Principles of Communication

ASSIGNMENT No. 4

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CLASS : D10A

ROLL NO : 62

PCOM Assignment 4

Question 1.

State and prove Sampling Theorem for low pass band limited signal.

Answer :

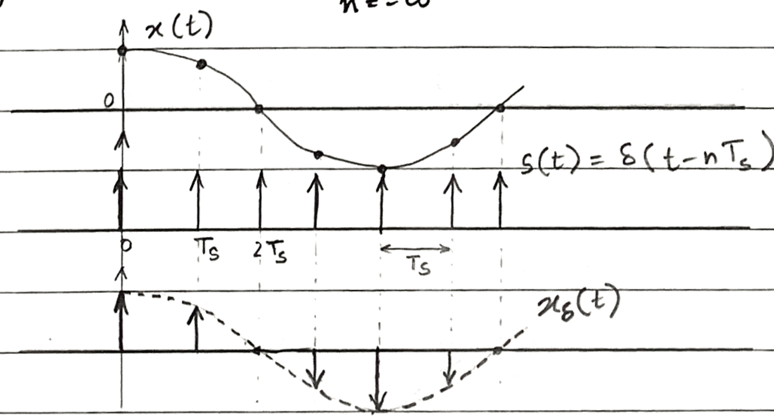
Sampling Theorem states that - "If a finite energy signal $x(t)$ contains no frequencies higher than 'W' Hz (i.e. it is a band limiting signal) then it is completely determined by specifying its values at the instants of time which are spaced $(1/2W)$ seconds apart." If a finite energy signal $x(t)$ contains no frequency components higher than 'W' Hz then it may be completely recovered from its samples which are space $(1/2W)$ seconds apart.

Proof :

Step 1 : Represent the sampling function $s(t)$ mathematically

The spacing between the adjacent unit impulses is T_s seconds, therefore the frequency of the sampling function is equal to the sampling frequency f_s. The sampled signal is denoted by $x_\delta(t)$

$\therefore$ Sample function $s(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$



Step 2: Represent the sampled signal $x_\delta(t)$ mathematically:

$x_\delta(t)$ is obtained by multiplying $x(t)$ and $s(t)$

$$\therefore \quad x_\delta(t) = x(t) \times s(t) = x(nT_s) \times s(t)$$

$$\therefore \quad x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \cdot \delta(t - nT_s)$$

Step 3: Obtain Fourier transform of sampled signal

$$\therefore \quad S(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)$$

$$\& \quad X(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$$

$$\therefore \quad X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f) * \delta(f - nf_s)$$

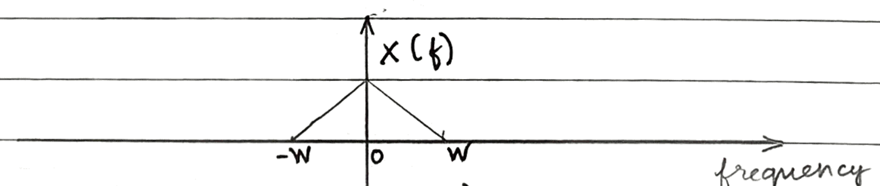
$$\because \quad \text{conv}\,(X(f), \delta(f - nf_s)) = X(f - nf_s)$$

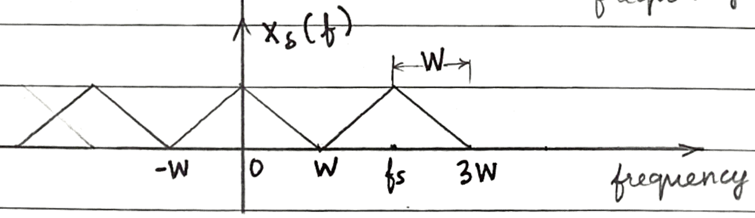
$$\therefore \quad X_\delta(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$$

Step 4: Sampled signal completely represents $x(t)$

$X(f - nf_s)$ represents the shifted version of $X(f)$ of original signal $x(t)$.

$$\therefore \quad X(f - nf_s) = X(f) \quad \text{at } f = 0, \pm f_s, \pm 2f_s$$





As seen above the spectrum of sampled signal $x_\delta(t)$ has frequency $f_s = 2W_s$. Thus sampling theorem is proved.

Question 2.

What do you mean by aliasing? How can it be avoided?

Answer:

Aliasing is the phenomenon of a high frequency in the spectrum of the original signal $x(t)$, taking on the identity of lower frequency in the spectrum of the sampled signal $x_\delta(t)$.

Aliasing can be avoided by doing the following:

i) Use a band-limiting low pass filter and pass the signal $x(t)$ through it before sampling. This filter is also called anti-aliasing filter or prealias filter.

ii) Increase the sampling frequency f_s greater than $2W$

Question 3.

The signal $v(t) = \cos 5\pi t + 0.5 \cos 10\pi t$ is instantaneously sampled. Calculate the Nyquist rate and Nyquist interval.

Answer:

$$f_1 = \frac{5\pi}{2\pi} = 2.5 \text{ Hz}$$

$$f_2 = \frac{10\pi}{2\pi} = 5 \text{ Hz}$$

$$\therefore \text{Nyquist rate} = 2f_2 = 10 \text{ Hz}$$

$\therefore$ Nyquist interval $= \frac{1}{2f_2} = \frac{1}{10} = 0.1$ seconds

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Question 4.

A bandpass signal has a spectral range that extends from 30 KHz to 75 KHz. Calculate the sampling frequency f_s.

Answer:

Bandwidth $= f_2 - f_1 = 75 - 30 = 45$ KHz

$\therefore$ sampling frequency $f_s = \frac{2 f_{max}}{B} = \frac{2 \times 75}{45}$

$\therefore f_s = 3.333$ kHz

Question 5.

Differentiate between analog and digital pulse modulation techniques.

