AKGEC/IAP/FM/02

Ajay Kumar Garg Engineering College, Ghaziabad

Department of ECE

Sessional Test-2

Course:	B.Tech	Semester: VII	
Session:	2017-18	Section:	EN-1,2
Subject:	Analog & Digital Communication	Sub. Code:NEC-702 A	
Max Marks: 50		Time:	2 hour

Note : Answer **all** the sections.

Section A.

Ques1. Differentiate NBFM & WBFM.

Soln. NBFM. small values of the modulation index 'β' (compared to one radian) represents Narrow Band FM. It consist of a carrier, an upper side band frequency component and a Lower side band freq. component.

WBFM: Wide Band FM for larger values of 'β' as compared to one radian. This FM contains a carrier and an infinite no. of side freq components located symmetrically around the carrier.

Ques2. Define frequency sensitivity of frequency modulator & a Phase modulator.

Soln: K_p is defined as the Phase sensitivity of the modulator which is expressed in Radians/volt.

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

K_f is defined as the Frequency sensitivity of a modulator, expressed in Hertz per volt.

$$f_i(t) = f_c + K_f m(t).$$

Ques 3. An analog signal is sampled at 36 kHz and quantized into 256 levels. Find the time duration of a bit of the binary coded signal is?

Soln: Sampling freq = 36 kHz
= 36,000 Hz.

i.e. 36,000 samples per second.

No. of quantization levels = 256 i.e. M = 256

$2^n = M \Rightarrow n = 8$.

i.e. Eight bit encoder is used, i.e. per second

no. of bits = 36,000 × 8 = 288,000 bits per second

∴ Bit duration = $\frac{1}{288,000}$ = 3.47×10^{-6} sec

i.e. 3.47 μsec.

Ques. 4. Define Carson's Rule.

Soln. Carson's Rule defines an approximate rule for the transmission bandwidth of an FM signal generated by a single tone modulating signal of frequency f_m as follows:

$$B_T \simeq 2\Delta f + 2f_m = 2\Delta f \left(1 + \frac{1}{\beta}\right)$$

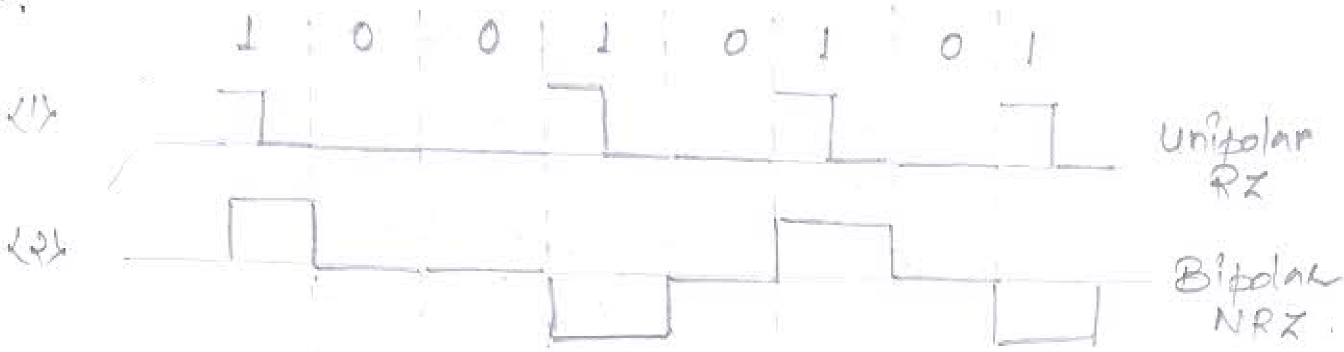
This emperical relation is known as Carson's rule.

Ques 5. Apply (1) Unipolar Return-to-Zero and
(2) Bipolar NRZ

18000

Line coding techniques on bit string [1 0 0 1 0 1 0 1]

Soln.



"Section B"

Ques.6. What do you mean by Angle modulation? Derive a relationship between FM and PM using suitable expressions and Block diagrams.

Soln. In angle modulation, either the phase or frequency of the carrier wave is varied according to the message signal.

$$s(t) = A_c \cos[\theta(t)]$$

$A_c \rightarrow$ constant, angular argument $\theta(t)$ is varied by a message signal $m(t)$.

