PES University, Bengaluru

(Established under Karnataka Act 16 of 2013)

**END SEMESTER ASSESSMENT (ESA) - JULY - 2023**

**UE17EE353 - Digital Signal Processing**

**Total Marks : 100.0**

1.a.

A mobile handset is been used for recording a concert been heard on FM radio station and presume that the signal is limited to 18kHz. The flash memory size is 2GB. Which sampling frequency do you select for the recording to playback later to get as closest as possible to original contents recorded: 1) 10 kHz 2) 24kHz 3) 44.1kHz 4) 33.2kHz. Presume 12 bits for each sample. Considering no compression, how long can the recording be done **in hours** before memory goes full.

(3.0 Marks)

1.b.

There are two sinusoidal signals from oscillator - $x_1(t) = \sin(2\pi t)$ and $x_2(t) = \sin(12\pi t)$ sampled at sampling time $T_s = 0.2s$. If it is sampled using A/D and CPU processes and prepares $y(n) = x(n)$ (if $x(n)$ is sampled input) to output the same to D/A at the sampling time $Ts$.

Represent outputs $y_1(n)$ and $y_2(n)$ corresponding to inputs $x_1(t)$ and $x_2(t)$

Are the outputs aliased? Substantiate the same. If they are aliased, find another input signal which will give an aliased output corresponding to input $x_1(t)$?

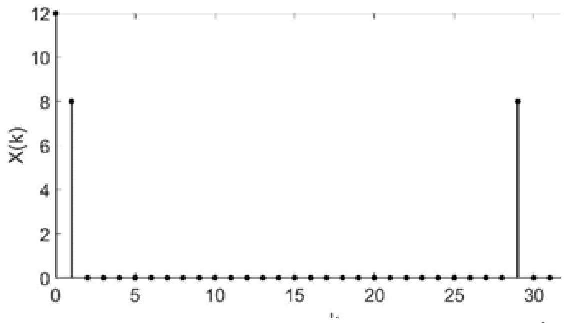
(6.0 Marks)

1.c.

What frequency components does a periodic impulse signal $\delta(n)$ with periodicity N=8, comprise considering DFT?

Below is the plot of 32-point DFT, $X(k)$ for an input sequence with sampling frequency at $f_s = 22.1kHz$

1) What is the DC value of the signal?
2) Which are the dominant frequencies of the signal based on DFT?




(5.0 Marks)

1.d.

Consider the periodic input sequence $x(n) = \{1 \;\; 2 \quad 4\}$ and periodic unit impulse response $h(n) = \{0 \;\; 1 \;\; 0\}$.

1) Determine a sequence $y(n)$ such that $Y(k) = X(k)H(k)$ .
2) Find out periodic impulse response $h(n)$ to get $y(n) = \{15 \;\; 16 \;\; 11\}$ for the same input sequence.

(6.0 Marks)

2.a.

Compute 4-point DFT of the sequence $x(n) = \cos(\frac{n\pi}{2})$ using Radix-2 DIF FFT algorithm.

(6.0 Marks)

2.b.

A microcontroller is used for speech processing. Speech input from microphone is sampled at a rate of 8 kHz and is to be processed in real time. The microcontroller performs 1024-point DFT and IDFT computations in addition to computation required to collecting blocks of 1024 speech values. If it takes $1\mu s$ for each real multiply, how much time remains for processing the data after the DFT and the inverse DFT are computed?

(6.0 Marks)

2.c.

Find the output $y(n)$ of the filter using overlap-save method. Input signal sequence $x(n) = \{3, -1, 0, 1, 3, 2, 0, 1, 2, 1\}$ and impulse response is $h(n) = \{1, 1, 1\}$.

(8.0 Marks)

3.a.

Show the plot of ripple in pass band for Chebyshev filter by plotting $|H_3(j\Omega)^2|$ with $\epsilon = 0.50$, $|H_3(j\Omega)^2 = 1/(1 + \epsilon^2\, T_3^2(\Omega))$ for $\Omega = 0$ to $1$. Note: You can find $T_3^2(\Omega)$ directly and no need to show details. Plot the boundary values and ripples.

(4.0 Marks)

3.b.

Design an Analog high pass Butterworth filter that has 3dB attenuation at 500rad/s and 15dB attenuation at 1000rad/s with the following steps:

1) Mark the frequency response desired
2) Formulate mathematically the needed gains
3) Determine H(s). (No need to simplify completely)

(10.0 Marks)

3.c.

A low pass filter of 1 rad/s bandwidth filter with the following characteristics

a) Acceptable passband ripple of 2 dB
b) Cut off radian frequency of 1 rad/s
c) Stop band attenuation of 20 dB or greater beyond 1.3 rad/s.

Which type of filter is suitable for this? Determine the order of filter needed.

(6.0 Marks)

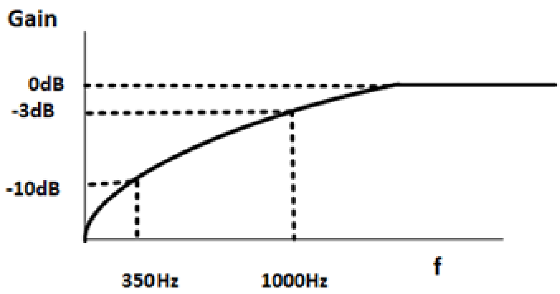
4.a.

Design a digital filter for Analog filter transfer function $H(s) = 10/(s^2 + 7s + 10)$. Use the impulse invariant technique with T=0.2s.

(7.0 Marks)

4.b.

In an application, there is a need to eliminate a few lower frequency perturbations and get a cleaner signal for processing. The frequency response for analogue domain is as below:




Design a digital filter using bilinear transformation that is monotonic in pass band. Consider sampling frequency is 5kHz.

(8.0 Marks)

4.c. Realise the digital filter using either Direct form-1 OR Direct form 2

$$H(z) = \frac{8z^3 - 4z^2 + 11z - 2}{\left(z - \frac{1}{4}\right)\left(z^2 - z + \frac{1}{2}\right)}$$

after representing it as a difference equation for $y(n)$

(5.0 Marks)

5.a.

Answer the following in one or two lines to articulate your concepts

1) Why is linear phase needed for filtering? Does FIR filter provide it?
2) Comment on magnitude and phase of Differentiator and Hilbert transformer. Just indicate its property.

(6.0 Marks)

5.b. Design a low pass linear phase FIR filter using frequency sampling method having cutoff frequency of $\frac{\pi}{2}$ and N =7.

(6.0 Marks)

5.c. Design a FIR filter to determine unit impulse response using windowing method (8.0 Marks)

$$H_d(e^{j\omega}) = e^{-(j3\omega)} \qquad -\frac{\pi}{4} \le \omega \le \frac{\pi}{4}$$
$$= 0 \qquad \frac{\pi}{4} \le |\omega| \le \pi$$

Use Hanning window with $N = 7$.

Hanning window: $0.5\ (1 - \cos(\frac{2\pi n}{M-1})) \quad 0 \le n \le (M-1)$