Advantages of PWM

1.Less effect of noise i.e., very good noise immunity.

2.Synchronization between the transmitter and receiver is not essential (Which is essential in PPM).

3.It is possible to reconstruct the PWM signal from a noise, contaminated PWM, as discussed in the detection circuit. Thus, it is possible to separate out signal from noise (which is not possible in PAM).

Disadvantages of PWM

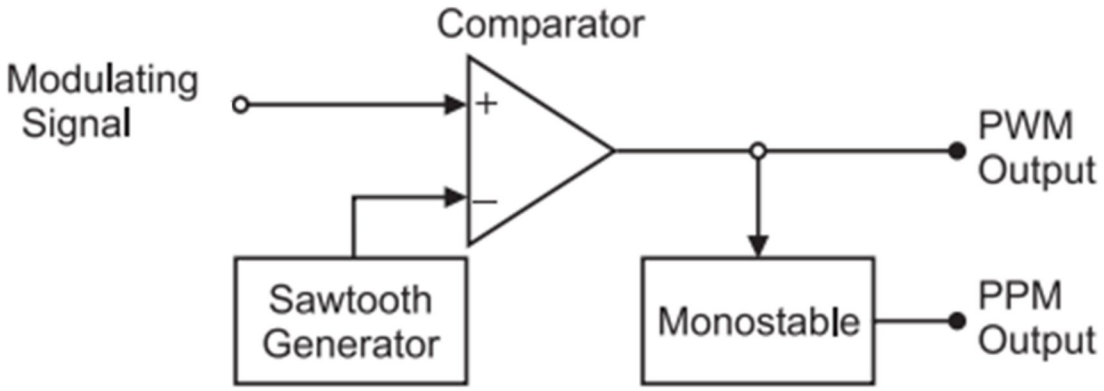
1.Due to the variable pulse width, the pulses have variable power contents. Hence, the transmission must be powerful enough to handle the maximum width.

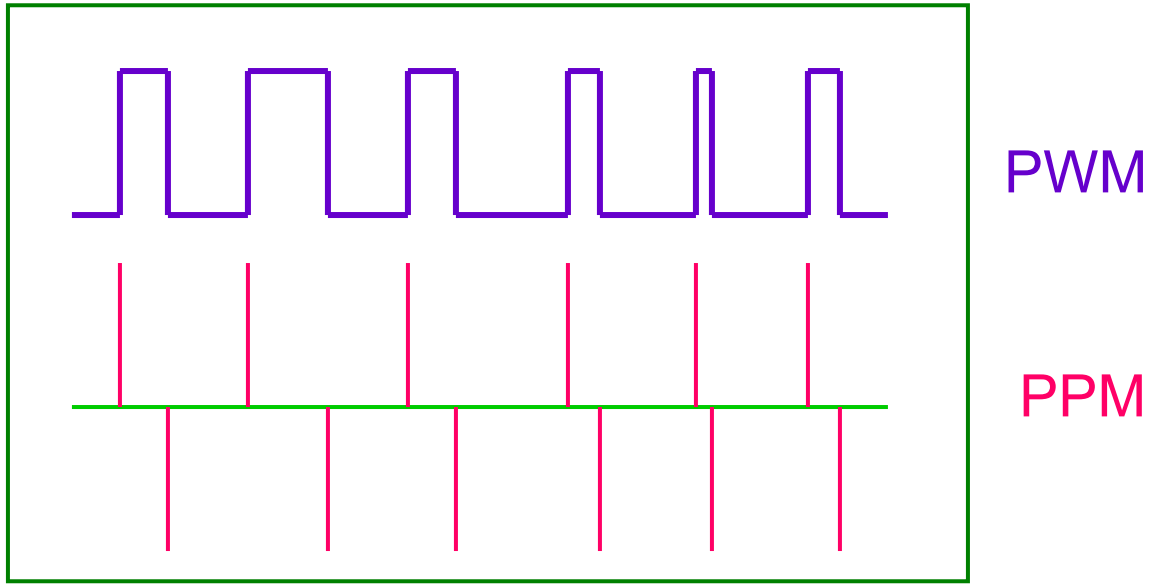
2.In order to avoid any waveform distortion, the bandwidth required for the PWM communication is large as compared to bandwidth of PAM.

Pulse Position Modulation (PPM)

In this type, the sampled waveform has **fixed amplitude and width** whereas the position of each pulse is varied as per instantaneous value of the analog signal.

