# DIGITAL SIGNAL PROCESSING LAB (EL-302)

# LABORATORY MANUAL

# ENGR. MUHAMMAD IBRAR KHAN BASIC CONCEPTS OF DIGITAL SIGNAL PROCESSING (LAB # 02)

Student Name:		
	Roll No:	Section:
	Date performed:	. 2019



#### NATIONAL UNIVERSITY OF COMPUTER AND EMERGING SCIENCES, ISLAMABAD

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# Lab # 02: BASIC CONCEPTS OF DIGITAL SIGNAL PROCESSING

## **Learning Objectives**

- A. Relative Error
- B. Circular Shift and Circular Time Reversal
- C. Sequence Generation from Continuous Time Signals (Sampling)
- D. Sampling Rate Alteration

### **Equipment Required**

- 1. PC
- 2. MATLAB

#### **Relative Error**

Whenever a signal/sequence is corrupted by some noise then it is important to find the difference between the original signal and the corrupted signal and this normalized difference is called relative error. Relative error is defined as the ratio of the  $L_2$  – norm of the difference signal and the  $L_2$  – norm of original signal  $\{x[n]\}$ .

$$E_{rel} = \left(\frac{\sum_{n=0}^{N-1} |y[n] - x[n]|^2}{\sum_{n=0}^{N-1} |x[n]|^2}\right)^{1/2}$$

Where  $\{y[n]\}$  is the corrupted sequence and  $\{x[n]\}$  is the original sequence.

#### **Task 01:**

- 1. Generate a sinusoidal sequence  $x[n] = \sin(2 * pi * 0.5 * n) -10 < n \le 10$
- 2. Add noise to the above signal using command "awgn" to generate a signal y[n]. You can use 'help awgn'.
- 3. Find Relative error  $E_{rel}$  using the above given formula.
- 4. Plot in time domain using 'stem'.

#### **Task 02:**

- 1. Find frequency response of x[n] & y[n] of Task 01 and plot these response with respect to frequency in Hz.
- 2. Use the 'freqz' command to find the frequency response.

#### **Circular Shift and Time Reversal**

Circular shift is different form linear shift. In case of infinite length sequence, linear shift can be used because our samples are infinite and appended zeros istead of a samples may not effect the operations. But in case of finite length sequences, it is mandatory to use such a shifting operation such that our samples are not lost. To do this we use modulo operation.

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#### **Modulo Operation:**

Let 0,1,....... N-1 be a set of N positive integers and let m be any integer. The integer r obtained by evaluating m modulo N is called the residue. The residue r is an integer with a value between 0 and N-1.

#### **Circular Time Reversal:**

The circular time-reversal version  $\{y[n]\}\$  of a length-N sequence  $\{x[n]\}\$  defined for  $N-1 \le n \le 0$  is given by  $\{y[n]\} = \{x[\langle -n \rangle_N]\}$ .

#### **Circular Shift:**

Let x[n] be a length-N sequence defined for  $N-1 \le n \le 0$ . Its circularly shifted version  $x_c[n]$  shifted by  $n_0$  samples is given by

$$\{x_c[n]\} = \{x[\langle n - n_0 \rangle_N]\}$$

#### **Task 03:**

1. Generate and plot the following exponential sequence

$$x[n] = 0.5^n, -3 < n \le 10$$

- 2. Find and plot the following
  - $x_1[n] = x[n+2]$
  - $x_2[n] = x[\langle n+2 \rangle_N]$
  - $\bullet \quad x_3[n] = x[-n]$
  - $x_4[n] = x[\langle -n \rangle_N]$

#### **Task 04:**

- 1. Write a generic matlab function for circular shift
- 2. Write a generic matlab function for circular time reversal.

