USN

Internal Assessment Test 4 – Feb. 2022

Sub	Principles of Communication Systems				Sub Code:	18EC53	Branch:	ECE
Date	02-02-2022	Duration:	90 Minutes	Max Marks:	50	Sem / Sec:	5/A, B, C, D	OBE

	<u>Answer any FIVE FULL Questions</u>	MARKS	CO	RBT
1	Derive the figure of merit for DSBSC receiver.	[10]	CO2	L3
2	With FM receiver model derive the figure of merit of FM receiver based on discriminator.	[10]	CO2	L3
3	Briefly explain the following as applicable to FM. (i) Capture effect (ii) Threshold effect. (iii) Pre-emphasis (iv) De-emphasis	[10]	CO2	L2
4	With neat diagram explain the basic elements of a PCM Transmitter	[10]	CO4	L2
5	I) Explain granular noise and slope overload distortion with reference to Delta modulation II) What is the significance of regenerative repeater in a PCM system. With the help of neat diagram explain the working of repeater.	[05] [05]	CO4 CO4	L2 L2
6	I) A compact disc (CD) records audio signals digitally using PCM. Assume the audio signal bandwidth to be 15 KHz. a. What is the Nyquist rate? b. If the Nyquist samples are quantized to L = 65, 536 levels and then binary coded, determine the number of bits required to encode a sample. c. Assuming that the signal is sinusoidal and that the maximum signal amplitude is 1 volt; determine the quantization step and the signal-to-quantization noise ratio. II) Explain the following line codes with suitable examples a) Manchester Code b) differential encoding	[05] [05]	CO4 CO4	L3 L2

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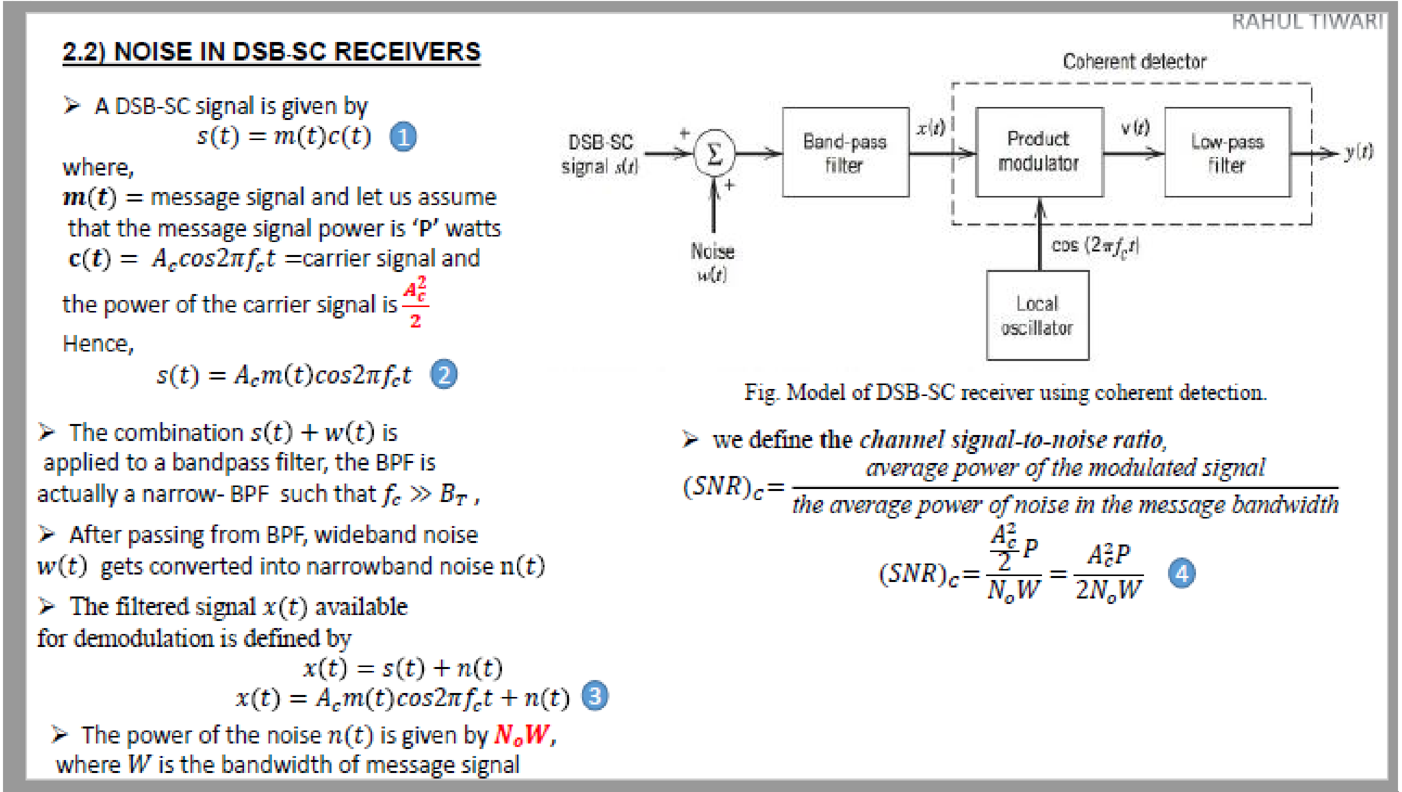
(1)



2.2) NOISE IN DSB-SC RECEIVERS

- A DSB-SC signal is given by

$$s(t) = m(t)c(t) \quad ①$$

where,

$\boldsymbol{m(t)}$ = message signal and let us assume that the message signal power is 'P' watts

$\mathbf{c(t)} = A_c cos2\pi f_c t$ =carrier signal and the power of the carrier signal is $\frac{A_c^2}{2}$

Hence,

$$s(t) = A_c m(t)cos2\pi f_c t \quad ②$$

- The combination $s(t) + w(t)$ is applied to a bandpass filter, the BPF is actually a narrow- BPF such that $f_c \gg B_T$,
- After passing from BPF, wideband noise $w(t)$ gets converted into narrowband noise $n(t)$
- The filtered signal $x(t)$ available for demodulation is defined by

