PRINCIPLES OF COMMUNICATIONS

ASSIGNMENT 04

Aamir Ansari
D10A 01

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PCOM assignment 4

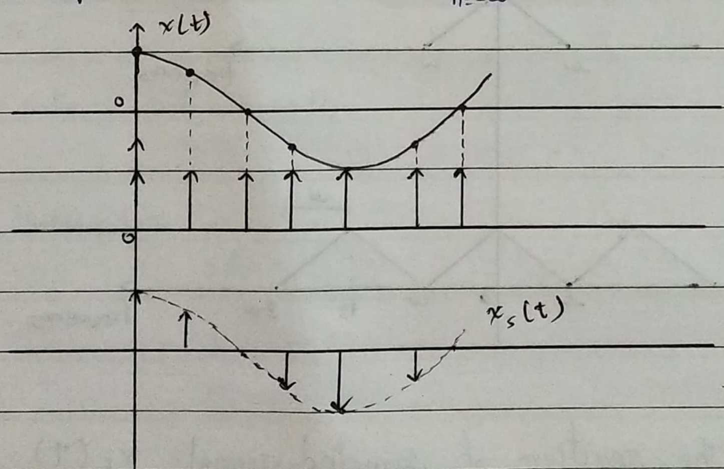
1) State and prove sampling theorem for low pass band limited signal

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

Proof:

Step1:- Represent the sampling function $s(t)$ mathematically the spacing between the adjacent unit impulses is T_s seconds, Therefore the frequency of sampling function is equal to the sampling frequency f_s. The sampled signal is denoted by $x_s(t)$

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



Step 2:- Represent the sampled signal $x_s(t)$ mathematically $x_s(t)$ is obtained by multiplying $x(t)$ and $s(t)$

$\therefore x_s(t) = x(t) \times s(t) = x(nT_s) \times s(t)$

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

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Step 3: Obtain fourier transform of sampled signal.

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

and $x(f) = f_0 \sum_{n=-\infty}^{\infty} \delta(f - nf_0)$

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

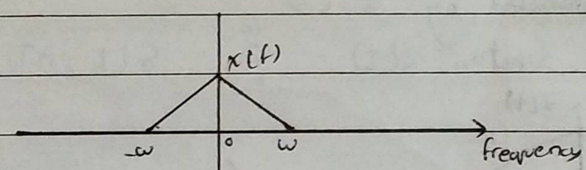
$\because \text{conv}(X(f)), \delta(f - nf_s) = X(f - nf_s)$

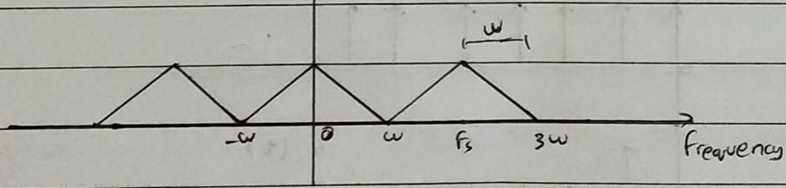
$\therefore X_s(f) = f_s \sum_{n=-\infty}^{\infty} X(f - nf_s)$

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

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

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





As seen above the spectrum of sampled signal $x_s(t)$ has frequency $f_s = 2W$. Thus sampling theorem is proved.

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

→ Aliasing is a phenomenon of a high frequency in the spectrum of original signal $x(t)$, taking on the identity of lower frequency in the spectrum of the sampled signal $x_s(t)$. 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 or prealias filter.

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

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

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

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

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

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

$= \frac{1}{10}$

$= \underline{\underline{0.1 \text{ seconds}}}$

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4) A Bandpass signal has a spectral range that extends from 30 kHz - 75 kHz. Calculate sampling frequency f_s.

→ Band width $= f_2 - f_1 = 75 - 30 = 45$ kHz

∴ Sampling frequency $f_s = \frac{2 f_{max}}{B} = \frac{2 \times 75}{45}$

∴ $f_s = 3.3333$ kHz

5) Differentiate between analog and digital pulse modulation techniques.

