II/IV B.Tech DEGREE EXAMINATION, AUG 2023

SCHEME OF EVALUATION

Analog Communication (20EC404)

SUBMITTED BY

K.KALPANA

ASSISTANT PROFESSOR

DEPARTMENT OF ECE

HEAD OF THE DEPARTMENT

Dr B.CHANDRA MOHAN

PROFESSOR & H.O.D

DEPARTMENT OF ECE

Name and Signature of External Examiner

II/IV B.Tech (Regular\Supplementary) DEGREE EXAMINATION

July/August,2023 | **Electronics and Communication Engineering**

Fourth Semester | **Analog Communications**

Time: Three Hours | **Maximum:** 70 Marks

Answer question 1 compulsory. **(14X1 = 14Marks)**

Answer one question from each unit. **(4X14=56 Marks)**

			CO	BL	M
1	a)	Define Amplitude Modulation (AM) ?	CO1	L1	1M
	b)	Write expression for total power in AM signal	CO1	L2	1M
	c)	How much power will be saved in DSB-SC compared to Conventional AM?	CO1	L2	1M
	d)	How much bandwidth is saved if we employ SSB-SC modulation scheme compare to AM?	CO2	L2	1M
	e)	Write the major application of VSB and SSB-SC modulation schemes?	CO2	L2	1M
	f)	Write the time domain equation of SSB-SC signals?	CO2	L2	1M
	g)	Write the mathematical equation for single tone Narrow Band FM signal ?	CO3	L1	1M
	h)	If the frequency deviation of FM is 75KHz, calculate the modulation index for the modulating signal with frequency 3KHz ?	CO3	L3	1M
	i)	Define Frequency modulation?	CO3	L2	1M
	j)	Define PWM signal with a neat diagram?	CO4	L2	1M
	k)	Draw De-emphasis circuit used in FM?	CO4	L2	1M
	l)	Define Capture effect in FM?	CO4	L2	1M
	m)	Draw the circuit diagram of Envelope detector?	CO1	L2	1M
	n)	Define Multiplexing?	CO2	L2	1M
		Unit-I			
2	a)	Describe the generation of AM wave by using square law modulator?	CO1	L2	7M
	b)	The output of an AM is given by $S(t) = \cos(3800\pi t) + 16\cos(4000\pi t) + 4\cos(4200\pi t)$. Find the message frequency, carrier frequency, modulation index	CO1	L3	7M
		(OR)			
3	a)	Draw the circuit for Envelope detector and explain its operation? Draw the necessary waveforms?	CO1	L2	7M
	b)	Describe the detection of DSB-SC using COSTAS loop by using neat diagram.	CO1	L2	7M
		Unit-II			
4	a)	Using pre-envelope concept derive the time domain equation of SSB-SC equation?	CO2	L3	8M
	b)	Explain Frequency Division Multiplexing with a neat block diagram?	CO2	L2	6M
		(OR)			
5	a)	With neat block diagram describe the generation methods of SSB-SC signals	CO2	L2	8M
	b)	Describe the principle of VSB transmission. What are its advantages over SSB?	CO2	L3	6M
		Unit-III			
6	a)	With the help of neat diagrams, explain Armstrong method for generation of WBFM.	CO3	L3	7M
	b)	An angle modulated signal is given by $s(t)=5\cos[2\pi 10^6 t+5\sin 2000\pi t]$. Determine (i) Phase deviation and frequency deviation (ii) Bandwidth	CO3	L3	7M
		(OR)			
7	a)	Deduce the expression for single tone WBFM signals and draw its spectrum?	CO3	L4	9M
	b)	Compare AM and NBFM?	CO3	L4	5M
		Unit-IV			
8	a)	Describe the Generation and detection of PPM signals with neat waveforms ?	CO4	L2	7M
	b)	Derive the expression for figure of merit of DSB-SC system using Coherent detector?	CO4	L3	7M
		(OR)			
9	a)	Compare PAM,PWM and PPM	CO4	L4	5M
	b)	Derive the Expression for Signal to Noise Ratio of FM?	CO4	L4	9M

