Department of Telecommunications Engineering
NED University of Engineering & Technology

LABORATORY WORKBOOK

For the Course

DIGITAL SIGNAL PROCESSING
(TC-212)

Instructor Name:

Student Name:

Roll Number: **Batch:**

Semester: **Year:**

Department:

LABORATORY WORKBOOK

For the Course

<u>DIGITAL SIGNAL PROCESSING</u>

<u>(TC-212)</u>

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Approved By:
The Board of Studies of Department of Telecommunications Engineering

CONTENTS

LAB SESSION 01

Objective:-

To analyze the effect of Sampling and Aliasing in discrete time sinusoid.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Signals are physical quantities that carry information in their patterns of variation. Continuous-time signals are continuous functions of time, while discrete-time signals are sequences of numbers. If the values of a sequence are chosen from a finite set of numbers, the sequence is known as a digital signal. Continuous-time, continuous-amplitude signals are also known as analog signals.
Signal processing is concerned with the acquisition, representation, manipulation, transformation, and extraction of information from signals. In analog signal processing these operations are implemented using analog electronic circuits. Digital signal processing involves the conversion of analog signals into digital, processing the obtained sequence of finite precision numbers using a digital signal processor or general purpose computer, and, if necessary, converting the resulting sequence back into analog form. When stored in a digital computer, the numbers are held in memory locations, so they would be indexed by memory address. In order to bridge the gap between analog and digital domain, we have to perform two basic operations Sampling and Quantization.

Sampling: It is the process of acquiring data at discrete (finite) instants of time

Sampling Theorem: A continuous time signal $x(t)$ can be reconstructed exactly from its samples $x(n)=x(nT_s)$, if the samples are taken at a rate $F_s=1/T_s$ that is greater than 2xFmax.

Aliasing: A common problem that arises when sampling a continuous signal is aliasing, where a sampled signal has replications of its sinusoidal components which can interfere with other components. It is an effect that causes two discrete time signals to become indistinct due to improper sampling ($f_d>1/2$). Aliasing also occurs on television whenever we see a car whose tires appear to be spinning in the wrong direction. A television broadcast can be thought of as a series of images, sampled at a regular rate, appearing on screen. If the wheels happen to rotate less than a full circle between frames (images), then they appear to be turning slowly in the opposite direction.

Laboratory Task:-

1. Simulate and plot two continuous time (CT) sinusoids of 10 Hz and 110 Hz for $0 < t < 0.2$ sec.
2. Sample both sinusoids at $F_s = 100$ samples/sec and plot them in discrete form.
3. Observe and note the aliasing effects.
4. Explore and learn.

Task-1:
Consider the following CT signal: $x(t) = \sin(2\pi F_o t)$. The sampled version will be:
$x(n) = \sin(2\pi F_0/F_s \, n)$, where n is a set of integers and sampling interval $T_s=1/F_s$.

Plot the signal x(n) for n = 0 to 99 for F_s = 5 kHz and F_1 = 0.5, 2, 3 and 4.5 kHz. Explain the similarities and differences among various plots. Also mention that whether aliasing occurs or not.

Task-2:
Consider the given CT signal: x(t) = sin ($2\pi F_o t$). Suppose that F_0 = 2 kHz and F_s = 50 kHz.
i) Plot the signal x(n). What will be the discrete frequency f_d of the signal x(n)?
ii) Plot the signal y(n) created by taking the even numbered samples of x(n). Is this a sinusoidal signal? Why? If so, what is the frequency?

Result:-

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>01</u>   Date: _____

Software Use Rubric					
Criterion	**Level of Attainment**				
	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	

LAB SESSION 02

Objective:-

To study the Discrete time convolution.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

A ***discrete-time system*** is a computational process or algorithm that transforms or maps a sequence x(n), called the input signal, into another sequence y(n), called the output signal. In practice, a discrete-time system is a numerical algorithm that processes an input sequence x(n), to produce an output sequence y(n).

A system is called ***causal*** if the present value of the output does not depend on future values of the input, that is, $y(n_0)$ is determined by the values of $x(n)$ for $n \leq n_0$, only. If the output of a system depends on future values of its input, the system is ***noncausal***.

Causality implies that if x(n) = 0 for $n < n_0$, then y(n) = 0 for $n < n_0$; that is, a causal system cannot produce an output before the input is applied. Clearly, this problem does not exist if the entire input sequence is already stored in memory. Although causality is necessary for the real-time implementation of discrete-time systems, it is not really a problem in off-line applications where the input signal has been already recorded.