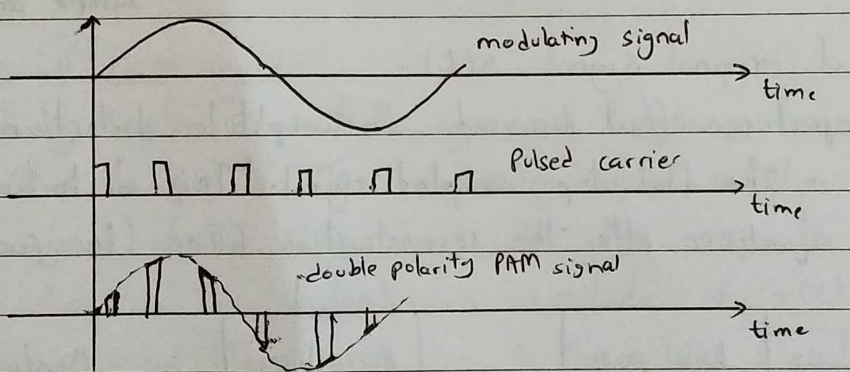
Analog pulse modulation	Digital pulse modulation
1) Amplitude, width or position of pulsed carrier is varied with message signal	1) Message is sampled, quantized and ecoded into codes
2) Output is a train of analog pulses	2) Output is train of digital bits that forms the codeword
3) It is simple and less expensive	3) It is complex and expensive
4) Repeaters can not be used	4) Repeaters can be used

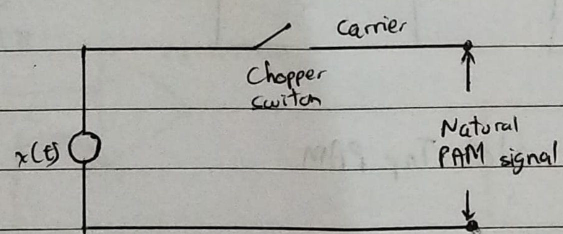
6) Explain with neat diagram and waveform generation and detection of of natural and flat top PAM

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

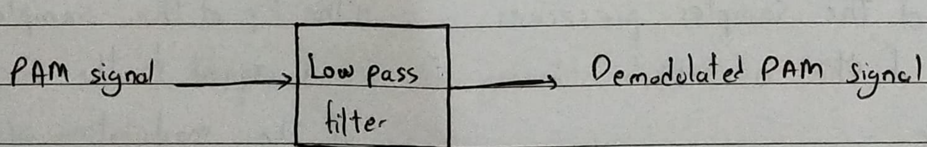
1) Natural PAM

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





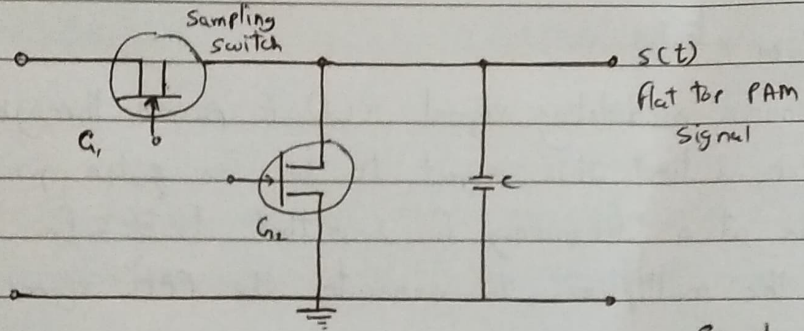
This PAM signal can be detected by passing it through a low pass filter.



PAM detector

Flat top PAM :-

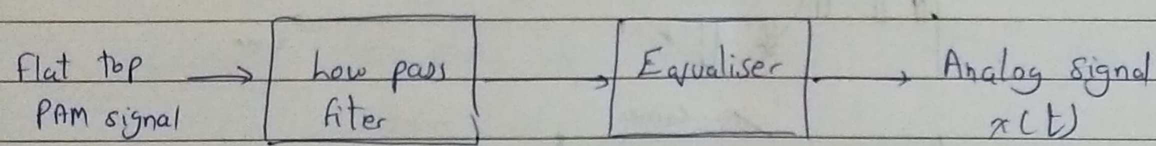
The sample and hold circuit consists of two FET switches and a capacitor
The analog signal $x(t)$ is applied at the input of this circuit and the flat topped PAM signal $s(t)$ is obtained accross the capacitor
The flat top PAM can be obtained by using the sample and hold circuit as shown below



Sample and hold circuit

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 an equalizer after the reconstruction filter (low pass filter)



7) Compare natural PAM and flat Top PAM

Natural PAM	FLAT Top PAM
1) The top of the samples preserves the shape of the waveform amplitude	1) The top of the sample remains constant and equal to the instantaneous value of the modulating signal at the start of the sampling

2) SNR is worse than that of flat top PAM signal | 2) Better SNR due to increased signal power