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

$$x(t) = A_c m(t)cos2\pi f_c t + n(t) \quad ③$$

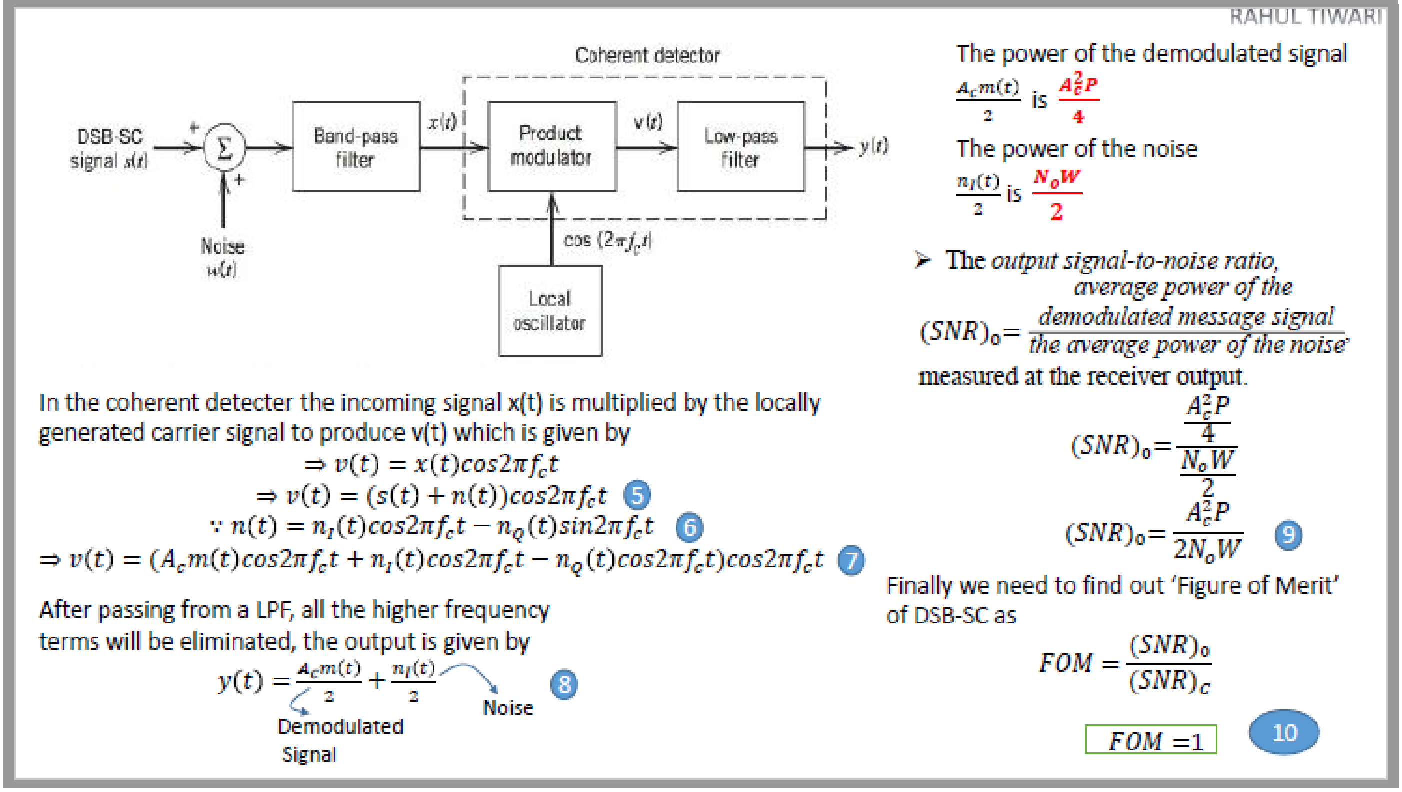
- The power of the noise $n(t)$ is given by $N_o W$, where W is the bandwidth of message signal

Fig. Model of DSB-SC receiver using coherent detection.

- we define the *channel signal-to-noise ratio,*

$$(SNR)_C = \frac{\textit{average power of the modulated signal}}{\textit{the average power of noise in the message bandwidth}}$$

$$(SNR)_C = \frac{\frac{A_c^2}{2}P}{N_o W} = \frac{A_c^2 P}{2N_o W} \quad ④$$



In the coherent detecter the incoming signal x(t) is multiplied by the locally generated carrier signal to produce v(t) which is given by

$$\Rightarrow v(t) = x(t)cos2\pi f_c t$$

$$\Rightarrow v(t) = (s(t) + n(t))cos2\pi f_c t \quad ⑤$$

$$\because n(t) = n_I(t)cos2\pi f_c t - n_Q(t)sin2\pi f_c t \quad ⑥$$

$$\Rightarrow v(t) = (A_c m(t)cos2\pi f_c t + n_I(t)cos2\pi f_c t - n_Q(t)cos2\pi f_c t)cos2\pi f_c t \quad ⑦$$

After passing from a LPF, all the higher frequency terms will be eliminated, the output is given by

$$y(t) = \underbrace{\frac{A_c m(t)}{2}}_{\text{Demodulated Signal}} + \underbrace{\frac{n_I(t)}{2}}_{\text{Noise}} \quad ⑧$$

The power of the demodulated signal $\frac{A_c m(t)}{2}$ is $\frac{A_c^2 P}{4}$

The power of the noise $\frac{n_I(t)}{2}$ is $\frac{N_o W}{2}$

- The *output signal-to-noise ratio,*

$$(SNR)_o = \frac{\textit{average power of the demodulated message signal}}{\textit{the average power of the noise}},$$

measured at the receiver output.

$$(SNR)_o = \frac{\frac{A_c^2 P}{4}}{\frac{N_o W}{2}}$$

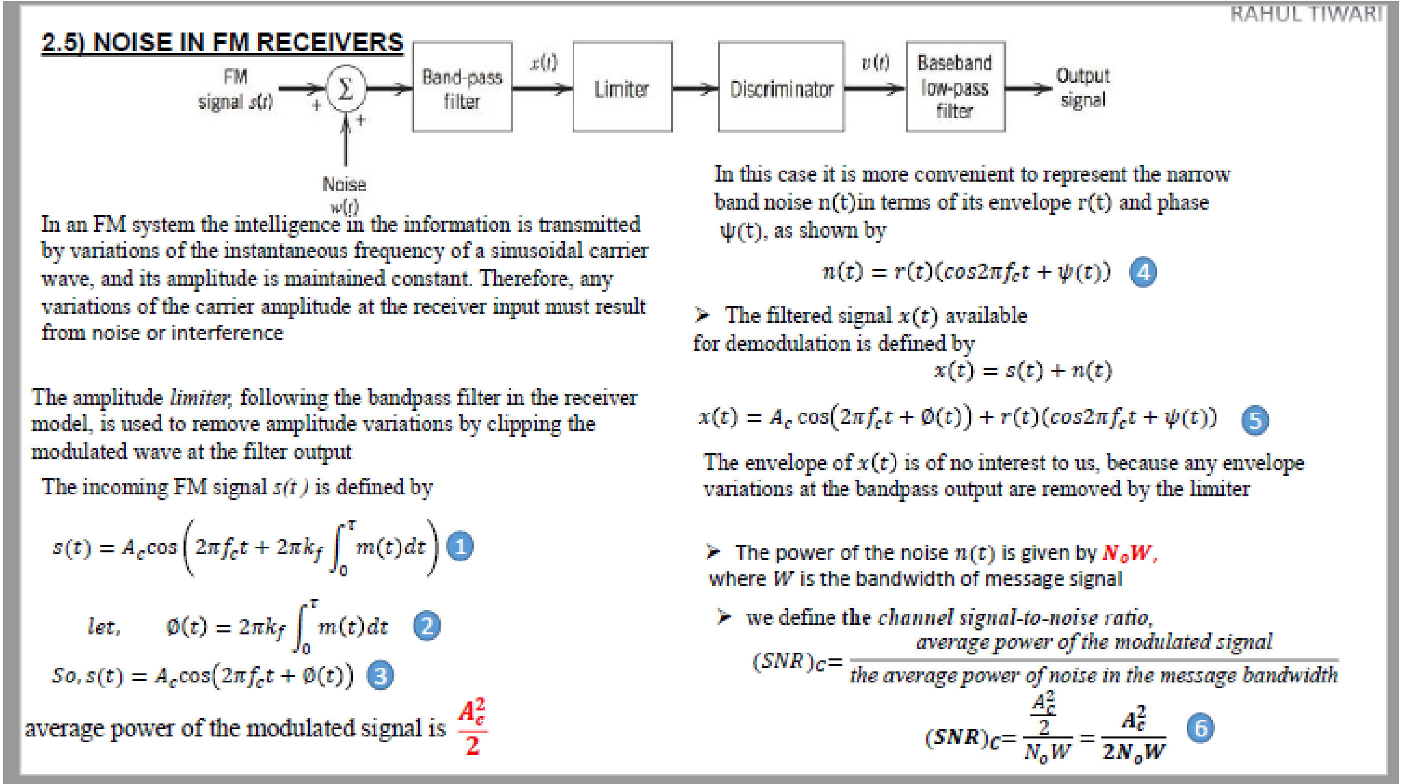
$$(SNR)_o = \frac{A_c^2 P}{2N_o W} \quad ⑨$$

