MODULE - II

Multiple Choice Type Questions

1. The use of non-uniform quantization leads to [WBUT 2009, 2012, 2013, 2017]
   a) reduction in transmission bandwidth
   b) increase in maximum SNR
   c) increase in SNR for low level signals
   d) simplification of quantization process

Answer: (c)

2. In electrical telemetry system the transmission system is so designed that the interference due to noise remains such that [WBUT 2009]
   a) $S/N=1$
   b) $S/N=2$
   c) $S/N<<2$
   d) $S/N>>2$

Answer: (d)

3. In digital telemetry, commonly used modulation is [WBUT 2009, 2011, 2013, 2016]
   a) AM
   b) PCM
   c) PDM
   d) PWM

Answer: (b)

4. Guard band is essential in [WBUT 2009, 2013, 2017]
   a) FDM system
   b) TDM system
   c) CDM system
   d) both TDM and CDM systems

Answer: (b)

5. Which of the following system is digital? [WBUT 2009]
   a) PPM
   b) PCM
   c) PWM
   d) Pulse-frequency modulation

Answer: (b)

6. A 20 bit data format consists of 16 for bits of information and 4 bits for parity and sign. What is the efficiency? [WBUT 2010]
   a) 100%
   b) 80%
   c) 75%
   d) 60%

Answer: (b)

7. What is the function of WDM? [WBUT 2010]
   a) Change the transmission speed of the input signal
   b) Combine signals of different wavelengths to pass through a single fibre
   c) Separate signals of different wavelengths and couple them to different detectors
   d) Tap off part of energy of the incoming signal

Answer: (b)

8. Which one is the fastest type of ADC? [WBUT 2010, 2016]
a) Slope integration type
b) Successive approximation type
c) Parallel type
d) Counter type

Answer: (c)

9. Companding means [WBUT 2011]
a) compressing and expanding
b) compressing or expanding
c) companding and expanding
d) none of these

Answer: (a)

10. The complementary error function, erfc(.) is known as [WBUT 2011]
a) the probability of misinterpreting a bit, a'0' to be '1' and '1' to be '0'
b) the probability of misinterpreting a bit, a '0' to be '0' and '1' to be '1'
c) the probability of misinterpreting a bit, a '0' to be '1/2' and '1' to be '–1/2'
d) all of these

Answer: (a)

11. In PLL, the phase difference between two signals remains [WBUT 2011]
a) constant
b) varied
c) both (a) and (b)
d) none of these

Answer: (a)

12. At the receiving side of FDM/FM system, the filter used is [WBUT 2011]
a) LPF
b) BPF
c) Band Reject Filter
d) BPF and LPF both

Answer: (b)

13. In telemetering system, the S/N ratio should be [WBUT 2011, 2015, 2018]
a) S/N >> 1
b) S/N > = 1
c) S/N >> 2
d) S/N << 2

Answer: (c)

14. According to the IRIG standard, the channel-C should have Low Band Edge Frequency (LBEF) [WBUT 2011]
a) 189.75
b) 18.7
c) 34.0
d) none of these

Answer: (c)

15. If there are 2 channels and sampling frequency of each channel is 8 kHz, then line speed is [WBUT 2011]
a) 64 kbps
b) 128 kbps
c) 256 kbps
d) 512 kbps

Answer: (a)

16. Which of the following pulse modulation systems is analog? [WBUT 2012, 2017, 2018]
a) PCM
b) DPCM
c) PWM
d) DELTA

Answer: (c)

17. If there are 5 information pulses in 0.125 seconds, then the bit per second ratio (bpsr) is [WBUT 2012]

a) 0.025 b) 0.625 c) 1.6 d) 40

Answer: (d)

18. If there are two channels and the sampling frequency is 8kHz then frame time is [WBUT 2012, 2018]

a) 225 sec b) 125 sec c) 125 μs d) 64 sec

Answer: (c)

19. Which of the following systems is digital? [WBUT 2013, 2016]

a) PPM b) PCM c) PWM d) PAM

Answer: (b)

20. In PLL the phase difference between two signals remains [WBUT 2013]

a) constant b) varied
c) both (a) and (b) d) none of these

Answer: (a)

21. Using an oscilloscope to display overlayed received data bits that provide information on noise, jitter and linearity is called a(n) [WBUT 2014]

a) eye pattern b) constellation pattern
c) statistical concentration d) loopback

Answer: (a)

22. The acronym OFDM refers to [WBUT 2014]

a) Orthogonal Frequency Division Modulation
b) Optional Frequency Division Modulation
c) Over Frequency Division Multiplexing
d) Orthogonal Frequency Division Multiplexing

Answer: (d)

23. ARQ stands for [WBUT 2014]

a) Accelerated Redirection Facility
b) Amplitude Ratiodetector Quantizing Noise
c) Automatic Repeat Request
d) Aerial Range Quartz Crystal

Answer: (c)

24. Shannon's law relates [WBUT 2014]

a) antenna gain to bandwidth
b) frequency to antenna gain
c) antenna gain to transmission losses
d) information carrying capacity to S/N ratio

Answer: (d)

25. The main characteristic of thermal noise is that **[WBUT 2014]**
a) it has a flat frequency spectrum
b) it has a slot frequency spectrum
c) it has a thick frequency spectrum
d) none of these

Answer: (a)

26. In an AM system, the total power radiated 600 W. The power of the carrier is 400 W. The modulation index is **[WBUT 2015]**
a) 1
b) 0.5
c) 0.75
d) 0.6

Answer: (a)

27. The relation between number of quantization level q and number of bits per sample n is **[WBUT 2015]**
a) $q = 2^n$
b) $n = 2^q$
c) $n = q - 1$
d) $q = n - 1$

Answer: (a)

28. If there are two channel and the sampling frequency is 8 kHz then frame time is **[WBUT 2015]**
a) 225 sec
b) 125 sec
c) 125 μs
d) 64 sec

Answer: (c)

29. In a PCM system the number of quantization levels is 16 and the maximum signal frequency is 4 kHz. The bit transmission rate is **[WBUT 2015, 2018]**
a) 64 kbites/sec
b) 16 kbites/sec
c) 32 kbites/sec
d) 32 bites/sec

Answer: (b)

30. In order to reduce Quantization noise, one must **[WBUT 2015]**
a) increase the number of samples per second
b) send pulse whose sides are more nearly vertical
c) use an RF amplifier in the receiver
d) increase the number of standard amplitudes

Answer: (d)

Short Answer Type Questions

1. What do you understand by intersymbol interference? How can intersymbol interference be eliminated? Explain with appropriate diagrams. **[WBUT 2009, 2016]**

OR,

Write short note on Inter symbol interference (ISI) **[WBUT 2012]**

OR,

What do you mean by Inter Symbol Interference (ISI)? How it is being removed? **[WBUT 2014, 2018]**

Answer:

The intersymbol interference ISI arises due to the imperfections in the overall frequency response of the system. When a short pulse of duration T_b seconds is transmitted through a bandlimited system, then the frequency components contained in the input pulse are differentially attenuated and more importantly differentially delayed by the system. Due to this, the pulse appearing at the output of the system will be dispersed over an interval which is longer than T_b seconds. Due to this dispersion, the symbols each of duration T_b will interfere with each other when transmitted over the communication channel. This will result in the intersymbol interference (ISI). The transmitted pulse of duration T_b seconds and the dispersed pulse of duration more than T_b seconds are shown in figure below.

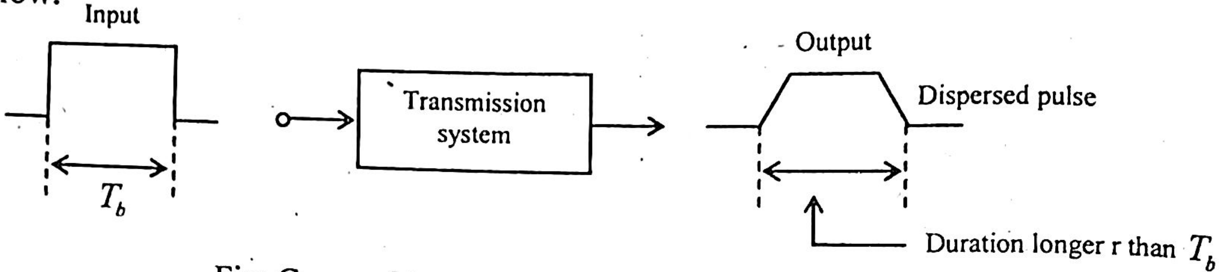


Fig: Cause of Intersymbol Interference (ISI)

Following are the effects of ISI:

(i) In the absence of ISI and noise, the transmitted bit can be decoded correctly at the receiver.

(ii) The presence of ISI will introduce errors in the decision at the receiver output.

