Unit – I

Objectives:

- To introduce the concepts of modulation and AM modulation technique

Syllabus:

Introduction, Need for modulation, Amplitude Modulation- Definition, Time domain and frequency domain description, power relations in AM waves, Generation and detection of AM Waves.

Outcomes:

Students will be able to

- Understand need of modulation
- analyze AM system
- Determine power for amplitude modulation scheme.
- Understand generation and detection of AM modulated waves

Introduction

Communication is a process of conveying message at a distance. If the distance is involved is beyond the direct communication, the communication engineering comes into the picture. The branch of engineering which deals with communication systems is known as telecommunication engineering.

Telecommunication engineering is classified into two types based on Transmission media. They are:

- Line communication
- Radio communication

In Line communication the media of transmission is a pair of conductors called transmission line. In this technique signals are directly transmitted through the transmission lines. The installation and maintenance of a transmission line is not only costly and complex, but also overcrowds the open space.

In radio communication transmission media is open space or free space. In this technique signals are transmitted by using antenna through the free space in the form of EM waves.

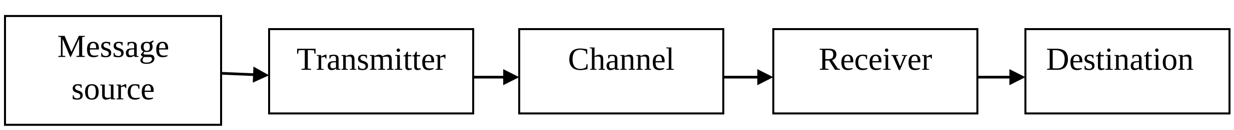


Fig.: Block diagram of Communication system

The communication system consists of three basic components.

- Transmitter
- Channel
- Receiver

Transmitter is the equipment which converts physical message, such as sound, words, pictures etc., into corresponding electrical signal.

Receiver is equipment which converts electrical signal back to the physical message.

Channel may be either transmission line or free space, which provides transmission path between transmitter and receiver.

Modulation

Modulation is defined as the process by which some characteristics (i.e. amplitude, frequency, and phase) of a carrier are varied in accordance with a modulating wave.

Demodulation

Demodulation is the reverse process of modulation, which is used to get back the original message signal. Modulation is performed at the transmitting end whereas demodulation is performed at the receiving end.

In analog modulation sinusoidal signal is used as carrier where as in digital modulation pulse train is used as carrier.

Need for modulation:

The reason why low frequency signals cannot be transmitted over long distances through space is listed below:

1. Short Operating Range – When a wave has a large frequency, the energy associated with it will also be large. Thus low frequency signals have less power that does not enable them to travel over long distances.
2. Poor Radiation Efficiency – The radiation efficiency becomes very poor for low frequency signals.
3. Mutual Interference – If all audio frequencies are send continuously from different sources, they would all get mixed up and cause erroneous interference air. If modulation is done, each signal will occupy different frequency levels and can be transmitted simultaneously without any error.
4. Huge Antenna Requirement – For a effective signal transmission, the sending and receiving antenna should be at least 1/4th of the wave length of the signal. Thus, for small frequencies, the antenna will have kilometres of length. But if the signal has the range of MegaHertz frequency, then the antenna size would be less. The carrier wave cannot be used alone for transmission purposes. Since its amplitude, frequency, and phase angle are constant with respect to some preference.

TYPES OF MODULATION

Continuous wave modulation (CW): When the carrier wave is continuous in nature the modulation process is known as continuous wave modulation. Pulse modulation: When the carrier wave is a pulse in nature the modulation process is known as continuous wave modulation Amplitude modulation (AM): A modulation process in which the amplitude of the carrier is varied in accordance with the instantaneous value of the modulating signal.

Amplitude Modulation- Time Domain and Frequency Representation

Amplitude modulation is defined as the process in which the amplitude of the carrier signal is varied in accordance with the modulating signal or message signal.

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

$$C(t) = A_c \cos(2\pi f_c t + \varphi)\, t$$

Where Ac = Amplitude of the carrier signal

f_c = frequency of the carrier signal

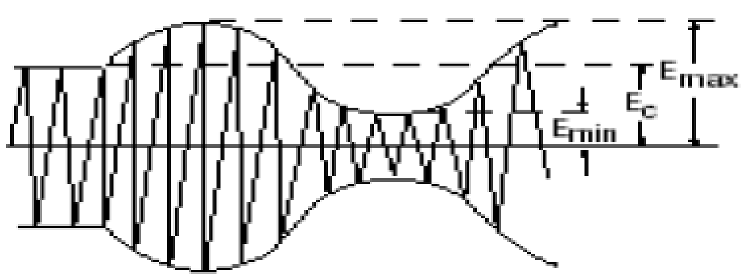
φ= Phase angle.

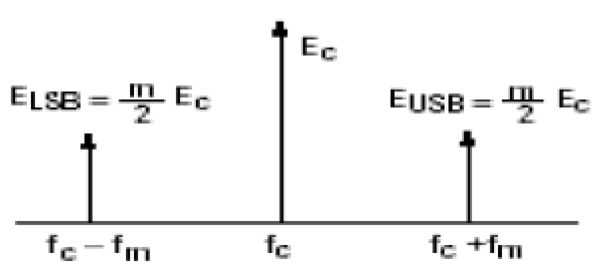
For our convenience, assume the phase angle of the carrier signal is zero. An amplitude-modulated (AM) wave S(t) can be described as function of time is given by

$$S(t) = A_c\,[1+k_a m(t)] \cos 2\pi f_c t$$

Where k_a = Amplitude sensitivity of the modulator.

The amplitude modulated (AM) signal consists of both modulated carrier signal and un modulated carrier signal.





There are two

requirements to maintain the envelope of AM signal is same as the shape of base band signal.

- The amplitude of the $k_a m(t)$ is always less than unity i.e., $|k_a m(t)|<1$ for all 't'
- The carrier signal frequency f_c is far greater than the highest frequency component *W* of the message signal m (t) i.e., $f_c >> W$

Assume the message signal m (t) is band limited to the interval $-W \leq f \leq W$ M (f)

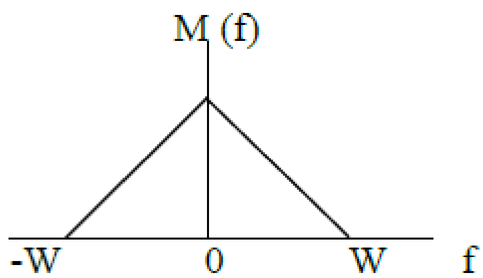


Fig. : Spectrum of message signal

The Fourier transform of AM signal S (t) is

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

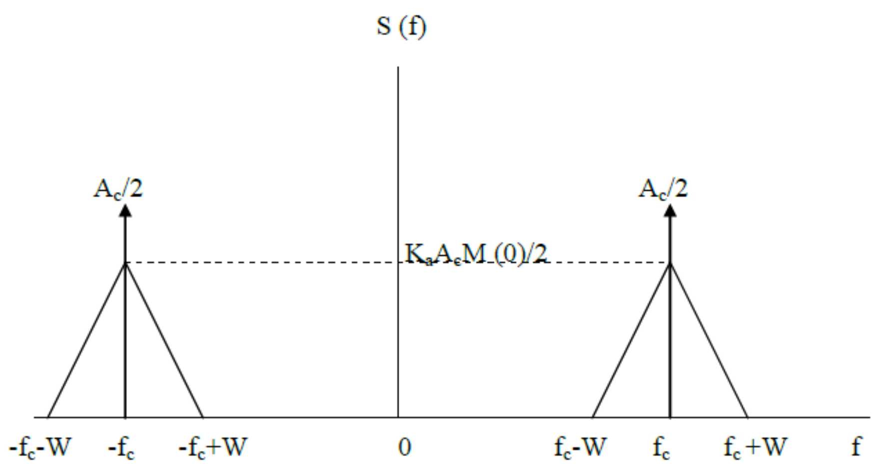


Fig : Spectrum of AM signal

The AM spectrum consists of two impulse functions which are located at f_c and $-f_c$ and weighted by $A_c/2$, two USBs, band of frequencies from f_c to f_c +W and band of frequencies from $-f_c$-W to $-f_c$, and two LSBs, band of frequencies from f_c-W to f_c and $-f_c$ to $-f_c$+W.

The difference between highest frequency component and lowest frequency component is known as transmission bandwidth. i.e., $B_T = 2W$

The envelope of AM signal is $A_c\,[1+k_a m(t)]$.

SINGLE TONE MODULATION

In single-tone modulation modulating signal consists of only one frequency component where as in multi-tone modulation modulating signal consists of more than one frequency component.

$$S(t) = A_c\,[1+k_a m(t)] \cos 2\pi f_c t \quad \ldots\ldots\ldots\text{(i)}$$

Let $m(t) = A_m \cos 2\pi f_m t$

Substitute m (t) in equation (i)

$$S(t) = A_c\,[1+k_a A_m \cos 2\pi f_m t] \cos 2\pi f_c t$$

Replace the term $k_a A_m$ by m which is known as modulation index or modulation factor.

Modulation index is defined as the ratio of amplitude of message signal to the amplitude of carrier signal. i.e.,

$$\mu = A_m/A_c$$

$$S(t) = A_c \cos(2\pi f_c t) + A_c\pi/2[\cos 2\pi(f_c+f_m)t] + A_c\pi/2[\cos 2\pi(f_c-f_m)t]$$

Fourier transform of S (t) is

$$S(f) = A_c/2[\pi(f-f_c) + \pi(f+f_c)] + A_c\pi/4[\pi(f-f_c-f_m) + \pi(f+f_c+f_m)]$$
$$+ A_c\pi/4[\pi(f-f_c+f_m) + \pi(f+f_c-f_m)]$$

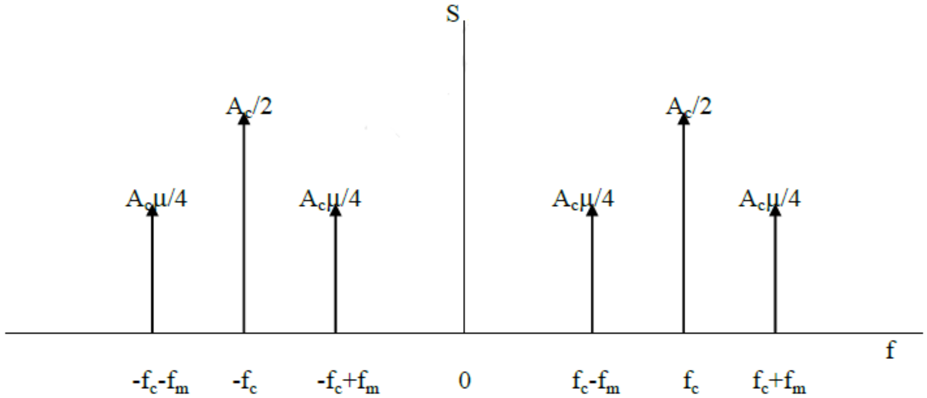


Fig. :Spectrum of Single tone AM signal

POWER RELATIONS IN AM WAVES

The standard time domain equation for single-tone AM signal is given by

$S(t) = A_c\cos(2\pi f_c t) + A_c m/2[\cos 2\pi(f_c+f_m)t] + A_c/2[\cos 2\pi(f_c-f_m)t]$

Power of any signal is equal to the mean square value of the signal

Carrier power $P_c = A_c^2/2$

Upper Side Band power $P_{USB} = A_c^2m^2/8$

Lower Side Band power $P_{LSB} = A_c^2m^2/8$

Total power $P_T = P_c + P_{LSB} + P_{USB}$

Total power $P_T = A_c^2/2 + A_c^2m^2/8 + A_c^2m^2/8$

$P_T = P_c[1+m^2/2]$

Transmission efficiency (η):-

Transmission efficiency is defined as the ratio of total side band power to the total transmitted power.

i.e., $P_T = P_{SB}/P_T$

Advantages of Amplitude modulation:-

- Generation and detection of AM signals are very easy
- It is very cheap to build, due to this reason it I most commonly used in AM radio broad casting

Disadvantages of Amplitude of modulation:-

- Amplitude modulation is wasteful of power
- Amplitude modulation is wasteful of band width

Application of Amplitude modulation: -

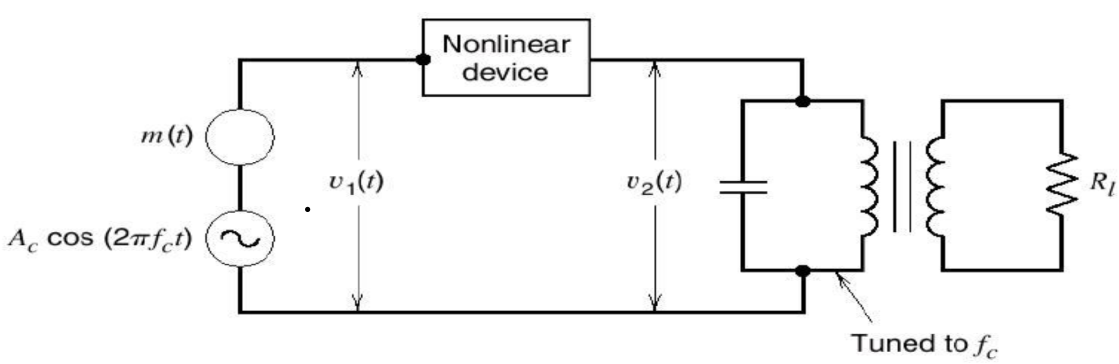
AM Radio Broadcasting

Generation of AM waves: Square-Law Modulator and Switching Modulator

There are two methods to generate AM waves

- Square-law modulator

Square-law modulator: -



A Square-law modulator requires three features: a means of summing the carrier and modulating waves, a nonlinear element, and a band pass filter for extracting the desired modulation products. Semi-conductor diodes and transistors are the most common nonlinear devices used for implementing square law modulators. The filtering requirement is usually satisfied by using a single or double tuned filters.

When a nonlinear element such as a diode is suitably biased and operated in a restricted portion of its characteristic curve, that is ,the signal applied to the diode is relatively weak, we find that transfer characteristic of diode-load resistor combination can be represented closely by a square law :

$V_0 (t) = a_1 V_i (t) + a_2 V_i^2(t)$ ……………….(i)

Where a_1, a2 are constants

Now, the input voltage V_i (t) is the sum of both carrier and message signals i.e.,

$V_i (t) = A_c \cos 2\pi f_c t + m (t)$ ……………. (ii)

Substitute equation (ii) in equation (i)we get

$V_0 (t) = a_1 A_c [1+k_a m (t)] \cos 2\pi f_c t + a_1 m (t) + a_2 A_c^2 \cos^2 2\pi f_c t + a_2 m^2 (t)$ ……….. (iii)

Where $k_a = 2a_2/a_1$

Now design the tuned filter /Band pass filter with center frequency f_c and pass band frequency width 2W.We can remove the unwanted terms by passing this output voltage $V_0(t)$ through the band pass filter and finally we will get required AM signal.

$$V_0 (t) = a_1 A_c [1+2a_2/a_1\, m (t)] \cos 2\pi f_c t$$

Assume the message signal m (t) is band limited to the interval $-W \le f \le W$

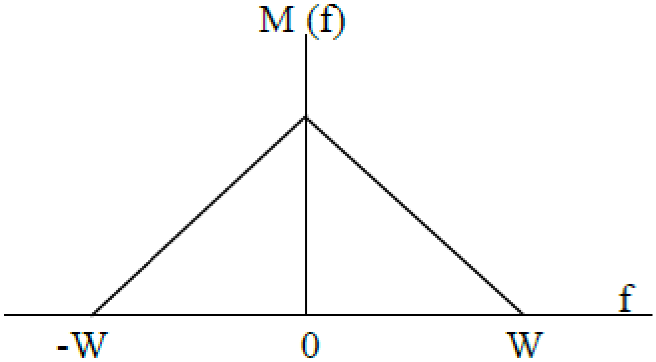


Fig .Spectrum of message signal

The Fourier transform of output voltage V_O (t) is given by

$V_O(f) = a_1 A_C/2[\delta(f-f_c) + \delta(f+f_c)] + a_2 A_C [M(f-f_c) + M(f+f_c)]$

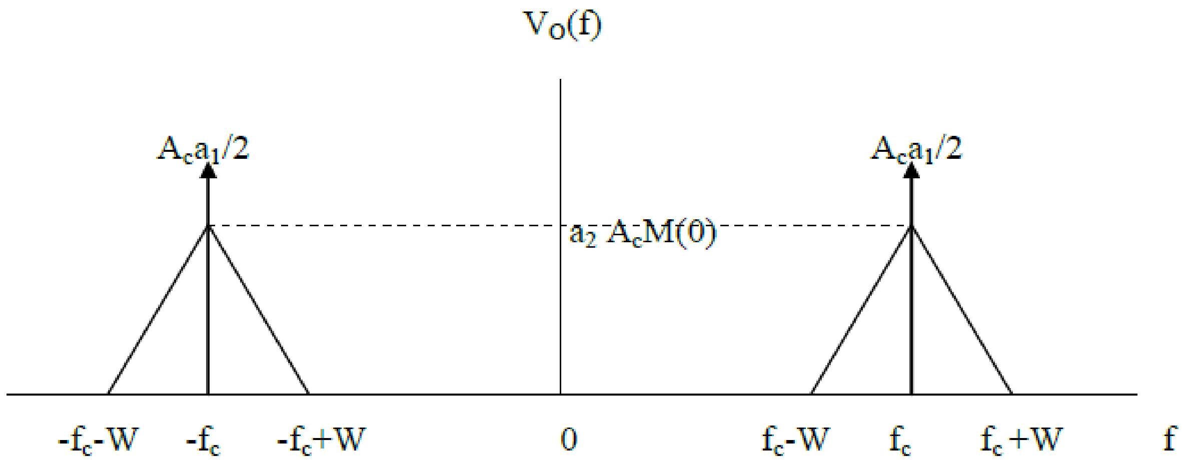


Fig: Spectrum of AM signal

The AM spectrum consists of two impulse functions which are located at f_c & $-f_c$ and weighted by $A_c a_1/2$ & $a_2 A_c/2$, two USBs, band of frequencies from f_c to $f_c + W$ and band of frequencies from $-f_c - W$ to $-f_c$, and two LSBs, band of frequencies from $f_c - W$ to f_c & $-f_c$ to $-f_c + W$.

Demodulation of AM waves

There are two methods to demodulate AM signals. They are:

- Square-law detector
- Envelope detector

Square-law detector:-

A Square-law modulator requires nonlinear element and a low pass filter for extracting the desired message signal. Semi-conductor diodes and transistors are the most common nonlinear devices used for implementing square law modulators. The filtering requirement is usually satisfied by using a single or double tuned filters.

When a nonlinear element such as a diode is suitably biased and operated in a restricted portion of its characteristic curve, that is ,the signal applied to the diode is relatively weak, we find that transfer characteristic of diode-load resistor combination can be represented closely by a square law :

$V_0(t) = a_1 V_i(t) + a_2 V_i^2(t)$ ……………….(i)

Where a_1, a2 are constants

Now, the input voltage V_i (t) is the sum of both carrier and message signals

i.e.,$V_i (t) = A_c [1+k_a m (t)] \cos 2\pi f_c t$ …………….(ii)

Substitute equation (ii) in equation (i) we get

$V_0 (t) = a_1 A_c [1+k_a m (t)] \cos 2\pi f_c t + 1/2\ a_2 A_c^2 [1+2 k_a m (t) + k_a^2 m^2 (t)] [\cos 4\pi f_c t]$……….(iii)

Now design the low pass filter with cutoff frequency f is equal to the required message signal bandwidth. We can remove the unwanted terms by passing this output voltage V_0 (t) through the low pass filter and finally we will get required message signal.