There are two ways to vary carrier angle w.r.t. message.

(i) Phase modulation – Is that form of angle modulation in which the angular argument $\theta(t)$ is varied linearly with the message signal $m(t)$

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

$K_p \rightarrow$ phase sensitivity of the modulation (Radians/Volts)

(2) Frequency modulation (FM) - is that form of angle modulation in which the instantaneous freq $f_i(t)$ is varied linearly with the message signal $m(t)$

$$f_i(t) = f_c + k_f\, m(t)$$

where k_f → frequency sensitivity of the modulator (Hertz per volt)

⇒ A Block diagram to represent relationship b/w FM & PM

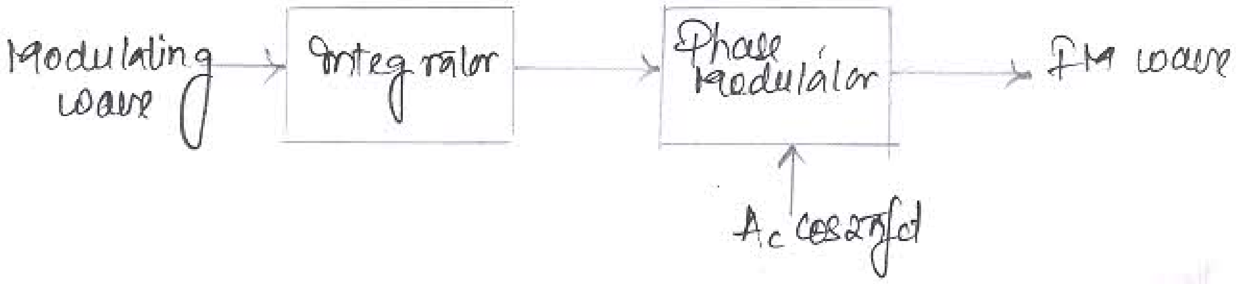


fig 6.1 Block diag. of FM modulator

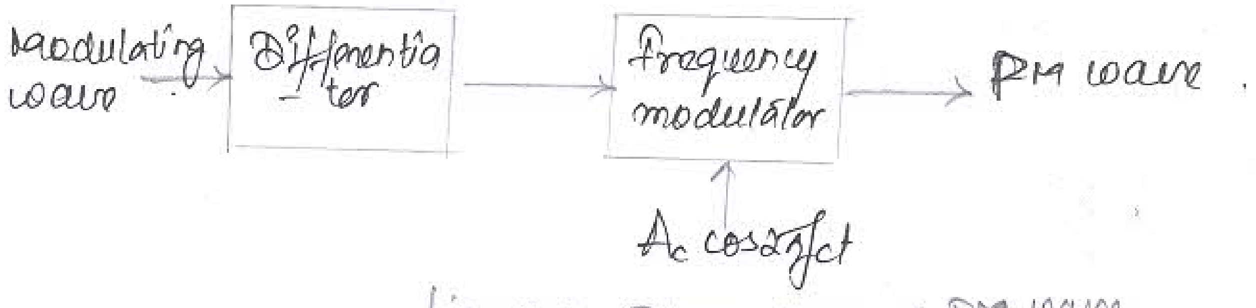


fig. 6.2 Block diag. of PM wave modulator

Que 7. What are the advantages of Digital communication over Analog communication?

An FM signal has a resting frequency of 210 MHz and highest frequency of 210.04, when modulated by a signal of frequency of 20 kHz. Determine

(i) Frequency Deviation
(ii) Carrier swing
(iii) Modulation index
(iv) Percent modulation
(v) Lowest frequency reached by the FM.

Soln. Advantages of Digital communication system over Analog

i) Digital communication systems are comparitively Immune to noise.

ii) Digital communication systems are simpler & cheaper compared to analog communication systems and require lesser memory for storage.

iii) Data encryption & multiplexing can easily be implemented.

iv) Channel coding usage helps in error detection & correction.

v) Digital communication is adaptive to other branches of data processing such as digital signal processing, image processing & data compression etc.