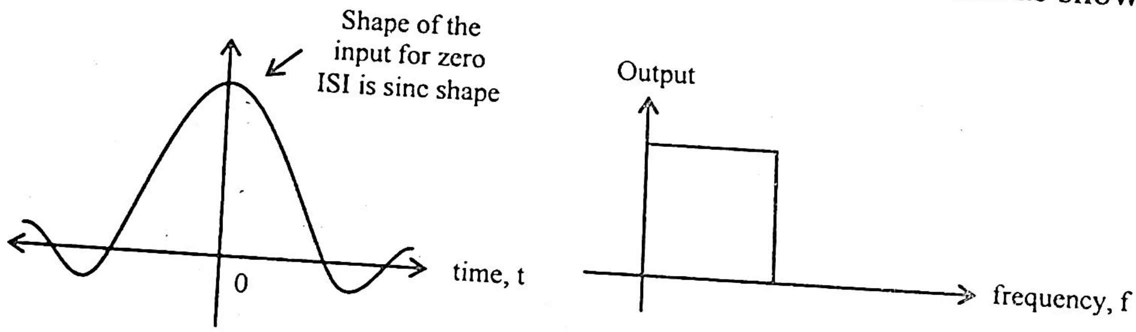
(iii) Hence, the receiver can make an error in deciding whether it has received a logic 1 or a logic 0.

Remedy to Reduce ISI

(i) It has been proved that the function which produces a zero intersymbol interference is a sinc function. Hence, instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero.

(ii) This is known as Nyquist Pulse Shaping. The sinc pulse transmitted to have a zero ISI has been shown below

(iii) Further, we know that Fourier transform of a sinc pulse is a rectangular function. Hence, to preserve all the frequency components, the frequency response of the fiber must be exactly flat in the pass band and zero in the attenuation band as shown in the figure below.



(a) Ideal pulse shape for zero ISI

(b) Frequency response of the filter

2. Schematically explain the operation of a PLL and its application as a frequency divider. **[WBUT 2010, 2012]**

OR,

Write short note on PLL. **[WBUT 2012, 2013]**

OR,

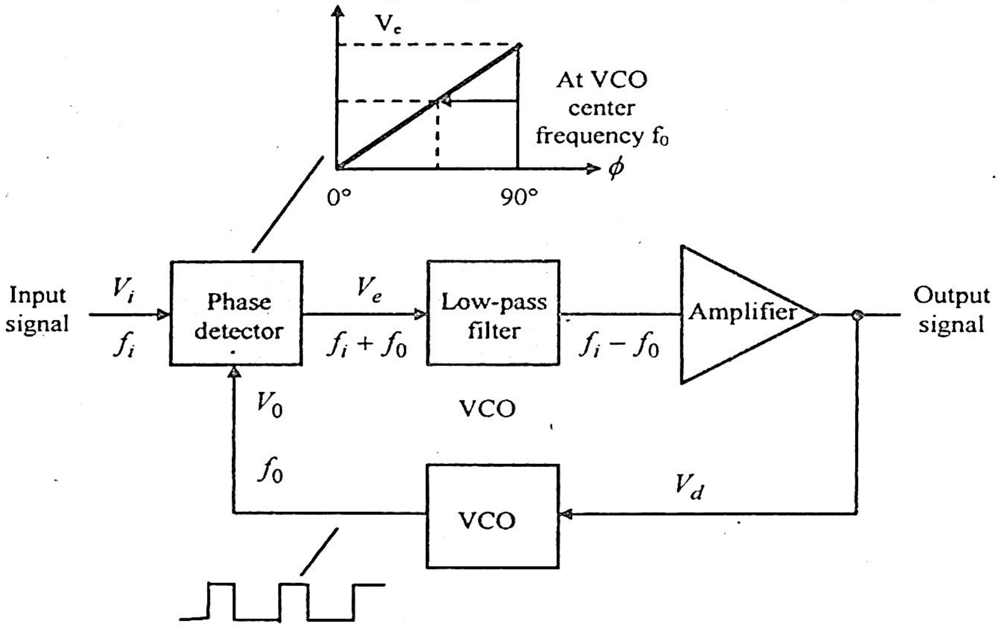
Explain the operation of PLL and its application as a frequency divider. **[WBUT 2013, 2014]**

Answer:

Phase-Locked Loop

A phase-locked loop (PLL) is an electronic circuit that consists of a phase detector, a, low-pass filter, and a voltage-controlled oscillator connected as shown in Fig. Common applications of a PLL include (1) frequency synthesizers that provide multiples of a reference signal frequency (2) FM demodulation networks for FM operation with excellent linearity between the input signal frequency and the PLL output voltage, (3) demodulation of the two data transmission or carrier frequencies in digital-data transmission used in frequency-shift keying (FSK) operation, and (4) a wide variety of areas including modems, telemetry receivers and transmitters, tone decoders, AM detectors and tracking filters.

An input signal V_i and that from a VCO, V_o, are compared by a phase comparator (refer to Fig.), providing an output voltage V_e that represents the phase difference between the two signals. This voltage is then fed to a low-pass filter, which provides an output voltage (amplified if necessary) that can be taken as the output voltage from the PLL and is used internally as the voltage to modulate the VCO's frequency. The closed-loop operation of the circuit is to maintain the VCO frequency locked to that of the input signal frequency.



Basic PLL Operation

The basic operation of a PLL circuit can be explained using the circuit of Fig. above as reference. We will first consider the operation of the various circuits in the phase-locked

2. Schematically explain the operation of a PLL and its application as a frequency divider. **[WBUT 2010, 2012]**

OR,

Write short note on PLL. **[WBUT 2012, 2013]**

OR,

Explain the operation of PLL and its application as a frequency divider. **[WBUT 2013, 2014]**

Answer:

Phase-Locked Loop

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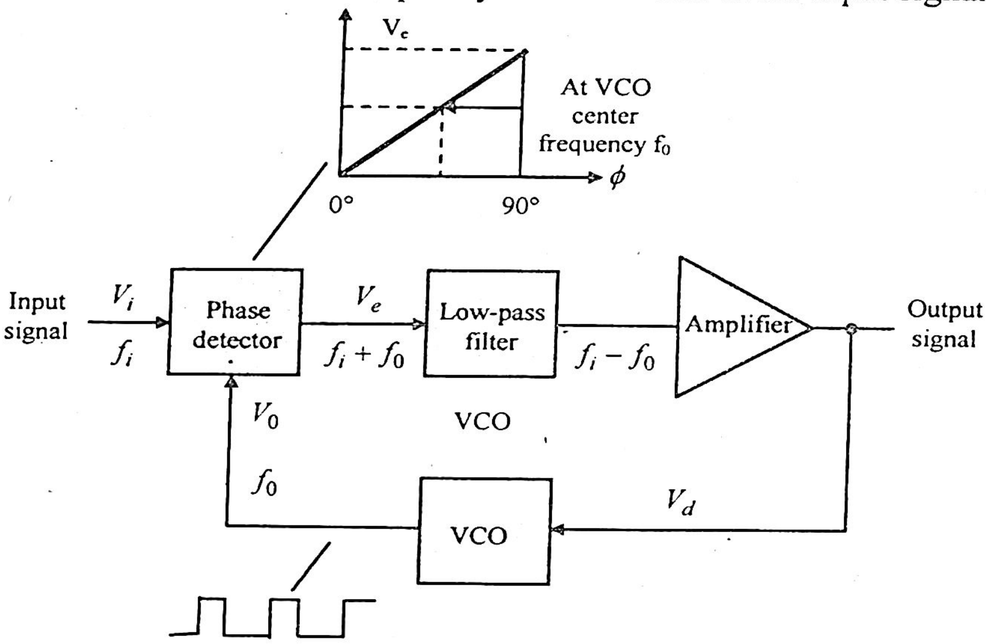
An input signal V_i and that from a VCO, V_o, are compared by a phase comparator (refer to Fig.), providing an output voltage V_e that represents the phase difference between the two signals. This voltage is then fed to a low-pass filter, which provides an output voltage (amplified if necessary) that can be taken as the output voltage from the PLL and is used internally as the voltage to modulate the VCO's frequency. The closed-loop operation of the circuit is to maintain the VCO frequency locked to that of the input signal frequency.



Basic PLL Operation

The basic operation of a PLL circuit can be explained using the circuit of Fig. above as reference. We will first consider the operation of the various circuits in the phase-locked loop when the loop is operating in lock (the input signal frequency and the VCO frequency are the same). When the input signal frequency is the same as that from the VCO to the comparator, the voltage V_d taken as output is the value needed to hold the VCO in lock with the input signal. The VCO then provides output of a fixed-amplitude square-wave signal at the frequency of the input. Best operation is obtained if the VCO center frequency f_o is set with the dc bias voltage midway in its linear operating range. The amplifier allows this adjustment in dc voltage from that obtained as output of the filter circuit. When the loop is in lock, the two signals to the comparator are of the same frequency, although not necessarily in phase. A fixed phase difference between the two signals to the comparator results in a fixed dc voltage to the VCO. Changes in the input signal frequency then result in change in the dc voltage to the VCO. Within a capture-and-lock frequency range, the dc voltage will drive the VCO frequency to match that of the input.

While the loop is trying to achieve lock, the output of the phase comparator contains frequency components at the sum and difference of the signals compared. A low-pass filter passes only the lower frequency component of the signal, so that the loop can obtain lock between input and VCO signals.

Owing to the limited operating range of the VCO and the feedback connection of the PLL circuit, there are two important frequency bands specified for a PLL. The capture range of a PLL is the frequency range centered about the VCO free-running frequency f_o over which the loop can acquire lock with the input signal. Once the PLL has achieved capture, it can maintain lock with the input signal over a somewhat wider frequency range called the lock range.