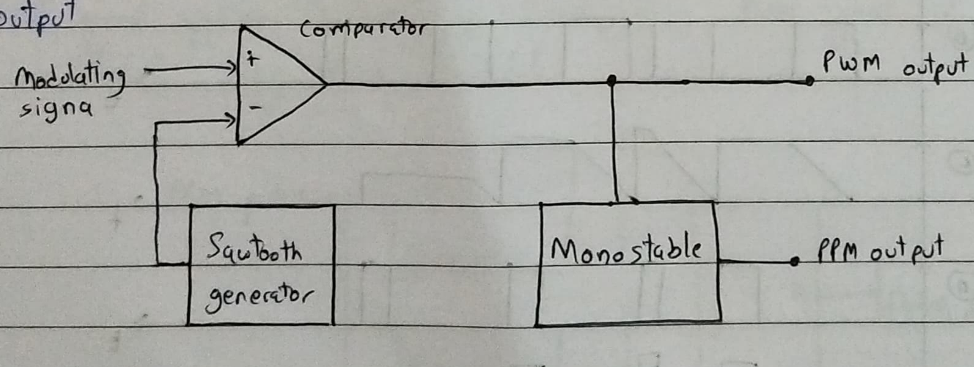
3) Reconstruction of original signal requires only a low pass filter | 3) Reconstruction of original signal requires lowpass filter and equiliser.

4) Very little distortion | 4) Aperture effect introduces distortion

8) Draw the block diagram of PWM signal generator and detector. Explain the working, giving waveform at the output of each block.

Generation of PWM signal

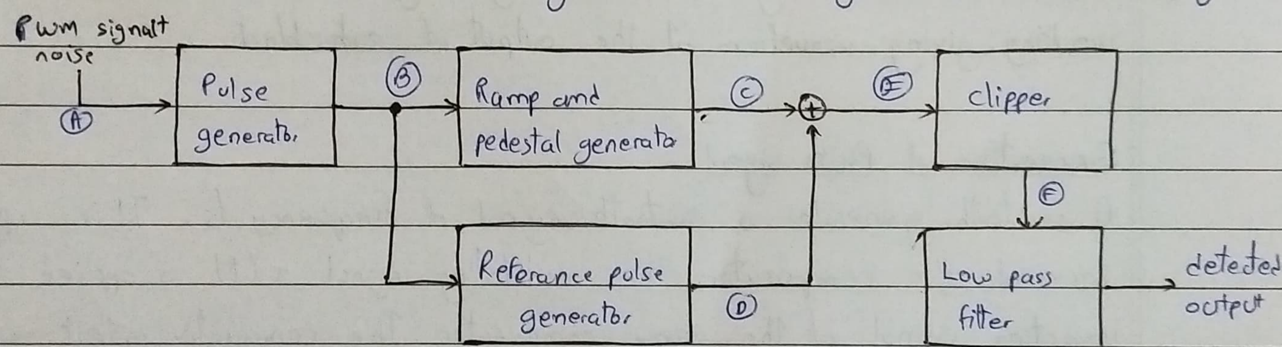
A sawtooth generates a sawtooth signal of frequency f_s. It is applied to inverting terminal of a comparator. The modulating signal $x(t)$ is applied to the non inverting signal of the same comparator. The comparator output will be remain haigh as long as the instantaneous amplitude of $x(t)$ is higher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