The main premise of this lab is that the response of a linear time-invariant (LTI) system to any input can be determined from its response $h(n)$ to the unit sample sequence $\delta(n)$, using a formula known as ***convolution summation***. The sequence $h(n)$, which is known as ***impulse response***, can also be used to infer all properties of a linear time-invariant system.

If we know the impulse response of an LTI system, we can compute its response to any input without using the actual system $y(n) = x(n)*h(n)$. Furthermore, if we have no access to the internal implementation of the system (that is, we treat the system as a black box), we can try to "reverse-engineer" the system from its impulse response. Convolution describes how a linear time-invariant system modifies the input sequence to produce its output.
Convolving two waveforms in time domain is equivalent to multiply their spectra (frequency content) in frequency domain.

Laboratory Task:-

1. Consider the impulse response of a system as $h(n) = \{\mathbf{3}, 2, 1, -2, 1, 0, -4, 0, 3\}$
2. For $x(n) = \{\mathbf{1}, -2, 3, -4, 3, 2, 1\}$, find the output $y(n)$.
3. Modify the code such that $h(n) = \{3, 2, \mathbf{1}, -2, 1, 0, -4, 0, 3\}$ (origin is shifted), and check the causality property .

Task 1:
What will happen if we provide input x(n) = {0,0,1,0,0} into the given system.

Task-2
a) Can you prove commutative property of the convolution?
b) Modify the code to prove Associative and Distributive properties of the convolution.

Task-3: What is Convolution reverb? Also explain how special effects can be created in sound using convolution?

Task-4: Discuss the difference between echo and reverberation?

Result:-
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Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>02</u>   Date: _____

Software Use Rubric					
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LAB SESSION 03

Objective:-

To analyze the effects of Quantization in Discrete Time Continuous valued signal

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

The digitization of any real world signal usually involves two stages: sampling, i.e. the measurement at discretely spaced time intervals, and quantization, i.e. the transformation of the measurements (amplitudes) into finite-precision numbers (allowed discrete levels), such that they can be represented in computer memory. Quantization leads to a loss in the signal quality, because it introduces a "quantization error" or "quantization noise" or "distortion". The number of bits used for each sample, called the bit depth (bits/sample), determines the precision with which the sample amplitudes can be represented. Each bit in a binary number holds either a 1 or a 0. The largest decimal number that can be represented with an b-bit binary number is 2^b - 1, and the number of different values that can be represented is 2b. For example, the decimal values that can be represented with an 8-bit binary number range from 0 to 255, so there are 256 different values. A bit depth of 8 allows 256 different discrete levels at which samples can be approximated or recorded. A bit depth of 16 allows 2^{16} = 65,536 discrete levels, which in turn provides much higher precision than a bit depth of 8. With a bit depth of 3, 2^3 = 8 quantization levels ranging from -4 to 3 are possible. By convention, half of the quantization levels are below the horizontal axis (that is 2^b-1, of the quantization levels). One level is the horizontal axis itself (level 0), and 2^b-1 -1 levels are above the horizontal axis. Quantization error represents the quality of quantization process but the total error may also turn out to be zero, so another quality deciding parameter is used which is called signal-to-quantization-noise-ratio (SQNR) and it can be defined as SQNR = 10 log10 (P_x/P_e) where P_x and P_e are average powers of the DTCV and quantization error signal.

$$P_x = \frac{1}{N}\sum_{n=0}^{N-1}|x(n)|^2 \text{ and } P_e = \frac{1}{N}\sum_{n=0}^{N-1}|x_q(n) - x(n)|^2$$

Laboratory Task:-

1. Simulate a DTCV sinusoid of 1/50 cycles/sample with length of the signal be 500 samples.
2. Choose the method of Quantization (round-off, floor & ceil) and apply to the signal generated above.
3. Compute the error signals and SQNR.
4. Explore and observe.

Task 1: Takes method of quantization from user as an input.

Task-2: Consider the sampled signal: x(n)= sin(2πf$_d$n)
a) For f$_d$ = 1/50 cycles/sample and N = 200 samples, write a program to quantize the signal x(n), using round-off to 64, 128 and 256 quantization levels. In each case plot the signals x(n), x$_q$(n)and x$_e$(n)and compute the corresponding SQNR.
b) Repeat step (a) by using floor instead of rounding.

c) Comment on the results obtained in steps (a) and (b).

d) Compare the experimentally measured SQNR with the theoretical SQNR predicted by the following formula and comment on the differences and similarities.

