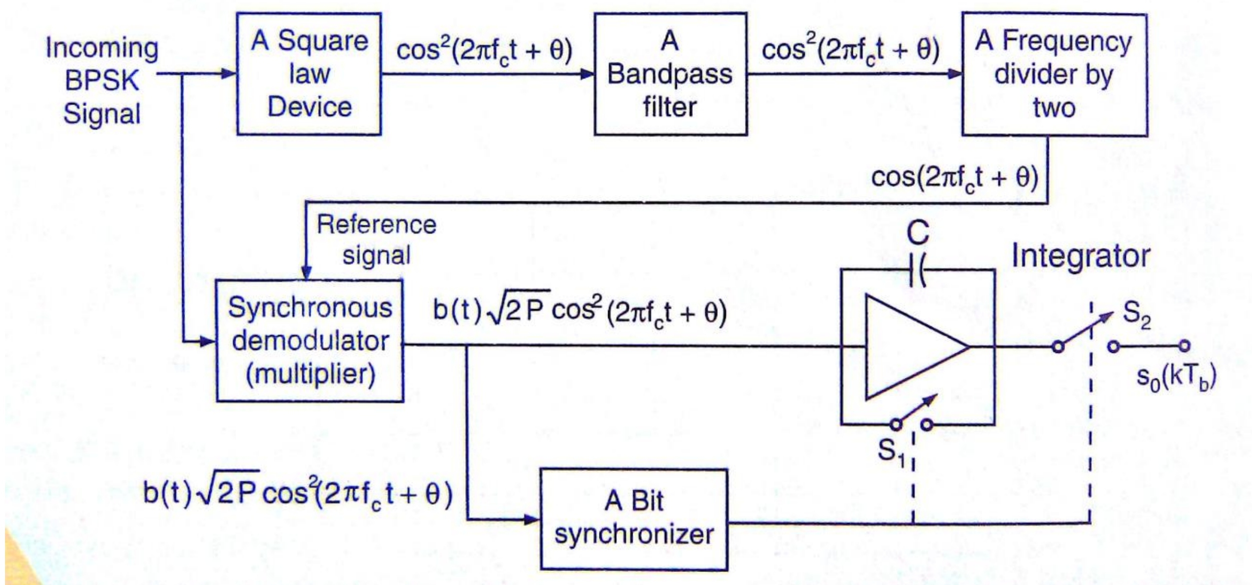
A NRZ level encoder converts binary data sequence into bipolar NRZ signal.

$$P = \frac{E_b}{T_b}\ ,where\ Eb\ is\ the\ signal\ energy\ and\ Tb\ is\ the\ bit\ duration$$

Also$\omega_c = 2\pi f_c$

RECEPTION OF BPSK SIGNAL; COHERENT DETECTION

Figure shows the BD and scheme to recover baseband signal from BPSK signal.



The transmitted BPSK signal is

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t)$$

This signal undergoes the phase change depending upon the time delay from transmitter end to receiver end . This phase change is usually a fixed phase shift in the transmitted signal.

Let us consider that phase shift is θ. Because of this the signal at the input of receiver is

$$s(t) = b(t)\sqrt{2P}\cos(2\pi f_c t + \theta)$$

Now from received signal a carrier is separated because this is coherent detection. This received signal is allowed to pass through square law device .At the output of square law device we get signal which is given as

$$cos^2(2\pi f_c t + \theta)$$

Here we have neglected the amplitude because we are only interested in the carrier signal.

We know that

$$cos^2\theta = \frac{1+\cos 2\theta}{2}$$

Therefore

$$cos^2(2\pi f_c t + \theta) = \frac{1 + \cos 2(2\pi f_c t + \theta)}{2} = \frac{1}{2} + \frac{1}{2}\cos 2(2\pi f_c t + \theta)$$

Here ½ represents DC level. This signal is then allowed to pass through a BPF whose passband is centered around 2fc. BPF removes the DC level of ½ and at the output we obtain

$$\cos 2(2\pi f_c t + \theta)$$

This signal having frequency 2fc.Hence it is passed through a frequency divider by two. Thus at the output of frequency divider, we get a carrier signal whose frequency is fc . ie. $\cos(2\pi f_c t + \theta)$

The coherent demodulator multiplies the input signal and recovered carrier. Hence at the output of multiplier we get

$$b(t)\sqrt{2P}\cos(2\pi f_c t + \theta) \times \cos(2\pi f_c t + \theta) = b(t)\sqrt{2P}\cos^2(2\pi f_c t + \theta)$$

$$= b(t)\sqrt{2P} \times \frac{1}{2}\,[1 + \cos 2(\,2\pi f_c t + \theta)$$

$$= b(t)\sqrt{\frac{P}{2}}\,[\,1 + \cos 2(2\pi f_c t + \theta)]$$

This signal is then applied to bit synchronizer and integrator. The integrator integrates the signal over one bit period. At the end of bit duration Tb bit synchronizer closes Switch S2 temperately.

