PAPER SOLUTION

Class: S.E. EXTC SEM. IV (Rev.) Winter 2010

Subject: Principles of Communication Engineering

PAPER CODE: GT-6519

Q.1] a)Compare between noise & interference. Explain different types of noise.

Ans- Comparison:-

1) Noise is electrical disturbance in communication ckt. Where interference is disturbance created among two or more signals.
2) Eg. Of noise- Noise created in ckt. Due to temperature variation.
   Eg. Of interference- interference created due to multipath propagation.

Different types of noise:-

1) Thermal noise:-

   Generated due to variation in random motion of electrons because of temperature changes. The avg voltage resulting from this nonuniform distribution is zero but the avg power is not zero.As this power result from thermal energy it is called thermal noise power which is given by

   Pn= KTB watts.

   Where k=Bolzmann’s constant= $1.38*10^{-23}$ J/K , B= B/W of noise spectrum

   T= temperature of conductor in ^{0}K

2) Shot noise:-

   Produced due to the random variation in arrival of electrons or holes at o/p electrode of any amplifying device.It sounds like shower of lead shots falling on metal sheet.The mean square shot noise current is given as,

   $\overline{I_n^2} = 2(I+I_0)qB$ amp^2

   Where

   I= direct current across the junction (n amp)

   I_0=reverse saturation current (in amp)

   q=electron charge=$1.6*10^{-19}$ C

   B=effective noise bandwidth.

3) Partition noise:-

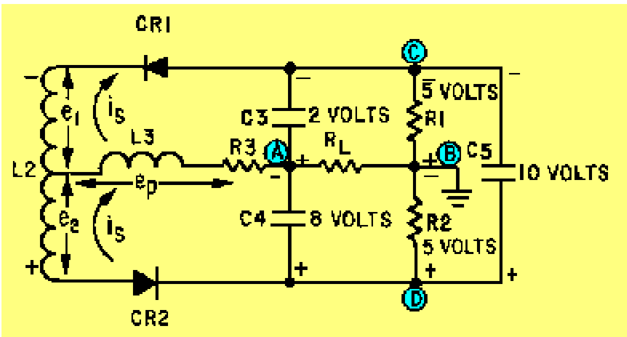
Generated when current gets divided into two or more paths. It is generated due to random fluctuations in division.

4) Low frequency flicker noise.

5) High frequency transit time noise

B] How ratio detector provides amplitude limiting?

Ans-



The ratio detector is not affected by amplitude variations on the fm wave. The output of the detector adjusts itself automatically to the average amplitude of the input signal. C5 charges to the sum of the voltages across R1 and R2 and, because of its time constant, tends to filter out any noise impulses. Before C5 can charge or discharge to the higher or lower potential, the noise disappears. The difference in charge across C5 is so slight that it is not discernible in the output. Ratio detectors can operate with as little as 100 millivolts of input. This is much lower than that required for limiter saturation and less gain is required from preceding stages.

C] What is quantization? Explain types of quantization.

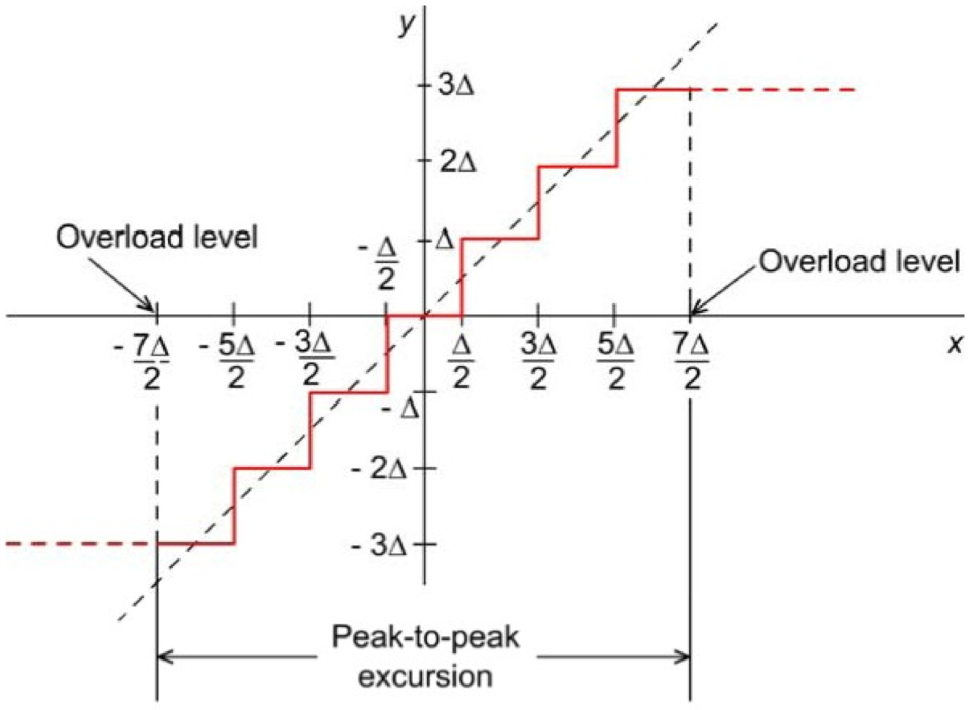
Ans-

Quantization-

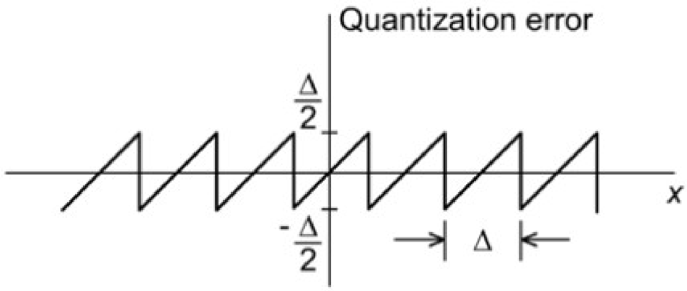
The process of conversion of analog samples of a signal into a set of discrete (digital) values is called quantization. Note however, that quantization is inherently information lossy. For a given peak-to-peak range of the analog signal, smaller the value of Δ , larger is the number of discrete amplitudes and hence, finer is the quantization. Sometimes one may resort to somewhat coarse quantization, which can result in some noticeable distortion; this may, however be acceptable to the end receiver.

The quantization process can be illustrated graphically. This is shown inFig. a. The variable x denotes the input of the quantizer, and the variable y represents the output. As can be seen from the figure, the quantization process implies that a straight line relation between the input and output (broken line through the origin) of a linear continuous system is replaced by a staircase characteristic.