Numerical

i) Frequency deviation, $\Delta f = 210.04 - 210\ MHz$
$= 0.04\ MHz$.

ii) Carrier swing, $\Rightarrow 2\Delta f \Rightarrow 2 \times 0.04$
$= 0.08\ MHz$

iii) Modulation index, $\beta = \frac{\Delta f}{f_m} = \frac{0.04\ MHz}{20\ KHz}$
$\boxed{\beta = 2}$

iv) Percent modulation $= \frac{\Delta f}{(\Delta f)_{max}} \times 100 = \frac{40}{75} \times 100$
$= 53.3\ \%$

v) Lowest frequency reached $= 210 - 0.04\ MHz$
$= 209.96\ MHz$.

Ques 8. What do you mean by SNR? Analyze noise performance of AM receiver using Envelope Detection.

Soln. SNR - signal to noise ratio is defined as the ratio of the average power of the message signal to the average power of the noise, both measured at the receiver output.
SNR is considered for analysing noise performance of a system.

Noise in AM Receivers using Envelope Detection

$$s(t) = A_c(1 + K_a m(t)) \cos(2\pi f_c t)$$

Front end of receiver is identical to that of coherent receiver.

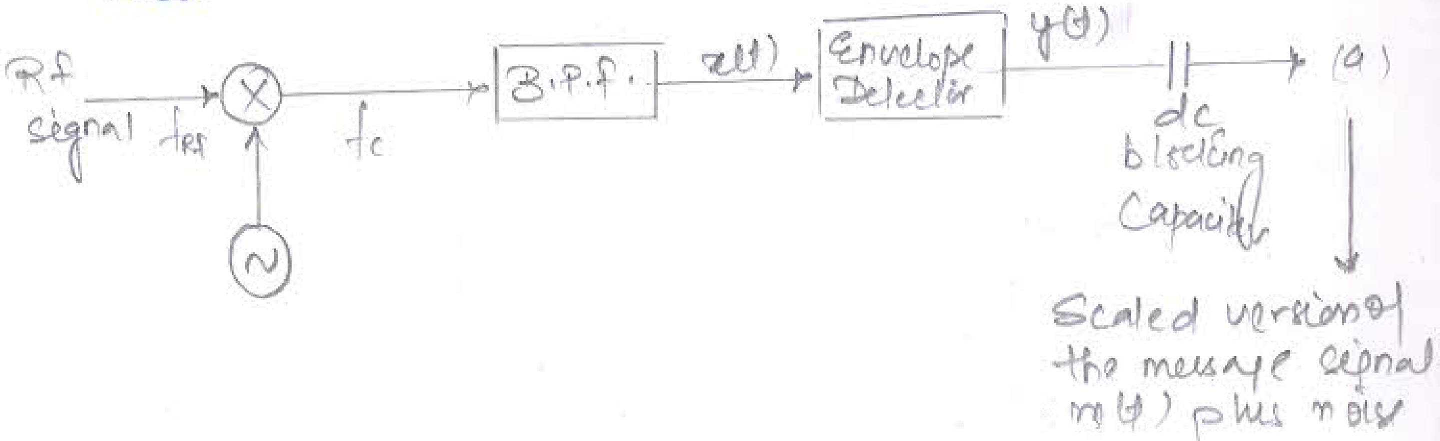


fig 8.1 model for AM rx using envelope detection.

Pre detection SNR

Carrier power $= \frac{A_c^2}{2}$, power in modulated part of the signal.

$$E[(1 + k_a m(t))^2] = E[1 + 2K_a m(t) + k_a^2 m^2(t)]$$
$$= 1 + 2K_a E[m(t)] + K_a^2 E[m^2(t)]$$
$$= 1 + k_a^2 P$$

We assume that the message signal has zero mean.
$E[m(t)] = 0$

Received signal power $A_c^2(1+K_a^2P)/2$. As linear rx. we assume that, without loss of generality that the gain of BPF is unity.

$$SNR^{AM}_{pre} = \frac{A_c^2(1+K_a^2P)}{2N_0B_T}$$, where $B_T \rightarrow$ noise Bandwidth of the BPF