$V_0 (t) = A_c^2 a_2 m (t)$

The Fourier transform of output voltage V_O (t) is given by
$V_O (f) = A_c^2 a_2 M (f)$

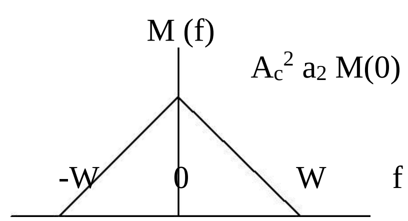


Fig: Spectrum of Output signal

Envelope Detector:

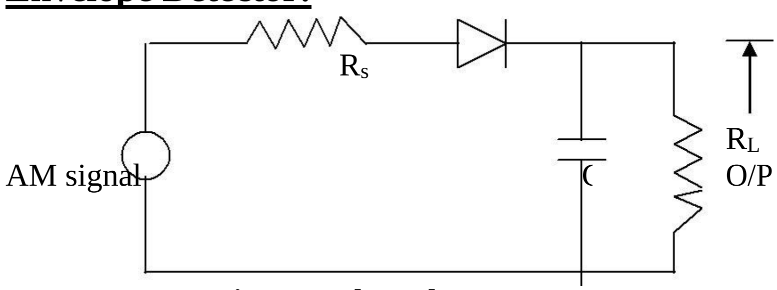


Fig.: Envelope detector

Envelope detector is used to detect high level modulated levels, whereas square-law detector is used to detect low level modulated signals (i.e., below 1v). The operation of the envelope detector is as follows. On a positive half cycle of the input signal, the diode is forward biased and the capacitor C charges up rapidly to the peak value of the input signal. When the input signal falls below this value, the diode becomes reverse biased and the capacitor C discharges slowly through the load resistor R_l . The discharging 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 charging time constant $R_s C$ is very small when compared to the carrier period $1/f_c$ i.e.,

$$R_sC << 1/f_c$$

Where R_s = internal resistance of the voltage source.
C = capacitor

i.e., the capacitor C charges rapidly to the peak value of the signal.

The discharging time constant R_lC is very large when compared to the charging time constant i.e.,

$1/f_c << R_lC << 1/W$, Where R_l = load resistance value

W = message signal bandwidth i.e., the capacitor discharges slowly through the load resistor.

Advantages

- It is very simple to design
- It is inexpensive
- Efficiency is very high when compared to Square Law detector

Disadvantage:

Due to large time constant, some distortion occurs which is known as diagonal clipping i.e., selection of time constant is somewhat difficult

Application:

It is most commonly used in almost all commercial AM Radio receivers.

Assignment-Cum-Tutorial Questions

A. Questions testing the remembering / understanding level

I) Objective Questions:

1. The maximum power efficiency of an AM modulator is

(a) 25% (b) 50% (c) 33% (d) 100%

2. Square law modulator is

a. Is used for AM generation
b. Consists of four diodes connected in the form of ring
c. Is a product modulator
d. Is used for DSB SC ganeration

3. Frequency components of an AM wave (m = modulation index) are

a.Carrier frequency (ω_c) with amplitude A
b. Upper side band ($\omega_c + \omega_m$) having amplitude mA/2
c. Lower side band ($\omega_c - \omega_m$) having amplitude mA/2
d. All of the above

4. The requirements of carrier frequency are

a) precise frequency value b) low frequency drift

c) both (a) and (b) d) neither (a) nor (b)

5. In an Amplitude Modulation
   a.Amplitude of the carrier varies constant b. Frequency of the carrier remains
   c. Phase of the carrier remains constant d. All of the above

6. If modulation index is greater than 1
   a.The baseband signal is not preserved in the envelope of the AM signal
   b. The recovered signal is distorted
   c. It is called over modulation
   d. All of the above

7. Consider sinusoidal modulation in an AM systems. Assuming no over modulation , the modulation index (μ) when the maximum and minimum values of the envelope, respectively, are 3V and 1V is _____________.

8. The most suitable method for detecting a modulated signal (6+5cosw$_m$t) cos(w$_c$t) is

   A. Envelope detector B. Synchronous detector

   C. Ratio detector D. Both A and B

9. AM demodulation techniques are
   a.Square law demodulator b. Envelope detector
   c. PLL detector d. Both a and b are correct

10. Suppose that the modulating signal is m(t)=2cos($2\pi fmt$) and the carrier signal is c(t)=ACcos($2\pi fct$).Which one of the following is a conventional AM signal without over-modulation?
    (a) (t)=$ACm(t)$cos($2\pi fct$) b) (t)=$AC[1+m(t)]$cos($2\pi fct$)
    (c) (t)=ACcos($2\pi fct$)+$AC4m(t)$cos($2\pi fct$)
    (d) (t)=ACcos($2\pi fmt$)cos($2\pi fct$)+ACsin($2\pi fmt$)sin($2\pi fct$)

II) Descriptive Questions

1. Defines modulation and explains need of modulation.
2. Represent single tone AM modulated wave in time and frequency domain.
3. Derive the total power in AM modulated signal.
4. Explain generation of square law AM modulated wave.
5. Explain the detection of the AM waves using envelope detector.
6. Explain generation of AM modulated wave using switching modulator.

B. Question testing the ability of students in applying the concepts.

I) Multiple choice Questions:

1. What is the carrier frequency in an AM wave when its highest frequency component is 850Hz and the bandwidth of the signal is 50Hz?

a. 800 Hz b. 695 Hz c. 625 Hz d. 825 Hz

2. Calculate the power in one of the side band in Am modulation when the carrier power is 124W and there is 80% modulation depth in the amplitude modulated signal.

a. 89.33W b. 64.85W c. 79.36W d. 102 W

3. A 1 MHz sinusoidal carrier is amplitude modulated by a symmetrical square wave of period 100 μsec. Which of the following frequencies will not be present in the modulated signal?

(a) 990 kHz (b) 1010 kHz

(c) 1020 kHz (d) 1030 kHz

4. A message signal given by $m(t)=(1/2)\cos\omega 1t-(1/2)\sin\omega 2t$ is amplitude modulated with a carrier of frequency ωcto generate $s(t)=[1+m(t)]\cos\omega ct$. What is the power efficiency achieved by this modulation scheme?

(a) 8.33% (b) 11.11% (c) 20% (d) 25%

5 Which of the following demodulator (s) can be used for demodulating the signal $s(t)=5(1+2\cos 200\ \pi t)cos20000\pi t$

(a) Envelope demodulator (b) Square-law demodulator

(c) Synchronous demodulator (d) None of the above

7. An AM signal is detected using an envelope detector. The carrier frequency and modulation signal frequency are 1 MHz and 2 KHz respectively. An appropriate value for the time constant of the envelope detector is

(a) 500 μsec (b) 20 μsec (c) 0.2 μsec (d) 1 μsec

8. A carrier is simultaneously modulated by two sine waves having modulation indices of 0.4 and 0.3. The total modulation index will be

A. 0.1 B. 0.7 C. 0.5 D. 0.35

II) Problems:

1. A 2000 Hz audio signal having amplitude if 15V modulates a 100KHz carrier which has a peak value of 25V when not modulated. Calculate

i) Modulation index

ii) Total power required for transmission

iii) What frequencies would show up in a spectrum analysis of modulated wave.

2. The output signal from an AM modulator is given by

$$s(t) = 5\cos(1800\pi t) + 20\cos(2000\pi t) + 5\cos(2200\pi t).$$

(i) Determine the message signal m (t) and the carrier c (t).
(ii) Determine the modulation index.
(iii) Determine the ratio of the power in the sidebands to the power in the carrier.

3. A carrier signal c (t) =5cos (2π106t) is modulated by message signal m (t) =cos (4π103t) to generate an AM signal with μ=1. Calculate bandwidth and total power.

4. An AM modulator is $s(t) = 25(1+0.7\cos 5000\pi t - 0.3\cos 10000\pi t)\cos 5\text{x}10^6\pi t$.
(i) Determine the amplitudes and frequencies of the carrier and sidebands
(ii) Determine the effective modulation index.
(iii) Determine the bandwidth.

5. The AM signal S(t) = (1+μcos(2πfct) is to be detected by envelope detection. What is the possible value of μ for which such detection possible? Sketch the envelop detector output ofr μ=2.

6. The signal s(t)=[1+0.2 cos(2π(fm/3)t] cos(2πfct) is demodulated using a square law demodulator having the characteristics $S_o(t)=[s(t)+2]^2$. The output is then filtered by an ideal LPF having cut off frequency at fm. determine the spectrum of the demodulated signal and plot it in the frequency range -fm ≤ f≤ fm.

7. Using the message signal $\mathbf{m(t)=1/(1+t^2)}$

Determine and sketch the modulated waves for the following methods of modulation:

a) Amplitude modulation with 50 percent modulation.
b) Double sideband- suppressed carrier modulation.

C) Previous GATE/IES Questions

1. A message signal m(t) = cos200pt + 4cos pt modulates the carrier c(t) = cos2πft where f = 1 M Hz to produce an AM signal. For demodulating the generated AM signal using an envelope detector, the time constant RC of the detector circuit should satisfy. ( )

Gate 2011

(A) 0.5 ms < RC < 1ms (B) 1μs << RC < 0.5 ms
(C) RC << μs (D) RC >> 0.5 ms

2. Consider sinusoidal modulation in an AM system. Assuming no over modulation, the modulation index (μ) when the maximum and minimum values of the envelope, respectively, are 3 V and 1 V, is ________.

Gate 2014

3. Consider the signal s(t)= m(t)cos(w_ct)+ m̂(t)cos(w_ct) where m̂ t denotes the Hilbert transform of m(t) and the bandwidth of m(t) is very small compared to w_c. The signal s(t) is a

(A) high-pass signal **Gate 2015**

(B) low-pass signal (C) band-pass signal

(D) double sideband suppressed carrier signal

4. Which of the following demodulator(s) can be used for demodulating the signal

S(t) = 5(1 + 2cos200 $\pi t) cos 20000\pi t$

(a) Envelope demodulator (b) Square-law demodulator

(c) Synchronous demodulator (d) None of the above

GATE 1993

5. The amplitude modulated wave form (t) = [1 + K(t)]cosωCt is fed to an ideal envelope detector. The maximum magnitude of (t) is greater than 1. Which of the following could be the detector output ?

(a) m(t) (b) Ac 2[1 + K(t)]2

(c) |AC[1 + Kam(t)]| (d) AC|1 + Kam(t)|2

GATE 2000

6. An AM signal is detected using an envelope detector. The carrier frequency and modulation signal frequency are 1 MHz and 2 KHz respectively. An appropriate value for the time constant of the envelope detector is

(a) 500 μsec (b) 20 μsec (c) 0.2 μsec (d) 1 μsec

GATE 2004

Unit – II
CONTINUOUS WAVE MODULATION – II

Objectives:

- To introduce the concepts of DSBSC and SSBSC modulation techniques and also to describe the effect of noise on analog modulated signals

Syllabus:

AM DSBSC Modulation - Time domain and frequency domain description, Generation of DSBSC Waves, Coherent detection of DSBSC Modulated waves, Costas loop.

AM SSBSC Modulation- Frequency domain description, Time domain description, Generation AM-SSB Modulated Waves-frequency discrimination, phase discrimination, Demodulation of SSB Waves, Noise in AM Systems, Comparison of AM techniques.

Outcomes:

Students will be able to

- Analyze AM DSB-SC systems
- Understand need of SSB modulation technique
- Determine power for various modulation schemes.
- Analyze the effect of noise in AM

DSBSC MODULATION

Double sideband-suppressed Carrier (DSB-SC) modulation, in which the transmitted wave consists of only the upper and lower sidebands. Transmitted power is saved through the suppression of the carrier wave, but the channel bandwidth requirement is same as in AM (i.e. twice the bandwidth of the message signal).

Basically, double sideband-suppressed (DSB-SC) modulation consists of the product of both the message signal m (t) and the carrier signal c(t),as follows:

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

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

The modulated signal s (t) undergoes a phase reversal whenever the message signal m (t) crosses zero. The envelope of a DSB-SC modulated signal is different from the message signal.

The transmission bandwidth required by DSB-SC modulation is the same as that for amplitude modulation which is twice the bandwidth of the message signal, 2W. Assume that the message signal is band-limited to the interval $-W \leq f \leq W$

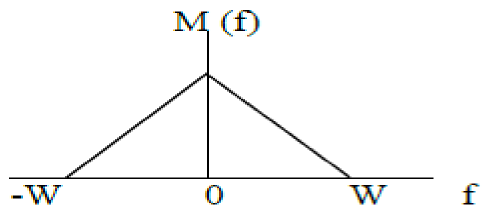


Fig: Spectrum of message signal

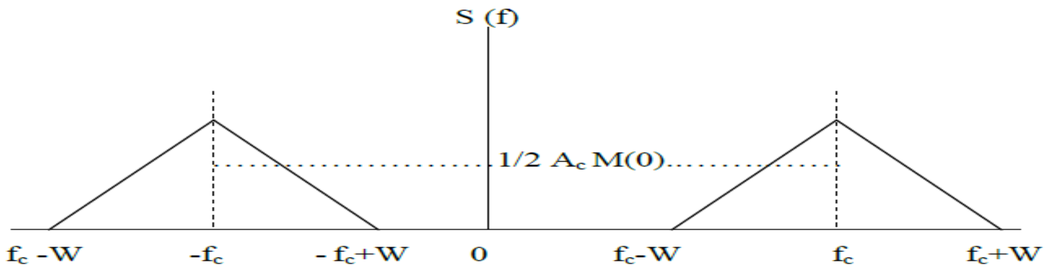


Fig: Spectrum of DSBSC wave

In single-tone modulation modulating signal consists of only one frequency component where as in multi-tone modulation modulating signal consists of more than one frequency components.

The standard time domain equation for the DSB-SC modulation is given by

$$S\ (t) = A_c \cos(2\pi fct)\ m\ (t) \ldots\ldots\ldots\ldots\ldots\ldots\ (1)$$

$$\text{Assume } m\ (t) = A_m \cos(2\pi fmt) \ldots\ldots\ldots\ldots\ldots\ldots\ (2)$$

Substitute equation (2) in equation (1) we will get

$$S\ (t) = A_c\ A_m \cos\ (2f_ct) \cos\ (2f_mt)$$

$$S\ (t) = A_c\ A_m/2[\cos 2\pi(f_c-f_m)t + \cos 2\pi(f_c+f_m)\ t] \ldots\ldots\ldots\ldots\ (3)$$

The Fourier transform of s (t) is

$$S\ (f) = A_c\ A_m/4[\delta\ (f-f_c-f_m) + \delta\ (f+f_c+f_m)] + A_c\ A_m/4[\delta\ (f-f_c+f_m) + \delta\ (f+f_c+f_m)]$$

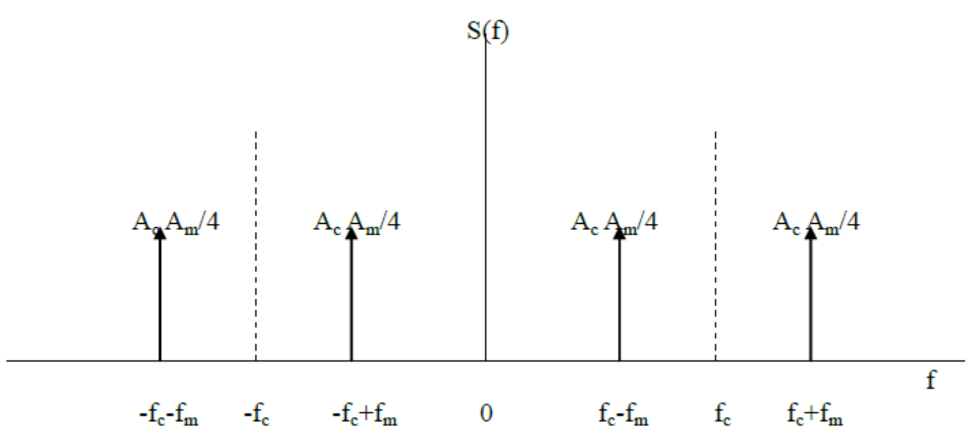


Fig: Spectrum of Single tone DSBSC wave

Generation of DSB-SC waves- Balanced Modulator and Ring modulator

There are two methods to generate DSB-SC waves. They are

- Balanced modulator
- Ring modulator

Balanced Modulator:-

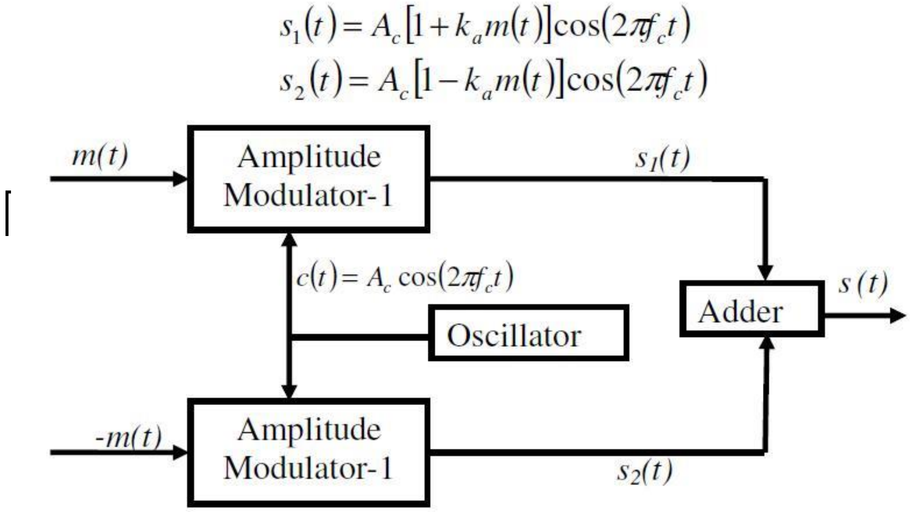


Fig: Balanced Modulator

One possible scheme for generating a DSBSC wave is to use two AM modulators arranged in a balanced configuration so as to suppress the carrier wave, as shown in above fig. Assume that two AM modulators are identical, except for the sign reversal of the modulating signal applied to the input of one of the modulators. Thus the outputs of the two AM modulators can be expressed as follows:

S_1 (t) = A_c [1+k_am (t)] cos 2πf_ct and

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

Subtracting S2 (t) from S_1 (t), we obtain

$$S (t) = S_1 (t) – S_2 (t)$$

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

Hence, except for the scaling factor 2k_a the balanced modulator output is equal to product of the modulating signal and the carrier signal

The Fourier transform of s (t) is

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

Assume that the message signal is band-limited to the interval $–W \le f \le W$

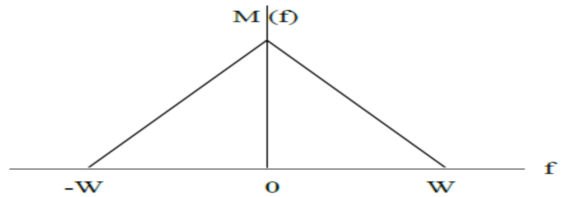


Fig: Spectrum of Baseband signal

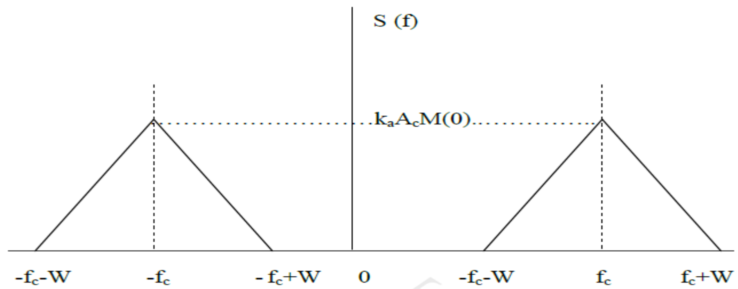


Fig: Spectrum of DSBSC wave

Ring modulator:-

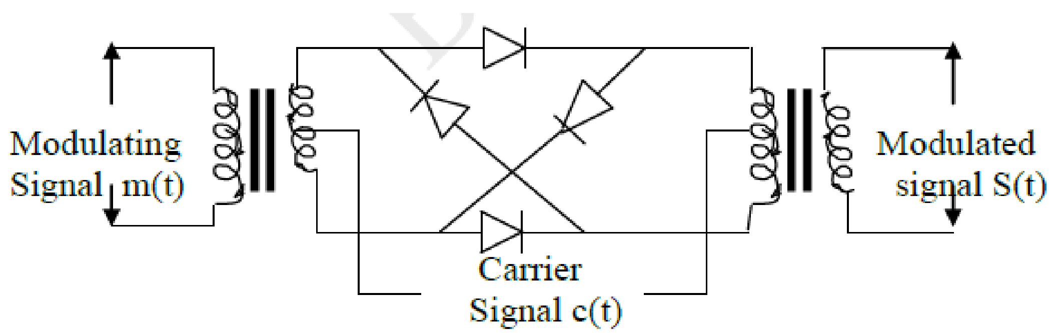


Fig: Ring modulator

One of the most useful product modulator, well suited for generating a DSB-SC wave, is the ring modulator shown in above figure. The four diodes form ring in which they all point in the same way-hence the name. The diodes are controlled by a square-wave carrier c (t) of frequency f_c, which applied longitudinally by means of to center-tapped transformers. If the transformers are perfectly balanced and the diodes are identical, there is no leakage of the modulation frequency into the modulator output. On one half-cycle of the carrier, the outer diodes are switched to their forward resistance r_f and the inner diodes are switched to their backward resistance r_b .on other half-cycle of the carrier wave, the diodes operate in the opposite condition.