The difference between two adjacent discrete values, Δ , is called the step size of the quantizer. The error signal, that is, difference between the input and the quantizer output has been shown in Fig. 6.20(b). We see from the figure that the magnitude of the error is always less than or equal to Δ/2



(a)



(b)

Types of Quatnization:-

a)Uniform quatization:-
An analog signal, even it is limited in its peak-to-peak excursion, can in general assume any value within this permitted range. If such a signal is sampled, say at uniform time intervals, the number of different values the samples can assume is unlimited. Any human sensor (such as ear or eye) as the
ultimate receiver can only detect finite intensity differences. If the receiver is not
able to distinguish between two sample amplitudes, say v1 and v2 such that
$|v1 - v2| < \Delta/2$
then we can have a set of discrete amplitude levels separated by

Δ and the original signal with continuous amplitudes, may be approximated by a
signal constructed of discrete amplitudes selected on a minimum error basis from
an available set. This ensures that the magnitude of the error between the actual
sample and its approximation is within Δ/2

and this difference is irrelevant from the
receiver point of view. The realistic assumption that a signal m(t) is (essentially)
limited in its peak-to-peak variations and any two adjacent (discrete) amplitude
levels are separated by Δ will result in a finite number (say L) of discrete
amplitudes for the signal.

b) Non Uniform quantization :-
A non-uniform quantizer is characterized by a variable
step size.

D] Explain the need of modulation.
Ans-

1) To reduce Height of antenna-

For efficient radiation, the size of the antenna should be λ /10 or more
(preferably around λ/ 4), where λ is the wavelength of the signal to be radiated.
Take the case of audio, which has spectral components almost from DC upto 20
kHz. Assume that we are designing the antenna for the mid frequency; that is,10
kHz. Then the length of the antenna that is required, even for the λ /10 situation
Even an antenna of the size of 3 km, will not be able to take care of the entire
spectrum of the signal because for the frequency components around 1 kHz, the
length of the antenna would be λ/ 100. Hence, what is required from the point of
view of efficient radiation is the conversion of the baseband signal into a
narrowband, bandpass signal. Modulation process helps us to accomplish this;
besides, modulation gives rise to some other features which can be exploited for
the purpose of efficient communication.

2) Modulation for efficient transmission

Quite a few wireless channels have their own appropriate passbands. For efficient transmission, it would be necessary to shift the message spectrum into the passband of the channel intended. Ground wave propagation (from the lower atmosphere) is possible only up to about 2 MHz. Long distance ionospheric propagation is possible for frequencies in the range 2 to 30 MHz. Beyond 30 MHz, the propagation is line of sight. Preferred frequencies for satellite communication are around 3 to 6 GHz. By choosing an appropriate carrier frequency and modulation technique, it is possible for us to translate the baseband message spectrum into a suitable slot in the passband of the channel intended. That is, modulation results in frequency translation.

3)Modulation for multiplexing

Several message signals can be transmitted on a given channel, by assigning to each message signal an appropriate slot in the passband of the channel. Take the example of AM broadcast, used for voice and medium quality music broadcast. The passband of the channel used to 550 kHz to 1650 kHz. That is, the width of the passband of the channel that is being used is 1100 kHz. If the required transmission bandwidth is taken as 10 kHz, then it is possible for us to multiplex, atleast theoretically, 110 distinct message signals on the channel and still be able to separate them individually as and when we desire because the identity of each message is preserved in the frequency domain.

4)Modulation for frequency assignment

Continuing on the broadcast situation, let us assume that each one of the message signals is being broadcast by a different station. Each station can be assigned a suitable carrier so that the corresponding program material can be received by tuning to the station desired.

5)Modulation to improve the signal-to-noise ratio

Certain modulation schemes (notably frequency modulation and phase modulation) have the feature that they will permit improved signal-to-noise ratio at the receiver output, provided we are willing to pay the price in terms of increased transmission bandwidth (Note that the transmitted power need not be increased). This feature can be taken advantage of when the quality of the receiver output is very important.

Q.2] a] Derive equation for total transmitted power, total side band power & single sideband power for AM wave & draw frequency spectrum for DSBFC.

Ans-

Total Power-

We know that AM wave has three components 1.Unmodulated Carrier,2.Lower Side band ,3.Upper sideband.

Therefor,

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

$$= Ecarr^2/R + E_{USB}^2/R + E_{LSB}^2/R$$

Where all three represent R.M.S. values, & Resistance R is a characteristic impedance of antenna in which power is dissipated.

CARRIER POWER-

$$P_c = E_{carr}^2/R$$

$$= (E_c/\sqrt{2})^2/R$$

$$\underline{P_c = E_c^2/2R}$$

Power in sideband-

$$P_{lsb} = P_{usb} = E_{sb}^2/R$$

$$= ((mE_c/2)/\sqrt{2})^2 * 1/R \qquad \text{since } E_{sb} = mE_c/2$$

$$\underline{P_{lsb} = P_{usb} = m^2E_c^2/8R}$$

We know that $E_c^2/2R = P_c$

$$\underline{P_{lsb} = P_{usb} = P_c(1+m^2/2)}$$

Total Power-

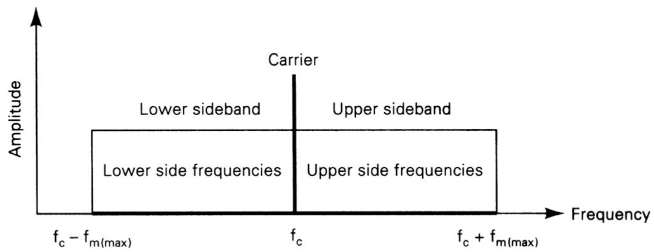
$$P_{Total} = E_c^2/2R + m^2E_c^2/8R + m^2E_c^2/8R$$

$$= E_c^2/2R(1 + m^2/4 + m^2/4)$$

$$= E_c^2/2R(1 + m^2/2)$$

$$\underline{P_{Total} = P_c(1 + m^2/2)}$$

Frequency Spectrum Of DSBFC:-

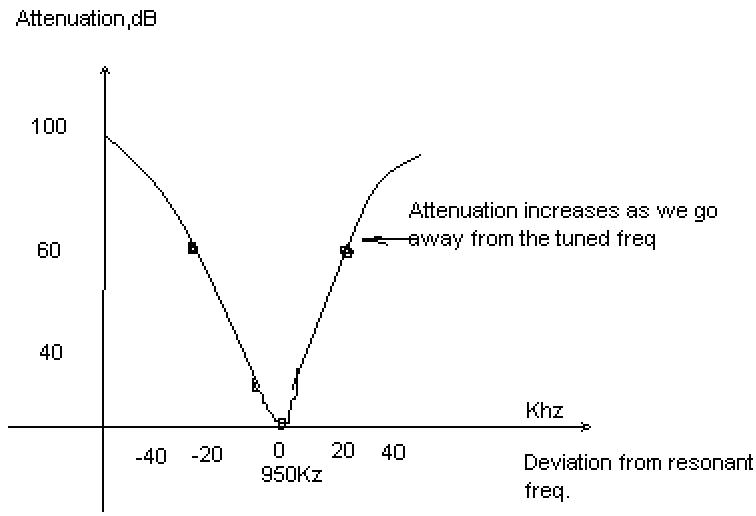


Q.2] b] Explain the following with respect to radio receivers:-
1.sensitivity 2.Fidelity 3.Selectivity 4.Dynamic Range.