3. Describe the operation of sample and hold circuit with suitable circuit diagram. [WBUT 2010, 2014]

Answer:

Natural Sampling

Natural sampling is a practical method of sampling which is discussed in this section. In natural sampling the pulse has a finite width equal to τ.

Consider an analog continuous-time signal $x(t)$ to be sampled at the rate of f_s Hertz. Here it is assumed that f_s is higher than Nyquist rate such that sampling theorem is satisfied.

Again, let us consider a sampling function $c(t)$ which is a train of periodic pulses of width τ and frequency equal to $f_s\ Hz$.

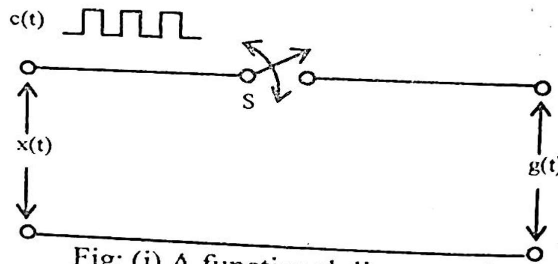


Fig: (i) A functional diagram of a Natural Sampler

Figure (i) shows a functional diagram of a natural sampler. With the help of this natural sampler, a sampled signal $g(t)$ is obtained by multiplication of sampling function $c(t)$ and input signal $x(t)$.

Now, according to figure (i), we have
When $c(t)$ goes high, the switch 'S' is closed.
Therefore

$$g(t)=x(t) \quad \text{when} \quad c(t)=A \qquad \ldots(a)$$

and $$g(t)=0 \quad \text{when} \quad c(t)=0 \qquad \ldots(b)$$

where A is the amplitude of $c(t)$.

The waveforms of signals $x(t)$, $c(t)$ and $g(t)$ have been illustrated in figure 1 (a), (b) and (c) respectively.

Now, the sampled signal $g(t)$ may also be described mathematically as

$$g(t)=c(t).x(t) \qquad \ldots(c)$$

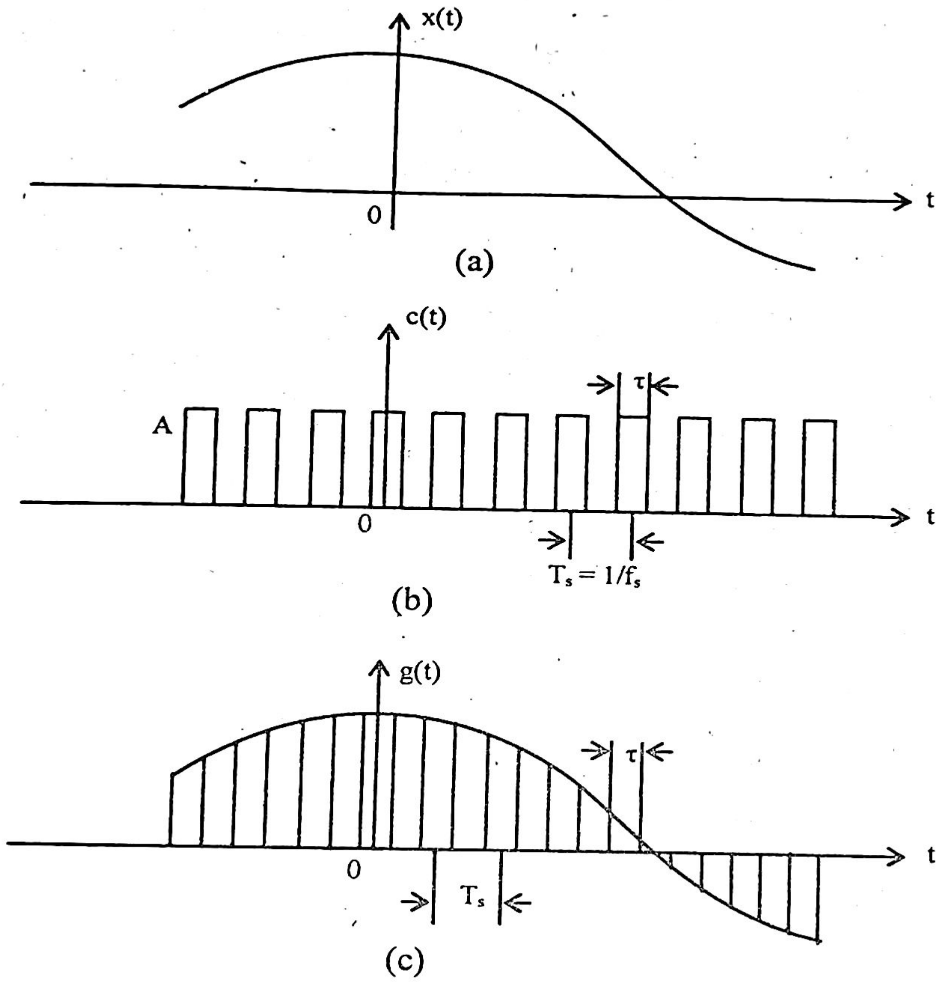


Fig: 1. (a) Continuous time signal x(t)
(b) Sampling function waveform i.e., periodic pulse train
(c) Naturally sampled signal waveform g(t)

4. What are the two telemetry standards of baseband configuration in terms of frequency as stipulated by IRIG? **[WBUT 2011, 2015]**

Answer:
PBW (Proportional bandwidth)
CBW (Constant Bandwidth) configurations.

5. A system has BER = 10^{-5}. What does it signify? **[WBUT 2012]**

Answer:
A system has BER = 10^{-5} means 1 error is tolerated in 10^5 bit sent.

6. What do you mean by CBW and PBW configurations? **[WBUT 2017]**

Answer:
CBW (Constant bandwidth) and PBW (Proportional bandwidth) are the standards of baseband configuration developed by Inter Range Instrumentation Group (IRIG) in 1975.

7. What do you mean by BER? A system has BER $=10^{-5}$. What does it signify? **[WBUT 2017, 2018]**

Answer:
1st Part: *Refer to Question No. 1(a) (1st part) of Long Answer type Questions.*

2nd Part: *Refer to Question No. 5 of Short Answer Type Questions.*

Long Answer Type Questions

1. a) What is BER? **[WBUT 2009, 2012, 2015]**
What is the highest allowed BER in speech transmission?
b) In a digital data transmission system the code word is of 8-bit and the bit error probability is 10^{-2}. Calculate the probability that the code word would have 2 errors and 3 errors. **[WBUT 2009, 2015]**

Answer:
a) In digital transmission, the number of **bit errors** is the number of received bits of a data stream over a communication channel that has been altered due to noise, interference, distortion or bit synchronization errors.
The **bit error rate** or **bit error ratio (BER)** is the number of bit errors divided by the total number of transferred bits during a studied time interval. BER is a unitless performance measure, often expressed as a percentage.
Highest allowed BER in speech signal transmission is 10^{-5} which means that there is an error of 1bit in 10^5 bits.

b) Let the probability is P_{be}^{r}
From given data we get
$P_{be}^{r}=(10^{-1})^{3}$ in the two cases whereas the values of ${}^{n}C_{r}$ are 28 & 112. Hence
$P(2,8)=10^{-4}.28=28\times10^{-4}$
$P(3,8)=10^{-6}.112=1.12\times10^{-4}$

2. a) Draw a schematic arrangement to show that PWM and PPM can be generated from PAM signals.
b) Explain the principle of operation of PCM transmitter and receiver.
[WBUT 2012, 2017, 2018]

Answer:

a) Pulse Modulation Methods

There are several major classes of pulse modulation suitable for lightwave communications, and they are compared in Fig 2. Here is a brief description of each method:

1. Pulse-Amplitude Modulation (PAM): In this modulation scheme the amplitude of the pulses is directly proportional to the amplitude of the modulating signal. PAM is closely related to amplitude modulation in that PAM can be achieved by simply sampling at a uniform interval brief segments of an AM signal. An obvious application of PAM is the transmission of two or more signals over a single lightwave channel.

2. Pulse-Width Modulation (PWM): This method is also known as Pulse-Duration Modulation (PDM). The duration of individual pulses within a pulse train is made proportional to the amplitude of the modulating signal.

3. Pulse-Position Modulation (PPM): Here the amplitude of the input signal controls the relative position of individual pulses in a pulse stream. Unlike PAM and PWM, all the pulses in PPM have precisely the same amplitude and duration. This means the PPM receiver can be optimized for the processing of identically shaped pulses. This gives a higher degree of noise immunity than provided by the PWM and especially PAM. Another advantage is that high peak power optical sources such as injection lasers can be used to full advantage.

The detection of PPM pulses by a receiver requires synchronization with the transmitter. This implies the necessity to transmit a clock signal along with the data or on a separate channel.