# **Sequence Generation from Continuous Time Signals (Sampling)**

In signal processing, sampling is the reduction of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous signal) to a sequence of samples (a discrete-time signal). A sample is a value or set of values at a point in time and/or space. A sampler is a subsystem or operation that extracts samples from a continuous signal.

```
clc, clear;
F=4000; Dt=0.01; f=1000/F; %Hz
t=0:Dt:12; % sec
y t = sin(2*pi*f*t);
%% Oversampling
y = y t(1:1/((2*f/Fs)*Dt):end);
fs=4*f;Ts1=1/fs;
n1=t(1)/Ts1:1:t(end)/Ts1;
y n1=sin(2*pi*f*n1*Ts1);
%% Under sampling
fs=1.5*f; Ts2=1/fs;
n2=t(1)/Ts2:1:t(end)/Ts2;
```

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```
y_n2=sin(2*pi*f*n2*Ts2);
^-8n=0:1/(2*f/Fs):12;
subplot(2,3,1)
plot (t,y t)
title('CT Plot of Sinusoidal signal');
xlabel('Time t');ylabel('CT Signal y(t)');grid on;
subplot(2,3,2)
stem (n1*Ts1, y n1); hold on
plot (t,y t,'g')
title('DT PlotT Oversampled');
xlabel('Time t');ylabel('DT Signal');grid on;
subplot(2,3,3)
stem (n2*Ts2, y n2); hold on
plot (t,y t,'g')
title('DT PlotT Undersampled');
xlabel('Time t');ylabel('DT Signal');grid on;
%% Computing frequency response of y n and plotting %%%%%%%%%%%
subplot(2,3,4)
[Y n, omega n] = freqz(y n1);
plot (omega n*F/(2*pi), (abs(Y n)))
title('Oversampling A sinusoidal signal in frequency domain');
xlabel('Frequency in Hz'); ylabel('Magnitude response Y')
grid on;
subplot(2,3,5)
[Y n, omega n] = freqz(y n2);
plot (omega n*F/(2*pi), (abs(Y n)))
title('Oversampling A sinusoidal signal in frequency domain');
xlabel('Frequency in Hz'); ylabel('Magnitude response Y')
grid on;
```

#### **Task#05:**

Generate a multi-tone sinusoidal signal with  $f_1$ =2000Hz and  $f_2$ =4000Hz, then by sampling generate the Oversampled and Undersampled version of it and display the frequency response also.

# **Sampling Rate Altertaion**

If we want to generate a new sequence from x[n] with a different sampling rate  $F_T$  higher or lower than the samling rate  $F_T$  of original sequence x[n] then sampling alteration ratio is given by  $R = \frac{F_T'}{F_T}$ . Now If R>1 then it is called interpolation (Upsampling) and if R<1 then it is called decimation or downsampling. Generally we mostly append zeros in case of up sampling.

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**Task 06:** 

1. Generate and plot the follwing sinusoidal sequence

$$x[n] = \sin(2 * pi * 0.25 * n) -15 < n \le 20$$

- 2. Generate and plot  $x_1[n]$  by upsampling the above sequence x[n] by 3.
- 3. Generate and plot  $x_2[n]$  by downsampling the above sequence x[n] by 3.
- 4. Help: Use 'upsample' and 'downsample 'commands
- 5. Now, generate and plot  $x_3[n]$  by upsampling  $x_2[n]$  by 3.
- 6. Now, generate and plot  $x_4[n]$  by downsampling  $x_1[n]$  by 3.

Question

Does  $x_3[n]$  and x[n] are similar? If Yes/No then Why?

Question

Does  $x_4[n]$  and x[n] are similar? If Yes/No the Why?

#### Task 07(Bonus Task):

1. Write your own matlab function for upsampling and downsampling.

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**Student's feedback:** Purpose of feedback is to know the strengths and weaknesses of the system for future improvements. This feedback is for the 'current lab session'. Circle your choice:

[-3 = Extremely Poor, -2 = Very Poor, -1 = Poor, 0 = Average, 1 = Good, 2 = Very Good, 3 = Excellent]: The following table should describe your experience with:

S#	Field	Rating	Describe in words if required
1	Overall Session	-3   -2   -1   0   1   2   3	
2	Lab Instructor	-3   -2   -1   0   1   2   3	
3	Lab Staff	-3   -2   -1   0   1   2   3	
4	Equipment	-3   -2   -1   0   1   2   3	
5	Atmosphere	-3   -2   -1   0   1   2   3	
Any	y other valuable fe	edback:	

Student's Signature:

MARKS AWARDED	Attitude	Neatness	Correctness of results	Initiative	Originality	Conclusion	TOTAL
TOTAL	10	10	10	20	20	30	100
EARNED							

Lab Instructor's Comments:_	 	 
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