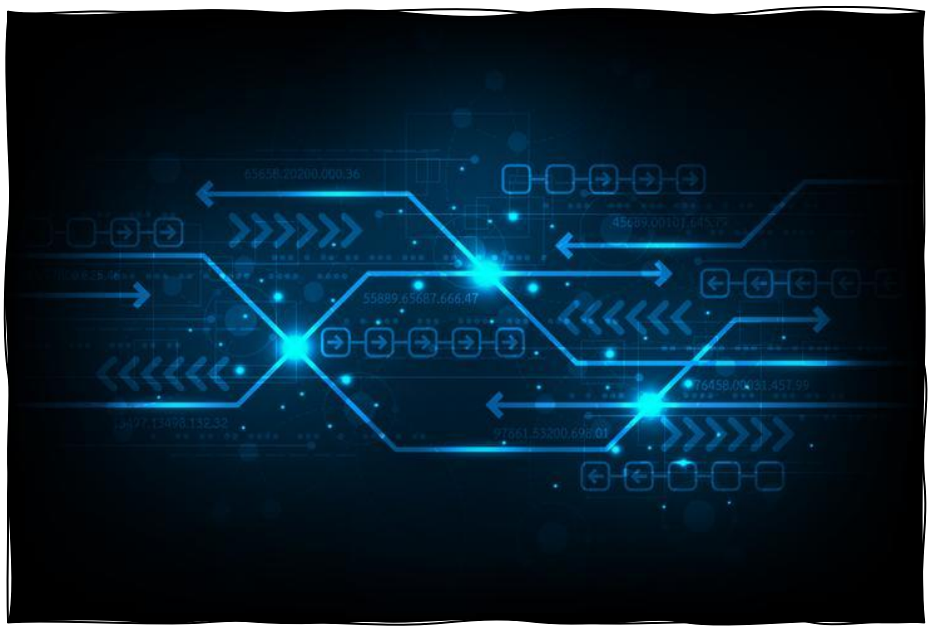
Al-Mustaqbal University
Department of Medical Instrumentation Techniques Engineering
Class: Third
Subject: Medical Communication Systems
Lecturer: Prof. Dr. Bayan Mahdi Sabbar & M.Sc. Huda Wasfi Hassoon
Lecture:11

Lecture 11

Digital Pulse Modulation



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Digital Pulse Modulation

In modern digital communication systems, efficient signal transmission and data integrity are essential for high-quality communication. One of the key techniques used to achieve this is Pulse Digital Modulation, which involves converting analog signals into digital form for reliable transmission over digital networks. This method ensures improved resistance to noise, minimal signal degradation, and efficient bandwidth utilization, making it a fundamental aspect of modern telecommunications.

Pulse digital modulation techniques allow analog signals, such as voice, video, and sensor data, to be represented in a discrete digital format. Unlike traditional analog transmission, digital modulation enhances the quality and security of transmitted signals by encoding them into binary values (**0s and 1s**). This makes it possible to efficiently transmit information over fiber-optic networks, satellite communications, and wireless systems with minimal distortion.

There are several key types of pulse digital modulation, one of them:

Pulse Code Modulation (PCM) – Converts an analog signal into a digital sequence using sampling, quantization, and encoding.

Pulse Code Modulation (PCM)

In modern communication systems, the accurate transmission of signals is crucial for ensuring high-quality data transfer. One of the most widely used techniques for converting analog signals into digital form is **Pulse Code Modulation (PCM)**. This method plays a fundamental role in digital signal

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processing, allowing analog signals, such as voice, music, or sensor data, to be efficiently transmitted, stored, and processed in digital communication systems.

PCM works by representing continuous analog signals using discrete digital values. Unlike analog transmission, which is susceptible to noise, distortion, and signal degradation over long distances, PCM ensures a more reliable and interference-resistant communication process. It does this by converting the analog waveform into a series of binary numbers, consisting of only two states: **high (1) and low (0)**. This transformation enables data to be transmitted accurately over digital networks, such as fiber-optic cables, wireless communication systems, and computer networks.

The PCM process consists of three essential steps:

1. **Sampling** – The continuous analog signal is measured at fixed time intervals to capture key data points.
2. **Quantization** – Each sampled value is approximated to the closest available discrete level, reducing infinite possibilities into a finite set of values.
3. **Encoding** – The quantized values are converted into a binary format, making them suitable for digital storage and transmission.