$SQNR = 1.76 + 6.02 \times b$ [Note: b = No. of bits of Quantizer or bit depth]

Task-3:

a) Develop a MATLAB code to quantize both speech and audio signals (other than speech) to the desired number of levels.

b) Play both the original and the quantized sounds in each case when rounding and floor are used. Which quantization method leads to the best quality sound after quantization?

c) For both recorded speech signal and any ringtone, is there a point at which the signal quality deteriorates drastically? At what point (if any) does it become incomprehensible?

d) Which signal's quality deteriorate faster as the number of levels decreases?

e) What improvement in terms of SQNR can we expect from adding one bit in the quantizer?

f) Do you think 4 bits/sample is acceptable for telephone systems? What about 2 bits/sample?

Task-4:

Develop a MATLAB code to import any image file and quantize it to the fewer bits/pixel than original and observe the images. For example an 8-bit image can be quantized to 7, 6, 5, 4 bits/pixel. Print hard copies of the 7, 6, 5, and 4 bits/pixel images, as well as the original. Describe the artefact (errors) that appear in the image as the number of bits is lowered? Also note the number of bits/pixel at which the image quality noticeably deteriorates.

Result:-

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>03</u>   Date: _____

Criterion	Level of Attainment				
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Above table title: **Software Use Rubric**

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

LAB SESSION 4

Objective:-

To analyze the Discrete Time Correlation with Application

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Correlation is given as:

$$r_{xy}(l) = \sum_{n=-\infty}^{\infty} x(n)y(n-l) = \sum_{n=-\infty}^{\infty} x(n+l)y(n); \text{ Where 'l' is the lag.}$$

This is called cross-correlation and it gives the magnitude and location of similarity between two signals. The correlation between y(n) and x(n) is not necessarily the same as the correlation between x(n) and y(n).It is given as:

$$r_{yx}(l) = \sum_{n=-\infty}^{\infty} y(n)x(n-l) = \sum_{n=-\infty}^{\infty} y(n+l)x(n)$$

Generally, $r_{xy}(l) = r_{yx}(-l)$.

These two are the same when x(n) and y(n) are the same signals or when x(n) and y(n) are even symmetric signals. The length of the resulting correlation sequence is N+M-1, where, N and M are the lengths of the two signals. Correlation may also be computed using convolution algorithm with a modification that we need to fold one of the signals before applying the convolution.

Mathematically, $r_{xy}(n) = x(n) * y(-n)$.

Laboratory Task:-

Task 1:
Detection of periodic signal buried in noise. Given a sine wave signal (amplitude and frequency of your choice and 100 samples on x-axis).
   a) Apply auto-correlation of the sine wave signal and plot the output
   b) Generate AWGN signal of equal number of samples (100)
   c) Plot the noise signal
   d) Add the sine wave signal (generated previously) to the noisy signal and plot the graph showing the corrupt signal
   e) Now, apply auto-correlation of the sine + noise signal and plot the output. What do you observe? Write your conclusion.

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>04</u>   Date: _____

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Remarks	
Instructor's Signature with Date	

LAB SESSION 5

Objective:-

To study the Discrete Fourier Transform using an audio signal.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

With the increased use of digital computers and specialized hardware in digital signal processing (DSP), we are normally interested in those transforms that are suitable for digital computations. Because of the continuous nature of spectrum provided by DTFT, direct implementation of the DTFT is not suitable on such digital devices. In this lab we apply the Discrete Fourier Transform (DFT), which can be computed efficiently on digital computers and other DSP boards. The most distinguishing feature of DFT is that it can be applied to any type of signal whether periodic or non-periodic. It simply treats all signals as the time limited or finite length sequence which corresponds to signal of interest/region of interest/sample interval denoted by N.

The DFT is an extension of the DTFT for time-limited sequences with an additional restriction that the spectrum is discretized to a finite set of values. The number of the frequency samples can have any value, but is typically set equal to the length of the time-limited sequence x(n).

DFT plays a central role in spectral analysis, the implementation of digital filters, and a variety of other signal processing applications. DFT is a powerful reversible operation for finite segments of discrete time sequences. Many "fast" algorithms have been developed for computing the DFT, and collectively these are known as Fast Fourier Transform (FFT) algorithms. The discovery of FFT algorithms established the DFT as one of the fundamental tools in digital signal processing.

DFT analysis equation: $X(k) = \sum_{n=0}^{N-1} x(n)e^{\frac{-j2\pi kn}{N}} = \sum_{n=0}^{N-1} x(n)W_N^{nk}$; $k = 0, 1, 2, \cdots, N\text{-}1$

DFT synthesis equation: $x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{\frac{-j2\pi kn}{N}} = \sum_{n=0}^{N-1} X(K)W_N^{-nk}$; $n = 0, 1, 2, \cdots, N\text{-}1$

Frequency resolution is given by $\Delta F = \dfrac{F_s}{N}$ (Hz)

Analysis frequencies or bin frequencies are given by $F_k = k \times \Delta F$; $k = 0, 1, 2, \cdots, N\text{-}1$

Note: Always keep in mind that an FFT algorithm is not a different mathematical transform: it is simply an efficient means to compute the DFT.

