Basic Digital Communication Transformations

- ✓Formatting/Source Coding
- ✓Transforms source info into digital symbols (digitization)
- ✓ Selects compatible waveforms (matching function)
- ✓Introduces redundancy which facilitates accurate decoding despite errors

Practical Aspects of Sampling

1. Sampling Theorem
2. Methods of Sampling
3. Significance of Sampling Rate
4. Anti-aliasing Filter
5. Applications of Sampling Theorem – PAM/TDM

Sampling

Sampling is the processes of converting continuous time analog signal, $x_a(t)$, into a discrete-time signal by taking the "samples" at discrete-time intervals

– Sampling analog signals makes them discrete in time but still continuous valued

– If done properly (***Nyquist theorem*** is satisfied), sampling does not introduce distortion

Sampled values:

– The value of the function at the sampling points

Sampling interval:

– The time that separates sampling points (interval b/w samples), ***Ts***

– If the signal is slowly varying, then fewer samples per second will be required than if the waveform is rapidly varying

– So, the optimum sampling rate depends on *maximum frequency* component present in the Signal.

Sampling

- **Analog-to-digital conversion** is (basically) a 2 step process:

– Sampling

Convert from continuous-time analog signal $x_a(t)$ to discrete-time continuous value signal x

– Is obtained by taking the "samples" of $x_a(t)$ at discrete-time intervals, T_s

- **Quantization**

– Convert from discrete-time continuous valued signal to discrete time discrete valued signal

- **Sampling Rate (or sampling frequency f_s):**

The rate at which the signal is sampled, expressed as the number of samples per second reciprocal of the sampling interval), $1/T_s = f_s$

Sampling

- **Nyquist Sampling Theorem (or Nyquist Criterion):**

the sampling is performed at a proper rate, no info is lost about the original signal and it can be properly reconstructed later on

- **Statement:** "If a signal is sampled at a rate at least, but not exactly equal to twice the max frequency component of the waveform, then the waveform can be exactly reconstructed from the samples without any distortion"

$$f_s \geq 2f_{max}$$

- **Sampling Theorem for Bandpass Signal :**

➢ If an analog information signal containing no frequency outside the specified bandwidth W Hz, it may be reconstructed from its samples at a sequence of points spaced 1/(2W) seconds apart with zero-mean squared error.

➢ The minimum sampling rate of (2W) samples per second, for an analog signal bandwidth of W Hz, is called the **Nyquist rate**.

➢ The reciprocal of Nyquist rate, 1/ (2W) is called **Nyquist interval** . i.e., Ts = 1/(2W),

➢ The phenomenon of the presence of high-frequency component in the spectrum of the original analog signal aliasing or simply foldover.

Sampling Theorem

- A baseband signal having no frequency components higher than f_m Hz may be completely recovered from the knowledge of its samples taken at a rate of at least $2f_m$ samples per second, that is, sampling frequency

