# ANALOG-TO-DIGITAL CONVERSION

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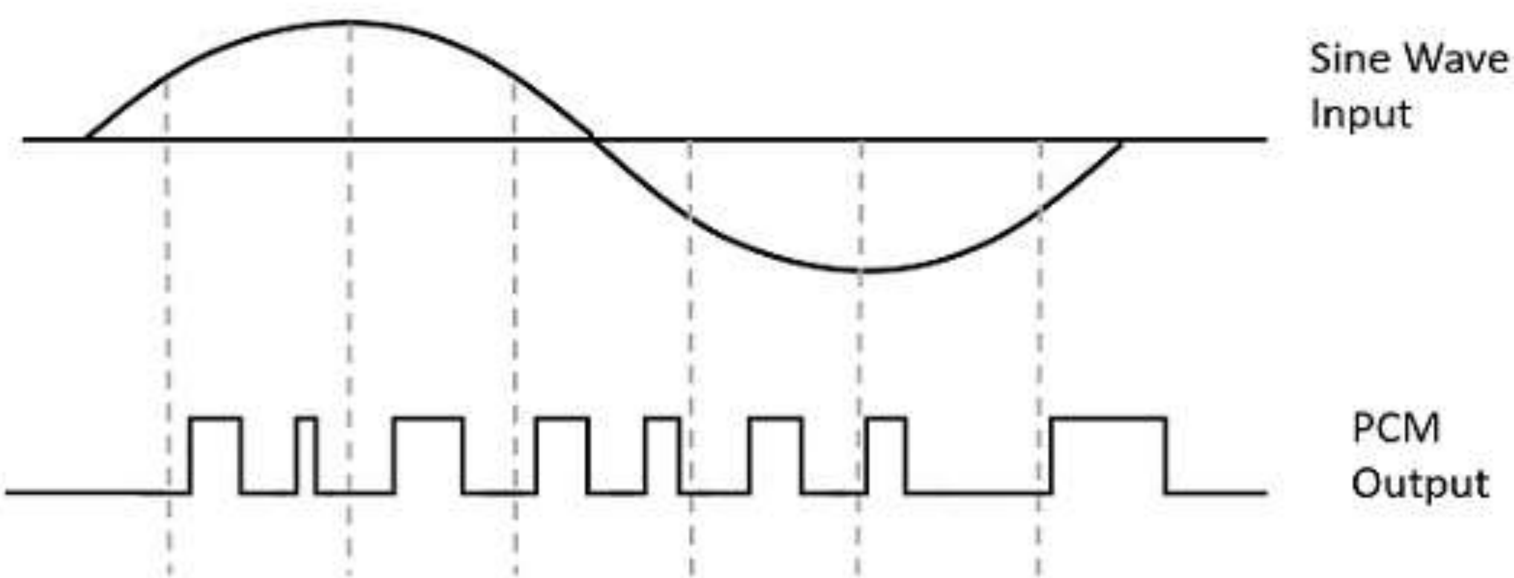
- Two techniques are there to convert the analog signal to digital signal

1. Pulse code modulation
2. Delta modulation.

# Pulse Code Modulation (PCM)

- The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
- A signal is pulse code modulated to convert its analog information into a binary sequence, i.e., **1s** and **0s**. The output of a PCM will resemble a binary sequence.