Post Detection SNR

i/p to envelop detector

$$x(t) = s(t) + n(t)$$
$$= [A_c + A_cK_a m(t) + n_I(t)]\cos(2\pi f_c t) - n_Q(t)\sin 2\pi f_c t$$

o/p of envelope detector is the amplitude of the phasor representing $x(t)$

$$y(t) = \text{envelope of } x(t)$$
$$= \left\{ [A_c(1+K_a m(t)) + n_I(t)]^2 + n_Q^2(t) \right\}^{1/2}$$

assume signal is much larger than noise,

$$\therefore y(t) \simeq A_c + A_c K_a m(t) + n_I(t)$$

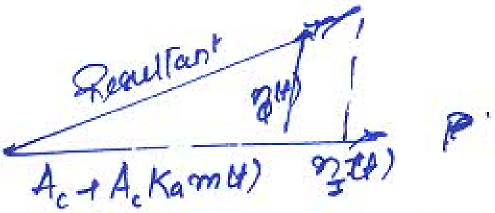


fig. Phasor diag. for AM wave + narrowband noise

$$SNR^{AM}_{post} = \frac{A_c^2 K_a^2 P}{2N_0W}$$

$$\text{Figure of merit} = \frac{SNR_{post}}{SNR_{pre}} = \frac{K_a^2P}{1+K_a^2P}$$

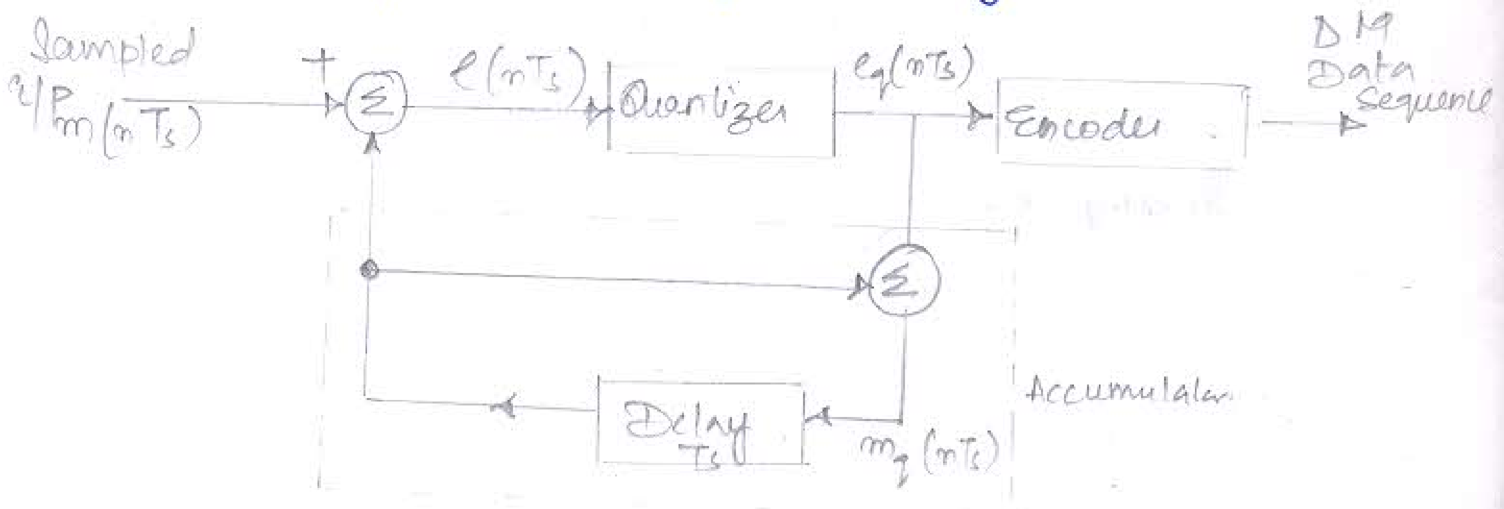
as $K_a^2 P < 1$. (otherwise the signal will be over modulated)