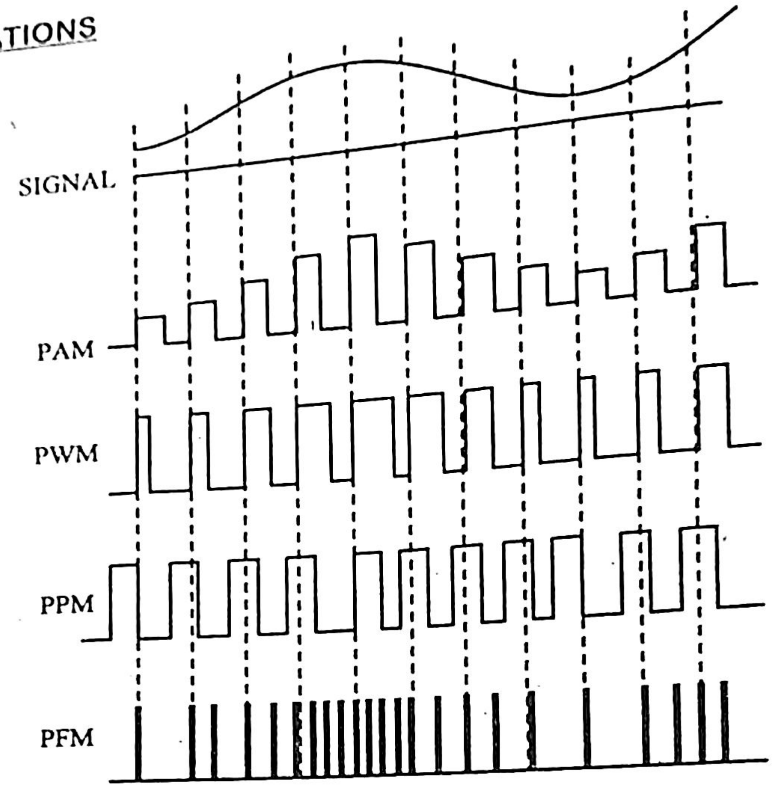
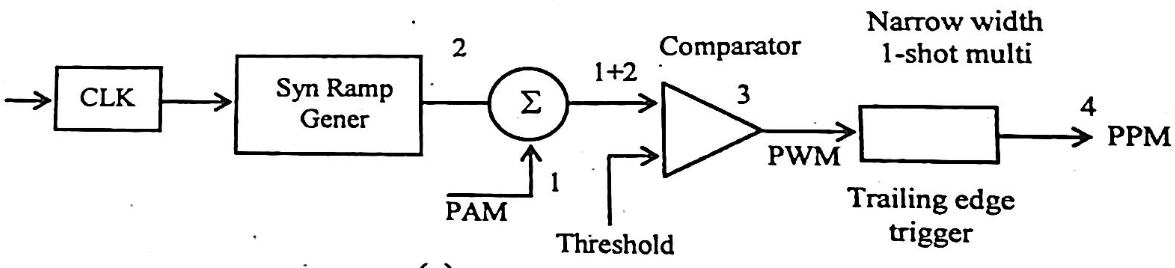
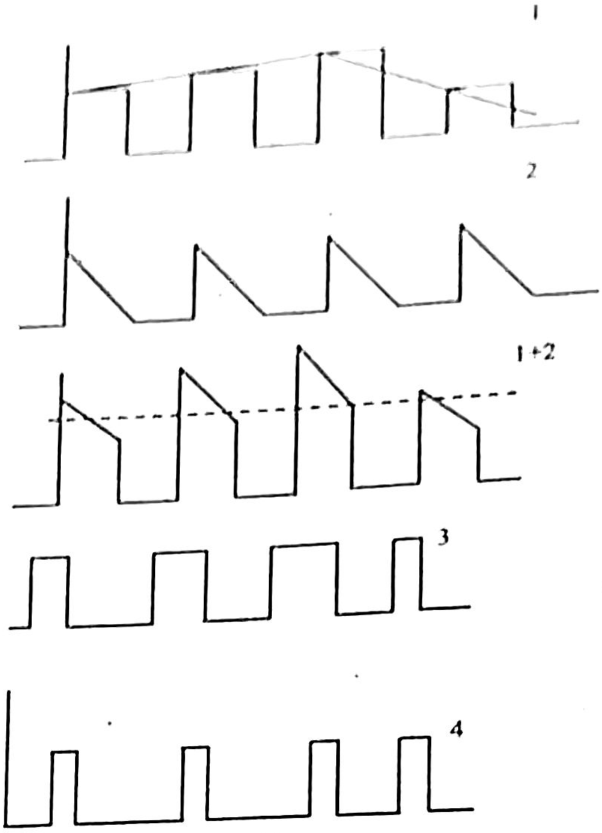


Fig1: Pulse modulation methods

The PWM & PPM signals are obtained from PAM signals by a scheme of the type show in fig 2 (a). Fig (b) shows stepwise pulse forms obtained from the scheme.



(a)



(b)

Fig 2: (a) Scheme of obtaining PWM and PPM pulses from PAM system, (b) Pulse waveforms at different stages

b) A PCM Generator or Transmitter

In PCM generator of figure below, the signal $x(t)$ is first passed through the low-pass filter of cutoff frequency f_m Hz. This low-pass filter blocks all the frequency components which are lying above f_m Hz.

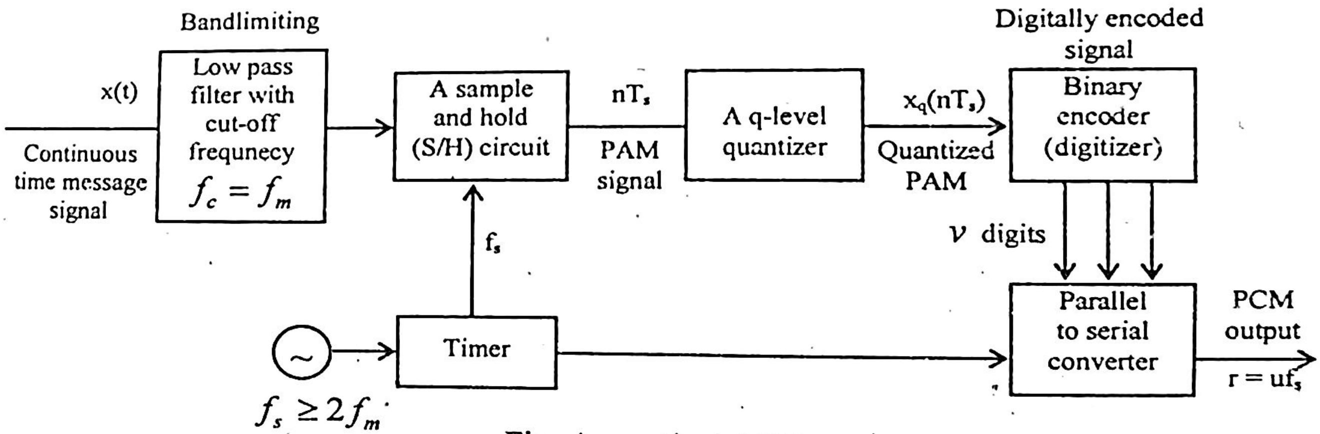


Fig. A practical PCM generator

This means that now the signal $x(t)$ is bandlimited to f_m Hz. The sample and hold circ then samples this signal at the rate of f_s. Sampling frequency f_s is selected sufficien above nyquist rate to avoid aliasing i.e., $f_s \geq 2f_m$
In figure above, the output of sample and hold circuit is denoted by $x(nT_s)$. The sign $x(nT_s)$ is discrete in time and continuous in amplitude. A q-level quantizer compar input $x(nT_s)$ with its fixed digital levels. It then assigns any one of the digital level $x(nT_s)$ which results in minimum distortion or error. This error is called **quantizatio error**. Thus, output of quantizer is a digital level called $x_q(nT_s)$. Now, the quantiz signal level $x_q(nT_s)$ is given to binary encoder. This encoder converts input signal to ' digits binary word. Thus $x_q(nT_s)$ is converted to 'v' binary bits. This encoder is al known as digitizer.

PCM Transmission Path
The path between the PCM transmitter and PCM receiver over which the PCM sign travel, is called as PCM transmission path and it is as shown in figure below. The mo important feature of PCM system lies in its ability to control the effects of distortion ar noise when the PCM wave travels on the channel. PCM accomplishes this capacity b means of using a chain of regenerative repeaters as shown in figure below. Such repeate are spaced close enough to each other on the transmission path. The regenerati performs three basic operations namely equalization, timing and decision making. Henc each repeater actually reproduces the clean noise free PCM signal from the PCM signa distorted by the channel noise. This improves the performance of PCM in presence c noise.

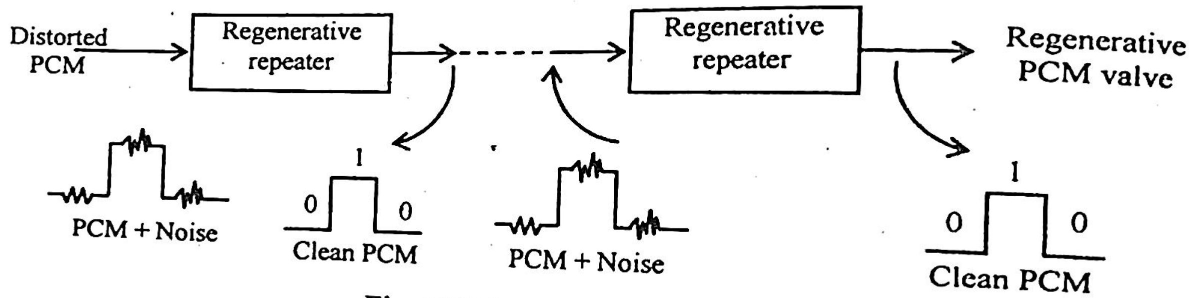


Fig: PCM transmission path

PCM Receiver
Figure 1 (a) below shows the block diagram of PCM receiver and 1 (b) shows the reconstructed signal. The regenerator at the start of PCM receiver reshapes the pulse and removes the noise. The signal is then converted to parallel digital words for each sample.