Finally we need to find out 'Figure of Merit' of DSB-SC as

$$FOM = \frac{(SNR)_o}{(SNR)_C}$$

$$\boxed{FOM = 1} \quad ⑩$$

2)



2.5) NOISE IN FM RECEIVERS

In an FM system the intelligence in the information is transmitted by variations of the instantaneous frequency of a sinusoidal carrier wave, and its amplitude is maintained constant. Therefore, any variations of the carrier amplitude at the receiver input must result from noise or interference

The amplitude *limiter*, following the bandpass filter in the receiver model, is used to remove amplitude variations by clipping the modulated wave at the filter output

The incoming FM signal $s(t)$ is defined by

$$s(t) = A_c\cos\left(2\pi f_c t + 2\pi k_f\int_0^{\tau} m(t)dt\right)$$ ①

$$let,\quad \emptyset(t) = 2\pi k_f\int_0^{\tau} m(t)dt$$ ②

$$So, s(t) = A_c\cos(2\pi f_c t + \emptyset(t))$$ ③

average power of the modulated signal is $\frac{A_c^2}{2}$

In this case it is more convenient to represent the narrow band noise n(t)in terms of its envelope r(t) and phase ψ(t), as shown by

$$n(t) = r(t)(cos2\pi f_c t + \psi(t))$$ ④

- The filtered signal $x(t)$ available

for demodulation is defined by

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

$$x(t) = A_c\cos\big(2\pi f_c t + \emptyset(t)\big) + r(t)\big(cos2\pi f_c t + \psi(t)\big)$$ ⑤

The envelope of $x(t)$ is of no interest to us, because any envelope variations at the bandpass output are removed by the limiter

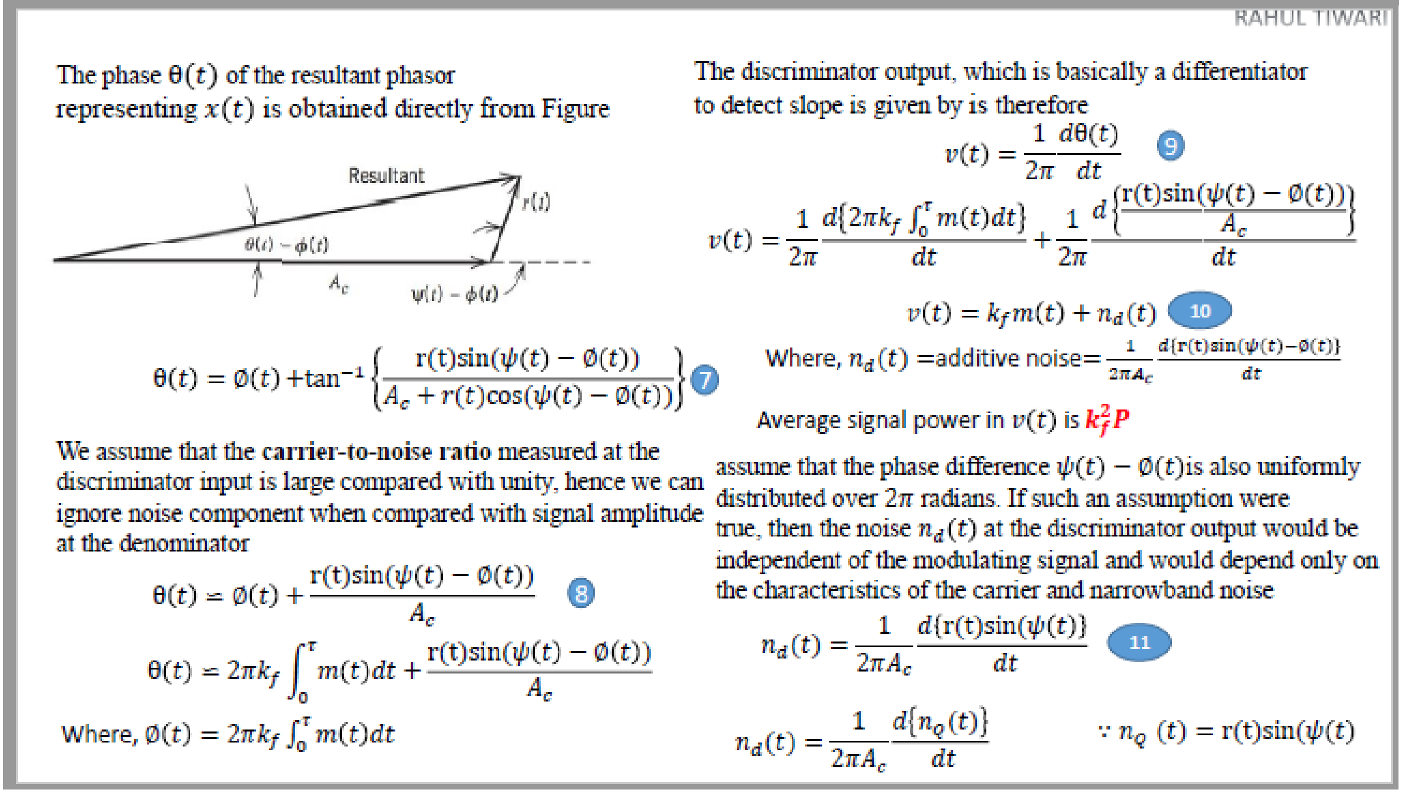
- The power of the noise $n(t)$ is given by N_0W,

where W is the bandwidth of message signal

- we define the *channel signal-to-noise ratio*,

$$(SNR)_C = \frac{average\ power\ of\ the\ modulated\ signal}{the\ average\ power\ of\ noise\ in\ the\ message\ bandwidth}$$

$$(SNR)_C = \frac{\frac{A_c^2}{2}}{N_0W} = \frac{A_c^2}{2N_0W}$$ ⑥



The phase θ(t) of the resultant phasor representing $x(t)$ is obtained directly from Figure

$$\theta(t) = \emptyset(t) + \tan^{-1}\left\{\frac{r(t)\sin(\psi(t) - \emptyset(t))}{A_c + r(t)\cos(\psi(t) - \emptyset(t))}\right\}$$ ⑦

We assume that the **carrier-to-noise ratio** measured at the discriminator input is large compared with unity, hence we can ignore noise component when compared with signal amplitude at the denominator

$$\theta(t) = \emptyset(t) + \frac{r(t)\sin(\psi(t) - \emptyset(t))}{A_c}$$ ⑧

$$\theta(t) = 2\pi k_f\int_0^{\tau} m(t)dt + \frac{r(t)\sin(\psi(t) - \emptyset(t))}{A_c}$$