Laboratory Task:-

Task 1:
1. Load an audio file 'noisy.wav' into MATLAB.
2. There is a tone added to the speech in this file. The objective is to find the frequency of this tone.
3. Computing the DFT of this signal
4. Generating frequency vector in Hz.
5. Displaying the spectrum (DFT) of audio file and observing the frequency of the added tone.

Task 2:
1. Try to remove the tone spikes by observing the spectra of the noisy signal.
2. Take Inverse DFT (ifft) of the modified spectra.
3. Listen to this new time-domain signal and see if the tone noise is removed.

Result:-

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>05</u>   Date: _____

Software Use Rubric					
Criterion	**Level of Attainment**				
	Below Average (1)	**Average (2)**	**Good (3)**	**Very Good (4)**	**Excellent (5)**
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Instructor's Signature with Date	

LAB SESSION 06

Objective:-

To study the Discrete Fourier Transform & Circular convolution.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Circular convolution is another way of finding the convolution sum of two input signals. It resembles the linear convolution, except that the sample values of one of the input signals is folded and right shifted before the convolution sum is found.

Also note that circular convolution could also be found by taking DFT of the two input signals and finding the product of two frequency domain signals. The inverse DFT of the product would give the output of the signal in the time domain which is the circular convolution output.

Laboratory Task:-

Task 1:

Consider the following given sequences, $x_1(n) = \{\underset{\uparrow}{1}, 2, 1, 2\}$ and $x_2(n) = \{\underset{\uparrow}{1}, 2, 3, 4\}$ and apply circular convolution. Include relevant figures/plots in your results.

Task 2:
Verify the result using DFT method. Include relevant figures/plots in your results.

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>06</u>   Date: _____

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LAB SESSION 07

Objective:-

To analyze spectral analysis of Windowing, Zero-Padding, and FFT

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Spectral Analysis of Signals:
It is very common for information to be encoded in the sinusoids that form a signal. This is true of naturally occurring signals, as well as those that have been created by humans. Many things oscillate in our universe. For example, speech is a result of vibration of the human vocal cords; stars and planets change their brightness as they rotate on their axes and revolve around each other; ship's propellers generate periodic displacement of the water, and so on. The *shape* of the time domain waveform is not important in these signals; the key information is in the *frequency*, *phase* and *amplitude* of the component sinusoids. The DFT is used to extract this information.

Windowing function in Signal Processing:
In signal processing, a window function (also known as an apodization function or tapering function) is a mathematical function that is zero-valued outside of some chosen interval. For instance, a function that is constant inside the interval and zero elsewhere is called a rectangular window, which describes the shape of its graphical representation. When another function or waveform/data-sequence is multiplied by a window function, the product is also zero-valued outside the interval: all that is left is the part where they overlap, the "view through the window".
In typical applications, the window functions used are non-negative, smooth, "bell-shaped" curves. Rectangle, triangle, and other functions can also be used. A more general definition of window functions does not require them to be identically zero outside an interval, as long as the product of the window multiplied by its argument is square integrable, and, more specifically, that the function goes sufficiently rapidly toward zero.

Zero Padding:
Zero padding is a simple concept; it simply refers to adding zeros to end of a time- domain signal to increase its length.
MATLAB CODE
   * **w = blackman(M)** returns the M-point Blackman window function in array w.

Laboratory Task:-

Task 1:
Assume M = 64, Zero padding factor = 8. Plot the spectrum (normalized frequency) using Blackman Window.

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

Course Code and Title: <u>(TC-212) Digital Signal Processing</u>

Laboratory Session No. <u>07</u>   Date: _____

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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 8(a)

Objective:-

Examine the relationship between Laplace and CTFT.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

The Laplace Transform of a general continuous time signal x (t) is defined as;

$$X(s) = \int_{-\infty}^{\infty} x(t)e^{-st}dt$$

Where the complex variable s=δ + j w, with δ and w the real and imaginary parts. CTFT is a subset of Laplace when δ =0. Since 'δ' information is not present in CTFT, therefore information about stability can only be obtained from Laplace. If pole lies on L.H.S of s-plane, system is stable. If pole lies on R.H.S of s-plane, system is unstable. If pole lies on y(jw)-axis, system is marginally stable or oscillatory. If system has FIR, it is stable. If system is IIR, it can be stable or unstable.

