Principles of Communication (BEC-28)

Unit-4

Pulse Modulation and Digital Transmission of Analog Signal

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Content of Unit-IV

Pulse Modulation and Digital Transmission of Analog Signal: Sampling Theorem and its applications, Concept of Pulse Amplitude Modulation, Pulse width modulation and pulse position modulation, PCM, Pulse Time Modulation, TDM and FDM. Line Coding, Quantizer, Quantization Noise, Compounding multiplexer.

Pulse Modulation

- 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)

Advantage of Pulse modulation:

(i) Transmitted power is no longer continuous as in CW Modulation, but pulsed in nature

(ii) Vacant time between pulse occurrence filled by interleaving/multiplexing pulse waveforms of some other Message (TDM)

Sampling Theorem

This provides a mechanism for representing a **continuous time signal** by a **discrete time signal** , taking **sufficient number of samples of signal** so that **original signal is represented in its samples completely.** It can be stated as:

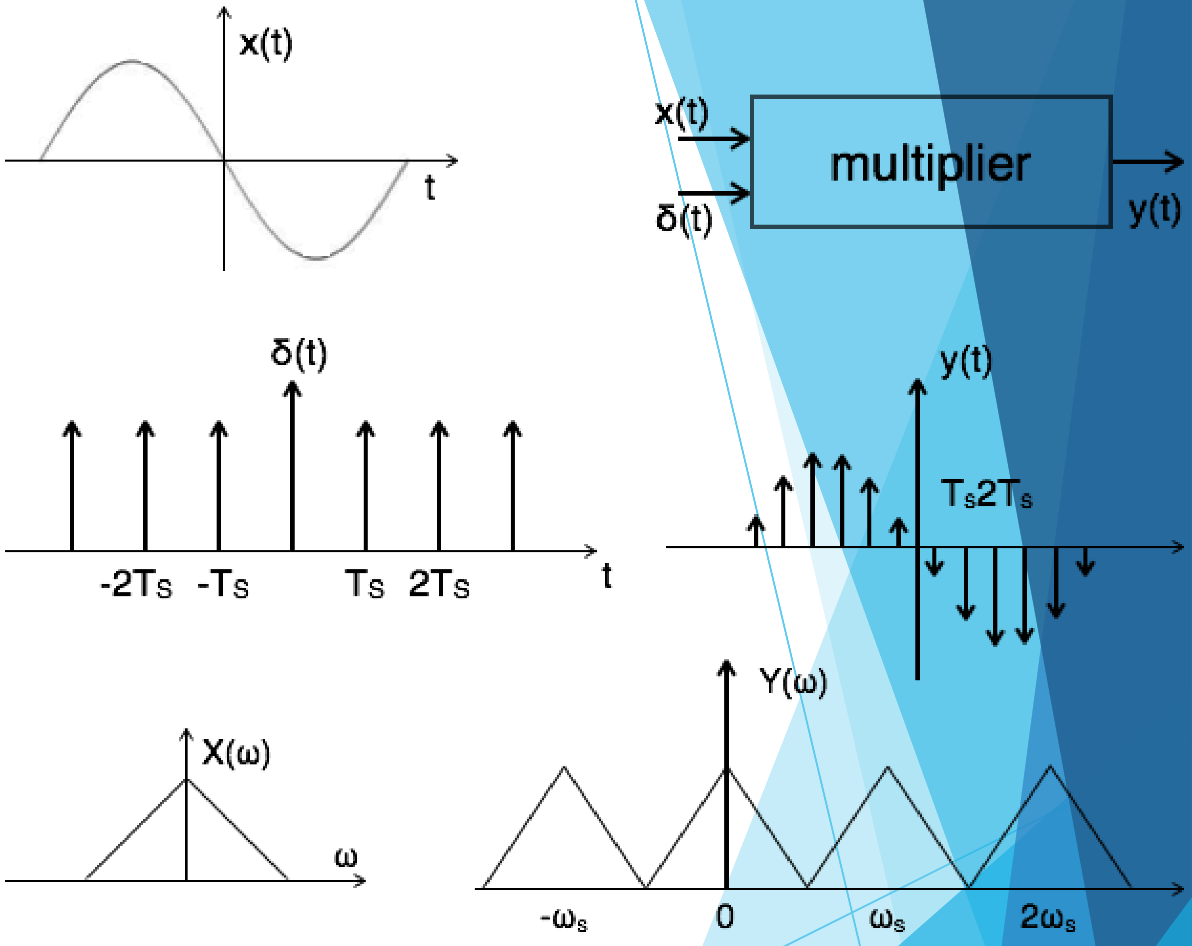
(i) A band-limited signal of finite energy with no frequency component higher **than f_m Hz**, is completely described by **its sample values** which are at uniform intervals **less than or equal to $1/2f_m$** seconds apart. [$T_s = \frac{1}{2fm}$]where T_s is sampling time.