Ans-
Sensitivity: minimum RF signal level that the receiver can detect at the RF input.
AM broadcast receivers
10 dB signal to noise ratio
½ watt (27 dBm) of power at the audio output
50 uV Sensitivity
Microwave receivers
40 dB signal to noise ratio
5 mw (7 dBm) of power at the output

Fidelity-
Ability to produce an exact replica of the original signal.
Forms of distortion
Amplitude
Results from non-uniform gain in amplifiers and filters.
Output signal differs from the original signal
Frequency: frequencies are in the output that were not in the orginal signal
Phase
Not important for voice transmission
Devastating for digital transmission

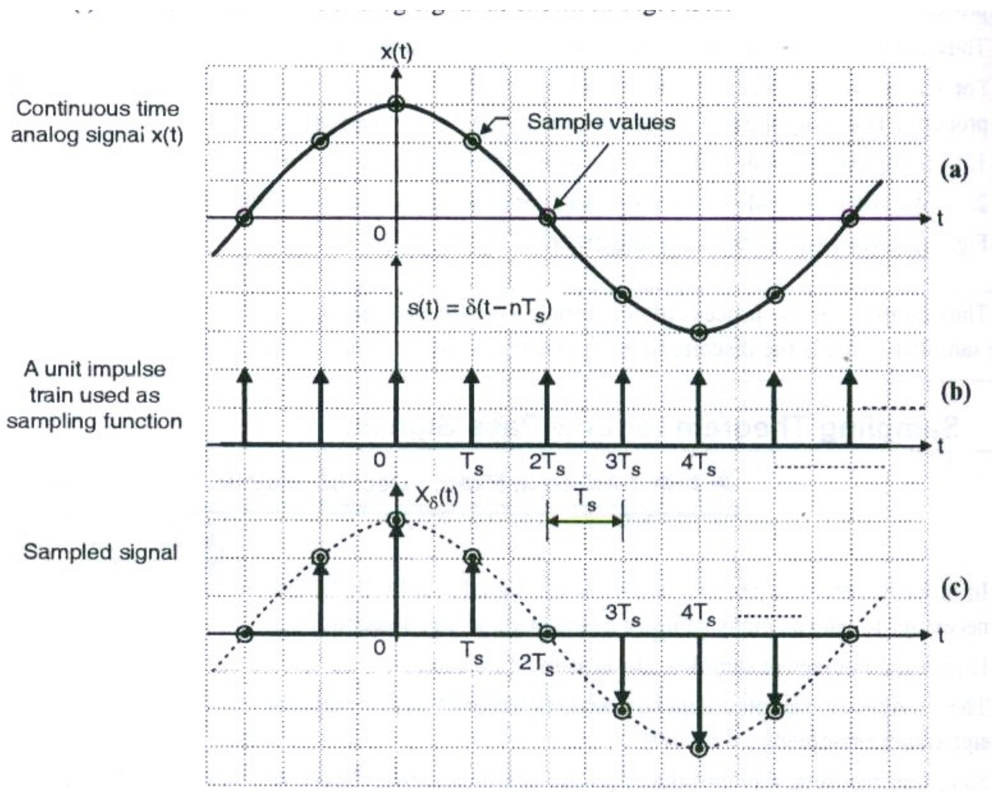
Selectivity:-
The selectivity of the receiver is it’s ability to reject unwanted signals. The selectivity is expressed as curve as shown in following fig.



It shows that receiver offers minimum ewjection at 950kHz. i.e. at tuned freq.
But the rejection increases as the received signal frequency deviates on both the sides of 950KHz. Selectivity is determined by freq. response characteristics of IF amplifier.

Dynamic Range:-
Difference in dB between the minimum input level and the level that will over drive the receiver (produce distortion).
Input power range that the receiver is useful.
100 dB is about the highest posible.
Low Dynamic Range
Causes desensitizing of the RF amplifiers
Results in sever inter-modulation distortion of weaker signals.

Q.3] a] State & prove sampling theorem for low pass signal.
Ans-
Statement:
1. If a finite energy signal x (t) contains no frequencies higher than "W" Hz (i.e. it is a band limited signal) then it is completely determined by specifying its values at the instants of time which are spaced (1/2W) seconds apart.
2. If a finite energy signal x (t) contains no frequency components higher than "W" Hz then it may be completely recovered from its samples which are spaced (I /2W) seconds apart



$$S(t) = \partial(t+2Ts)+ \partial(t+Ts)+ \partial(t)+ \partial(t-Ts)+ \partial(t-Ts)+\ldots$$

$$S(t)= \sum_{n=-\infty}^{\infty} \partial(t-nTs)$$

$$x\partial(t)= \sum_{n=-\infty}^{\infty} x(nTs) . \partial(t-nTs)$$

Fourier Transform:

$$X(f) =fo \sum_{n=-\infty}^{\infty} \partial(f-nf0)$$

$$S(f) =fs \sum_{n=-\infty}^{\infty} \partial(f-nfs)$$

$$X\partial(t)=x(t)S(t)$$

$$X\partial(f)=x(f)s(f)$$

$$S(f) = x(f) \left[fs \sum_{n=-\infty}^{\infty} \partial(f-nfs) \right]$$

$$X\partial(f) = fs \sum_{n=-\infty}^{\infty} \partial(f-nfs)$$

Q.3] b] An electronic device operating at temperature of 17^0c with a bandwidth of 10KHz. Calculate-

1. Thermal noise power. 2. RMS noise voltage for 100Ω internal & 100Ω load resistance.

Ans-

Thermal noise power'

P_n=KTB

=

=

Q.4] a] Explain indirect method of FM generation.

Ans-

Armstrong Method of FM generation:-

In this method FM is obtained through phase modulation. A crystal oscillator can be used hence freq stability is very high.