# Pulse Code Modulation (PCM)

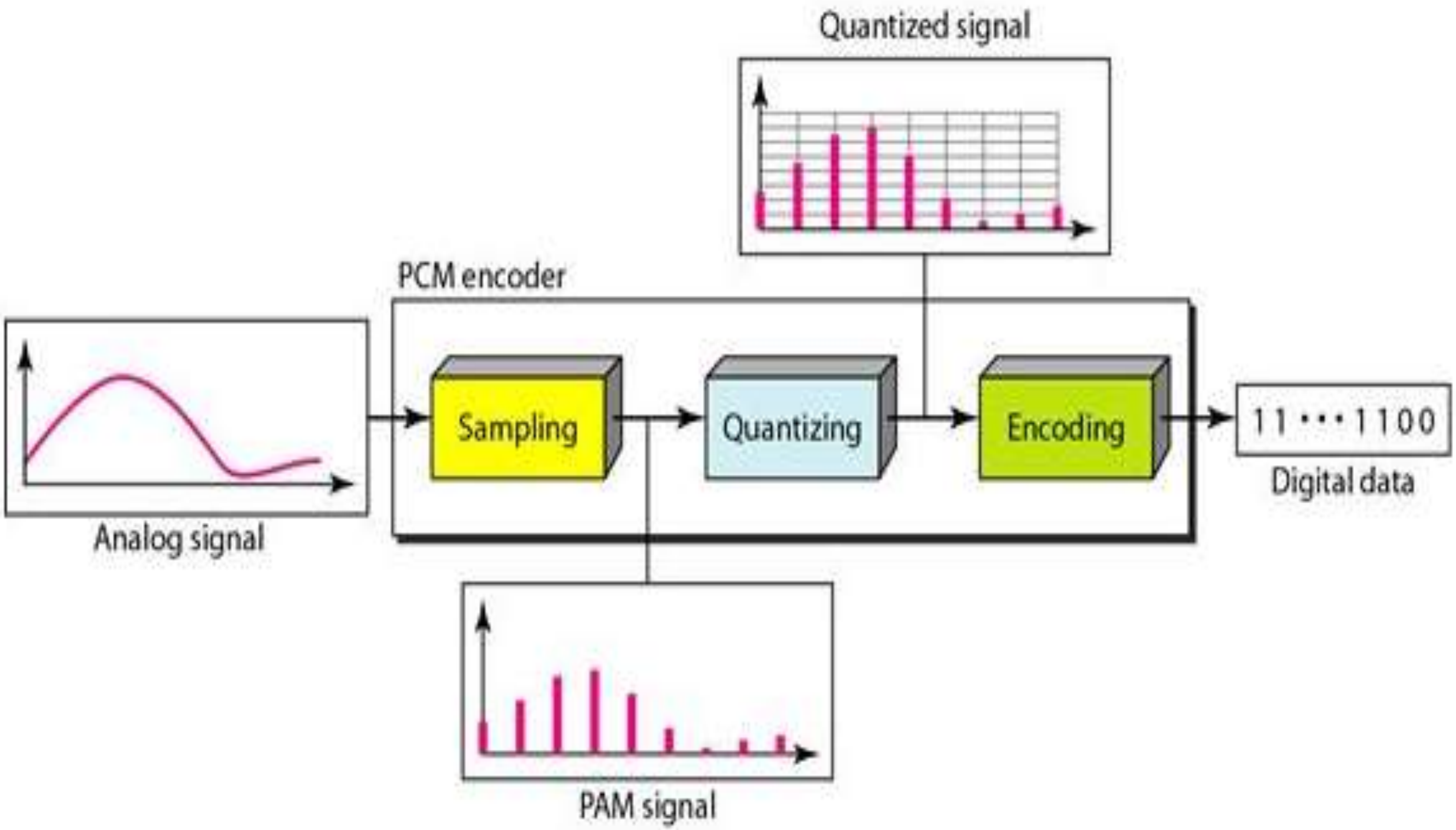


# Pulse Code Modulation (PCM)

- A PCM encoder has three processes

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits

# Pulse Code Modulation (PCM)



# Sampling

- The first step in PCM is sampling.
- The analog signal is sampled every Ts s, where Ts is the sample interval or period.
- The inverse of the sampling interval is called the sampling rate or sampling frequency and denoted by ƒs, Where ƒs = 1/ Ts.