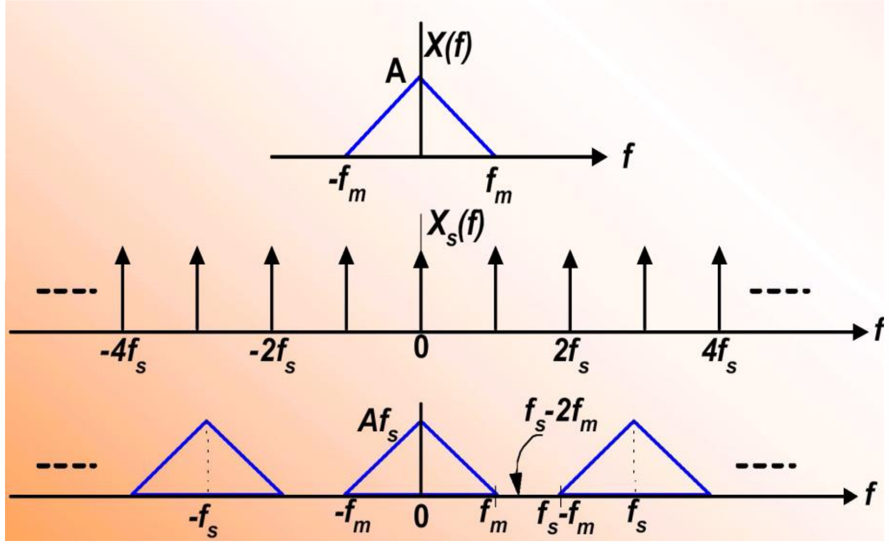
$$f_s \geq 2f_m$$

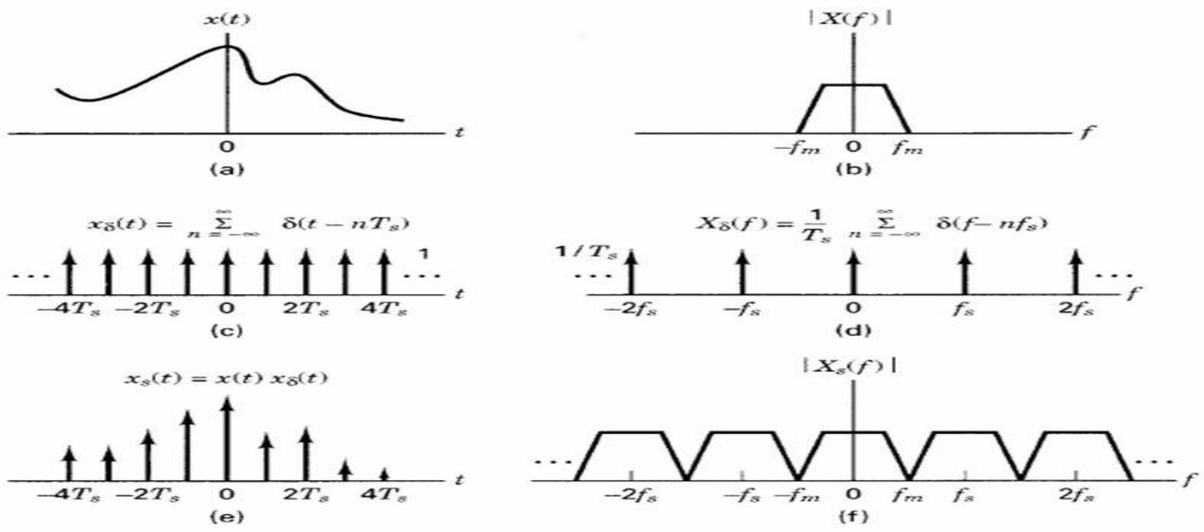
- The minimum sampling rate $f_s = 2f_m$ samples per second is called the Nyquist sampling rate.

- A baseband signal having no frequency components higher than f_m Hz is completely described by its sample values at uniform intervals less than or equal $1/(2f_m)$, seconds apart that is, the sampling interval $T_s \leq 1/(2f_m)$ seconds.

Methods of Sampling

- Ideal Sampling (or Impulse Sampling): This means that the output is simply the replication of the original signal at discrete intervals