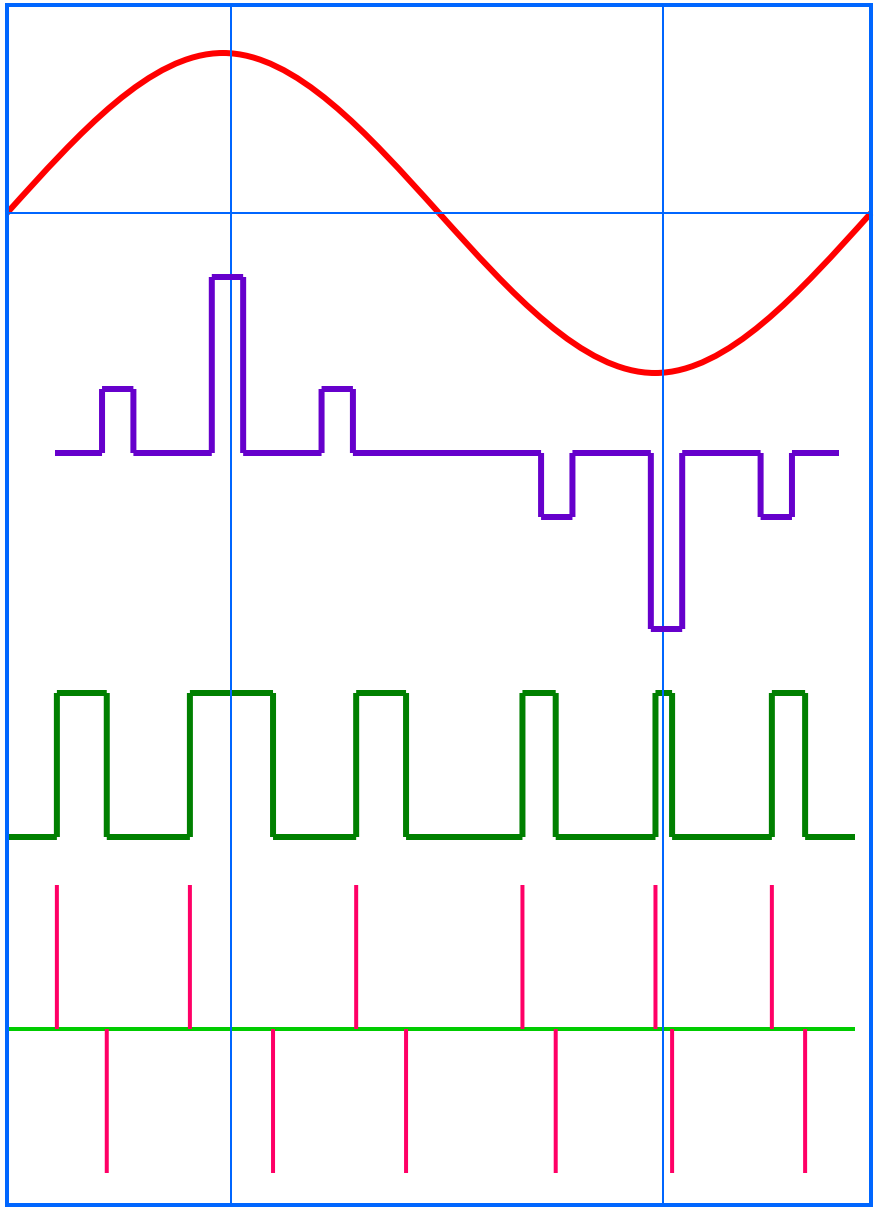
PPM signal is further modification of a PWM signal. It **has positive thin pulses (zero time or width) corresponding to the starting edge of a PWM pulse** and **negative thin pulses corresponding to the ending edge of a pulse**.