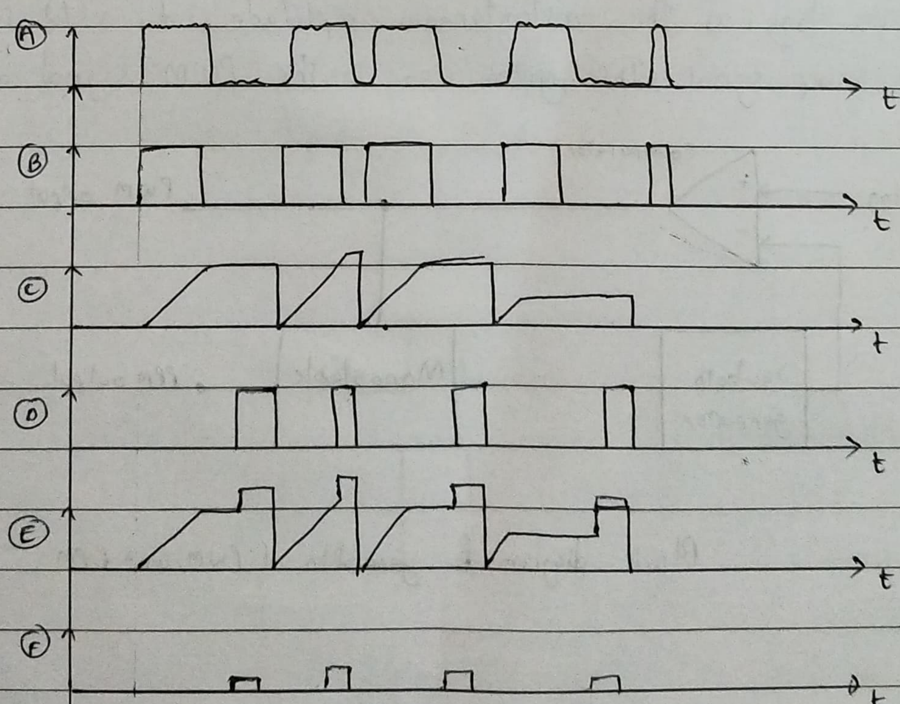


Block diagram for generation of PWM and PPM

Detection of PWM signal

The PWM signal recieved at input of detection circuit contains noise, hence it is first passed through pulde generator, to remove noise. Hence it is applied to a referance pulse generator which produces a train of constant amplitude constant width pulse. The regenerated pulse are also applied to the 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 referance 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 modulation signal back from the signal.





waveform for PWM detection

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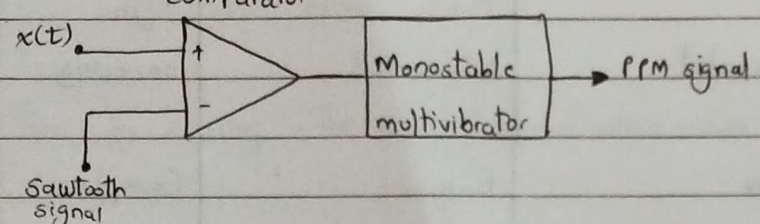
9) With the help of neat block diagram, and waveform, explain the working of PPM generation and degeneration

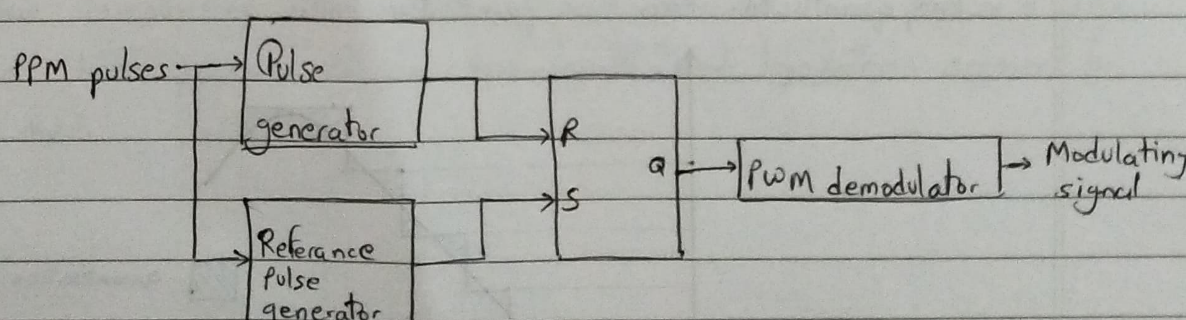
→ Generation of PPM signal:

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



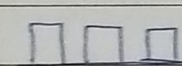
Demodulation of PPM signal

Noise corrupted PPM signal is fixed by the pulse generator and referance pulse generator which output the respective pulses at SR flipflop. Due to the set and reset signal applies to the flip f, we get a PWM signal at its output which is demodulated using PWM demodulator.



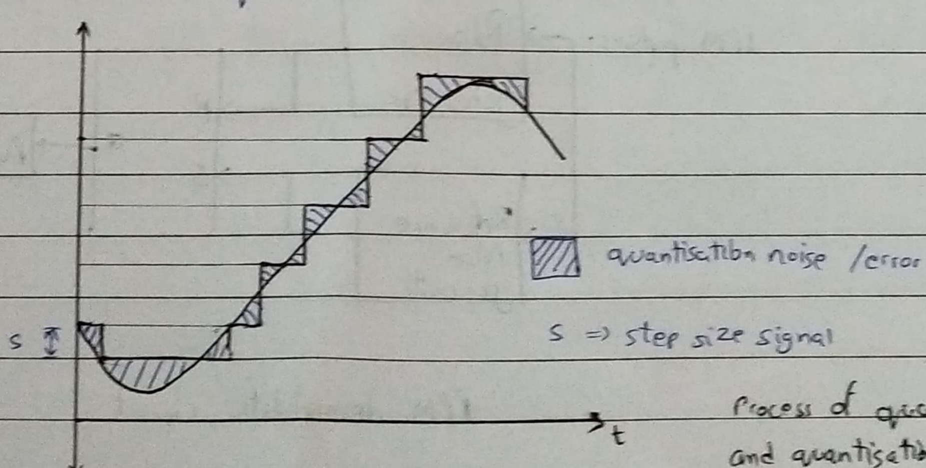
PPM demodulator circuit

10 Compare PAM, PWM and PPM

Parameter	PAM	PWM	PPM
i) Type of carrier	Train of pulses	Train of pulses	Train of pulses
ii) Variable characteristics of pulsed carrier	Amplitude	width	position
iii) B.W requirement	Low	High	High
iv) Noise immunity	Low	High	High
v) Need to transmit synchronised pulses	Not needed	Not needed	necessary
vi) Complexity of generation and detection	Complex	Easy	Complex
vii) Output waveform			