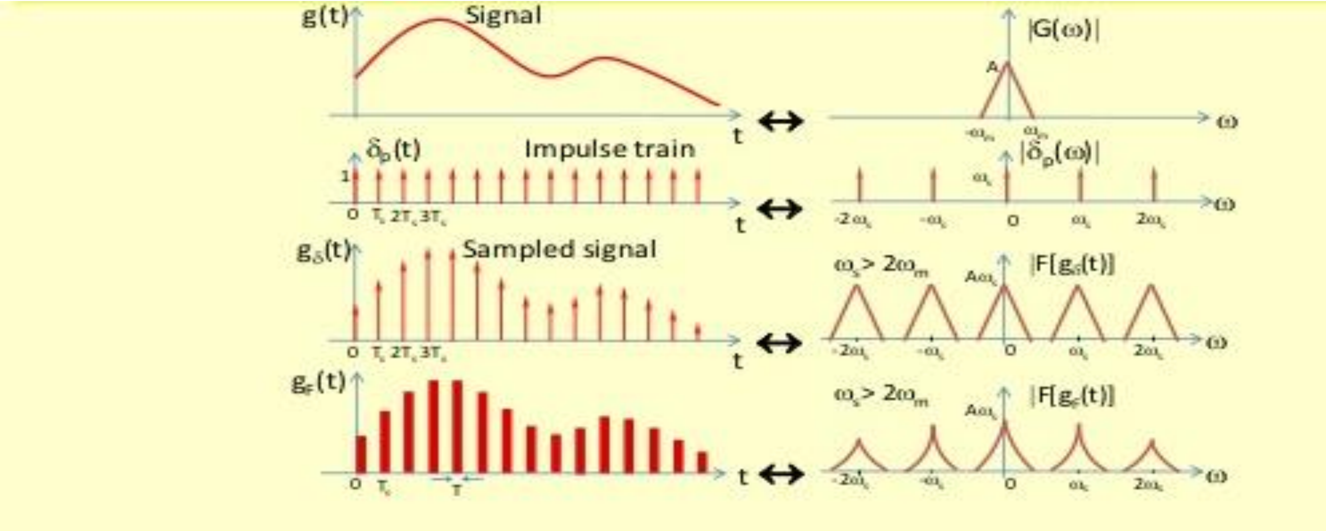




T_s is called the ***Nyquist interval:*** It is the longest time interval that can be used for sampling a band limited signal and still allow reconstruction of the signal at the receiver without distortion

Methods of Sampling

- Natural Sampling is a practical method of **sampling** in which pulse have finite width equal to τ. **Sampling** is done in accordance with the carrier signal which is digital in **nature**. **Natural Sampled** Waveform.
- The amplitude of the **flat top** signal must be constant, but sometimes it is not constant due to the high frequency roll off of the **sampling** signal.
- Thus the **sampled** signal in the **flat top sampling** consists of attenuated high frequency components and this **effect** is known as **Aperture effect**.



Flat top sampling, introduces aperture effect as per sample function

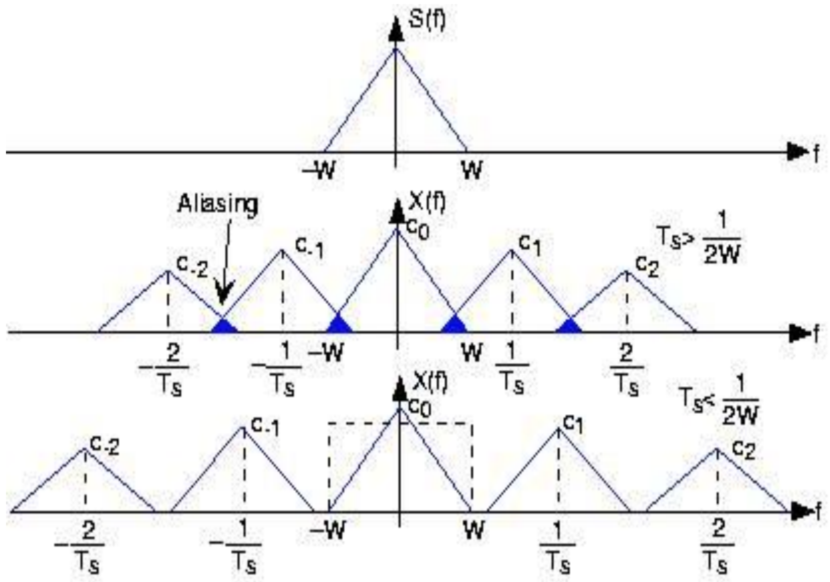
Contents

- Proof of Sampling Theorem

Proof of Sampling Theorem

- Representation of $x(t)$ in terms of its samples.
- Reconstruction of $x(t)$ from its samples.

