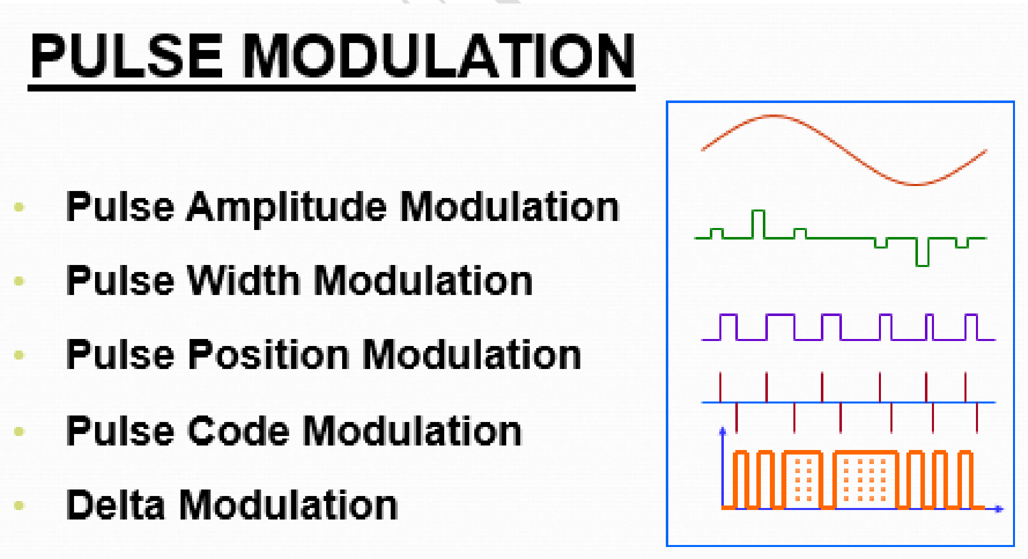
Analog and Digital Pulse Modulation Techniques

Need?

- Many Signals in Modern Communication Systems are digital
- Also, analog signals are transmitted digitally.
- Reduced distortion and improvement in signal to noise ratios.
- PAM , PWM , PPM , PCM and DM.

- In CW modulation schemes some parameter of modulated wave varies continuously with message.
- In Analog pulse modulation some parameter of each pulse is modulated by a particular sample value of the message.
- Pulse modulation of two types
    - Analog Pulse Modulation
        - Pulse Amplitude Modulation (PAM)
        - Pulse width Modulation (PWM)
        - Pulse Position Modulation (PPM)
    - Digital Pulse Modulation
        - Pulse code Modulation (PCM)
        - Delta Modulation (DM)



- PAM: In this scheme high frequency carrier (pulse) is varied in accordance with sampled value of message signal.

- PWM: In this width of carrier pulses are varied in accordance with sampled values of message signal.

  Example: Speed control of DC Motors.

PPM: In this scheme position of high frequency carrier pulse is changed in accordance with the sampled values of message signal.

PCM

Three steps

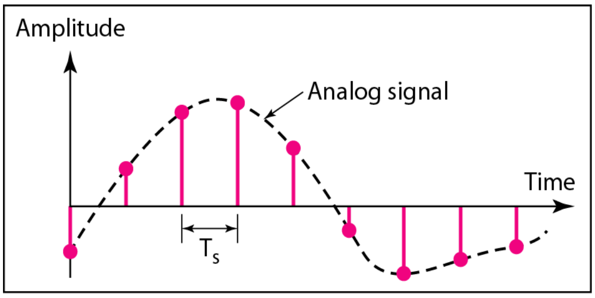
- Sampling
- Quantization
- Binary encoding

Before sampling the signal is filtered to limit bandwidth.