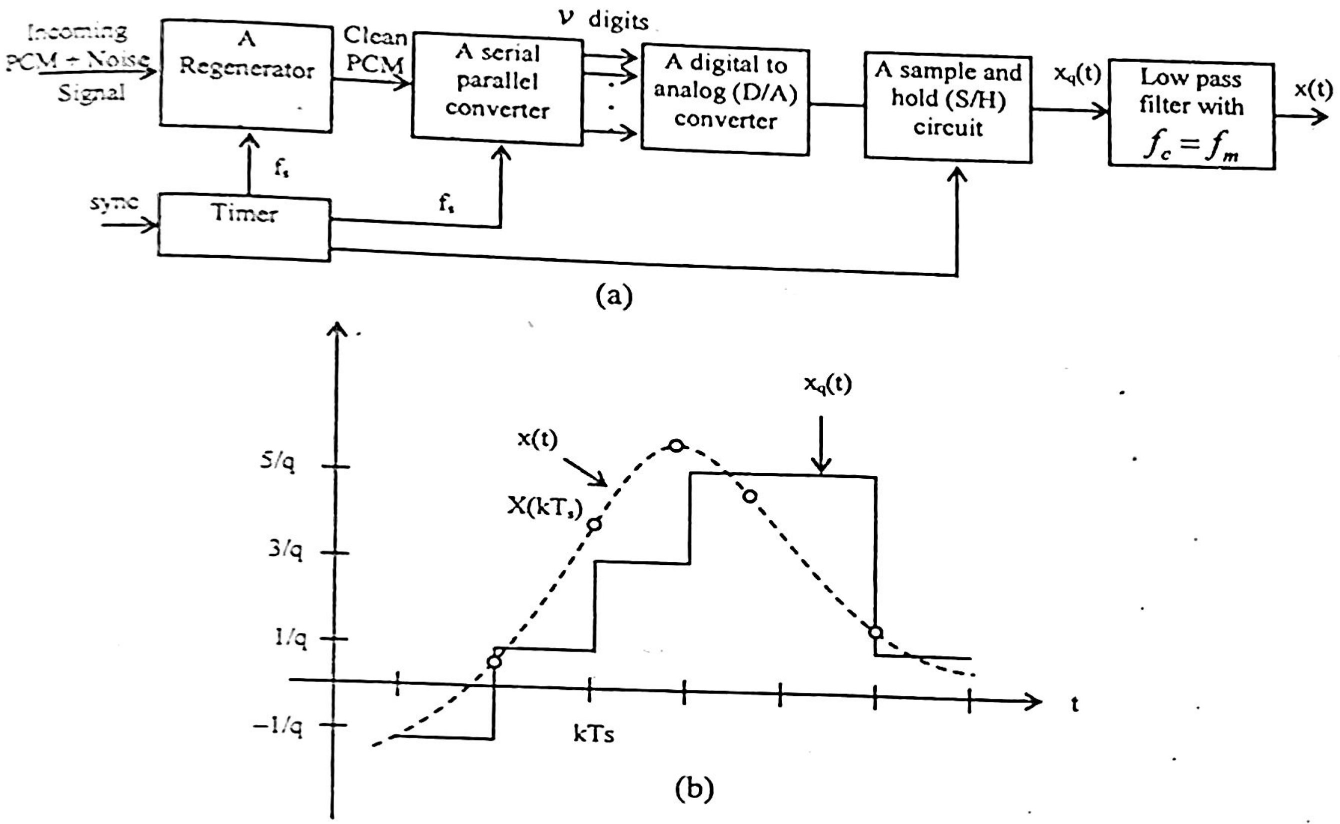


Fig: 1 (a) PCM receiver (b) Reconstructed waveform

Now, the digital word is converted to its analog value denoted as $x_q(t)$ with the help of a sample and hold circuit. This signal, at the output of sample and hold circuit, is allowed to pass through a low pass reconstruction filter to get the appropriate original message signal denoted as $y(t)$.

3. Give the block diagram of a clock recovery circuit and also explain the operation.

[WBUT 2013, 2018]

Answer:

Fig. 1 shows the block diagram of a clock recovery circuit for an NRZ s.gnal, where a narrowband filter selects the fundamental frequency component at the half bit rate, $f_b/2$ and a non-linear circuit doubles this frequency to f_b. Then this single-frequency signal is compared with a sinusoid signal generated by a voltage-controlled oscillator (VCO) and their frequency difference is used as the error signal to control the VCO. This ensures that the sinusoid generated by the VCO has exactly the same frequency and phase as the clock extracted from the incoming signal. This also allows the clock to be recovered from PRBS signals with long zeroes.

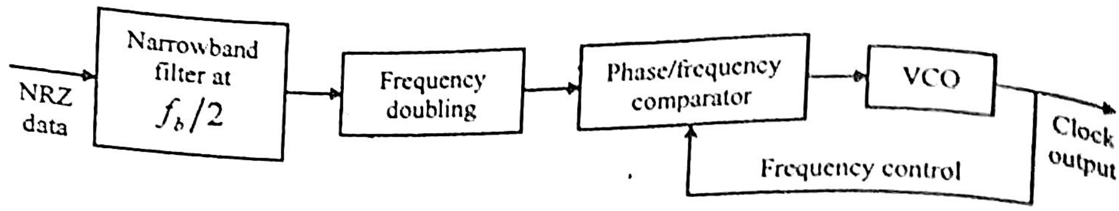


Fig: 1 Block diagram of a clock recovery circuit for NRZ signal.
VCO: Voltage Controlled Oscillator

4. Prove that the average error probability $P(E) = Q(V_P / \sigma_n)$, where V_P is the received pulse amplitude and σ_n is the r.m.s. value of the noise. [WBUT 2015]

Answer:

The probability of misinterpreting a bit, in terms of V_d and V_n, can be written as

$$P_{be} = \frac{1}{2} erfc\left(\frac{V_d}{\sqrt{2}V_n}\right)$$

Often a Q-function is defined to denote the bit error probability. The relation is

$$P_{be} = Q\left(\frac{V_d}{V_n}\right)$$

This shows that

$$Q(x) = \frac{1}{2} erfc\left(\frac{x}{\sqrt{2}}\right)$$

It must be noted that $erfc(x) = 1 - erf(x)$ and the series form of $erfc(x)$ is given as

$$erfc(x) = \left(\frac{e^{-x^2}}{(\sqrt{\pi} \cdot x)}\right)\left(1 - \frac{1}{(2x^2)} + \frac{(1\cdot 3)}{(2x^2)^2} - \frac{(1\cdot 3\cdot 5)}{(2x^2)^3} \cdots\cdots\right)$$

and $erf(x)$ is given as

$$erf(x) = (2\sqrt{\pi})\int_0^x e^{-u^2} du$$

Q-function, applicable for Gaussian distribution of a random variable, is defined in terms of the ratio of the variable (x) and the standard deviation σ. Thus $Q(3)$ is the probability that the random variable will have a positive value more than thrice its standard deviation, i.e., probability that $\frac{x}{\sigma} > 3$. Hence, if a noise has zero mean, the probability that the noise exceeds a threshold V_{th}, is $Q\left(\frac{V_{th}}{\sigma}\right)$. For $\left(\frac{x}{\sigma}\right)_{th} > 4$, the approximate relation is given by

$$Q\left\{\left(\frac{x}{\sigma}\right)_{th}\right\} = \exp\frac{\left\{\frac{-\left(\frac{x}{\sigma}\right)_{th}^{2}}{2}\right\}}{\left(\left(\frac{x}{\sigma}\right)_{th}\sqrt{2\pi}\right)}$$

and hence

$$Q\left(\frac{V_{th}}{\sigma}\right) = Q\left(\frac{V}{2\sigma}\right)$$

Hence, P_t is obtained as

$$P_t = 0.5 \times Q\left(\frac{V}{2\sigma}\right) + 0.5 \times Q\left(\frac{V}{2\sigma}\right) = Q\left(\frac{V}{2\sigma}\right)$$

This is often written in terms of the signal-to-noise ratio of the system. one very common technique of writing this ratio is in terms of the voltage ratio or power ratio. The peak signal-to-noise voltage ratio is defined as

$(S/N)_v$ = Peak signal voltage/rms noise voltage, so that

$(S/N)_v = V/\sigma$ or A/σ

and $\quad P_t = Q\left(\frac{V}{2\sigma}\right) = Q\left[\frac{1}{2}(S/N)_v\right]$

In power form, $(S/N)_p$ = (peak voltage)^2/mean sq. noise voltage $= (S/N)_v^2$

For a bipolar system the excursion is between $+\frac{V}{2}$ to $-\frac{V}{2}$ instead of V and 0.

