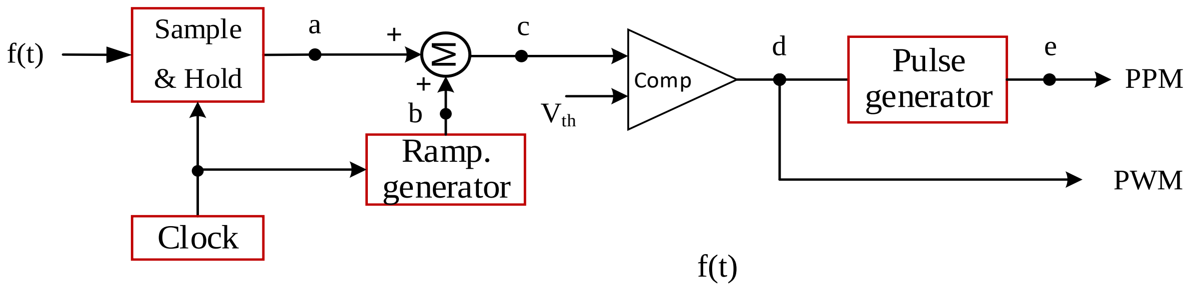
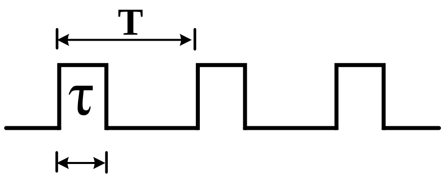
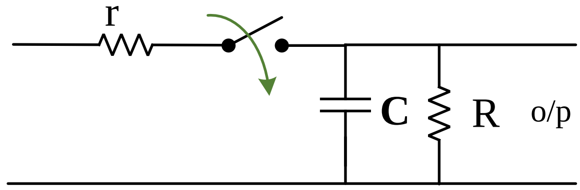
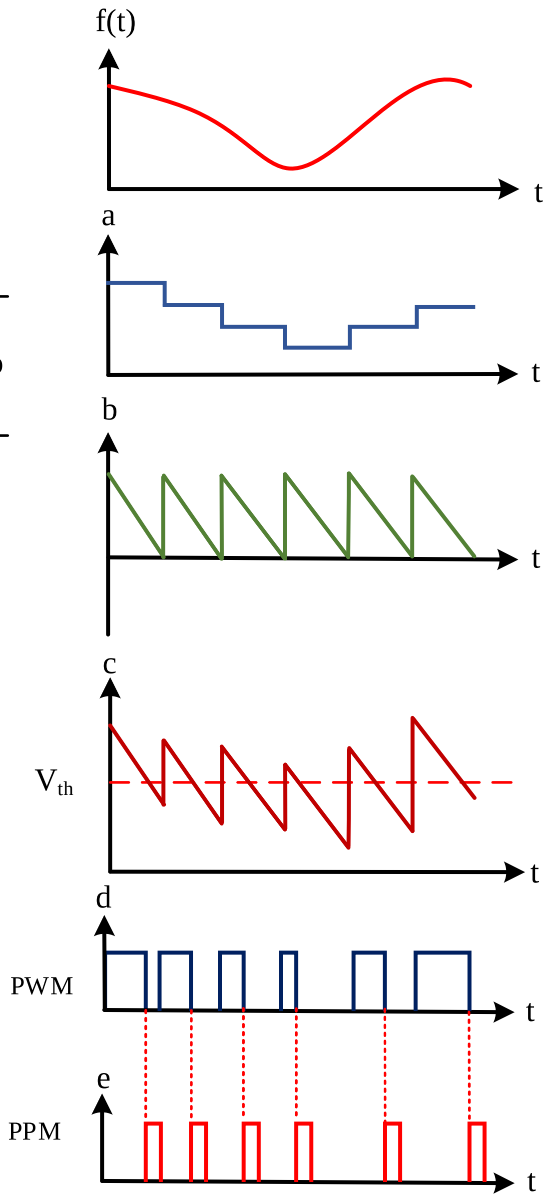
2- PWM and PPM:

Generation:







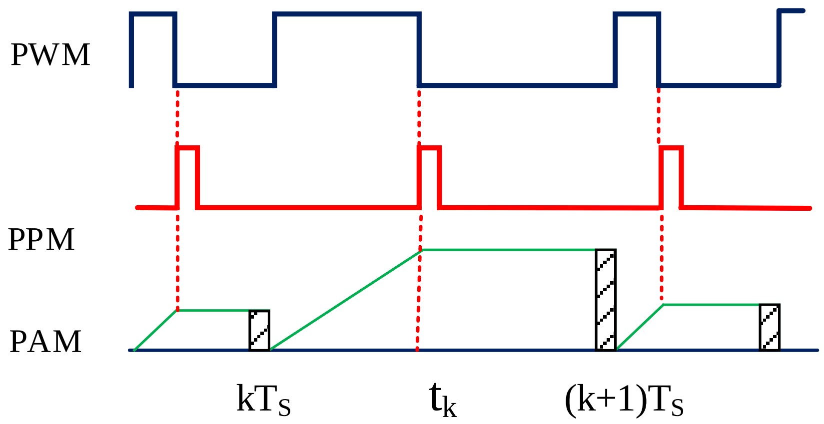


$rc \ll \tau$

$RC \gg T$ Sample & Hold circuit

Detection:

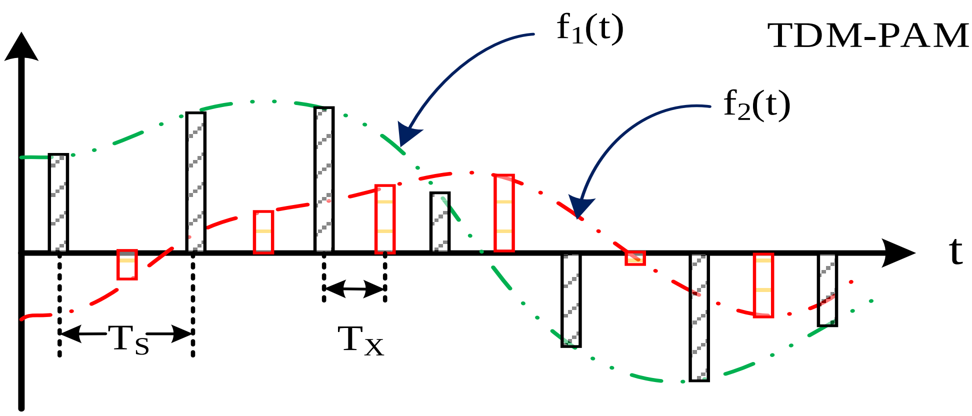
One method of detection is to convert PWM or PPM signals to PAM ones using ramp generator starts at kT_S, stops at t_k, restarts at $(k+1)T_S$ and so forth.



PWM and PPM, are rarely used now in communication systems.

Time Division Multiplexing (TDM):

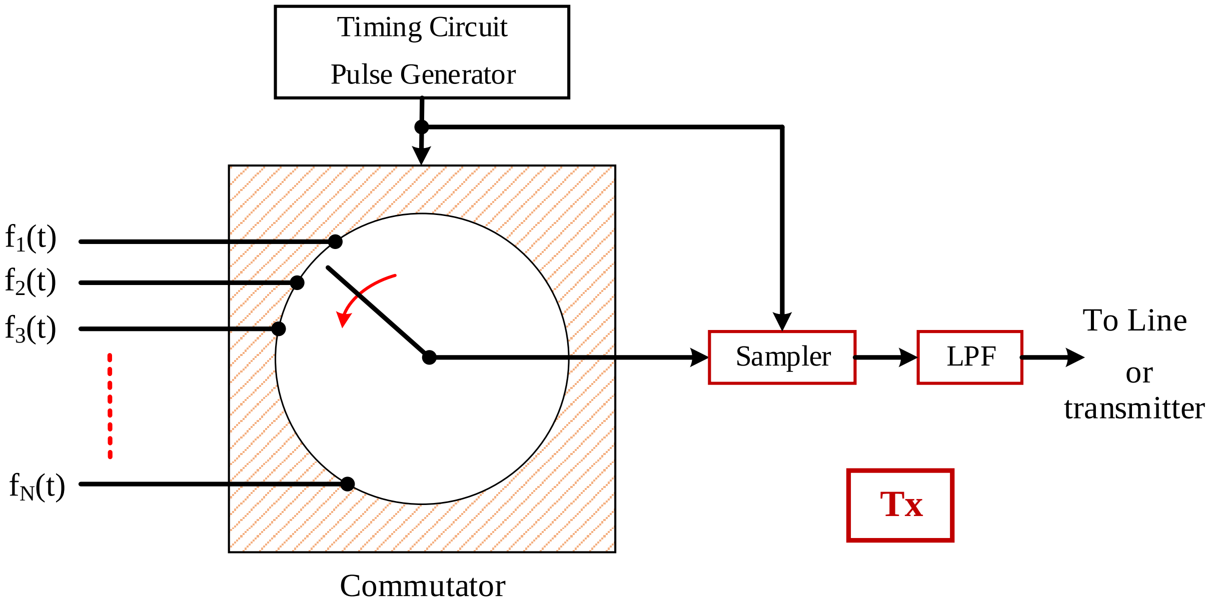
A mode of transmission in which simultaneous transmission of several baseband signals on time-sharing basis is possible.