1Mark Questions

a) Amplitude modulation is formally defined as a process in which the amplitude of the carrier wave c(t) is varied about a mean value, linearly with the message signal m(t).

b) $P_t=P_c(1+\mu^2/2)$

c) 66%

d) AM= $2f_m$ SSB-SC=f_m

e) VSB- TV signal SSB- Telephone or Voice Signals

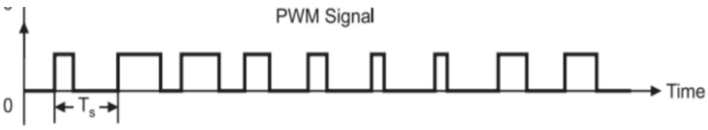
f) $s(t)=A_c/2[m(t)\cos(2\pi f_c t) - m(t) \sin(2\pi f_c t)]$

g) $s(t)=A_c \cos[2\pi f_c t + \beta\sin(2\pi f_m t)]$

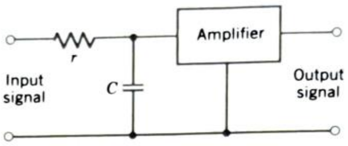
h) $\beta=75/3= 25$

i) Frequency modulation is defined as a process in which the frequency of the carrier wave c(t) is varied linearly with the message signal m(t).

j) Width of the pulse is varied with amplitude of message signal.

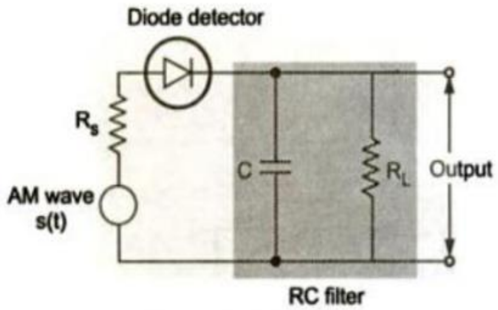


k)



l) When the interference is the stronger one of the two, the receiver locks on to the stronger signal and thereby suppresses the desired FM input. When they are of nearly equal strength, the receiver fluctuates back and forth between them. This phenomenon is known as the capture effect

m)



n) Multiplexing is a technique whereby a number of independent signals can be combined into a composite signal suitable for transmission over a common channel.

Unit – I

2a) Diagram-3M Explanation-4M

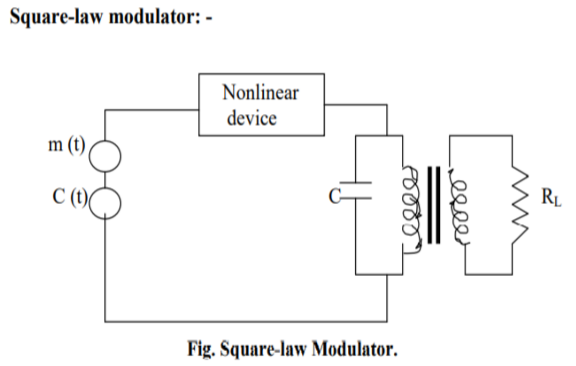


Fig. Square-law Modulator.

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.

$v_2(t)=a_1v_1(t)+a_2v_1^2(t)$ Where a_1, a_2 are constants

$v_1(t)= A_c\cos(2\pi f_ct)+m(t)$

The output of the non linear device is given by $v_2(t)=a_1v_1(t)+a_2v_1^2(t)$

$v_2(t)=a_1 [A_c\cos(2\pi f_ct)+m(t)]+a_2 [A_c\cos(2\pi f_ct)+m(t)]^2$

$[A_c\cos(2\pi f_ct)+m(t)]^2 =Ac^2\cos^2(2\pi f_ct)+m^2(t)+ 2A_c\cos(2\pi f_ct)m(t)$