Answer:

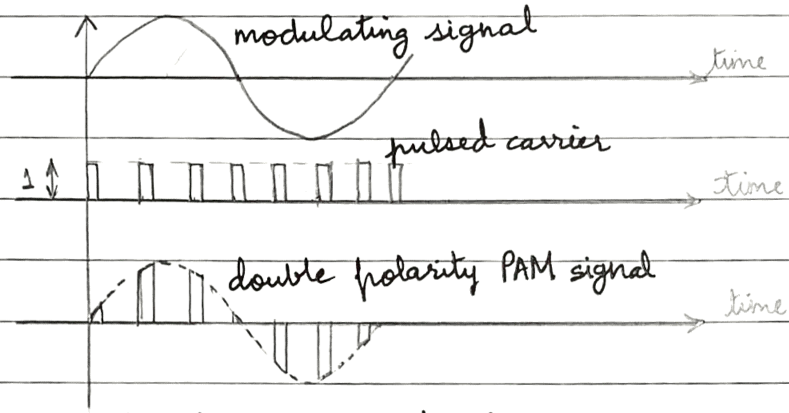
Analog pulse modulation	Digital pulse modulation
i) Amplitude, width or position of pulsed carrier is varied with message signal.	i) Message is sampled, quantized and encoded into codes.
ii) Output is a train of analog	ii) Output is a train of digital bits

pulses.	that form the codewords
iii) It is simple and less expensive	iii) It is complex and expensive.
iv) PAM shows bad whereas PWM & PPM show good performance in presence of noise	iv) All systems show excellent performance in presence of noise.
v) Repeaters cannot be used	v) Repeaters can be used.

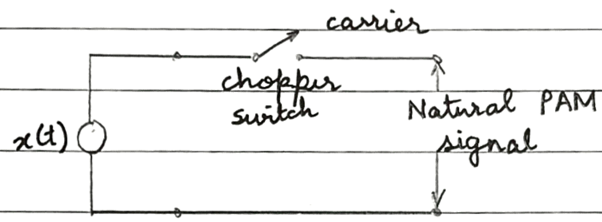
Question 6.
Explain with neat diagram and waveforms generation and detection of natural and flat top PAM.

Answer:
In PAM system, the amplitude of the pulsed carrier is changed in proportion with the instantaneous amplitude of the modulating signal $x(t)$.

I] Natural PAM
The continuous modulating signal $x(t)$ is passed through a low pass filter. It will bandlimit this signal to f_m. The pulse generator generates a pulse train at a frequency f_s such that $f_s \geq 2f_m$ (Nyquist criteria). Sampling takes place in the multiplier, to generate the PAM signal.

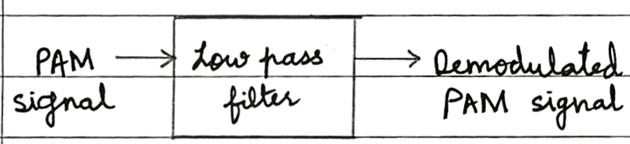


Waveform of natural PAM

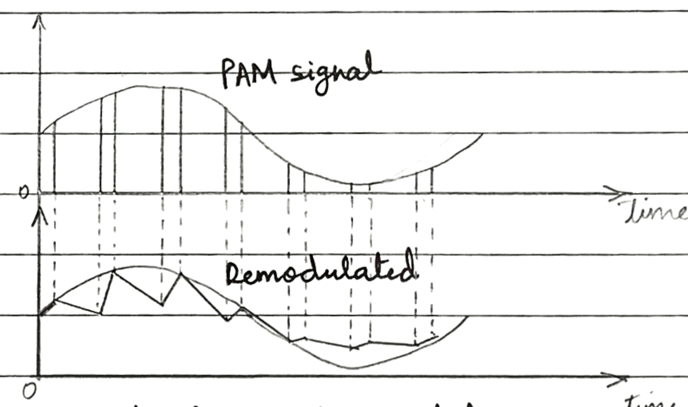


circuit arrangement for natural PAM

This PAM signal can be detected by passing it through a low pass filter.



PAM detector

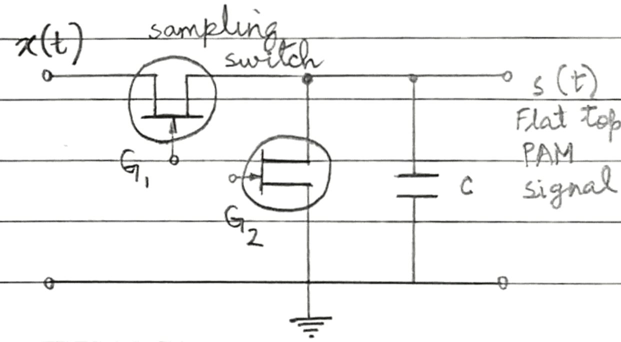


Waveform of detected signal

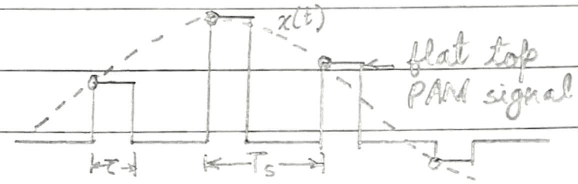
II] Flat Top PAM

The sample and hold circuit consists of two FET switches and a capacitor. The analog signal $x(t)$ is applied at the input of this circuit and the flat topped PAM signal $s(t)$ is obtained across the capacitor.

The flat top PAM can be obtained by using the sample and hold circuit as shown below :



sample and hold circuit



Flat top PAM signal

Reconstruction of original signal x(t) :

Due to the aperture effect discussed, an amplitude distortion as well as a delay is introduced in the flat top sampled signal. This distortion can be corrected by connecting an equalizer after the reconstruction filter (low pass filter).

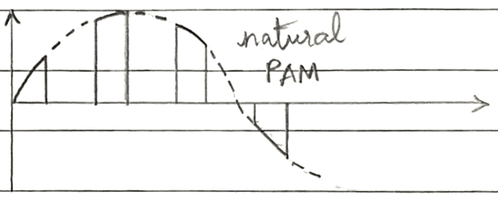
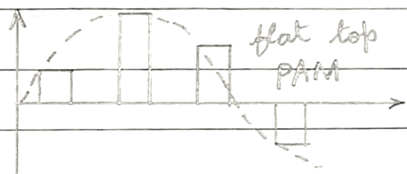
Flat top PAM signal → | low pass filter | → | Equalizer | → Analog signal x(t)

Question 7.