Where, $\emptyset(t) = 2\pi k_f\int_0^{\tau} m(t)dt$

The discriminator output, which is basically a differentiator to detect slope is given by is therefore

$$v(t) = \frac{1}{2\pi}\frac{d\theta(t)}{dt}$$ ⑨

$$v(t) = \frac{1}{2\pi}\frac{d\{2\pi k_f\int_0^{\tau} m(t)dt\}}{dt} + \frac{1}{2\pi}\frac{d\left\{\frac{r(t)\sin(\psi(t) - \emptyset(t))}{A_c}\right\}}{dt}$$

$$v(t) = k_f m(t) + n_d(t)$$ ⑩

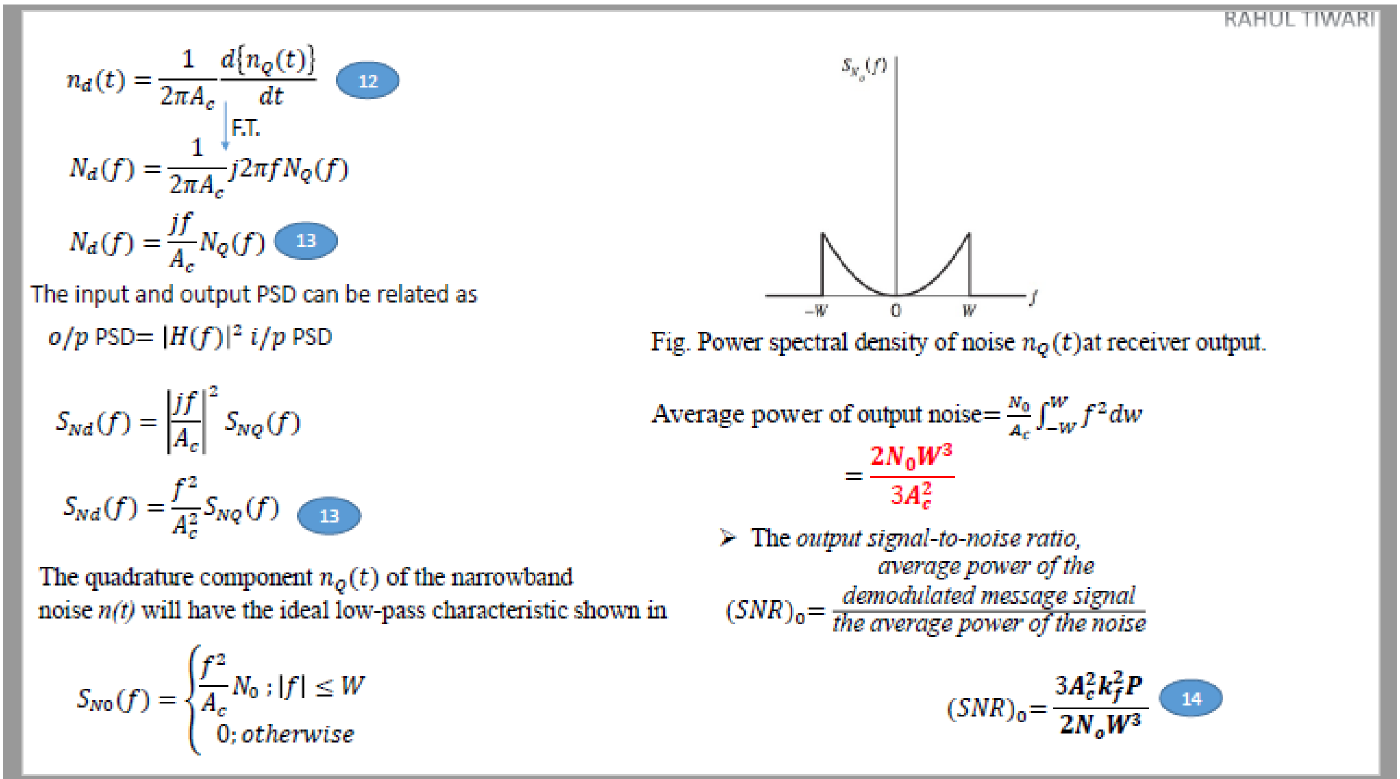
Where, $n_d(t)$ =additive noise= $\frac{1}{2\pi A_c}\frac{d\{r(t)\sin(\psi(t) - \emptyset(t)\}}{dt}$

Average signal power in $v(t)$ is k_f^2P

assume that the phase difference $\psi(t) - \emptyset(t)$is also uniformly distributed over 2π radians. If such an assumption were true, then the noise $n_d(t)$ at the discriminator output would be independent of the modulating signal and would depend only on the characteristics of the carrier and narrowband noise

$$n_d(t) = \frac{1}{2\pi A_c}\frac{d\{r(t)\sin(\psi(t)\}}{dt}$$ ⑪

$$n_d(t) = \frac{1}{2\pi A_c}\frac{d\{n_Q(t)\}}{dt} \qquad \because n_Q\ (t) = r(t)\sin(\psi(t)$$



$$n_d(t) = \frac{1}{2\pi A_c}\frac{d\{n_Q(t)\}}{dt}$$ ⑫

F.T.

$$N_d(f) = \frac{1}{2\pi A_c} j2\pi f N_Q(f)$$

$$N_d(f) = \frac{jf}{A_c} N_Q(f)$$ ⑬

The input and output PSD can be related as

o/p PSD= $|H(f)|^2$ i/p PSD

$$S_{Nd}(f) = \left|\frac{jf}{A_c}\right|^2 S_{NQ}(f)$$

$$S_{Nd}(f) = \frac{f^2}{A_c^2} S_{NQ}(f)$$ ⑬

The quadrature component $n_Q(t)$ of the narrowband noise *n(t)* will have the ideal low-pass characteristic shown in

$$S_{N0}(f) = \begin{cases} \frac{f^2}{A_c} N_0 \,;\, |f| \le W \\ 0; otherwise \end{cases}$$

Fig. Power spectral density of noise $n_Q(t)$at receiver output.

Average power of output noise= $\frac{N_0}{A_c}\int_{-W}^{W} f^2 dw$

$$= \frac{2N_0W^3}{3A_c^2}$$

- The *output signal-to-noise ratio,*

$$(SNR)_0 = \frac{\textit{average power of the demodulated message signal}}{\textit{the average power of the noise}}$$

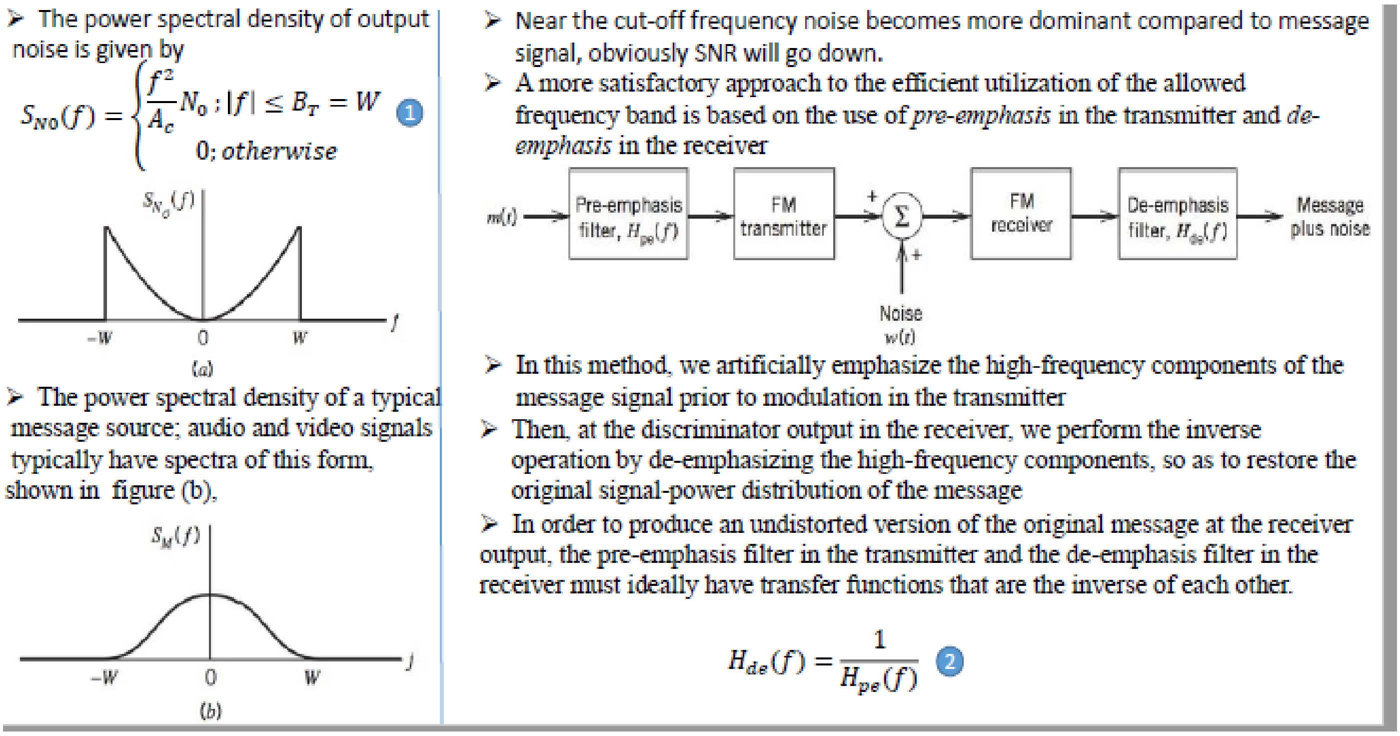
$$(SNR)_0 = \frac{3A_c^2k_f^2P}{2N_oW^3}$$ ⑭