The output at filter is given by $s(t)= a_1A_c\cos(2\pi f_ct)[1+2a_2 /a_1 m(t)]$

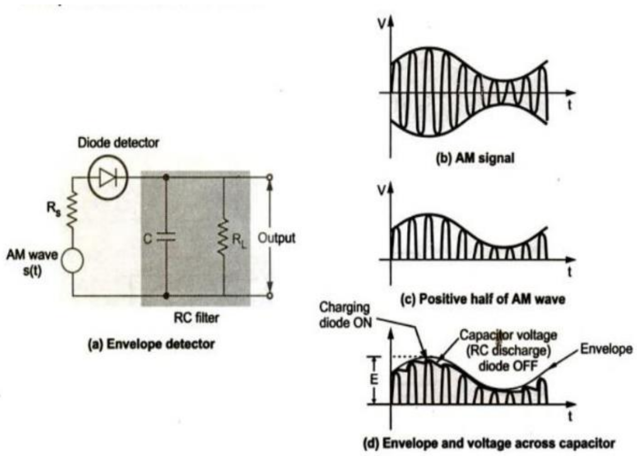
2b) Formulas-3M Answers-4M

$s(t) = \cos(3800\pi t) +16\cos(4000\pi t)+4\cos(4200\pi t)=A_c\cos(2\pi f_ct)[1+\mu\cos2\pi fm)t]$

message frequency=100Hz carrier frequency= 2000Hz modulation index=0.5

(OR)

3a) Diagram-3M Explanation-4M

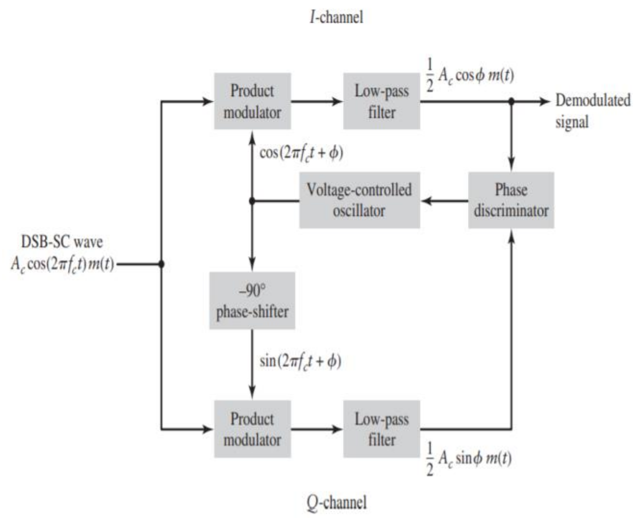


On positive half-cycle of RF input signal s(t) the diode is forward biased and the capacitor C charges up rapidly to the peak value of RF input signal. The charging time constant (R_s +r_f)C is very small when compared to the carrier period 1/fc i.e. (R_s +r_f)C<< $1/f_c$

When RF input falls below the output voltage then the diode becomes reverse-biased and the capacitor C discharges slowly through the load resistor R_l

If $1/ f_c << R_lC << 1 /W$ then the average value of output voltage is equal to the message signal.

3b)(Diagram-3M Explanation-4M)



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.

The frequency of the local oscillator is adjusted to be the same as the carrier frequency.

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. These two detectors are coupled together to form a negative feedback system designed in such a way as to maintain the local oscillator in synchronism with the carrier wave. Local oscillator signal is of the same phase as the carrier wave used to generate the incoming DSBSC wave.

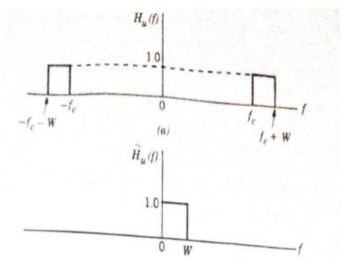
Unit – II

4a)(Expression-4M Explanation-4M)

The task of developing the time-domain description of SSB modulated waves is mathematically more difficult than that of standard AM or DSBSC modulated waves. To solve the problem, we use the idea of a complex envelope.
The DSBSC modulated wave is defined by $s(t)= A_c\cos(2\pi f_c t)m(t)$

Low-pass complex envelope of the DSBSC modulated wave is given by $\check{s}(t)= A_c m(t)$

Low-pass signal denote the complex envelope of then write $s_u(t)= \mathrm{Re}[\check{s}_u(t)\exp(j2\pi f_c t)]$



The band-pass filter of a transfer function $H_u(f)$ is replaced by an equivalent low-pass filter of transfer function $\check{H}_u(f)=1/2[(1+\mathrm{sgn}(f)]$ $0<f<W$. where sgn(f) is the signum function.

