FACULTY OF ENGINEERING, UNIVERSITY OF JAFFNA

DIGITAL SIGNAL PROCESSING – EC5011

LABORATORY SESSION 1
DIGITAL SIGNAL PROCESSING THEORY AND APPLICATION

PRE LAB PREPARATION

INTRODUCTION

This session focuses on understanding the basics of Sampling, Aliasing, Fourier transform, filtering, and convolution.

PART1: SAMPLING, TIME DOMAIN & FREQUENCY DOMAIN REPRESENTATION

1. Refer the digital signal x[n] = cos(2*pi*100*t)+cos(2*pi*500*t)+cos(2*pi*2000*t) +cos(2*pi*2750*t) with frequencies 100Hz, 500Hz, 2000Hz, and 2750Hz, sampled by a sampling frequency of 4000Hz. Find the aliased frequency components

PART2: CONVOLUTION AND FILTERING IN LTI SYSTEM

Consider the following filter:

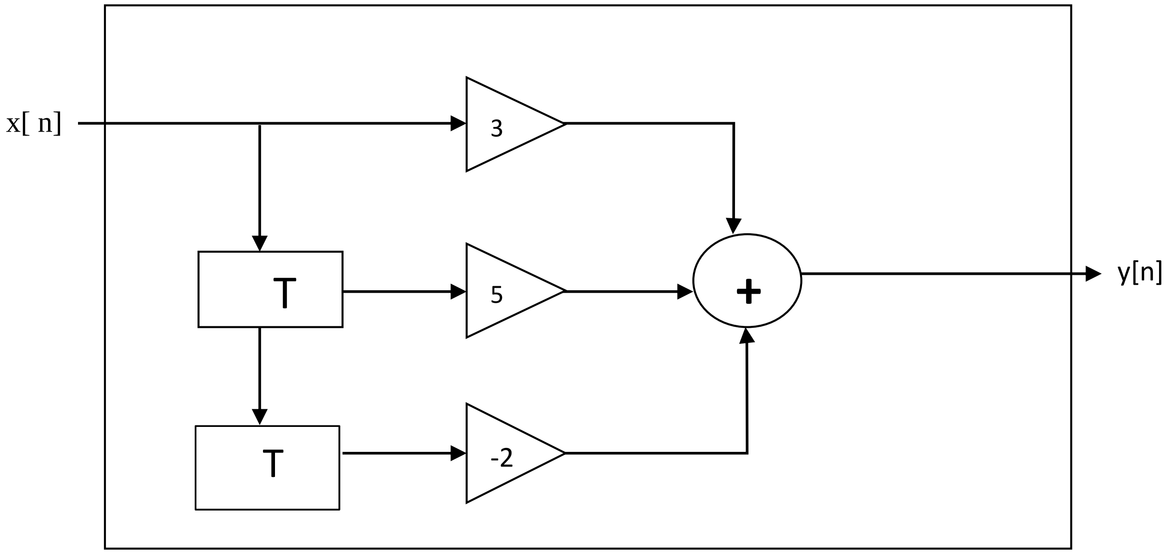

Fig 01. Linear time invariant system

1. If a signal x[n] is passed through this filter calculate the output of filter using the block diagram with zero initial condition.

   x[n]={1, 0, 4, 0, 3,…}   for n=0, 1, 2, 3, 4,…

2. Calculate the output of the filter using the block diagram when initial condition Zi=[1,2]
3. Calculate the impulse response (without using the Z-Transform) of the filter **.**
4. If a signal x(n) is passed through this filter calculate the output of the filter using convolution. You can use the impulse response obtained in Question 3, then find and sketch the output y (n) , state the assumptions needed

   x[n]={1, 0, 4, 0, 3,…}   for   n=0, 1, 2, 3, 4,…

5. Write pseudo code for the implementation of convolution using 'for' or 'while' loop.