Compare natural PAM and flat top PAM

Answer :

Natural PAM	Flat Top PAM
i) The top of the samples preserve the shape of the	i) The top of the samples remains constant and equal to the

waveform amplitude.	instantaneous value of the modulating signal at the start of the sampling.
ii) 	ii) 
iii) SNR is worse than that of flat top PAM signal.	iii) Better SNR due to increased signal power.
iv) Reconstruction of original signal requires only a low pass filter	iv) Reconstruction of original signal requires low pass filter as well as an equalizer.
v) Very little distortion.	v) Aperture effect introduces distortion.

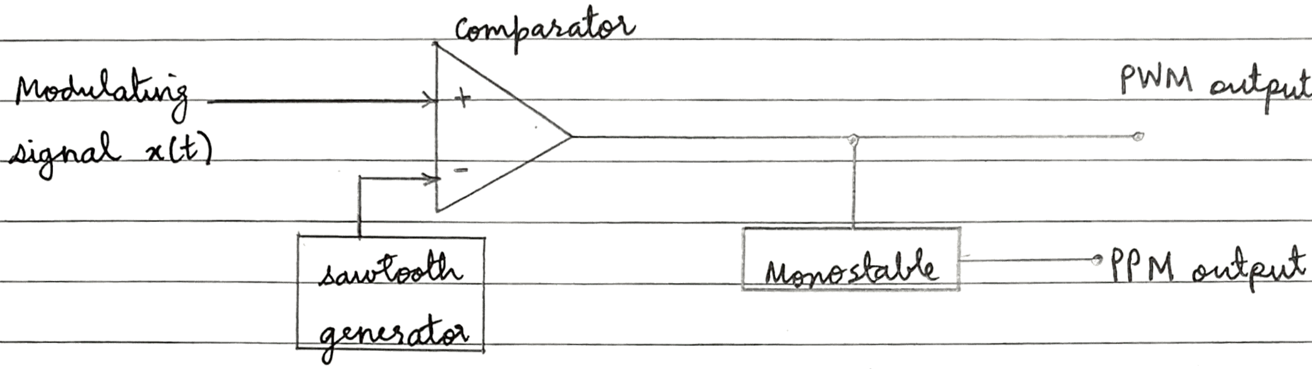
Question 8.

Draw the block diagram of PWM generator and detector. Explain the working giving waveforms at the output of each block.

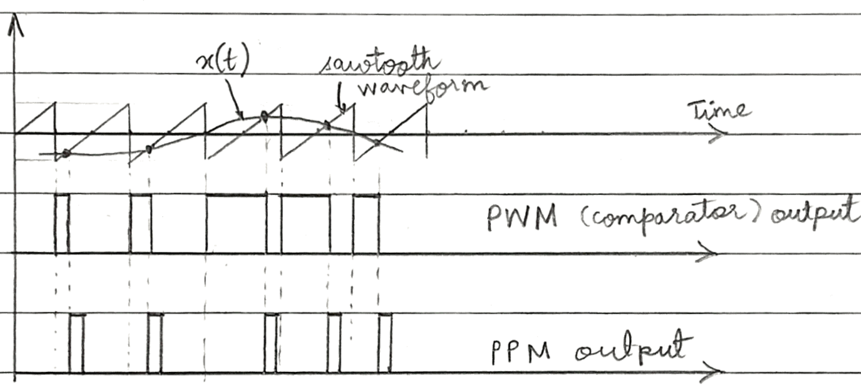
Answer:

Generation of PWM signal:

A sawtooth generates a sawtooth signal of frequency f_s. It is applied to the inverting terminal of a comparator. The modulating signal $x(t)$ is applied to the non-inverting terminal of the same comparator. The comparator output will remain high as long as the instantaneous amplitude of $x(t)$ is higher than that of the ramp signal. This gives rise to the PWM signal at the comparator output.



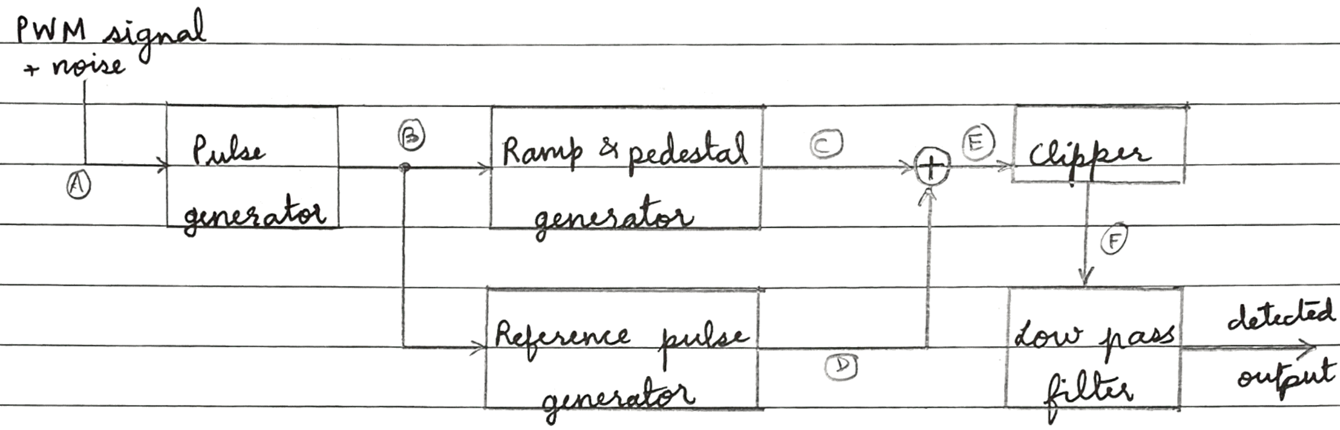
Block diagram for generation of PWM & PPM