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

$$c(t) = 4/\pi \sum_{n=1}^{\infty} (-1)^{n-1}/(2n-1) \cos[2\pi f_c t(2n-1)]$$

When the carrier supply is positive, the outer diodes are switched ON and the inner diodes are switched OFF, so that the modulator multiplies the message signal by +1

When the carrier supply is positive, the outer diodes are switched ON and the inner diodes are switched OFF, so that the modulator multiplies the message signal by +1.when the carrier supply is negative, the outer diodes are switched OFF and the inner diodes are switched ON, so that the modulator multiplies the message signal by -1.

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

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

$$S(t) = 4/\pi \sum_{n=1}^{\infty} (-1)^{n-1}/(2n-1) \cos[2\pi f_c t(2n-1)]\, m(t)$$

For n=1

$$S(t) = 4/\pi \cos(2\pi f_c t)\, m(t)$$

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

The Fourier transform of s (t) is $S(f) = 2/\pi\, [M(f-f_c) + M(f+f_c)]$

Coherent Detection of DSB-SC Waves:-

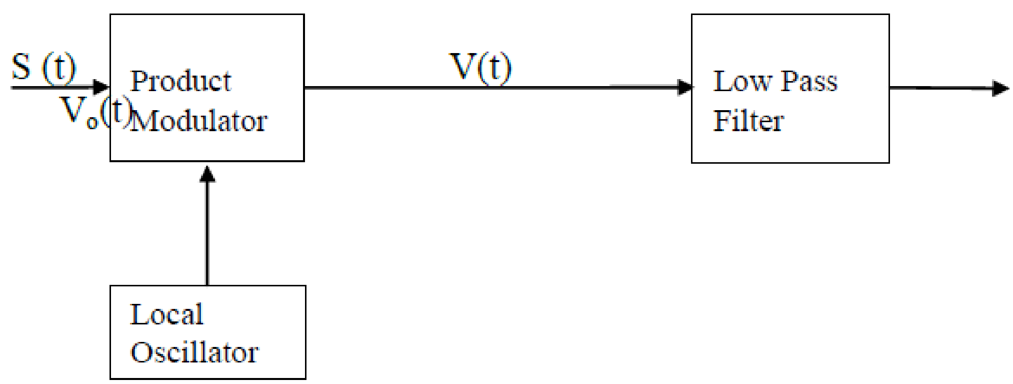


Fig: Coherent detection of DSBSC waves.

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

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

$$v(t) = \acute{A}_c \cos(2\pi f_c t + \phi)\, s(t)$$

$$v(t) = \acute{A}_c \cos(2\pi f_c t + \phi)\, A_c \cos 2\pi f_c t\, m(t)$$

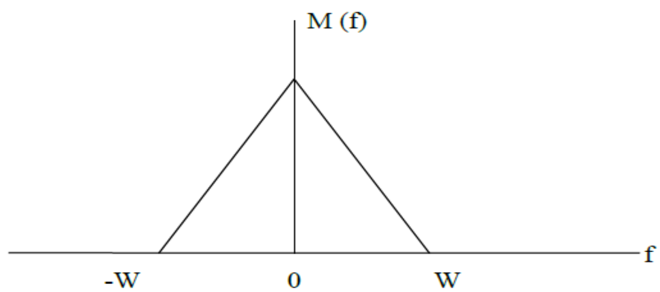
$$v(t) = A_c \acute{A}_c /2 \cos(2\pi f_c t + \phi)\, m(t) + A_c \acute{A}_c/2 \cos\phi\, m(t)$$

The high frequency can be eliminated by passing this output voltage to the Low Pass Filter. Now the Output Voltage at the Low pass Filter is given by

$$v_0(t) = A_c \acute{A}_c /2 \cos\phi\, m(t)$$

The Fourier transform of vo (t) is

$$V_O(f) = A_c \acute{A}_c /2 \cos\phi\, M(f)$$



The demodulated signal is proportional to the message signal m (t) when the phase error is constant. The Amplitude of this Demodulated signal is maximum when $\phi=0$, and it is minimum (zero) when $\phi=\pm\pi/2$ the zero demodulated signal, which occurs for $\phi=\pm\pi/2$ represents quadrature null effect of the coherent detector.

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<u>COSTA’S Loop</u>

Coherent detection of a DSB-SC modulated wave requires that the locally generated carrier in the receiver be synchronous in both frequency and phase with the oscillator responsible for generating the carrier in the transmitter. This is a rather demanding requirement, all the more so since the carrier is suppressed from the transmitted DSB-SC signal Ac m(t) cos (2πfct). One method of satisfying this requirement is to use the Costas receiver shown in Fig. This receiver consists of two coherent detectors supplied with the same input signal—namely, the incoming DSB-SC wave but with two local oscillator signals that are in phase quadrature with respect to each other.

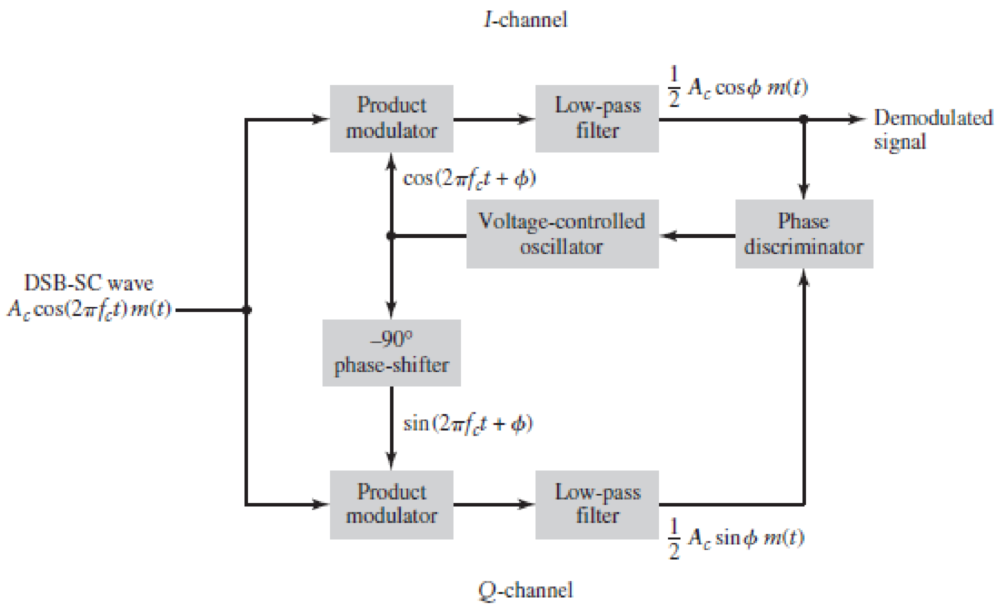


Fig: COSTA'S Loop

The detector in the upper path is referred to as the in phase coherent detector or I-channel, and the detector in the lower path is referred to as the quadrature-phase coherent detector or Q-channel. To understand the operation of this receiver, suppose that the local oscillator signal is of the same phase as the carrier wave used to generate the incoming DSBSC wave. Under these conditions, we find that the I-channel output contains the desired demodulated signal m(t).

From the discussion on coherent detection in Section, we know that the I-channel output is proportional to and for small hence, the I-channel output remains essentially unchanged so long as is small. But there will now be some signal, albeit small, appearing at the Q-channel output, which is proportional to for small This Q-channel output will have the same polarity as the I-channel output for one direction of local oscillator phase drift and the opposite polarity for the opposite direction of Thus, by combining the I- and Q-channel outputs in a phase discriminator (which consists of a multiplier followed by a time-averaging unit), a dc control signal proportional to the phase drift is generated. With negative feedback acting around the Costas receiver, the control signal tends to automatically correct for the local phase error in the voltage-controlled oscillator.

SINGLE SIDE BAND (SSB) Modulation Wave:-

In suppressing the carrier, DSB-SC modulation takes care of a major limitation of AM that pertains to the wastage of transmitted power. To take care of the other major limitation of AM that pertains to channel bandwidth, we need to suppress one of the two sidebands in the DSB-SC modulated wave. This modification of DSB-SC modulation is precisely what is done in single sideband (SSB) modulation. In effect, SSB modulation relies solely on the lower sideband or upper sideband to transmit the message signal across a communication channel. Depending on which particular sideband is actually transmitted, we speak of lower SSB or upper SSB modulation. Specifically, we start the study of SSB modulation by first considering the simple case of a sinusoidal modulating wave, and then we generalize the results to an arbitrary modulating signal in a step-by-step manner.

To proceed then, consider a DSB-SC modulator using the sinusoidal modulating wave

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

With the carrier $c(t) = A_c \cos(2\pi f_c t)$, the resulting DSB-SC modulated wave is defined by

$$\begin{aligned} S_{\mathrm{DSB}}(t) &= c(t)m(t) \\ &= A_c A_m \cos(2\pi f_c t)\cos(2\pi f_m t) \\ &= \frac{1}{2}A_c A_m \cos[2\pi(f_c + f_m)t] + \frac{1}{2}A_c A_m \cos[2\pi(f_c - f_m)t] \end{aligned} \tag{3.13}$$

which is characterized by two *side-frequencies*, one at $f_c + f_m$ and the other at $f_c - f_m$. Suppose that we would like to generate a sinusoidal SSB modulated wave that retains the upper side-frequency at $f_c + f_m$. Then, suppressing the second term in Eq. (3.13), we may express the upper SSB modulated wave as

$$S_{\mathrm{USSB}}(t) = \frac{1}{2}A_c A_m \cos[2\pi(f_c + f_m)t] \tag{3.14}$$

The cosine term in Eq. (3.14) includes the sum of two angles—namely, $2\pi f_c t$ and $2\pi f_m t$. Therefore, expanding the cosine term in Eq. (3.14) using a well-known trigonometric identity, we have

$$S_{\mathrm{USSB}}(t) = \frac{1}{2}A_c A_m \cos(2\pi f_c t)\cos(2\pi f_m t) - \frac{1}{2}A_c A_m \sin(2\pi f_c t)\sin(2\pi f_m t) \tag{3.15}$$

If, on the other hand, we were to retain the lower side-frequency at $f_c - f_m$ in the DSB-SC modulated wave of Eq. (3.13), then we would have a lower SSB modulated wave defined by

$$S_{\mathrm{LSSB}}(t) = \frac{1}{2}A_cA_m\cos(2\pi f_ct)\cos(2\pi f_mt) + \frac{1}{2}A_cA_m\sin(2\pi f_ct)\sin(2\pi f_mt) \qquad (3.16)$$

Examining Eqs. (3.15) and (3.16), we see that they differ from each other in only one respect: the minus sign in Eq. (3.15) is replaced with the plus sign in Eq. (3.16). Accordingly, we may combine these two equations and thereby define a sinusoidal SSB modulated wave as follows:

$$S_{\mathrm{SSB}}(t) = \frac{1}{2}A_cA_m\cos(2\pi f_ct)\cos(2\pi f_mt) \mp \frac{1}{2}A_cA_m\sin(2\pi f_ct)\sin(2\pi f_mt) \qquad (3.17)$$

where the plus sign applies to lower SSB and the minus sign applies to upper SSB.

the cosine term cos(2πfct) is replaced by the sine term sin(2πfct) and we may reformulate the SSB modulated wave

$$S_{\mathrm{SSB}}(t) = \frac{A_c}{2}m(t)\cos(2\pi f_ct) \mp \frac{A_c}{2}\hat{m}(t)\sin(2\pi f_ct)$$

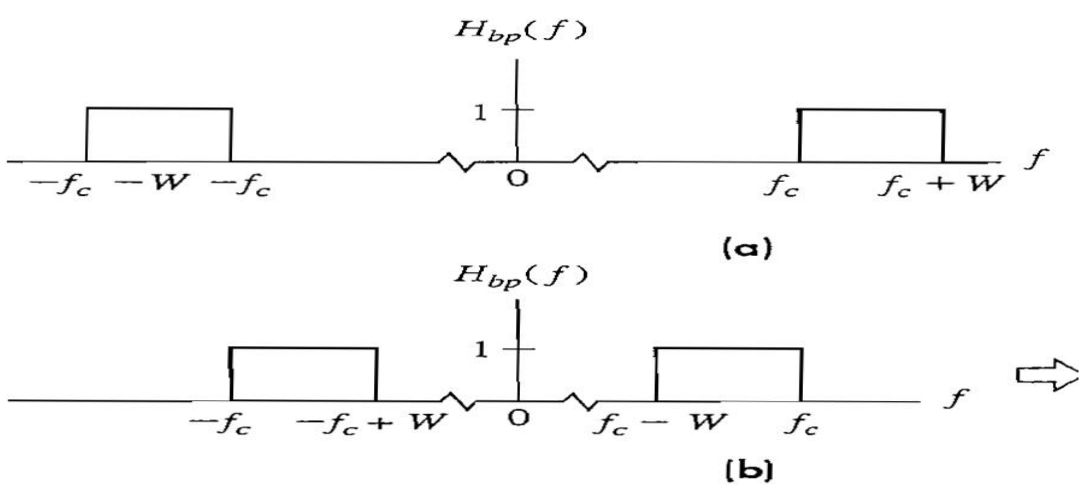
We observe that the periodic signal $m^{\wedge}$(t)can be derived from the periodic modulating signal m(t) simply by shifting the phase of each cosine term by -90^{o}

The signal $\hat{m}(t)$ is the Hilbert transform of the signal $m(t)$. Basically, a *Hilbert transformer* is a system whose transfer function is defined by

$$H(f) = -j\,\mathrm{sgn}(f)$$

$$s(t) = \frac{A_c}{2}m(t)\cos(2\pi f_ct) \mp \frac{A_c}{2}\hat{m}(t)\sin(2\pi f_ct) \qquad (3.23)$$

where $A_c\cos(2\pi f_ct)$ is the carrier, $A_c\sin(2\pi f_ct)$ is its $-90°$ phase-shifted version; the plus and minus signs apply to the lower SSB and upper SSB, respectively. In Eq. (3.23), we have omitted the use of SSB as a subscript for $s(t)$, with it being understood that this equation refers to SSB modulation in its most generic form.



(a) USSB; (b) LSSB.

Fig: SSB Frequency Spectrum

SSB Generations:-

1. Frequency Discrimination Method

The frequency-domain description presented for SSB modulation leads us naturally to the frequency discrimination method for generating as SSB modulated wave. Application of the method, however, requires that the message signal satisfy two conditions.

1. The message signal m(t) has little or no low-frequency content; that is, the message spectrum M(f) has "holes" at zero frequency. An important type of message signal with such a property is an audio signal (speech or music). In telephone, for example, the useful Frequency content of a speech signal is restricted to the band 0.3-3.4kHz, thereby creating an energy gap from zero to 300Hz

2. The highest frequency component W of the message signal m(t) is much less that the carrier frequency f_c. Then, under these conditions, the desired sideband will appear in a non-overlapping interval in the spectrum in such a way that it may be selected by an appropriate filter.

Thus an SSB modulator based on frequency discrimination consists basically of product modulator and a filter designed to pass the desired sideband of the DSBSC modulated wave at the product modulator output and reject the other sideband. A block diagram of this modulator is shown in Fig 1. The most severe requirement of this method of SSB generation usually arises from the unwanted sideband, the nearest frequency component of which is separated from the desired sideband by twice the lowest frequency component of the message signal. In designing the band-pass filter in the SSB modulation scheme must satisfy two basic requirements:

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.

We usually find that this kind of frequency discrimination can be satisfied only by using highly selective filters, which can be realized using crystal resonators with a Q factor per resonator in the range of 1000 to 2000.

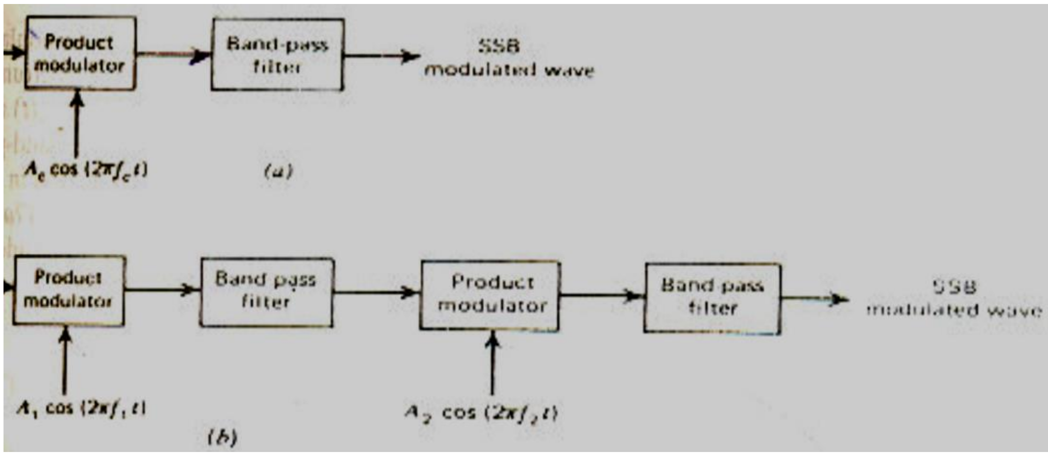


Fig: Block diagram of Frequency Discrimination Method

In the above figure, Frequency discrimination method of SSB modulated wave a) single stage b)two stage When it is necessary to generate an SSB modulated wave occupying a frequency band that is much higher than that of the message signal (e.g., translating a voice signal to the high-frequency region of the radio spectrum), it becomes very difficult to design an appropriate filter that will pass the desired sideband and reject the other using the simple arrangement of Fig.2. 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 Fig 1b involving two stages of modulation. The SSB modulated wave at the first filter output is used as the modulation 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_c2. The frequency separation between the sidebands of this DSBSC modulated wave is effectively twice the first carrier frequency f_c1, thereby permitting the second filter to remove the unwanted sideband.

2. Phase Discrimination Method

The second method for SSB generation, called the phase discrimination method, is depicted in Fig. its implementation follows from the time-domain description of SSB waves defined in Equations. This second SSB modulator consists of two parallel paths, one called the in-phase path and the other called the quadrature path. Each path involves a product modulator. The sinusoidal carrier waves applied to the two product modulators are in phase quadrature, which is taken care of by simply using a -90^0 phase-shifter as shown in Fig.3 However, it requires special attention is the wide-band phase-shifter, which is designed to produce the Hilbert transform m^(t) in response to the incoming message signal m(t). The role of the quadrature path embodying the wide-band phase shifter is merely to interfere with the in-phase path so as to eliminate power in one of the two sidebands, depending on whether upper SSB or lower SSB is the requirement. The two modulators are clearly quite different in their structures. In terms of design challenge, the band-pass filters in the frequency discriminator of stands out as the functional block that requires special attention. On the other hand, in the phase discriminator, it is the wide-band phase shifter that requires special attention.

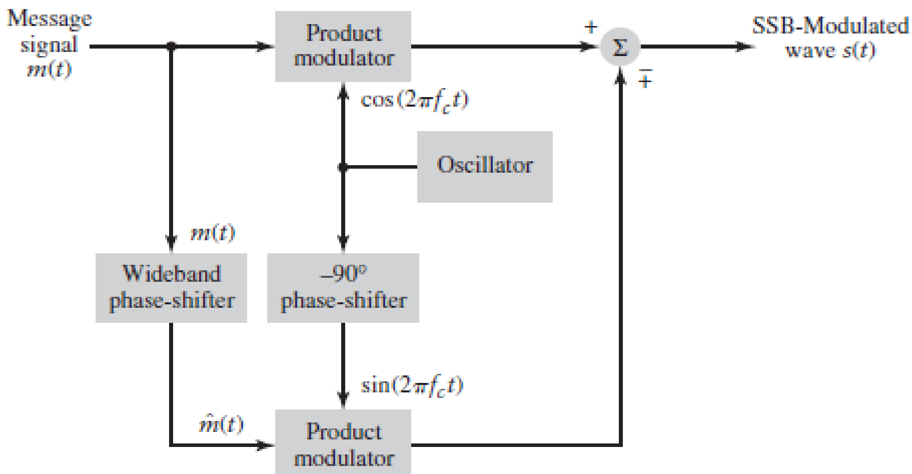


Fig: Phase Discrimination Method

Demodulation technique of SSB wave:-

Coherent Detection of SSB wave

The product modulator is a type of coherent SSB demodulator. To recover the modulating signal from the SSB-SC signal, we require a phase coherent or synchronous demodulator. The received SSB signal is first multiplied with a locally generated carrier signal. The locally generated carrier should have exactly the same frequency as that of the suppressed carrier. The product modulator multiplies the two signals at its input and the product signal is passed through a low pass filter with a bandwidth equal to f_m. At the output of the filter, we get the modulating signal back.

