Problem Statement for DSPLab

Lab- 1

1a) Generate and plot a Unit Step sequence of length 20;

b) Generate and plot a unit sample after delay 10 of total length 20;

c) Generate and plot the complex exponential sequence $x(n) = -3.6\exp(-0.5 + j\frac{\pi}{4})n$,
for $0 \le n \le 20$.

d) Generate and plot addition of two sinusoidal sequences
$x(n) = \cos(0.2\pi n) + \cos(0.5\pi n)$, $0 \le n \le 20$.

e) Generate and plot exponential sequence $x(n) = \alpha^n$ for $\alpha = 0.1, 0.7, 1.1$, $0 \le n \le 20$

f) Generate and plot square sequence $x(n) = 10\Pi(0.2\pi n)$, $0 \le n \le 20$ in polar and unipolar form.

Lab-2a

2 Assume a clean data sequence is given by $s(n) = 2.n.(0.9)^n$, $0 \le n \le 20$

a) Plot data sequence $s(n)$
Generate a noise corrupted signal $x(n) = s(n) + w(n)$.
where, $w(n)$ in the a random noise in the range [-0.5 0.5]. Multiply the noise by a factor 5.

b) Find the ensemble average after 50 (=M) measurements and plot that.
Show that ensemble average is nearly the same as the original uncorrupted data in part(a).
Also check the ensemble average for M=10.

Lab-2b (HW)

Repeat the problem Lab-2a for $x(n) = 2.\Pi(0.2\pi n)$, for unipolar square sequence.
(Note this is a square wave of frequency 1 KHz at sampling frequency 10 KHz.

Lab-2c (before Lab 4d)

a) Noise corrupted signal $x(n) = s(n) + w(n)$. in problem part(a) in Lab-2a is passed through a Moving Average(MA) filter. Observe the output of the filter of length M=3 and M=10.

b) Given $s(n)$, $w(n)$, $x(n)$ implement the MA filter output from the following recursive equation (IIR filter):

$$y(n) = y(n-1) + \frac{1}{M}(x(n) - x(n-M))$$

c) Do the above problem part(b), Lab(2c) by the following algorithm

$$y(n) = \alpha\, y(n-1) + \frac{1}{M}(x(n) - x(n-M)); \text{ where } 0 \le \alpha \le 1,$$

This average weighted filter places more emphasis on current data samples and less emphasis on past data samples by exponentially weighting the data samples.

Lab-3a

a) A discrete time signal $s(n)$, $0 \le n \le 100$, is obtained by uniformly sampling a continuous time signal $s(t) = 5\cos(200\pi t)$ at a sampling rate of 1 KHz. Find normalized frequency 'f' and plot $s(t)$ and $s(n)$.

b) Demonstrate that all continuous time signal frequencies 1100 Hz, 2100 Hz and 3100 Hz are aliased.

c) What should be the pre-filter frequency before sampling to avoid aliasing.

Lab-3b (H.W.)

$0 \le n \le 100$

a) Generate the modulated sequence $y(n) = x_1(n).x_2(n)$, where $x_1(n) = \cos(0.2\pi n)$ and $x_2(n) = \cos(0.02\pi n)$. Find out the new normalized frequencies from the product of two sinusoids.

b) From the plot of $y(n)$, obtain the beat frequency.

Lab-4a

Given the sequences,

	x(n) =[-4 5 1 -2↑ -3 0 2],	$-3 \le n \le 3$
	y(n) =[6 -3↑ -1 0 8 7 -2],	$-1 \le n \le 5$
	w(n)=[3 2 2 -1 0 -2 5],	$2 \le n \le 8$,

and sample values of each of he above sequences outside the range specified are all ze

a) Evaluate the auto correlation sequences [$r_{xx}(l), r_{yy}(l), r_{ww}(l)$] of each of the abov sequences and plot those w.r.to lag 'l'.
b) Evaluate the cross correlation sequences [$r_{xy}(l), r_{xw}(l)$] of the above sequences a plot those w.r.to lag 'l'.
c) Using 'conv' command determine auto-correlation and cross-correlation sequenc obtained in part (a) and (b).