For a bipolar scheme, therefore,

$$\left(\frac{S}{N}\right)_v = \frac{V}{2\sigma}$$

and $\quad P_t = Q\left(\frac{V}{2\sigma}\right) = Q\left(\frac{S}{N}\right)_v$

5. Draw and explain the block diagram of FDM / FM / FM telemetry system for both the transmitter and receiver. [WBUT 2017]

Answer:

The figure below shows the block diagram of the FDM/FM telemetry system that has two sections i) transmitter, and ii) receiver.

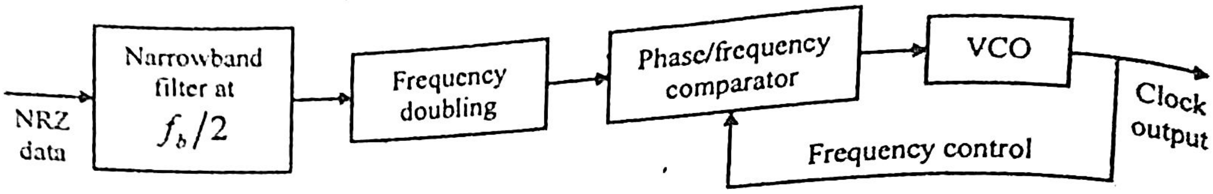


Fig: 1 Block diagram of a clock recovery circuit for NRZ signal.
VCO: Voltage Controlled Oscillator

4. Prove that the average error probability $P(E_r)=Q(V_P/\sigma_n)$, where V_P is the received pulse amplitude and σ_n is the r.m.s. value of the noise. [WBUT 2015]

Answer:

The probability of misinterpreting a bit, in terms of V_d and V_n, can be written as

$$P_{be}=\frac{1}{2}erfc\left(\frac{V_d}{\sqrt{2}V_n}\right)$$

Often a Q-function is defined to denote the bit error probability. The relation is

$$P_{be}=Q\left(\frac{V_d}{V_n}\right)$$

This shows that

$$Q(x)=\frac{1}{2}erfc\left(\frac{x}{\sqrt{2}}\right)$$

It must be noted that $erfc(x)=1-erf(x)$ and the series form of $erfc(x)$ is given as

$$erfc(x)=\left(\frac{e^{-x^2}}{(\sqrt{\pi}\cdot x)}\right)\left(1-\frac{1}{(2x^2)}+\frac{(1\cdot 3)}{(2x^2)^2}-\frac{(1\cdot 3\cdot 5)}{(2x^2)^3}\cdots\cdots\right)$$

and $erf(x)$ is given as

$$erf(x)=(2\sqrt{\pi})\int_0^x e^{-u^2}du$$

Q-function, applicable for Gaussian distribution of a random variable, is defined in terms of the ratio of the variable (x) and the standard deviation σ. Thus $Q(3)$ is the probability that the random variable will have a positive value more than thrice its standard deviation, i.e., probability that $\frac{x}{\sigma}>3$. Hence, if a noise has zero mean, the probability that the noise exceeds a threshold V_{th}, is $Q\left(\frac{V_{th}}{\sigma}\right)$. For $\left(\frac{x}{\sigma}\right)_{th}>4$, the approximate relation is given by

$$Q\left\{\left(\frac{x}{\sigma}\right)_{th}\right\} = \frac{\exp\left\{\dfrac{-\left(\dfrac{x}{\sigma}\right)_{th}^{2}}{2}\right\}}{\left(\left(\dfrac{x}{\sigma}\right)_{th}\sqrt{2\pi}\right)}$$

and hence

$$Q\left(\frac{V_{th}}{\sigma}\right) = Q\left(\frac{V}{2\sigma}\right)$$

Hence, P_t is obtained as

$$P_t = 0.5\times Q\left(\frac{V}{2\sigma}\right) + 0.5\times Q\left(\frac{V}{2\sigma}\right) = Q\left(\frac{V}{2\sigma}\right)$$

This is often written in terms of the signal-to-noise ratio of the system. one very common technique of writing this ratio is in terms of the voltage ratio or power ratio. The peak signal-to-noise voltage ratio is defined as

$(S/N)_v$ = Peak signal voltage/rms noise voltage, so that

$(S/N)_v = V/\sigma$ or A/σ

and $\quad P_t = Q\left(\dfrac{V}{2\sigma}\right) = Q\left[\dfrac{1}{2}(S/N)_v\right]$

In power form, $(S/N)_p$ = (peak voltage)^2/mean sq. noise voltage $= (S/N)_v^2$

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$$\left(\frac{S}{N}\right)_v = \frac{V}{2\sigma}$$

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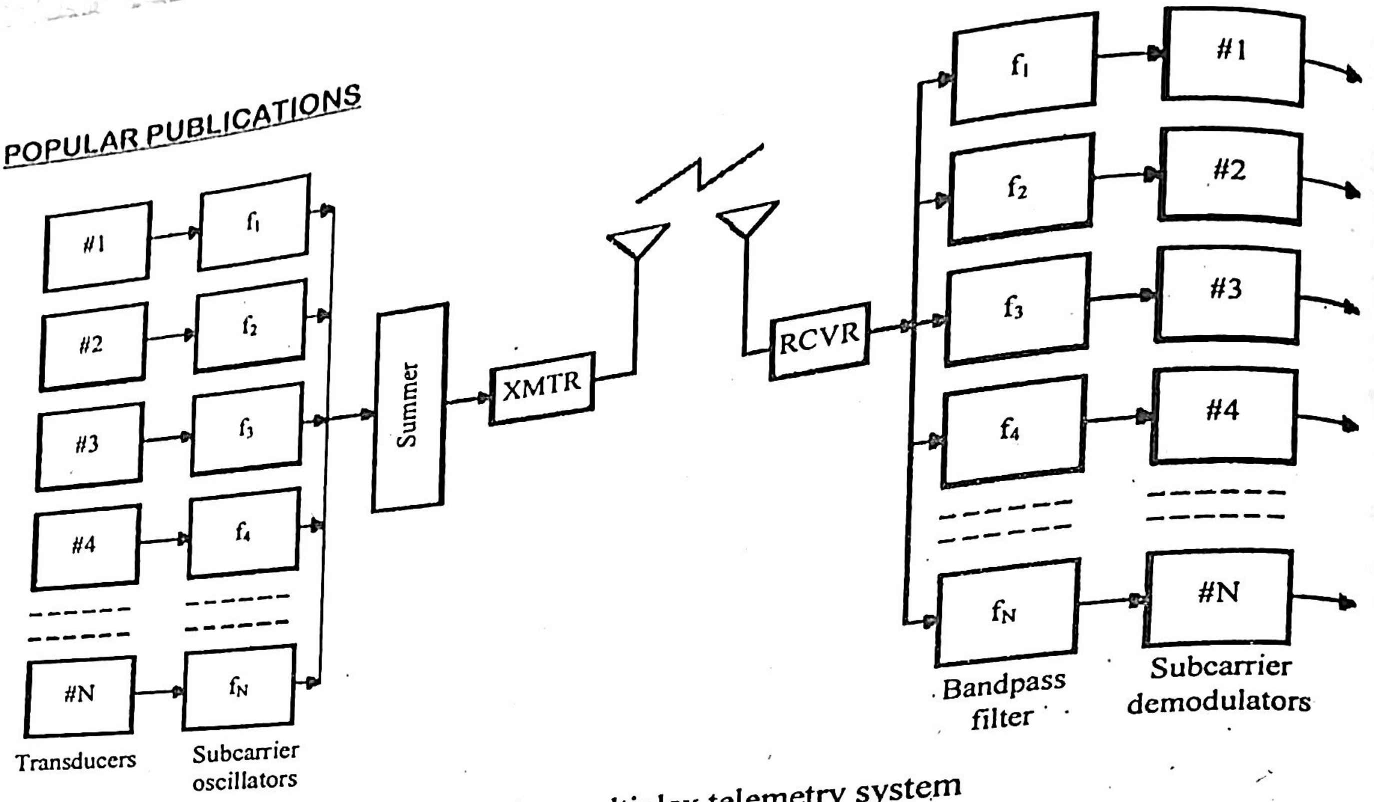
5. Draw and explain the block diagram of FDM / FM / FM telemetry system for both the transmitter and receiver. **[WBUT 2017]**

Answer:

The figure below shows the block diagram of the FDM/FM telemetry system that has two sections i) transmitter, and ii) receiver.



Frequency-division multiplex telemetry system

The measurement signals from transducers modulate "subcarrier" oscillators tuned to different frequencies. The output voltages from the subcarrier oscillators are then summed linearly in the "summer block. The composite signal is used to modulate the downlink transmitter in the "XMTR" block.

In the receiving station, the composite signal is available at the output of the receiver demodulator, which is then fed to" bandpass filters" that are tuned to the center frequencies of the subcarrier oscillators. The outputs from the filters are then demodulated in the "subcarrier demodulators" section to recover and the original transducer signals.