Laboratory Task:-

Make a folder at desktop and name it as your current directory within MATLAB.

Task 1:
Generate pole zero constellation in s-plane.
Plot corresponding Frequency (Bode magnitude) response.
Plot impulse response and determine that the system is FIR or IIR.

Task 2:
Modify location of poles in s-plane to observe the corresponding change in frequency and impulse response*.*

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

LAB SESSION 8(b)

Objective:-

Examine the relationship between z-transform and DTFT.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

The z-transform of a general discrete time signal x(n) is defined as;

$$X(z) = \sum_{n=-\infty}^{\infty} x(n)z^{-n}$$

Where the complex variable $z = r < \omega$, with r the radius and w the angle. DTFT is a subset of z-transform when r =1. Since 'r' information is not present in DTFT, therefore information about stability in discrete time can only be obtained from z-transform. If pole lies inside the unit circle, system is stable. If pole lies outside the unit circle, system is unstable. If pole lies at the unit circle, system is marginally stable or oscillatory. If system has FIR, it is stable. If system is IIR, it can be stable or unstable.

Laboratory Task:-

Make a folder at desktop and name it as your current directory within MATLAB.

Task 1:
Generate pole zero constellation in z-plane.
Plot corresponding Frequency (Bode magnitude) response.
Plot impulse response and determine that the system is FIR or IIR.

Task 2:
Modify location of poles in z-plane to observe the corresponding change in frequency and impulse response. Change the location of poles from inside the unit circle to outside and at the unit circle and observe the effects.

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

Course Code and Title: (TC-212) Digital Signal Processing

Laboratory Session No. 08   Date: _____

Software Use Rubric					
Criterion	**Level of Attainment**				
	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)
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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 09

Objective:-

To design FIR filters using MATLAB.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Digital filters refers to the hard ware and software implementation of the mathematical algorithm which accepts a digital signal as input and produces another digital signal as output whose wave shape, amplitude and phase response has been modified in a specified manner.

Digital filter play very important role in DSP. Compare with analog filters they are preferred in number of application due to following advantages.
- Truly linear phase response
- Better frequency response
- Filtered and unfiltered data remains saved for further use.

There are two type of digital filters.
- FIR (finite impulse response) filter
- IIR (infinite impulse response) filter

1. FIR filters are Finite Impulse Response filters with no feedback, whereas IIR contains feedback.
2. Transfer function of FIR filter does not contain any non-trivial poles. Their frequency response is solely dependent on zero locations. IIR filters contain poles as well as zeros.
3. As there are no poles, FIR filters cannot become unstable; however, IIR filters can become unstable if any pole lies outside the unit circle in z-plane.
4. More number of coefficients is needed to design the desired filter in FIR than IIR.

Commands used in FIR Filter Design

FIR1 designs FIR filter using the window method. B = FIR1(N,W_n) designs an Nth order low pass FIR digital filter and returns the filter coefficients in length N+1 vector B. **The cutoff frequency W_n must be between $0 < W_n < 1.0$, with 1.0 corresponding to half the sample rate**. The filter B is real and has linear phase. The normalized gain of the filter at Wn is -6 dB.

B = FIR1(N,W_n,'high') designs an Nth order highpass filter.

You can also use B = FIR1(N,W_n,'low') to design a lowpass filter.
If Wn is a two-element vector, $W_n = [W_1 \ W_2]$, FIR1 returns an order N bandpass filter with passband $W_1 < W < W_2$.

B = FIR1(N,Wn,'stop') is a bandstop filter if Wn = $[W_1 \ W_2]$. You can also specify If W_n is a multi-element vector, $W_n = [W_1 \ W_2 \ W_3 \ W_4 \ W_5 \ ... \ W_N]$, FIR1 returns an order N multiband filter with bands $0 < W < W_1$, $W_1 < W, < W_2$, ..., $W_N < W < 1$.

B = FIR1(N,W_n,'DC-1') makes the first band a passband.
B = FIR1(N,W_n,'DC-0') makes the first band a stopband.
By default FIR1 uses a Hamming window. Other available windows, including Boxcar, Hann, Bartlett, Blackman, Kaiser and Chebwin can be specified with an optional trailing argument. For example, B = FIR1(N,W_n,kaiser(N+1,4)) uses a Kaiser window with beta=4.
B = FIR1(N,W_n,'high',chebwin(N+1,R)) uses a Chebyshev window.

freqz returns the frequency response based on the current filter coefficients.

<u>**Laboratory Task:-**</u>

Task 1:
- Designing a Low Pass Filter that passes all frequencies below 1200 Hz. Take F_s = 8000 Hz, F = 1200 Hz and N = 64. Take time vector as [0 : $1/F_s$: 1].
- Display the Magnitude Plot of the filter.
- Display the pole-zero constellation.