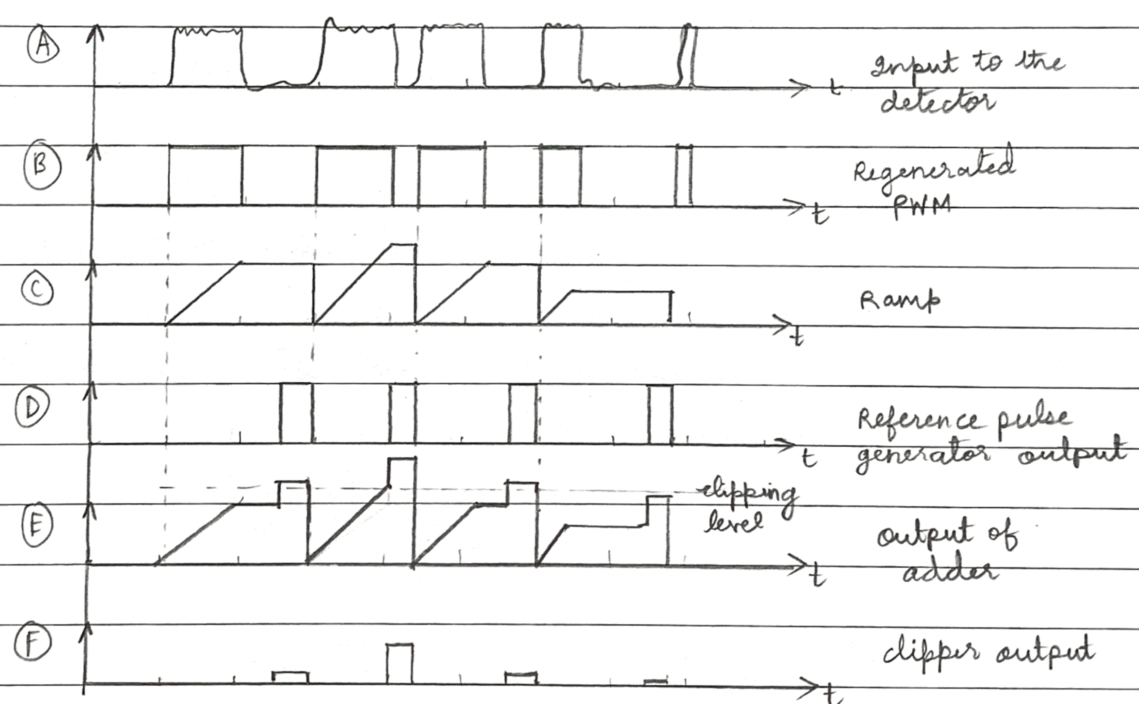
Waveforms of generation of PWM and PPM

Detection of PWM Signal:

The PWM signal received at input of detection circuit contains noise, hence it is first passed through pulse generator, to remove noise. This is applied to a reference pulse generator which produces a train of constant amplitude, constant width pulses. The regenerated PWM pulses are also applied to a ramp generator, the output of which is a constant slope ramp for the duration of pulse. At the end of the pulse a sample and hold amplifier retains the final ramp voltage. The output pulses of reference pulse generator are then added to the ramp signal. The output of adder is then clipped off by a clipper. A low pass filter is used to recover the original modulating signal back from the PAM signal.



PWM detection circuit



Waveforms for PWM detection circuit

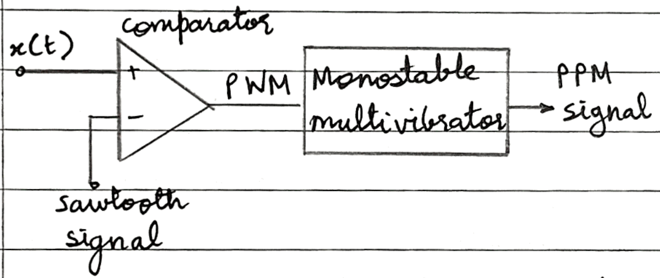
Question 9.

With the help of neat block diagram and waveforms demonstrate the working of PPM generation and degeneration.

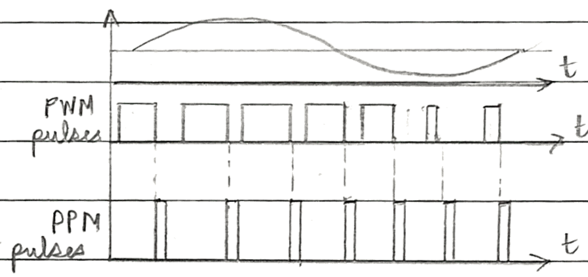
Answer:

Generation of PPM signal:

The PWM pulses obtained at comparator output are applied to a monostable multivibrator. The monostable is negative edge triggered. Corresponding to each trailing edge of PWM signal, the output goes high for a fixed time.



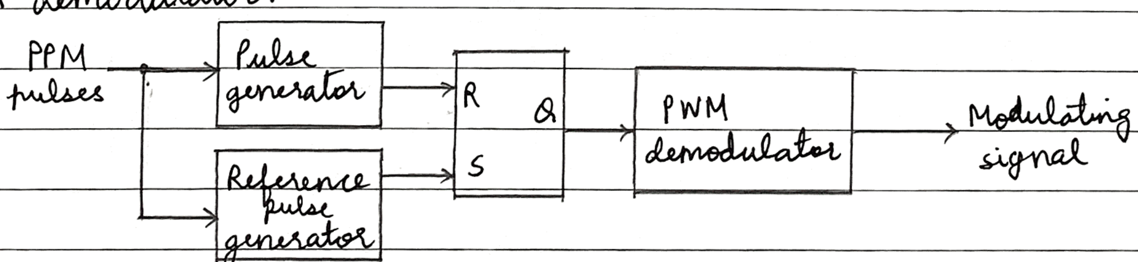
Block diagram for PPM generation



PPM pulses waveform

Demodulation of PPM signal:

Noise corrupted PPM waveform is fixed by the pulse generator and reference pulse generator which output the respective pulses at SR flip-flop. Due to the set and reset signals applied to the flip-f we get a PWM signal at its output, which is demodulated using the PWM demodulator.

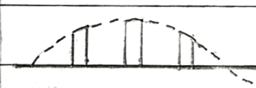
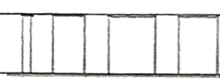
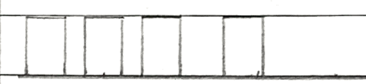


PPM demodulator circuit

Question 10.

Compare PAM, PWM and PPM.