11) Describe quantisation noise or quantisation error

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



Process of quantisation and quantisation error

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12) What is companding? Show how companding reduce the quantization error

→ Companding is the process of compression and then expansion of the signal. With companded signal. Wi the higher amplitude analog signals are compressed prior to transmission and then explanded in the reciever. Companding is non uniform quantisation. Due to the inverse nature of compressor and expandor charateristics. The compander is a staight line. This indicates that all boosted signals are broght back to their original amplitudes

$$\text{Quntisation error} \quad N_q = \frac{s^2}{12}$$

Thus by boosting, The SNR of quantisation noise is better

13) Write A-law and μ-law of companding

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

Mathematically,

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

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

In the A-law companding, the compressor characteritic is of precise nature, made up of linear segment for low level inpults and logarithmic segment for high level inputs.

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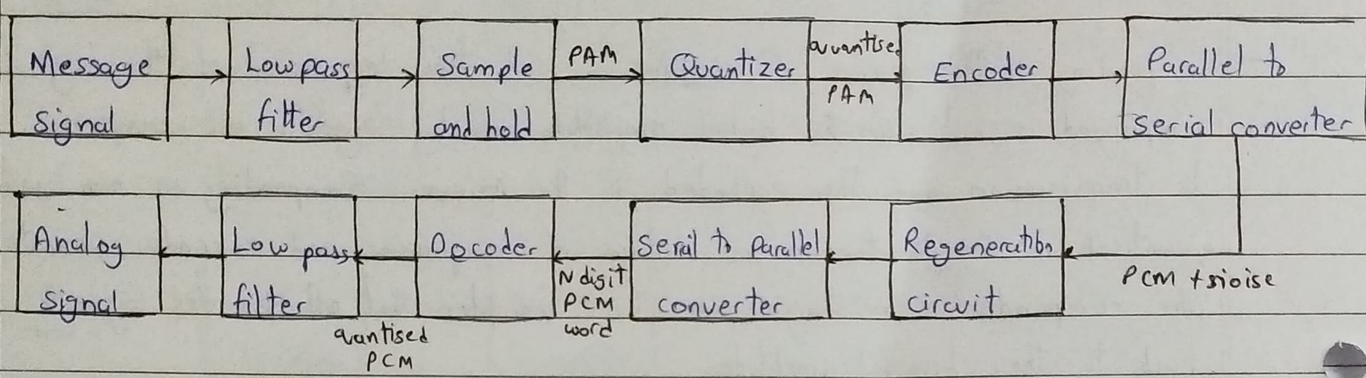
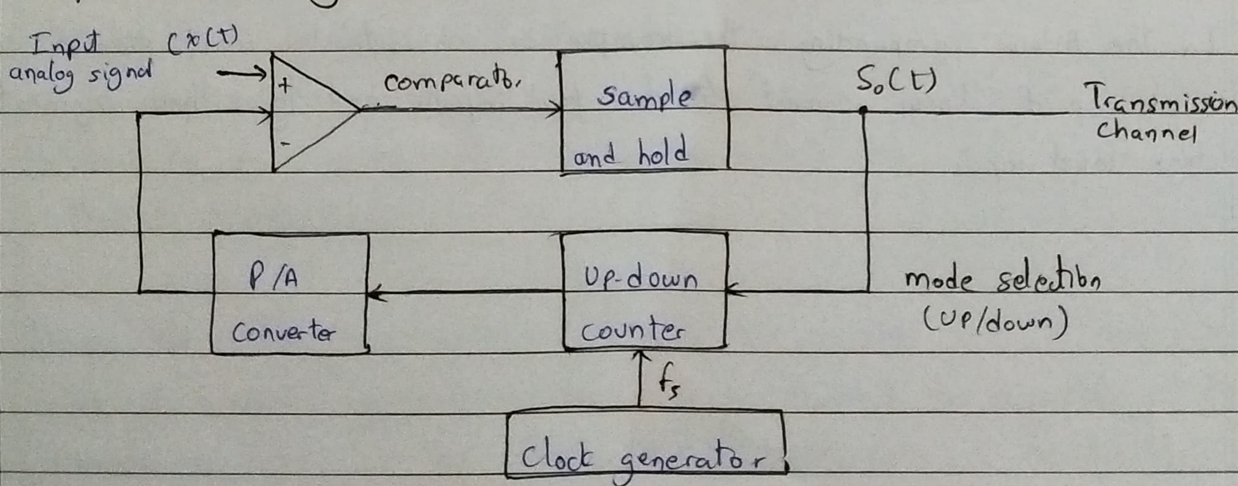
14) Draw the block diagram of PCM system, and explain function of each block

