Department of Electrical Engineering
Sessional Assignment
Date: 05/05/2020

Course Details

Course Title: Signals & Systems **Module:** 04
Instructor: Engineer Mujtaba Ehsan **Total Marks:** 20

Student Details

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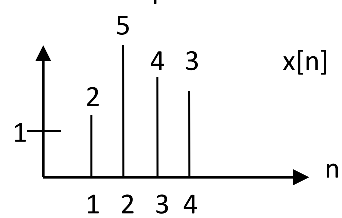
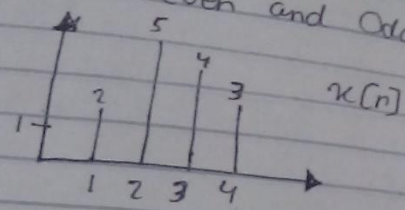
Q1.		**Evaluate** the even and odd components for the given function. 	Marks 05 CLO 1
Q2.		**Calculate** the inverse Laplace transform of the given equation. $\mathbf{Y(s)} = \frac{\mathbf{s+4}}{\mathbf{s^2 + 4s - 12}}$	Marks 07 CLO 3
Q3.	i. ii.	**Discuss** the procedure of converting an analog signal into a digital one. Suppose an analog signal has a highest frequency of 60Hz. **Outline** the steps that will ensure that no aliasing occurs.	Marks 02+02 CLO 2
Q4.		**Show** that: $x[n] * [h_1[n] * h_2[n]] = [x[n] * h_1[n]] * h_2[n]$	Marks 04 CLO 2