Task 2:
Repeat similar procedure to implement:
- High Pass Filter that pass all frequencies above 1200 Hz.
- Band Pass Filter that passes frequencies between 1200 and 1800 Hz.
- Band stop Filter that rejects frequencies between 1200 and 1800 Hz.
- Notch filter that reject frequency between 1500 and 1550 Hz.

Task 3:
1. Create a signal vector containing two frequencies as: i) 100 Hz. and ii) 150 Hz, with F_s = 1000 Hz.
2. Design two band pass FIR filters with 64 coefficients and with pass bands as:
   i) 75 to 125 Hz. and ii) 125 to 175 Hz.
3. Use both filters on the created signal and observe their outputs.
4. Plot frequency responses and pole-zero constellations of both filters and note observations.

<u>**Result:-**</u>
The code, results / output of all tasks must be attached with this lab.

NED University of Engineering & Technology
Department of Telecommunications Engineering

Course Code and Title: (TC-212) Digital Signal Processing

Laboratory Session No. 09   Date: _____

Criterion	Level of Attainment				
	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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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.

The table title "Software Use Rubric" spans the entire table header above.

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

LAB SESSION 10

Objective:-

 To design of IIR filters using MATLAB.

Equipment Required:

PC along with software, Oscilloscope.

Theory:-

Matlab contains various routines for design and analyzing digital filter IIR. Most of these are part of the signal processing tool box. A selection of these filters is listed below.

- Buttord (for calculating the order of filter)
- Butter (creates an IIR filter)
- Ellipord (for calculating the order of filter)
- Ellip (creates an IIR filter)
- Cheb1ord (for calculating the order of filter)
- Cheyb1 (creates an IIR filter)

Buttord- Butterworth filter order selection $[N, W_n] = BUTTORD(W_p, W_s, R_p, R_s)$
Ellipord- Elliptic filter order selection $[N, W_n] = ELLIPORD(W_p, W_s, R_p, R_s)$
Cheb1ord- Chebyshev Type I filter order selection $[N, W_n] = CHEB1ORD(W_p, W_s, R_p, R_s)$

returns the order N of the lowest order digital Butterworth, elliptic and Chebyshev Type I filter respectively that loses no more than R_p dB in the pass band and has at least R_s dB of attenuation in the stop band. W_p and W_s are the pass band and stop band edge frequencies, normalized from 0 to 1 (where 1 corresponds to π radians/sample).

BUTTORD also returns W_n, the Butterworth natural frequency (or, the "3 dB frequency") to use with BUTTER to achieve the specifications.

ELLIPORD also returns W_n, the elliptic natural frequency to use with ELLIP to achieve the specifications.

CHEB1ORD also returns W_n, the Chebyshev natural frequency to use with CHEBY1 to achieve the specifications.

Butter- Butterworth digital and analog filter design

$[B,A] = BUTTER(N,W_n)$ designs an Nth order lowpass digital Butterworth filter and returns the filter coefficients in length N+1 vectors B (numerator) and A (denominator). The coefficients are listed in descending powers of z. The cutoff frequency W_n must be $0.0 < W_n < 1.0$, with 1.0 corresponding to half the sample rate.

If W_n is a two-element vector, $W_n = [W_1 \; W_2]$, BUTTER returns an order 2N bandpass filter with passband $W_1 < W < W_2$. [B,A] = BUTTER(N,W_n,'stop') is a bandstop filter.

[B,A] = BUTTER(N,W_n,'high') designs a highpass filter.

When used with three left-hand arguments, as in [Z,P,K] = BUTTER(...), the zeros and poles are returned in length N column vectors Z and P, and the gain in scalar K.

Ellip- Elliptic or Cauer digital and analog filter design

[B,A] = ELLIP(N,R_p,R_s,W_n) designs an Nth order low pass digital elliptic filter with R_p decibels of peak-to-peak ripple and a minimum stop band attenuation of R_s decibels. ELLIP returns the filter coefficients in length N+1 vectors B (numerator) and A (denominator).The cutoff frequency W_n must be $0.0 < W_n < 1.0$, with 1.0 corresponding to half the sample rate. Use $R_p = 0.5$ and $R_s = 20$ as starting points, if you are unsure about choosing them.

If W_n is a two-element vector, $W_n = [W_1 \; W_2]$, ELLIP returns an order 2N band pass filter with pass band $W_1 < W < W_2$. [B,A] = ELLIP(N, R_p,R_s,W_n,'stop') is a band stop filter if $W_n = [W_1 \; W_2]$

[B,A] = ELLIP(N, R_p,R_s,W_n,'high') designs a high pass filter.