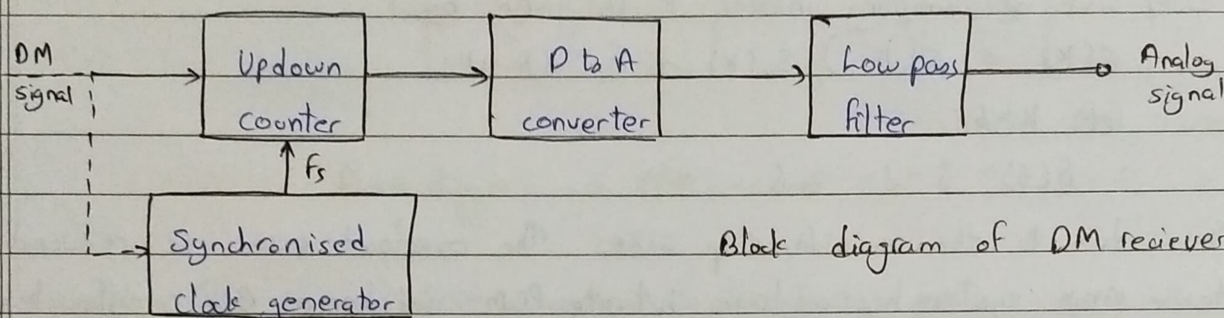


Analog signal $x(t)$ is passed through low pass filter having a cut off frequency $f_c = W$ Hz. This is then passed through sample and hold circuit where it is sampled at high sampling rate. These samples are then subjected to quantisation process in quantizer. The quantized PAM pulses are applied to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

15) Explain the working of delta modulator transmitter and reciever with neat lablock diag



x (t) and its quantised 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 entire clock cycle period, and its output is transmitted as the output of DM transmitter. The transmitted signal is also used to decide the mode of operation of an up/down counter.



Block diagram of DM reciever

The block diagram of DM reciever is shown above, on comparing it with the Transmitter block, we find that its identical to the chain of block producing signal x'(t). The original modulating can be recovered by passing this ginal throgh low pass filter

16) What are the limitations of linear delta modulator?, How do they overcome in adaptive delta modulator.

Disadvantages of linear Delta Modulators

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 of PCM

During the working of adaptive delta modulator, in order to the k^{th} form pulse rating edge, the processor generates a step which is equal to the magnitude to the sper generator in response to the previous $(k-1)^{th}$ clock edge, if the direction of both steps is same, the processor will increase in magnitude and decrease it otherwise