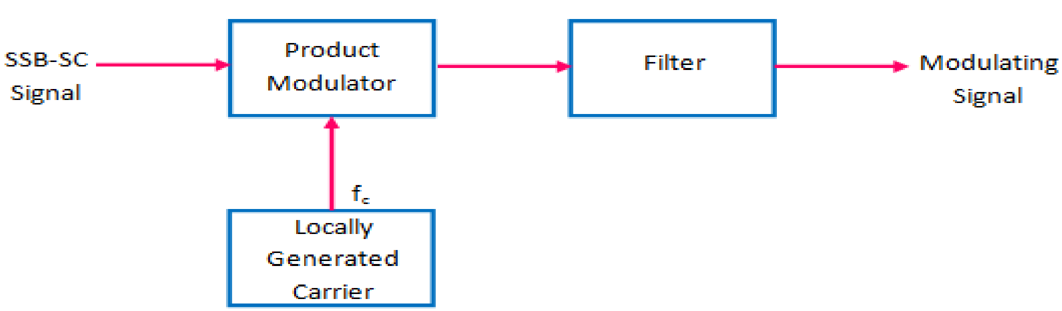


Fig: Block Diagram of coherent SSB demodulator

we find that the product modulator output is given by

$$\begin{aligned} v(t) &= \cos(2\pi f_c t)s(t) \\ &= \frac{1}{2}A_c \cos(2\pi f_c t)[m(t)\cos(2\pi f_c t)] \pm m(t)\sin(2\pi f_c t) \\ &= \underbrace{\frac{1}{4}A_c m(t)}_{\textit{Scaled message signal}} + \underbrace{\frac{1}{4}A_c [m(t)\cos(4\pi f_c t)] \pm m(t)\sin(4\pi f_c t)}_{\textit{Unwanted component}} \end{aligned}$$

The combination of the remaining terms represents an SSB modulated wave with a carrier frequency of 2fc; as such, it represents an unwanted component in the product modulator output that is removed by low-pass filtering. The detection of SSB modulated waves, just presented, assumes ideal conditions, namely, perfect synchronization between the local carrier and that in the transmitter both in frequency and phase. The effect of a phase error in the local generated carrier wave is to modify the detector output as follows.

$$v_o(t) = \frac{1}{4}A_c m(t)\cos\phi \pm \frac{1}{4}A_c m(t)\cos\phi$$

Where the plus sign applies to an incoming SSB modulated wave containing only the upper sideband and the minus sign applies to one containing only the lower sideband. Owing to the phase error, the detector output Vo(t) contains not only the message signal m(t) but also its Hilbert transform m(t). Consequently, the detector output suffers from phase distortion. This phase distortion is usually not serious with voice communications because the human ear is relatively in sensitive to phase distortion. The presence of phase distortion gives rise to what is called the Donald Duck voice effect. In the transmission of music and video signals, on the other hand, phase distortion in the form of a constant phase difference in all components can be intolerable.

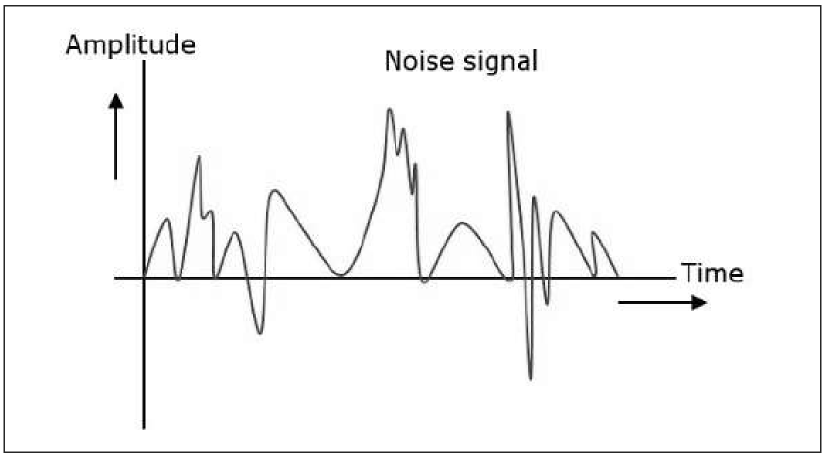
Noise:

In any communication system, during the transmission of the signal or while receiving the signal, some unwanted signal gets introduced into the communication, making it unpleasant for the receiver, and questioning the quality of the communication. Such a disturbance is called as **Noise**.

What is Noise?

Noise is an **unwanted signal**, which interferes with the original message signal and corrupts the parameters of the message signal. This alteration in the communication process, leads to the message getting altered. It most likely enters at the channel or the receiver.

The noise signal can be understood by taking a look at the following figure.



Hence, it is understood that the noise is some signal which has no pattern and no constant frequency or amplitude. It is quite random and unpredictable. Measures are usually taken to reduce it, though it can't be completely eliminated.

Most common examples of noise are:

- Hiss sound in radio receivers
- Buzz sound amidst of telephone conversations
- Flicker in television receivers, etc.

Types of Noise

The classification of noise is done depending on the type of the source, the effect it shows or the relation it has with the receiver, etc.

There are two main ways in which noise is produced. One is through some **external source** while the other is created by an **internal source**, within the receiver section.

External Source

This noise is produced by the external sources, which may occur in the medium or channel of communication usually. This noise cannot be completely eliminated. The best way is to avoid the noise from affecting the signal.

Examples

Most common examples of this type of noise are

- Atmospheric noise (due to irregularities in the atmosphere)
- Extra-terrestrial noise, such as solar noise and cosmic noise
- Industrial noise

Internal Source

This noise is produced by the receiver components while functioning. The components in the circuits, due to continuous functioning, may produce few types of noise. This noise is quantifiable. A proper receiver design may lower the effect of this internal noise.

Examples

Most common examples of this type of noise are

- Thermal agitation noise (Johnson noise or Electrical noise)
- Shot noise (due to the random movement of electrons and holes)
- Transit-time noise (during transition)
- Miscellaneous noise is another type of noise which includes flicker, resistance effect and mixer generated noise, etc.

Effects of Noise

Noise is an inconvenient feature, which affects the system performance. Following are the effects of noise.

Noise limits the operating range of the systems.

Noise indirectly places a limit on the weakest signal that can be amplified by an amplifier. The oscillator in the mixer circuit may limit its frequency because of noise. A system's operation depends on the operation of its circuits. Noise limits the smallest signal that a receiver is capable of processing.

Noise affects the sensitivity of receivers

Sensitivity is the minimum amount of input signal necessary to obtain the specified quality output. Noise affects the sensitivity of a receiver system, which eventually affects the output.

In this chapter, let us calculate Signal to Noise Ratios and Figure of Merits of various modulated waves, which are demodulated at the receiver.

Signal to Noise Ratio

Signal-to-Noise Ratio (SNR) is the ratio of the signal power to noise power. The higher the value of SNR, the greater will be the quality of the received output.

Signal-to-Noise Ratio at different points can be calculated using the following formulas.

$$\textbf{Input SNR} = (\mathrm{SNR})_I = \frac{\textbf{Average power of modulating signal}}{\textbf{Average power of noise at input}}$$

$$\textbf{Output SNR} = (\mathrm{SNR})_O = \frac{\textbf{Average power of demodulated signal}}{\textbf{Average power of noise at output}}$$

$$\textbf{Channel SNR} = (\mathrm{SNR})_C = \frac{\textbf{Average power of modulated signal}}{\textbf{Average power of noise in message bandwidth}}$$

Figure of Merit

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

$$\mathbf{F} = \frac{(\mathrm{SNR})_O}{(\mathrm{SNR})_I}$$

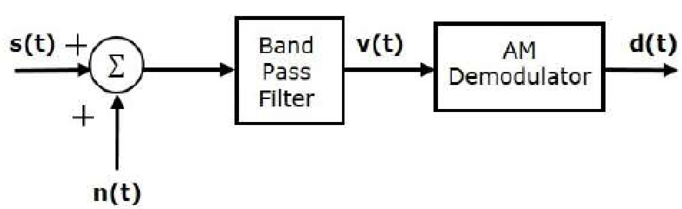
Figure of merit of a receiver is

$$\mathbf{F} = \frac{(\mathrm{SNR})_O}{(\mathrm{SNR})_c}$$

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

SNR Calculations in AM System

Consider the following receiver model of AM system to analyze noise.



We know that the Amplitude Modulated (AM) wave is

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

$$=> s(t) = A_c\cos(2\pi f_c t) + A_c k_a\,\mathrm{m}(t)\cos(2\pi f_c t)$$

Average power of AM wave is

$$P_s = \left(\frac{A_c}{\sqrt{2}}\right)^2 + \left(\frac{A_c k_a m(t)}{\sqrt{2}}\right)^2 = \frac{A_c{}^2}{2} + \frac{A_c{}^2 k_a{}^2 P}{2}$$

$$=> P_s = \frac{A_c{}^2\left(1 + k_a{}^2 P\right)}{2}$$

Average power of noise in the message bandwidth is

$$P_{nc} = WN_O$$

Substitute, these values in **channel SNR** formula.

$$(SNR)_{C,AM} = \frac{\text{Average power of AM wave}}{\text{Average power of noise in message bandwidth}}$$

$$=> (SNR)_{C,AM} = \frac{A_c{}^2\left(1 + k_a{}^2 P\right)}{2WN_O}$$

Where,

- **P** is the power of the message signal $= \frac{A_m{}^2}{2}$
- **W** is the message bandwidth

Assume the band pass noise is mixed with AM wave in the channel as shown in the above figure. This combination is applied at the input of AM demodulator. Hence, the input of AM demodulator is

$$v(t) = s(t) + n(t)$$

$$=> v(t) = A_c[1 + k_a m(t)]cos(2\pi f_c t) + \left[n_I(t)cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)\right]$$

$$=> v(t) = [A_c + A_c k_a m(t) + n_I(t)]cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)$$

Where $n_I(t)$ and $n_Q(t)$ are in phase and quadrature phase components of noise.

The output of AM demodulator is nothing but the envelope of the above signal.

$$d(t) = \sqrt{[A_c + A_c k_a m(t) + n_I(t)]^2 + \left(n_Q(t)\right)^2}$$

$$=> d(t) \approx A_c + A_c k_a m(t) + n_I(t)$$

Average power of the demodulated signal is

$$P_m = \left(\frac{A_c k_a m(t)}{\sqrt{2}}\right)^2 = \frac{A_c{}^2 k_a{}^2 P}{2}$$

Average power of noise at the output is

$$P_{no} = WN_O$$

Substitute, these values in **output SNR** formula.

$$(SNR)_{O,AM} = \frac{\textbf{Average power of demodulated signal}}{\textbf{Average power of noise at output}}$$

$$=> (SNR)_{O,AM} = \frac{A_c{}^2 k_a{}^2 P}{2WN_O}$$

Substitute, the values in **Figure of merit** of AM receiver formula

$$F = \frac{(SNR)_{O,AM}}{(SNR)_{c,AM}}$$

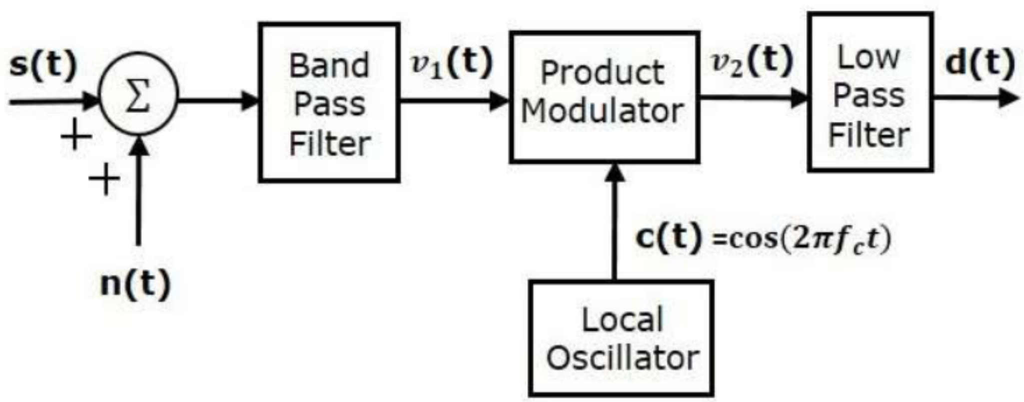
$$=> F = \left(\frac{A_c{}^2 k_a{}^2 P}{2WN_O}\right) \Big/ \left(\frac{A_c{}^2(1 + k_a{}^2 P)}{2WN_O}\right)$$

$$=> F = \frac{K_a^2 P}{1 + K_a^2 P}$$

Therefore, the Figure of merit of AM receiver is less than one.

SNR Calculations in DSBSC System

Consider the following receiver model of DSBSC system to analyze noise.



We know that the DSBSC modulated wave is

$$s(t) = A_c\,\mathrm{m}(t)\cos(2\pi f_c t)$$

Average power of DSBSC modulated wave is

$$P_s = \left(\frac{A_c m(t)}{\sqrt{2}}\right)^2 = \frac{A_c{}^2 P}{2}$$

Average power of noise in the message bandwidth is

$$P_{nc} = WN_O$$

Substitute, these values in **channel SNR** formula.

$$(SNR)_{C,DSBSC} = \frac{\textbf{Average power of DSBSC modulated wave}}{\textbf{Average power of noise in message bandwidth}}$$

$$=> (SNR)_{C,DSBSC} = \frac{A_c{}^2 P}{2WN_O}$$

Assume the band pass noise is mixed with DSBSC modulated wave in the channel as shown in the above figure. This combination is applied as one of the input to the product modulator. Hence, the input of this product modulator is

$$v_1(t) = s(t) + n(t)$$

$$=> v_1(t) = A_c m(t)cos(2\pi f_c t) + \left[n_I(t)cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)\right]$$

$$=> v_1(t) = [A_c m(t) + n_I(t)]cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)$$

Local oscillator generates the carrier signal $c(t) = cos(2\pi f_c t)$. This signal is applied as another input to the product modulator. Therefore, the product modulator produces an output, which is the product of $v_1(t)$ and $c(t)$**.**

$$v_2(t) = v_1(t)c(t)$$

Substitute, $v_1(t)$ and $c(t)$ values in the above equation.

$$=> v_2(t) = \left([A_c m(t) + n_I(t)]cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)\right)\ cos(2\pi f_c t)$$

$$=> v_2(t) = [A_c m(t) + n_I(t)]\ cos^2(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)cos(2\pi f_c t)$$

$$=> v_2(t) = [A_c m(t) + n_I(t)]\left(\frac{1 + cos(4\pi f_c t)}{2}\right) - n_Q(t)\frac{sin(4\pi f_c t)}{2}$$

When the above signal is applied as an input to low pass filter, we will get the output of low pass filter as

$$d(t) = \frac{[A_c m(t) + n_I(t)]}{2}$$

Average power of the demodulated signal is

$$P_m = \left(\frac{A_c m(t)}{2\sqrt{2}}\right)^2 = \frac{A_c{}^2 P}{8}$$

Average power of noise at the output is

$$P_{no} = \frac{WN_O}{4}$$

Substitute, these values in **output SNR** formula.

$$(SNR)_{O,DSBSC} = \frac{\textbf{Average power of demodulated signal}}{\textbf{Average power of noise at output}}$$

$$=> (SNR)_{O,DSBSC} = \left(\frac{A_c{}^2 P}{8}\right) \bigg/ \left(\frac{WN_O}{4}\right) = \frac{A_c{}^2 P}{2WN_O}$$

Substitute, the values in **Figure of merit** of DSBSC receiver formula

$$F = \frac{(SNR)_{O,DSBSC}}{(SNR)_{c,DSBSC}}$$

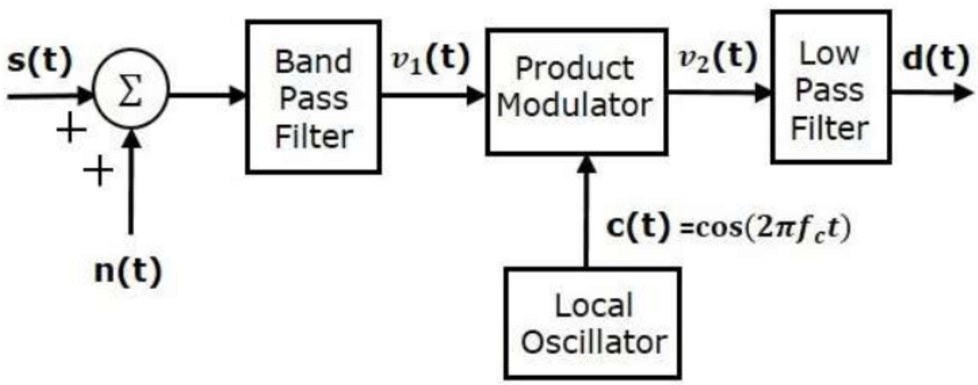
$$=> F = \left(\frac{A_c{}^2 P}{2WN_O}\right) \bigg/ \left(\frac{A_c{}^2 P}{2WN_O}\right)$$

$$=> F = 1$$

Therefore, the Figure of merit of DSBSC receiver is 1.

SNR Calculations in SSBSC System

Consider the following receiver model of SSBSC system to analyze noise.



We know that the SSBSC modulated wave having lower sideband is

$$s(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t]$$

Average power of SSBSC modulated wave is

$$P_s = \left(\frac{A_m A_c}{2\sqrt{2}}\right)^2 = \frac{A_m{}^2 A_c{}^2}{8}$$

Average power of noise in the message bandwidth is

$$P_{nc} = WN_O$$

Substitute, these values in **channel SNR** formula.

$$(SNR)_{C,SSBSC} = \frac{\text{Average power of SSBSC modulated wave}}{\text{Average power of noise in message bandwidth}}$$

$$=> (SNR)_{C,SSBSC} = \frac{A_m{}^{\sim} A_c{}^{\sim}}{8WN_0}$$

Assume the band pass noise is mixed with SSBSC modulated wave in the channel as showr in the above figure. This combination is applied as one of the input to the producl modulator. Hence, the input of this product modulator is

$$v_1(t) = s(t) + n(t)$$

$$=> v_1(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t] + n_I(t)cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)$$

The local oscillator generates the carrier signal $c(t) = cos(2\pi f_c t)$. This signal is applied as another input to the product modulator. Therefore, the product modulator produces ar output, which is the product of $v_1(t)$ and $c(t)$.

$$v_2(t) = v_1(t)c(t)$$

Substitute, $v_1(t)$ and $c(t)$ values in the above equation.

$$=> v_2(t) = \left(\frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t] + n_I(t)cos(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)\right) cos(2\pi f_c t)$$

$$=> v_2(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t] \, cos(2\pi f_c t) + n_I(t)cos^2(2\pi f_c t) - n_Q(t)sin(2\pi f_c t)cos(2\pi f_c t)$$

$$=> v_2(t) = \frac{A_m A_c}{4} \{\cos[2\pi(2f_c - f_m)t] + \cos(2\pi f_m t)\} + n_I(t)\left(\frac{1 + cos(4\pi f_c t)}{2}\right) - n_Q(t)\frac{sin(4\pi f_c t)}{2}$$

When the above signal is applied as an input to low pass filter, we will get the output of low pass filter as

$$d(t) = \frac{A_m A_c}{4} \cos(2\pi f_m t) + \frac{n_I(t)}{2}$$

Average power of the demodulated signal is

$$P_m = \left(\frac{A_m A_c}{4\sqrt{2}}\right)^2 = \frac{A_m{}^2 A_c{}^2}{32}$$

Average power of noise at the output is

$$P_{no} = \frac{WN_0}{4}$$

Substitute, these values in **output SNR** formula

$$(SNR)_{O,SSBSC} = \frac{\text{Average power of demodulated signal}}{\text{Average power of noise at output}}$$

$$=> (SNR)_{O,SSBSC} = \left(\frac{A_m{}^2 A_c{}^2}{32}\right) \Big/ \left(\frac{WN_0}{4}\right) = \frac{A_m{}^2 A_c{}^2}{8WN_0}$$

Substitute, the values in **Figure of merit** of SSBSC receiver formula

$$F = \frac{(SNR)_{O,SSBSC}}{(SNR)_{C,SSBSC}}$$

$$=> \boldsymbol{F} = \left(\frac{\boldsymbol{A_m}^2 \boldsymbol{A_c}^2}{\boldsymbol{8WN_0}}\right) \Big/ \left(\frac{\boldsymbol{A_m}^2 \boldsymbol{A_c}^2}{\boldsymbol{8WN_0}}\right)$$

$$=> \boldsymbol{F} = \mathbf{1}$$

Therefore, the Figure of merit of SSBSC receiver is 1.