When used with three left-hand arguments, as in [Z,P,K] = ELLIP(...), the zeros and poles are returned in length N column vectors Z and P, and the gain in scalar K.

Cheby1- Chebyshev Type I digital and analog filter design

[B,A] = CHEBY1(N,R,W_n) designs an Nth order lowpass digital Chebyshev filter with R decibels of peak-to-peak ripple in the passband. CHEBY1 returns the filter coefficients in length N+1 vectors B (numerator) and A (denominator). The cutoff frequency W_n must be $0.0 < W_n < 1.0$, with 1.0 corresponding to half the sample rate. Use R=0.5 as a starting point, if you are unsure about choosing R.

If W_n is a two-element vector, Wn = [$W_1 \; W_2$], CHEBY1 returns an order 2N bandpass
filter with passband $W_1 < W < W_2$.

[B,A] = CHEBY1(N,R,W_n,'high') designs a highpass filter.

[B,A] = CHEBY1(N,R,W_n,'stop') is a bandstop filter if $W_n = [W_1 \; W_2]$.

When used with three left-hand arguments, as in [Z,P,K] = CHEBY1(...), the zeros and poles are returned in length N column vectors Z and P, and the gain in scalar K.

Laboratory Task:-

Task 1:
Buttord and Butter Filter,
Design IIR Low Pass Filter: Suppose our target is to design a filter to pass all frequencies below 1200 Hz with pass band ripples = 1 dB and minimum stop band attenuation of 50 dB at 1500 Hz. The sampling frequency for the filter is 8000 Hz;

Task 2:
Design IIR High Pass Filter: We will consider same filter but our target now is to pass all frequencies above 1200 Hz. The sampling frequency for the filter is 8000 Hz.

Task 3:
Designing IIR Band Pass Filter: Now we wish to design a filter to pass all frequencies between 1200 Hz and 2800 Hz with pass band ripples = 1 dB and minimum stop band attenuation of 50 dB. The sampling frequency for the filter is 8000 Hz.

Task 4:
Design IIR Band Stop Filter: Now we wish to design a filter to stop all frequencies between 1200 Hz and 2800 Hz. The sampling frequency for the filter is 8000 Hz.

<u>Result:-</u>

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

Course Code and Title: (TC-212) Digital Signal Processing

Laboratory Session No. 10   Date: _____

Criterion	Level of Attainment				
	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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(Software Use Rubric)

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

LAB SESSION 11

Objective:-

To explore the multirate sampling using MATLAB.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

Multirate simply means "multiple sampling rates". A multirate DSP system uses multiple sampling rates within the system. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so. Therefore "Multirate DSP" really refers to the art or science of *changing* sampling rates.

Laboratory Task:-

Task 1:
Generate a time domain signal and its spectrum.

Task 2:
Down sample the generated signal by a factor and plot its spectrum. Pass signal by low pass filter and observe its spectrum by down sampling it.

Task 3:
Up sample the generated signal by a factor and repeat task 2 for it.

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

Course Code and Title: (TC-212) Digital Signal Processing

Laboratory Session No. 11   Date: _____

Software Use Rubric					
Criterion	**Level of Attainment**				
	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	

LAB SESSION 12

Objective:-

Designing filters using MATLAB Signal Processing Toolkit.

Equipment Required:-

PC along with software, Oscilloscope.

Theory:-

The goal of filtering is to perform frequency-dependent alteration of a signal. A simple design specification for a filter might be to remove noise above a certain cutoff frequency. A more complete specification might call for a specific amount of passband ripple (R_p, in decibels). Stopband attenuation (R_s, in decibels), or transition width (W_p-W_s, in hertz).

Filter Configurations:
All the filter design functions in the Signal Processing Toolbox operate with normalized frequencies, so that they do not require the system sampling rate as an extra input argument. The normalized frequency is always in the interval 0≤f≤1. For example, with a 1000 Hz sampling frequency, 300 Hz is 300=500 = 0.6. To convert normalized frequency to angular frequency around the unit circle, multiply by π. To convert normalized frequency back to Hertz, multiply by half the sample frequency.

