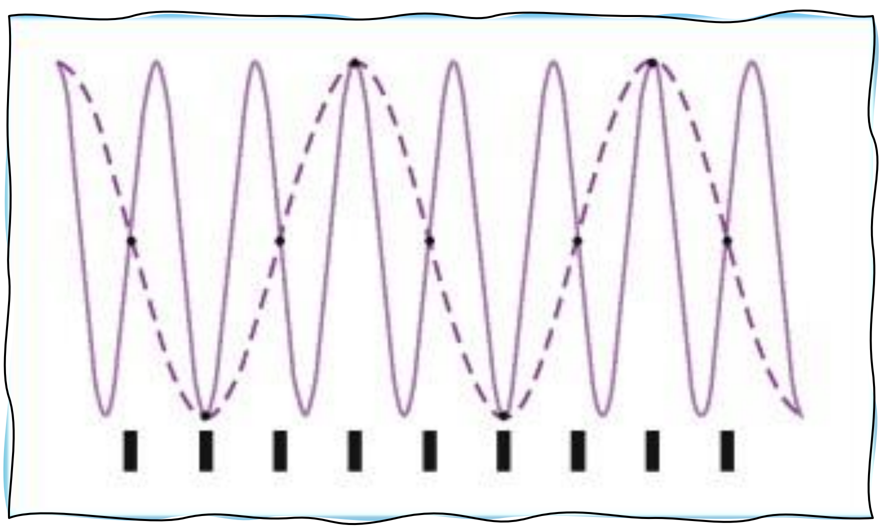
Al-Mustaqbal University
Department of Medical Instrumentation Techniques Engineering
Class: Third
Subject: Medical Communication Systems
Lecturer: Prof. Dr. Bayan Mahdi Sabbar & M.Sc. Huda Wasfi Hassoon
Lecture: 8

Lecture 8

Sampling



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Sampling refers to the process of selecting a subset from a large dataset. In digital communication, it specifically involves measuring the instantaneous values of an analog signal and converting them into discrete form.

A sampler is the device responsible for this process. Sampling is a key component in almost all digital communication systems, as it facilitates the conversion of analog signals, which are continuous and time-varying, into digital signals represented in discrete form. The sampler measures the instantaneous values of the continuous analog signal and transforms them into discrete values.

In digital communication, the combined operation of a quantizer and a sampler function as an **Analog-to-Digital (A/D)** Converter, which converts an incoming analog signal into a digital signal. This process consists of two main steps:

1. Sampling: The sampler converts the analog signal into discrete values.
2. Quantization: The quantizer assigns these discrete values to a fixed set of discrete levels.

Therefore, analog-to-digital conversion is a two-step process, where the sampler performs the first step, and the quantizer completes the conversion. Below is a visual representation of the waveforms of a continuous analog signal and its corresponding digital signal.