FOM of FM receiver is given by

$$FOM = \frac{(SNR)_0}{(SNR)_C}$$

$$FOM = \frac{\frac{3A_c^2k_f^2P}{2N_oW^3}}{\frac{A_c^2}{2N_oW}} = \frac{3k_f^2P}{W^2}$$ ⑮

3) (iii)Pre emphasis and (iv) Deemphasis



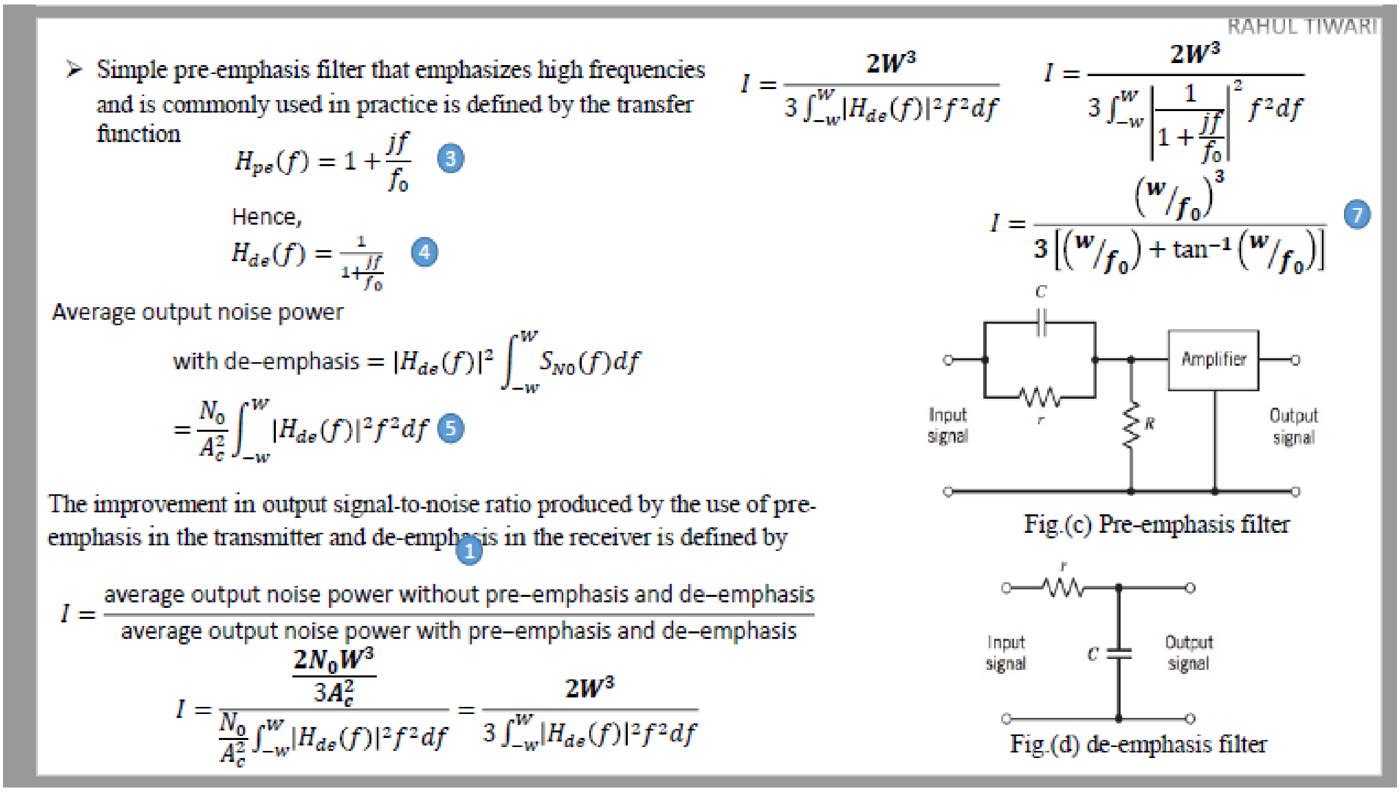
- The power spectral density of output noise is given by

$$S_{N0}(f) = \begin{cases} \frac{f^2}{A_c} N_0 \,;\, |f| \le B_T = W \\ 0; otherwise \end{cases}$$ ①

- The power spectral density of a typical message source; audio and video signals typically have spectra of this form, shown in figure (b),

- Near the cut-off frequency noise becomes more dominant compared to message signal, obviously SNR will go down.
- A more satisfactory approach to the efficient utilization of the allowed frequency band is based on the use of *pre-emphasis* in the transmitter and *de-emphasis* in the receiver
- In this method, we artificially emphasize the high-frequency components of the message signal prior to modulation in the transmitter
- Then, at the discriminator output in the receiver, we perform the inverse operation by de-emphasizing the high-frequency components, so as to restore the original signal-power distribution of the message
- In order to produce an undistorted version of the original message at the receiver output, the pre-emphasis filter in the transmitter and the de-emphasis filter in the receiver must ideally have transfer functions that are the inverse of each other.

$$H_{de}(f) = \frac{1}{H_{pe}(f)}$$ ②



- Simple pre-emphasis filter that emphasizes high frequencies and is commonly used in practice is defined by the transfer function

$$H_{pe}(f) = 1 + \frac{jf}{f_0} \quad ③$$

Hence,

$$H_{de}(f) = \frac{1}{1+\frac{jf}{f_0}} \quad ④$$

Average output noise power

$$\text{with de–emphasis} = |H_{de}(f)|^2 \int_{-W}^{W} S_{NO}(f)df$$

$$= \frac{N_0}{A_c^2} \int_{-W}^{W} |H_{de}(f)|^2 f^2 df \quad ⑤$$

The improvement in output signal-to-noise ratio produced by the use of pre-emphasis in the transmitter and de-emphasis in the receiver is defined by ①

$$I = \frac{\text{average output noise power without pre–emphasis and de–emphasis}}{\text{average output noise power with pre–emphasis and de–emphasis}}$$

$$I = \frac{\frac{2N_0W^3}{3A_c^2}}{\frac{N_0}{A_c^2}\int_{-W}^{W}|H_{de}(f)|^2 f^2 df} = \frac{2W^3}{3\int_{-W}^{W}|H_{de}(f)|^2 f^2 df}$$