Due to its ability to maintain signal integrity and resist noise interference, PCM is extensively used in various fields, including **telecommunications, audio recording, radar systems, and medical imaging**. From traditional telephone networks to modern digital audio formats like CDs and VoIP, PCM has been a cornerstone of digital communication technology, enabling efficient and high-fidelity signal transmission.

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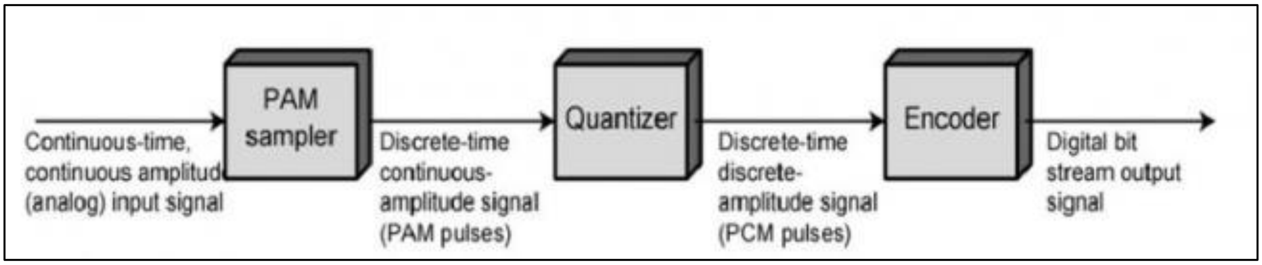

Figure 1: Block Diagram of Pulse Code Modulation.

The block diagram of the Pulse Code Modulation (PCM) process

The block diagram of the Pulse Code Modulation (PCM) process, which is used to convert analog signals into digital signals for efficient transmission over digital communication networks. The diagram consists of three main sections:

1. **Transmitter**
2. **Transmission Path**
3. **Receiver**

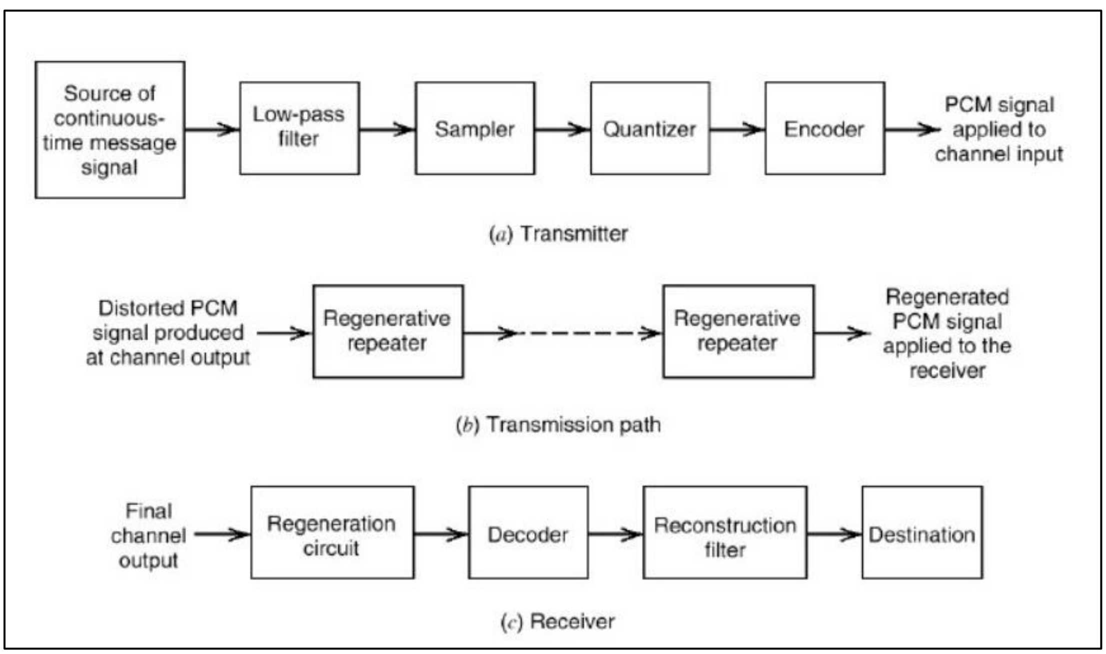

Figure 2: Basic Elements of Pulse Code Modulation System.