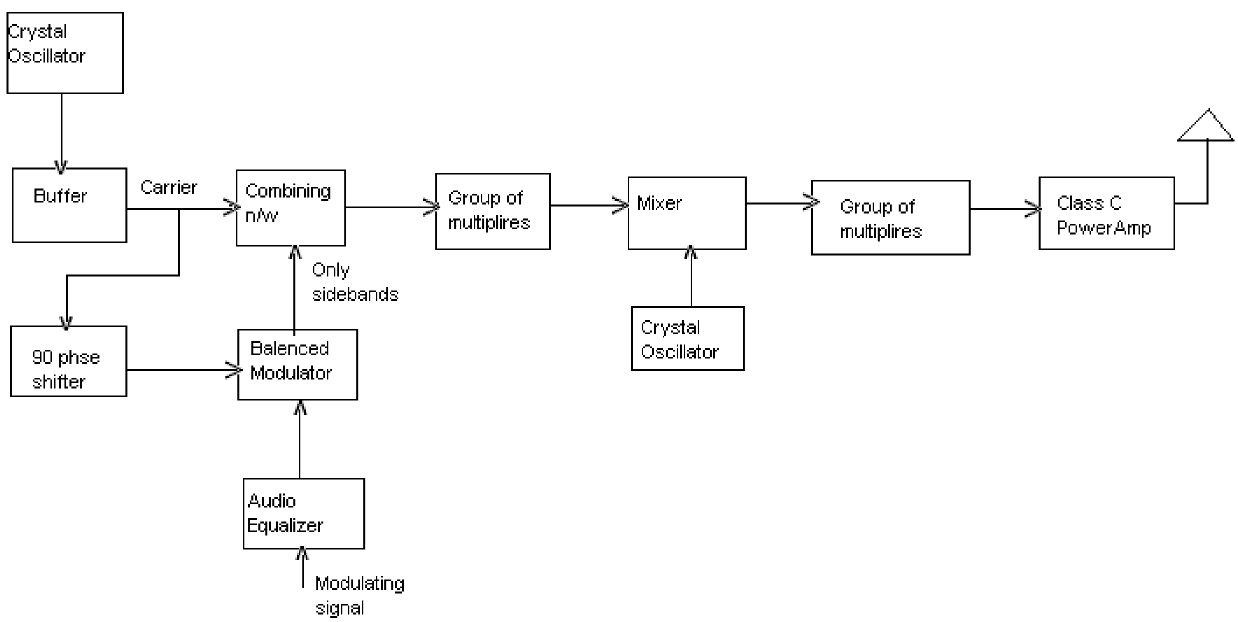
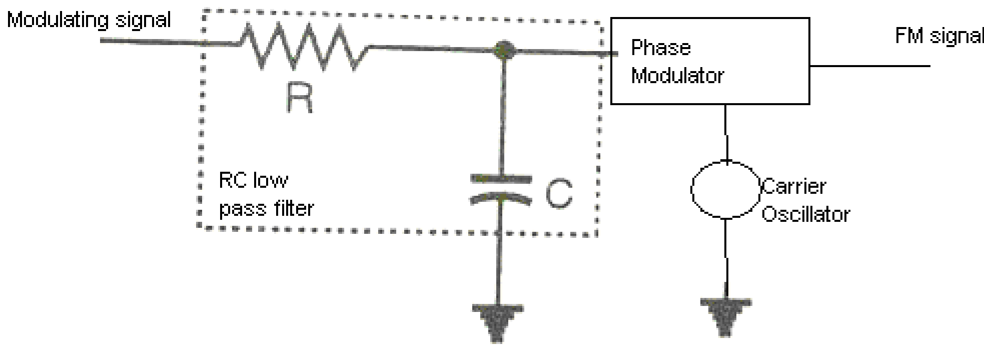


Fig- Block diagram of Armstrong method.

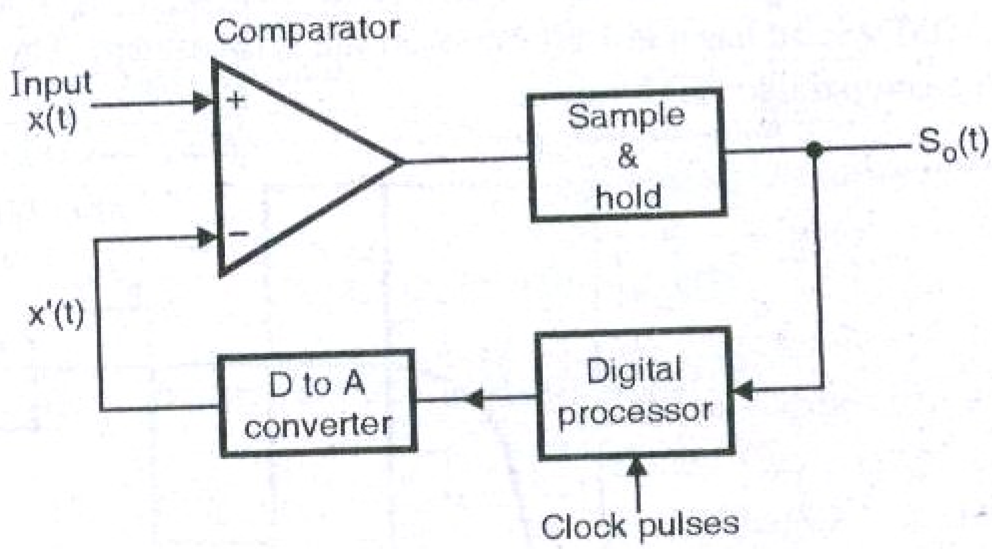
1. In PM along with phase variation some freq variation also take place.Higher modulating voltage produces greater phase shift which in turn produces large freq deviation.

2. And higher modulating frequencies produce a faster rate of change of modulating voltage hence they also produce greater freq deviation.
3. Thus in PM the carrier freq deviation is proportional to modulating voltage as well as modulating frequency.
4. But in FM frequency deviation is only proportional to modulating voltage.
5. To correct this problem modulating signal is passed through a low pass filter.
6. Filter o/p is then applied to phase modulator alongwith carrier



Q.4]b] Explain the block diagram of adaptive delta modulation with waveforms. How does it reduce slope overload error.

Ans-



S0(t)= + 1 If x(t)>x’(t) just before kth clock edge.
= -1 If x(t)<x’(t) just before kth clock edge.

• In response to the kth clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous i.e. (k- 1))" clock edge.

• If the direction of both the steps is same, then the processor will increase the magnitude of the present step by "8". If the directions are opposite then the processor will decrease the magnitude of the present step by "8",

• So(t)in the Fig. 10.11.1, i.e. the output of the ADM system is given as,

So (t) = + 1 if x (t) > x' (t) just before kth clock edge, and

So (t) = - 1 if x (t) < x' (t) just before the kth clock edge.