The DSBSC modulated wave is replaced by its complex envelope $\check{S}_{DSBSC}(f)= A_c M(f)$.The desired complex envelope $\check{s}_u(t)$ is determined by evaluating the inverse Fourier transform of the product $\check{H}_u(f)\ \check{S}_{DSBSC}(f)$. The message spectrum M(f) is zero outside the frequency interval $-W<f<W$. Therefore $\check{H}_u(f)\check{S}_{DSBSC}(f)= A_c M(f)/2[(1+\mathrm{sgn}(f)]$.

The Hilbert transform of m(t) = -jsgn(f) M(f)

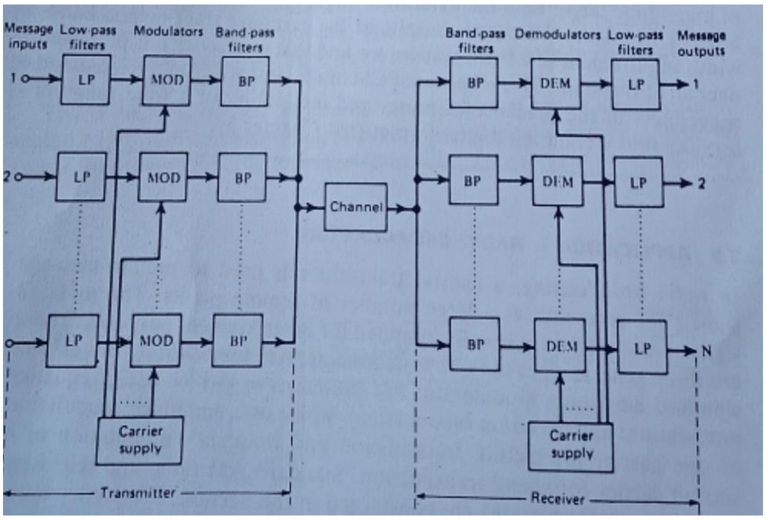
Accordingly, the inverse Fourier transformation $\check{s}_u(t)= A_c[m(t)+jm(t)]$

$$s_u(t) = \frac{A_c}{2}[m(t)\cos(2\pi f_c t) - \hat{m}(t)\sin(2\pi f_c t)]$$

Except for a scaling factor, a modulated wave containing only an upper sideband has an in-phase component equal to the message signal m(t) and a quadrature component equal to m(t)

4b)(Expression-3M Explanation-3M)

Separating the signals in frequency is referred as frequency division multiplexing (FDM).



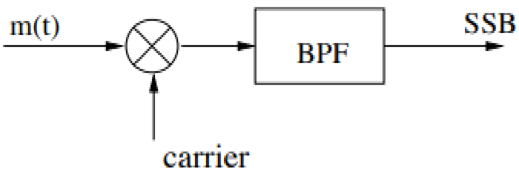
The incoming message signals are assumed to be of the low-pass type, but their spectra do not necessarily have nonzero values all the way down to zero frequency. Following each signal input, we have shown a low-pass filter, which is designed to remove high-frequency components that do not contribute significantly to signal representation but are capable of disturbing other message signals that share the common channel.

The filtered signals are applied to modulators that shift the frequency ranges of the signals so as to occupy mutually exclusive frequency intervals. The necessary carrier frequencies, to perform these frequency translations, are obtained from a carrier supply. For the modulation single-sideband modulation, which requires a bandwidth that is approximately equal to that of the original message signal is used. The band-pass fitters following the modulators are used to restrict the band of each modulated wave to its prescribed range. The resulting band-pass filter outputs are combined in parallel to form the input to the common channel. At the receiving terminal. a bank of band-pass filters, with their inputs connected in parallel. is used to separate the message signals on a frequency-occupancy basis. Finally. the original message signals are recovered by individual demodulators.