Answer:

	Parameter	PAM	PWM	PPM
i)	Type of carrier	Train of pulses	Train of pulses	Train of pulses
ii)	Variable characteristic of pulsed carrier	Amplitude	Width	Position
iii)	Bandwidth requirement	Low	High	High
iv)	Noise immunity	Low	High	High
v)	Need to transmit synchronized pulses	Not needed	Not needed	Necessary
vi)	Complexity of generation & detection	Complex	Easy	Complex
vii)	Output waveforms			

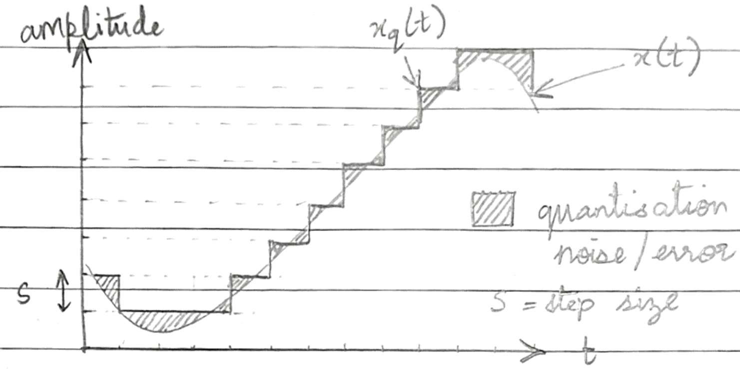
Question 11.

Describe quantisation noise or quantisation error.

Answer:

The difference between the instantaneous values of the quantised signal and input signal is called as quantisation error or quantisation noise.

$$\epsilon = x_q(t) - x(t)$$



Process of quantisation & quantisation error

Question 12.

What is companding? Explain. Show how companding reduces the quantization error.

Answer:

Companding is the process of compression and then expansion of the signal. With companded system, the higher amplitude analog signals are compressed prior to transmission and then expanded in the receiver. Companding is non-uniform quantisation. Due to the inverse nature of compressor and expander characteristics, the compander is a straight line. This indicates that all boosted signals are brought back to their original amplitudes.

$$\text{Quantization error } N_q = \frac{S^2}{12}$$

Thus by boosting weak signals, the SNR of quantisation noise is better.

Question 13.

Write A-law and μ-law companding.

Answer:

In the μ-law companding, the compressor characteristic is continuous. It is approximately linear for smaller values of input levels and logarithmic for high levels of input signal.

Mathematically,

$$z(x) = (\operatorname{sgn} x) \frac{\ln(1 + \mu |x| / x_{max})}{\ln(1+\mu)}$$

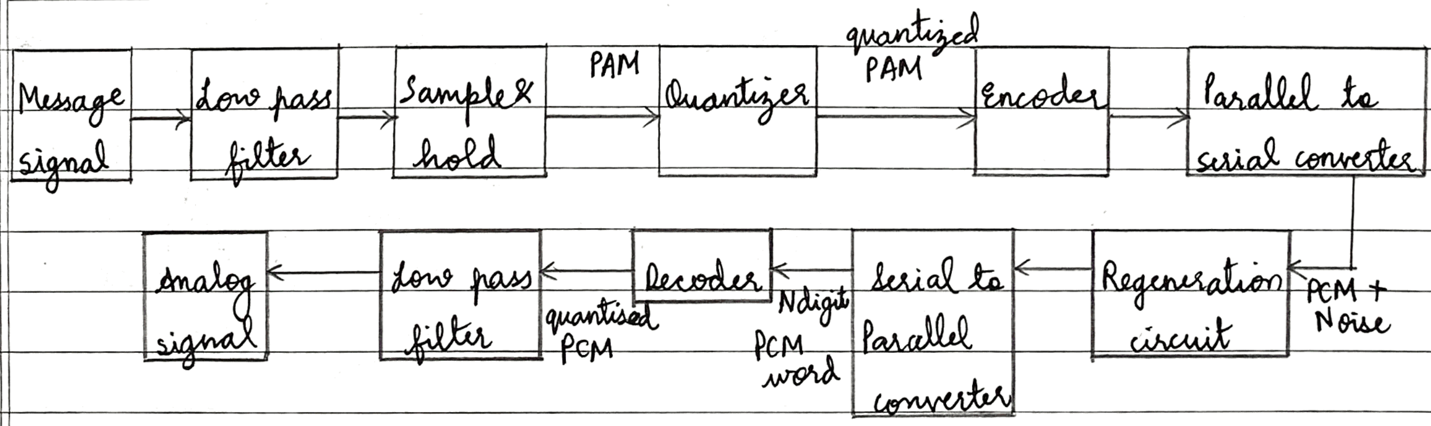
$$\text{where } 0 \leq |x| / x_{max} \leq 1$$

In the A-law companding, the compressor characteristic is of piecewise nature, made up of a linear segment for low level inputs and a logarithmic segment for high level inputs.

Question 14.

Draw the block diagram of a PCM system and explain the function of each block.

Answer:



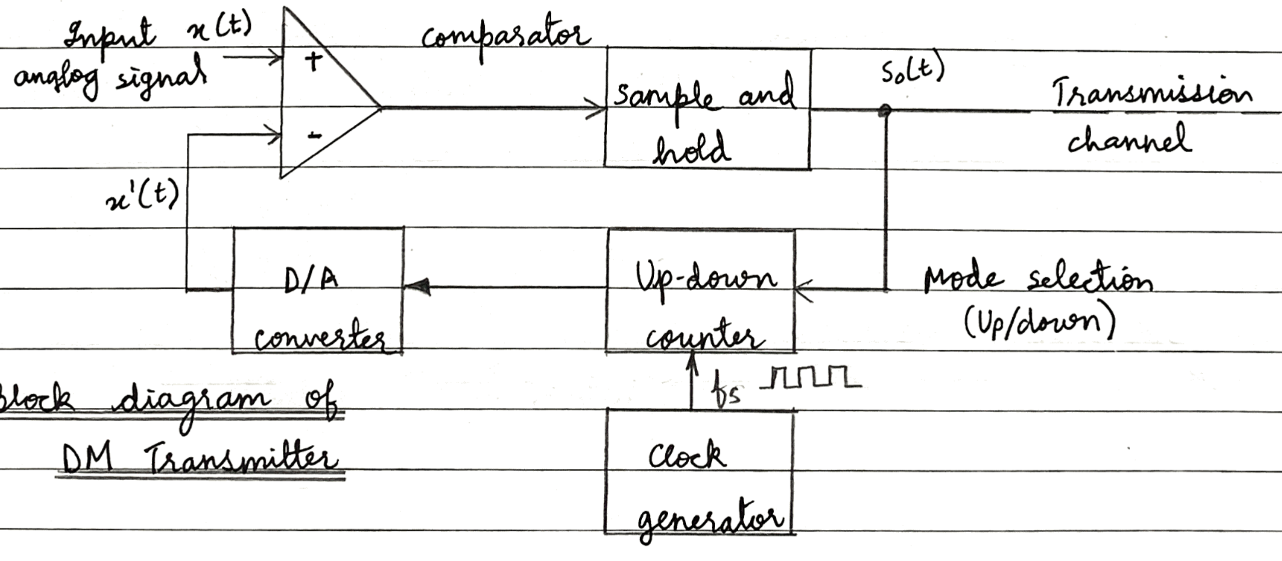
The analog signal $x(t)$ is passed through low pass filter having a cut-off frequency $f_c = W$ Hz. This is then passed through sample and hold circuit where it is sampled at high sampling rate. These samples are then subjected to the quantization process in the quantizer. The quantized PAM pulses are applied to an encoder. The regeneration circuit is a part of the receiver and it seperates the PCM pulses from noise. This clean PCM signal is passed through a serial to parallel converter, which is then applied to a decoder, which is a D to A converter. The quantized PAM obtained from decoder is passed through a low passed filter to recover the analog signal $x(t)$.