Comparison of AM techniques:-

SI.No:	Parameter	DSB-FC (AM)	DSB-SC	SSB-SC
1	Power	High	Medium	Less
2	Band Width	$2f_m$	$2f_m$	f_m
3	Carrier Suppression	No	Yes	Yes
4	Side band Transmission	No	No	One Side band Completely
5	Transmission Efficiency	Minimum	Moderate	Maximum
6	Receiver Complexity	Simple	Complex	Complex
7	Modulation Type	Non-Linear	Linear	Linear

Unit – III

Objectives:

- To introduce the concepts of AM Transmission and multiplexing

UNIT – III AM Transmitters

Classification of Transmitters, AM Transmitters: high level and low level AM transmitters, Frequency division multiplexing and Time division multiplexing

Outcomes:

Students will be able to

- understand various blocks in AM transmitters
- Know the difference between the TDM and FDM

AM TRANSMITTERS

CLASSIFICATIONOF TRANSMITTERS

Transmitters are mainly classified into two types

- Low level transmitters
- High level transmitters

AM TRANSMITTERS

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:

1. High Level
2. Low Level

HIGH LEVEL TRANSMITTER

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. High-Level and Low-Level Transmitters Below figure 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 signal.

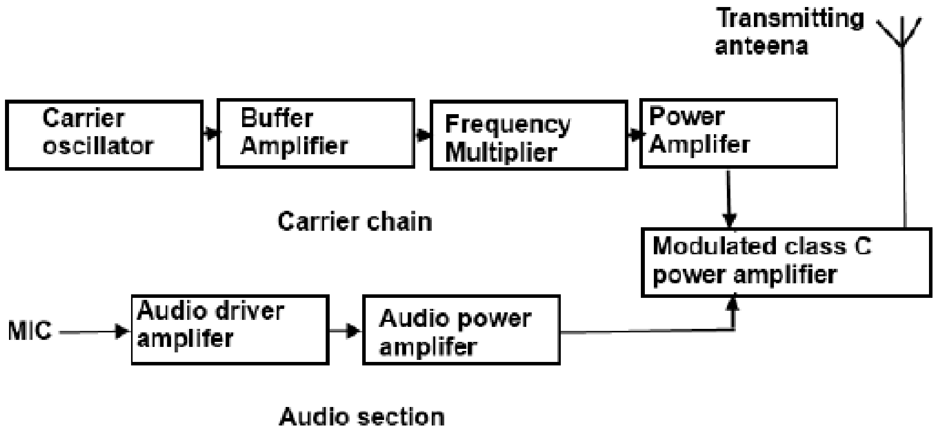


Fig. Block diagram of High level AM transmitter

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 are:

1. Carrier oscillator.
2. Buffer amplifier.
3. Frequency multiplier.
4. Power amplifier
5. Audio chain.
6. 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 requirements.

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. 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. This radiates the signal into space of transmission.

<u>LOW LEVEL TRANSMITTER</u>

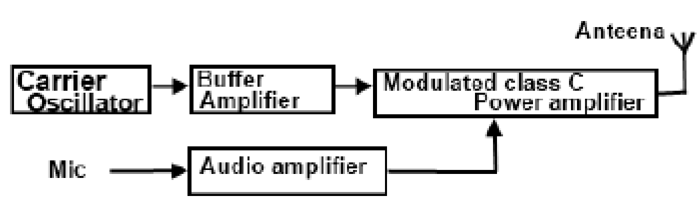


Fig. Block diagram of low level AM transmitter

The low-level AM transmitter shown in the above figure 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.

Coupling of Output Stage and Antenna

The output stage of the modulated class C power amplifier feeds the signal to the transmitting antenna. To transfer maximum power from the output stage to the antenna it is necessary that the impedance of the two sections match. For this, a matching network is required. The matching between the two should be perfect at all transmitting frequencies. As the matching is required at different frequencies, inductors and capacitors offering different impedance at different frequencies are used in the matching networks.

Multiplexing

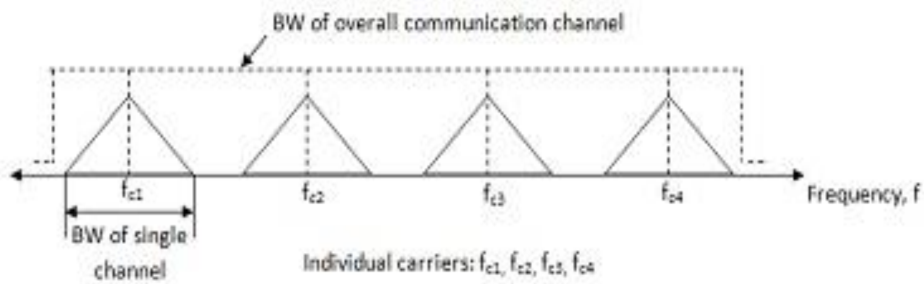
Multiplexing is a method by which multiple analog or digital signals are combined into one signal over a shared medium. There are three types of multiplexing techniques.

1. **Time-division multiplexing**
2. **Frequency-division multiplexing**
3. **Wave length-division multiplexing**

Frequency-division multiplexing (FDM)

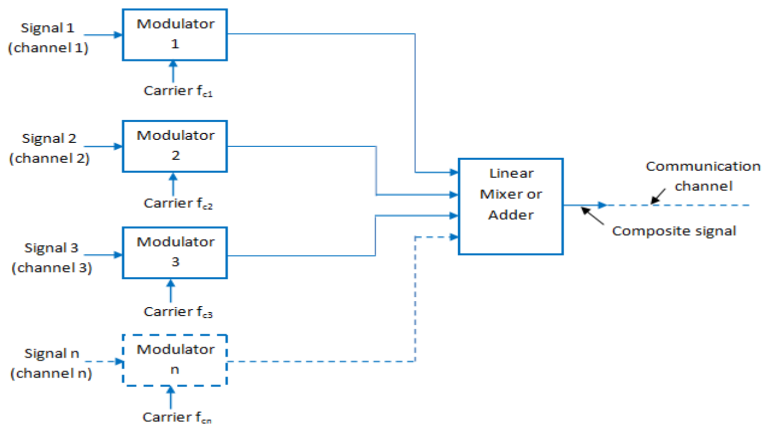
FDM is an analog technique which is applied only when the bandwidth of the link is greater than the combined bandwidth of the signals to be transmitted. Each sending device generates signals which modulate at different carrier frequencies. To hold the modulated signal, the carrier frequencies are separated by sufficient bandwidth.

The operation of frequency division multiplexing (FDM) is based on sharing the available bandwidth of a communication channel among the signals to be transmitted.



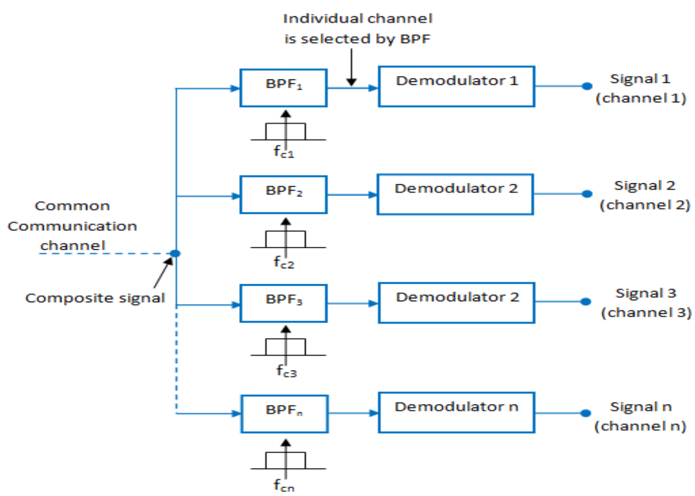
FDM Transmitter

Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the corresponding signals. The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place.



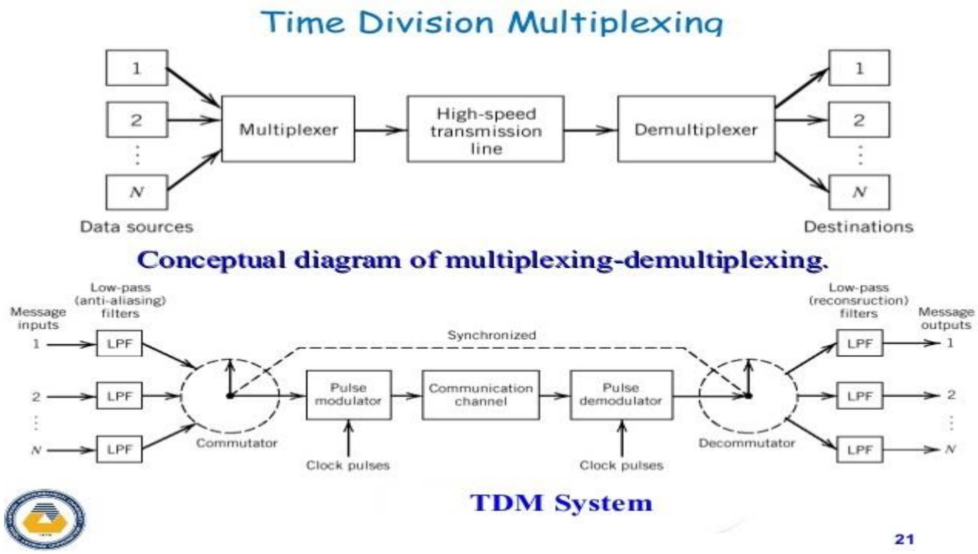
FDM Receiver

Each BPF has a center frequency corresponding to one of the carriers. The BPFs have an adequate bandwidth to pass all the channel information without any distortion. Each filter will pass only its channel and rejects all the other channels. The channel demodulator then removes the carrier and recovers the original signal back.



Time division multiplexing

TDM is a communications process that transmits two or more streaming digital signals over a common channel. In TDM, incoming signals are divided into equal fixed-length time slots. After multiplexing, these signals are transmitted over a shared medium and reassembled into their original format after de-multiplexing. Time slot selection is directly proportional to overall system efficiency.



Since the synchronization of the signals is a must for a time division multiplexing system, a synchronization pulse is used at the receiver side. It is used to decode the information from individual channels in the proper way. The absence of synchronization will lead to severe interference among the channels in almost all cases. The output is decoded and by using suitable filters, information from each channel is separated.

Difference between TDM and FDM

S.No:	FDM	TDM
1	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signal which is to be multiplexed, they occupy the entire bandwidth in the time domain.
2	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signal.
3	Synchronization is not required.	Synchronization is required is required.
4	The FDM requires a complex circuitry at transmitter and receiver.	TDM circuitry is not very complex.
5	FDM suffers from the problem of	In TDM the problem of crosstalk is not

	crosstalk due to imperfect Band Pass Filter	severe.
6	Due to bandwidth fading in the transmission medium, all the FDM channels are affected.	Due to fading only a few TDM channels will be affected.

ASSIGNMENT CUM TUTORIAL QUESTIONS

A) Questions testing the understanding and remembering level of students.

I. Objective Questions

1. How many types of multiplexing techniques are

(a) One (b) Two (c) Three (d) Four

2. Time-Division multiplexing (TDM) is a digital technique of

(a) Encoding (b) Decoding (c) Multiplexing (d) De-multiplexing

3. Applications of Frequency-Division Multiplexing (FDM) are

(a) Television broadcasting (b) AM and FM radio stations

(c) Cellular Telephones (d) All of the above

4. FDM is an analog multiplexing technique that combines

(a) Digital signals (b) Analog signals (c) Both a and b (d) None

II.<u>Descriptive Questions</u>

1. Explain the operation of high level AM transmitter
2. Define multiplexer
3. Explain the concept of Time Division Multiplexing.
4. Explain the concept of Frequency Division Multiplexing
5. Explain the operation of low level AM transmitter
6. Distinguish between the TDM & FDM multiplexing techniques?

Problems

1. Assume that a voice channel occupies a bandwidth of 4 kHz. We need to multiplex 10 voice channels with gaurd bands of 500 Hz using FDM. Calculate the required bandwidth.
2. Twenty-four voice signals are to be multiplexed and transmitted over twisted pair. What is the bandwidth required for FDM?
3. A signal $m_1(t)$ is band-limited to 3.6khz and 3 other signals $m_2(t)$, $m_3(t)$ and $m_4(t)$ are band-limited to 1.2khz each. These signals are to be transmitted by means of TDM
   (a) Set up a scheme for accomplishing this multiplexing requirement, with each signal sampled at its Nyquist rate.
   (b) What must be the speed of commutator (in samples per second)?
   (c) Determine the minimum transmission bandwidth of the channel.

Unit – IV

Objectives:

- To introduce the concepts of AM Receivers

UNIT - IV:AM Receivers

Receiver types – tuned radio frequency receiver, super heterodyne receiver,image frequency and rejection ratio,RF section and receiver characteristics,AGC**.**

Outcomes:

Students will be able to

- Understand various blocks in AM receivers
- Analyze image frequency and rejection ratio
- Compare TRF receiver with super heterodyne receiver

RECEIVERS

This process is used in the receivers to recover the original signal coming from the sender end in modulating form. We can say that its function is opposite to that of modulation process.

As we have discussed earlier that signal transmission is done by superimposing the signal on a carrier wave. This is done by using any one method from the number of available modulation methods. Then after modulation these amplified signals are transmitted successfully with the help of a transmitting antenna.

When the signals reach the destination i.e. at the receiver end, then the signal strength will be very less. This weak signal is amplified with the help of other signals. After amplification this signal is filtered from the other signals which were used earlier to modify it. When the signal becomes ready for demodulation process, then the below steps are performed for demodulation. These steps are basically the functions of the receiver:

1. Demodulating and amplifying the received signal.
2. Filtering of the original received signal from the non necessary signals.
3. Proper display of the received signal after the completion of demodulation process.

TYPES OF RECEIVERS

Receivers are basically available in two types:

1. Tuned Radio Frequency Receiver.
2. Super Heterodyne Receiver.

TUNED RADIO FREQUENCY RECEIVER

The RF signal (RF energy) is picked up by the antenna, as it is in all receivers, and goes into the tuned circuit. The circuit diagrams above shows a parallel tuned circuit, which can be tuned to the desired radio station (frequency) by the variable capacitor.

At the resonant frequency, the parallel tuned circuit has a high **impedance** (or opposition to AC / RF currents), the effect of this is to generate a maximum AC / RF potential difference across the tuned circuit.

The potential difference is rectified by the diode and DC is smoothed out by the capacitor C. The end result is that the recovered audio is only the overall shape of the signal which is applied to the ear piece / headphones which would have been the same shaped audio signal used to modulate the transmitter hence you hear what was transmitted.

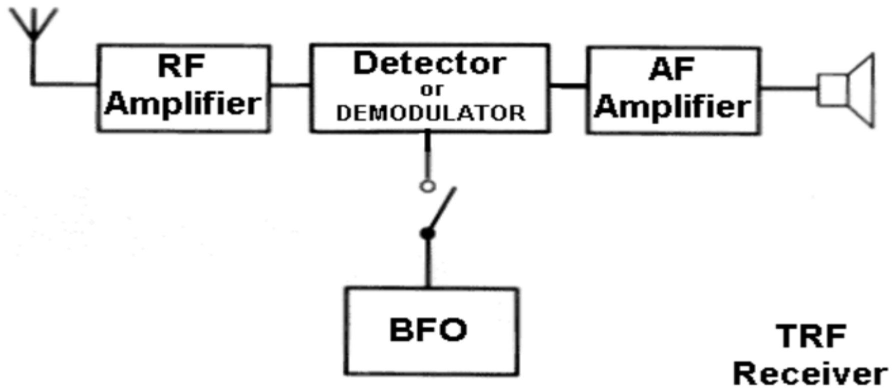


Fig: TRF Receiver Block Diagram

The Tuned Radio Frequency receiver is for AM signals and with the BFO (beat frequency oscillator switched in can also receive CW signals.

RF Amplifier

TRF has a RF Amplifier (also called a Radio Frequency Stage or (RF) Stage). This stage amplifies the incoming signal from the antenna before it passes it to the detector or demodulator stage.

The detector (also call the demodulator)

The detector recovers the AM signal and operates on the same principal as the crystal set but has the advantage that the amplified signal is provided by the RF amplifier stage. Then the recovered audio is passed to the Audio Amplifier stage.

Audio Amplifier

The Audio Amplifier stage simply makes the recovered audio signal louder so that it is sufficient to hear in a loudspeaker.

BFO

The detector mixing with the BFO recovers the CW signal. As mentioned above the detector recovers the AM signal and with the BFO switched in a CW signal can be resolved. Then the recovered audio is passed to the Audio Amplifier stage.

SUPERHETERODYNE RADIO RECEIVER

One of the most common forms of radio receiver is the superhet or superheterodyne radio receiver. Virtually all broadcast radio receivers, as well as televisions, short wave receivers and commercial radios use the superheterodyne principle as the basis of their operation. The superheterodyne radio technique is used in most radios found around the home. Virtually all transistor portable radios as well as television sets, hand portable radios and many others use the superheterodyne principle.

Basic superheterodyne concepts

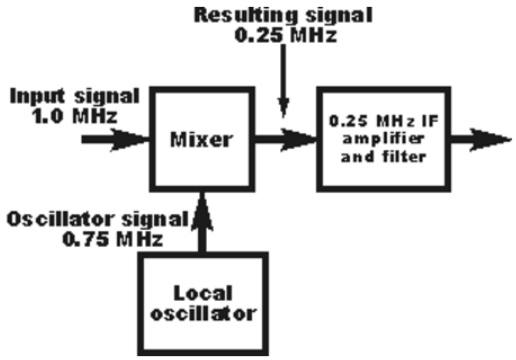
The superhet or superheterodyne radio operates by taking the signal on the incoming frequency, mixing it with a variable frequency locally generated signal to convert it down to a frequency where it can pass through a high performance fixed frequency filter before being demodulated to extract the required modulation or signal. It is obviously necessary to look ay this in more detail to understand the concept behind what goes on, but the main process in the superheterodyne radio is that of mixing.

How the superheterodyne receiver works

In order to look at how a superhet or superheterodyne radio works, it is necessary to follow the signal through it. In this way the processes it undergoes can be viewed more closely. The signal that is picked up by the antenna passes into the receiver and enters a mixer. Another locally generated signal, often called the local oscillator, is fed into the other port on the mixer and the two signals are mixed. As a result new signal are generated at the sum and difference frequencies.

The output from the mixer is passed into what is termed the intermediate frequency or IF stages where the signal is amplified and filtered. Any of the converted signals that fall within the pass band of the IF filter will be able to pass through the filter and they will also be amplified by the amplifier stages. Any signals that fall outside the pass band of the filter will be rejected.

It is often helpful to look at a real example to illustrate how the process works. To see how this operates in reality take the example of two signals, one at 1.0 MHz and another at 1.1MHz. If the IF filter is centred at 0.25 MHz, and the local oscillator is set to 0.75 MHz, then the two signals generated by the mixer as a result of the 1.0 MHz signal fall at 0.25 MHz and 1.75MHz. Naturally the 1.75 MHz signal is rejected, but the one at 0.25 MHz passes through the IF stages. The signal at 1.1 MHz produces a signal at 0.35 MHz and another at 1.85MHz. Both of these fall outside bandwidth of the IF filter so the only signal to pass through the IF is that from the signal on 1.0 MHz.