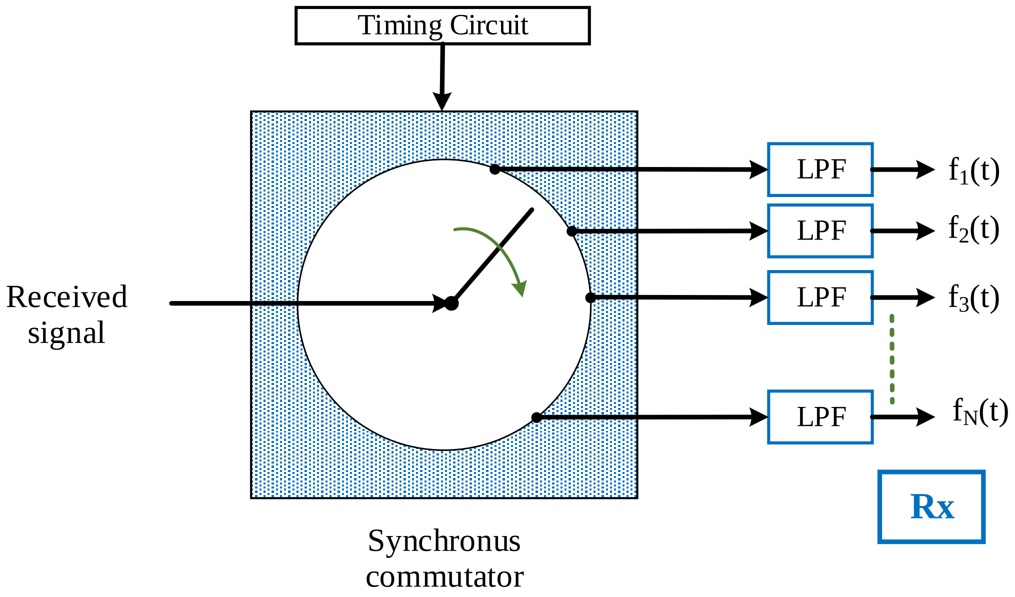


T_S: Sampling time for each signal ($T_s \leq \frac{1}{2f_{max}}$ $for\ nyquist\ sampling$)

T_x: Clock frequency for PAM/TDM system.

$T_x = \frac{T_s}{N}$; where N: number of messages (channels)





If N identical messages have the same f_{max} , then $f_s \geq 2Nf_{max}$ at the channel the sample rate is at least ($2Nf_{max}$).

The minimum required bandwidth of the channel is half the sampling frequency or

$$BW_{min} \geq Nf_{max}$$

Hz for TDM-PAM ….(6-4)

Ex 6-3:

Twelve speech signals are TDM-PAM transmitted, find minimum sample rate at the channel and minimum required BW.

Solution:

f_{max}=3.5 kHz (for speech)

$f_s \geq 2Nf_{max}$ or $f_s \geq 2 \times (12) \times (3.5\ kHz)$

$f_{s_{min}}$=84 kHz

$$BW_{min} = \frac{1}{2} f_{s_{min}} = 42\ kHz$$

Ex 6-4:

Determine the minimum transmission BW in a TDM system transmitting 20 different messages, each message signal have BW of 5 kHz; compare the result if FDM is used with AM & SSB techniques.