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A PCM GENERATION OR TRANSMITTER:

In a PCM generation shown in figure 3, the signal x(t) is first passed through the low-pass filter of cutoff frequency f_m Hz. This low-pass filter blocks all the frequency components that are lying above f_m Hz.

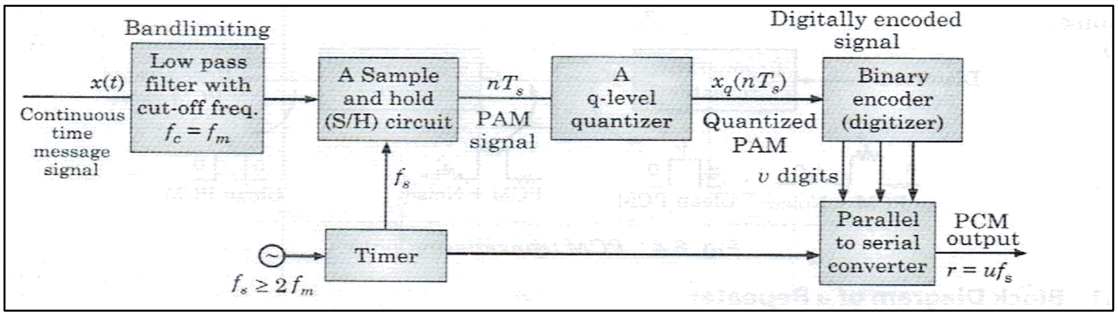

Figure 3: Practical PCM generation.

- Band-**limiting the Signal**
  - The original analog signal, represented as $x(t)$, is first **band-limited** to a maximum frequency f_m Hz.
  - This ensures that the signal does not contain frequency components higher than f_m , which is crucial for accurate sampling.
- Sampling **the Signal**
  - The **sample-and-hold circuit** takes periodic samples of the band-limited signal at a **sampling frequency** f_s.
  - To avoid aliasing (overlapping of frequency components), the sampling frequency must be **at least twice the highest frequency** in the signal, following Nyquist’s theorem: $f_s \geq 2f_m$
  - The result is a series of discrete-time samples, denoted as $x(nT_s)$, where $T_s = {}^{1}/_{f_s}$ is the sampling period.

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- Holding **the Sampled Value**
  - The sampled signal is held constant for a short duration before being processed further.
  - This ensures that each sample can be properly quantized and encoded without fluctuations.
- Quantization **Process**
  - The sampled values $x(nT_s)$ are **mapped to the nearest available digital level** from a predefined set of discrete levels.
  - Since an analog signal has infinite possible values, but a digital system can only represent a limited number of levels, the quantization process introduces a small difference between the actual sample value and the nearest digital level.
  - This difference is known as **quantization error** or **quantization noise**.
  - The output of the quantizer is denoted as $x_q(nT_s)$, which is now a **digitally represented signal**.
- Binary **Encoding**
  - The quantized values are $x_q(nT_s)$ then **converted into binary form** using a **binary encoder**.
  - Each quantized level is assigned a unique **binary word** consisting of ‘v’ bits.
  - This process ensures that the signal is now completely in digital form, suitable for transmission and storage.
  - The encoder used in this step is also known as a **digitizer**.
- Parallel**-to-Serial Conversion**

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- The binary words generated for each sample contain multiple bits, but it is inefficient to transmit each bit separately.
- Instead, a **parallel-to-serial converter** is used to transform the multiple-bit parallel data into a **serial bit stream**.
- This is commonly done using a **shift register**, which sequentially outputs the binary bits as a continuous stream.

- Generation **of the PCM Signal**
  - The final output of this process is a **PCM baseband signal**, consisting of a continuous stream of binary bits.
  - This digital signal is now ready to be transmitted over a communication channel, such as a fiber-optic cable, radio link, or telephone network.

PCM transmission path:

The channel through which the PCM signal travels from the PCM transmitter to the PCM receiver is known as the **PCM transmission path**. This path facilitates the transfer of the digital signal and is illustrated in Figure 4.