The basic concept of the superheterodyne radio
using a mixer to convert the frequency of the incoming signal

If the local oscillator frequency is moved up by 0.1 MHz to 0.85 MHz then the signal at 1.1 MHz will give rise to a signal at 0.25 MHz and another at 1.95 MHz. As a result the signal at 1.1 MHz giving rise to the 0.25 MHz signal after mixing will pass through the filter. The signal at 1.0 MHz will give rise to a signal of 0.15 MHz at the IF and another at 1.85 MHz and both will be rejected. In this way the receiver acts as a variable frequency filter, and tuning is accomplished by varying the frequency of the local oscillator within the superhet or superheterodyne receiver.

The advantage of the superheterodyne radio process is that very selective fixed frequency filters can be used and these far out perform any variable frequency ones. They are also normally at a lower frequency than the incoming signal and again this enables their performance to be better and less costly.

Complete superheterodyne receiver

Having looked at the concepts behind the superheterodyne receiver it is helpful to look at a block diagram of a basic superhet. Signals enter the front end circuitry from the antenna. This contains the front end tuning for the superhet to remove the image signal and often includes an RF amplifier to amplify the signals before they enter the mixer. Once the signals leave the mixer they enter the IF stages. These stages contain most of the amplification in the receiver as well as the filtering that enables signals on one frequency to be separated from those on the next. Filters may consist simply of LC tuned transformers providing inter-stage coupling, or they may be much higher performance ceramic or even crystal filters, dependent upon what is required.

Once the signals have passed through the IF stages of the superheterodyne receiver, they need to be demodulated. Different demodulators are required for different types of transmission, and as a result some receivers may have a variety of demodulators that can be switched in to accommodate the different types of transmission that are to be encountered. The output from the demodulator is the recovered audio. This is passed into the audio stages where they are amplified and presented to the headphones or loudspeaker.

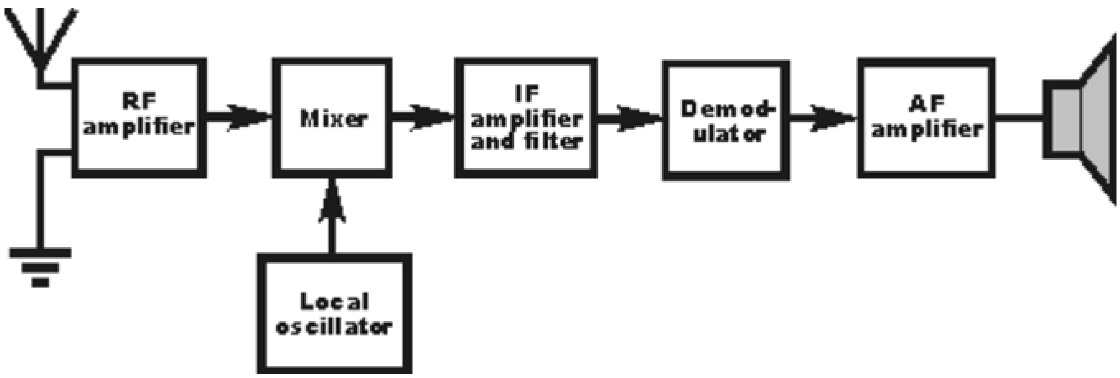


Fig. Block diagram of a basic super heterodyne receiver

The diagram above shows a very basic version of the superhet or superheterodyne receiver. Many sets these days are far more complicated.

RF SECTION AND RECEIVER CHARACTERISTICS

The RF stage provides

1. Increased sensitivity
2. Increased selectivity
3. Improved automatic volume control action.
4. Elimination of image frequency response.

Image Frequency and its rejection:

In a standard broadcast receiver the local oscillator frequency is made higher than the incoming signal frequency for reasons that will become apparent. It is made equal at all times to the signal frequency plus the intermediate frequency. Thus no $f_o = f_{si} + f_i$ or $f_s = f_o - f_i$ matter what the signal frequency may be. When f_s and are f_o mixed, the difference frequency , which is one of the by-products, is equal to f_i. As such , it is the only one passed and amplified by the IF stage.

If a frequency f_{si} manages to reach the mixer, such that, $f_{si} = f_o + f_i$ that is, $f_{si} = f_s + 2f_i$ then this frequency will also produce, when mixed with. The relationship of these frequencies is shown in above figure. This spurious intermediate-frequency signal will also be amplified by the IF stage and will therefore provided interference. This has the effect of two stations being received simultaneously and is naturally undesirable. The term f_o is called the image frequency and is defined as the signal frequency plus twice the intermediate frequency.

$$f_{si} = f_s + 2f_i \quad \text{----------(1)}$$

The rejection of image frequency by a single-tuned circuit i.e, the ratio is the gain at the signal frequency to the gain at the image frequency, is given by

$$\alpha = \sqrt{1 + Q^2 \rho^2} \quad \text{--------------(2)}$$

Where

$$\rho = \frac{f_{si}}{f_s} = \frac{f_s}{f_{si}} \quad \text{-------------(3)}$$

Automatic Gain Control:

The purpose of the Automatic Gain Control (AGC) / Automatic Voltage Control (AGC) is to maintain a constant voltage level at the output even though the input voltage may vary. The circuit can do this over a limited range eg. 30dB.

- Automatic gain control (AGC) is a mechanism wherein the overall gain of the radio receiver is automatically varied according to the changing strength of the received signal. This is done to maintain the output at a constant level.
- If the gain is not varied as per the input signal, consider a stronger input signal, then the signal might probably be distorted with some of the amplifiers reaching saturation level.
- AGC is applied to the RF, IF and mixer stages, which also helps in improving the dynamic range of the receiver antenna to 60-100 dB by adjusting the gain of the various stages in the radio receiver.

- The AGC derives dc bias voltage from the part of the detected signal to apply to the RF, IF and mixer stages to control their gains. The transconductance and hence the gain of the devices used in these stages of the receiver depends on the applied bias voltage or current.
- When the overall signal level increases, the value of the applied AGC bias increase leading to the decrease in the gain of the controlled stages.
- When there is no signal or signal with low value, there is minimum AGC bias which results in amplifier generating maximum gain.
- AGC facilitates tuning to varying signal strength stations providing a constant output.

There are two types of AGC circuits:

1. Simple AGC   2. Delayed AGC

Simple AGC:

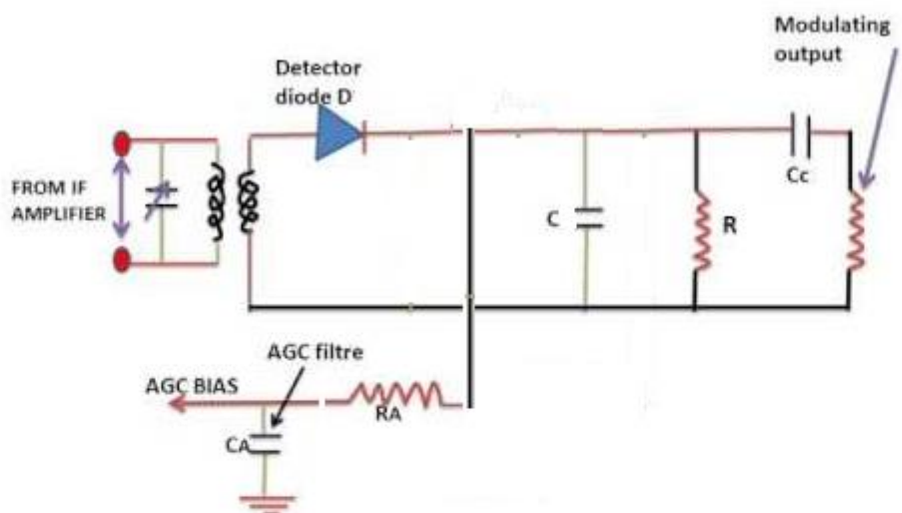


Fig: Simple AGC circuit

The above diagram shows the circuit diagram of linear diode detector with capacitor filter and simpleAGC. In this circuit half wave rectifier voltage is developed across the load instaneously positive bias voltage is applied . The capacitor ‘c’ is used to filter out RF components due to which the output of capacitor ‘c’ only consists of Dc and modulating voltage and obtain across the resistor R. The coupling capacitor Cc is used to require Dc components and finally at the output of coupling capacitor only modulating voltage components are available. To over come this frequency ccomponents we introduce an AGC filter .

The AGC filter mainly consists of two components

1.) Series resistance Ra
2.) Shunt capacitance Ca

The output of AGC only a positive DC voltage that is used in for AGC bias. If the time constant RaCa is very high some lowest modulating frequency components from AGC bias. If RaCa is very small the AGC bias follows the varying the series amplitude. Therefore, in practical the AGC filter having time constant is in the range of 0.1-0.2 seconds.

Drawbacks of simple AGC circuit:

1. The simple AGC clock operates even for very weak signals. Due to that the gain of receiver starts falling as soon as diode detector starts producing the output.
2. An ideal AGC signal must operates until the input carrier reaches to reasonable pre-determined voltage subsequently the AGC must come into operation to maintain the output level constant instead of variations of input voltage.

These two drawbacks are overcome by using delayed AGC

DELAYED AGC:

AGC bias is not applied to the amplifiers until signal strength crosses a predetermined level, after which AGC bias is applied.

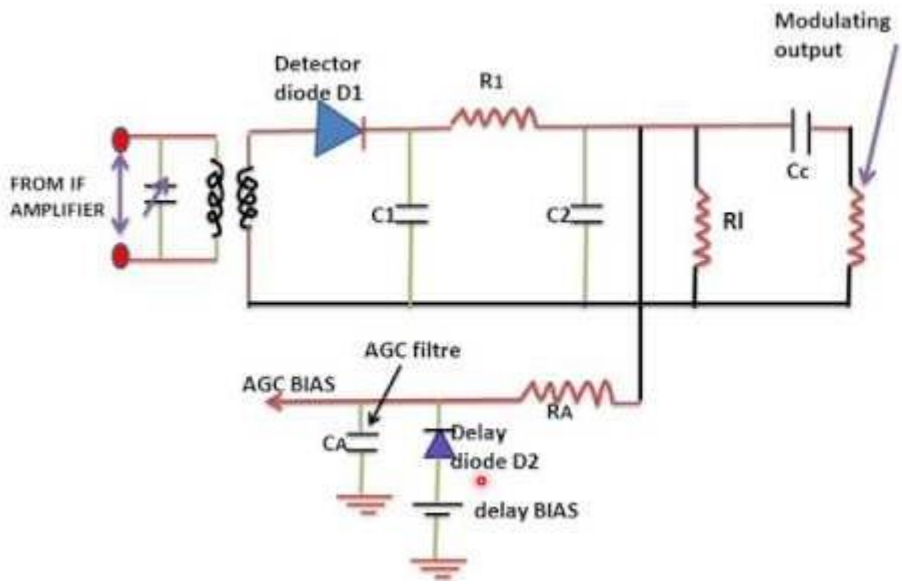


Fig: Delayed AGC circuit

Operation:

When input carrier voltage is increasing the AGC bias produce due to rectification of carrier voltage in diode detection D1 increases. When this rectifier bias magnitude exceeds the magnitude of the positive cathode voltage of diode D2. The diode D2 stops conduction hence the AGC works normally (works as simple AGC).

Unit – V

Objectives:

- To introduce the concepts of FM modulation and demodulation technique

UNIT – V: Angle Modulation

Introduction to angle modulation, relation between frequency and phase modulations, Single tone frequency modulation, narrow band FM, wide band FM, constant average power, transmission bandwidth of FM wave, generation of FM waves: direct FM and Armstrong method.

Outcomes:

Students will be able to

- Relate phase modulation and frequency modulation
- Explain single tone frequency modulation
- Identify the difference between NBFM & WBFM
- Classify methods of generation of FM.

ANGLE MODULATION

Angle modulation: there are two types of Angle modulation techniques namely

- Phase modulation
- Frequency modulation

Phase modulation (PM) is that of angle modulation in which the angular argument θ (t) is varied linearly with the message signal m(t), as shown by

$$\theta(t) = 2\pi f_c t + k_p m(t)$$

where $2\pi f_c t$ represents the angle of the unmodulated carrier

k_p represents the phase sensitivity of the modulator(radians/volt) The phase modulated wave $s(t)=A_c\cos[2\pi f_c t+k_p m(t)]$.

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

$$fi(t) = fc + k_f\, m(t)$$

Where fc represents the frequency of the unmodulated carrier

k_f represents the frequency sensitivity of the modulator(Hz/volt) .

The frequency modulated wave $s(t)=A_c\cos[2\pi f_c t+2\pi k_f \int_0^t m(t)dt]$

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

PM wave can be generated by first differentiating m(t) and then using the result as the input to a frequency modulator. Frequency modulation is a Non-linear modulation process.

This means that an FM signal can be generated by first integrating *m*(*t*) and the using the result as the input to a phase modulator (see Fig 2.1).

Relation between Phase modulation and Frequency modulation:

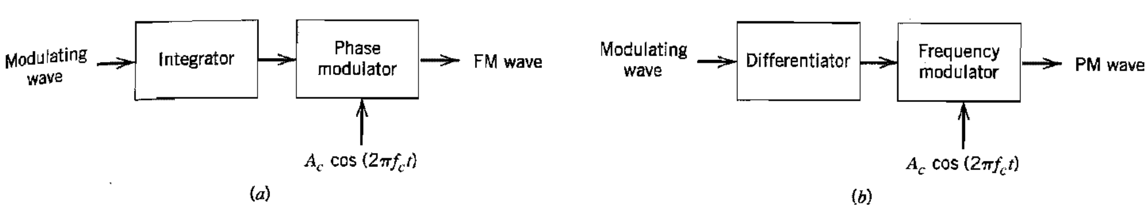


Fig.1 : Illustrating the relationship between frequency modulation and phase modulation.

Single tone FM:

Consider $m(t)=A_m\cos(2\pi f_m t)$

The instantaneous frequency of the resulting FM wave $f_i(t) = f_c + k_f A_m \cos(2\pi f_m t)$

$$= f_c + \Delta f \cos(2\pi f_m t)$$

where $\Delta f = k_f A_m$ is called as frequency deviation

$$\theta(t) = 2\pi \int f_i(t)dt = 2\pi f_c t + \Delta f / f_m \sin(2\pi f_m t) = 2\pi f_c t + \beta \sin(2\pi f_m t)$$

Where $\beta = \Delta f / f_m$ = modulation index of the FM wave

- When $\beta \ll 1$ radian then it is called as narrowband FM consisting essentially of a carrier, an upper side-frequency component, and a lower side-frequency component.
- When $\beta \gg 1$ radian then it is called as wideband FM which contains a carrier and an infinite number of side-frequency components located symmetrically around the carrier.

Narrowband Frequency Modulation:

Consider an FM Signal resulting from the use of a sinusoidal modulating signal, expanding the relation we get,

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

Assuming that the modulation index β is small compared to 1 radian , we may use the following approximation s

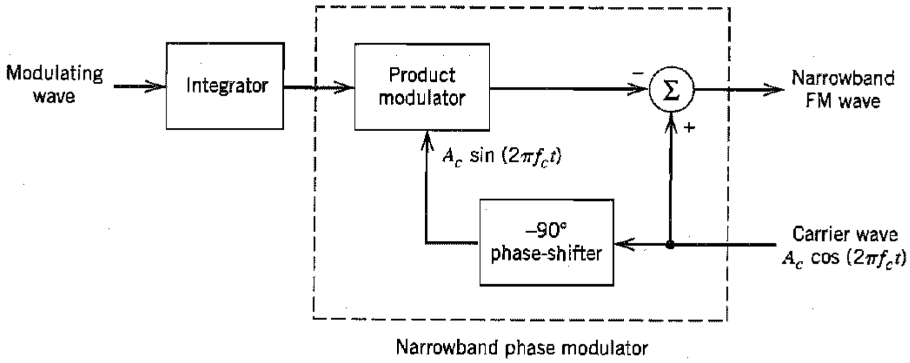
$$\cos[\beta \sin(2\pi f_m t)] \cong 1$$

$$\sin[\beta \sin(2\pi f_m t)] \cong \beta \sin(2\pi f_m t)$$

Hence, equation simplified to

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

Block diagram of a method for generating a Narrowband FM Signal :-



And the above equation is further simplied to

$S(t)= A_c \cos(2\pi f_c t) +0.5\ \beta A_c \{\cos(2\pi(f_c+f_m)t - \cos(2\pi(f_c-f_m)t\}$

And the corresponding narrowband FM signal with phasor diagram is shown below:

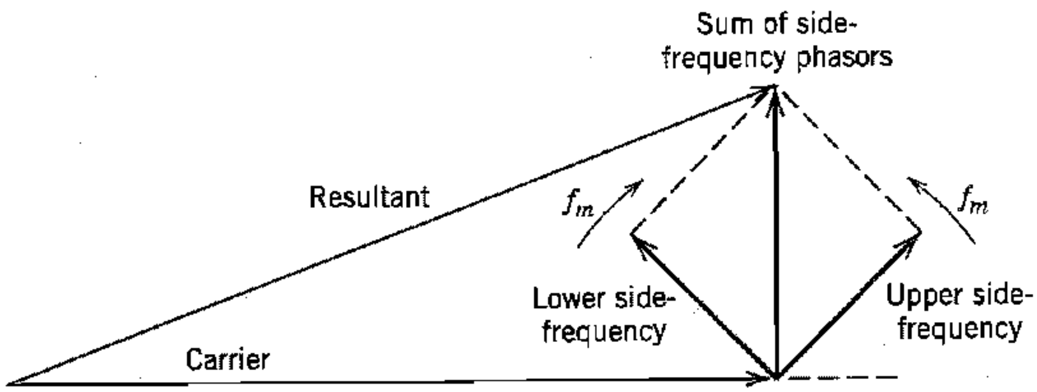


Fig.: Phasor diagram of Narrowband FM Signal

<u>Frequency Spectrum Analysis of sinusoidal FM wave</u>

The FM modulated signal in the time domain is given by

- From this equation it can be seen that the frequency spectrum of an FM waveform with a sinusoidal modulating signal is a discrete frequency spectrum made up of components spaced at frequencies of $\omega_c \pm n\omega m$
- By analogy with AM modulation, these frequency components are called sidebands.
- We can see that the expression for s(t) is an infinite series. Therefore the frequency spectrum of an FM signal has an infinite number of sidebands.

- The amplitudes of the carrier and sidebands of an FM signal are given by the corresponding Bessel functions, which are themselves functions of the modulation index

The following spectra show the effect of modulation index, β, on the bandwidth of an FM signal, and the relative amplitudes of the carrier and sidebands The envelope of an FM wave is constant, so that the average power of such a wave dissipated in a 1-ohm resistor is also constant.