(OR)

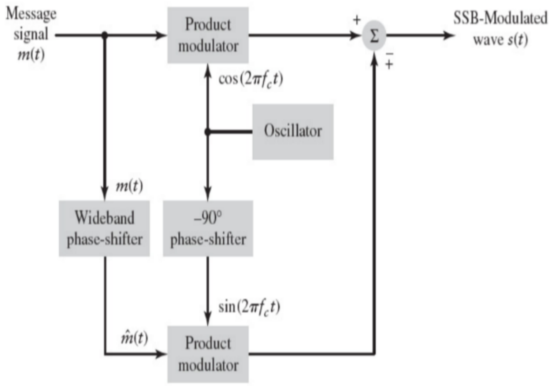
5a) Diagram-4M Explanation-4M

Frequency discrimination method: It is the most common method of generation SSB.



The basic idea is the following, Using m(t) generate DSB-SC then passing through BPF we can obtain SSB

For successful implementation of this method, we must have B << fc

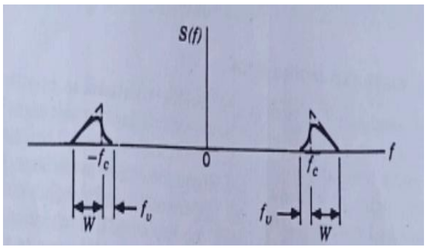


The second method for SSB generation, called the phase discrimination method.

- 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 phase-shifter
- However, the one functional block that requires special attention is the wide-band phase-shifter, which is designed to produce the Hilbert transform in response to the incoming message signal.

5b) Explanation-6M

The spectra of wideband signals contain significant low frequencies, which make it impractical to use SSB modulation. However, DSBSC requires a transmission bandwidth equal to twice the message bandwidth, which violates the bandwidth conservation requirement. To overcome these two practical limitations, VSB is used which is a compromise between SSB and DSB-SC .The transmission bandwidth of a VSB modulated signal is f_v+W where f_v is the vestige bandwidth 25 percent of W and W is the message bandwidth.



Advantages

Generation is easy as filter design is easy.

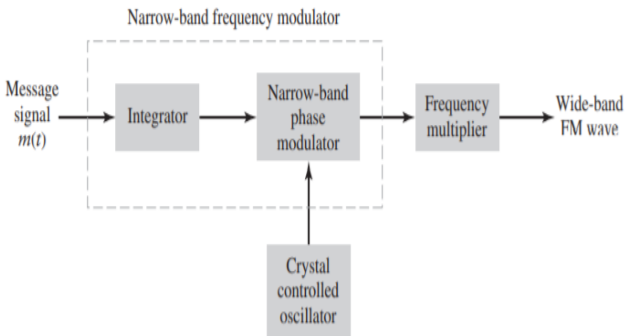
The transmission of low frequency components is possible, without difficulty.

Highly efficient.

Good phase characteristics.

Unit III

6a) Diagram-3M Explanation-4M



The message signal is first used to produce a narrow-band FM, which is followed by frequency multiplication to increase the frequency deviation to the desired level. The message signal is first integrated and then used to phase-modulate a crystal-controlled oscillator.Basically a frequency multiplier consists of a nonlinear device followed by a bandpass filter. At the output of frequency multiplier, there is a d.c.component and n frequency modulated waves with carrier frequencies f_c,2 f_c, n f_c and frequency deviations Δ f_1, 2Δ f_1................ nΔ f_1 respectively. The band-pass filter following the nonlinear device is used :To pass the FM wave centered at the carrier frequency n f_1 and with frequency deviation of nΔ f_1 and to attenuate completely all remaining FM spectra.

6b) Formulas-3M Answers-4M

An angle modulated signal is given by $s(t)=5Cos[2\pi10^6t+5Sin2000\pi t]$.