Question 15.

Explain the working of Delta Modulator Transmitter and Receiver with neat block diagrams.

Answer:

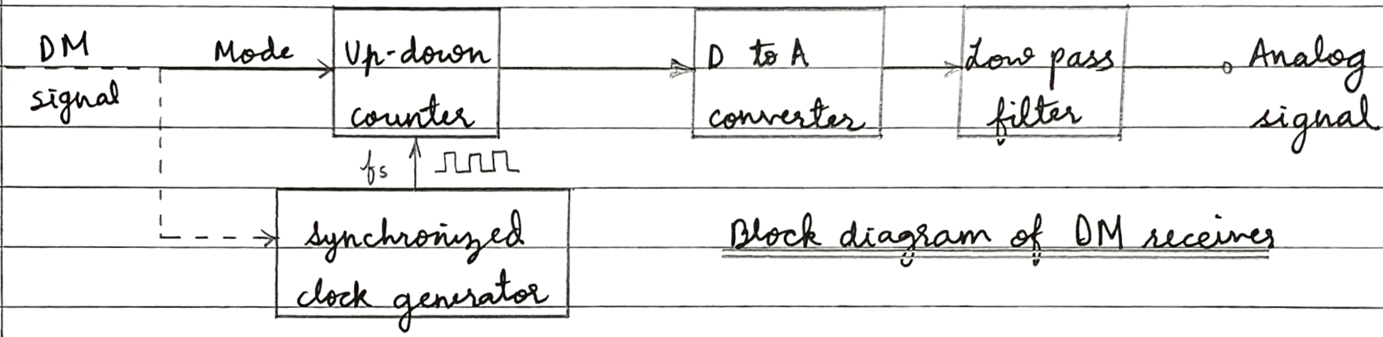
Delta Modulator Transmitter :



Block diagram of DM Transmitter

Operation : $x(t)$ and its quantized version $x'(t)$ are applied to the comparator, which results in either 1 or 0. The sample and hold circuit will hold this level for the entire clock cycle period, and its output is transmitted as the output of the DM transmitter. The transmitted signal is also used to decide the mode of operation of an up/down counter.

Delta Modulator Receiver :



Block diagram of DM receiver

Operation :
The block diagram of a DM receiver is as shown above. On comparing it with the transmitter block, we find that it is identical to the chain of blocks producing signal $x'(t)$. The original modulating signal can be recovered by passing this signal through a low pass filter.

Question 16.
What are the limitations of linear Delta Modulator ? How they are overcome in Adaptive Delta Modulator ?

Answer :

Disadvantages of linear Delta Modulator:

i) The two distortions, namely slope overload error and granular noise, are present.

ii) Practically the signalling rate with no slope overload error will be much higher than that of PCM

During the working of the Adaptive Delta Modulation, In response to the K^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous $(K-1)^{th}$ clock edge. If the direction of both steps is same, the processor will increase magnitude, and decrease it otherwise. S_o i.e. output of the ADM is given as

$$S_o(t) = +1 \quad \text{if } x(t) > x'(t) \text{ before } K^{th} \text{ edge} \quad \&$$

$$S_o(t) = -1 \quad \text{if } x(t) < x'(t) \text{ before } K^{th} \text{ edge}$$

$\therefore$ step size at sampling instant K is given by

$$\delta(K) = \delta(K-1)\, S_o(K) + \delta\, S_o(K-1)$$

Let $K = 6$

$$\therefore \delta(6) = \delta + \delta = 2\delta$$

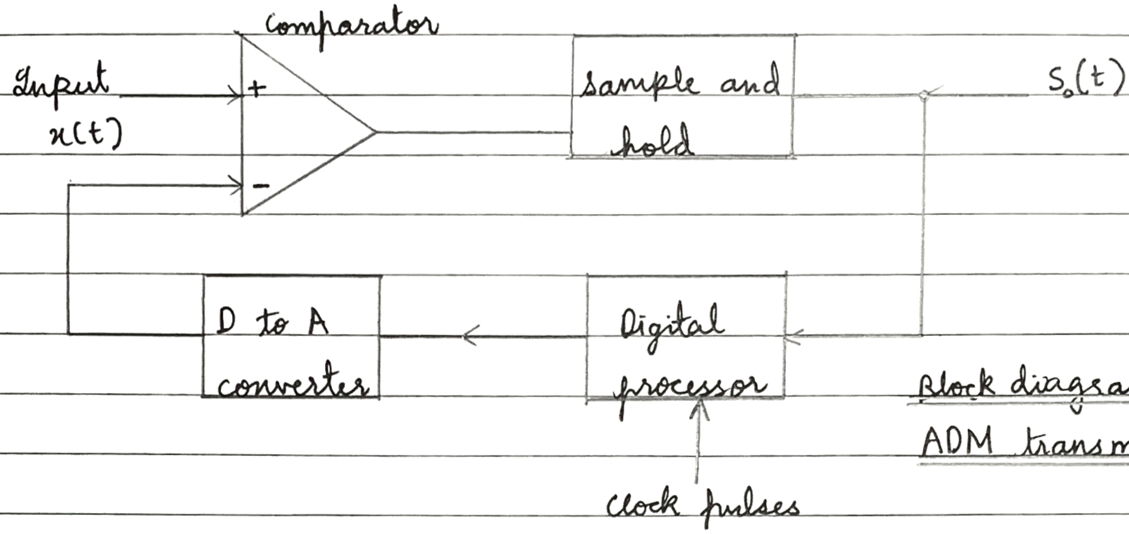
Thus due to the variable step size the slope overload error is reduced. Hence ADM system has a lower bit rate than PCM, & hence the BW required is also less.

