

## **Department of Telecommunications** Engineering **NED University of Engineering & Technology**

# LABORATORY WORKBOOK

## For the Course

## **SIGNALS & SYSTEMS** <u>(TC-202)</u>

Instructor Name:

Student Name:

Roll Number: Batch:

Semester: Year:

Department:

## LABORATORY WORK BOOK

# For The Course <u>SIGNALS AND SYSTEMS</u> <u>(TC-202)</u>

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**Reviewed By:** 

Dr. Sundus Ali (Assistant Professor)

**Approved By:** 

**Board of Studies of Department of Telecommunications Engineering** 

## **INTRODUCTION**

This laboratory manual contains exercises based on MATLAB and EV kits. The purpose of these exercises is to help reestablish what is and how to points of view on signals and systems. The exercises integrate the basic concepts for both continuous-time and discrete-time signals and systems. This laboratory manual focuses on an imperative style, where signals and systems are constructed procedurally.

MATLAB distributed by The Math Works, Inc., is chosen as the basis for these exercises because it is widely used by practitioners in the field, and because it is capable of realizing interesting systems.

The exercises in this manual cover many of the properties of linear time-invariant (LTI) systems. It provides an introduction to the basic concepts involved in using MATLAB to represent signals and systems. Also, the necessary tools for dealing with both numerical and symbolic expressions are learned.

The manual covers a variety of exercises that includes signal and system representations for both time and frequency domains, basic properties of signals and systems, the processing of signals by linear systems, Fourier series and transforms, discrete-time processing of continuous-time signals.

The lab exercises introduce the fundamental ideas of signal and system analysis that will help the students to understand the engineering systems in many diverse areas, including seismic data processing, communications, speech processing, image processing, and defense electronics.

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#### **OBJECTIVE:**

To simulate the continuous and discrete times signals and perform basic operations on signals in MATLAB.

#### THEORY:-

#### **Generation of Signals:**

The MATLAB Signal Processing Toolbox has a large variety of functions for generating continuoustime and discrete-time signals. In this lab, we shall learn how to generate some commonly used signals. We shall also perform some basic operations on signals.

#### Impulse Signal:

An impulse (Dirac delta function) is defined as a signal that has an infinite magnitude and an infinitesimally narrow width with an area under it of one, centered at zero. An impulse can be represented as an infinite sum of sinusoids that includes all possible frequencies. It is not, in reality, possible to generate such a signal, but it can be sufficiently approximated with large amplitude, narrow pulse, to produce the theoretical impulse response in a network to a high degree of accuracy. The symbol for an impulse is  $\delta(t)$ . If an impulse is used as an input to a system, the output is known as the impulse response. The impulse response defines the system because all possible frequencies are represented in the input.

To generate impulse signal in MATLAB, provide the number of samples, and define the impulse signal array. stem (n,x) command will be used to plot the impulse signal.

#### **Unit Step Signal:**

A unit step function is a signal that has a magnitude of zero before zero and a magnitude of one after zero. The symbol for a unit step is u(t). If a step is used as the input to a system, the output is called the step response. The step response can be multiplied with other signals to show how the system responds when an input is suddenly turned on.

A discrete time unit step function may be created by defining the number of samples, and discrete unit step function. The plotting can be done using stem (n,x) command.

#### Periodic Signals:

It is an easy matter to generate periodic signals such as square waves, triangular waves and sinusoidal waves using MATLAB.

Consider first the generation of a sawtooth wave of amplitude A, fundamental frequency fo (measured in radians per second) and phase shift w (measured in radians). The basic command for generating this signal is:

#### A\*sawtooth(fo\*t + w);

The sawtooth wave (or saw wave) is a kind of non-sinusoidal waveform. It is named a sawtooth based on its resemblance to the teeth on the blade of a saw. The convention is that a sawtooth wave ramps upward and then sharply drops. However, there are also sawtooth waves in which the wave ramps downward and then sharply rises. The latter type of sawtooth wave is called a 'reverse sawtooth wave' or 'inverse sawtooth wave'.

Now consider generating a square wave of amplitude A, fundamental frequency fo and phase shift w (in radians). To generate such a signal in MATLAB, we use the command:

#### A\*square(fo\*t + W);

#### **Aperiodic Signals:**

#### **Exponential Signals:**

There are two types of exponential signals: decaying exponentials and growing exponentials. The MATLAB command for decaying exponential Be-at is

#### **B\*exp(-a\*t);**

To generate a growing exponential Be<sup>at</sup>, we use command

**B\*exp(a\*t);** 

In both cases, the exponential parameter 'a' is positive.

#### **Basic Operations on Signals:**

#### **Time Scaling:**

Time scaling compresses and expands a signal by multiplying the time variable by some amount. If that amount is greater than one, the signal becomes narrower and the operation is called compression, while if the amount is less than one, the signal becomes wider and is called expansion. The time scaling property is used in our day-to- day life. For example, in tape recording, the linear changing of scale into twice, speeds up the recording, whereas scaling down reduces the speed of recording.

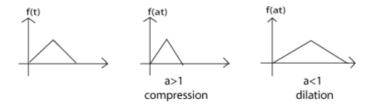


Figure 1: Time scaling of triangular signal

Function x(t)=sawtooth(wt) is plotted in MATLAB by defining the appropriate values of frequency, however, its compressed version can be plotted using x(2t), and expanded version with x(0.5t).

#### Time Reversal

When the time variable is multiplied by a negative number, this operation is the reversal of the time axis, or flipping the signal over the y-axis. The resulting signal is called Time Reversed Signal.

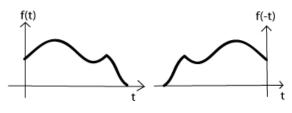


Figure 2: Time Reversal Operation on Signal

Function x(t)=et can be generated in MATLAB, and its time reversed version can be plotted using x(-t).