(ii) **Sampling frequency** must be **equal to or higher than $2f_m$** Hz. **[$fs \geq 2fm$]**

A continuous time signal may be completely represented in samples and recovered back, if $f_s \geq 2f_m$, where f_s is sampling frequency and f_m is maximum frequency component of message signal

Proof of sampling theorem

- **Sampling** of input signal **x(t)** can be obtained by **multiplying x(t)** with an **impulse train δ(t)** of period T_s.
- The output of multiplier is a discrete signal called **sampled signal** which is represented with y(t) in the diagrams,
- **y(t)=x(t).δ(t)......(1)**



The Fourier series representation of δ(t) :

- $\boldsymbol{\delta(t)=a_0+\sum_{n=1}^{\infty}(a_n\cos n\omega_s t + b_n \sin n\omega_s t)......(2)}$

where $a_0=\frac{1}{T_s}\int_{-T/2}^{T/2}\delta(t)\,dt = \frac{1}{Ts}\delta(0) = \frac{1}{T_s}$

$a_n=\frac{2}{T_s}\int_{-T/2}^{T/2}\delta(t)\cos n\omega_s dt = \frac{2}{T_s}\delta(0)\cos n\omega_s 0=\frac{2}{T_s}$

$b_n= \frac{2}{\ }\int\delta(t)\sin n\omega_s t\, dt = \frac{2}{\ }\delta(0)\sin n\omega_s 0=0$

$$\delta(t)=\frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)$$

$$\therefore\delta(t) = \frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)$$

Substitute δ(t) in equation 1.

$\rightarrow y(t)=x(t).\delta(t)$

$$= x(t)\left[\frac{1}{T_s}+\sum_{n=1}^{n=\infty}\left(\frac{2}{T_s}\cos n\omega_s t+0\right)\right]$$

$$=\frac{1}{T_s}\left[x(t)+2\sum_{n=1}^{n=\infty}(\cos n\omega_s t)x(t)\right]$$

$y(t)= \frac{1}{T_s}[x(t)+2\cos\omega_s t.x(t)+2\cos 2\omega_s t.x(t)+2\cos 3\omega_s t.x(t)......]$

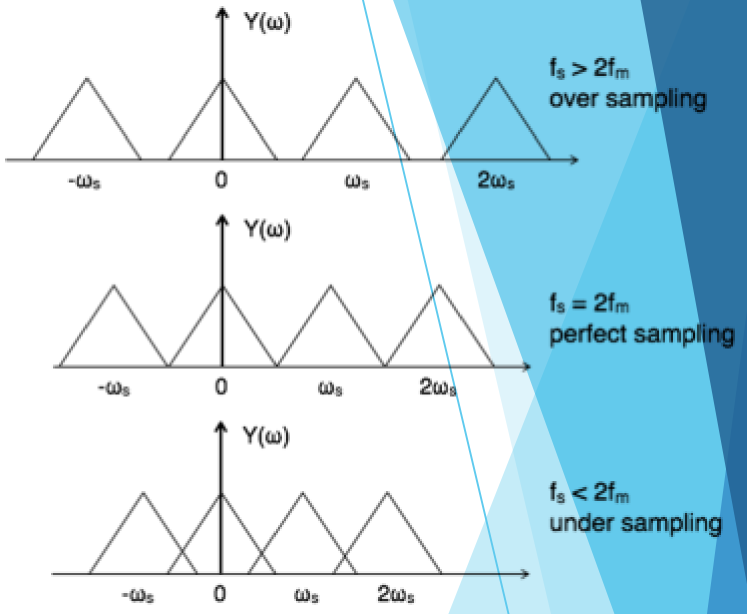
Take Fourier transform on both sides.

$$Y(\omega) = \frac{1}{T_s}[X(\omega)+ X(\omega-\omega_s) +X(\omega+\omega_s)+X(\omega-2\omega_s)+X(\omega+2\omega_s)+ X(\omega+3\omega_s)+..........]$$

$$Y(\omega)=\frac{1}{T_s}\sum_{-\infty}^{+\infty} X(\omega - n\omega_s)$$

To reconstruct x(t), one has to recover input signal spectrum X(ω) from sampled signal spectrum Y(ω), which is possible when there **is no overlapping between the cycles of Y(ω) which is possible if**

$f_s \geq 2f_m$



For **$f_s=2f_m$,** is known as **Nyquist rate.**

$T_s = \frac{1}{2fm}$ **is known as Nyquist interval**

Aliasing Effect

The overlapped region in case of **under sampling**

represents **Aliasing effect**. It can be termed as "**the phenomenon of a high-frequency component in the spectrum of a signal, taking on the identity of a lower-frequency component in the spectrum of its sampled version.**

This effect can be removed by considering

(i) fs >$2f_m$ or

(ii) by using anti aliasing filters which are low pass filters and eliminate high frequency components


Thank You