# Sampling

- There are three sampling methods-

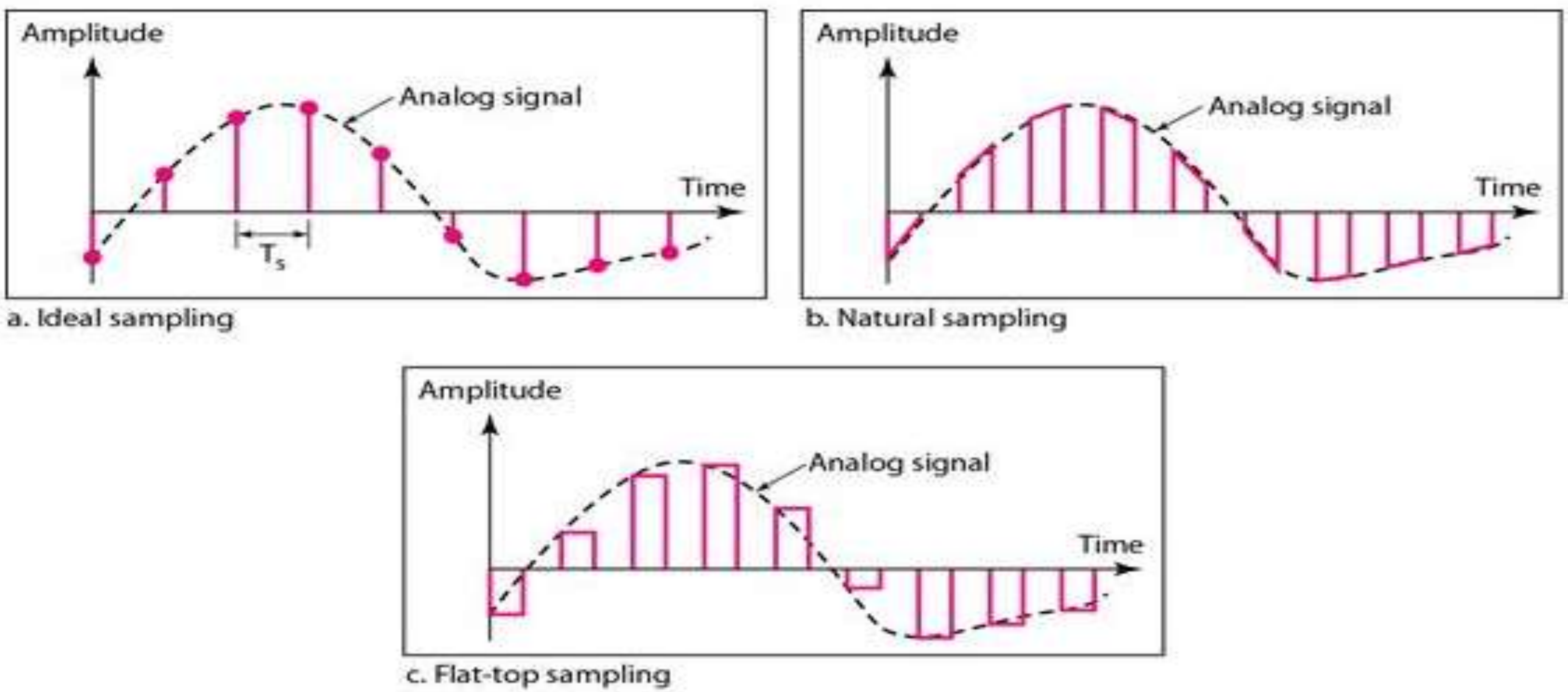
1. Ideal
2. Natural
3. Flat-top.

# Sampling

- In ideal sampling, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented.
- In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs.
- The result is a sequence of samples that retains the shape of the analog signal. The most common sampling method, called sample and hold, however, creates flat-top samples by using a circuit.

# Sampling

- The sampling process is sometimes referred to as pulse amplitude modulation (PAM).



# *Sampling Rate*

- One important consideration is the sampling rate or frequency.
- What are the restrictions on Ts?
- According to the Nyquist theorem, to reproduce the original analog signal, one necessary condition is that the sampling rate be at least twice the highest frequency in the original signal.

# *Sampling Rate*

- As for this Theorem, First, we can sample a signal only if the signal is band-limited i.e a signal with an infinite bandwidth cannot be sampled.
- Second, the sampling rate must be at least 2 times the highest frequency, not the bandwidth.
- If the analog signal is low-pass, the bandwidth and the highest frequency are the same value.
- If the analog signal is bandpass, the bandwidth value is lower than the value of the maximum frequency.

# *Quantization*

- The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal.
- The following are the steps in quantization:

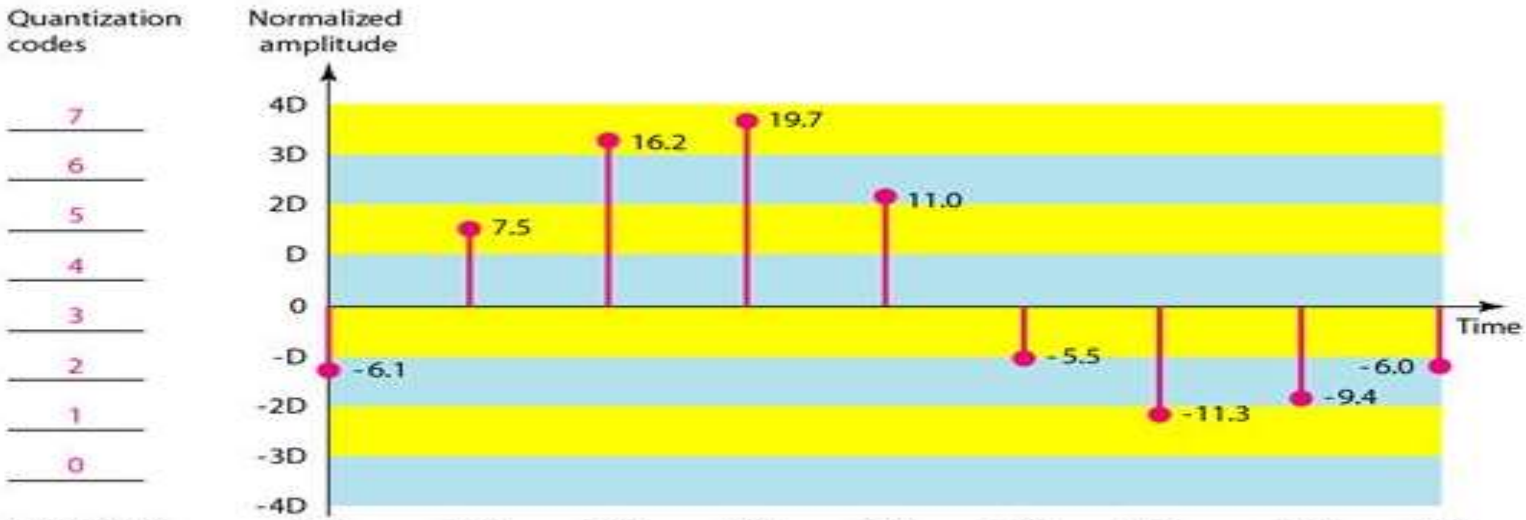
1. We assume that the original analog signal has instantaneous amplitudes between Vmin and Vmax