FOM is always less than 0.5

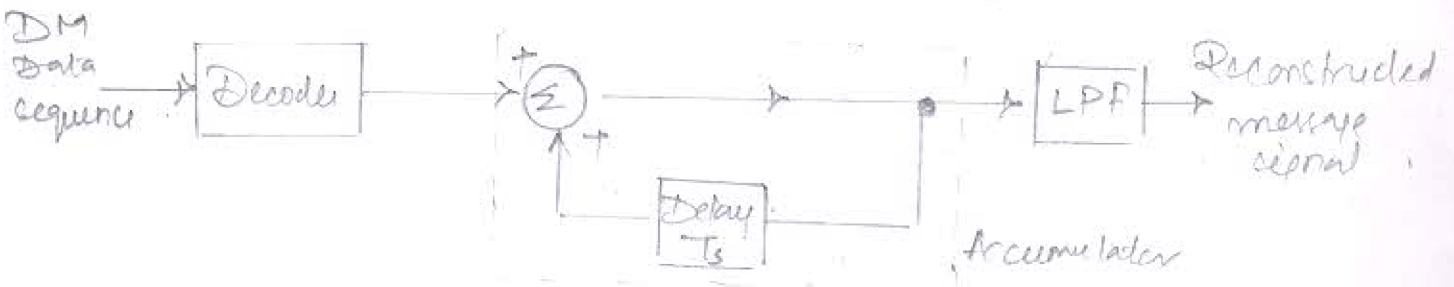
Ques 9. What is Delta modulation. Explain about the types of Quantization noise.

Soln: In Delta modulation, an incoming message signal is oversampled at a rate higher than Nyquist rate, to purposely increase the correlation between adjacent samples of the signal.

DM provides a staircase approximation to the oversampled version of the message signal.



(a) Transmitter



(b) Receiver

Quantization Noise

is defined as the difference between the input signal 'm' and the output signal 'v'. The error is called quantization noise. These are of two types

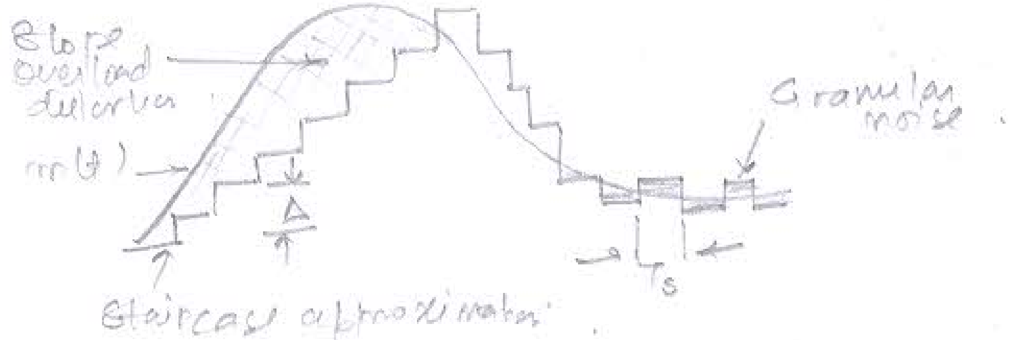


fig: Illustration of Noise in Delta modulation.

i) Slope overload distortion:
when step-size Δ is too small for the staircase approximation $m_q(t)$ to follow a steep segment of the input waveform $m(t)$, with that result that $m_q(t)$ falls behind $m(t)$.
This condition is called slope overload.

ii) Granular noise: It occurs when the step size Δ is too large relative to the local slope characteristics of the input waveform $m(t)$, thereby causing the staircase approximation $m_q(t)$ to hunt around a relatively flat segment of the input waveform.

Ques 10. Explain Sampling theorem. Draw & explain the working of Sample & Hold circuit.

Solu. Sampling Theorem:

A continuous - time signal may be completely represented in its samples & recorded back, if the sampling frequency is $f_s \geq 2f_m$ where f_s is the sampling frequency and f_m is the maximum frequency present in the signal.