Digital filter specifications are often given is terms of the loss function (in dB),

$$A(\omega) = -20\log 10 \left| G(e^{j\omega}) \right|$$

Filter Design with SPTool:
The Filter Designer in the SPTool allows you to design and edit IIR and FIR filters of various lengths and types, with lowpass, highpass, bandpass, and bandstop, and multiband configurations. To activate the Filter Designer, click either the New button or the Edit button under the Filters list box in SPTool.
Try: sptool

Filter Design with FDATool:
The FDATool, supports many advanced techniques not available in SPTool. It also allow realization of Simulink models of quantized direct form FIR filters. Perform digital frequency transformations of filters.
Try: fdatool

Laboratory Task:-

Task 1:
Design a Lowpass, FIR Eqiripple filter, Minimum order, F_s=1000 Hz, F_{pass}=60 Hz, F_{stop}=200 Hz , A_{pass} =1dB, A_{stop} =8dB. Note the following observations:
- Magnitude and phase response
- Phase and group delay
- Filter impulse response
- Filter information, filter coefficient , filter parameters
- Pole-zero plot

Export the filter parameters (Num) to your workspace. Do size (Num) to obtain the size of your FIR filter. Do freqz(Num,1). In the command window, do sptool. In sptool, Import Num to sptool. Change sampling frequency from the default 1 to 1000. Now under SPTool: startup.spt, view the signal (Recall that Num is just the values of the 18 point FIR filter; i.e. the FIR impuls response). Under Spectra, use\ create and do a 1024 point fft. Under options > Magnitude > Linear. Observe the difference between observing on a linear and log scales. Do you see the equiripples in the stopband.

Task 2:
Now in fdatool, for the same specs as the FIR filter, design an IIR Butterworth filter. (Match exactly the passband and stopbands). Look at the magnitude response. Linear phase? Look at the impulse response. (Note that it is NOT symmetric). Look at the filter information. Did it meet specs? How many poles and zeros? Does it have any properties dissimilar to that of FIR filter ? Explain.

<u>Result:-</u>
The results / output of all tasks must be attached with this lab.

Course Code and Title: (TC-212) Digital Signal Processing

Laboratory Session No. 12   Date: _____

Software Use Rubric					
Criterion	**Level of Attainment**				
	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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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:

To explore, analyse and develop experience in designing a digital system using the software platform of MATLAB.

Problem Statement:

Design of digital filter and its application to remove noise from a real-world signal.

Background/Pre-Requisite:

To perform this activity student must have the knowledge and understanding of
- Signal processing and digital filters.
- MATLAB

Outcome:

This activity is mapped to CLO 3 of the course and results in the attainment of PLO 5 (Modern Tool Usage)

CLO	Taxonomy Level	PLO	Outcome
3	P2	5	The students must be able to build the model of digital system and analyze various aspects of the system through experimental work/simulation following instructor guidance and operating manual instructions.

Instructions for Problem Solution:

You have to design a digital filter (using MATLAB) which can separate noise from a real-world signal (i.e. ECG signal). Optimize the parameters: stop band attenuation, pass-band ripple and width of transition band. You are not allowed to use any in-built functions of FIR filter or filter design toolbox. The design should be tested to:
1. Verify the removal of noise by Frequency analysis.
2. Verify the optimization of mentioned parameters by time domain and frequency domain analysis of filter characteristics.

Deliverables:

Write a detailed report on the design of your digital filter. The report should be in the standard format including abstract, objectives, background, system working/description, results/code/discussions and conclusions.

NED University of Engineering & Technology
Department of Telecommunications Engineering
Course Code & Title: (TC-212) Digital Signal Processing
Assessment Rubric for Open Ended Lab

Criterion	Level of Attainment				
	Below Average (1)	Average (2)	Good (3)	Very Good (4)	Excellent (5)
Problem Understanding	Did not try to understand the Problem	Tried to understand the problem	Partially understood the problem	Completely understood the problem but could not explain it properly	Completely understood the problem and was able to explain it properly
Solution Viability	No viability approach was adopted	Viability approach was inappropriate	Viability approach was appropriate but not convincing	Viability approach was appropriate and convincing	Viability approach was appropriate, convincing and appreciable
Data Analysis & Discussion	No submission/No effort exhibited.	Major results are notincluded. Plots and figures are poorly constructed or not present.	Misses several key results. Plots/Figures lack proper identification in the X and Y axis.	One or more key results are missing. Figures and tables are present but contain minor errors.	Key results are presented in an orderly and logical sequence using both text and illustrative materials (Plots and Figures).
Response to Questions	Neither understands the question, nor could reply.	Adequate understanding of the question and reply.	Seemed to understand the main points of the question and replied to those with ease.	Clearly understood the question and replied with ease.	Understood the question in-depth and replied confidently.
Report Writing	No submission/No effort exhibited	Directions were not followed.	Directions were not explicitly followed.	Minor errors in format or procedures were encountered	Directions were followed and questions were answered correctly

Student's Name: _____   Roll No.: _____

Total Score = _____   Instructor's Signature: _____

35