• Then the step size at the sampling instant k is given by,

δ (k) =[δ (k-1)] So(k)+ δ So(k-1)

let us take an example,

refer fig. below. Assume k=6

k-1=5,

δ(k-1) = δ(5)= δ

So(k)=So(6)=+1

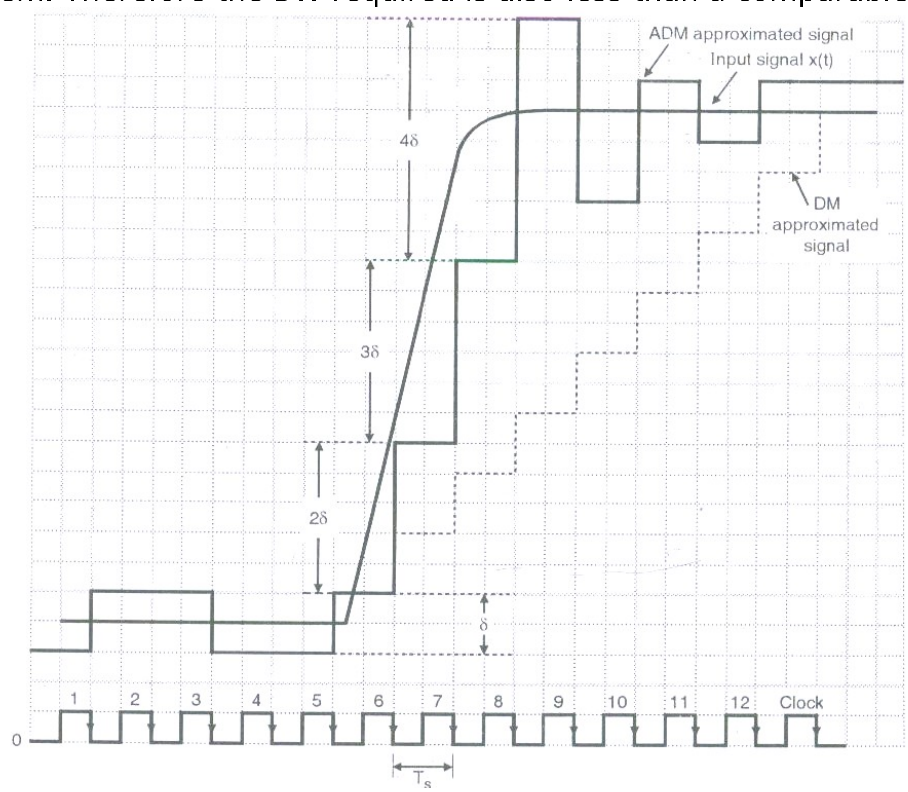
So(k-1)=So(5)=+1

"Substitute in Equation

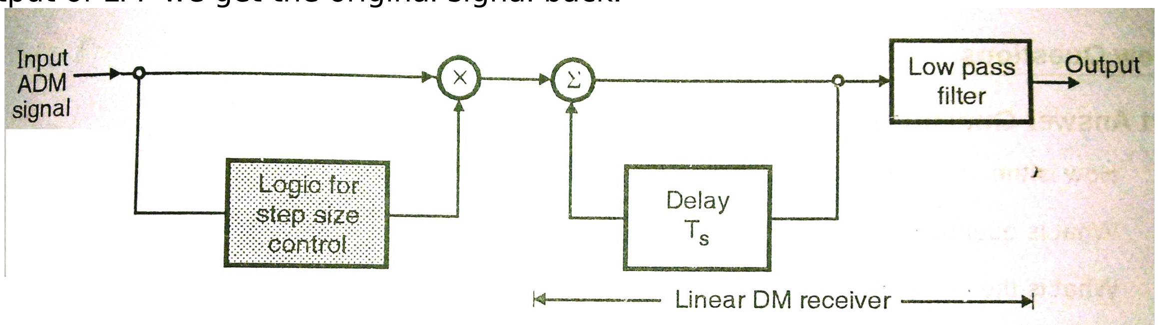
to get, δ(6)= δ+ δ=2 δ

Look at the Fig. , the step size at the 6th clock edge is 2 δ.

As shown in Fig. 10.11.2, due to variable step size, the slope overload error is reduced. But quantization error is increased. Due to the adjustable step size, the slope overload problem is solved. Hence ADM system has a low bit rale than the PCM system. Therefore the BW required is also less than a comparable PCM system.

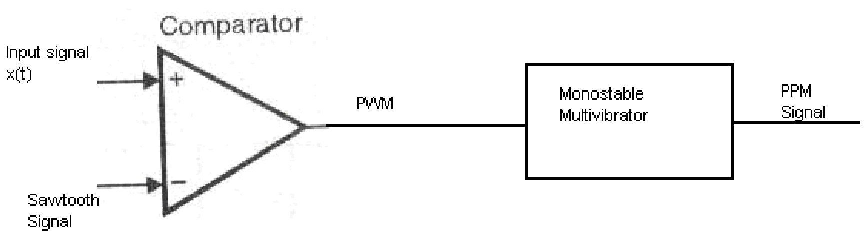


ADM signal is first converted into a D.M. signal with the help of a step size control logic and then applied to D.M. receiver.
At the output of LPF we get the original signal back.



Q.5]a] Explain how PPM is generated from PWM.

Ans-
The PPM signal can be generated from PWM signal as shown in following fig.

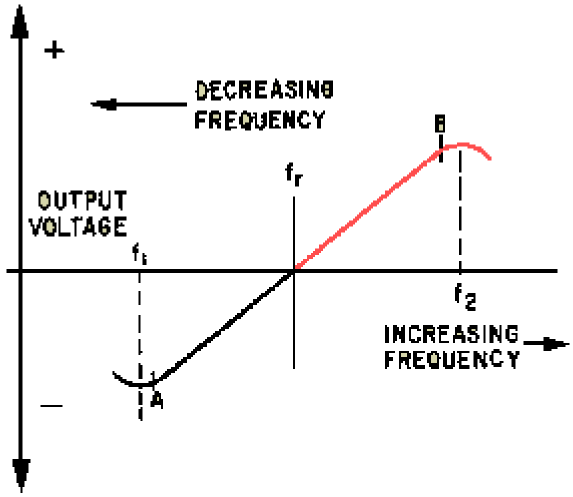


1. The PWM pulses obtained at the comparator o/p are applied to monostable multivibrator. The monostable multivibrator is –ve edge triggered.
2. Hence corresponding to each trailing edge of PWM s/g the monostable o/p goes high. It remains high for a fixed time decided by it's own RC components.
3. Thus as the trailing edge of the PWM signsl keep shifting in proportion with the modulating signal x(t).

Q.5] b] Explain the working of Foster-seeley discriminator with neat ckt. Diagram & phasor diagram.