#### Time Shifting:

Time shifting is the shifting of a signal in time. This is done by adding or subtracting the amount of the shift to the time variable in the function. Subtracting a fixed amount from the time variable will shift the signal to the right (delay) that amount, while adding to the time variable will shift the signal to the left (advance).

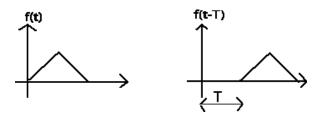


Figure 3: Time Shifting Operation

The function  $x(t)=e^{-3t}$  can be plotted and its time delayed and time advanced version can be plotted using x(t-1) and x(t+1) respectively.

#### LABORATORY TASKS:

- 1. Write a MATLAB code to generate impulse signal in MATLAB.
- 2. Write a MATLAB code for plotting the compressed and expanded version of Square wave by defining the appropriate values of frequency. The compressed version can be plotted using x(2t), and expanded version with x(0.5t).
- **3.** Write a MATLAB code to generate time delayed exponential signal.

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>01</u>			Date:			
	Software Use Rubric						
		Level of Attainment					
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)		
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.		
Skills to use software (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Rarely uses software efficiently.	Occasionally uses software efficiently.	Often uses software efficiently.	Efficiently uses software (syntax, commands, tools, layout)		
Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.		
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors		
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.		

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:-**

To simulate and analyze the Fourier synthesis of square and triangular waves and their spectrum by applying Fourier transform using MATLAB

#### THEORY:-

A square wave spectrum is made of the sum of all the sinusoidal harmonics being odd of the fundamental with decreasing amplitude according to the law of trigonometric Fourier series. In other words the square wave can be obtained by summing up the infinite sine waves as per the following relation:

```
S(t) = \frac{\sin(2\pi Ft)}{1} + \frac{\sin(2\pi 3Ft)}{3} + \frac{\sin(2\pi 5Ft)}{5} + \frac{\sin(2\pi 7Ft)}{7} + \frac{\sin(29Ft)}{9} + \dots
```

The triangular wave can be obtained by summing up the infinite sine waves as per the following relation:

```
S(t) = \cos(2\pi Ft)/1 + \cos(2\pi 3Ft)/32 + \cos(2\pi 5Ft)/52 + \cos(2\pi 7Ft)/72 + \cos(2\pi 9Ft)/92 + \dots
```

#### LABORATORY TASKS:-

Write the code on MATLAB to carry out Fourier synthesis of square wave and analyze its spectrum

- Initiate a row matrix variable, with total number of iterations in the column index, to save and plot the values, For example: out=zeros(1,t)
- Use commands sin(t) or cos(t) to add the harmonics and form a square or triangular wave within a looping command to obtain values at each iteration
- Use fft(out) command for computing Fourier transform to get the frequency components in the above generated square wave
- Finally plot the generated square wave and its spectra using subplot() command.

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.

#### POST LAB TASK:

Write the code on MATLAB to carry out Fourier synthesis of triangular wave and analyze its spectrum.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>02</u>			Date:			
Software Use Rubric							
		Level of Attainment					
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)		
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.		
Skills to use software (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Rarely uses software efficiently.	Occasionally uses software efficiently.	Often uses software efficiently.	Efficiently uses software (syntax, commands, tools, layout)		
Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.		
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors		
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.		

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:-**

To demonstrate the Fourier synthesis of square and triangular wave on hardware module

#### **EOUIPMENT:-**

- Modules T10H
- +/- 12Vdc Supply
- Oscilloscope

#### **THEORY:-**

A square wave spectrum is made of the sum of all the sinusoidal harmonics being odd of the fundamental with decreasing amplitude according to the law of trigonometric Fourier series. In other words the square wave can be obtained by summing up the infinite sine waves as per the following relation:

 $S(t) = \frac{\sin(2\pi Ft)}{1} + \frac{\sin(2\pi 3Ft)}{3} + \frac{\sin(2\pi 5Ft)}{5} + \frac{\sin(2\pi 7Ft)}{7} + \frac{\sin(29Ft)}{9} + \dots$ 

The triangular wave can be obtained by summing up the infinite sine waves as per the following relation:

```
S(t) = \cos(2\pi Ft)/1 + \cos(2\pi 3Ft)/32 + \cos(2\pi 5Ft)/52 + \cos(2\pi 7Ft)/72 + \cos(2\pi 9Ft)/92 + \dots
```

#### LABORATORY TASK:-

#### FOURIER SYNTHESIS OF SQUARE WAVE:

1- Odd harmonics (1, 3, 5, 7, 9): two way switches  $-\frac{0}{+}$  on + and two way switches  $\frac{\sin}{\cos 9}$  on sin.

2- Even harmonics (2, 4, 6, 8): two way switches  $-\frac{0}{+}$  on 0.

3- Connect the oscilloscope with the amplifier output of the fundamental (1st) and adjust the amplitude at 10Vp-p.

4- Connect the oscilloscope with the output of the third harmonic amplifier (3RD) and adjust the amplitude at 10/3

5- Connect the oscilloscope with the output of the 5TH harmonic amplifier (5TH) and adjust the amplitude at 10/5

6- Connect the oscilloscope with the output of the seventh harmonic amplifier (7TH) and adjust the amplitude at 10/7

7- Connect the oscilloscope with the output of the 9th harmonic amplifier (9TH) and adjust the amplitude at 10/9

8- Connect the oscilloscope with OUT and check that there is the signal corresponding to the components sum.

9-Remove some harmonics (put the relating two way switch on 0) and check the o/p signal.