Lab-4b

a) Repeat the problem Lab-4a , append with three extra zeros for each sequence x(n), y(n) and w(n).
Show that the results are same.

Lab-4c

A sequence y(n) is obtained by convolution of two finite length sequences x(n) and h(n) given by

$$x(n) = [-2\uparrow, 0, 1, 2, 3] \text{ and } h(n) = [1\uparrow, 2, 0, -1].$$

Plot x(n), h(n) and y(n).

~~Lab 5c~~

Lab-4d

1a) A sinusoidal sequence $s(t) = \cos(0.25\pi n)$, $0 \le n \le 100$ is corrupted by an additive uniformly distributed random noise w(n) of amplitude in the range [-0.5 0.5]. Find the auto-correlation sequence $r_{yy}(k)$ of the noise corrupted sinusoidal sequence y(n). y =

b) Find also the cross correlation sequences $r_{sw}(k)$, $r_{yw}(k)$ and auto correlation sequence $r_{ww}(k)$.

Lab5a

An input -output relation of a discrete LTI system is described by

$$y(n) = \frac{1}{M}[x(n) + x(n-1) + x(n-2) + \cdots\cdots\cdots x(n-M)];$$

$$= \frac{1}{M}\sum_{k=0}^{M} x(n-k), \qquad M = 40;$$

Determine the frequency response of the system $H(w)$ and plot its magnitude $|H(w)|$ and phase $\angle H(w)$ response for M =5 and 10

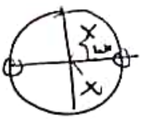
Lab-5b

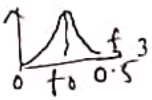
a) Determine $H(w)$ and plot its magnitude and phase response of an LTI system whose input output relationship is as follows:

$$y(n) = 0.15x(n) - 0.15x(n-2) + 0.5y(n-1) - 0.75y(n-2);$$

b) Determine the phase introduced by this system when a single sinusoidal frequency w is passed through the $H(w)$.

Draw pole-zero plot. (z plane (num, den))





c) Determine also the group delay introduced by the system when several sinusoidal components with different frequencies, that are not harmonically related, are passed through.

Lab-6a

a) Compute DFT of the following sequence: $x(t) = \cos(0.25\pi n)$, $0 \le n \le N$, for $N =$ 64 and 512 and plot it.
b) Verify he above program with the help of m-file FFT command.
c) Find IDFT of part (b) and compare with $x(t)$ for the first 60 samples.($0 \le n \le 60$)

Lab-6b

6a) Consider length 4-sequences x(n) and h(n). Find the circular convolution $Y_{c1}(n) = x(n) \otimes h(n)$ where, x(n) = [1 2 0 1] and h(n)= [2 2 1 1], $0 \le n \le 3$.
b) Also obtain circular convolution Y_{c2} of length 7 sequences obtained by zero padding the above sequences: xe(n) = [1 2 0 1 0 0 0] and he(n)= [2 2 1 1 0 0 0].
c) Obtain linear convolution Y_{l1} of part(a) and compare that with Y_{c2} in part(b).
d) Implement the linear convolution Y_{l1} of the two sequences x(n) and h(n) by DFT and IDFT operations.

Lab-7

A long incoming noisy signal x(n) = s(n) + w(n) , $0 \le n \le 128$, is passed through a MA-filter whose input output relationship is given by $y(n) = \sum_{k=0}^{M-1} x(n-k)$, $M = 3$.

Use overlap-save method (m file command 'fftfilt') for block length L=4 for filtering. Where, $s(n) = 2.n.(0.9)^n$, and w(n) is the a random noise in the range [-0.5 0.5].

a) Plot x(n);
b) Find the impulse response of the MA-filter.
c) Find the output y (n) of the MA-filter for noisy input x (n).
d) Deconvolve x (n) from the given output y (n) and h(n) [in part© and (b)] and compare it with x(n) obtained in part(a).

Lab-8

Transfer function of an IIR filter is given by $H(z) = \dfrac{3 + 4z^{-1}}{1 - 3.5z^{-1} + 1.5z^{-2}}$

a) Express $H(z)$ in partial fraction form.
b) Obtain pole-zero plot in the Z-plane for the above transfer function.Obtain impulse response h(n), $1 \le n \le 100$. Is h(n) is converging? If not, find h(n) for a stable causal system $H_1(z)$. Show the pole-zero of this stable system $H_1(z)$.

c) Compare frequency response plot of $H(z)$ and $H_1(z)$.

Lab-9

9) Effect of coefficient quantization on the sensitivity of an LTI, IIR digital system H(z) is studied where H(z)is given by

$$H(z) = \frac{1}{1 - 1.8456 z^{-1} + 0.849585 z^{-2}}$$

a) Find h(n).
b) Quantize the coefficients of z^{-1} and z^{-2} to -1.85 and 0.84 respectively. Show that the system $H(z)$ in this case is unstable.
c) Show he pole-zero plot for the stable and quantized system.

Lab10 (only a and b)

Pass band edge

a)Determine the order of the Butterworth filter that has a ~~3 dB bandwidth~~ of 1000 Hz. and stopband attenuation of 40 dB at 1500 Hz. , passband ripple = 1dB
b) Obtain gain-frequency response plot using m-file command.
c) Plot gain-frequency response of Chebyshev (type1) low-pass filter **of order 7** for passband edge frequency of 1000 Hz and stopband attenuation of 40 dB at 1500 Hz. Assume passband ripple is 1dB.
d) Repeat part c with Chebyshev (type 2) low-pass filter **of order 7 .**
e) Repeat part c with Elliptic low-pass filter **of order 7 .**
f) Compare al the filter types with order 7 other specifications being the same.
and pole-zero plot in each case. (s-plane (num, den))

Lab-11

Design IIR lowpass filter using the following specifications: (analog domain)
Pass band edge frequency = 800 Hz,
Stop band edge frequency = 1000 Hz,
Pass band ripple = 0.5 dB,
Stop band attenuation (or ripple)= 30 dB,
Sampling frequency= 8000 KHz.

a) Obtain frequency response plot of the filter.
b) Obtain group delay in the pass-band of the filter.
c) If group delay in the passband is not constant , provide group delay equalization of the IIR filter.

Lab-12

2 a lowpass   1112

Design a highpass IIR digital filter using bilinear transformation. Specifications of the HP digital filter is as follows:
Pass band edge frequency = 700 Hz,
Stop band edge frequency = 500 Hz,
Pass band ripple = 1 dB,
Stop band attenuation (or ripple)= 30 dB,
Sampling frequency= 2000 Hz.
Use Chebyshev (type 2) filter as analog low-pass prototype.
Plot the frequency response of the filter

Lab13

(a)Do the problem Lab-11 using Elliptic filter.
(b) Obtain group delay in the filter passband.

Lab14

Design the problem lab11 for a digital filter using frequency prewarping and bilinear transformation.

Lab-15

Design a FIR lowpass filter of the following specifications:

Pass band edge frequency = 1500 Hz,
Stop band edge frequency = 2000 Hz,
Stop band attenuation (or ripple) $\geq$ 50 dB,
Sampling frequency= 8000 Hz.
Use Hamming window;

a) Find the order of the filter.
b) Plot the window function and causal impulse response of the filter.
c) Obtain gain-frequency of the designed filter
d) Obtain group delay in the passband.

Lab-16

Use the same specification as in problem Lab15, design the filter using rectangular, Hamming, and Blackman window function and compare those filters.

Lab17

Use the same specification as in problem Lab15, design equiripple linear phase FIR lowpass filter using Ramez algorithm;

Lab18

Use the following specification, design equiripple linear phase FIR bandpass filter us
Ramez algorithm;

Pass band edge frequency = [600Hz to 1000 Hz]
Stop band edge frequency = [0 to 500Hz] and [1100Hz to 4000Hz],
Passband ripple= 0.5 dB,
Stop band attenuation (or ripple) = 40 dB,
Sampling frequency= 8000 KHz.