Ans-
FOSTER-SEELEY DISCRIMINATOR
The FOSTER-SEELEY DISCRIMINATOR is also known as the PHASE-SHIFT DISCRIMINATOR. It uses a double-tuned rf transformer to convert frequency variations in the received fm signal to amplitude variations. These amplitude variations are then rectified and filtered to provide a dc output voltage. This voltage varies in both amplitude and polarity as the input signal varies in frequency. A typical discriminator response curve is shown in figure 3-10. The output voltage is 0 when the input frequency is equal to the carrier frequency (fr). When the input frequency rises above the center frequency, the output increases in the positive direction. When the input frequency drops below the center frequency, the output increases in the negative direction.

Figure 3-10. - Discriminator response curve.

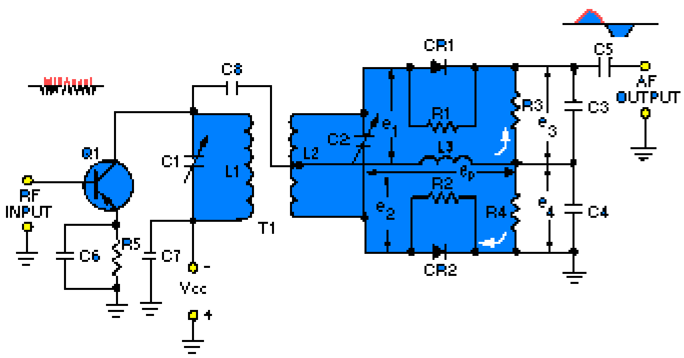


The output of the Foster-Seeley discriminator is affected not only by the input frequency, but also to a certain extent by the input amplitude. Therefore, using limiter stages before the detector is necessary.

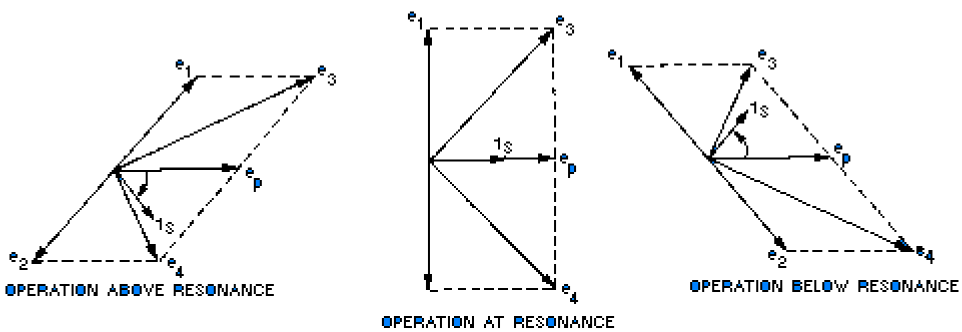
Circuit Operation of a Foster-Seeley Discriminator

View (A) of following figure shows a typical Foster-Seeley discriminator. The collector circuit of the preceding limiter/amplifier circuit (Q1) is shown. The limiter/amplifier circuit is a special amplifier circuit which limits the amplitude of the signal. This limiting keeps interfering noise low by removing excessive amplitude variations from signals. The collector circuit tank consists of C1 and L1. C2 and L2 form the secondary tank circuit. Both tank circuits are tuned to the center frequency of the incoming fm signal. Choke L3 is the dc return path for diode rectifiers CR1 and CR2. R1 and R2 are not always necessary but are usually used when the back (reverse bias) resistance of the two diodes is different. Resistors R3 and R4 are the load resistors and are bypassed by C3 and C4 to remove rf. C5 is the output coupling capacitor.

Figure - Foster-Seeley discriminator. FOSTER-SEELEY DISCRIMINATOR



(A) FOSTER - SEELEY DISCRIMINATOR



(B) VECTORS

CIRCUIT OPERATION AT RESONANCE. - The operation of the Foster-Seeley discriminator can best be explained using vector diagrams [figure 3-11, view (B)] that show phase relationships between the voltages and currents in the circuit. Let's look at the phase relationships when the input frequency is equal to the center frequency of the resonant tank circuit.

The input signal applied to the primary tank circuit is shown as vector ep. Since coupling capacitor C8 has negligible reactance at the input frequency, rf choke L3 is effectively in parallel with the primary tank circuit. Also, because L3 is effectively in parallel with the primary tank circuit, input voltage ep also appears across L3. With voltage ep applied to the primary of T1, a voltage is induced in the secondary which causes current to flow in the secondary tank circuit. When the input frequency is equal to the center frequency, the tank is at resonance and acts resistive. Current and voltage are in phase in a resistance circuit, as shown by is and ep. The current flowing in the tank causes voltage drops across each half of the balanced secondary winding of transformer T1. These voltage drops are of equal amplitude and opposite polarity with respect to the center tap of the winding. Because the winding is inductive, the voltage across it is 90 degrees out of phase with the current through it. Because of the center-tap arrangement, the voltages at each end of the secondary winding of T1 are 180 degrees out of phase and are shown as e1 and e2 on the vector diagram.

The voltage applied to the anode of CR1 is the vector sum of voltages ep and e1, shown as e3 on the diagram. Likewise, the voltage applied to the anode of CR2 is the vector sum of voltages ep and e2, shown as e4 on the diagram. At resonance e3 and e4 are equal, as shown by vectors of the same length. Equal anode voltages on diodes CR1 and CR2 produce equal currents and, with equal load resistors, equal and opposite voltages will be developed across R3 and R4. The output is taken

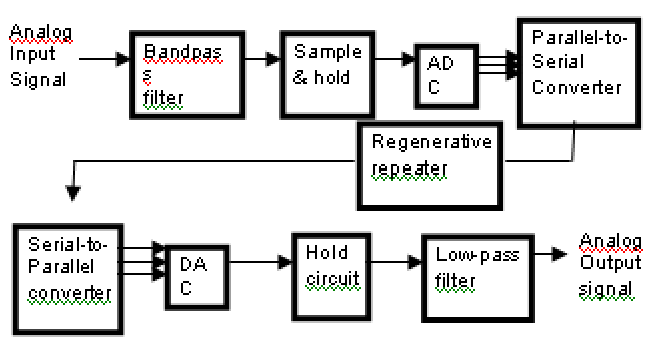
across R3 and R4 and will be 0 at resonance since these voltages are equal and of appositive polarity.
The diodes conduct on opposite half cycles of the input waveform and produce a series of dc pulses at the rf rate. This rf ripple is filtered out by capacitors C3 and C4.