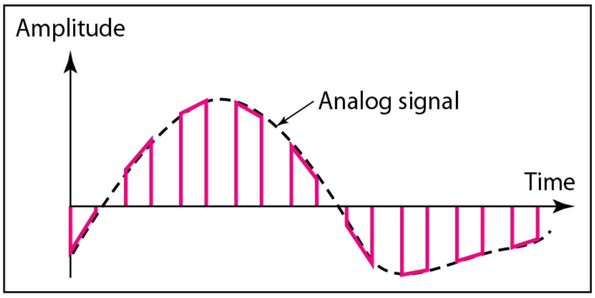
Sampling:

- Process of converting analog signal into discrete signal.
- Sampling is common in all pulse modulation techniques
- The signal is sampled at regular intervals such that each sample is proportional to amplitude of signal at that instant
- Analog signal is sampled every T_s $Secs$, called sampling interval. $f_s = {}^{1}/_{T_S}$ is called sampling rate or sampling frequency.
- $f_s = 2f_m$ is Min. sampling rate called **Nyquist rate.** Sampled spectrum $G(\omega)$ is repeating periodically without overlapping.
- Original spectrum is centered at $\omega = 0$ and having bandwidth of ω_m. Spectrum can be recovered by passing through low pass filter with cut-off ω_m.
- For $f_s < 2f_m$ sampled spectrum will overlap and cannot be recovered back. This is called **aliasing.**

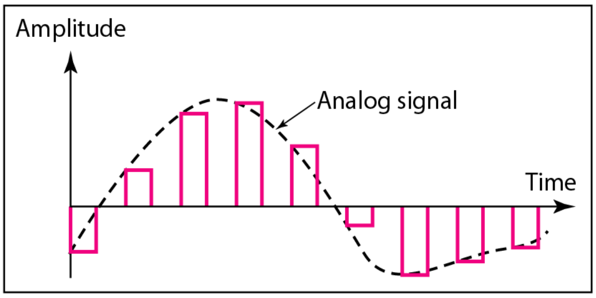
- Sampling methods:
  - Ideal – An impulse at each sampling instant.
  - Natural – A pulse of Short width with varying amplitude.
  - Flat Top – Uses sample and hold, like natural but with single amplitude value.



a. Ideal sampling



b. Natural sampling



c. Flat-top sampling

Sampling of band-pass Signals:

A band-pass signal of bandwidth $2f_m$ can be completely recovered from its samples.

Min. sampling rate $= 2 \times Bandwidth$

$$= 2 \times 2f_m = 4f_m$$

Range of minimum sampling frequencies is in the range of $2 \times BW \;\; to \; 4 \times BW$

Instantaneous Sampling or Impulse Sampling:

Sampling function is train of spectrum remains constant impulses throughout frequency range. It is not practical.

Natural sampling:

The spectrum is weighted by a **sinc** function.
Amplitude of high frequency components reduces.

Flat top sampling:

Here top of the samples remains constant.

In the spectrum high frequency components are attenuated due sinc pulse roll off. This is known as **Aperture effect**

If pulse width increases aperture effect is more i.e. more attenuation of high frequency components.

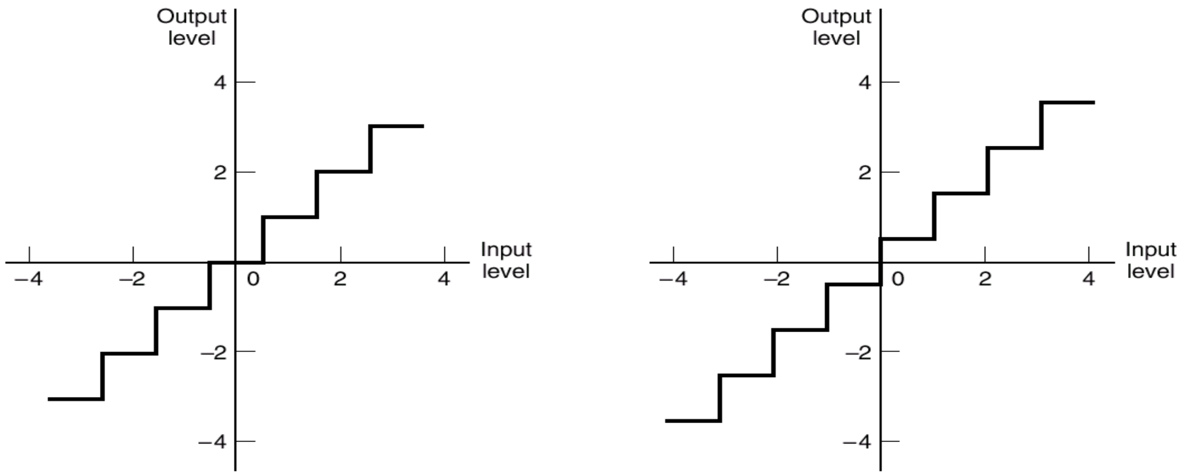
Quantization:

- Sampling results in series of pulse of varying amplitude between two limits.
- The amplitude values are infinite between two limits, we map these to finite set of values.
- This is achieved by dividing the distance between min and max into L zones each of height Δ

$$\Delta = {(max - min)}/{L}$$

Quantization Levels

- The midpoint of each zone is assigned a value from 0 to L-1 (resulting in L values)
- Each sample falling in a zone is then approximated to the value of the midpoint.



(a) (b)

Assigning Codes to Zones

- Each zone is then assigned a binary code.
- The number of bits required to encode the zones, or the number of bits per sample as follows:

$$n_b = log_2 L$$

- Say, $n_b = 3$
- The 8 zone (or level) codes are therefore: 000, 001, 010, 011, 100, 101, 110, and 111
- Assigning codes to zones:
  - 000 will refer to zone -20 to -15
  - 001 to zone -15 to -10, etc

Quantization Error

- When a signal is quantized, we introduce an error – the coded signal is an approximation of the actual amplitude value.
- The difference between actual and coded value (midpoint) is referred to as the quantization error.
- BUT, the more zones the more bits required to encode the samples so higher bit rate

Quantization Error and SQNR

- Signals with lower amplitude values will suffer more from quantization error as the error $\Delta/2$ is fixed for all signal levels.
- Non -linear quantization is used to alleviate this problem. Goal is to keep SN_QR fixed for all sample values.
- Two approaches:
  - The quantization levels follow a logarithmic curve. Smaller Δ's at lower amplitudes and larger Δ's at higher amplitudes.
  - **Companding:** The logarithmic zone, and then expanded at the receiver. The zones are fixed in height.

Bit rate and bandwidth requirements of PCM

- The bit rate of a PCM signal can be calculated form the number of bits per sample × the sampling rate. Bit rate $= n_b \times f_s$
- The bandwidth required to transmit this signal depends on the type of line encoding used.

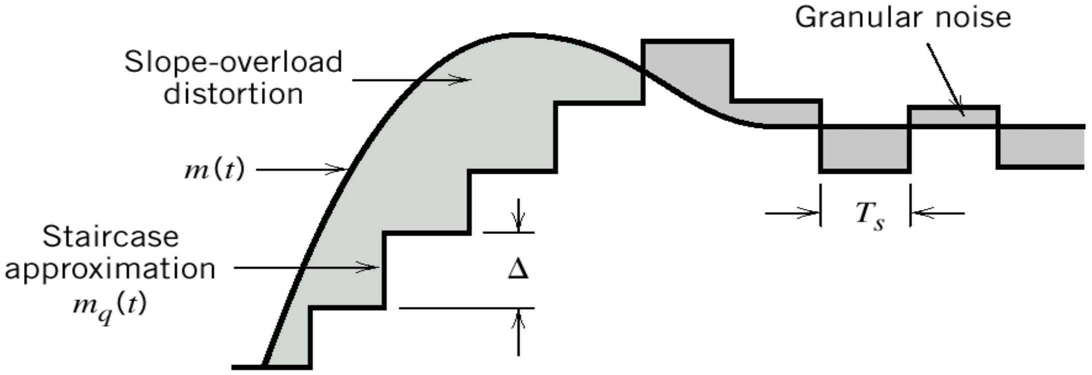
- A digitized signal will always need more bandwidth than the original analog signal. Price we pay for robustness and other features of digital transmission.