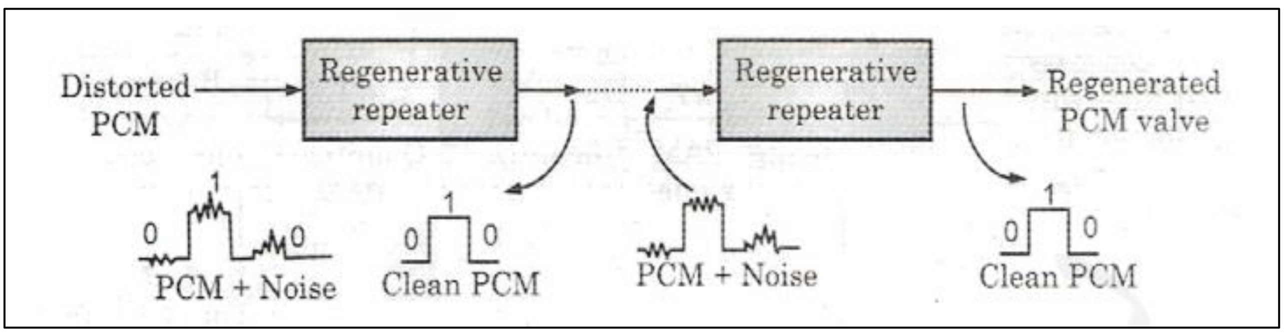

Figure 4: PCM transmission path.

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Block diagram of a repeater:

- **Amplitude Equalization**
  - The **Amplitude Equalizer** is responsible for correcting any distortions in the received PCM signal.
  - It compensates for both **amplitude and phase distortions**, which may have occurred during transmission.
  - This ensures that the signal maintains its integrity before further processing.
- **Timing Circuit Generation**
  - A **timing circuit** extracts a **periodic pulse train** from the received PCM signal.
  - This pulse train is crucial as it helps in synchronizing the receiver with the incoming data.
  - The extracted timing pulses are then sent to the next stage for precise decision-making.
- **Decision-Making Device**
  - The **Decision-Making Device** utilizes the extracted pulse train to **sample** the equalized PCM pulses.
  - Sampling is performed at specific time instances where the **Signal-to-Noise Ratio (SNR)** is at its maximum.
  - This ensures that the signal is sampled at the most optimal moments, minimizing errors caused by noise or distortion.
- **Determining Digital Values (0 or 1)**
  - At the moment of sampling, the decision device must determine whether the received **equalized PCM signal** corresponds to a **binary 0 or 1**.

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- This decision is made by comparing the received PCM signal against a **fixed reference level**, known as the **decision threshold**.
- If the signal amplitude is above the threshold, it is interpreted as **1**; otherwise, it is considered **0**.
- This threshold comparison ensures that the signal is accurately reconstructed.

- **Noise-Free PCM Output**
  - After passing through the decision-making process, the output of the device is a **clean PCM signal**.
  - At this stage, any noise or distortion present in the original received signal is effectively removed.
  - The final output is a **restored and noise-free PCM waveform**, which is ready for further processing or decoding.

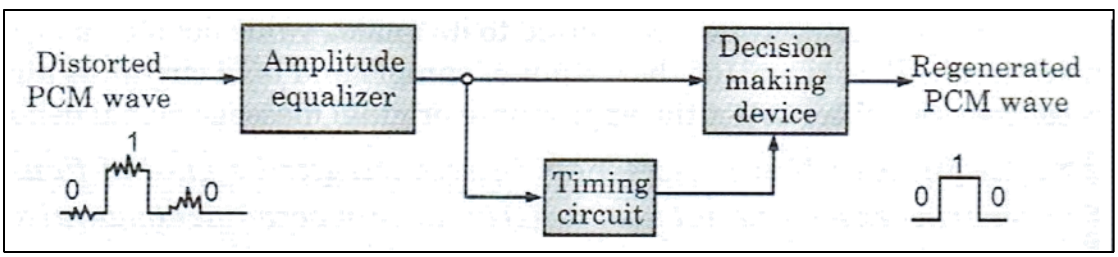

Figure 5: Block diagram of a repeater.

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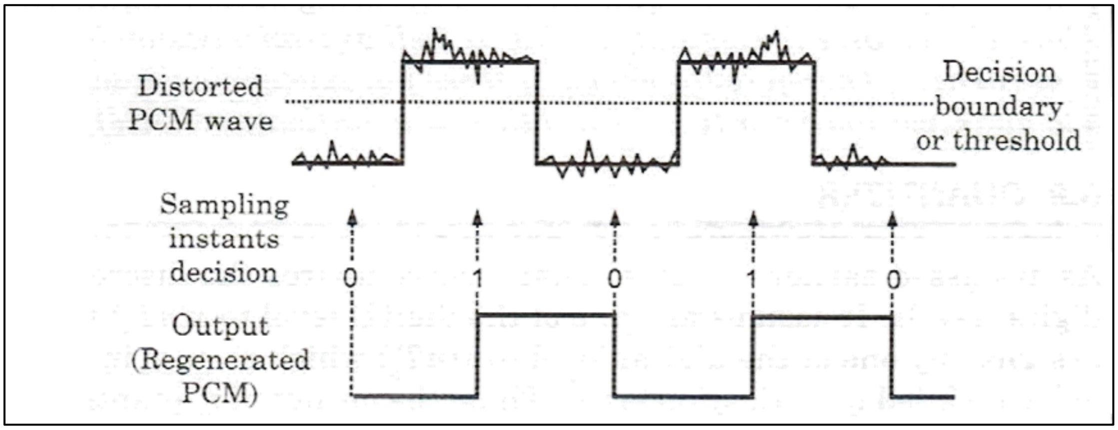


Figure 6: Waveform of Regenerated PCM.

PCM RECEIVER:

Figure 7(a) illustrates the block diagram of the PCM receiver, while Figure 7(b) shows the reconstructed signal. In the PCM receiver, the process begins with a regenerator. This regenerator's role is to reshape the received pulse signal, correcting any distortion and removing noise that may have been introduced during transmission. This ensures that the signal is as accurate as possible before further processing.

Once the pulse is regenerated, the signal is then converted into parallel digital words, where each digital word corresponds to a sample of the original signal. These digital words represent the quantized values of the continuous-time signal at specific intervals, which were taken during the sampling process.

Afterward, each of these digital words is converted back to its corresponding analog value, denoted as $x_q t$. This conversion is carried out using a sample-and-hold circuit, which takes each digital word and holds its value for a short period to allow for smooth reconstruction.

The output from the sample-and-hold circuit is still not a perfect analog signal, so it is passed through a low-pass reconstruction filter. This filter is essential for smoothing out the signal and removing any high-frequency components that might have been introduced during the quantization and sampling process. The output of this filter is the reconstructed signal, which closely matches the original message signal, denoted as y(t).

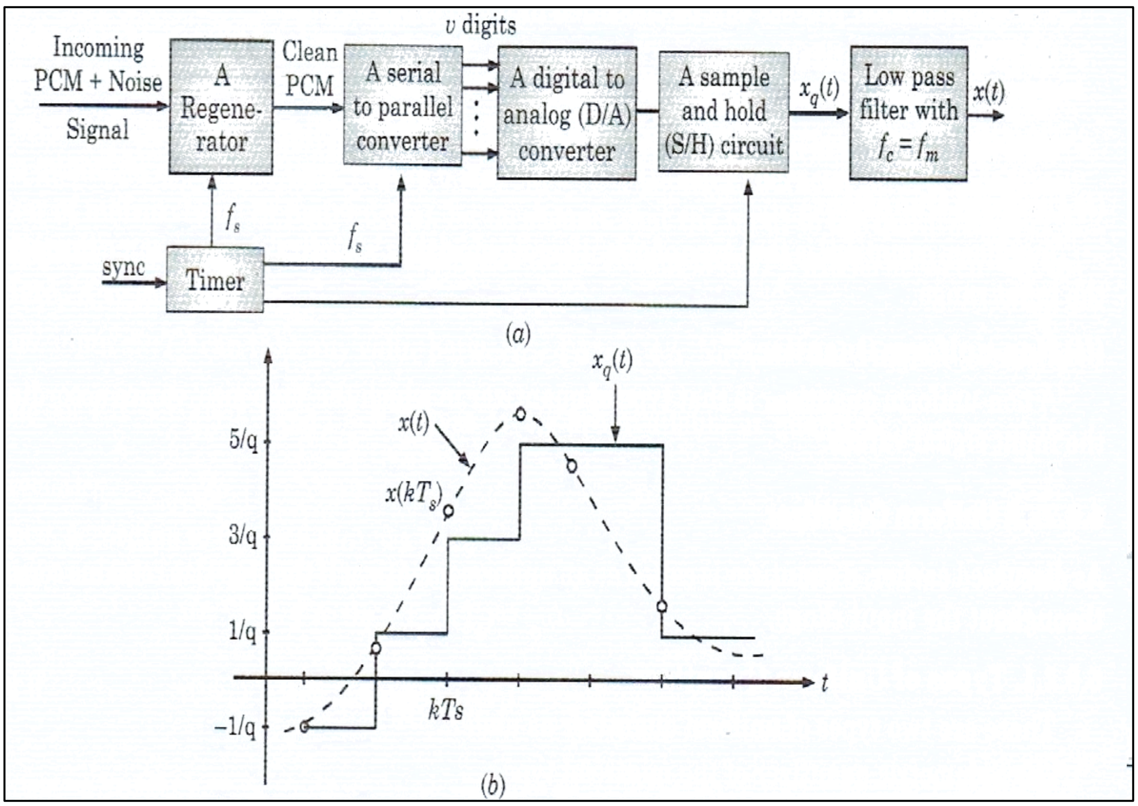


Figure 7: (a) PCM receiver, (b) the reconstructed signal.

Quantizer

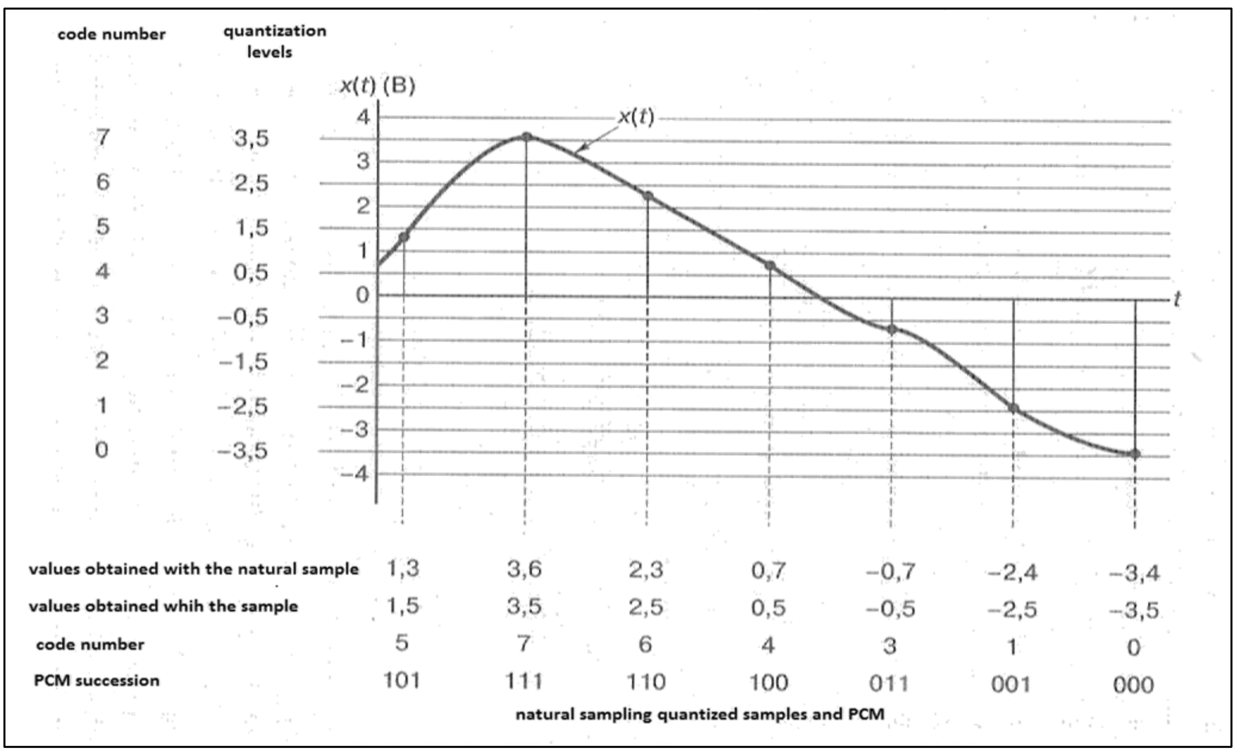
As mentioned earlier, a q-level quantizer compares the discrete-time input, $x(nT_s)$, with its predefined digital levels. It assigns the closest digital level to $x(nT_s)$, resulting in the least distortion or error. This distortion or error is referred to as **quantization error**. Therefore, the output of the quantizer is a digital level, denoted as, $x_q(nT_s)$.

Classification of the Quantization Process

The quantization process can be divided into two types:

a. **Uniform Quantization**

b. **Non-uniform Quantization**



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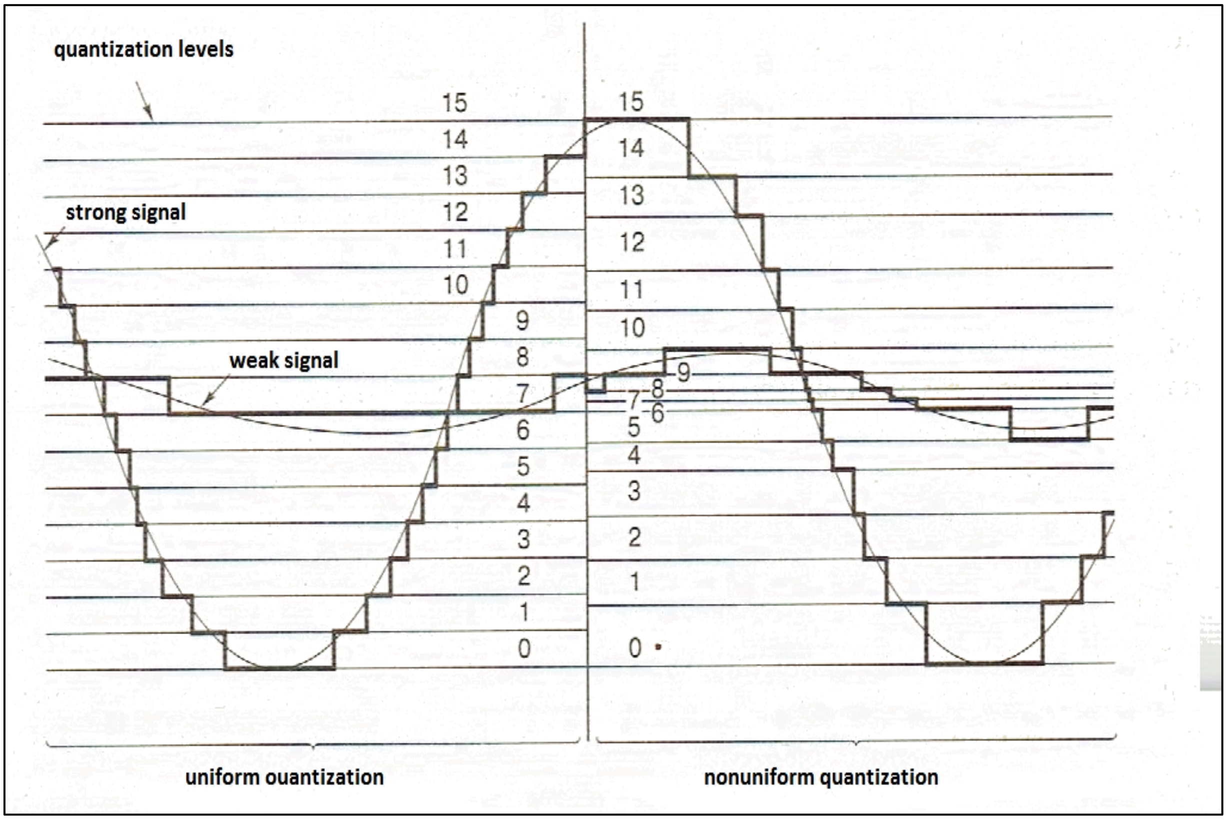

Figure 8: uniform and nonuniform quantization.

Transmission Bandwidth in a PCM System

In this section, we will evaluate the transmission bandwidth required for a **Pulse Code Modulation (PCM)** system.

Let us assume that the quantizer uses **'v'** binary digits (bits) to represent each level. The total number of quantization levels that can be represented using **'v'** bits is given by:

$$q = 2^v$$

where **'q'** represents the total number of digital levels in a **q-level quantizer**.

For example, if **v = 4** bits, the total number of levels will be:

$q = 2^4 = 16$ levels

Since each sample is converted into **'v'** binary bits, the **number of bits per sample** is: v

We also know that the **sampling rate** (number of samples per second) is: f_s

Thus, the **total number of bits per second** (or bit rate) is given by:

$$Bit\ rate = (Number\ of\ bits\ per\ sample) \times (Number\ of\ samples\ per\ second)$$

$$= v \times f_s$$

In PCM systems, the **bit rate (r)** is defined as:

$$r = vf_s$$

Since the sampling frequency must satisfy **Nyquist's criterion**:

$$f_s \geq 2f_m$$

where f_m is the maximum frequency of the input signal.

The **transmission bandwidth (BW)** required for PCM is given by **half of the bit rate**:

$$BW \geq \frac{1}{2}r$$

Substituting $r = vf_s$:

$$BW \geq \frac{1}{2}vf_s$$

Since $f_s \geq 2f_m$, we get:

$$BW \geq vf_m$$

This is the required expression for the **transmission bandwidth of a PCM system**.

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Quantization Noise/Error in PCM

In this section, we derive an expression for quantization noise (i.e., error) in a PCM system for linear or uniform quantization. Due to the quantization process, inherent errors are introduced into the signal, known as **quantization error**. The quantization error is defined as the difference between the original signal and its quantized version:

$$\varepsilon = x(nT_s) - x_q(nT_s)$$

Where:

- $x(nT_s)$ is the actual sampled signal value.
- $x_q(nT_s)$ is the quantized signal value.

Assume that the input $x(nT_s)$ to a linear or uniform quantizer has a continuous amplitude within the range $-x_{max}$ to $+x_{max}$. The total amplitude range can be expressed as:

$$Total\ Amplitude\ rang = x_{max} - (-x_{max}) = 2x_{max}$$

this total amplitude range is divided into quantization levels, the step size Δ is given by:

$$\Delta = \frac{2x_{max}}{q}$$

For a normalized signal where the amplitude is scaled to a range of -1 to 1:

$$x_{max} = 1\,, -x_{max} = -1$$

Thus, the step size simplifies to:

$\Delta = \frac{2}{q}$ ((for normalized signal)

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If the step size Δ is sufficiently small, the quantization error ε can be approximated as a uniformly distributed random variable. The maximum quantization error is given by:

$$|\varepsilon_{max}| = \frac{\Delta}{2}$$

Thus, the quantization error satisfies:

$$-\frac{\Delta}{2} \leq \varepsilon \leq \frac{\Delta}{2}$$

This assumption holds for uniform quantization when the signal is sufficiently dense across the quantization levels. However, in cases of non-uniform quantization, such as logarithmic companding (e.g., μ-law or A-law), the error distribution may differ.

Example: Transmission of a Television Signal Using Binary PCM

A television signal with a bandwidth of 10.2 MHz is transmitted using a binary PCM system. Given that the number of quantization levels is 512, determine:

a) Code word length

b) Transmission bandwidth

c) Final bit rate

Solution:

Given Data:

- Bandwidth of the signal: f_m=10.2MHz
- Number of quantization levels: q=512

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(a) Code Word Length

The relationship between the number of bits per sample v and the number of quantization levels q in binary PCM is given by:

$$q = 2^v$$

$$512 = 2^v$$

Taking the logarithm on both sides:

$$v = \frac{log_{10}512}{log_{10}2} = 9bits$$

(b) Transmission Bandwidth

The transmission bandwidth for PCM is given by:

$$BW \geq vf_m$$

$$BW \geq 9 \times 10.2 \times 10^6$$

$$BW \geq 91.8MHz$$

(c) Final Bit Rate

The final bit rate is given by:

$$r = vf_s$$

$$f_s \geq 2f_m = f_s \geq 2 \times 10.2M = 20.4MHz$$

$$r = 9 \times 20.4 \times 10^6 = 183.6 \times 10^6 bits/sec$$

$$= 183.6Mbps$$

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