Effects of UnderSampling



If we sample at too low of a rate (below the Nyquist rate), then problems will arise that will make perfect reconstruction impossible - this problem is known as aliasing.

Aliasing occurs when there is an overlap in the shifted, perioidic copies of our original signal's FT, i.e. spectrum.

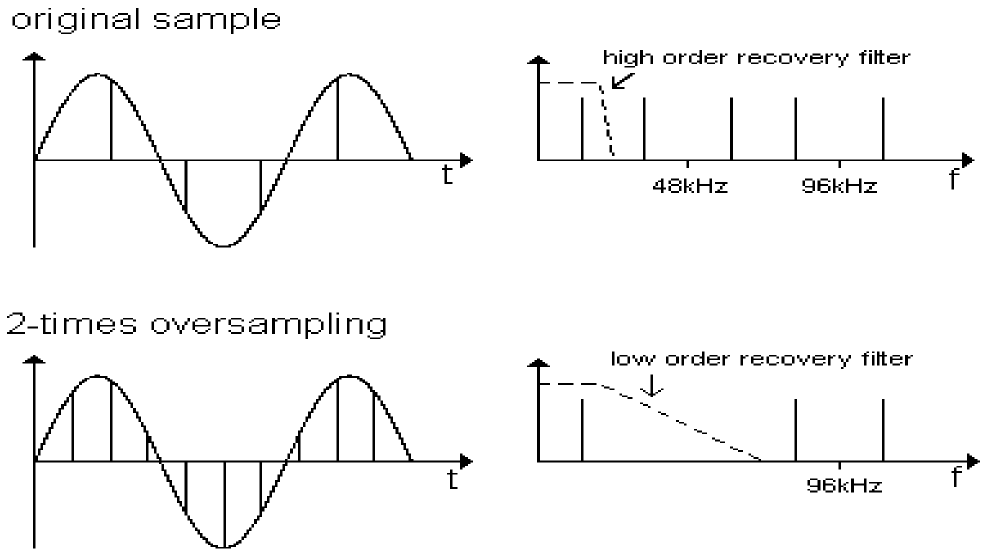
In the frequency domain, one will notice that part of the signal will overlap with the periodic signals next to it.

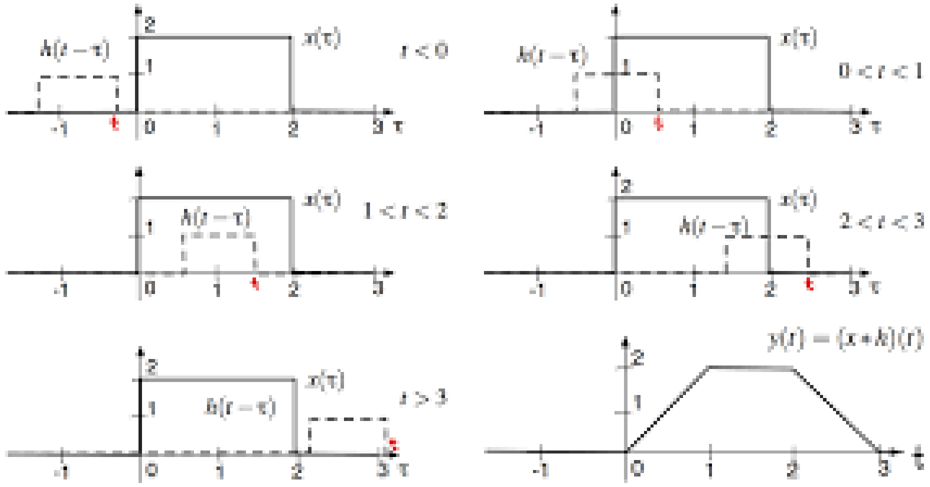
In this overlap the values of the frequency will be added together and the shape of the signals spectrum will be unwantingly altered.