#### FOURIER SYNTHESIS OF TRIANGULAR WAVE:

1- Odd harmonics (1, 3, 5, 7, 9): two way switches  $-\frac{0}{+}$  on + and two way switches  $\frac{\sin}{\cos 9}$  on  $\cos -\frac{1}{2}$ 

2- Even harmonics (2, 4, 6, 8): two way switches  $-\frac{0}{+}$  on 0.

3- Connect the oscilloscope with the amplifier output of the fundamental (1st) and adjust the amplitude at 10Vp-p.

4- Connect the oscilloscope with the output of the third harmonic amplifier (3RD) and adjust the amplitude at 10/32

5- Connect the oscilloscope with the output of the 5TH harmonic amplifier (5TH) and adjust the amplitude at 10/52

6- Connect the oscilloscope with the output of the seventh harmonic amplifier (7TH) and adjust the amplitude at 10/72

7- Connect the oscilloscope with the output of the 9th harmonic amplifier (9TH) and adjust the amplitude at 10/92

8- Connect the oscilloscope with OUT and check that there is the signal corresponding to the components sum.

9-Remove some harmonics (put the relating two way switch on 0) and check the o/p signal.

#### **RESULTS:**

Draw waves of each stage below for both square and triangular wave.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>03</u>			Date:	
	Psychomo	tor Domain Asses	sment Rubric-Le	vel P2	
01 '11 O 4		Extent of Achievement			
Skill Sets	0	1	2	3	4
Equipment identification Sensory skill to <i>identify</i> equipment and/or its component for a lab work.	Not able to identify the equipment.				Able to identify equipment as well as its components.
Equipment Use Sensory skills to <i>demonstrate</i> the use of the equipment for the lab work.	Doesn't demonstrate the use of equipment.	Slightly demonstrates the use of equipment.	Somewhat demonstrates the use of equipment.	Moderately de monstrates the use of equipment.	Fully demonstrates the use of equipment.
<b>Procedural Skills</b> <b>Displays</b> skills to act upon sequence of steps in lab work.	Not able to either learn or perform lab work procedure.	Able to slightly understand lab work procedure and perform lab work.	Able to somewhat understand lab work procedure and perform lab work.	Able to moderately understand lab work procedure and perform lab work.	Able to fully understand lab work procedure and perform lab work.
Safety Adherence Adherence to safety procedures.	Doesn't adhere to safety procedures.	Slightly adheres to safety procedures.	Somewhat adheres to safety procedures.	Moderately adheres to safety procedures.	Fully adheres to safety procedures.
Equipment Handling Equipment care during the use.	Doesn't handle equipment with required care.	Rarely handles equipment with required care.	Occasionally handles equipment with required care.	Often handles equipment with required care.	Handles equipment with required care.
Group Work Contributes in a group based lab work.	Doesn't participate and contribute.	Slightly participates and contributes.	Somewhat participates and contributes.	Moderately participates and contributes.	Fully participates and contributes.
Weighted CLO (Psychomo	tor Score)				
Remarks					
Instructor's Signature with	Date				

#### **OBJECTIVE:**

To represent and simulate LTI systems in MATLAB

#### **THEORY:**

Any LTI system can be characterized entirely by a single function called the system's impulse response. The output of the system is simply the convolution of the input to the system with the system's impulse response. This method of analysis is often called the time domain point-of-view. The same result is true of discrete-time linear shift-invariant systems in which signals are discrete-time samples, and convolution is defined on sequences.

Equivalently, any LTI system can be characterized in the frequency domain by the system's transfer function, which is the Laplace transform of the system's impulse response (or Z transform in the case of discrete-time systems). As a result of the properties of these transforms, the output of the system in the frequency domain is the product of the transfer function and the transform of the input. In other words, convolution in the time domain is equivalent to multiplication in the frequency domain.

LTI system theory is good at describing many important systems. Most LTI systems are considered "easy" to analyze, at least compared to the time-varying and/or nonlinear case. Any system that can be modeled as a linear homogeneous differential equation with constant coefficients is an LTI system. Examples of such systems are electrical circuits made up of resistors, inductors, and capacitors (RLC circuits). Ideal spring–mass–damper systems are also LTI systems, and are mathematically equivalent to RLC circuits.

On MATLAB the impulse response of the system can be seen using impulse(sys) command and the system response at any arbitrary input can be viewed by using *lsim(sys,Vin,t)* command.

#### **LABORATORY TASK:-**

Write MATLAB code for a first order passive Low pass filter (series RC circuit) to represent LTI systems.

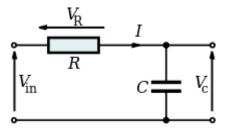


Figure 4: RC Low Pass Filter

Whose transfer function can be given by;

$$H_C(s)=rac{V_C(s)}{V_{
m in}(s)}=rac{1}{1+RCs}$$

Implement its transfer function, Plot it's bode diagram and impulse response and finally check the response when provided square wave as an input.

### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.

### POST LAB TASK:

Write the code on MATLAB to carry out the above mentioned task for first order passive high pass filter.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>04</u>			Date:	
Software Use Rubric					
		]	Level of Attainn	nent	
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.
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Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:**

To identify and validate the properties of LTI (Linear Time Invariant) systems

#### THEORY:

The defining properties of any LTI system are *linearity* and *time invariance*.

• Linearity means that the relationship between the input and the output of the system is a linear map: If input  $x_1(t)$  produces response  $y_1(t)$  and input  $x_2(t)$  produces response  $y_2(t)$  then the *scaled* and *summed* input  $a_1x_1(t)+a_2x_2(t)$  produces the scaled and summed response  $a_1y_1(t)+a_2y_2(t)$  where  $a_1$  and  $a_2$  are real scalars. It follows that this can be extended to an arbitrary number of terms, and so for real numbers  $c_1, c_2, \dots, c_k$ ,

Input  $\sum_k c_k x_k(t)$  produces output  $\sum_k c_k y_k(t)$ 

In particular,

Input  $\int_{-\infty}^{\infty} c_w x_w(t) dw$  produces output  $\int_{-\infty}^{\infty} c_w y_w(t) dw$ 

where  $c_w$  and  $x_w$  are scalars and inputs that vary over a continuum indexed by w. Thus if an input function can be represented by a continuum of input functions, combined "linearly", as shown, then the corresponding output function can be represented by the corresponding continuum of output functions, *scaled* and *summed* in the same way.