Important Relations

- Quantization Noise $\left(N_q\right) = \frac{\Delta^2}{12}$
- Signal to Noise ratio
  $(SQNR) = \frac{3}{2}.2^{2n}\ \ or\ \ SQNR\ in\ dB = 1.76 + 6.02n \cong (1.8 + 6n)dB$
- $Bit\ rate = No.of\ bits\ per\ sample \times sampling\ rate = nfs$
  Bandwidth for PCM signal = n.fm
  Where, n – No. of bits in PCM code
  Fm – signal bandwidth
  fs – sampling rate

Delta Modulation

- The present sample is compared with previous sample value and 1/0 is transmitted if it is greater/less than the previous sample value.
- Bandwidth requirement of DM is less on compared to PCM.
- DM needs simple circuity compared to PCM
- Quantization error is more.
- Drawbacks are
  - Slope overload – Magnitude of slope is greater than slope of staircase
  - Granular Noise – Signal variations with in step size
- In ADM step size is made adaptive to take care of above problems.
- Delta PDM: The difference between two successive samples is quantized, encoded and transmitted. Useful in voice transmission.



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Two types of quantization errors:
Slope overload distortion and **granular noise**

- So need for shaping binary data
- Line coding converts binary sequence into digital signal format which is more convenient for transmission over cable or other medium.
- It maximizes bit rate, reduces power of transmission and reduces dc component.

Various line code formats

RZ, NRZ, AMI, Manchester etc.

Unipolar NRZ: Requires only one power supply. It has DC value.

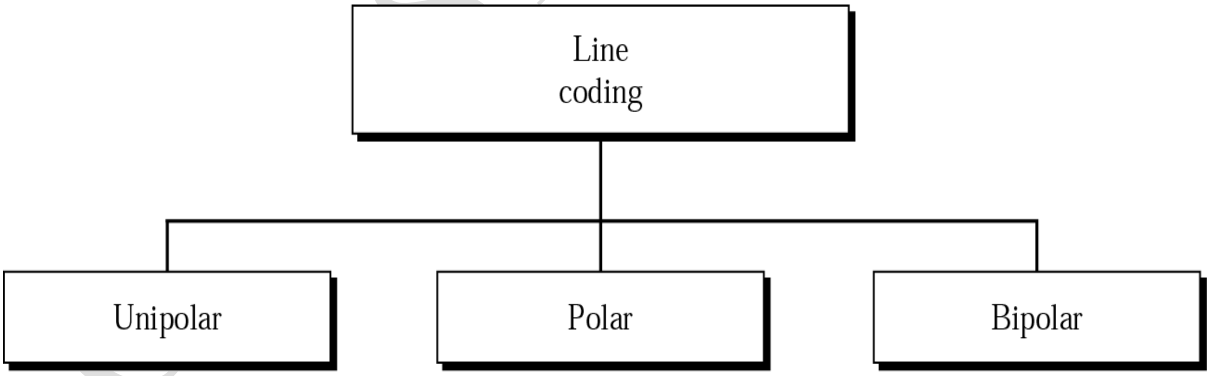
Polar NRZ: Both –ve & +ve power supply required.

