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 \( \omega_m \). Spectrum can be recovered by passing through low pass filter with cut-off \( \omega_m \).
• 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.
Sampling of band-pass Signals:

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

Min. sampling rate $= 2 \times \text{Bandwidth}$

$= 2 \times 2f_m = 4f_m$

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 $\text{sinc}$ 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 $\Delta$
  \[ \Delta = \frac{max - min}{L} \]

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.

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 \( \text{SNR}_{\text{Q}} \) fixed for all sample values.
- Two approaches:
  - The quantization levels follow a logarithmic curve. Smaller \( \Delta \)'s at lower amplitudes and larger \( \Delta \)'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 from the number of bits per sample \( \times \) the sampling rate. \( \text{Bit rate} = n_b \times f_s \)
- 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 \( (N_q) = \frac{\Delta^2}{12} \)
• Signal to Noise ratio \( (SQNR) = \frac{3}{2} \cdot 2^{2n} \) or \( SQNR \text{ in } dB = 1.76 + 6.02n \equiv (1.8 + 6n)dB \)
• Bit rate = No. of bits per sample \( \times \) sampling rate = \( n \cdot f_s \)
  Bandwidth for PCM signal = \( n \cdot f_m \)

Where,
- \( n \) – No. of bits in PCM code
- \( f_m \) – signal bandwidth
- \( f_s \) – 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 within 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.
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

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

Polar NRZ

Unipolar RZ

Bipolar RZ

Manchester NRZ