All types of modulation can be used for both the subcarrier oscillators and the prime carrier. The transmission system for frequency division multiplex systems is designated by first giving the modulation for the subcarriers and then the prime carrier. Thus FM/AM would indicate a frequency-division multiplex system in which the subcarriers are frequency modulated and the prime carrier is amplitude modulated by the composite subcarrier signal.

6. Write short notes on the following:

a) IRIG standards **[WBUT 2009]**
b) Importance of M2M system in modern society **[WBUT 2010]**
c) SCO **[WBUT 2014]**
d) Quantization **[WBUT 2015]**
e) PCM **[WBUT 2017]**
f) Sample and Hold circuit **[WBUT 2018]**
g) PLL **[WBUT 2016, 2018]**

Answer:

a) IRIG Standards

The IRIG standards specify the subcarrier frequencies, nominal peak frequency deviations, nominal modulation indices, maximum signal variations, and many other system parameters. IRIG standards specify two frequency modulation schmes for the subcarriers: *proportional bandwidth* (PBW) and *constant bandwidth* (CBW) channels. In the PBW channels, the peak frequency deviation of the subcarrier is proportional to the subcarrier frequency, whereas in the CBW channels the deviation is a constant. The standards are given in Table 1 and 2 with deviation allowed.

Table 1a – IRIG Standard PBW Telemetry Channels

Channel	Data cut-off frequency Hz	7.5% Deviation channel		
		LBEF	CF (kHz)	HBEF
1	6	0.370	0.400	0.430
2	8	0.518	0.560	0.602
3	11	0.675	0.730	0.785
4	14	0.888	0.960	1.032
5	20	1.202	1.300	1.398
6	25	1.572	1.700	1.828
7	35	2.127	2.300	2.473
8	45	2.775	3.000	3.225
9	59	3.605	3.900	4.193
10	81	4.995	5.400	5.805
11	110	6.799	7.350	7.901
12	160	9.712	10.500	11.288
13	220	13.412	14.500	15.588
14	330	20.350	22.000	23.650
15	450	27.750	30.000	32.250
16	600	37.000	40.000	43.000
17	790	48.562	52.500	56.438
18	1050	64.750	70.000	75.250
19	1395	86.025	93.000	99.975
20	1860	114.700	124.000	133.300
21	2475	152.625	165.000	177.375

Table 1b – IRIG Standard PBW Telemetry Channels

Channel	Data cut-off frequency Hz	15% Deviation channel		
		LBEF	CF (kHz)	HBEF
A	660	18.700	22.000	25.300
B	900	25.500	30.000	34.500
C	1200	34.000	40.000	46.000
D	1575	44.625	52.500	60.375
E	2100	59.500	70.000	80.500
F	2790	79.050	93.000	106.950
G	3720	105.400	124.000	142.000
H	4950	140.250	165.000	189.750

LBEF — Low Band Edge Frequency
HBEF — High Band Edge Frequency
CF — Centre Frequency

Table 2 – IRIG Standard CBW Telemetry Channels

Deviation Nominal Channel	2kHz 0.4 kHz CF (kHz)	Deviation Nominal Channel	4 kHz 0.8 kHz CF (kHz)	Deviation Nominal Channel	8 kHz 1.6 kHz CF (kHz)
1A	16				
2A	24				
3A	32	3B	32	3C	32
4A	40				
5A	48	5B	48		
6A	56				
7A	64	7B	64	7C	64
8A	72				
9A	80	9B	80		
10A	88				
11A	96	11B	96	11C	96
12A	104				
13A	112	13B	112		
14A	120				
15A	128	15B	128	15C	128
16A	136				
17A	144	17B	144		
18A	152				
19A	160	19B	160	19C	160
20A	168				
21A	176	21B	176		

b) Importance of M2M system in modern society:

Machine to machine (M2M) refers to technologies that allow both wireless and wired systems to communicate with other devices of the same type. M2M is a broad term as it does not pinpoint specific wireless or wired networking and communications technology. This broad term is particularly useful for business executives.

M2M can include the case of industrial instrumentation - where a *device* (such as a sensor or meter) to capture an *event* (such as temperature, inventory level, etc.), which is relayed through a *network* (wireless, wired or hybrid) to an *application* (software program), that translates the captured event into *meaningful information* (for example, items need to be restocked). Such communication was originally accomplished by having a remote network of machines relay information back to a central hub for analysis, which would then be rerouted into a system like a personal computer.

However, modern M2M communication has expanded beyond a one-to-one connection and changed into a system of networks that transmits data to personal appliances. The expansion of IP networks across the world has made it far easier for M2M communication to take place and has lessened the amount of power and time necessary

for information to be communicated between machines. These networks also allow an array of new business opportunities and connections between consumers and producers in terms of the products being sold.
M2M was originally used for automation and instrumentation but now has been also used in various telemetric applications.

c) SCO:
The conditioned transducer outputs are normally used to frequency-modulate a subcarrier. The varying direct or alternating current the frequency of an oscillator operating at the carrier frequency. Such a circuit is generally referred to as a voltage-controlled oscillator (VCO) or a sub-carrier oscillator (SCO). To produce FDM, each VCO operates at a different center or carrier frequency. The outputs of the subcarrier oscillators are added.
A diagram of such a system is shown in figure below.

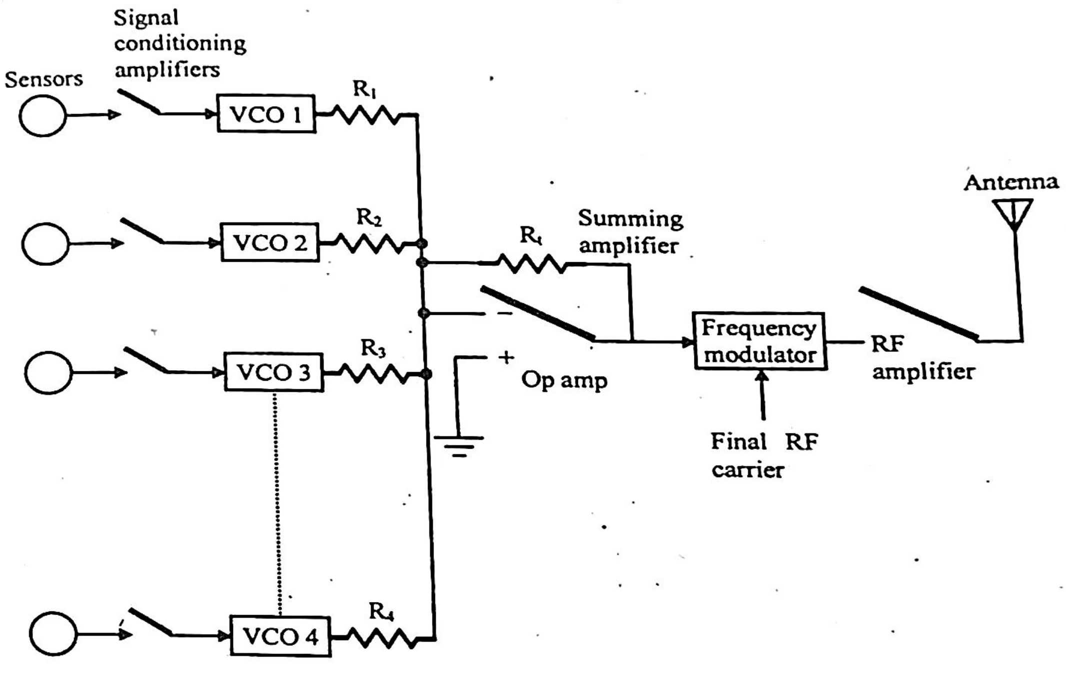


Fig: FDM telemetry transitting system

d) Quantization:
Quantization is an integral part of an analogue-to-digital converter and is done by the AD conversion technique using either the mid-tread form or the mid-rise form as shown in Figs. 1 (a) and (b), respectively.
The curves in Figs. 1 (a) and (b) actually show the input-output characteristics with the straight-line showing the linear relationship, while the steps represent quantization levels. The discrete (step) change produces what is known as quantization error, which is a function of the input voltage. For symmetric case, the steps are equally divided, $\pm\frac{\Delta V}{2}$.