OPERATION ABOVE RESONANCE. - A phase shift occurs when an input frequency higher than the center frequency is applied to the discriminator circuit and the current and voltage phase relationships change. When a series-tuned circuit operates at a frequency above resonance, the inductive reactance of the coil increases and the capacitive reactance of the capacitor decreases. Above resonance the tank circuit acts like an inductor. Secondary current lags the primary tank voltage, ep. Notice that secondary voltages e1 and e2 are still 180 degrees out of phase with the current (iS) that produces them. The change to a lagging secondary current rotates the vectors in a clockwise direction. This causes el to become more in phase with ep while e2 is shifted further out of phase with ep. The vector sum of ep and e2 is less than that of ep and e1. Above the center frequency, diode CR1 conducts more than diode CR2. Because of this heavier conduction, the voltage developed across R3 is greater than the voltage developed across R4; the output voltage is positive.

OPERATION BELOW RESONANCE. - When the input frequency is lower than the center frequency, the current and voltage phase relationships change. When the tuned circuit is operated at a frequency lower than resonance, the capacitive reactance increases and the inductive reactance decreases. Below resonance the tank acts like a capacitor and the secondary current leads primary tank voltage ep. This change to a leading secondary current rotates the vectors in a counterclockwise direction. From the vector diagram you should see that e2 is brought nearer in phase with ep, while el is shifted further out of phase with ep. The vector sum of ep and e2 is larger than that of e p and e1. Diode CR2 conducts more than diode CR1 below the center frequency. The voltage drop across R4 is larger than that across R3 and the output across both is negative.

Q.6] b] Draw the block diagram of pulse code modulation techniques & explain each block

Ans-
Simplified Block Diagram of a PCM Transmission System



PCM Sampling

Used to periodically sample the continually changing analog input voltage and convert it to a series of constant amplitude pulses that can be easily converted to binary PCM code.

- Natural Sampling

- Flat-top Sampling

Quantization – process of converting an infinite number of possibilities to a finite number of conditions (rounding off amplitudes)

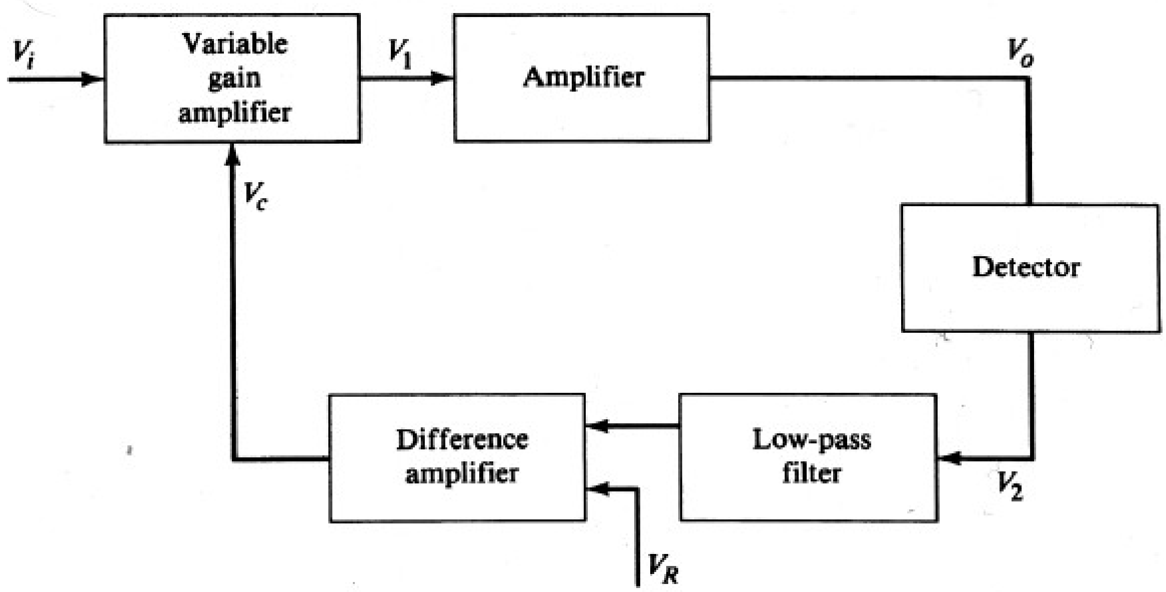
- Encoder- Each quantized level is converted into N bit digital word.
- Parellel to Serial- The encoder o/p is converted into a stream of pulses by parallel to serial converter.

Q.7] Explain the following (any two)

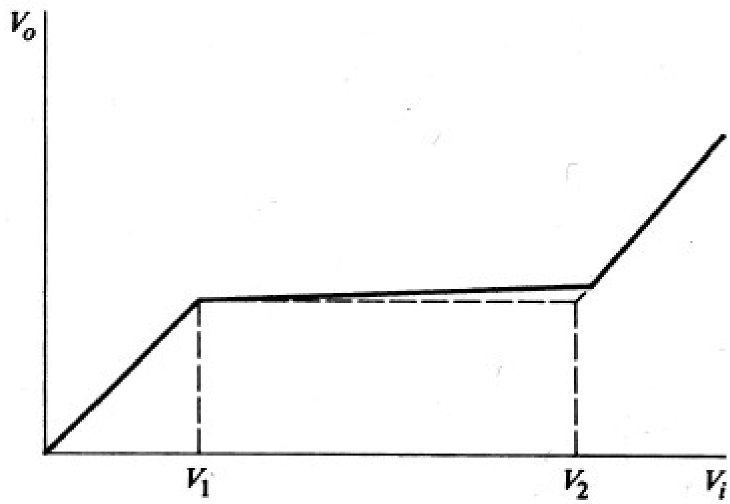
b] Automatic gain control:-

AGC was implemented in first radios for the reason of fading propagation (defined as slow
variations in the amplitude of the received signals) which required continuing adjustments in the receiver's gain in order to maintain a relative constant output signal.
Such situation led to the design of circuits, which primary ideal function was to maintain a constant signal level at the output, regardless of the signal's variations at the input of the system. Now AGC circuits can be found in any device or system where wide amplitude variations in the output signal could lead to a lost of information or to an unacceptable performance of the system. Automatic Gain Control (AGC) circuits are employed in many systems where the amplitude of an incoming signal can vary over a wide dynamic range. The role of the AGC circuit is to provide a relatively
constant output amplitude so that circuits following the AGC circuit require less dynamic range. If the signal level changes are much slower than the information rate contained in the signal, then an AGC circuit can be used to provide a signal with a well defined average level to downstream circuits. In most system applications, the time to adjust the gain in response to an input amplitude change should
remain constant, independent of the input amplitude level and hence gain setting of the amplifier. The large dynamic range of signals that must be handled by most receivers requires gain adjustment to
prevent overload or IM of the stages and to adjust the demodulator input level for optimum operation.
· A simple method of gain control would involve the use of a variable attenuator between the input and the first active stage. Such an attenuator, however, would decrease the signal level, but it would
also reduce the S/N of any but the weakest acceptable signal.