• **Time invariance** means that whether we apply an input to the system now or T seconds from now, the output will be identical except for a time delay of the T seconds. That is, if the output due to input x(t) is y(t), then the output due to input x(t-T) is y(t-T). Hence, the system is time invariant because the output does not depend on the particular time the input is applied.

Any LTI system can be characterized entirely by a single function called the system's impulse response. The output of the system is simply the convolution of the input to the system with the system's impulse response. This method of analysis is often called the time domain point-of-view.

The same result is true of discrete-time linear shift-invariant systems in which signals are discrete-time samples, and convolution is defined on sequences.

The response of linear time invariant system can be computed using the convolution integral by convolving the input x(t) with the impulse response h(t) to generate the response or output y(t).

#### LABORATORY TASK:-

Write the code on MATLAB to verify the additive and time invariant properties of LTI (Linear Time Invariant) systems

For additive property, you may initiate input and impulse response as;

t=-1.2:0.0001:1.2; x=zeros(size(t)); x(t>=-1 & t<=1)=1;</pre>

Use conv(x,h) command to get the output from an input. Also, for time invariant property, you may initiate input as;

x1(t>=-0.9 & t<=1.1)=1;

And introduce delay in the output by padding zeros in the first output variable as;

```
pad=zeros(1,0.1/0.0001);
y2=[pad y];
```

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.

#### **POST LAB TASK:**

Write the code on MATLAB to verify the scaling property of LTI system



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>05</u>			Date:		
	Software Use Rubric					
	Level of Attainment					
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)	
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.	
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Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:**

To Simulate and determine the time response of LTI systems to arbitrary inputs

### **THEORY:**

Any LTI system can be characterized entirely by a single function called the system's impulse response. The output of the system is simply the convolution of the input to the system with the system's impulse response. This method of analysis is often called the time domain point-of-view. The same result is true of discrete-time linear shift-invariant systems in which signals are discrete-time samples, and convolution is defined on sequences.

Equivalently, any LTI system can be characterized in the frequency domain by the system's transfer function, which is the Laplace transform of the system's impulse response (or Z transform in the case of discrete-time systems). As a result of the properties of these transforms, the output of the system in the frequency domain is the product of the transfer function and the transform of the input. In other words, convolution in the time domain is equivalent to multiplication in the frequency domain.

### LABORATORY TASK:-

Write a MATLAB code to Simulate and plot the time response of LTI systems to arbitrary inputs.

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>06</u>			Date:			
Software Use Rubric							
		Level of Attainment					
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)		
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.		
Skills to use software (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Rarely uses software efficiently.	Occasionally uses software efficiently.	Often uses software efficiently.	Efficiently uses software (syntax, commands, tools, layout)		
Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.		
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors		
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.		

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:**

To analyze an audio and image signals using Fast Fourier Transform

#### **THEORY:**

The Fourier transform is one of the most useful mathematical tools for many fields of science and engineering. The Fourier transform has applications in signal processing, physics, communications, geology, astronomy, optics, and many other fields. This technique transforms a function or set of data from the time or sample domain to the frequency domain. This means that the Fourier transform can display the frequency components within a time series of data. The Discrete Fourier Transform (DFT) transforms discrete data from the sample domain to the frequency domain. The Fast Fourier Transform (FFT) is an efficient way to do the DFT, and there are many different algorithms to accomplish the FFT. MATLAB uses the FFT to find the frequency components of a discrete signal.

This lab shows how to use the FFT to analyze an audio as well as image files in MATLAB. The audio and image files are provided with the lab. This shows how the Fourier transform works and how to implement the technique in MATLAB.

#### Analysis of Audio Signal:

The audio file is opened using the wavread function, which returns the sampled data from the file, the sampling frequency, and the number of bits used in the A/D converter. Note that the file extension ".wav" does not have to be specified in the function call. The sampling frequency is important for interpreting the data.

The FFT is performed using the "fft" function. MATLAB has no "dft" function, as the FFT computes the DFT exactly. Only the magnitude of the FFT is saved, although the phase of the FFT is useful is some applications. The "fft" function allows the number of points outputted by the FFT to be specified, but in this lab, we will use the same number of input and output points. In the next line, half of the points in the FFT are discarded. This is done for the purposes of this lab, but for many applications, the entire spectrum is interesting. In the following line, the data that will be used for the abscissa is prepared by using the sampling frequency and the number of samples in the time domain. This step is important to determine the actual frequencies contained in the audio data.

Next, the original data are plotted in the time domain and the FFT of the data is plotted. The x-axis is limited to the range [0, 1000] in the plot to show more detail at the peak frequency.

The Fourier transform is a useful tool in many different fields. Two-dimensional Fourier transforms are often used for images as well.

#### Analysis of Image Signal:

#### Steps:

- Read the image.
- Apply Fourier transformation.
- Display log and shift FT images.

#### **Functions Used:**

- imread() inbuilt function is used to read image.
- rgb2gray() converts the image into grayscale.

- fft2() inbuilt function is used to apply forward fourier transform on 2D signal.
- ifft2() inbuilt function is used to apply inverse Fourier transform on 2D signal.
- fftshift() inbuilt function is used to shift corners to center in FT image.
- log() inbuilt function is used to evaluate logarithm of FT complex signal.
- imtool() inbuilt function is used to display the image.