Sample & Hold circuit

circuit can be made by combination of basic 2, FET's and a capacitor

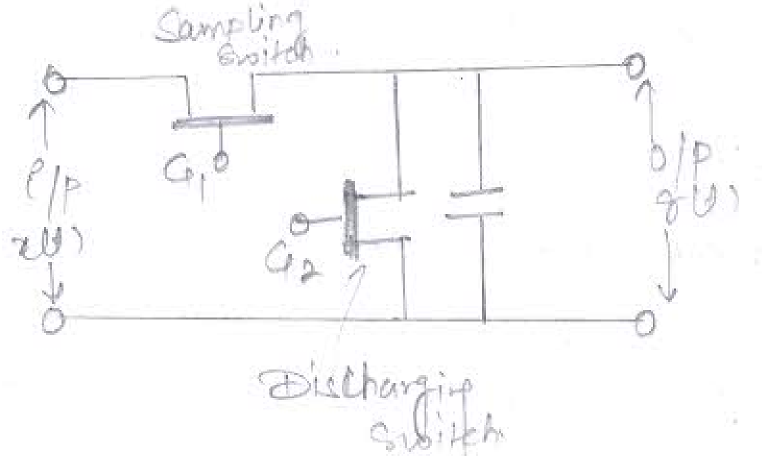


fig. Sample & Hold ckt

Operation: The sampling switch is closed for a short duration by a short pulse, applied to the gate G_1 of FET. During this period, the capacitor C is quickly charged upto a voltage equal to the instantaneous sample value of the incoming signal x(t)

- Now the sampling switch is opened & the capacitor 'C' holds the charge.

This discharge switch is then opened and thus capacitor C has no voltage.

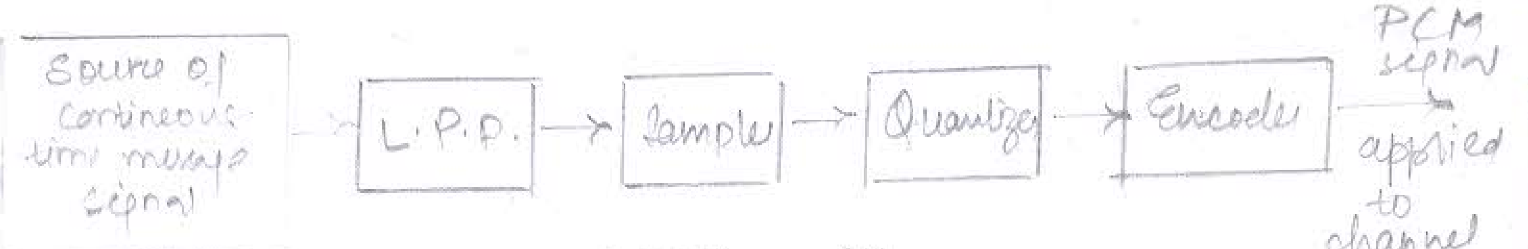
- Thus the output of S/H circuit consists of a sequence of flat top samples.

Section C.

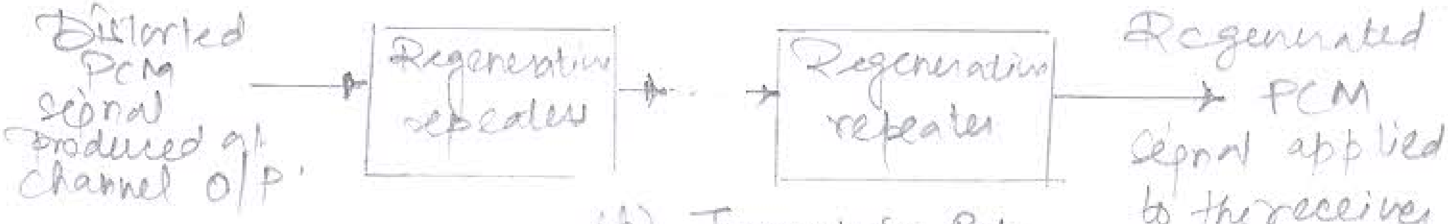
Ques 11. What is PCM? Draw the block diagram representing all the elements of a PCM system and explain functionality of each block.

Soln. PCM - Pulse code modulation, here message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time & amplitude.

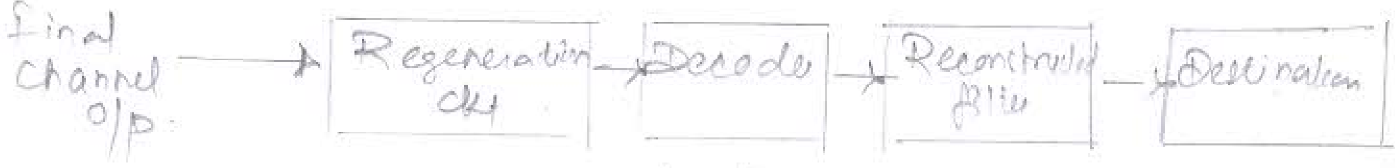
+ Basic operations performed in the transmitter of a PCM system are Sampling, Quantization & Encoding.