· Gain control is generally distributed over a number of stages, so that the gain in later stages (the IF amplifiers) is reduced first, and the gain in earlier stages (RF and first IF) is reduced only for signal levels sufficiently high to assure a large S/N.
· If the RF gain is small is enough switching in/out an attenuator at RF only for sufficiently high signal levels. Variable gain control for the later stages can operate from low signal levels. Variable-gain amplifiers are controlled electrically, and when attenuators are used in receivers, they are often operated electrically either by variable voltages for continuous attenuators or by electric switches
(relays or diodes) for fixed or stepped attenuators.
AGC Block

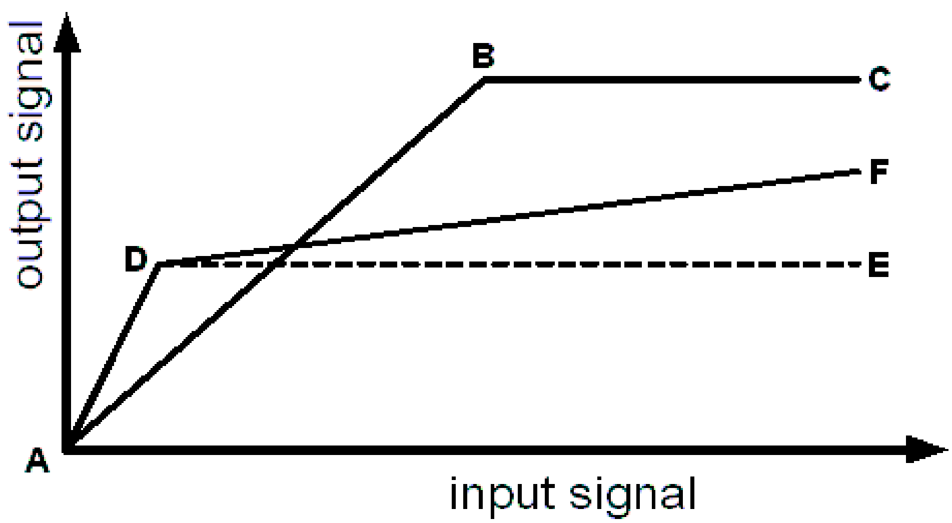


The input signal is amplified by a Variable Gain Amplifier (VGA), whose gain is controlled by an external signal VC. The output from the VGA can be further amplified by a second stage to generate and adequate level of Vo. Some the output signal's parameters, such as amplitude, carrier frequency, index of modulation or frequency, are sensed by the detector; any undesired component is filtered out and the remaining signal is compared with a reference signal. The result of the comparison is used to generate the control voltage (Vc) and adjust the gain of the VGA.
· If the control time constants are determined primarily by the detector circuit and the additional amplifier has a wider bandwidth than the detector, then the attack and decay times will be shortened by the amount of the post-amplification
· An AGC circuit in the receiver provides a substantially constant signal level to the demodulator independent of the input signal level.



For low input signals the AGC is disabled and the output is a linear function of the input, when the output reaches a threshold value (V1) the AGC becomes operative and maintains a constant output level until it reaches a second threshold value (V2). At this point, the AGC becomes inoperative again; this is usually done in order to prevent stability problems at high levels of gain.

· If the gain loop is much greater than 1, the steady state change in the input is greatly reduced.



The line A, B, C represents a system that has no AGC applied. The output increases linearly with the input signal until point B is reached, when some element in the signal chain overloads and becomes non-linear.

Generally from point B to C the output signal is distorted and, unless the input signal is reduced, the system is unusable. Increasing this value increases the slope of the line A to B and reduces the input signal level at which the signal distorts.

The line A, D, E represents a system that has AGC applied. The slope A, D is greater than unity and indicates that the AGC has gain prior to the AGC detector. The transition from a linear to constant output at

D is known as the AGC ‘knee’ or threshold. From D to E the output level does not increase in response to an increase in input signal. How flat the section of the line D to E is depends on the overall AGC loop gain and is called the “AGC Slope”.

Q.7] c] significance of signal to noise ratio.

Signal-to-noise ratio (often abbreviated SNR or S/N) is a measure used in science and engineering to quantify how much a signal has been corrupted by noise. It is defined as the ratio of signal power to the noise power corrupting the signal. A ratio higher than 1:1 indicates more signal than noise. While SNR is commonly quoted for electrical signals, it can be applied to any form of signal (such as isotope levels in an ice core or biochemical signaling between cells).

Signal-to-noise ratio is defined as the power ratio between a signal (meaningful information) and the background noise (unwanted signal):

$$\mathrm{SNR} = \frac{P_{\mathrm{signal}}}{P_{\mathrm{noise}}},$$

where P is average power. Both signal and noise power must be measured at the same or equivalent points in a system, and within the same system bandwidth. If the signal and the noise are measured across the same impedance, then the SNR can be obtained by calculating the square of the amplitude ratio:

$$\mathrm{SNR} = \frac{P_{\mathrm{signal}}}{P_{\mathrm{noise}}} = \left(\frac{A_{\mathrm{signal}}}{A_{\mathrm{noise}}}\right)^2,$$

where A is root mean square (RMS) amplitude (for example, RMS voltage). Because many signals have a very wide dynamic range, SNRs are often expressed using the logarithmic decibel scale. In decibels, the SNR is defined as

$$\mathrm{SNR_{dB}} = 10\log_{10}\left(\frac{P_{\mathrm{signal}}}{P_{\mathrm{noise}}}\right) = P_{\mathrm{signal,dB}} - P_{\mathrm{noise,dB}},$$

which may equivalently be written using amplitude ratios as

$$\mathrm{SNR_{dB}} = 10\log_{10}\left(\frac{A_{\mathrm{signal}}}{A_{\mathrm{noise}}}\right)^2 = 20\log_{10}\left(\frac{A_{\mathrm{signal}}}{A_{\mathrm{noise}}}\right).$$

The concepts of signal-to-noise ratio and dynamic range are closely related. Dynamic range measures the ratio between the strongest un-distorted signal on a channel and the minimum discernable signal, which for most purposes is the noise level. SNR measures the ratio between an arbitrary signal level (not necessarily the most powerful signal possible) and noise. Measuring signal-to-noise ratios requires the selection of a representative or reference signal. In audio engineering, the reference signal is usually a sine wave at a standardized nominal or alignment level, such as 1 kHz at +4 dBu (1.228 VRMS).

SNR is usually taken to indicate an average signal-to-noise ratio, as it is possible that (near) instantaneous signal-to-noise ratios will be considerably different. The concept can be understood as normalizing the noise level to 1 (0 dB) and measuring how far the signal 'stands out'.