#### **LABORATORY TASK:-**

Write MATLAB codes to analyze audio and image signals using Fast Fourier Transform.

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>07</u> Date:				
	S	oftware Use Ru	bric		
		1	Level of Attainn	nent	
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.
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Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:-**

To compute and display the frequency response of an LTI system (different types of filters) using the Laplace transform

### THEORY:-

The Laplace transform is used for solving differential and integral equations. In physics and engineering it is used for analysis of linear time-invariant systems such as electrical circuits, harmonic oscillators, optical devices, and mechanical systems. In such analyses, the Laplace transform is often interpreted as a transformation from the time-domain, in which inputs and outputs are functions of time, to the frequency-domain, where the same inputs and outputs are functions of complex angular frequency, in radians per unit time. Given a simple mathematical or functional description of an input or output to a system, the Laplace transform provides an alternative functional description that often simplifies the process of analyzing the behavior of the system, or in synthesizing a new system based on a set of specifications. The transfer function of the system can be obtained by taking the Laplace transform of its impulse response.

#### LABORATORY TASK:-

Write the following MATLAB code to compute and plot the frequency response of an LTI system using the Laplace transforms.

*First order Low Pass Filter:* Use RC series circuit with R=10K and C=1uF *First order High Pass Filter:* Use RC parallel circuit with R=10K and C=1uF *Second order Low Pass Filter:* Use cascaded RC series circuit with R1=R2=10K and C1=C2=1uF *Second order High Pass Filter:* Use cascaded RC parallel circuit with R1=R2=10K and C1=C2=1uF

Use laplace(x) command to convert a time domain signal into s-domain

For the transfer function definition use

- a denominator coefficients of transfer function [s^2 s^1 s^0]
- b numerator coefficients of transfer function
- z is zero, p is pole, k is gain

Use freqs(b,a,w) command to get the transfer function and use abs() to get magnitude and angle() to get phase.

Also use loglog(w,mag) command to plot magnitude v/s frequency and semilog(w,phase) command to plot phase v/s frequency.

#### **RESULTS:**

The code results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>08</u> Date:					
	S	oftware Use Ru	bric			
		Level of Attainment				
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)	
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.	
Skills to use software (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Rarely uses software efficiently.	Occasionally uses software efficiently.	Often uses software efficiently.	Efficiently uses software (syntax, commands, tools, layout)	
Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.	
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors	
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.	

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

#### **OBJECTIVE:-**

To identify the I/O characteristic of a low-pass system and analyze the cutting frequency

#### **EOUIPMENTS:-**

- Modules C 17/EV
- +/- 12 V dc Supply
- Oscilloscope
- Function generator
- Digital multimeter

#### **THEORY:-**

The low-pass filter is carried out with an operational amplifier as active element and resistors and capacitors as passive elements.

The diagram of a generic filter is the one of figure 5 which does not specifies the impedance used.

The characteristics of the filter are determined by the impedances used and their value.

The input / output relation for the generic diagram is given by

$$\frac{Vo}{Vin} = \frac{Z_2 Z_4 Z_5}{Z_1 Z_3 Z_5 + Z_1 Z_2 Z_5 + Z_1 Z_2 Z_3 + Z_2 Z_3 Z_5 - Z_2 Z_1 Z_4}$$

Where Vo and Vi respectively indicate the output and input voltage.

This relation is obtained supposing that the input impedance of the operational amplifier is infinite and that its input is not crossed by current and that the amplification is infinite. This fact implies that the differential voltage is null and that the inverting input can be considered as virtually grounded.

With this hypothesis and applying the superimposition theorem you can obtain the above relation after a few simple operations.

This filter is low-pass if Z1, Z3 and Z5 must be resistors while Z2 and Z4 are capacitors.

Changing the generic impedances Z1 of the general formula with the characteristic impedance of the components used we obtain the following general relation for the low-pass filter

D

$$\frac{Vo}{Vin} = \frac{\frac{R_3}{w^2 C_1 C_2}}{R_1 R_2 R_3 + \frac{R_1 R_2 R_3 + R_1 R_3 + R_1 R_2 + R_2 R_3}{jw C_1} + \frac{R_1}{w^2 C_1 C_2}}$$

Where W is the Pulse of the input signal.

The cutting frequency of the filter is determined by the value of the passive components. For the value marked in figure 6, this frequency is about 66 Hz.

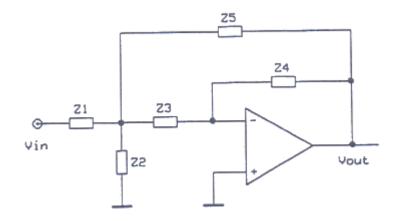


Figure 5: Operational Amplifier Based Low Pass Filter

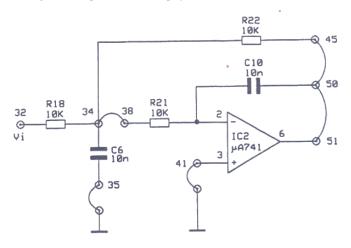


Figure 6: Circuit of OPAMP Based Low Pass Filter

### LABORATORY TASK:-

- Carry out the circuit of figure 6.
- Apply a sine voltage with amplitude of 1 Vpp and frequency of 10 Hz across the circuit.
- Connect one probe of the oscilloscope to the output of the filter and measure the signal
- amplitude
- Report this value in table 1
- Repeat the measurement of the output voltage ( and report it in the proper spaces) for the
- frequencies reported in table 1
- Repeat all values obtained in the table and draw the input / output characteristic of the filter connecting all points.
- From the graph above, calculate the cutting frequency of the filter, defined as the frequency at which the amplification drops to 0.707 times the maximum value.