This wave can be further amended by eliminating the whole positive narrow pulses. The remaining pulse is called **clipped PPM.**

PAM, PWM and PPM at a glance



Analog Signal

Amplitude Modulated Pulses (PAM)

Width Modulated Pulses (PWM)

Position Modulated Pulses (PPM)

Pulse Digital Modulation : Pulse Code Modulation (PCM)

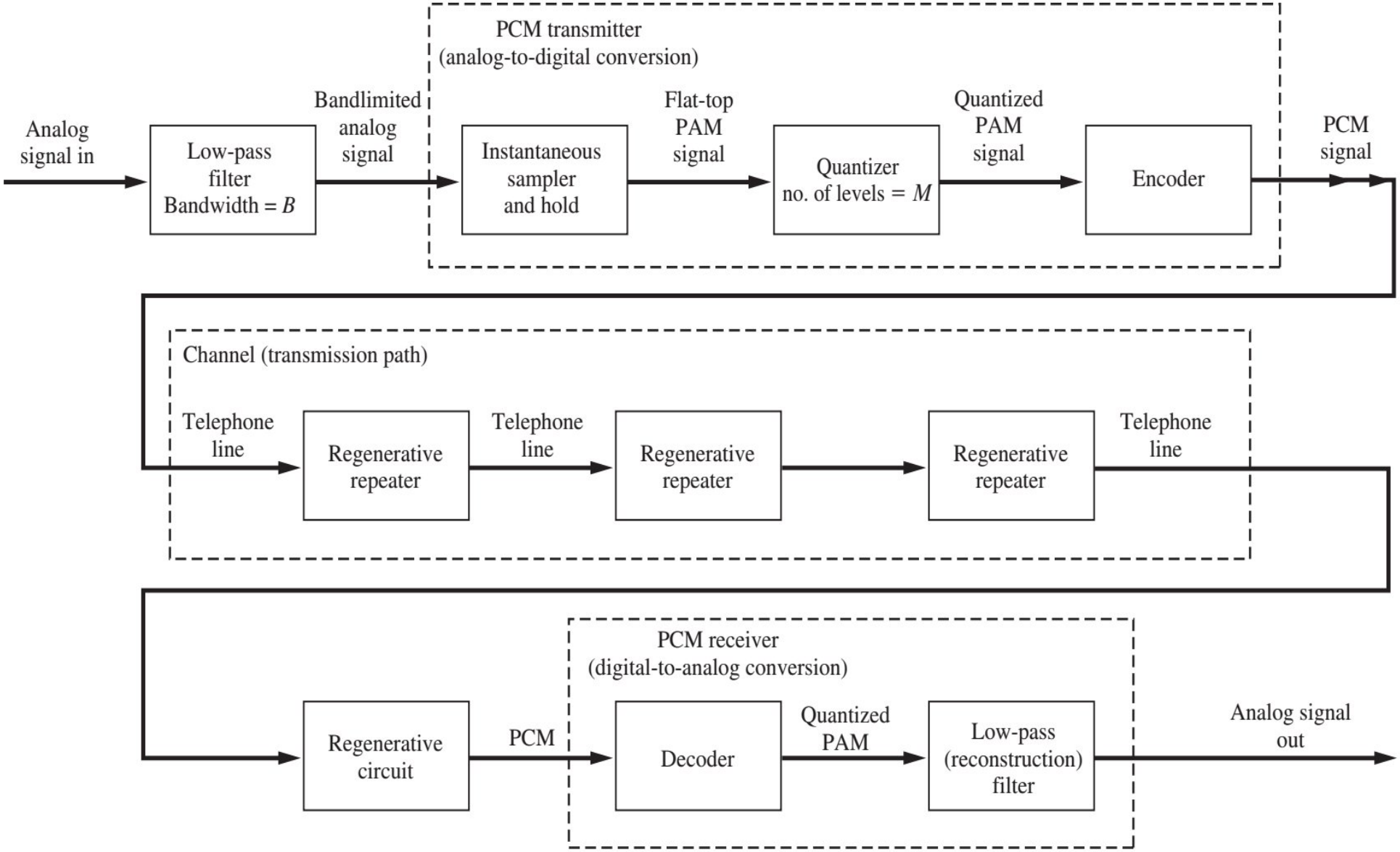
The modulation methods PAM, PWM, and PPM discussed in the previous lectures still represent analog communication signals since the height, width, and position of the PAM, PWM, and PPM, respectively, can take any value in a range of values.

One solution to reduce the disturbances and distortions on a transmission line is by using digital signals for transmission. The digital information can be transmitted directly as a baseband signal through a low pass filter channel or carried through a band pass channel. Thus, Pulse Code Modulation (PCM) is an Analog-to-digital conversion.