2. We divide the range into L zones, each of height Δ (delta).

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

3. We assign quantized values of 0 to L - I to the midpoint of each zone.
4. We approximate the value of the sample amplitude to the quantized values.

# *Quantization*




| | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|
| Normalized PAM values | -1.22 | 1.50 | 3.24 | 3.94 | 2.20 | -1.10 | -2.26 | -1.88 | -1.20 |
| Normalized quantized values | -1.50 | 1.50 | 3.50 | 3.50 | 2.50 | -1.50 | -2.50 | -1.50 | -1.50 |
| Normalized error | -0.38 | 0 | +0.26 | -0.44 | +0.30 | -0.40 | -0.24 | +0.38 | -0.30 |
| Quantization code | 2 | 5 | 7 | 7 | 6 | 2 | 1 | 2 | 2 |
| Encoded words | 010 | 101 | 111 | 111 | 110 | 010 | 001 | 010 | 010 |

# *Quantization*

- Assume that we have a sampled signal and the sample amplitudes are between -20 and +20 V.
- We decide to have eight levels (L = 8). This means that Δ =5 V.
- We have shown only nine samples using ideal sampling (for simplicity). The value at the top of each sample in the graph shows the actual amplitude. In the chart, the first row is the normalized value for each sample (actual amplitude/Δ).

# *Quantization*

- The quantization process selects the quantization value from the middle of each zone.
- This means that the normalized quantized values (second row) are different from the normalized amplitudes.
- The difference is called the normalized error (third row).
- The fourth row is the quantization code for each sample based on the quantization levels at the left of the graph.
- The encoded words (fifth row) are the final products of the conversion.

# *Quantization Levels*

- The choice of L, the number of levels, depends on the range of the amplitudes of the analog signal and how accurately we need to recover the signal.
- If the amplitude of a signal fluctuates between two values only, we need only two levels; if the signal, like voice, has many amplitude values, we need more quantization levels.
- In audio digitizing, L is normally chosen to be 256; in video it is normally thousands.
- Choosing lower values of L increases the quantization error if there is a lot of fluctuation in the signal.

# *Quantization Error*

- One important issue is the error created in the quantization process.
- Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values.
- The output values are chosen to be the middle value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error.
- In the example discussed, the normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26. The value of the error for any sample is less than $\Delta/2$. In other words, we have $\Delta/2 <= \text{error} <= \Delta/2$.

# *Uniform Versus Non uniform Quantization*

- For many applications, the distribution of the instantaneous amplitudes in the analog signal is not uniform.
- Changes in amplitude often occur more frequently in the lower amplitudes than in the higher ones.
- For these types of applications it is better to use nonuniform zones.
- In other words, the height of $\Delta$ is not fixed; it is greater near the lower amplitudes and less near the higher amplitudes.

# *Uniform Versus Non uniform Quantization*

- Nonuniform quantization can also be achieved by using a process called companding and expanding.
- The signal is companded at the sender before conversion; it is expanded at the receiver after conversion.
- Companding means reducing the instantaneous voltage amplitude for large values; expanding is the opposite process.
- Companding gives greater weight to strong signals and less weight to weak ones.
- It has been proved that nonuniform quantization effectively reduces the SNRdB of quantization.

# *Encoding*

- The last step in PCM is encoding. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an nb-bit code word.
- In the above figure the encoded words are shown in the last row. A quantization code of 2 is encoded as 010; 5 is encoded as 101; and so on.
- Note that the number of bits for each sample is determined from the number of quantization levels. If the number of quantization levels is L, the number of bits is nb=log2 L.
- In our example L is 8 and nb is therefore 3. The bit rate can be found from the formula.
- Bitrate = Sampling rate X Number of bites per sample= ʃs X nb.