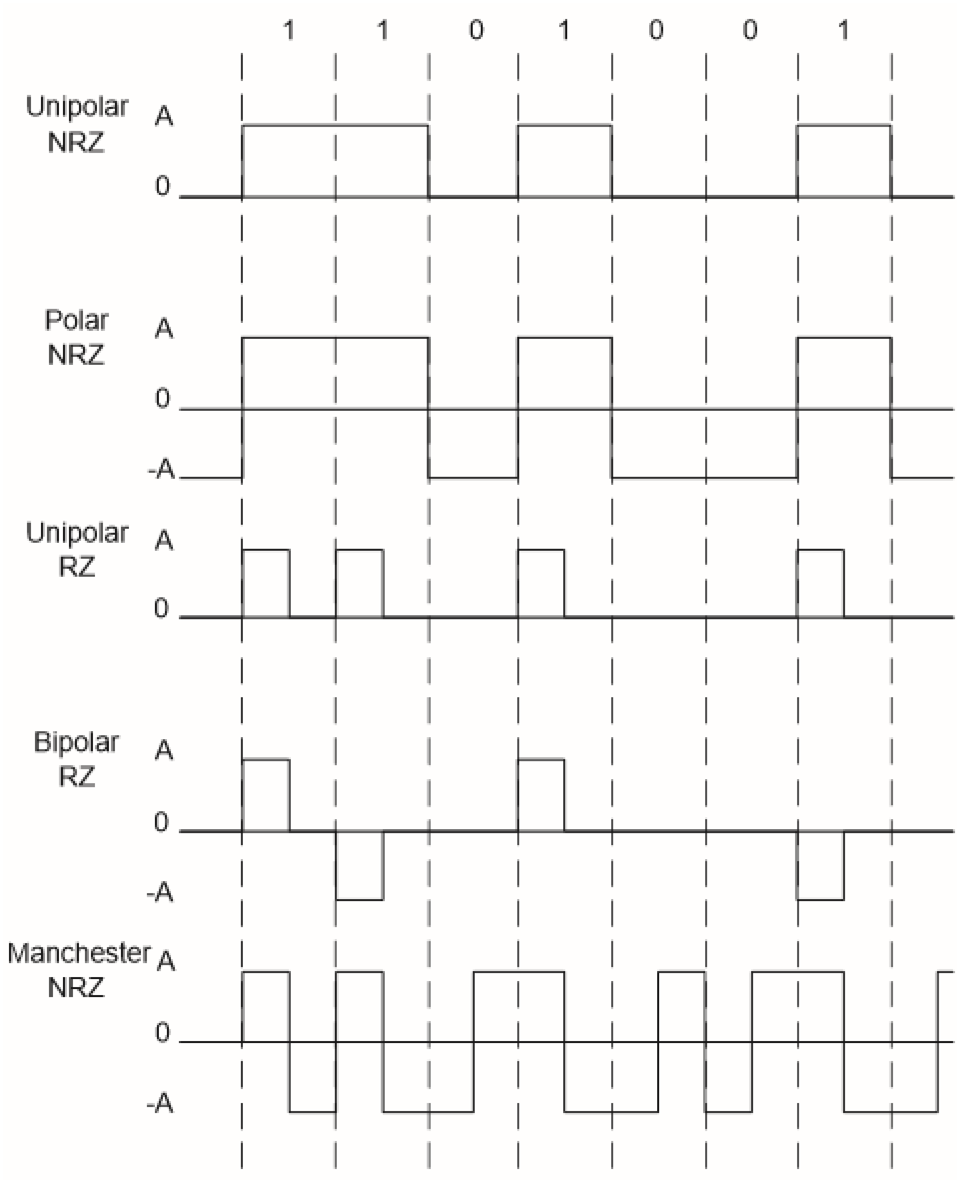
Bipolar: Binary 1, as alternate positive and negative value. Binary 0 by 0 level also called alternate mark inversion (AMI)

Manchester: Called split phase encoding

No DC voltage

Twice the BW of unipolar NRZ or polar NRZ (pulses) are half the width




1 1 0 1 0 0 1
Unipolar NRZ
A
0
Polar NRZ
A
0
-A
Unipolar RZ
A
0
Bipolar RZ
A
0
-A
Manchester NRZ
A
0
-A