The digital information is obtained by dividing the message signal into certain amplitude stages, the so-called quantization intervals. Each of these intervals is assigned a code word. The finer the resolution of the amplitude stages, the greater the number of quantization intervals and the greater the number of positions of the binary codes. For transmission, it is not necessary to quantize and code every instantaneous value of the information signal. Individual sampling values as supplied by the PAM technique are sufficient. The PAM is therefore used frequently as a pre-stage to PCM.

Pulse Code Modulation (PCM)

The principle of a pulse code modulation is illustrated below.



Pulse Code Modulation (PCM)

The analog message signal is limited in bandwidth by the low pass filter (requirement of the sampling theorem f_s min > 2 f_m). The frequency band limited information signal is sampled with the sampling frequency f_s . The amplitude of the sampling values constant until the next sample arrives. This is enabled by the hold circuit. Sample and hold form a unit and are often referred to as a sample and hold circuit.

PCM modifies the pulses created by PAM to create a completely digital signal. PCM quantizes the PAM pulses (assigns integer values in a specific range to sampled instances

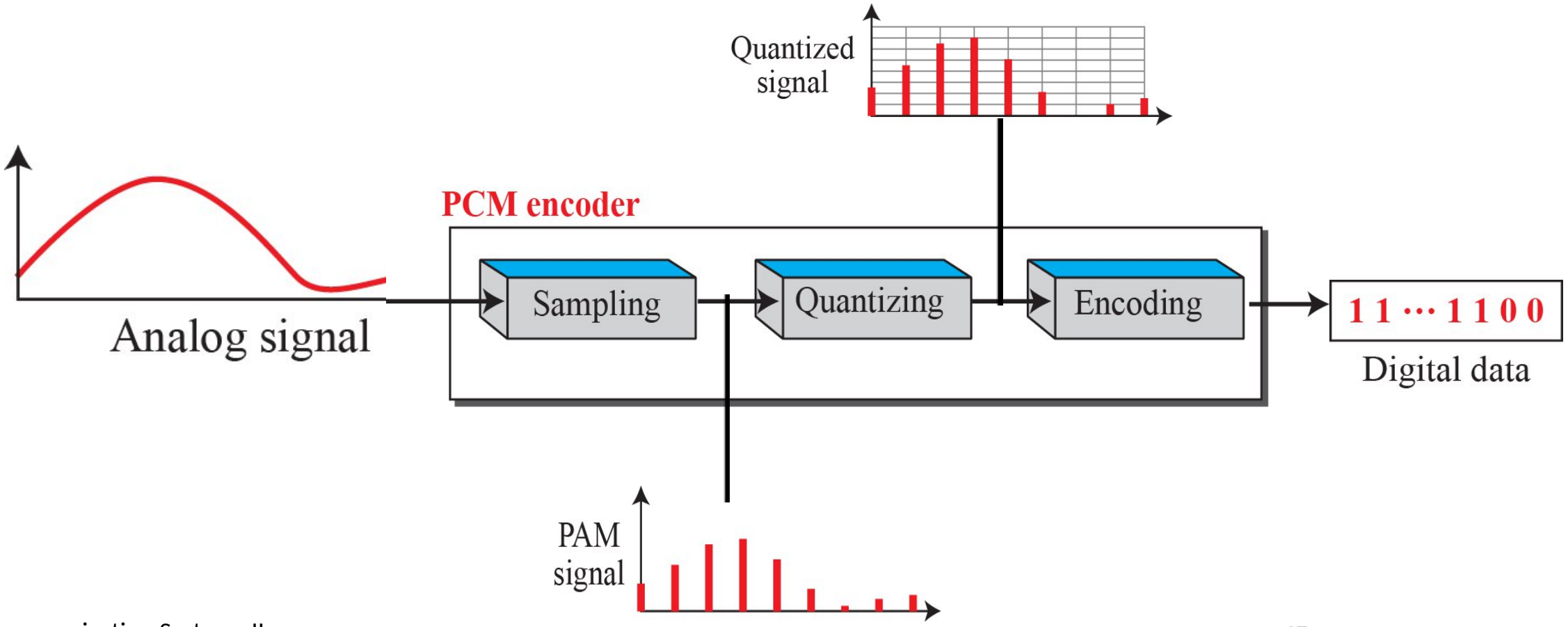
Advantages of PCM

- Inexpensive digital circuitry may be used in the system.
- Further digital signal processing such as *encryption* is possible.
- Errors may be minimized by appropriate coding of the signals.
- Signals may be regularly reshaped or regenerated using *repeaters* at appropriate intervals.