$$I = \frac{2W^3}{3\int_{-W}^{W}|H_{de}(f)|^2 f^2 df}$$

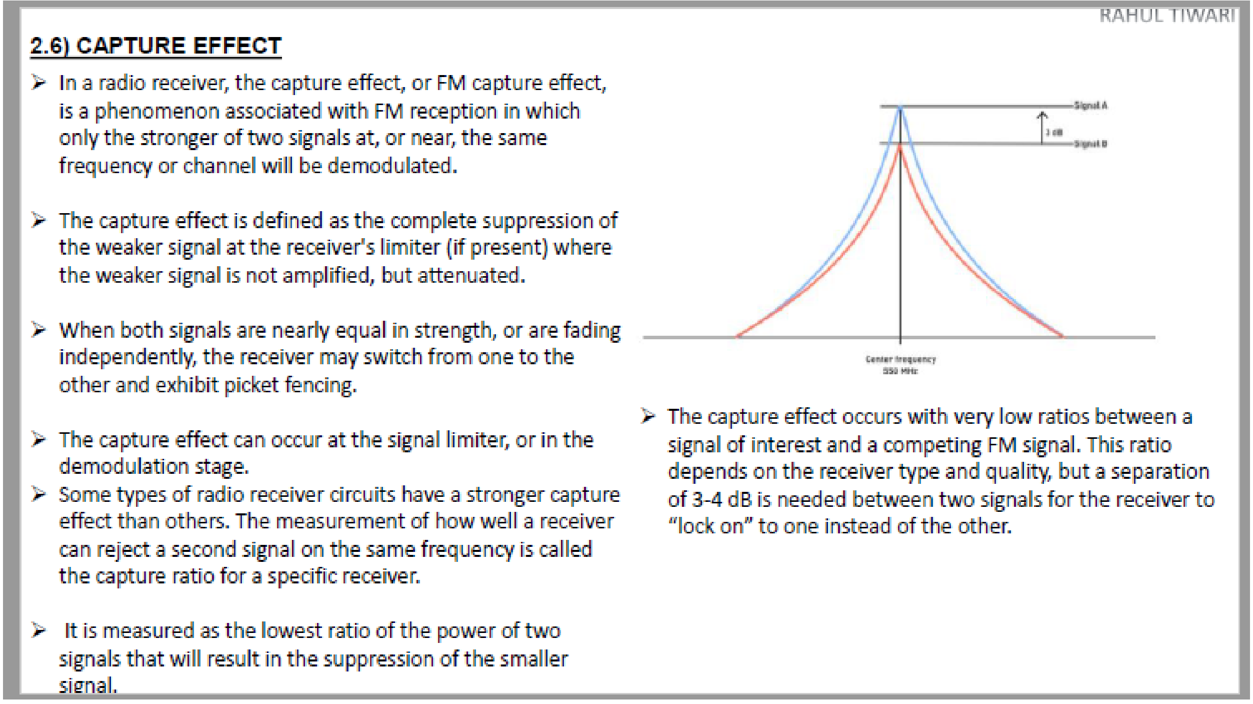
$$I = \frac{2W^3}{3\int_{-W}^{W}\left|\frac{1}{1+\frac{jf}{f_0}}\right|^2 f^2 df}$$

$$I = \frac{\left({}^{W}/_{f_0}\right)^3}{3\left[\left({}^{W}/_{f_0}\right) + \tan^{-1}\left({}^{W}/_{f_0}\right)\right]} \quad ⑦$$

Fig.(c) Pre-emphasis filter

Fig.(d) de-emphasis filter

i) Capture effect

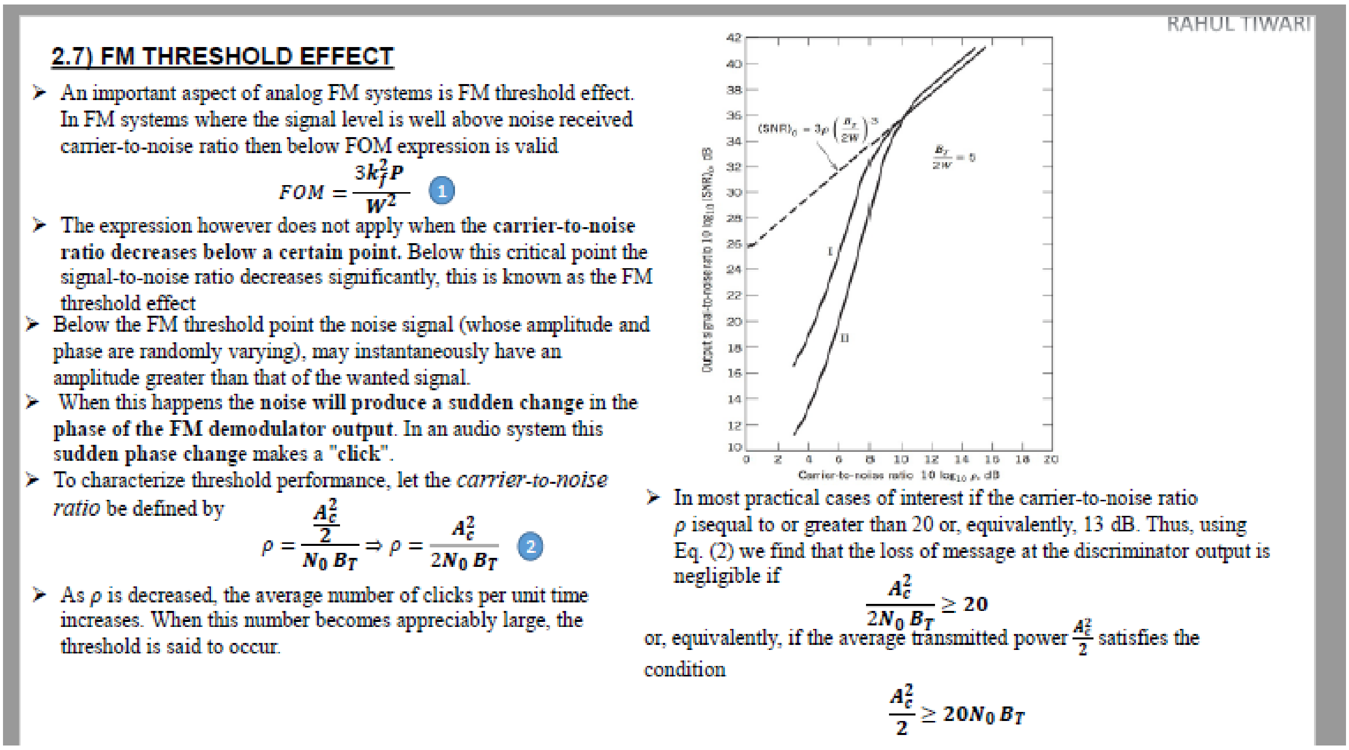


2.6) CAPTURE EFFECT

- In a radio receiver, the capture effect, or FM capture effect, is a phenomenon associated with FM reception in which only the stronger of two signals at, or near, the same frequency or channel will be demodulated.
- The capture effect is defined as the complete suppression of the weaker signal at the receiver's limiter (if present) where the weaker signal is not amplified, but attenuated.
- When both signals are nearly equal in strength, or are fading independently, the receiver may switch from one to the other and exhibit picket fencing.
- The capture effect can occur at the signal limiter, or in the demodulation stage.
- Some types of radio receiver circuits have a stronger capture effect than others. The measurement of how well a receiver can reject a second signal on the same frequency is called the capture ratio for a specific receiver.
- It is measured as the lowest ratio of the power of two signals that will result in the suppression of the smaller signal.