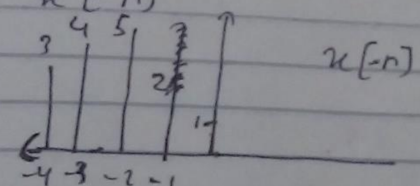
Q. Evaluate Even and Odd components



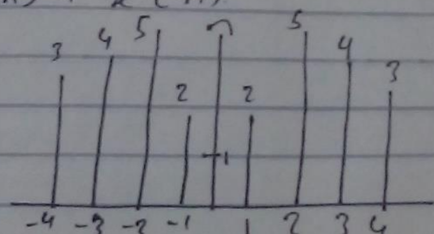
For even component the formula is

$$x_e[n] = \frac{x[n] + x[-n]}{2}$$

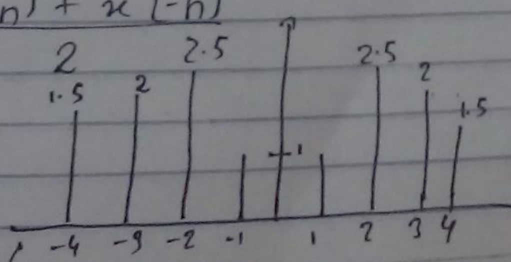
Now $x[-n]$



Now $x(n) + x(-n)$



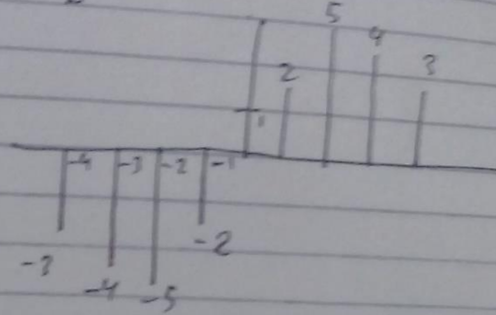
Now $\frac{x(n) + x(-n)}{2}$



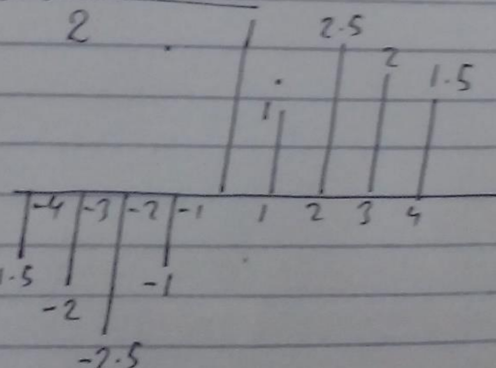
$x_e[n]$

Now odd component
Formula

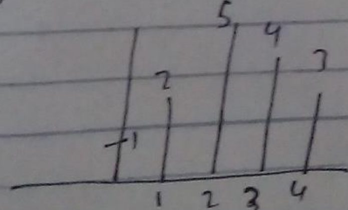
$$x_o[n] = \frac{x[n] - x[-n]}{2}$$



Now $\frac{x[n] - x[-n]}{2}$



Now, by adding $x_e[n]$ and $x_o[n]$ we will get $x[n]$



Q2 Calculate Laplace transform

$$Y(s) = \frac{s+4}{s^2+4s-12}$$

Sol $$Y(s) = \frac{s+4}{s^2+4s-12}$$

$$= \frac{s+4}{s^2+6s-2s-12}$$

$$= \frac{s+4}{s(s+6)-2(s+6)}$$

$$= \frac{s+4}{(s-2)(s+6)}$$

$$\frac{s+4}{(s-2)(s+6)} = \frac{A}{(s-2)} + \frac{B}{(s+6)}$$

$s+4 = A(s+6) + B(s-2)$ —*

Let $s = 6$ in *

$6+4 = A(6+6) + B(6-2)$

$10 = A(12) + B(4)$

$10 = 12A + 4B$ — ①

Let $s = 2$ in *

$2+4 = A(2+6) + B(2-2)$

$6 = A(8) + 0$

$A = 6/8$ — Put in ①

or $\frac{3}{4}$

$$10 = 12\left(\frac{36}{48}\right) + 4B$$

$$10 = \frac{36}{4} + 4B$$

$$10 = 9 + 4B$$

$$10 - 9 = 4B$$

$$1 = 4B$$

$$4B = 1$$

$$B = \frac{1}{4}.$$

$$Y(s) = \frac{3/4}{(s-2)} + \frac{1/4}{(s+6)}$$

$$= \frac{3}{4}e^{2t} + \frac{1}{4}e^{-6t}$$

Q3 ①.

An analog signal can be converted into digital through sampling and quantization.

Proper sampling:-
The definition of proper sampling is quite simple. Suppose you sample a continuous signal in some manner.

If you exactly reconstruct the analog signal from the samples, you must have done the sampling properly. Even if the sampled data appears confusing or incomplete, the key information has been captured if you can reverse the process.

A continuous signal is sampled properly if the samples contain all the information needed to recreate the original waveform.

Digitizing this same signal would produce virtually no increase in the noise, and nothing would be lost due to quantization.

We can simply avoid aliasing by sampling the signal at a higher rate than the Nyquist rate ($F_s >> F_m$). Or we can use anti-aliasing filters. These are special low-pass filters that are usually found in the initial stages of any digital signal processing operation. The anti-aliasing filters attenuate the unnecessary high-frequency components of a signal. They band limit the input signal by removing all frequencies higher than the signal frequencies. As a result, they help preserve a lot of information that is needed and remove unnecessary information.

Qu.

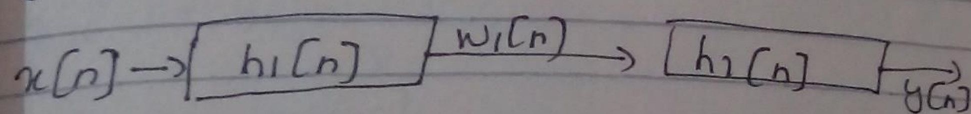
Consider

$y[n] = x[n] * h_1[n] * h_2[n]$

Let, $x[n] * h_1[n] = W_1[n]$

$y[n] = [x[n] * h_1[n]] * h_2[n]$ — ①

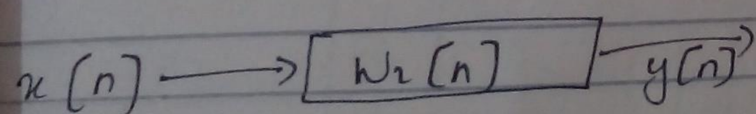
$y[n] = W_1[n] * h_2 n$



Now consider that

$W_2[n] = h_1[n] * h_2[n]$

$y[n] = x[n] * [h_1[n] * h_2[n]]$
$= x[n] * W_2[n]$



Both block diagrams give the same response

Hence

$x[n] * [h_1[n] * h_2[n]] = [x[n] * h_1[n]] * h_2[n]$