Components of PCM Encoder

Three main process in PCM transmission are

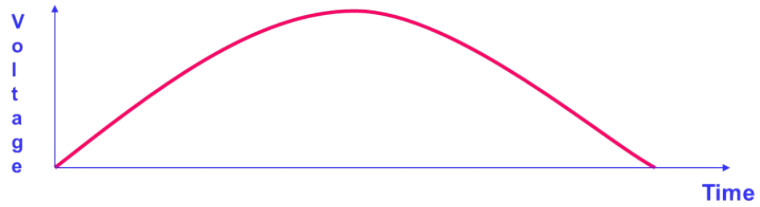
- **Sampling** :is done at higher than Nyquist rate
- **Quantization:** the sample level is rounded off to the closest allowed level (only a fixed finite number of levels are allowed)
- **Encoding:** each allowed (quantized) level is represented by a (unique) binary code word



Sampling

Process of taking samples of the analog signals at given interval of time. Only samples are being transmitted. If sufficient samples are sent and sampling theorem are met, the original signal can be re-constructed at the receiver. Usually –the flat-top PAM is the results of this step.

$$fs \geq 2f_m = 2\ \mathbf{B}$$

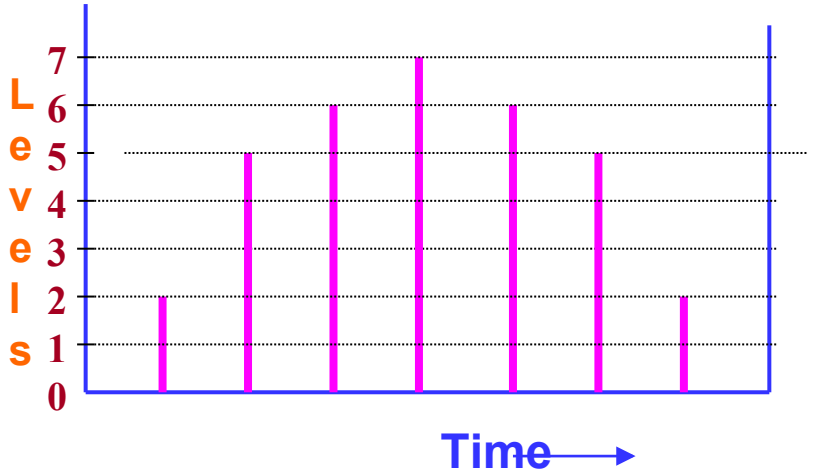


Analog-to-digital conversion needs always sampling of the data

Quantization

- Quantization is a process of assigning the analog signal samples to a pre-determined discrete levels.
- The number of quantization levels, ***L*** depends on the number of bits per sample, ***n***, used to code the signal where

$$L = 2^n$$



The above quantized signal instead of having any value between 0 to 7V, will have only discrete values ex. 0,1,2,3,4V etc.

The magnitude of the minimum step size of the quantization levels is called resolution ***Δv.*** The resolution depends on the maximum voltage, V_{max} and the minimum voltage, V_{min} of the information signal, where

$$\Delta v = \frac{V_{max} - V_{min}}{L}$$

Quantization error or quantization noise is the distortion introduced during the quantization process when the modulating signal is not an exact value of the quantization level.

✦ **The maximum quantization error**,

$$Qe = \frac{\Delta v}{2}$$

✦ Quantization error can be reduced **by increasing the number of quantization levels**, but **this will increase the bandwidth required**.

Encoding

✦ In this process, the samples that has been divided into various levels is coded into respective codes where the samples that are the same number of level are coded into the same code.

$$n = log_2 \ L$$

L = quantization level and n = Number of bits

Signal to Quantization noise ratio (SNR or SQNR) for PCM

Each additional coding bit, which increases the number of quantize levels (L), decrease the quantization error, and increases SNR. The relation between the SNR and Number of bits is given by

$$\mathbf{SNR_{dB} = 20log10(L) + 1.76 = 6.02(}n\mathbf{) + 1.76}$$

Example : What is the SNR_{dB} of 8- quantization level code word?

Solution

$n = log_2 \ L$ =3 bits per sample, so

SNR_{dB} = 6.02(3) + 1.76 = 19.82 dB.

Note that increasing the number of levels increases the SNR.