- The capture effect occurs with very low ratios between a signal of interest and a competing FM signal. This ratio depends on the receiver type and quality, but a separation of 3-4 dB is needed between two signals for the receiver to "lock on" to one instead of the other.

(ii) Threshold effect



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2.7) FM THRESHOLD EFFECT

- An important aspect of analog FM systems is FM threshold effect. In FM systems where the signal level is well above noise received carrier-to-noise ratio then below FOM expression is valid

$$FOM = \frac{3k_f^2 P}{W^2} \quad ①$$

- The expression however does not apply when the **carrier-to-noise ratio decreases below a certain point.** Below this critical point the signal-to-noise ratio decreases significantly, this is known as the FM threshold effect
- Below the FM threshold point the noise signal (whose amplitude and phase are randomly varying), may instantaneously have an amplitude greater than that of the wanted signal.
- When this happens the **noise will produce a sudden change** in the **phase of the FM demodulator output**. In an audio system this **sudden phase change** makes a **"click"**.
- To characterize threshold performance, let the *carrier-to-noise ratio* be defined by

$$\rho = \frac{\frac{A_c^2}{2}}{N_0\, B_T} \Rightarrow \rho = \frac{A_c^2}{2N_0\, B_T} \quad ②$$

- As ρ is decreased, the average number of clicks per unit time increases. When this number becomes appreciably large, the threshold is said to occur.

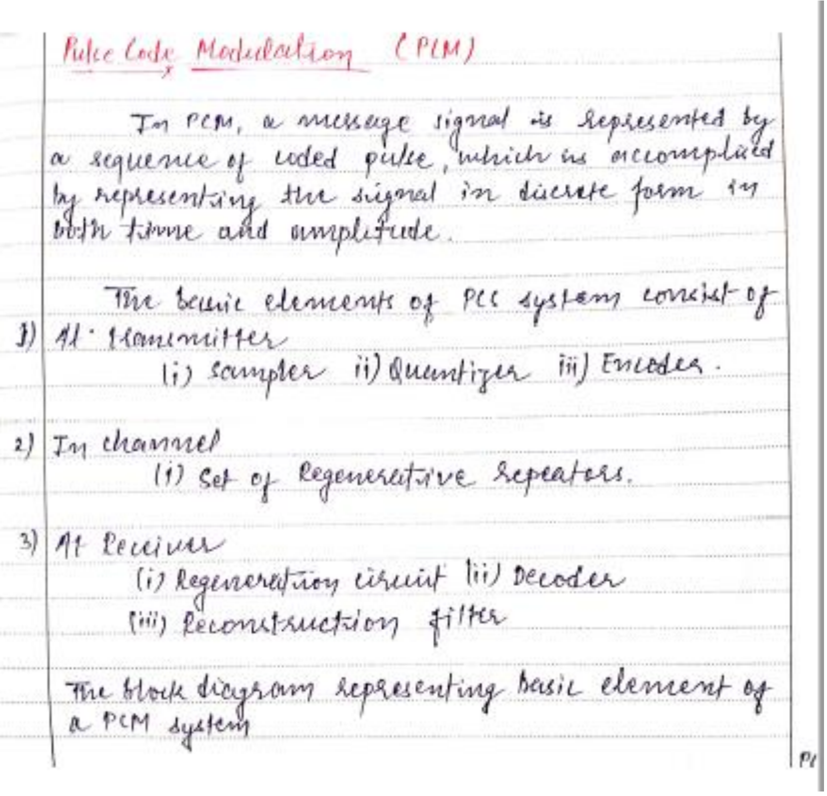
- In most practical cases of interest if the carrier-to-noise ratio ρ isequal to or greater than 20 or, equivalently, 13 dB. Thus, using Eq. (2) we find that the loss of message at the discriminator output is negligible if

$$\frac{A_c^2}{2N_0\, B_T} \geq 20$$

or, equivalently, if the average transmitted power $\frac{A_c^2}{2}$ satisfies the condition

$$\frac{A_c^2}{2} \geq 20 N_0\, B_T$$

4)



Pulse Code Modulation (PCM)

In PCM, a message signal is represented by a sequence of coded pulse, which is accomplied by representing the signal in discrete form in both time and amplitude.

The basic elements of PCC system consist of

1) At transmitter
    (i) sampler ii) Quantizer iii) Encoder.

2) In channel
    (1) Set of Regenerative repeaters.

3) At Receiver
    (i) Regeneration circuit (ii) Decoder
    (iii) Reconstruction filter

The block diagram representing basic element of a PCM system

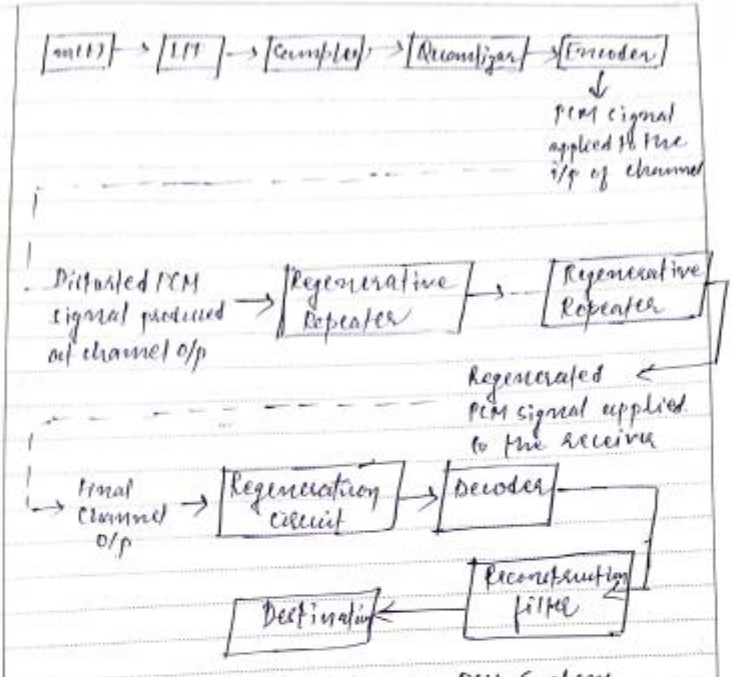
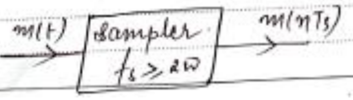
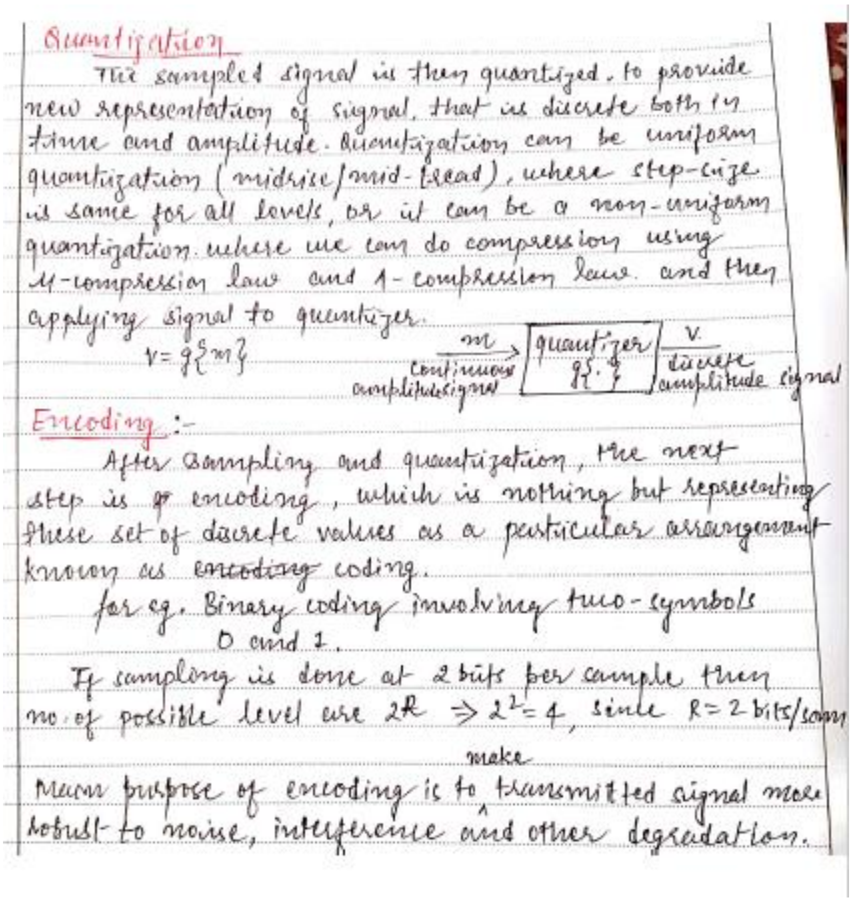