This overlapping, or aliasing, makes it impossible to correctly determine the correct strength of that frequency.

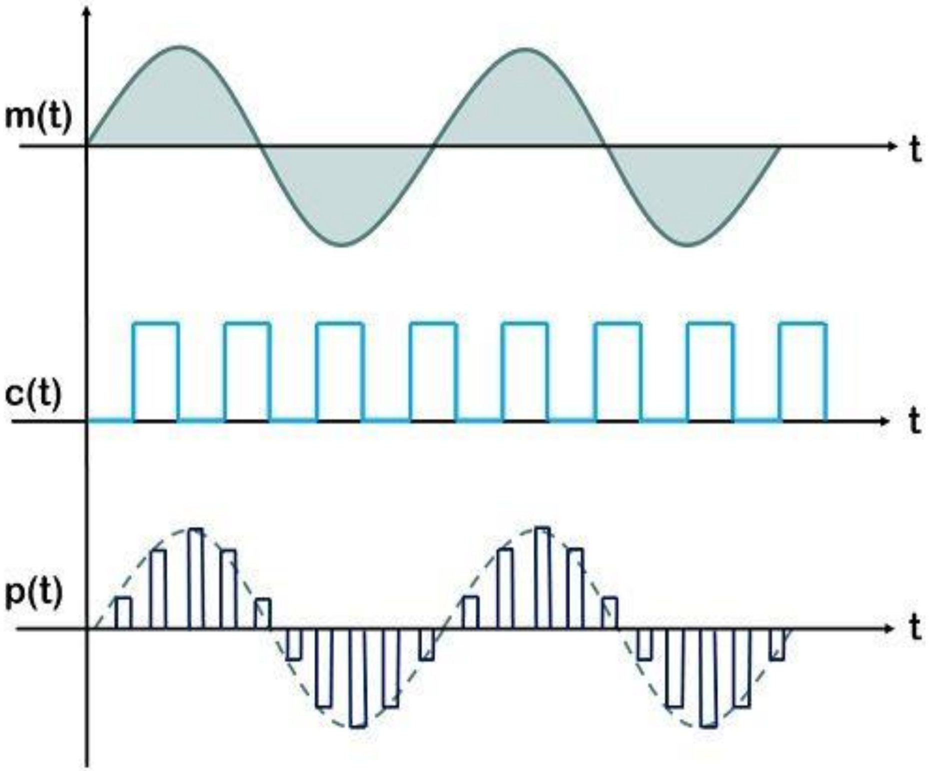
Effects of OverSampling

- **Oversampling** is the process of sampling a signal at a sampling frequency significantly higher than the Nyquist rate.
- Theoretically, a bandwidth-limited signal can be perfectly reconstructed if sampled at the Nyquist rate or above it.
- The Nyquist rate is defined as twice the bandwidth of the signal.
- Oversampling is capable of improving resolution and signal-to-noise ratio, and can be helpful in avoiding aliasing and phase distortion by relaxing anti-aliasing filter performance requirements





Pulse Amplitude Modulation



Pulse-amplitude modulation (**PAM**), is a form of signal modulation where the message information is encoded in the amplitude of a series of signal pulses.

It is an analog pulse modulation scheme in which the amplitudes of a train of carrier pulses are varied according to the sample value of the message signal.

Demodulation is performed by detecting the amplitude level of the carrier at every single period.