F (Vin)	Vin	Vo	Vo / Vin
10Hz			
20Hz			
30Hz			
40Hz			
50Hz			
60 Hz			
70 Hz			
100Hz			
200Hz			
500Hz			
1 KHz			
2 KHz			

### Table 1: Observation Table

#### **RESULTS:**

The code results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>09</u>			Date:	
	Psychomo	tor Domain Asses	sment Rubric-Le	vel P2	
01 '11 O (		Extent of Achievement			
Skill Sets	0	1	2	3	4
Equipment identification Sensory skill to <i>identify</i> equipment and/or its component for a lab work.	Not able to identify the equipment.				Able to identify equipment as well as its components.
Equipment Use Sensory skills to <i>demonstrate</i> the use of the equipment for the lab work.	Doesn't demonstrate the use of equipment.	Slightly demonstrates the use of equipment.	Somewhat demonstrates the use of equipment.	Moderately de monstrates the use of equipment.	Fully demonstrates the use of equipment.
<b>Procedural Skills</b> <b>Displays</b> skills to act upon sequence of steps in lab work.	Not able to either learn or perform lab work procedure.	Able to slightly understand lab work procedure and perform lab work.	Able to somewhat understand lab work procedure and perform lab work.	Able to moderately understand lab work procedure and perform lab work.	Able to fully understand lab work procedure and perform lab work.
Safety Adherence Adherence to <i>safety</i> procedures.	Doesn't adhere to safety procedures.	Slightly adheres to safety procedures.	Somewhat adheres to safety procedures.	Moderately adheres to safety procedures.	Fully adheres to safety procedures.
Equipment Handling Equipment care during the use.	Doesn't handle equipment with required care.	Rarely handles equipment with required care.	Occasionally handles equipment with required care.	Often handles equipment with required care.	Handles equipment with required care.
Group Work Contributes in a group based lab work.	Doesn't participate and contribute.	Slightly participates and contributes.	Somewhat participates and contributes.	Moderately participates and contributes.	Fully participates and contributes.
Weighted CLO (Psychomotor Score)					
Remarks					
Instructor's Signature with Date					

#### **OBJECTIVE:-**

To identify the I/O characteristic of a high-pass system and calculate the cutting frequency

#### EQUIPMENT:-

- Modules C 17/EV
- +/- 12 V dc Supply
- Oscilloscope
- Function generator
- Digital multimeter

#### THEORY:-

The ideal high-pass filter is a circuit which has a zero amplification for all those signals which frequency is inferior to a certain frequency f' while it has an amplification different from zero and constant for all those signals which frequencies are higher than f'.

In this case as for the low-pass filter, the circuit which carries out the filter is composed by an operational amplifier and by resistors and capacitors.

Obviously, the real circuit which carries out the high-pass filter can have ideal characteristics: it is enough that the transfer function has a behavior near the one of the ideal filter.

In this case too, we start from the relation between input and output. Consider the relation written for the generic filter which here we write as

$$\frac{Vo}{Vin} = \frac{Z_2 Z_4 Z_5}{Z_1 Z_3 Z_5 + Z_1 Z_2 Z_5 + Z_1 Z_2 Z_3 + Z_2 Z_3 Z_5 - Z_2 Z_1 Z_4}$$

This filter is high-pass if Z1, Z3 and Z5are capacitors while Z2 and Z4 are resistors.

Changing the values of the general relation above described we obtain.

$$\frac{Vo}{Vin} = \frac{\frac{R_1R_2}{jwC_3}}{\frac{1}{jwC_1 * jwC_2 * jwC_3} + \frac{R_1}{jwC_1 * jwC_3} + \frac{R_1}{jwC_1 * jwC_2} + \frac{R_1}{jwC_2 * jwC_3} - \frac{R_1R_2}{jwC_1}}$$

Now the characteristic of the filter, i.e. the cutting frequency and the transfer function will be determined by the value of the passive components used.

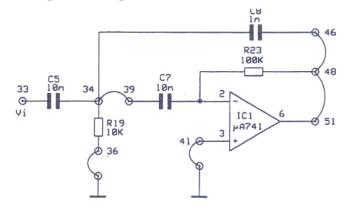


Figure 7: OPAMP Based High Pass Filter

### LABORATORY TASK:

- 1. Carry out the circuit of figure 7.
- 2. Apply a sine voltage with amplitude of 1 Vpp and frequency of 1 KHz across the input of the circuit.
- 3. Connect one probe of the oscilloscope to the output of the filter and measure the amplitude of the signal
- 4. Fill table 2 for the value. Repeat the output voltage measurement (and report it in the proper spaces) for all the frequencies shown in table.
- 5. Report the obtain values of figure 7 and connect all the points to obtain a graphic of the variation of the output voltage with frequency.
- 6. From the graph above, calculate the cutting frequency of the filter, defined as the frequency at which the amplification drops to 0.707 times the maximum.

F (Vin)	Vin	Vo	Vo / Vin
1KHz			
2KHz			
5KHz			
10KHz			
20KHz			
50KHz			
100 KHz			
200 KHz			