$s(t)=A_cCos[2\pi f_ct+\beta Sin2\pi f_mt]$

Phase deviation=β=5rad

Frequency deviation= $\beta*f_m=5*1000=5000$Hz Bandwidth $=2(\beta+1)f_m=12$KHz

(OR)

7a) Deivation-7M Spectrum-2M

The FM wave for sinusoidal modulation is given by $s(t)=A_cCos[2\pi f_ct+\beta Sin2\pi f_mt]$
Using a well-known trigonometric identity, $s(t)=A_c\cos(2\pi f_ct)\cos[\beta Sin2\pi f_mt]-A_c\sin(2\pi f_ct)\sin[\beta Sin2\pi f_mt]$
Hence. the complex envelope of the FM wave equals $\check{s}(t)= s_I(t)+js_Q(t)=A_c\exp[j\beta Sin2\pi f_mt]$

The complex envelope $\check{s}(t)$ retains complete information about the modulation process s(t) in terms complex envelope $\check{s}(t)=Re[A_c\exp(j2\pi f_ct\ )+j\beta\sin(2\pi f_mt)]=\ Re[\check{s}(t)\exp(j2\pi f_ct\ )]$

complex envelope is a periodic function with a fundamental frequency equal to the modulation frequency We may therefore

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j2\pi n f_m t)$$

where the complex Fourier coefficient

$$c_n = f_m \int_{-1/(2f_m)}^{1/(2f_m)} \tilde{s}(t) \exp(-j2\pi n f_m t)\, dt$$

$$= f_m A_c \int_{-1/(2f_m)}^{1/(2f_m)} \exp[j\beta \sin(2\pi f_m t) - j2\pi n f_m t]\, dt$$

Define the new variable:

$$x = 2\pi f_m t$$

:

$$c_n = \frac{A_c}{2\pi}\int_{-\pi}^{\pi} \exp[j(\beta \sin x - nx)]\, dx \qquad (4.25)$$

The integral on the right-hand side of Eq. (4.25), except for the carrier amplitude A_c, is referred to as the nth *order Bessel function* of the first kind and argument β. This function is commonly denoted by the symbol $J_n(\beta)$, so we may write

$$J_n(\beta) = \frac{1}{2\pi}\int_{-\pi}^{\pi} \exp[j(\beta \sin x - nx)]\, dx \qquad (4.26)$$

Accordingly, we may rewrite Eq. (4.25) in the compact form

$$c_n = A_c J_n(\beta) \qquad (4.27)$$

Substituting Eq. (4.27) into (4.22), we get, in terms of the Bessel function $J_n(\beta)$, the following expansion for the complex envelope of the FM wave:

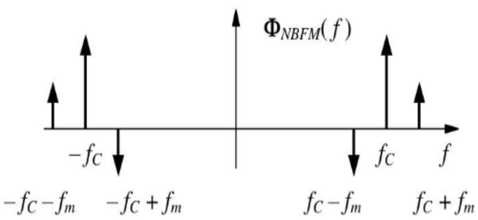
$$\tilde{s}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j2\pi n f_m t) \qquad (4.28)$$

Next, substituting Eq. (4.28) into (4.20), we get

$$s(t) = \mathbf{Re}\left[A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp[j2\pi(f_c + nf_m)t]\right] \qquad (4.29)$$

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

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)]$$



For narrowband FM $s(t)=A_c\cos(2\pi f_c t)+\beta A_c/2 \cos[2\pi(f_c+f_m)t]- \beta A_c/2 \cos[2\pi(f_c-f_m)t]$

7b) Differences-5M

For narrowband FM

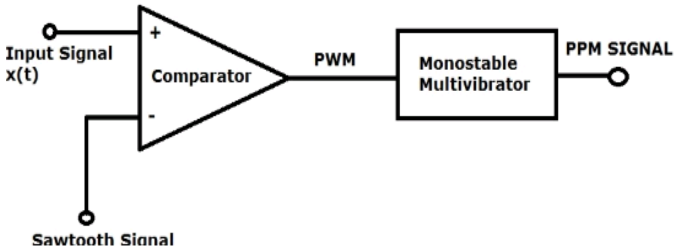
$$s(t) \approx A_c \cos(2\pi f_c t) + \frac{1}{2}\beta A_c\{\cos[2\pi(f_c + f_m)t] - \cos[2\pi(f_c - f_m)t]\}$$