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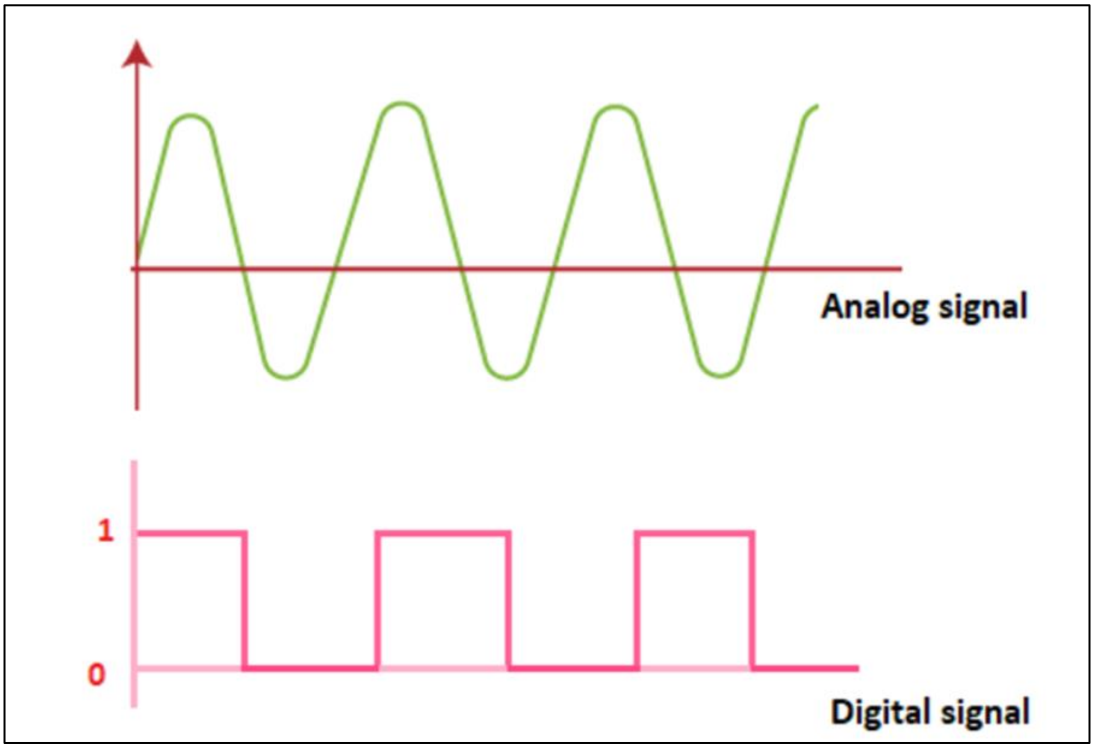

Figure 1: Analog and digital signal.

When a continuous signal is sampled at regular intervals and multiplied by a periodic pulse train, the resulting signal is referred to as the **sampled signal**. This process captures discrete values of the continuous signal at specific time intervals defined by the periodic pulse train. The sampled signal retains the essential information of the original signal while converting it into a form suitable for further digital processing.

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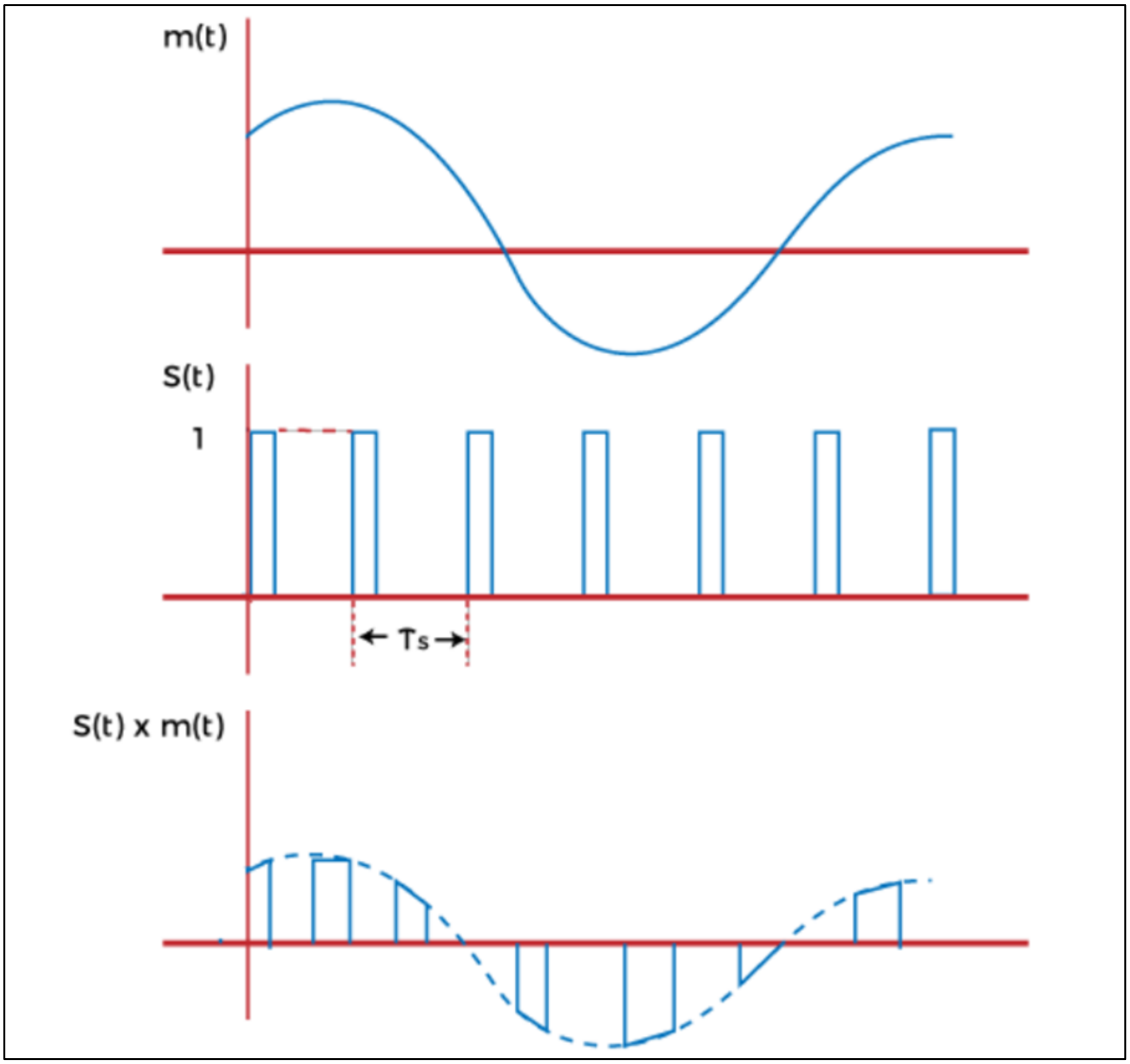