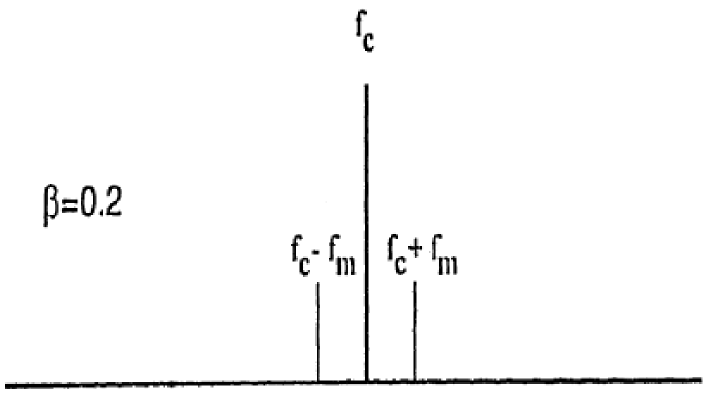


Fig. FM WAVE SIDE BANDS

WIDEBAND FM:

Parameters :

1. modulation index : Greater than 1
2. Maximum deviation: 75 kHz
3. Range of modulating frequency: 30 Hz to 15 kHz
4. Maximum modulation index: 5 to 2500
5. Bandwidth: large,about 15 times higher than BW of narrowband FM
6. Applications: entertainment broadcasting
7. Pre-emphasis and de-emphasis : is needed.

$$s(t) = A_c \left[\sum_{n=-\infty}^{\infty} j_n(\beta) \cos[2\pi(f_c + nf_m)t] \right]$$

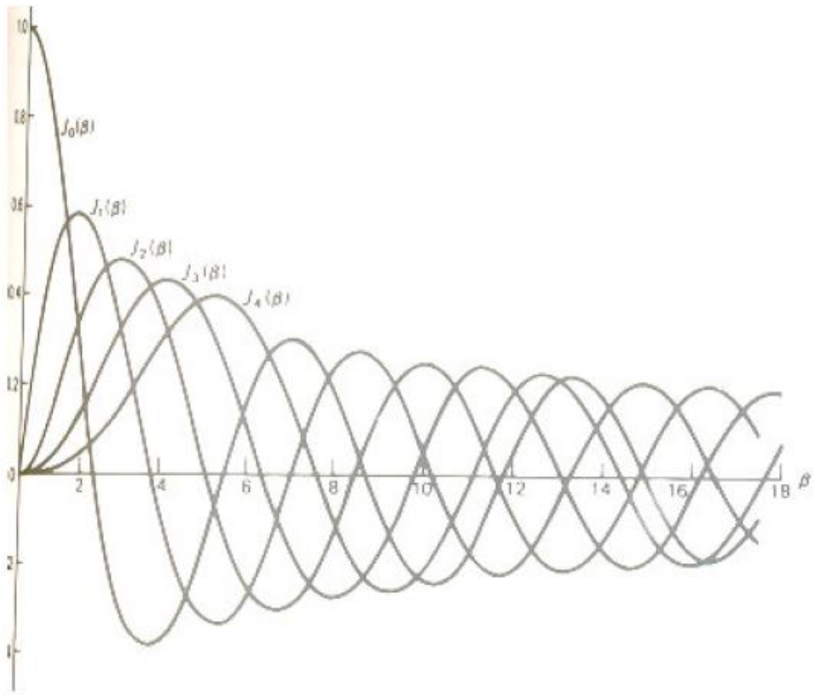


Fig 1 Plots of Bessels function of first kind

$$S(f)=A_c \sum_{n=-\infty}^{\infty} j_n(\beta)[\Delta((f - f_c - nf_m) + \Delta((f + f_c + nf_m)]$$

TRANSMISSION BANDWIDTH OF FM SIGNAL

In theory an FM signal contains an infinite number of side frequencies so that the band width required to transmit such a signal is infinite in extent.

In practice, we find that FM signal is effectively limited to a finite number of significant side frequencies.

In the case of an FM signal generated by a single tone modulating wave of frequency f_m , The side frequencies that are separated from the carrier frequency f_C by an amount greater than the frequency deviation f decrease rapidly toward zero.

Specifically for **large values of β,** the bandwidth approaches, and is only slightly greater than the total frequency **deviation**

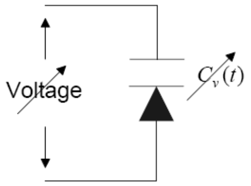
Carson’s Rule: Bandwidth is twice the sum of the maximum frequency deviation and the modulating frequency.

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

The nominal BW ≈ $2\Delta f = 2\,\beta f_m$.

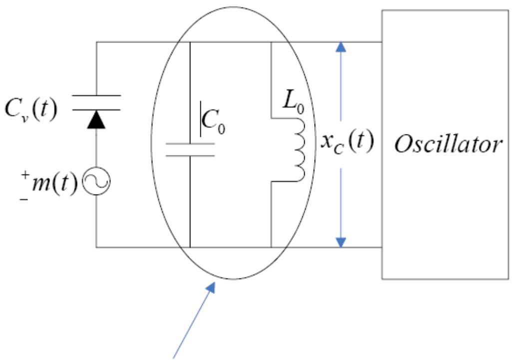
Direct Method of FM Generation:-

In this approach a “Varactor diode” is used. A Varactor diode is a capacitor whose capacitance changes with the applied voltage.



Varactor diode

Therefore , if this capacitor is used in a tuned circuit of the oscillator and the message signal is applied to it, the frequency of the tuned circuit ,and the oscillator will change in accordance with the message signal(see diagram below).



Tuned Circuit

Let the indicator in the tuned circuit be L_0 and the capacitance of the varactor diode is given by $C(t)=c_0+k_0m(t)$

When m (t) =0, the frequency of the tuned circuit is given by:

$$f_c = \frac{1}{2\pi\sqrt{L_0 C_0}}$$

In general if m(t) $\neq 0$,

$$\therefore f_i(t) = \frac{1}{2\pi\sqrt{L_0 C_0}} \cdot \frac{1}{\sqrt{1+\frac{h_0}{C_0}m(t)}} = f_c \cdot \frac{1}{\sqrt{1+\frac{k_0}{C_0}m(t)}}$$

Assuming that $\in = \frac{h_0}{c_0} m(t) << 1$

and using the approximations,

$$\boxed{\begin{array}{l} \sqrt{1+\varepsilon} = (1+\varepsilon)^{\frac{1}{2}} \approx 1 + \frac{\varepsilon}{2} \\ \frac{1}{1+\varepsilon} \approx 1 - \varepsilon \end{array}} \quad \boxed{\varepsilon < 1}$$

We obtain,

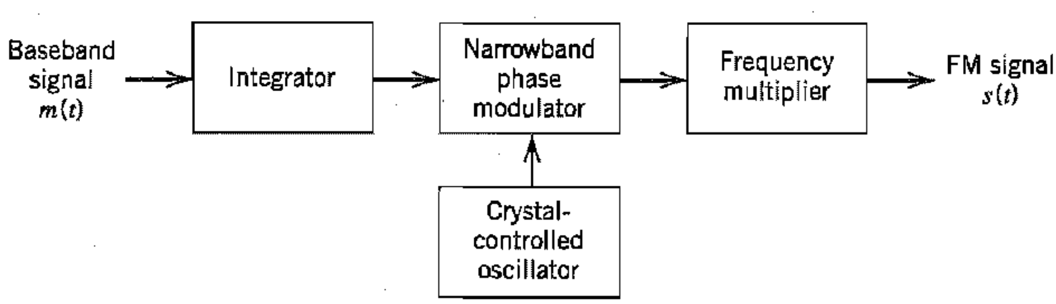
$$f_i(t) = f_c \cdot \left[1 - \frac{k_0}{2c_0} m(t) \right]$$

This is the relation for a frequency modulated signal.

$$x_c(t) = A_c \cos\theta_i(t) = A_c \cos(2\pi f_i(t))$$

$$\text{where } f_i(t) = f_c \left[1 - \frac{k_0}{2c_0} m(t) \right]$$

Indirect method of generation of FM:



In indirect FM , the baseband signal is first integrated and then used to phase modulate a crystal controlled oscillator. In order to minimize the distortion inherent in the phase modulator, the maximum phase deviation or modulation index β is kept small. Thereby resulting in a NBFM . this signal is next multiplied in frequency by means of a frequency multiplier so as to produce the desired wideband FM Wave.

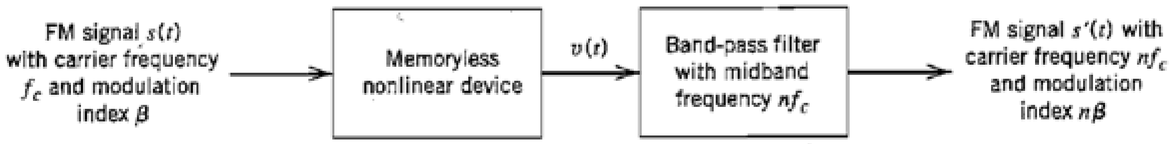
A frequency multiplier consists of a non-linear device followed by a Bandpass filter.
The instantaneous frequency in this method is $f_i(t)=f_c+k_f m(t)$
For now it suffices to say that after bandpass filtering of the non-linear device's output we have a new FM Signal defined by

$$s'(t) = A'_c \cos\left[2\pi n f_c t + 2\pi n k_f \int_0^t m(\tau)\, d\tau\right]$$

Whose instantaneous frequency is
$f_i(t)=nf_c+nk_f m(t)$.

We see that the non-linear processing circuit acts as a frequency multiplier. The frequency multiplication ratio is determined by the highest power n in the input-output relation , characterizing the memoryless nonlinear device.

Block diagram of a frequency multiplier:



Block diagram of frequency multiplier.

Unit – VI

Objectives:

- To introduce the concepts of FM modulation and demodulation technique

UNIT – IV: Angle Modulation

Detection of FM waves: balanced frequency discriminator, FM transmitters, FM receivers,noise in FM system, pre-emphasis and de-emphasis in FM. TDM.

Outcomes:

Students will be able to

- realize concept of frequency discrimination
- analyze FM modulation and demodulation schemes
- find the necessity of Pre-emphasis and De-emphasis.
- understand various blocks in FM transmitters and receivers

SLOPE DETECTOR AND BALANCED DETECTOR

The slope detector is the simplest type of FM detector. A schematic diagram of a slope detector appears below:

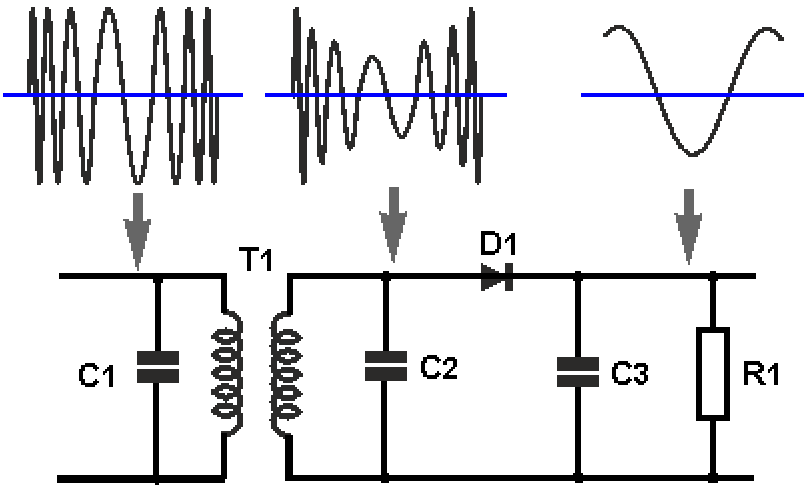
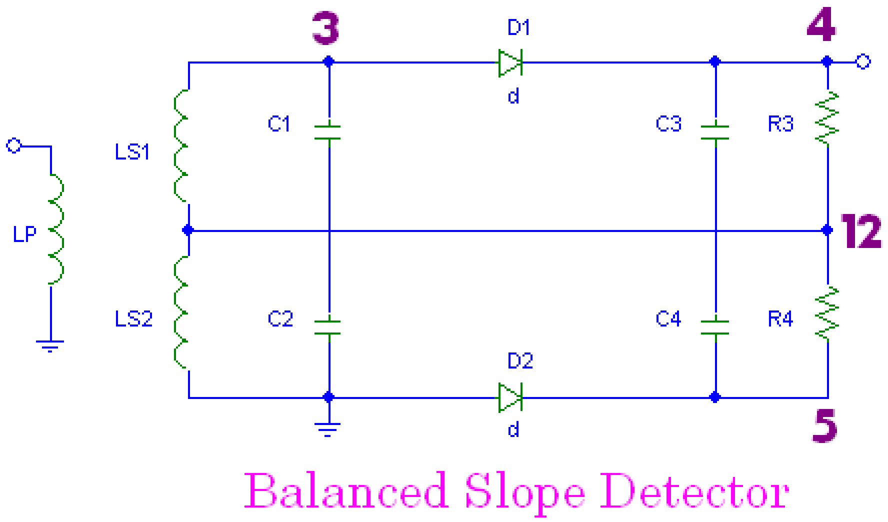


Fig. circuit diagram and waveforms



The operation of the slope detector is very simple. The output network of an amplifier is tuned to a frequency that is slightly more than the carrier frequency + peak deviation. As the input signal varies in frequency, the output signal across the LC network will vary in amplitude because of the band pass properties of the tank circuit. The output of this amplifier is AM, which can be detected using a diode detector.

The circuit shown in the diagram above looks very similar to the last IF amplifier and detector of an AM receiver, and it is possible to receive NBFM on an AM receiver by detuning the last IF transformer. If this transformer is tuned to a frequency of approximately 1 KHz above the IF frequency, the last IF amplifier will convert NBFM to AM.

In spite of its simplicity, the slope detector is rarely used because it has poor linearity.

To see why this is so, it is necessary to look at the expression for the voltage across the primary of the tuned transformer in the sloped detector:

The voltage across the transformer's primary winding is related to the square of the frequency. Since the frequency deviation of the FM signal is directly proportional to the modulating signal's amplitude, the output of the slope detector will be distorted. If the bandwidth of the FM signal is small, it is possible to approximate the response of the slope detector by a linear function, and a slope detector could be used to demodulate an NBFM signal

COMPARSION BETWEEN AM AND FM

In **radio communication**, the message signal wave (low frequency) is combined with a carrier signal (high frequency). In this combination, one or more characteristics of the carrier wave are varied with respect to message signal. This variation is termed as modulation and it is needed so that message can be transmitted over long distances and no undesired signal mixing takes place. Depending on several factors such as range, application and budget, modulation can be casted into three types: **Amplitude Modulation**, **Frequency Modulation** and **Phase Modulation**. Out of these three types, the former two are widely known as they form a major commercially applicative part of radio communication. In this article, we will discuss common **difference between AM and FM** which will enhance our learning in terms of these two technologies.

1. Evolution: Formulated in the 1870s, AM is a relatively older modulation process compared to FM which was found in the 1930s by Edwin Armstrong.

2. Technology: AM stands for **amplitude modulation** where the amplitude of the carrier is modulated as per the message signal. The other aspects of the carrier wave such as frequency phase etc. remain constant. On the other hand, **FM** means **frequency modulation** and in it only frequency of the carrier wave changes while amplitude, phase etc. remain constant.

3. Frequency range of working: Amplitude modulation works between 540-1650 KHz while FM works at 88-108MHz.

4. Power Consumption: FM based signal transmission consumes a higher amount of power than an equivalent AM based signal transmission system.

5. AM vs FM: Signal Quality: Signal quality is a **lot superior in FM than AM** as amplitude based signals are more susceptible to noise than those which use frequency.

Moreover, noise signals are difficult to filter out in AM reception whereas FM receivers easily filter out noise using the capture effect and pre-emphasis, de-emphasis effects. In capture effect, the receiver locks itself to catch stronger signal so that signals received are more synced with that at the transmitting end.

In pre-emphasis, de-emphasis process, the signal is further amplified to a higher frequency at sending end (pre-emphasis) and vice versa at receiver end (de-emphasis). These two processes reduce down the chances of a signal to get mixed with other signals and make FM more immune to noise than AM.

6. Fading: Fading refers to power variation during signal transmission. Due to fading, the power with the signal received can vary significantly and reception wouldn't be of a good quality. Fading is more prominent in amplitude modulation as compared to frequency modulation. That is why, AM radio channels often face the problem where sound intensity varies while FM radio channels have constant good reception.

7. Wavelength Difference between AM and FM: AM waves work in the range of KHz while in FM waves work in MHz range. As a result, AM waves have a higher wavelength than the FM ones. A higher wavelength increases the range of AM signals as compared to FM which have a limited area of coverage.

8. Bandwidth consumption: AM signals consume 30KHz of bandwidth for each while in FM 80KHz is the bandwidth consumed by each signal. Hence, over a limited range of bandwidth, more number of signals can be sent in AM than FM.

9. Circuit Complexity: Aforesaid, Amplitude Modulation is an older process and has a very simple circuitry. On the other hand, frequency modulation requires a complicated circuitry for transmission and reception of signal. The signals sent in FM are more modulated and emphasized at the transmitter and they are thoroughly checked and corrected at the receiving end. This is why circuitry for FM signals is very complicated.

10. Commercial Aspects: Setting up an AM based radio communication system is very economic as there is no complicated circuitry and processes are easy to understand

FM TRANSMITTER

INDIRECT METHOD (PHASE SHIFT) OF MODULATION

The part of the Armstrong FM transmitter (Armstrong phase modulator) which is expressed in dotted lines describes the principle of operation of an Armstrong phase modulator. It should be noted, first that the output signal from the carrier oscillator is supplied to circuits that perform the task of modulating the carrier signal. The oscillator does not change frequency, as is the case of direct FM. These points out the major advantage of phase modulation (PM), or indirect FM, over direct FM. That is the phase modulator is crystal controlled for frequency.

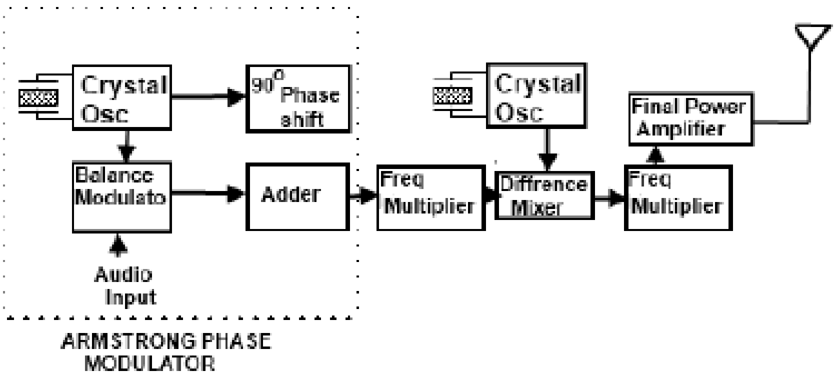


Fig. Armstrong Modulator

The crystal-controlled carrier oscillator signal is directed to two circuits in parallel. This signal (usually a sine wave) is established as the reference past carrier signal and is assigned a value 0°.The balanced modulator is an amplitude modulator used to form an envelope of double sidebands and to suppress the carrier signal (DSSC). This requires two input signals, the carrier signal and the modulating message signal. The output of the modulator is connected to the adder circuit; here the 90° phase-delayed carriers signal will be added back to replace the suppressed carrier. The act of delaying the carrier phase by 90° does not change the carrier frequency or its wave shape. This signal identified as the 90° carrier signal.

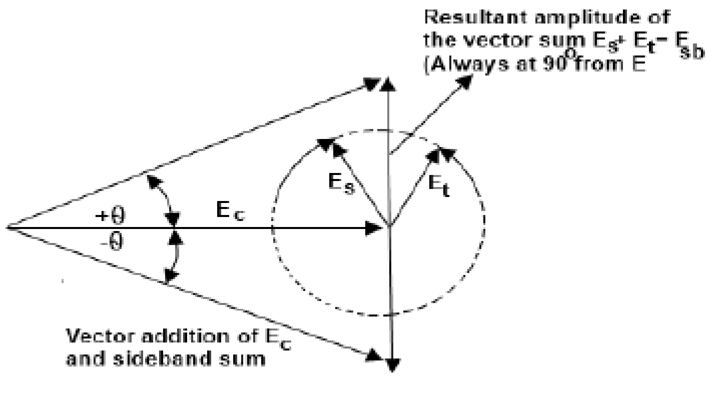


Fig. Phasor diagram of Armstrong Modulator

$$\% \text{ of modulation} = \frac{E_{max} - E_{min}}{E_{max} + E_{min}} \times 100$$

The carrier frequency change at the adder output is a function of the output phase shift and is found by. fc = ∆θfs (in hertz) When θ is the phase change in radians and fs is the lowest audio modulating frequency. In most FM radio bands, the lowest audio frequency is 50Hz. Therefore, the carrier frequency change at the adder output is 0.6125 x 50Hz = ± 30Hz since 10% AM represents the upper limit of carrier voltage change, then ± 30Hz is the maximum deviation from the modulator for PM.

The 90° phase shift network does not change the signal frequency because the components and resulting phase change are constant with time. However, the phase of the adder output voltage is in a continual state of change brought about by the cyclical variations of the message signal, and during the time of a phase change, there will also be a frequency change.

USING REACTANCE MODULATOR DIRECT METHOD

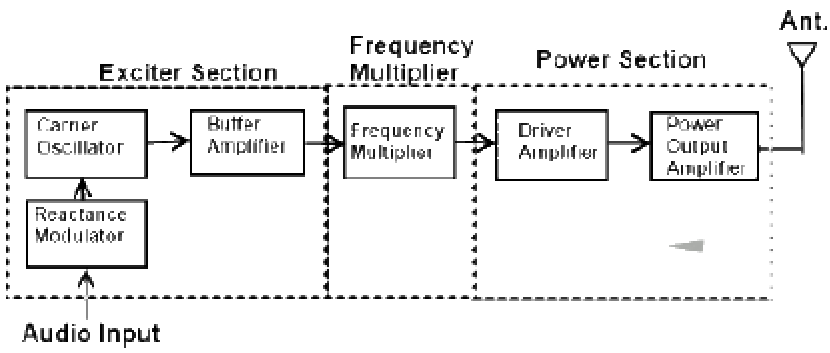


Fig. Reactance Modulator

The FM transmitter has three basic sections.