Solution:

- **TDM**

  $f_{min} = 2Nf_{max} = 2 \times (20) \times (5\ kHz)$=200 kHz

  $BW_{min} = 100\ kHz$

- **FDM**

  AM: $BW_{min} = 2(5 * 20)kHz = 200\ kHz$

  SSB: $BW_{min} = 5 * 20\ kHz = 100\ kHz$

  ∴ TDM/PAM is more efficient in terms of BW than FDM/AM

Digital Communication System:

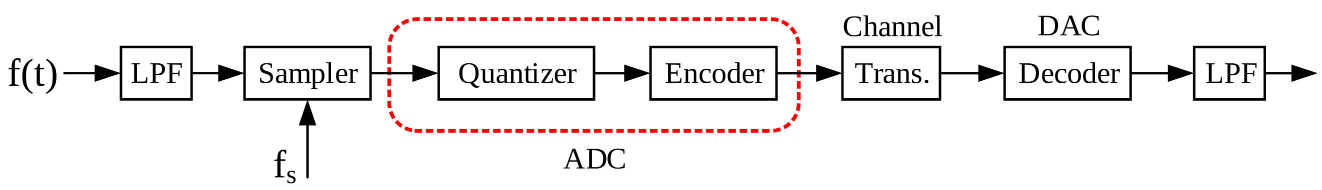
Digital communication has several advantages over analog communication:

1- Digital communication has high immunity to channel noise and channel distortion.
2- Regenerative repeaters along the transmission path can detect and retransmit a new, clean signal.
3- Digital hardware implementation is flex able (it may use microprocessors, digital switching and LSI-ICs)
4- Digital signals can be added to yield low error and high fidelity as well as privacy.
5- It is easier to multiplex digital signals.
6- Exchange of SNR and BW can be done more effectively.

Digital Transmission of Analogue Signals:

1- Pulse Code Modulation (PCM):

This is widely used in digital transmissions. Its block diagram is as shown below:

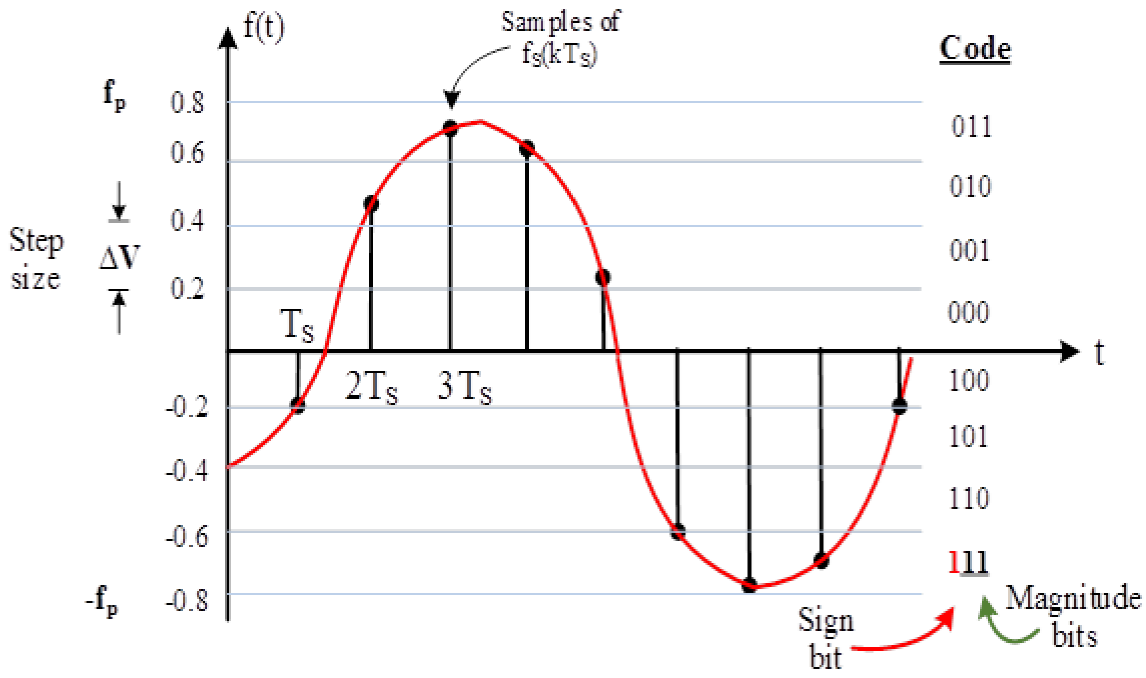


ADC: Analogue to digital converter.