 Table 2: Observation Table

#### **RESULTS:**

The code results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>10</u>			Date:	
	Psychomo	tor Domain Asses	sment Rubric-Le	vel P2	
01.11.0		Extent of Achievement			
Skill Sets	0	1	2	3	4
Equipment identification Sensory skill to <i>identify</i> equipment and/or its component for a lab work.	Not able to identify the equipment.				Able to identify equipment as well as its components.
Equipment Use Sensory skills to <i>demonstrate</i> the use of the equipment for the lab work.	Doesn't demonstrate the use of equipment.	Slightly demonstrates the use of equipment.	Somewhat demonstrates the use of equipment.	Moderately de monstrates the use of equipment.	Fully demonstrates the use of equipment.
<b>Procedural Skills</b> <b>Displays</b> skills to act upon sequence of steps in lab work.	Not able to either learn or perform lab work procedure.	Able to slightly understand lab work procedure and perform lab work.	Able to somewhat understand lab work procedure and perform lab work.	Able to moderately understand lab work procedure and perform lab work.	Able to fully understand lab work procedure and perform lab work.
Safety Adherence Adherence to safety procedures.	Doesn't adhere to safety procedures.	Slightly adheres to safety procedures.	Somewhat adheres to safety procedures.	Moderately adheres to safety procedures.	Fully adheres to safety procedures.
Equipment <u>Handling</u> Equipment care during the use.	Doesn't handle equipment with required care.	Rarely handles equipment with required care.	Occasionally handles equipment with required care.	Often handles equipment with required care.	Handles equipment with required care.
Group Work Contributes in a group based lab work.	Doesn't participate and contribute.	Slightly participates and contributes.	Somewhat participates and contributes.	Moderately participates and contributes.	Fully participates and contributes.
Weighted CLO (Psychomotor Score)					
Remarks					
Instructor's Signature with Date					

#### **OBJECTIVE:-**

To identify the I/O characteristic of a band-pass system and calculate the cutting frequency

#### EOUIPMENT:-

- Modules T 10A-T 10B
- +/- 12 V dc Supply
- Oscilloscope

#### THEORY:-

#### **Ceramic Filter:-**

A ceramic filter is a band pass filter using a piezoelectric ceramic material. Important parameters of ceramic filter are input & output impedance. The center frequency of ceramic is 455 KHz. The response curve can be obtained by applying a variable freq across i/p & detecting amplitude at o/p. The attenuation measured at different frequency is given by:

### LABORATORY TASK:-

- 1. Supply the modules with dc supply. Carryout the following presetting: VCO1: switch on 500 KHz, level about 2Vp-p, Freq.450KHz.
- 2. Apply a signal of 455 KHz corresponding to the central frequency of the ceramic filter.
- If Vo and Vi are the peak-to-peak voltages measured across the output and the input of the filter. The attenuation A of the filter at 455 KHz is given by. A=Vo/Vi and dB= 20 log (Vo/Vi)
- 4. Repeat the measurement carried out in the last step varying the frequency from 445 to 465 KHz at step of 1 KHz.
- 5. Calculate AdB in correspondence to each frequency and report all in the following table.

FREQUENCY (KHz)	OUT PUT VOLTAGE (Vo)	INPUT VOLTAGE (Vi)	AdB=20log (Vo/Vi)
445			
446			
447			
448			
449			
450			
451			
452			
453			

Table 3: Observation Table

454 455 456		
455		
456		
457 458 459		
458		
459		
460		
461		
462		
463 464		
464		

6. With the data in the table 3 and plot a graph setting  $A_{dB}$  on the Y axis and frequency on the x-axis, you obtain the frequency response curve of the filter.

#### **RESULTS:**

The code results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>11</u>			Date:		
	Psychomo	tor Domain Asses	sment Rubric-Le	vel P2		
Skill Sets		Extent of Achievement				
Skill Sets	0	1	2	3	4	
Equipment identification Sensory skill to <i>identify</i> equipment and/or its component for a lab work.	Not able to identify the equipment.				Able to identify equipment as well as its components.	
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Group Work Contributes in a group based lab work.	Doesn't participate and contribute.	Slightly participates and contributes.	Somewhat participates and contributes.	Moderately participates and contributes.	Fully participates and contributes.	
Weighted CLO (Psychomotor Score)						
Remarks						
Instructor's Signature with	Date					

#### **OBJECTIVE:**

To analyze LTI system using z-transform and Inverse z-transform

#### **THEORY:**

The Z-transform converts a time domain signal, which is a sequence of real or complex numbers, into a complex frequency domain representation. It can be considered as a discrete-time equivalent of the Laplace transform.

Due to its convolution property, the z-transform is a powerful tool to analyze LTI systems.

$$y[n] = h[n] * x[n] \xrightarrow{\mathcal{Z}} Y(z) = H(z)X(z)$$

**Zplane:** zplane(z, p) plots the zeros specified in column vector z and the poles specified in column vector p in the current figure window. The symbol 'o' represents a zero and the symbol 'x' represents a pole. The plot includes the unit circle for reference.

Find the poles and zeros of the following pulse transfer function and plot them onto the z-plane.

$$H(z) = \frac{2 - z^{-1}}{1 - 0.1 z^{-1} - 0.02 z^{-2}}$$

The pole-zero plot of a rational z-transform H(z) can be readily obtained using the function zplane and command **zplane(num,den)** will be used where num and den are row vectors containing the coefficients of the numerator and denominator polynomials of H(z) in ascending powers of z - 1. The function tf2zp can be used to determine the zeros and poles of a rational z-transform H(z). The program statement to use is [z,p,k] = tf2zp(num,den) where num and den are row vectors containing the coefficients of the numerator and denominator polynomials of H(z) in ascending powers of z - 1 and the output file contains the gain constant k and the computed zeros and poles given as column vectors z and p, respectively.

This function displays the poles and zeros of discrete-time systems in MATLAB:

syms z n a=ztrans(1/16^n)

Inverse Z-Transform: It can be calculated using the functions:

syms Z n

iztrans(3\*Z/(Z+1))

#### **Pole Zero Diagrams For a Function In Z Domain:**

Z plane command computes and display the pole-zero diagram of Z function using **Zplane(b,a)**.

To display the pole value, **use root(a)** 

To display the zero value, **use root(b)** 

#### Factored Z-transform / zeros, poles to rational Z-transform:

It is the inverse of the above case, when the transfer function is given in factored form and it is required to convert in rational form then a single 'MATLAB' command can serve the purpose.