Figure 2: Sampled signal.

we will cover the following key topics related to sampling:

- **Sampling Theorem**: The fundamental principle that defines the conditions for accurate signal sampling.
- **Sampling Rate**: The frequency at which the continuous signal is sampled.
- **Nyquist Rate**: The minimum sampling rate required to avoid aliasing.

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- **Methods of Sampling**: Various techniques used to sample a signal, such as uniform sampling, non-uniform sampling, and others.
- **Anti-Aliasing Filter**: A filter applied before sampling to eliminate high-frequency components and prevent aliasing.
- **Advantages of Sampling**: The benefits of converting continuous signals into discrete forms.
- **Disadvantages of Sampling**: The potential challenges or limitations associated with sampling.
- **Applications of Sampling**: Practical uses of sampling in fields such as digital communication, audio processing, and medical imaging.

Sampling theorem

The sampling theorem is based on a fixed sampling rate, known as the **Nyquist rate**. For this reason, the sampling theorem is also referred to as the **Nyquist theorem**. It relies on the theory of bandlimited signals. Let us explore the sampling theorem as it applies to **bandpass signals** and **baseband signals**.

Sampling Theorem for Bandpass Signals:

According to the sampling theory for bandpass signals, a signal can be accurately reconstructed if its sampling rate is not greater than the maximum frequency W. The samples must be spaced at intervals of Ts seconds, where Ts is defined as:

$$T_s = \frac{1}{2\omega}$$

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where ω is the maximum angular frequency of the signal. Reconstruction without any mean square error is possible under this condition.

Sampling Theorem for Baseband Signals:

For baseband signals, a signal can be successfully reconstructed if the samples are separated by a uniform interval Ts, which must be less than or equal to $(1/2F_m)$, where F_m is the maximum frequency of the baseband signal. This can be represented mathematically as:

$$F_s \leq \frac{1}{2F_m}$$

This condition ensures that the sampling process retains all the necessary information for perfect signal reconstruction.

Sampling Rate

The sampling rate refers to the number of samples captured per second from a continuous signal to create a finite set of discrete values. It can also be expressed as the sampling frequency, which is the inverse of the sampling time:

$$F_s = \frac{1}{T_s}$$

Where:

- F_s : sampling frequency
- T_s: sampling time

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As previously explained, the sampling rate is a critical parameter in the sampling process, enabling the accurate recovery of the digital signal at the receiving end. To ensure this accuracy, a fixed parameter called the Nyquist rate is established as a standard for the sampling rate.