(a) Transmitter



(b) Transmission Path



(c) Receiver.

Sampling : The incoming message signal is sampled with a train of narrow rectangular pulses.

$f_s \geq 2f_m$.

An anti-aliasing filter is used at the front end.

Quantization : Sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time & amplitude.

Quantization & Encoding are in same same circuit called Analog to Digital converter.

Encoding : it translates the discrete set of sample values to a more appropriate form of signal. Different line coding techniques can be used for the same like unipolar, Bipolar, Manchester etc.

Receiver → Regeneration of impaired signals, decoding and reconstruction of the train of quantized samples occurs here.

Regeneration requires intermediate point, hence regenerative repeaters are used along the transmission path.

Ques 12. Explain the Indirect method for FM generation. Support your answers with suitable block diagram.

Soln. Generation of FM waves using indirect method

Modulating wave is first used to produce a narrow-band FM wave, & freq multiplication is next used to increase the frequency deviation to the desired level.

Indirect FM:

$$s(t) = A_1 \cos[2\pi f_1 t + \phi_1(t)]$$

$f_1 \rightarrow$ carrier freq
$A_1 \rightarrow$ carrier amplitude.

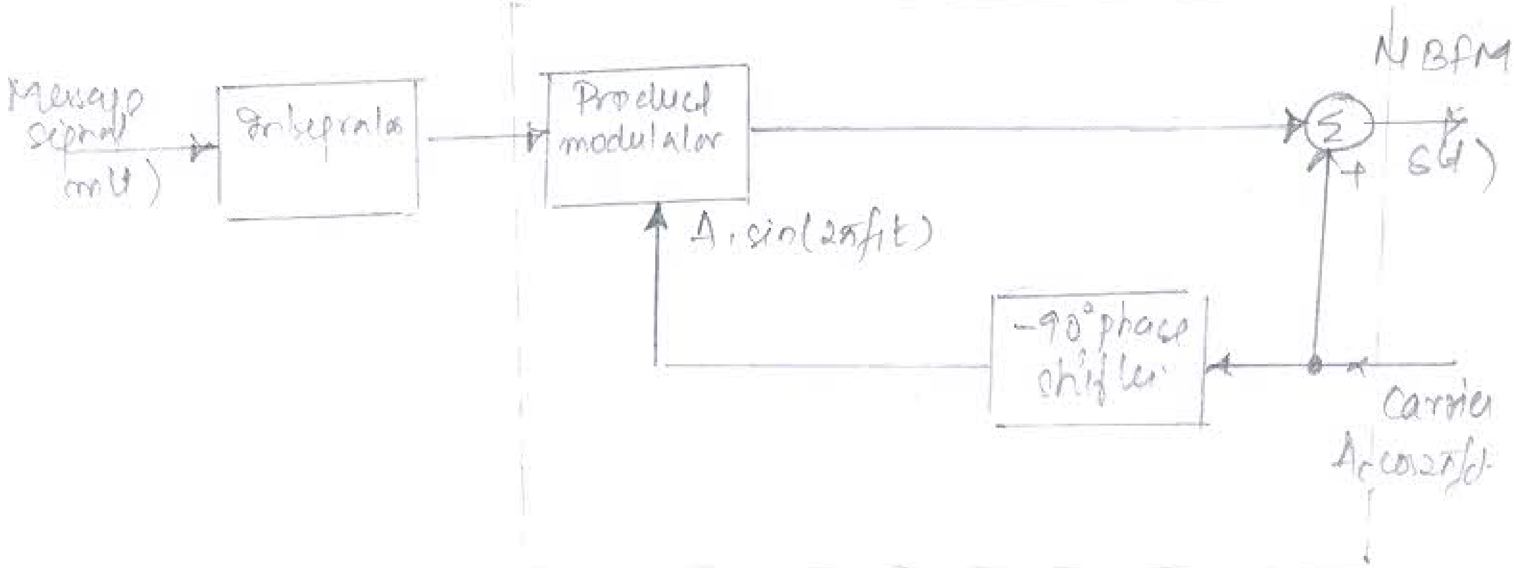
Angular argument $\phi(t)$ of $s_1(t)$ is related to $m(t)$ by $\phi_1(t) = 2\pi k_1 \int_0^t m(t)\,dt$, $k_1 \rightarrow$ freq. sensitivity of the modulator