This causes output of integrator to the decision device. Infect this is equivalent to sampling the output of integrator.The synchronizer then opens S and closes switch S1 temperately. This resets integrator voltage to zero. Integrator then integrates next bit .Let us assume one bit period Tb contains integral number of cycles of carrier. This means that the phase change occurs in the carrier only at zero crossing.

In the kth bit interval we can write output signal as

$$s_0(kTb) = b(kTb)\sqrt{\frac{P}{2}}\int_{(k-1)Tb}^{KTb} [cos2(2\pi f_c + \theta]dt$$

This integration gives the output of an interval for k^th^ bit.Hence integration is performed from (k-1)Tb to kTb. Tb is the one bit period.

We can write above equation as

$$s_0(kTb) = b(kTb)\sqrt{\frac{P}{2}}\left[\int_{(k-1)Tb}^{kTb} 1dt + \int_{(k-1)Tb}^{kTb} \cos 2(2\pi f_c t + \theta)dt\right]$$

Where $\int_{(k-1)Tb}^{kTb} \cos 2(2\pi f_c t + \theta)dt = 0$ since average value of sinusoidal wave form is zero if integration is done over full cycles. Hence we can write above equation as

$$s_0(kTb) = b(kTb)\sqrt{\frac{P}{2}}\int_{(k-1)Tb}^{kTb} 1dt = b(kTb)\sqrt{\frac{P}{2}}\ \{kTb - (k-1)Tb\} = b(kTb)\sqrt{\frac{P}{2}}\ Tb$$

The last equation shows that output of receiver depends on input.

$$s_0(kTb)\alpha\ b(kTb)$$

This signal is then applied to decision device which decides whether the transmitted symbol was zero or one.

SALIENT FEATURES OF BPSK

1. BPSK has bandwidth wich is lower than BFSK signal.
2. BPSK has best performance of all three digital modulation techniques in presence of noise. It yields minimum value of probability of error.
3. BPSK has good noise immunity.

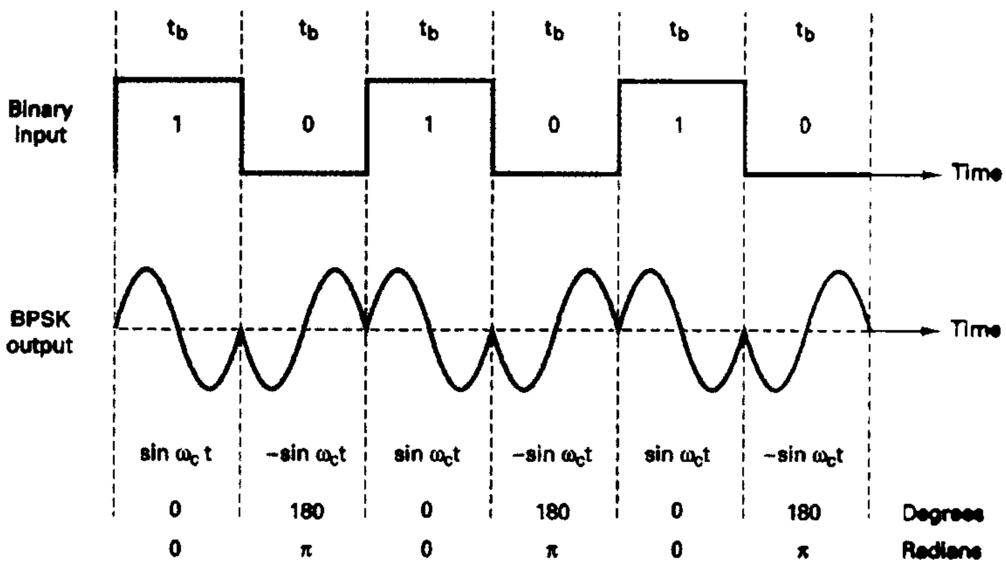


FIGURE : Output phase-versus-time relationship for a BPSK modulator

ANGLE MODULATOR CIRCUITS