Question 17.

Draw the block diagram of ADM Transmitter and Receiver and explain their working.

Answer:

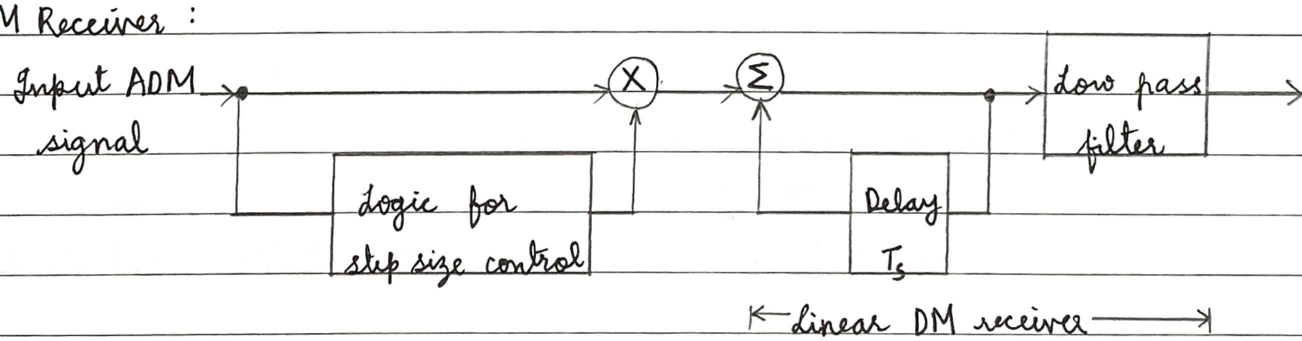
ADM Transmitter :



Block diagram of ADM transmitter

Operation: On comparing this diagram with that of delta modulator, we find that except for the counter replaced by digital processor, the remaining blocks are identical. In response to the k^{th} clock pulse trailing edge, the processor generates a step which is equal in magnitude to the step generated in response to the previous $(k-1)^{th}$ clock edge. If direction of both step is same then the processor will increase the magnitude of the present step by S.

ADM Receiver :



Block diagram of ADM receiver

Operation : The received ADM signal is applied to a digital process as well as the synchronised clock generator, which produces the clock pulses synchronized with those at the transmitter. The digital processor produces a digital signal in the parallel form which is converted into an analog signal by a D to A converter. The output of DAC is in the form of quantised PAM which is passed through a low pass filter to obtain the analog signal.

Question 18.

Compare PCM, DM and ADM.

Answer:

	Parameter	PCM	DM	ADM
i)	No. of bits per sample	N = 4, 8, 16, 32, etc.	N = 1	N = 1
ii)	Step size	Depends on no. of Q levels	Step size is fixed	Step size is variable
iii)	Distortions / errors	Quantization error	Slope overload & granular noise	Granular noise
iv)	Signalling rate & Bandwidth	Highest	Low, if input is slow varying	Lowest
v)	System complexity	Complex	Simple	Simple
vi)	Feedback from output	No feedback	Feedback present	Feedback present

Question 19.

State and explain different types of line codes

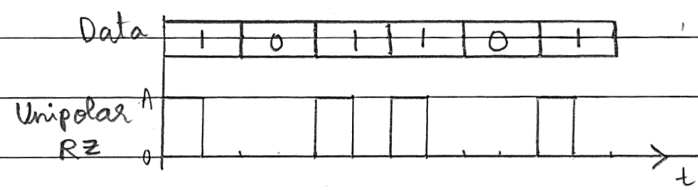
Answer:

Line codes are classified as following:

Line codes
- Unipolar
  - RZ
  - NRZ
- Polar
  - RZ
  - NRZ
  - Manchester
  - Differential manchester
- Bipolar
  - AMI
- Multilevel
  - 2B/1Q
  - 8B/6T
- Multitransition
  - MLT-3

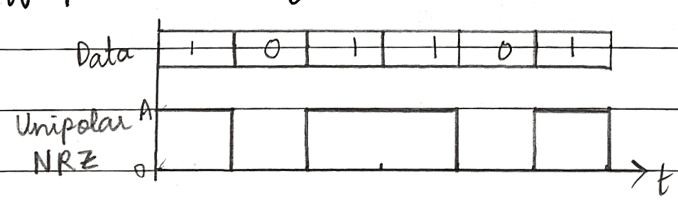
i) **Unipolar RZ format**

In this each "0" is represented by an off pulse (0) and each "1" by an on pulse with amplitude A and a duration of $T_b/2$ followed by a return to zero level.



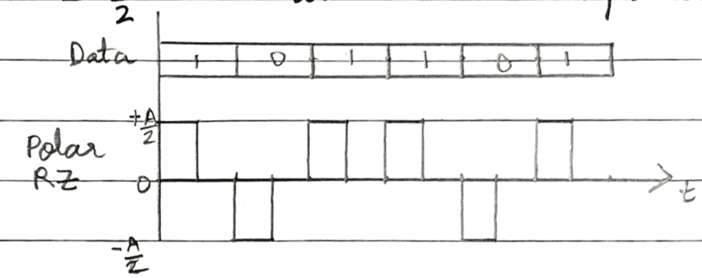
ii) **Unipolar NRZ format**

In this format a logic "1" is represented by a pulse of full bit duration T_b and amplitude A while a logic "0" is represented by an off pulse or zero amplitude.



