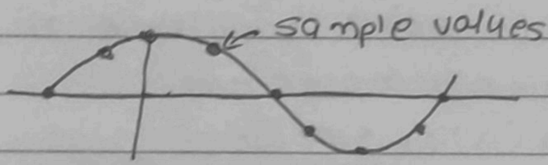
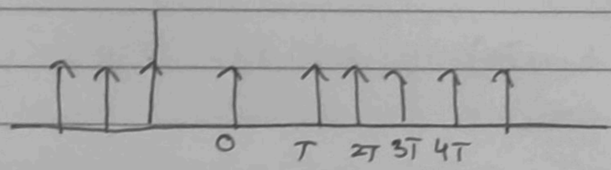
Assignment 5

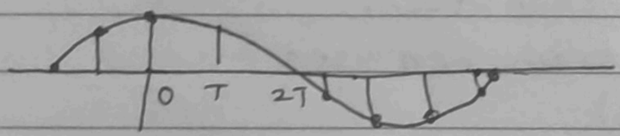
Q1] State and prove sampling theorem for low pass band limited signal

→ Let $x(t)$ be continuous time analog signal.





A unit impulse train used as sampling function.



sampled signal

Let $x(t)$ be a signal with finite energy & infinite duration

Let $x(t)$ be a strictly bandlimited signal.

- Let $s(t)$ be sampling function. It is train of unit pulses spaced by a period of T_s. This function samples the original signal at rate of f_s samples per second.

$T_s = \frac{1}{f_s}$ = sampling period.

$f_s = \frac{1}{T_s}$ = sampling rate

Step 1: Represent $s(t)$ mathematically

$s(t)$ is sampling function which is a train of impulses spacing between them is T_s seconds, frequency f_s.

The func $s(t)$ can be represented as:

$s(t) = \dots \delta(t+2T_s) + \delta(t+T_s) + \delta(t) + \delta(t-T_s) + \delta(t-2T_s) + \dots$

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

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step 2: Represent sampled signal $x_s(t)$ mathematically

It is present only at sampling instants i.e T_s, $2T_s$ etc and instantaneous amplitude is equal to amplitude of orginal signal $x(t)$ at sampling instants

- Let us represent instantaneous amplitude of $x(t)$ at various sampling points $t = nT_s$ as $x(nT_s)$

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

Substituting for $s(t)$

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

step 3: obtain fourier transform

fourier transform of train of impulses

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

Fourier transform of sampling func

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

$\therefore$ Sampling sig in time domain $= x_s(t) = x(t) \times s(t)$

Take fourier transform $x_s(f) = x(f) * S(f)$

$$x_s(f) = x(f) * \left[f_s \sum_{-\infty}^{\infty} \delta(f - nf_s) \right]$$

$$\therefore x_s(f) = f_s \sum_{-\infty}^{\infty} x(f) * \delta(f - nf_s)$$

$$\therefore \text{FF of } x_s(f) = f_s \sum_{-\infty}^{\infty} x(f - nf_s)$$

$x(f - nf_s)$ represents shifted version of $x(f)$ of $x(t)$

Thus depending on n, we get infinite no of spectrum.

$$\therefore x_s(f) = f_s\, x(f) + \sum_{-\infty}^{\infty} f_s\, x(f - nf_s)$$

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Q2] What do you mean by alising, How it can be avoided?

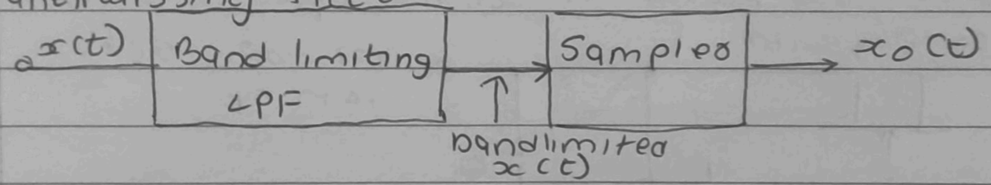
→ The phenomenon of a high frequency in the spectrum of orginal signal $x(t)$. taking on identity of lower freq in spectrum of the sampled signal $xs(t)$, is called as aliasing or fold over error.

Due to aliasing some of the information contained in orginal signal $x(t)$ is lost in the process of sampling.

Aliassing can be eliminated by:-

① Using a bandlimited low pass filter and pass signal $x(t)$ through it before sampling.

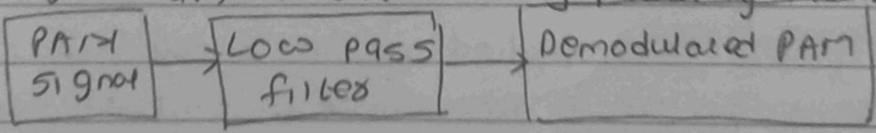
② This filter has a cutoff frequency at $fc = \omega$, it will strictly bandlimit the signal $x(t)$ before sampling. This is also called as antialissing filter

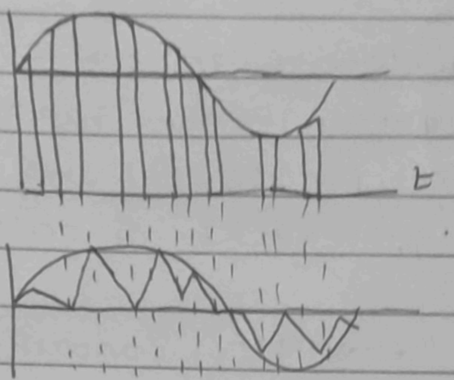


Q4] Explain with digram and waveforms generation and detection of natural 2 flat top PAM.

→ In PAM, The amplitude of the pulsed carrier is changed in prop with instanteneous amplitude of modulating signal $x(t)$.

PAM signal can be detected by passing through low pass filter

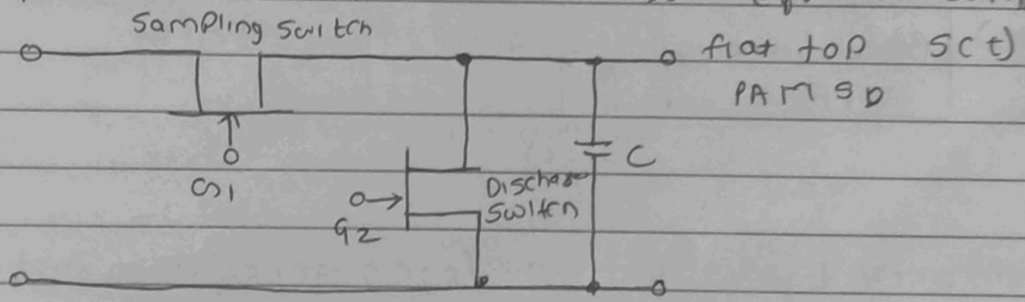


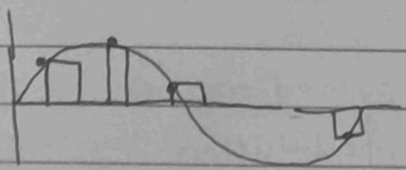


The low pass filter cutoff freq is adjusted to fm so the all the high frequency ripple is removed and original modulating signal is recovered.

② Flat top PAM

The analog signal x(t) is sampled instantaneously at rate fs = 1/Ts and duration of each sample is lengthened to τ. amplitude of these pulses are const & equal to sampled value





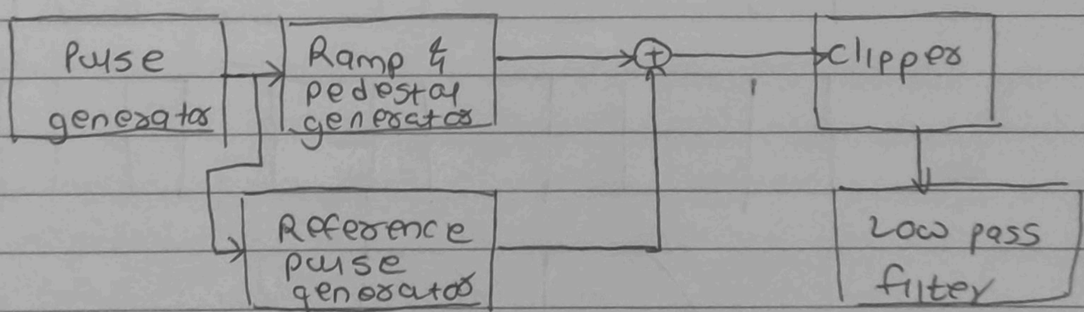
sample & hold ckt consists of two FET switches & a capacitor analog signal x(t) is applied at I/P & flat top PAM at O/P gate pulse is applied to gate G1 at instant of sampling. the sampling switch will turn on & capacitor changes through to sample val x(nT) The sampling switch will turn off Both FET will remain off for duration of τ sec & both capacitors will hold the vtg const for this period The pulse is stretched to τ seconds & we get flat top pls

Q5] Compare Natural & flat top PAM.

Natural PAM	Flat top PAM
① Train of finite duration pulses.	① Train of finite duration pulses
② Uses a chopper.	② Uses a sample & hold ckt.
③ Pratically realizable	③ Pratically realizable.
④ Satisfies nyquist critera	④ Satisfies nyquist criteria.
⑤ Signal power increase with increase in pulse width.	⑤ Signal power increase with increase in puse width.
⑥ Bandwidth increases with reduction in pulse width	⑥ Bandwidth increases with reduction in pulse width

Q9 PWM Detection

→



The PWM signal recieved at the input of the detection ckt is contaminated with noise. This signal is applied to the pulse generator circuit which regenerates PWM. Noise is removed & pulses are squared.

The regenerated pulses are applied to a reference pulse generator. It produces a train of constant amplitude constand width pulses. These are synchronized to the leading edges of regenerated PWM delayed by fix interval.

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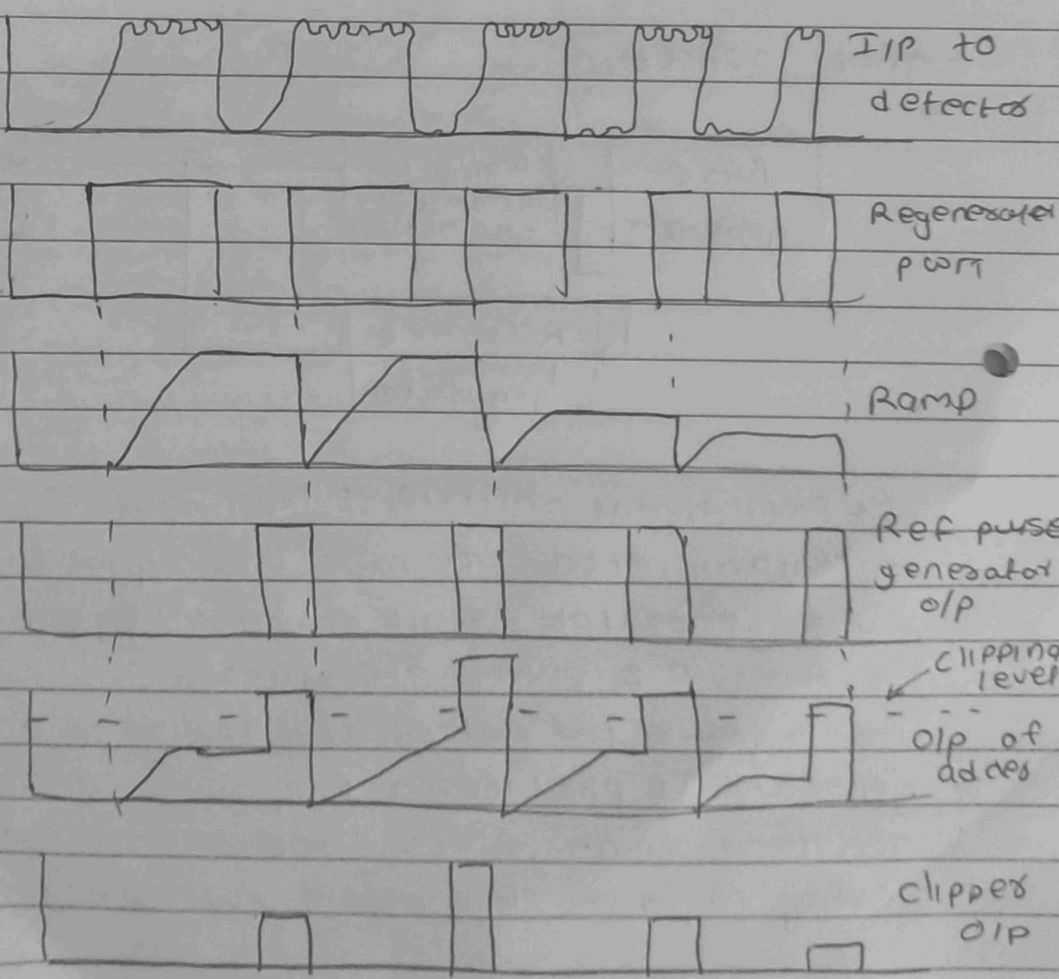
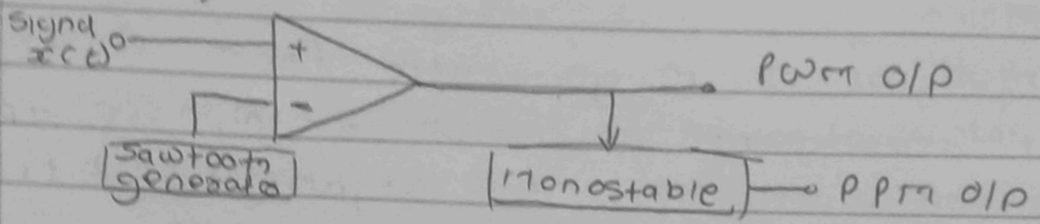
The regenerated PWM pulses are also applied to a ramp generator. At o/p of it we get constant slope ramp for duration of pulse. height of ramp is thus proportional to widths of PWM pulses. At end of the pulse a sample & hold amplifier retains the final ramp vtg until it reset at end of pulse.

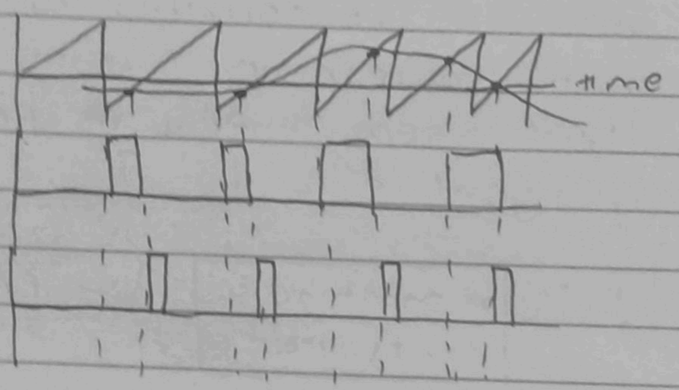
Constant amp pulses at o/p of refrence pulse generator are then added to ramp signal. o/p of adder is then clipped off at a threshold level to generate a PAM signal at o/p of clipper.

A low pass filter is used to recover org modulating sig



Q8 PPM PWM generation



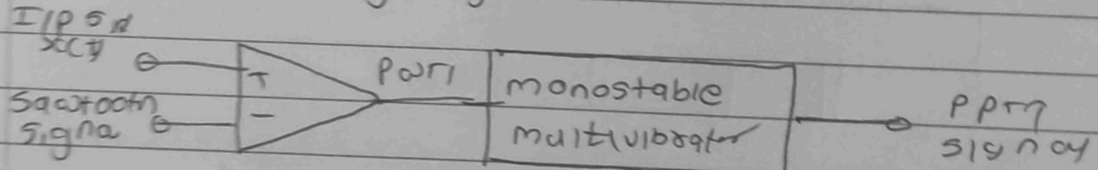


A sawtooth generates a sawtooth signal of freq f_0, it is the sampling signal, it is given to inv terminal of comparator. modulating signal $x(t)$ is applied to non-inv terminal. The comparator o/p will remain high as long as the instantaneous amp of $x(t)$ is higher than the ramp signal. This generates PWM signal at comparator op.

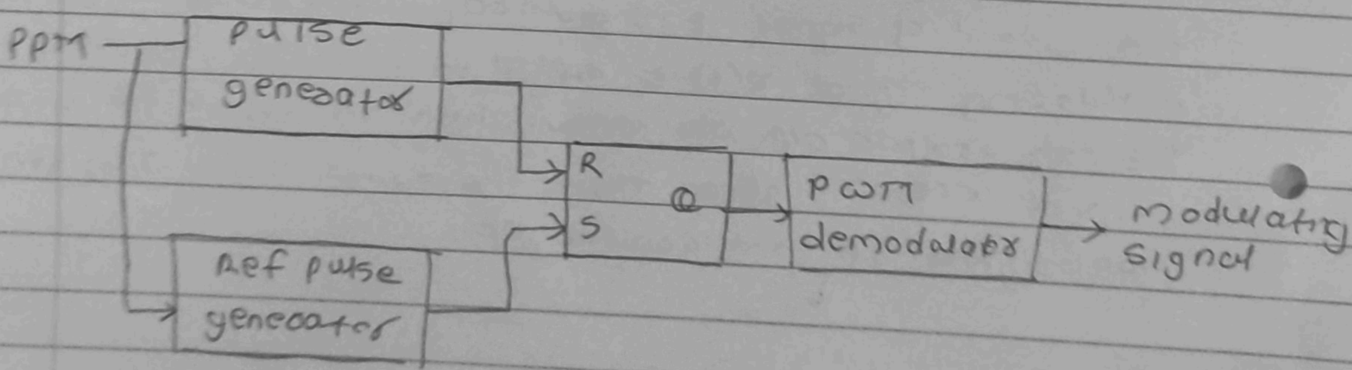
Q7] PPM

① Generation

The PPM signal can be generated from PWM signal. PWM pulses obtained at comparator o/p are applied to monostable multivibrator. It is negatively edge triggered. Hence corresponding to each trailing edge of PWM signal monostable o/p goes high. It remains high for a fixed time by its own RC components. Thus as trailing edge of PWM signal keeps shifting in proportion with modulating signal x(t).



② Demodulation



The noise corrupted PPM waveform is recieved by the PPM demodulator ckt. The pulse generator develops a pulsed waveform at its o/p of fixed duration & apply these pulses to reset pin of SR flip-flop.
A fixed period reference pulse generated from incoming PPM waveform & SR flip-flop is set by ref pulses

Due to the set and reset signals applied to flip-flop we get a PWM signal at output. The PWM signal can be demodulated using PWM demodulator

Q8]

PAM	PWM	PPM
① carrier is train of pulses.	① carrier is train of pulses.	① carrier are train of pulses.
② Amplitude of pulsed carrier is varied	② width of pulsed carrier is varied.	② Position of pulsed carrier is varied.
③ Low bandwidth requirement	③ High bandwidth requirement	③ High bandwidth requirement.
④ Low noise immunity	④ High noise immunity	④ High noise immunity.
⑤ Information is in amp variations	⑤ ~~Info~~ Information is stored in width variations	⑤ Information is in position variation.

Q9] Describe Quantization noise or quantization error

→ The difference between the instantaneous values of the quantized signal and input signal is called as quantization error or quantization noise.

$\epsilon = x_q(t) - x(t)$. The max value of quantization is $\pm S/2$ where S is step size. To reduce the quantization error we have to reduce step size by increasing no of quantization levels.

mean sq value of quantization error $= \frac{S^2}{12}$

Q10] What is companding? Show how companding reduce quantization error.

→ Companding is non-uniform quantization. It is required to be implemented to improve signal to quantization ratio.

Quantization noise is $N_q = s^2/12$

This shows that in uniform quantization once the step size is fixed, quantization noise remains constant. But signal power is not constant. The signal to quantization noise ratio for the weak signal is very poor. This will affect the quality of signal. The remedy is to use companding.

Companding = compressing + Expanding.

The compression of signal at transmitter & expansion at reciever is companding.

Due to inverse nature of compressor and expander characteristics, overall characteristics of compander is a straightline.

This indicates that all the boosted signals are brought back to original amplitudes.

Q11] Write A law & μ-law companding

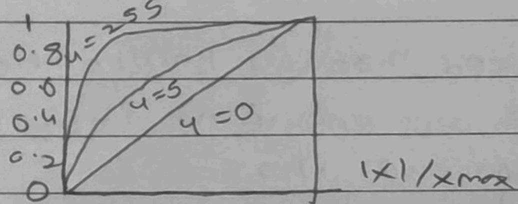
(1) μ-law companding

The compressor characteristic is continuous. It is approximately linear for smaller values of input levels & logarithmic for high levels of i/p signal.

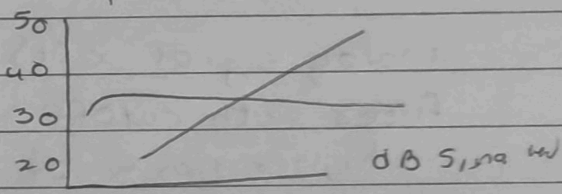
$$z(x) = (\operatorname{sgn} x) \frac{\ln(1+\mu|x|/x_{max})}{\ln(1+\mu)}$$

$$0 \le |x|/x_{max} \le 1$$

z(x) is o/p and x is i/p to compressor. $|x|/x_{max}$ is normalized value of i/p with respect to max value x_{max}. The (sgn x) term represents ±, ie +ve & -ve value of i/p & o/p. The μ law companding is used for speech & music signals. It is used for PCM telephone systems in USA, Canada, Japan.



pcm performance

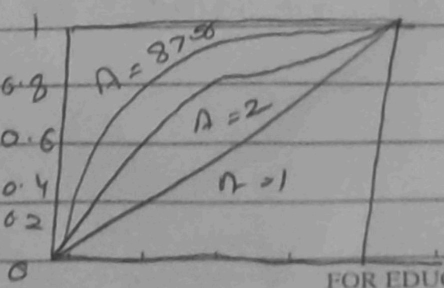


② A-law companding

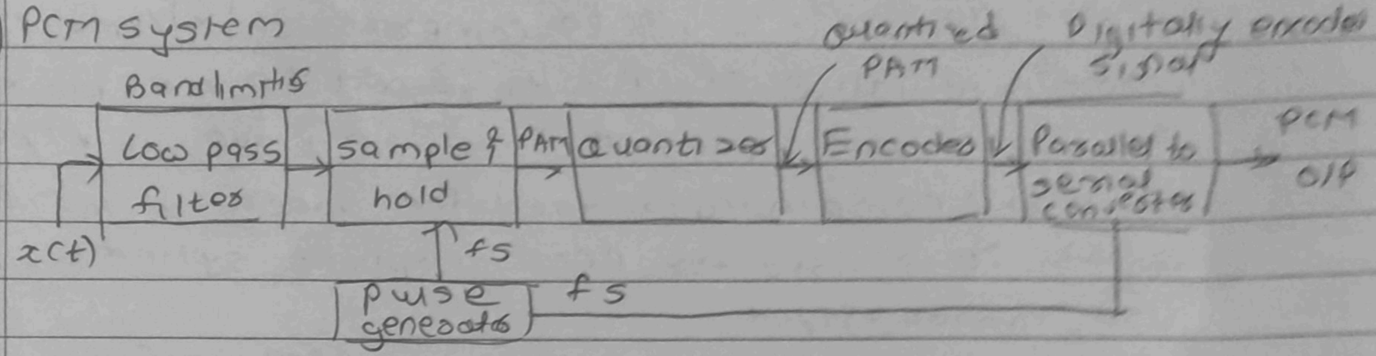
In the A law, the compressor characteristic is of piecewise nature, made up of a linear segment for low level inputs & logarithmic segment for high level inputs. At A=1 characteristic is linear which corresponds to uniform quantization. Practically used value of A is 87.56. A law is used for PCM telephone system in Europe

$$\frac{z(x)}{x_{max}} = \frac{A|x|/x_{max}}{1+\log_e A} \qquad 0 \le \frac{|x|}{x_{max}} \le 1$$

$$\frac{1+\log_e[A|x|/x_{max}]}{1+\log_e A} \qquad \frac{1}{A} \le \frac{|x|}{x_{max}} \le 1$$



Q12) PCM system



analog signal x(t) is passed through bandlimiting low pass filter with cutoff W Hz This will ensure no freq compo— is higher than W. It will eliminate aliasing

- It is then applied to sample & hold ckt where it is sampled at adequately hig sampling rate. o/p is flat topped PAM signal
- These samples are quantized in quantizer to reduce the effect of noise. It generate quantized PAM at o/p.
- Quantized PAM pulses are applied to encoder which is an A to D converter each level is converted into N bit digital word by converter, N is 8, 16, 32, 64 ---
- Encoder o/p is converted into a stream of pulses by parallel to serial convertor. at o/p we get a train of digital pulses.
- A pulse generator produces a train of rectangular pulses with each pulse of duration of τ sec. fs of is fs Hz. It act as sampling signal for sample & hold block also as clock signal for parallel to serial converter. fs is adjusted to satisfy the Nyquist criteria