fig. Basic Elements of PCM System.

Sampling: The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate instantaneous sampling. in order for proper reconstruction of signal at the Rx the sampling rate (f_s) must be chosen greater than 2W

$\boxed{f_s \geq 2(\text{Msg. freq}).}$





Quantization

The sampled signal is then quantized. to provide new representation of signal. that is discrete both in time and amplitude. Quantization can be uniform quantization (midrise/mid-tread), where step-size is same for all levels, or it can be a non-uniform quantization. where we can do compression using μ-compression law and A-compression law. and then applying signal to quantizer.

$v = g\{m\}$

Encoding :-

After sampling and quantization, the next step is encoding, which is nothing but representing these set of discrete values as a particular arrangement known as ~~encoding~~ coding.

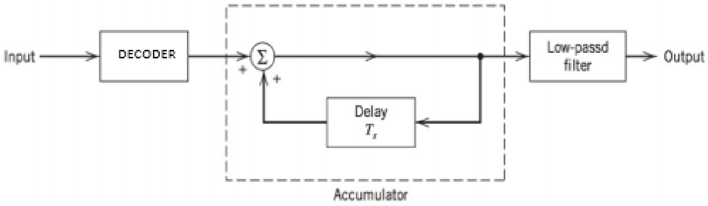
for eg. Binary coding involving two-symbols 0 and 1.

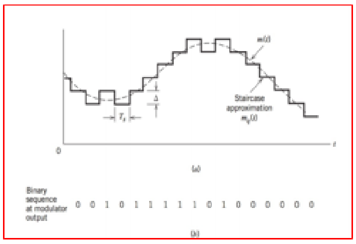
If sampling is done at 2 bits per sample then no. of possible level are $2^R \Rightarrow 2^2 = 4$, since R = 2 bits/sam

Main purpose of encoding is to make transmitted signal more robust to noise, interference and other degradation.

5)

DM(D/A) decoder

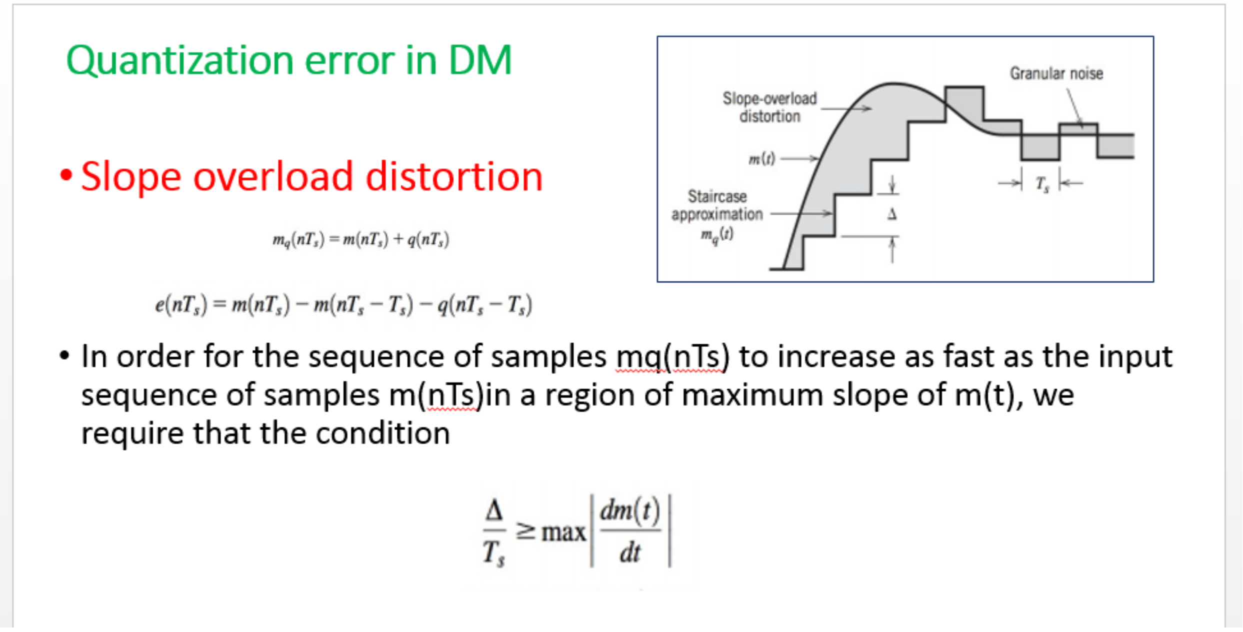




$$m_q(nT_s) = m_q(nT_s - T_s) + e_q(nT_s)$$

$$m_q(nT_s) = \Delta \sum_{i=1}^{n} \operatorname{sgn}[e(iT_s)] = \sum_{i=1}^{n} e_q(iT_s)$$

- Staircase approximation mq(t) is reconstructed by passing the sequence of positive and negative pulses, produced at the decoder output, through an accumulator in a manner similar to that used in the transmitter.
- The out-of-band quantization noise in the high-frequency staircase waveform mq(t) is rejected by passing it through a low-pass filter.



Quantization error in DM

- Slope overload distortion

$$m_q(nT_s) = m(nT_s) + q(nT_s)$$

$$e(nT_s) = m(nT_s) - m(nT_s - T_s) - q(nT_s - T_s)$$

- In order for the sequence of samples mq(nTs) to increase as fast as the input sequence of samples m(nTs)in a region of maximum slope of m(t), we require that the condition

$$\frac{\Delta}{T_s} \geq \max\left|\frac{dm(t)}{dt}\right|$$

Note

- Slope overload:
  - If the step-size Δ is too small for the staircase approximation mq(t) to follow a steep segment of the input waveform m(t), with the result that mq(t) falls behind m(t), this condition is called slope overload.
  - The resulting quantization error is called slope-overload distortion
- Granular noise
  - the step-size Δ is too large relative to the local slope characteristics of the input waveform m(t)
  - the staircase approximation mq(t) hunts around a relatively flat segment of the input waveform.
- There is a need to have a large step-size to accommodate a wide dynamic range, whereas a small step-size is required for the accurate representation of relatively low-level signals.
- To improve performance, we need to make the delta modulator adaptive
  - the step-size is made to vary in accordance with the input signal.