For AM

$$s_{\mathrm{AM}}(t) = A_c \cos(2\pi f_c t) + \frac{1}{2}\mu A_c\{\cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t]\}$$

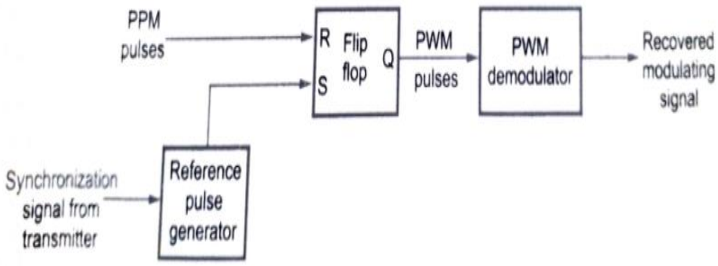
The basic difference between an AM wave and a narrow-band FM wave is that the algebraic sign of the lower side-frequency in the narrow-band FM is reversed. Narrow-band FM wave requires essentially the same transmission bandwidth as the AM wave.

8a) Diagrans-3M Explanation-4M

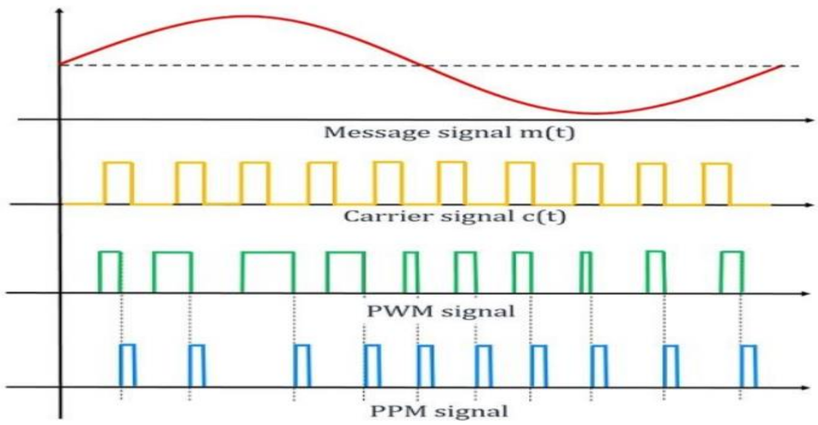


The trailing edge of the PWM signal is used as trigger input for the monostable multivibrator to generate PPM Signal.

To convert the received pulses that vary in position to pulses that vary in length PPM demodulator is shown below



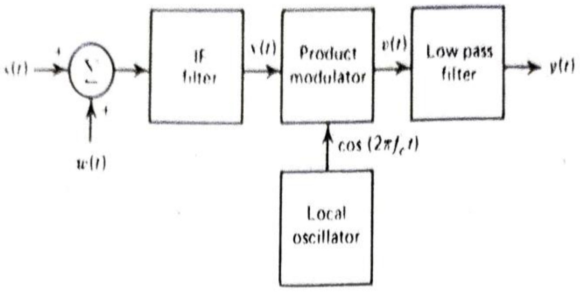
Flipflop sets when pulse arrives. This reference pulse is generated by reference pulse generator of the receiver with the synchronization signal from the transmitter. The flipflop is reset at the leading edge of the PPM pulse which generates PWM signal which is then demodulated by PWM demodulator.



8b) Diagrans-2M Expression-5M

Consider a DSBSC wave s(t) = $A_c\cos(2\pi f_c t)m(t)$

$$\mathrm{SNR}^{\mathrm{DSB}}_{\mathrm{pre}} = \frac{A_c^2 P}{2N_0 B_T}$$



Using the narrow-band representation of the filtered noise n(t), the total signal at the coherent detector input may be expressed as $x(t) = A_c\cos(2\pi f_c t)m(t) + n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t)$

The low pass filter output $y(t) = 1/2A_c m(t) + 1/2n_I(t)$

$$\mathrm{SNR}_{\mathrm{post}}^{\mathrm{DSB}} = \frac{\frac{1}{4}(A_c^2)P}{\frac{1}{4}(2N_0W)} = \frac{A_c^2P}{2N_0W}$$