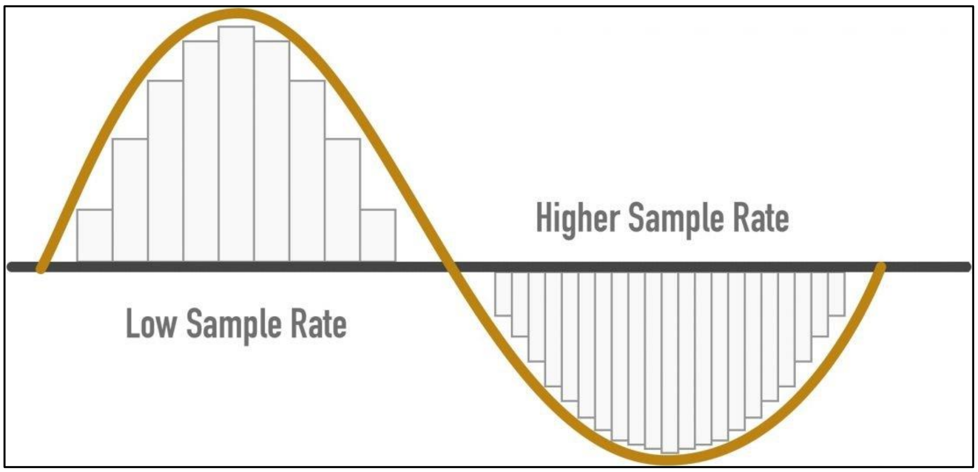

Figure 3: The sampling rate.

Nyquist Rate

Let W represent the highest frequency component of a bandlimited signal. To accurately reconstruct the original signal without distortion, the sampling rate must be at least twice the highest frequency. This is expressed mathematically as:

$$F_s = 2\omega$$

Where:

- F_s : Sampling rate
- ω: Highest frequency

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This specific sampling rate is referred to as the Nyquist rate. Sampling at the Nyquist rate ensures there is no distortion introduced during the signal reconstruction process.

The Nyquist rate is also commonly defined as the minimum sampling rate, satisfying the condition:

$$F_s = 2F_m$$

Where:

- F_s : Sampling frequency or sampling rate
- F_m: Maximum frequency of the input or message signal

Nyquist Interval

The **Nyquist interval** is defined as the reciprocal of the Nyquist rate. It represents the time interval between successive samples required for accurate signal reconstruction. Mathematically, it is expressed as:

$$T_s = \frac{1}{2W}$$

Where:

- T_s : Nyquist interval
- W: Highest frequency

This interval is a critical parameter in the sampling process, ensuring that the signal is sampled at the appropriate rate to avoid distortion or loss of information.

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Methods of Sampling

Sampling is the process of converting a continuous-time signal into a discrete-time signal by taking periodic samples. The methods of sampling are categorized into three main types:

1. **Ideal Sampling**
2. **Natural Sampling**
3. **Flat-Top Sampling**

1. Ideal Sampling

Ideal sampling, also referred to as **instantaneous sampling** or **impulse sampling**, is a theoretical sampling approach. In this method, the input signal is multiplied by a carrier signal, which consists of a train of very narrow pulses (essentially impulses). These pulses sample the input signal at discrete intervals, creating an output that is a series of weighted impulses.

This method is considered "ideal" because it assumes the impulses are infinitely narrow and occur at precise intervals, ensuring no loss of information. However, due to practical limitations, ideal sampling is not directly implemented in real-world systems but serves as a theoretical model for understanding the sampling process.

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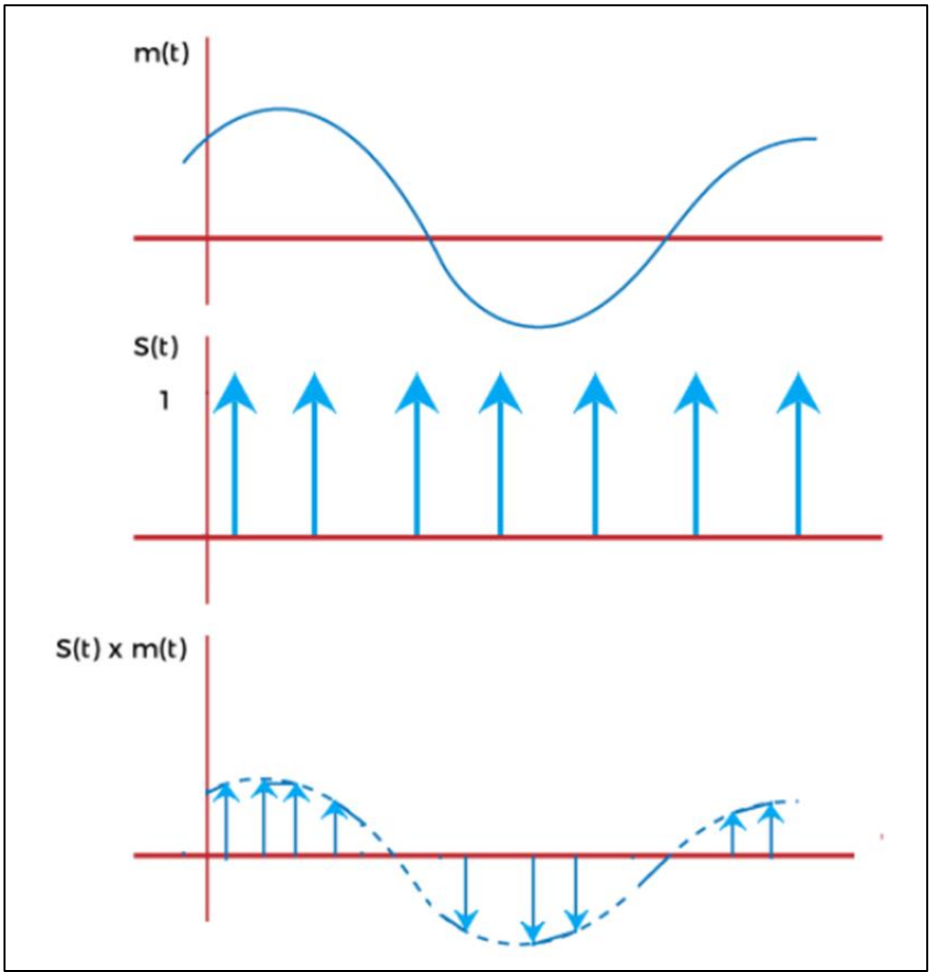

Figure 4: The ideal sampling.

In the diagram:

1. **The first graph** represents the input signal or message signal, $m(t)$, which is a continuous waveform.
2. **The second graph** shows the sampling signal, $S(t)$, which is a train of narrow pulses. These pulses are spaced at regular intervals, corresponding to the sampling period.
3. **The third graph** illustrates the sampled signal, $S(t) \cdot m(t)$, which is obtained by multiplying the message signal and the sampling signal. The result is a

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series of pulses where the amplitude of each pulse is proportional to the value of the message signal at the corresponding time.

This sampling process is governed by the **multiplication principle**, where each sample corresponds to the instantaneous product of the input signal m(t) and the sampling signal S(t). The resulting sampled signal contains all the essential information about the original signal, provided the sampling theorem is satisfied.

2. Natural Sampling

Natural sampling is an efficient method used in Pulse Amplitude Modulation (PAM) and is particularly effective for multiplexing purposes. In this technique, the analog input signal is multiplied by a series of uniformly spaced rectangular pulses. Unlike ideal sampling, where impulses are used, natural sampling maintains the shape of the signal within each sampling interval, creating a more practical representation of the original signal.

In the diagram below:

1. **The first graph** shows the message signal, $m(t)$, which is the continuous analog input signal to be sampled.
2. **The second graph** illustrates the sampling signal, $S(t)$, which consists of a train of rectangular pulses. These pulses are uniformly spaced, with a period denoted as Ts, representing the sampling interval.
3. **The third graph** depicts the sampled signal, $S(t) \cdot m(t)$, which is the product of the message signal and the sampling signal. In this output, the amplitude of the rectangular pulses follows the shape of the message signal within each interval.

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Natural sampling is particularly significant because it provides a more realistic representation of the sampling process, as it accounts for the finite width of the pulses, unlike ideal sampling. This approach is widely used in practical systems and is a foundational concept in communication theory.

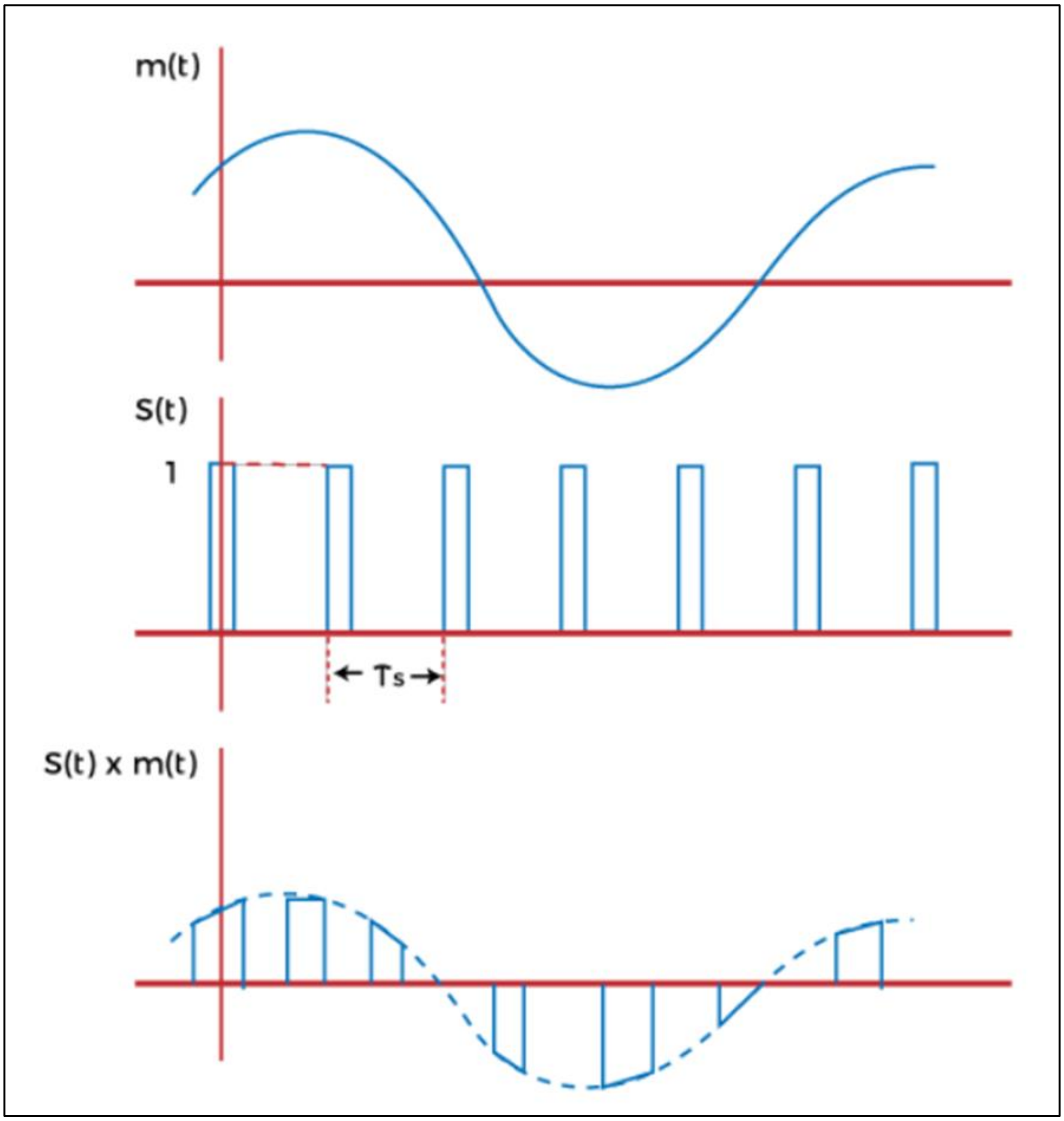

Figure 5: The Natural Sampling.

3. Flat-Top Sampling

Flat-top sampling is a widely used sampling technique because it simplifies both the design and reconstruction processes compared to natural sampling. In this method, the sampling pulses have a flat, rectangular shape at the top, and their amplitude is held constant during the sampling interval.

This means that the samples are represented as "flat" pulses, where the amplitude of each pulse remains constant and corresponds to the value of the analog signal at the moment of sampling. The flat-top nature of the pulses ensures that the sampled signal is easier to process and reconstruct, as it eliminates any variations or slopes within the pulse duration.

Flat-top sampling is particularly beneficial in practical systems due to its simplicity and efficiency. However, it introduces a phenomenon called **aperture effect**, where the sampled signal experiences slight attenuation due to the finite width of the flat-top pulses. This effect can be corrected during reconstruction using techniques such as equalization.

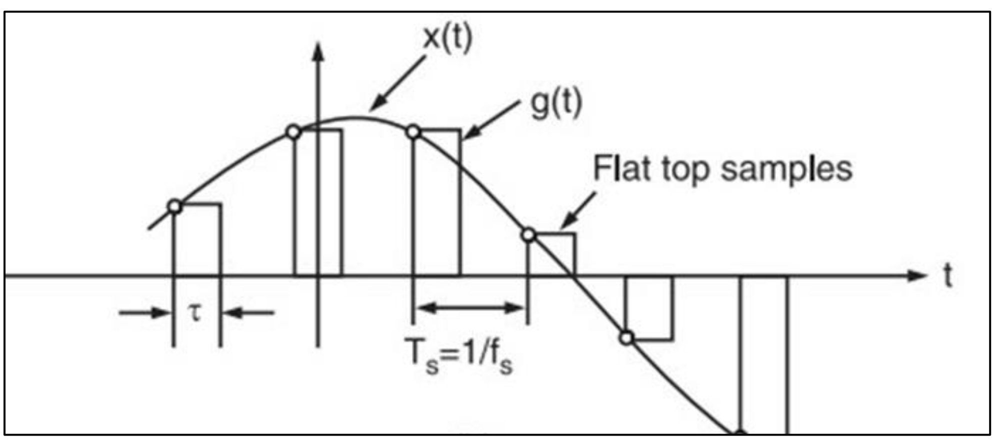

Figure 6: The Flat-Top Sampling.

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Why is sampling required?

Sampling plays a crucial role in the conversion of analog signals to digital signals. This process enables data transmission in digital form, which offers numerous advantages over analog transmission. These benefits include high efficiency, faster speeds, reduced costs, minimal interference, low distortion, and enhanced security. As a result, sampling is essential for improving the quality of signals and optimizing their transmission over communication channels.

Advantages of Sampling

The key advantages of sampling stem from the conversion of analog signals to digital form, as digital signals provide numerous benefits. Sampling converts a continuous-time analog signal into discrete values, making it suitable for efficient processing and transmission. The main advantages of sampling include:

- **Low Cost**: Digital systems are generally more cost-effective compared to analog systems.
- **High Accuracy**: The discrete representation minimizes errors during processing.
- **Ease of Implementation**: Digital systems are simpler to design and implement.
- **Time Efficiency**: Processing and transmitting digital signals are faster.
- **Low Signal Loss**: Digital systems experience minimal signal degradation during transmission.
- **Wide Applicability**: Sampling is foundational in various modern communication systems.

Additionally, sampling helps preserve the original signal's integrity. By converting the incoming data to a suitable rate for transmission, it ensures no information is lost. For instance, signals with high-frequency components are sampled at higher rates to ensure effective transmission. Typically, the sampling process follows the **Nyquist theorem**, which states that the sampling rate must be at least twice the highest frequency of the input signal to preserve all the information.

Applications of Sampling

Sampling is critical in representing analog signals as digital values, reducing information loss during transmission and enhancing system accuracy. It finds applications in several communication and signal processing methods, including:

- **Pulse Amplitude Modulation (PAM)**
- **Pulse Code Modulation (PCM)**
- **Time Division Multiplexing (TDM)**

By ensuring efficient and accurate signal representation, sampling serves as the foundation for modern digital communication systems.

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