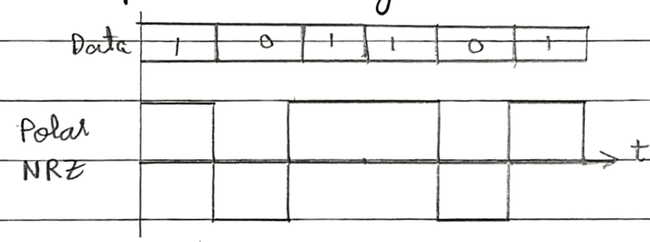
iii) **Polar RZ format**

In this format, the signal shows opposite polarity pulses of amplitude $\pm \frac{A}{2}$ that are used to represent logic "1" and "0".



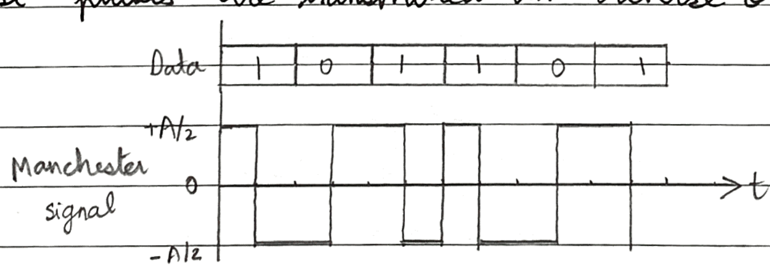
iv) <u>Polar NRZ format</u>

In this format, pulse of amplitude $+\frac{A}{2}$ of duration T_b is used to represent a logic "1" and a pulse of amplitude "$-\frac{A}{2}$" of the same duration represents a logic "0".



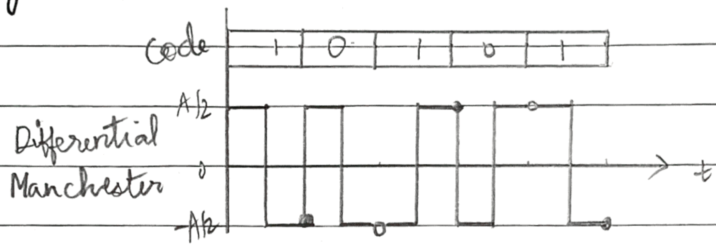
v) <u>Split Phase Manchester Format</u>

In this format, logic "1" is represented by transmitting a positive pulse of $+\frac{A}{2}$ for one half of the symbol duration, followed by a negative pulse of amplitude $-\frac{A}{2}$ for remaining duration. For logic "0" these pulses are transmitted in reverse order.



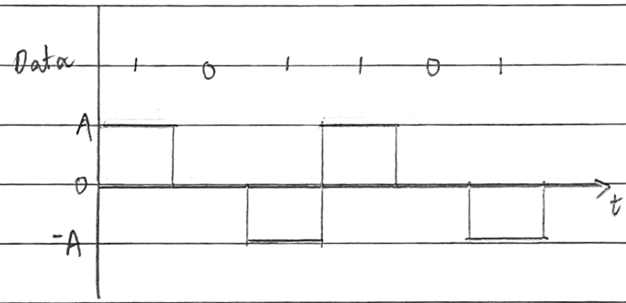
vi) <u>Differential Manchester Format</u>

In this code there is always a transition in the middle of a bit interval. The binary zero has an additional transition at the beginning of the bit interval.



vii) Bipolar NRZ format (AMI):

In this format the successive "1"s are represented by pulses which alternating polarity, and no pulse is transmitted for a logic "0".



Question 20.

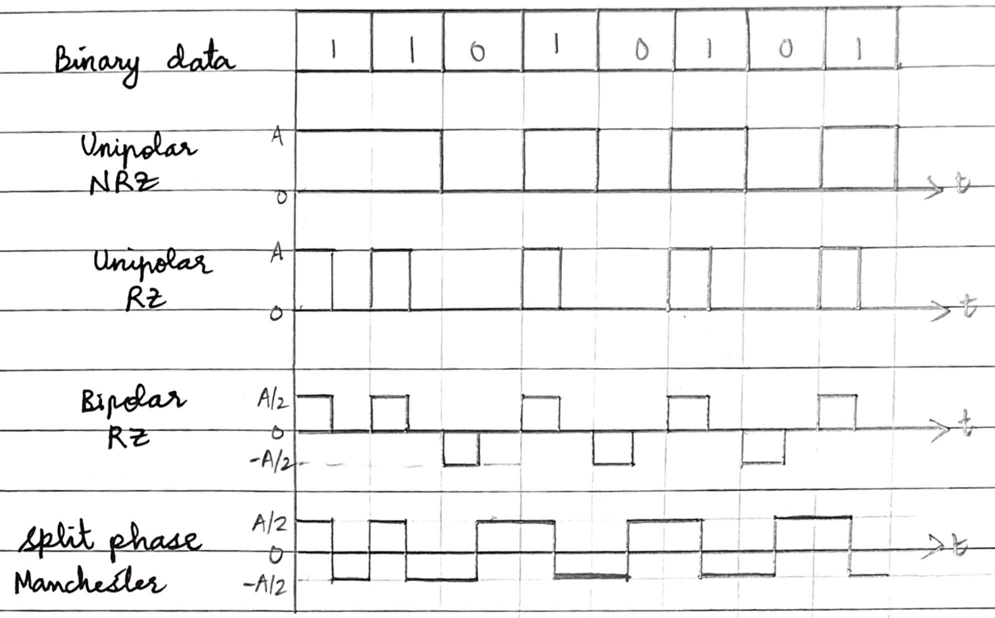
The binary data 11010101 is transmitted over a baseband channel. Draw the waveform for transmitted data using the following data formats : i) Unipolar NRZ

ii) Unipolar RZ

iii) Bipolar RZ

iv) Split phase Manchester

Answer:



Question 21.

What do you mean by ISI ? What are the remedies ?

Answer :

In communication system when data is being transmitted in the form of pulses (bits), the output produced at the receiver due to the other bits or symbols interferes with the output produced by the desired bit. This is called inter-symbol interference or ISI.

Remedies to reduce ISI :

Since it has been proved that sinc function produces zero ISI, we transmit a sinc pulse instead of a rectangular pulse. Using sinc pulse for transmission is known as Nyquist Pulse Shaping.