S_o ie output of ADM is given as

$S_o(t) = 1$ if $(x(t) > x'(t))$ before k^{th} edge

and $S_o(t) = -1$ if $x(t) < x'(t)$ before k^{th} edge

∴ Step size at sampling instant k is given by

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

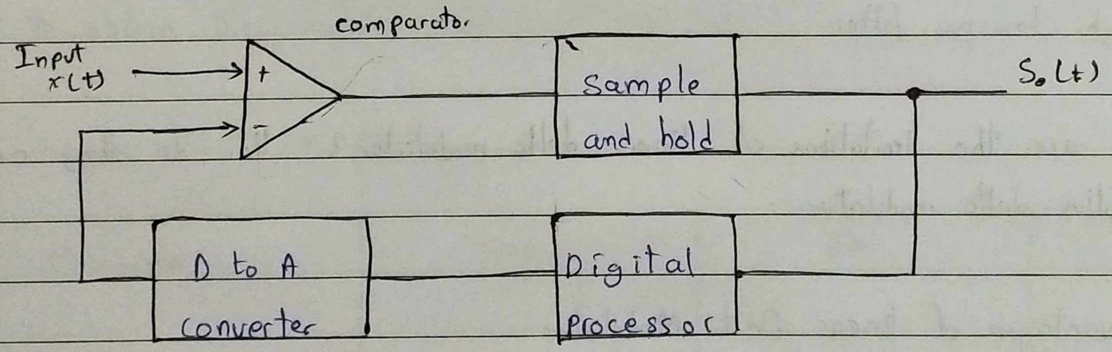
let $k = 6$

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

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

17) Draw the block diagram of ADM Transmitter and reciever and explain working

ADM transmitter

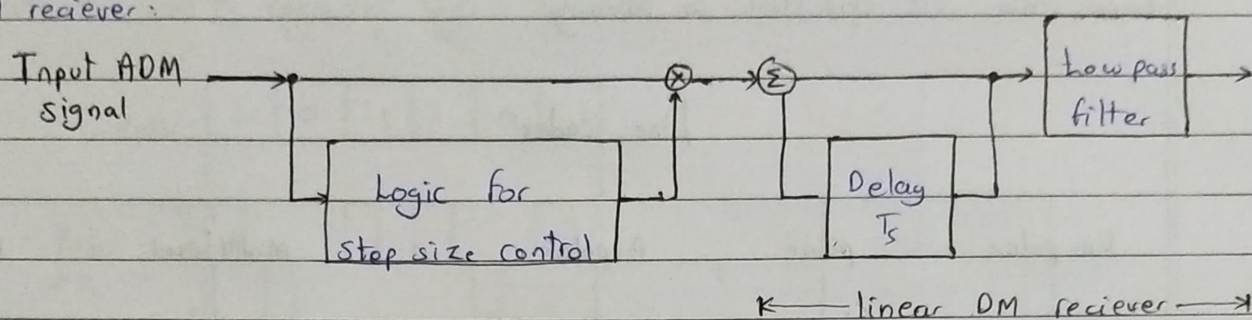


Block diagram of ADM transmitter

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 railing edge. The processor generates a step which is equal in magnitude to the step generated in respones to previous $(k-1)^{th}$ clock edge. If diretion of both step is same, then there is processor will

will increase the magnitude of the present step by δ

ADM reciever:



Block diagram of ADM reciever

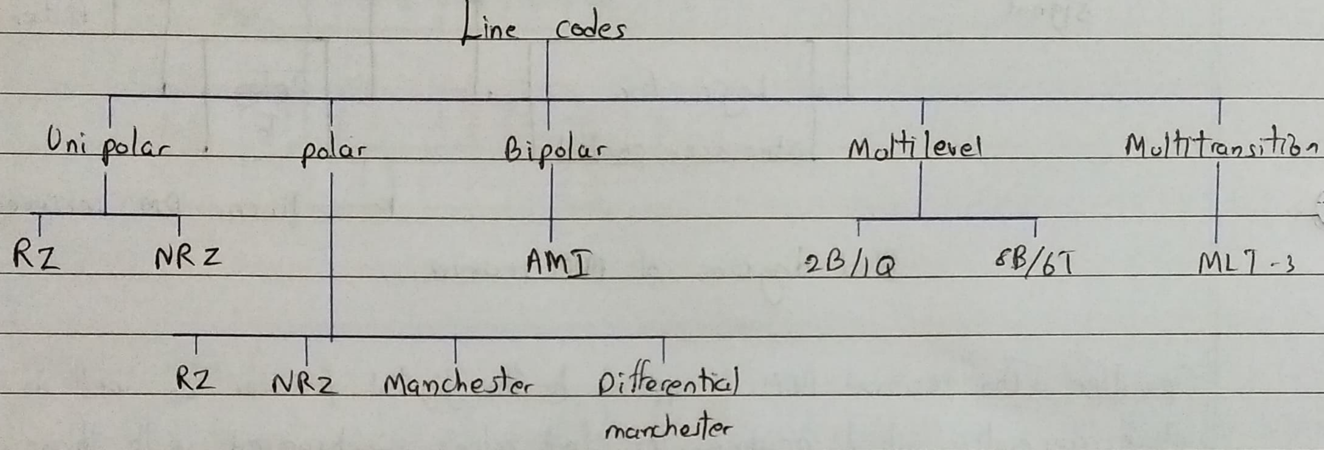
Operation:- The recieved ADM is applied to the digital process as well as sychronised clock generator, which produces the clock pulses synchronised with those at the transmitter. The digital processors produces a digital signal in parallel form which is converted into an analog signal by D to A converter.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

18) Compare PCM, DM, ADM

Parameter	PCM	DM	ADM
i) No of bits per sample	N = 4, 8, 16, 32	N = 1	N = 1
ii) Step size	Depends on number of Q levels	Step size is fixed	step size is variable
iii) Distortion /errors	Quantization error	Slope overload and granular noise	Granular noise
iv) System complexity	Complex	simple	simple
v) feedback from output	No feedback	feed back present	Feedback present

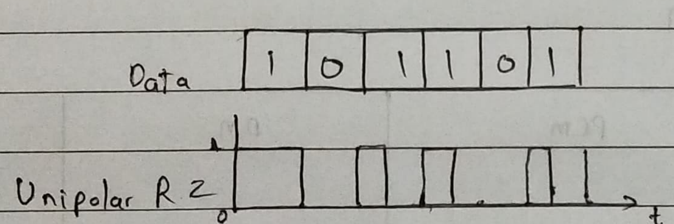
19) State and explain different types of line codes.

Line codes are classified as following



i) Unipolar RZ format

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

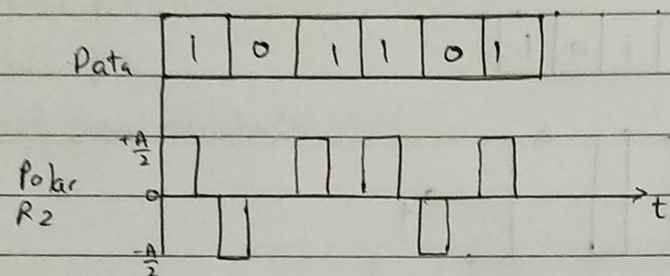


ii) Unipolar NRZ format

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

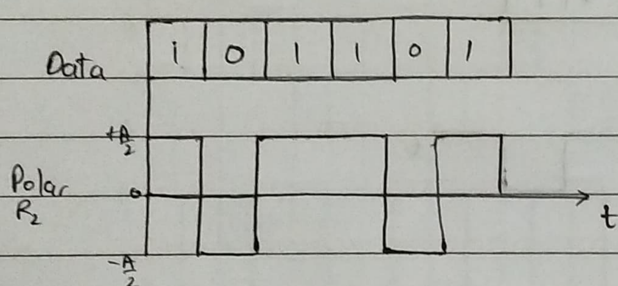
iii) Polar RZ format

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



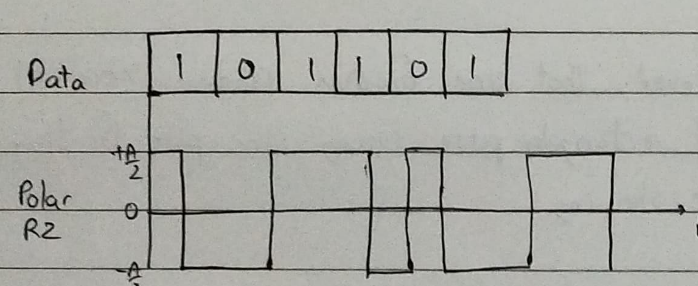
iv) Polar NRZ format

In this format, pulses of amplitude $+A/2$ of duration T_b is used to represent a logic "1" and pulse of amplitude $-A/2$ of same duration represented by "0"



v) Split phase manchester format

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



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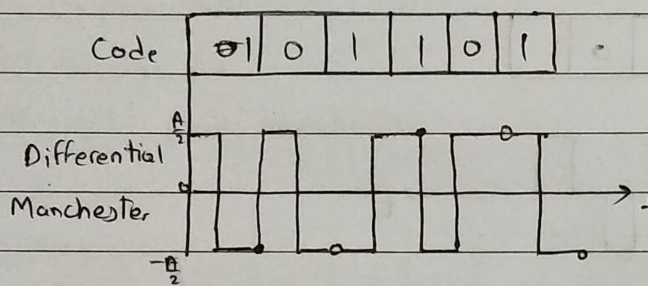
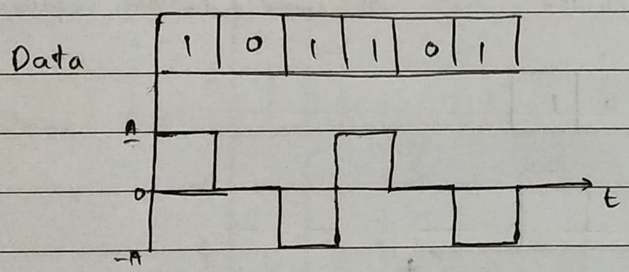
vi) Differential Manchester format

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



vii) Bipolar NRZ format (AMI)

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



20) What do you mean by ISI

In communication system when data is being transmitted in the form of pulses the o/put produced at the reciever due to the other bits or symbols interferes with output produced by the desired bit. This is called inter-symbol interferance or ISI

Remedies to reduce ISI

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

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2) The binary data 1 1 0 1 0 1 0 1 is transmitted over a baseband channel draw the waveform for transmitted data using the following data formats

i) Unipolar

ii) Unipolar RZ

iii) Bipolar RZ

iv) Split phase manchester