DAC: Digital to Analogue converter.

The output of the sampler $f_s(kT_s)(ADC)$. Assuming that f(t) has $\pm f_p$ peak voltage level, (ADC full scale), the quantizer will divide the $+f_p$ to $-f_p$ range into L equally spaced intervals of size ΔV (step size) then:

$$\Delta V = \frac{2f_p}{L} \quad \text{volt} \qquad \text{.... (6-5)}$$



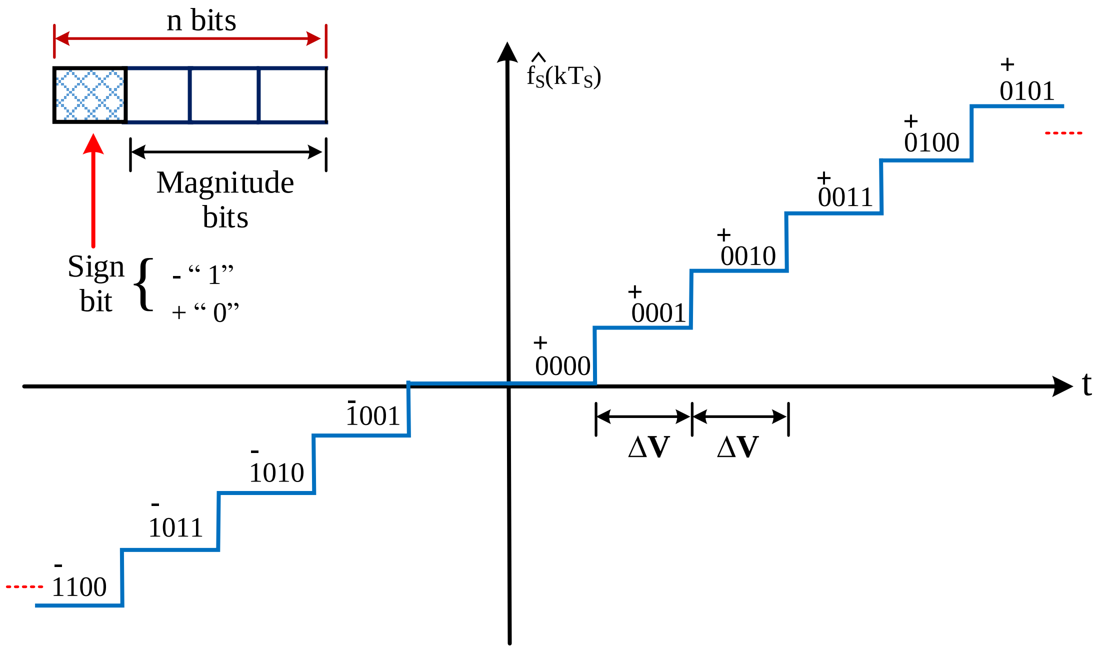
Quantizing Noise:

Since the quantization process introduces some fluctuations about the true value, these fluctuations can be regarded as noise. As the number of quantization levels L increases, the quantization noise decreases.

$$N_q = \frac{f_p^2}{3L^2} \quad \text{Volt}^2 \qquad \text{.... (6-6)}$$

Encoding:

ADC will then encode the quantized values according to a certain binary code. The uniform PCM with equal step size mostly uses the **signed binary code** of n bits.



For n=4, then the $\pm f_p$ values will be encoded as shown above, this is called transfer characteristic of the PCM encoder. The relation between number of quantizing levels and number of bits of encoder is:

$$L = 2^n \quad \text{Or} \quad N = log_2 L \qquad \ldots \text{(6-7)}$$

Note:

If n for a given value of L is *not integer* number,

Then n is computed using $n = int(log_2 L) + 1$,

and L is corrected using $L = 2^n$