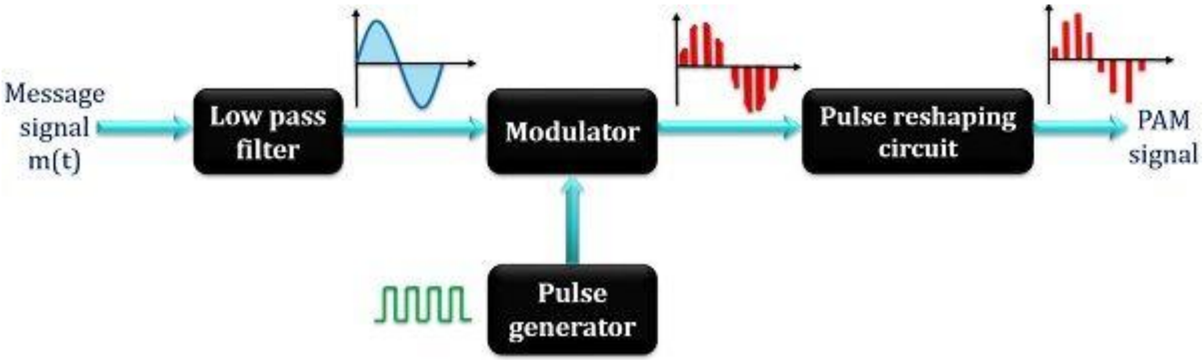
Flat Top PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

Natural PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. Then follows the amplitude of the pulse for the rest of the half-cycle.

Pulse Amplitude Modulation

Applications of PAM

- It is used in Ethernet communication.
- It is used in many micro-controllers for generating the control signals.
- It is used in Photo-biology.
- It is used as an electronic driver for LED lighting.



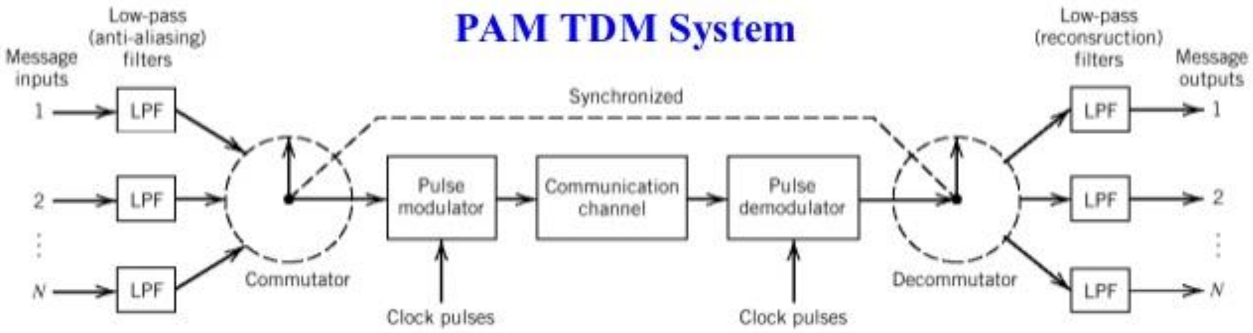
Advantages

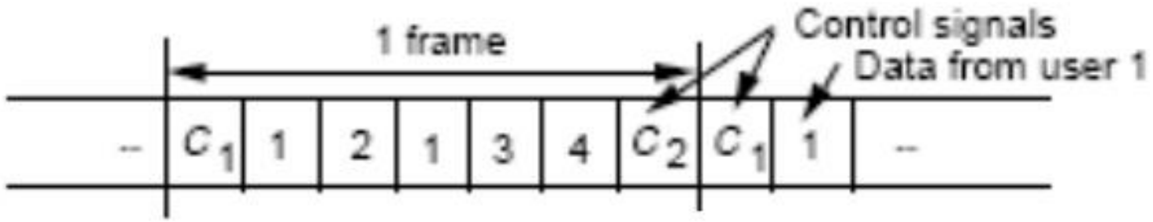
- It is a simple process for both modulation and demodulation.
- Transmitter and receiver circuits are simple and easy to construct.
- PAM can generate other pulse modulation signals and can carry the message at the same time.

Disadvantages

- Bandwidth should be large for transmission PAM modulation.
- Noise will be great.
- Pulse amplitude signal varies so the power required for transmission will be more.

TDM





A Typical Framing Structure for TDM