The quantization error is the difference between level of quantization, say V_{Qk}, and analogue input V_{ik}. The maximum difference can be $\pm\frac{\Delta}{2}$ or $-\frac{\Delta}{2}$ so that the error can lie between $\pm\frac{\Delta V}{2}$ and is known as the mean square quantization error.

$$E_{mq}^2 = \frac{(\Delta V)^2}{12}$$

which also is the mean square quantization noise voltage. If there are n_L number of levels, the mean square signal voltage is $E_{m\cdot sig}^2 = \frac{(n_L \Delta V)^2}{12}$

giving the S/N quantization noise ratio as $(S/N)_q = \frac{E_{m\cdot sig}^2}{E_{mg}^2} = n_L^2$

supporting the idea that levels should be as high in number as possible.
If bits per code is n, then $n_L = 2^n$, so that $(S/N)_q = 2^{2n}$

If one has $\frac{E_{peak}}{E_{rms}} = \frac{1}{\rho}$, for low distortion the maximum peak signal level should be less than or equal to half the total input voltage range $\left(n_L \frac{\Delta V}{2}\right)$, i.e., $E_{peak} \le \frac{n_L \Delta V}{2}$

so that $(S/N)_q = \frac{E_{rms}^2}{E_{mq}^2} = \left(\frac{\rho n_L \Delta V}{2}\right)^2 \times \frac{12}{(\Delta V)^2} = 3\rho^2 n_L^2$

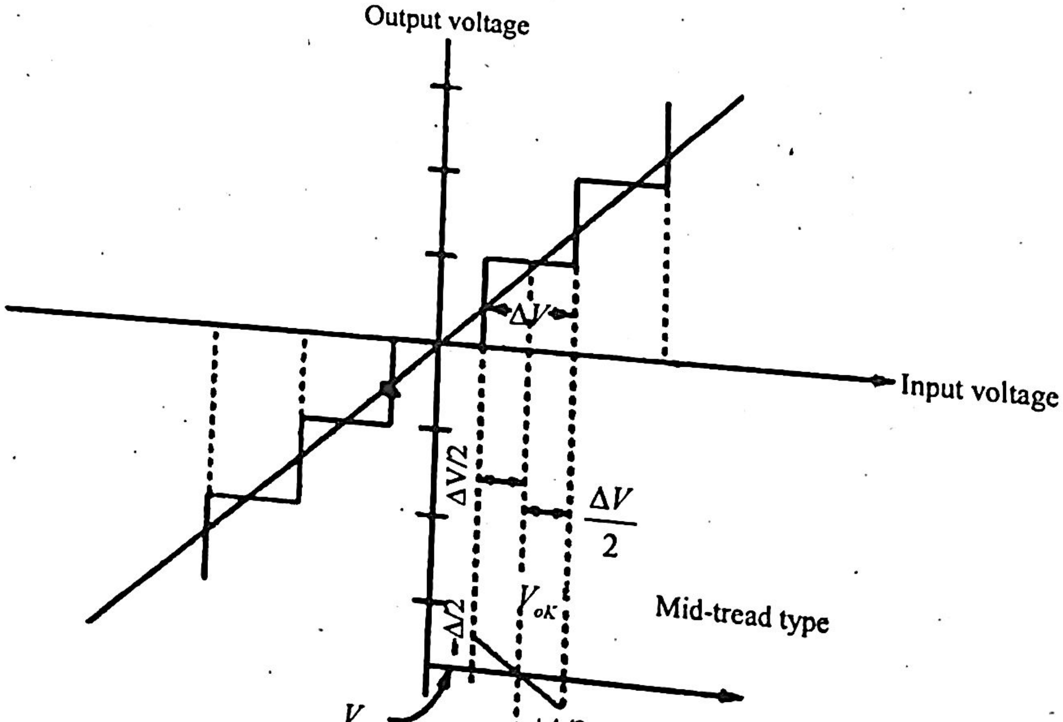


Fig: 1(a) Mid-tread AD conversion technique

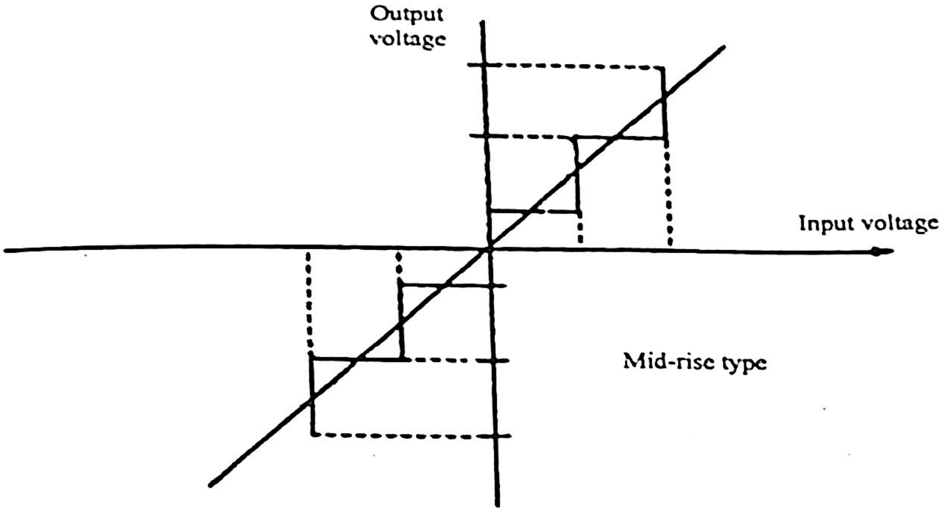
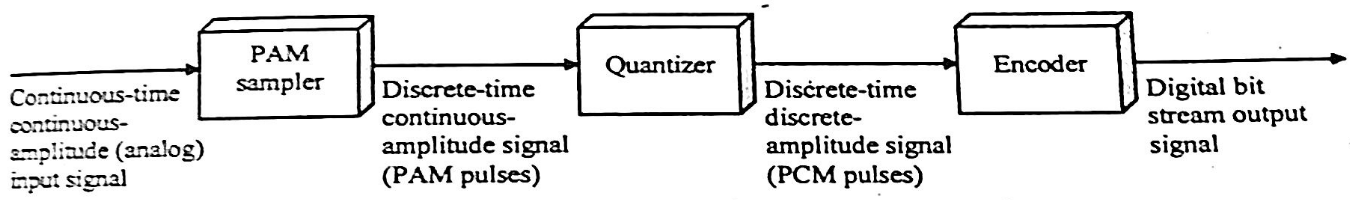


Fig: 1(b) Mid-rise AD conversion technique

e) PCM:

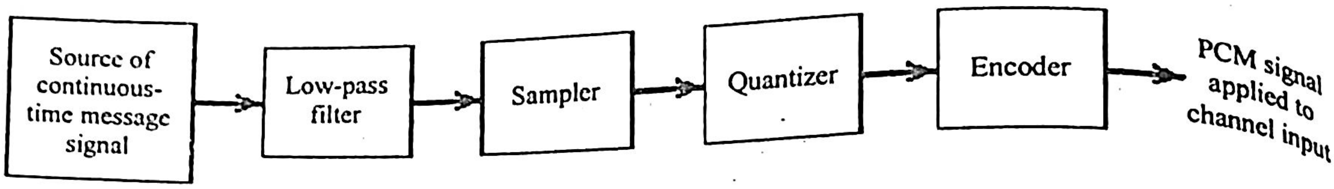
Pulse code modulation (PCM) is a digital scheme for transmitting analog data. The signals in PCM are binary; that is, there are only two possible states, represented by logic 1 (high) and logic 0 (low). It samples of the amplitude of the analog signal at regular intervals. The sampled analog data is changed to binary data. PCM requires a very accurate clock. The number of samples per second, ranging from 8,000 to 192,000, is usually several times the maximum frequency of the analog waveform in Hertz (Hz), or cycles per second, which ranges from 8 to 192 kHz.

The functional block diagram of the PCM is shown below:

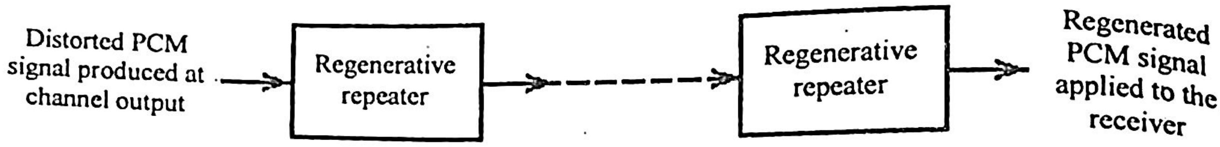


The Pulse Code Modulation process is done in three steps Sampling, Quantization, and Coding. There are two specific types of pulse code modulations such as differential pulse code modulation (DPCM) and adaptive differential pulse code modulation(ADPCM)

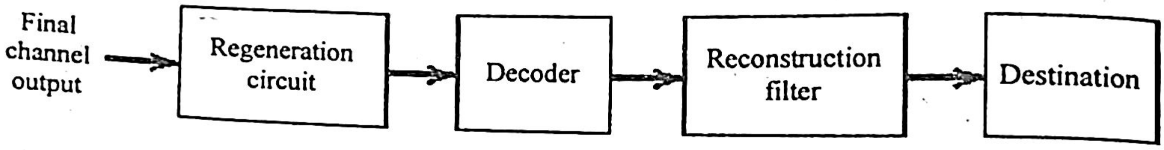
The basic elements in a PCM mechanism are (i) Transmitter, (ii) Transmission path, and (iii) Receiver as shown in the following figure.



(a) Transmitter



(b) Transmission path



(c) Receiver

To get a pulse code modulated waveform from an analog waveform at the transmitter end (source) of a communications circuit, the amplitude of the analog signal samples at regular time intervals. The sampling rate or number of samples per second is several times the maximum frequency. The message signal converted into binary form will be usually in the number of levels, which is always to a power of 2. This process is called quantization.

At the receiver end, a pulse code demodulator decodes the binary signal back into modular replicated pulses with same quantum levels. By further processes, we can restore the original analog waveform.

f) Sample and Hold circuit:
Refer to Question No. 3 of Short Answer Type Questions.

g) PLL:
Refer to Question No. 2 of Short Answer Type Questions.