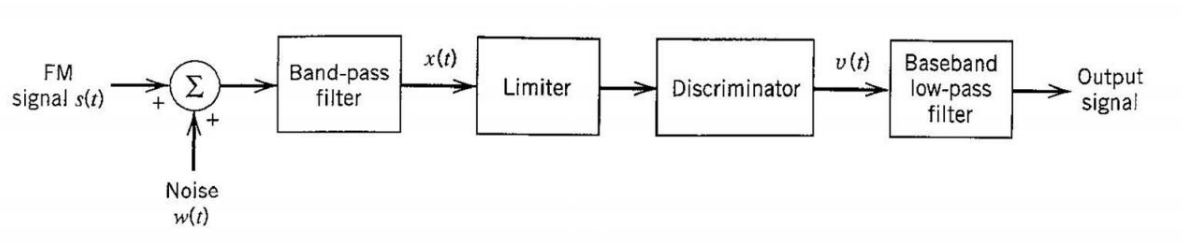
$$\text{Figure of merit} = \frac{\mathrm{SNR}_{\mathrm{post}}^{\mathrm{DSB}}}{\mathrm{SNR}_{\mathrm{ref}}} = 1$$

(OR)

9a) Any 5 differences-5M

PAM	**PWM/PDM**	**PPM**
Amplitude of the pulse is proportional to the amplitude of modulating signal	Width of the pulse is proportional to amplitude of modulating signal.	The relative position of the pulse is proportional to the amplitude of modulating signal.
The bandwidth of the transmission channel depends on width of the pulse	Bandwidth of transmission channel depends on rise time of the pulse.	Bandwidth of transmission channel depends on rise time of the pulse.
The instantaneous power of the transmitter varies with amplitude of pulses.	The instantaneous power of the transmitter varies with width of pulses	The instantaneous power of the transmitter remains constant with width of pulses.
Noise interference is high	Noise interference is minimum	Noise interference is minimum
System is complex	Simple is implement	Simple is implement
Similar to Amplitude modulation	Similar to frequency modulation	Simple to Phase modulation

9b) Diagrans-2M Expression-4M



$$\mathrm{SNR}_{\mathrm{pre}}^{\mathrm{FM}} = \frac{A_c^2}{2N_0 B_T}$$

The noisy FM signal after band-pass filtering may be represented as

$$x(t) = s(t) + n(t) \tag{9.41}$$

where $s(t)$ is given by Eq. (9.40). In previous developments, we have expressed the filtered noise $n(t)$ at the band-pass filter output in Fig. 9.13 in terms of its in-phase and quadrature components

$$n(t) = n_I(t)\cos(2\pi f_c t) - n_Q(t)\sin(2\pi f_c t) \tag{9.42}$$

We may equivalently express $n(t)$ in terms of its envelope and phase as (see Problem 4.3.)

$$n(t) = r(t)\cos[2\pi f_c t + \phi_n(t)] \tag{9.43}$$

where the envelope is

$$r(t) = [n_I^2(t) + n_Q^2(t)]^{1/2} \tag{9.44}$$

and the phase is

$$\phi_n(t) = \tan^{-1}\left(\frac{n_Q(t)}{n_I(t)}\right) \tag{9.45}$$

$$v(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt}$$
$$= k_f m(t) + n_d(t)$$

where the noise term $n_d(t)$ is defined by

$$n_d(t) = \frac{1}{2\pi A_c}\frac{dn_Q(t)}{dt}$$

$$\mathrm{SNR}_{\mathrm{post}}^{\mathrm{FM}} = \frac{3A_c^2 k_f^2 P}{2N_0 W^3}$$

$$\text{Figure of merit} = \frac{\mathrm{SNR}_{\mathrm{post}}^{\mathrm{FM}}}{\mathrm{SNR}_{\mathrm{ref}}} = \frac{\dfrac{3A_c^2 k_f^2 P}{2N_0 W^3}}{\dfrac{A_c^2}{2N_0 W}}$$
$$= 3\left(\frac{k_f^2 P}{W^2}\right)$$
$$= 3D^2$$