1. The exciter section contains the carrier oscillator, reactance modulator and the buffer amplifier.
2. The frequency multiplier section, which features several frequency multipliers.
3. The power output section, which includes a low level power amplifier, the final power amplifier, and the impedance matching network to properly load the power section with the antenna impedance.

The essential function of each circuit in the FM transmitter may be described as follows.

The Exciter

1. The function of the carrier oscillator is to generate a stable sine wave signal at the rest frequency, when no modulation is applied. It must be able to linearly change frequency when fully modulated, with no measurable change in amplitude.
2. The buffer amplifier acts as a constant high-impedance load on the oscillator to help stabilize the oscillator frequency. The buffer amplifier may have a small gain.
3. The modulator acts to change the carrier oscillator frequency by application of the message signal. The positive peak of the message signal generally lowers the oscillator's frequency to a point below the rest frequency, and the negative message peak raises the oscillator frequency to a value above the rest frequency. The greater the peak-to-peak message signal, the larger the oscillator deviation.

Frequency multiplier

Frequency multipliers are tuned-input, tuned-output RF amplifiers in which the output resonant circuit is tuned to a multiple of the input frequency. Common frequency multipliers are 2x, 3x and 4x multiplication. A 5x Frequency multiplier is sometimes seen, but its extreme low efficiency forbids widespread usage. Note that multiplication is by whole numbers only. There can not a 1.5x multiplier, for instance.

Power Output Section

The final power section develops the carrier power, to be transmitted and often has a low-power amplifier driven the final power amplifier. The impedance matching network is the same as for the AM transmitter and matches the antenna impedance to the correct load on the final over amplifier.

FM RECEIVER

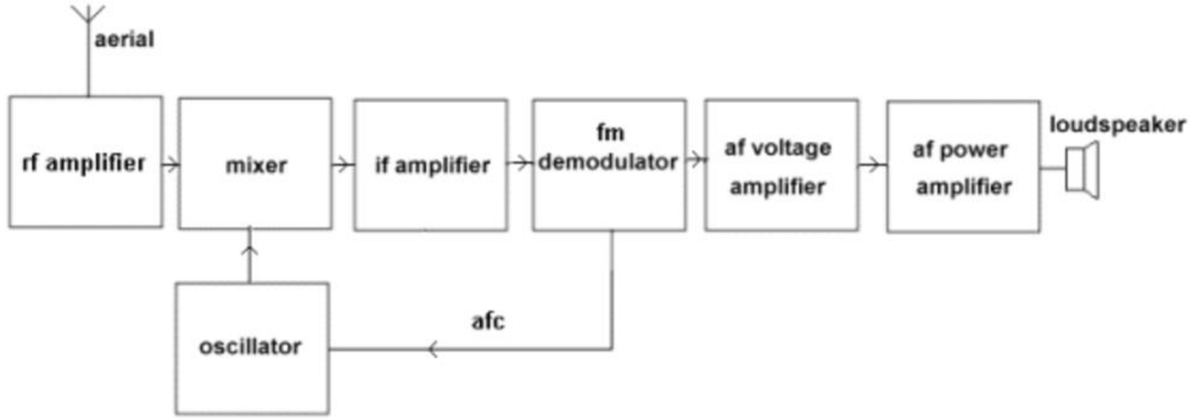


Fig.: FM Receiver.

See filters, mixers, frequency changers, am modulation and amplifiers. The f.m. band covers 88-108 MHz. There are signals from many radio transmitters in this band inducing signal voltages in the aerial. The rf amplifier selects and amplifies the desired station from the many. It is adjustable so that the selection frequency can be altered. This is called TUNING. In cheaper receivers the tuning is fixed and the tuning filter is wide enough to pass all signals in the f.m. band. The selected frequency is applied to the mixer. The output of an oscillator is also applied to the mixer. The mixer and oscillator form a FREQUENCY CHANGER circuit. The output from the mixer is the intermediate frequency (i.f.) The I.F.is a fixed frequency of 10.7 MHz. No matter what the frequency of the selected radio station is, the I.F.is always 10.7 MHz. The I.F.signal is fed into the I.F.amplifier. The advantage of the I.F. amplifier is that its frequency and bandwidth are fixed, no matter what the frequency of the incoming signal is. This makes the design and operation of the amplifier much simpler. The amplified I.F.signal is fed to the demodulator. This circuit recovers the audio signal and discards the r.f. carrier. Some of the audio is fed back to the oscillator as an AUTOMATIC FREQUENCY CONTROL voltage. This ensures that the oscillator frequency is stable in spite of temperature changes. The audio signal voltage is increased in amplitude by a voltage amplifier. The power level is increased sufficiently to drive the loudspeaker by the power amplifier.

Pre-emphasis and De-emphasis

There is another way in which the SNR of an FM system may be increased . We saw in the previous subsection that the PSD of the noise at the detector output has a square-law dependence on frequency. On the other hand, the PSD of a typical message source is not uniform, and typically rolls off at around 6 dB per decade. We note that at high frequencies the relative message power is quite low, whereas the noise power is quite high (and is rapidly increasing). It is possible that this situation could be improved by reducing the bandwidth of the transmitted message (and the corresponding cutoff frequency of the baseband LPF in the receiver), thus rejecting a large amount of the out-of-band noise. In practice, however, the distortion introduced by low-pass filtering the message signal is unsatisfactory

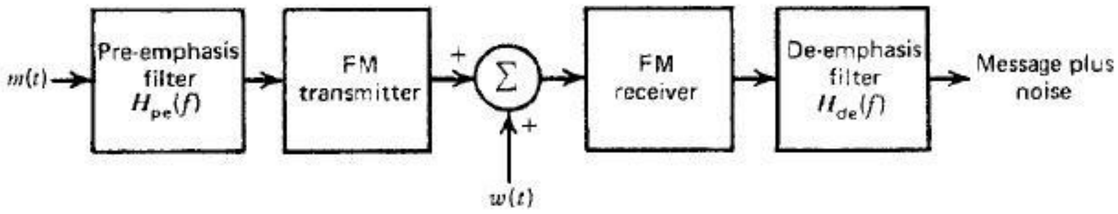


Figure: Pre-emphasis and Deemphasis in FM System

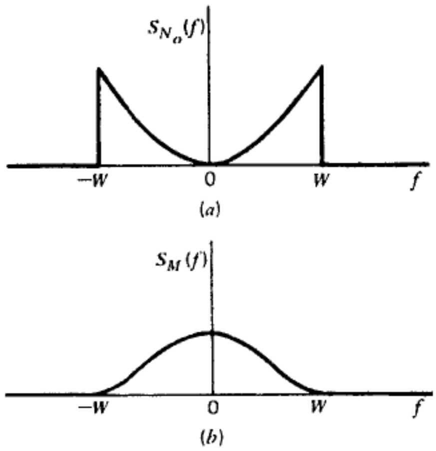
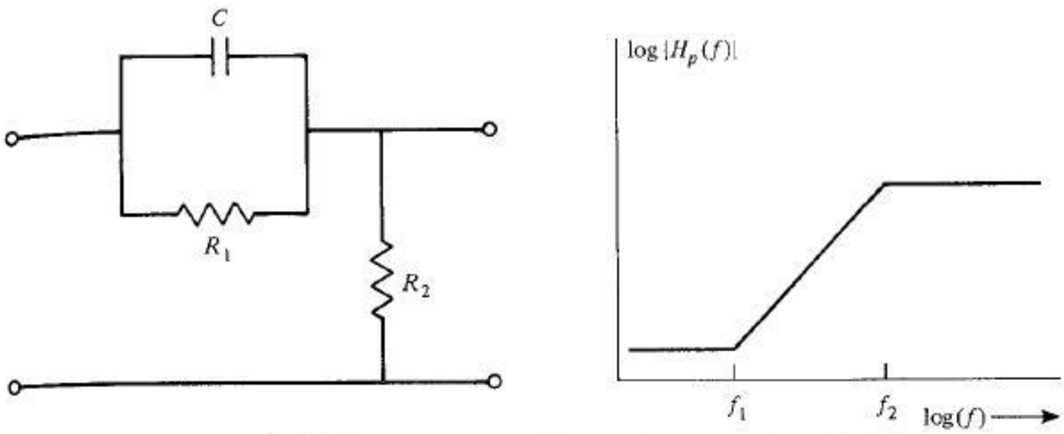


Figure: PSD of: (a) noise at the output of FM receiver, (b) a typical message signal



$$H_p(f) = K\,\frac{1 + j(f/f_1)}{1 + j(f/f_2)} \quad \text{where } f_1 = \frac{1}{2\pi\tau_1} = \frac{1}{2\pi R_1 C},\ f_2 = \frac{1}{2\pi\tau_2} = \frac{R_1 + R_2}{2\pi R_1 R_2 C}$$

(a) Preemphasis Filter

(b) Bode Plot of Preemphasis Frequency Response

A better solution is obtained by using the *pre-emphasis* and *de-emphasis* stages shown in Fig. . The intention of this scheme is that (*f*) is used to artificially emphasize the high frequency components of the message prior to modulation, and hence, before noise is introduced. This serves to effectively equalize the low- and high-frequency portions of the message PSD such that the message more fully utilizes the bandwidth available to it. At the receiver, (*f*) performs the inverse operation by de-emphasizing the high frequency components, thereby restoring the original PSD of the message signal.

Simple circuits that perform pre- and de-emphasis are shown in Fig. , along with their respective frequency responses. Haykin shows that these circuits can improve the output SNR by around 13 dB. In closing this section, we also note that Dolby noise reduction uses an analogous pre-emphasis technique to reduce the effects of noise.

Noise in FM:-

The model of an FM receiver[1] is shown in Fig., where $s(t)$ is the FM signal , and $w(t)$ is white Gaussian noise with power spectral density $No/2$. The bandpass filter is used to remove any signals outside the bandwidth of $f_c+B_T/2$, and thus, the predetection noise at the receiver is bandpass with a bandwidth of B_T. Since an FM signal has a constant envelope, the limiter is used to remove any amplitude variations. The discriminator is a device whose output is proportional to the deviation in the instantaneous frequency (i.e., it recovers the message signal), and the final baseband low-pass filter has a bandwidth of W and thus passes the message signal and removes out-of-band noise.

The predetection signal is

$x(t) = A\cos[2\pi f_c dt+2\pi k_f \int m(T)dT]+nc(t)]\cos(2\pi f_c t)-n_s(t)\sin(2\pi f_c t)$

First, let us consider the signal power at the receiver output. When the predetection SNR is high, it can be shown that the noise does not affect the power of the signal at the output.c Thus,

ignoring the noise, the instantaneous frequency of the input signal is

$fi = fc + k_f m(t)$

and the output of the discriminator (which is designed to simply return the deviation of the instantaneous frequency away from the carrier frequency) is $K_f m(t)$

The output signal power is therefore

$Ps = K_f^2 P$

where P is the average power of the message signal.

Now, to calculate the noise power at the receiver output, it turns out that for high predetection SNR the noise output is approximately independent of the message signal. In this case, we only

have the carrier and noise signals present. Thus,

$$\tilde{x}(t) = A\cos(2\pi f_c t)+n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t)$$

The phasor diagram of this is shown in Fig. From this diagram, we see that the instantaneous phase is

$\theta_i(t) = tan^{-1}\frac{n_s(t)}{A+n_c(t)}$,

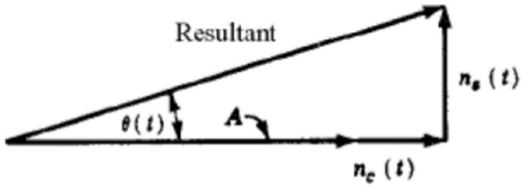


Figure: Phasor diagram of the FM carrier and noise signals.
For large carrier power, then most of the time

$$\theta_i(t) = tan^{-1}\frac{n_s(t)}{A} \approx \frac{n_s(t)}{A}$$

On solving the noise equations finally, we have that at the output the SNR is

$$SNR_O = \frac{3A^2K_f^2\, p}{2N_O W^3}$$

Since the transmitted power of an FM waveform is $\mathbf{P_T=A^2/2}$

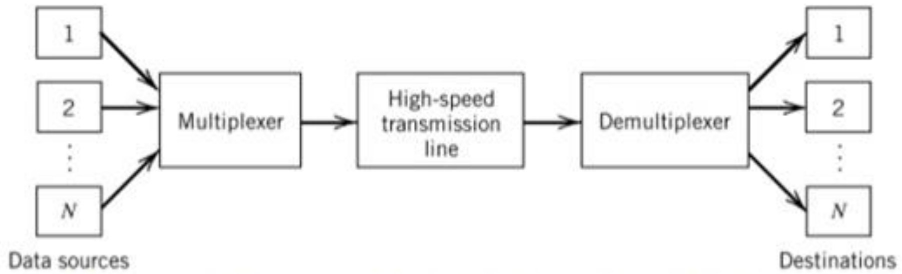
$$\text{SNR}_{\text{FM}} = \frac{3K_f^2 p}{w^2} SNR_{baseband} = 3\beta^2 \frac{P}{m_p^2} SNR_{baseband}$$

One should note that whereas equation suggests that output SNR for an FM system can be increased arbitrarily by increasing β while keeping the signal power fixed, inspection of equation shows this not to be strictly true. The reason is that if β increases too far, the condition of equation that we are above threshold may no longer be true, meaning that equation no longer provides an expression for the true SNR.

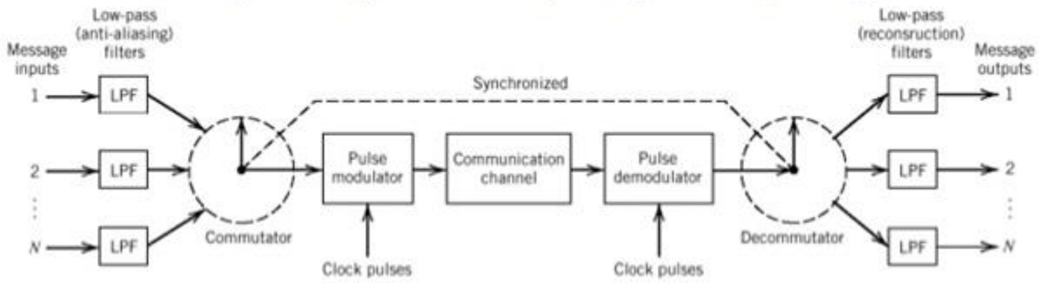
Time Division Multiplexing:-

TDM is widely used in digital communications, for example in the form of pulse code modulation in digital telephony. In TDM, each message signal occupies the channel (*e.g.* a transmission line) for a short period of time. The principle is illustrated below:

Time Division Multiplexing



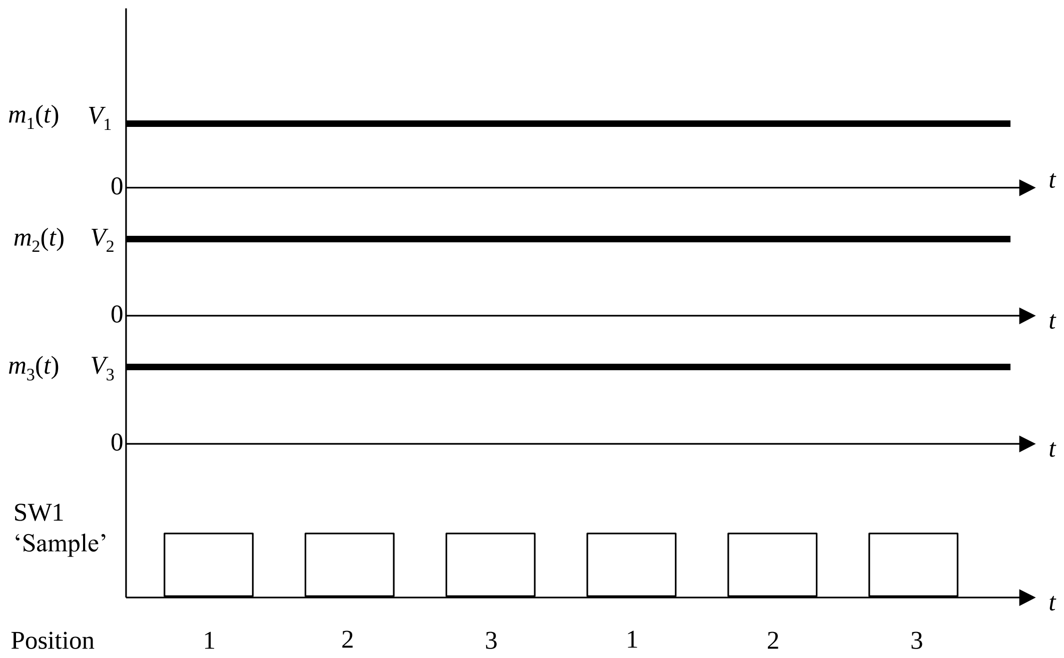
Conceptual diagram of multiplexing-demultiplexing.

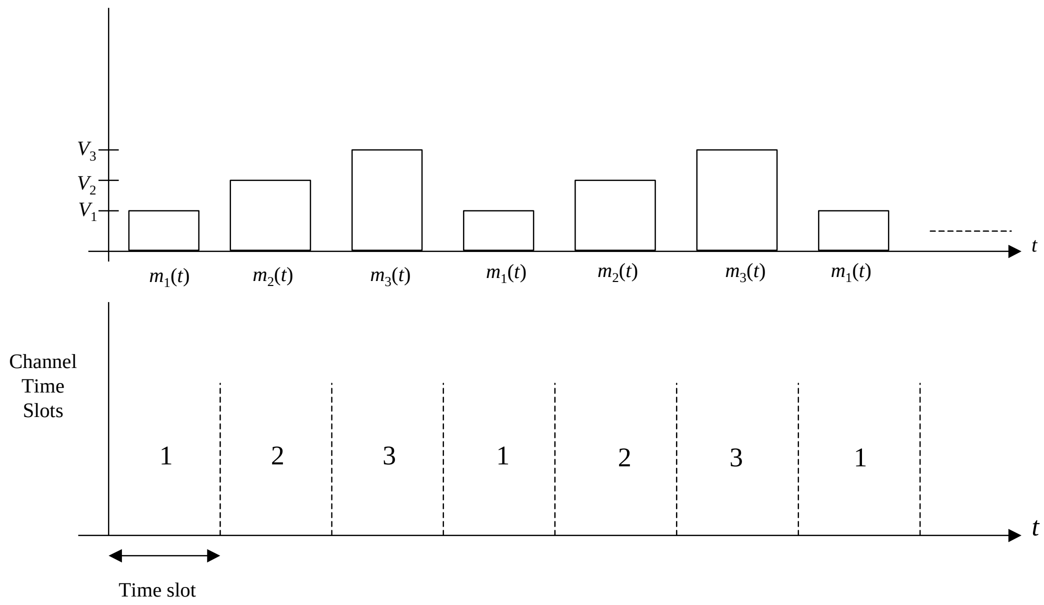


PAM TDM System

Switches SW1 and SW2 rotate in synchronism, and in effect sample each message input in a sequence $m_1(t)$, $m_2(t)$, $m_3(t)$, $m_4(t)$, $m_5(t)$, $m_1(t)$, $m_2(t)$,…

The sampled value (usually in digital form) is transmitted and recovered at the ‘far end’ to produce output $m_1(t)$…$m_5(t)$. For ease of illustration consider such a system with 3 messages, $m_1(t)$, $m_2(t)$ and $m_3(t)$, each a different DC level as shown below.





In this illustration the samples are shown as levels, *i.e.* V_1, V_2 or V_3. Normally, these voltages would be converted to a binary code before transmission as discussed below.

Note that the channel is divided into time slots and in this example, 3 messages are time-division multiplexed on to the channel. The sampling process requires that the message signals are a sampled at a rate $f_s \geq 2B$, where f_s is the sample rate, samples per second, and B is the maximum frequency in the message signal, $m(t)$ (*i.e.* Sampling Theorem applies). This sampling process effectively produces a pulse train, which requires a bandwidth much greater than B.

Thus in TDM, the message signals occupy a wide bandwidth for short intervals of time. In the illustration above, the signals are shown as PAM (Pulse Amplitude Modulation) signals. In practice these are normally converted to digital signals before time division multiplexing.