Assume $\phi_1(t)$ is small,

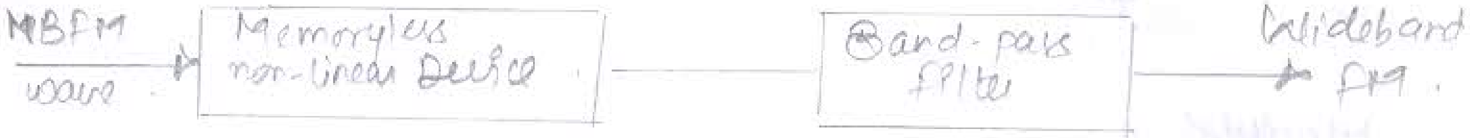
$$\cos[\phi(t)] \simeq 1$$
$$\sin[\phi(t)] \simeq \phi(t)$$

$$s(t) \simeq A\cos(2\pi f_1 t) - A_1 \sin(2\pi f_1 t)\,\phi_1(t)$$
$$= A_1\cos(2\pi f_1 t) - 2\pi k_1 A_1 \sin(2\pi f_1 t)\int_0^t m(t)\,dt$$

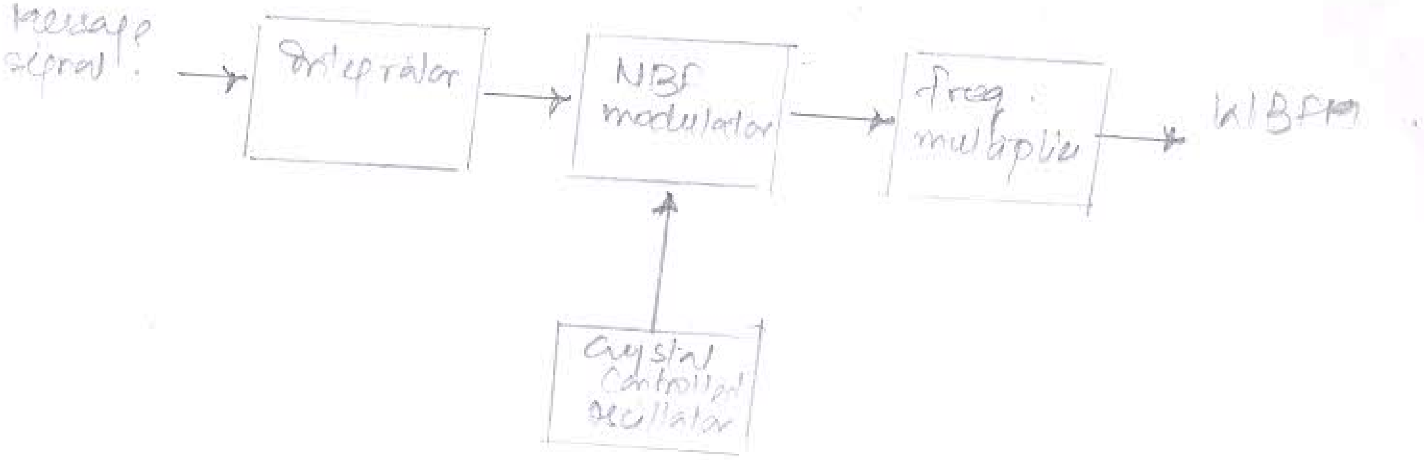
NBFM



(a) NBFM modulator



(b) freq. multiplier.



(c) WBFM modulator

Operation.

→ Envelope contains a residual amplitude modulation, & it varies with time for a sinusoidal modulating wave. The phase of the FM wave contains "Harmonic distortion" in the form of third & higher-order harmonics of the modulation freq. f_m.

→ Freq. multiplier consists of a non linear device followed by a BPF.

Non linear device is memoryless.

$s_o(t) = a_1 s_1(t) + a_2 s_1^2(t) + \dots + a_n s_1^n(t)$

expanding we find $s_2(t)$ has a dc component

FM has $f_1, 2f_1, \dots nf_1$

$a_1, a_2 \dots a_n$ are constant. coef. & deviations Δf_1 & $2\Delta f_1$.

The value of Δf_1 is determined by sensitivity,