For building up transfer function in rational form we find the poles and zeros of above system simply by using MATLAB 'root' command. Or simply we have poles and zeros of the given system. We can find the transfer function in factored form.

MATLAB command that converts poles and zeros of the system in to transfer function is 'zp2tf'

#### **Rational Z-transform to partial fraction form:**

This technique is usually used while taking the inverse Z-transform and when the order 'H(z)' is high so that it is quite difficult to solve it mathematically.

MATLAB command that converts rational z-transform in to partial fraction form is 'residuez'

#### Partial fraction form to Z-transform:

This technique is used when it is required to convert partial fraction expression in to rational Z-transform.

MATLAB command that converts partial fraction form into rational z-transform is 'residuez'.

#### LAB TASKS:

Task#1: Find poles and zeros of the following pulse transfer function and plot them onto the z-plane.

$$H(z) = \frac{2.25 - 2.1z^{-1} - 3.95z^{-2} - 1.6z^{-3} - 0.2z^{-4}}{4 - 2.96z^{-1} + 0.8z^{-2} - 0.1184z^{-3} - 0.0064z^{-4}}$$

Task#2: Determine the partial fraction expansion of the z-transform G(z) given by

$$G(z) = \frac{18z^3}{18z^3 + 3z^2 - 4z - 1}$$

#### **RESULTS:**

The code, results / output of all tasks must be attached with this lab.



Course Code and Title:

TC-202 - Signals and Systems

Laboratory Session No.	<u>12</u> Date:				
	S	oftware Use Rul	bric		
		1	Level of Attainn	nent	
Criterion	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)
Identification of software menu (syntax, components, commands, tools, layout etc.).	Can't identify software menus.	Rarely identifies software menus.	Occasionally identifies software menus.	Able to identify software menus.	Perfectly able to identify software menus.
Skills to use software (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Rarely uses software efficiently.	Occasionally uses software efficiently.	Often uses software efficiently.	Efficiently uses software (syntax, commands, tools, layout)
Adherence to safety procedures and handling of equipment (computing unit, peripheral devices, and other equipment in lab).	Doesn't handle equipment with required care and safety.	Rarely handles equipment with required care and safety.	Occasionally handles equipment with required care and safety.	Often handles equipment with required care and safety.	Handles equipment with required care and safety.
Ability to troubleshoot software errors (detection and debugging).	Not able to troubleshoot the errors	Rarely able to troubleshoot the errors	Occasionally able to troubleshoot the errors	Often able to troubleshoot the errors	Fully able to troubleshoot the errors
Analysis and interpretation of results/outputs.	Not able to analyze and interpret results/outputs.	Rarely able to perform the analysis and interpretation.	Occasionally able to perform the analysis and interpretation.	Often able to perform the analysis and interpretation.	Perfectly able to perform the analysis and interpretation.

Weighted CLO (Score)	
Remarks	
Instructor's Signature with Date	

# LAB SESSION 13 (Open Ended Lab)

### **OBJECTIVE:**

The aim of this open ended lab is to model and investigate LTI system by designing and analyzing the RLC filter transfer function and its frequency response

#### THEORY:

Simulink is a software package for modeling, simulating, and analyzing dynamic systems. It supports linear and nonlinear systems, modeled in continuous time, sampled time, or a hybrid of the two. One can easily build, analyze and visualize models.

An RLC circuit is a combination of many resistors, inductors, and capacitors. This kind of circuits have a wide range of uses. Oscillators, tuners for televisions and radio receivers, and filters are a few of the more crucial ones. Band-pass and band-stop filters may be made with RLC circuits as well.

The RLC filter is typically referred to as a second order circuit, which indicates that circuit characteristics like voltage and current may be represented by a second-order differential equation.

#### **DELIVERABLES:**

- You are required to model the differential equations for the characteristics of RLC second order Band Pass and Band Stop filter and design its circuitry with suitable values of the components
- Evaluate their Transfer Function
- Provide pole-zero information and plot them
- Simulate them using Simulink environment of MATLAB
- Analyze their Transfer function and frequency response of the LTI system
- Calculate their cut off frequency and Bandwidth
- Provide the Bode plots for phase and frequency response

#### **RESULTS:**

The circuit, results / output of task must be attached with this lab



### NED University of Engineering & Technology Department of <u>Telecommunications</u> Engineering Course Code & Title: <u>TC-202-Signals and Systems</u> Assessment Rubric for OEL

	Level of Attainment			
Criterion	Below Average (0)	Good (1)	Excellent (2)	
Modelling of differential equations for the particular filter designing.	Unable to model the differential equations properly.	Somewhat modelled the equations but still have some mistakes.	Perfectly modelled the differential equations as per the filter designed.	
Evaluate transfer function and plot the pole zero locations.	Unable to estimate the transfer function and thus poorly plotted pole zero locations.	Transfer function has few discrepancies hence did not plot the pole zero correctly.	Correctly evaluated the transfer function and also the plot is accurate.	
Skills to use Simulink software to simulate the filter (schematic, syntax, commands, tools, layout) efficiently.	Can't use software efficiently.	Occasionally uses software efficiently.	Efficiently uses software (syntax, commands, tools, and layout).	
Literature Review and Report Writing.	No background information is provided. No organization, sequencing, or structured content.	All major sections of the relevant content are included, but not covered in as much depth, or as explicit, as expected.	Well organized and appropriate content is covered in depth without being redundant. Sources are cited where required.	
Interpretation of results/graphs.	Not able to analyze and interpret bode plot for phase and frequency and incorrectly calculated the Bandwidth.	Able to perform the analysis and interpretation but with inefficiency.	Perfectly able to do the graphical analysis and Bandwidth calculations.	

Student's Name: \_\_\_\_\_ Roll No.: \_\_\_\_\_

Total Score = \_\_\_\_\_

Instructor's Signature: