PVP 14

Code: EE6T1

III B.Tech - II Semester – Regular/Supplementary Examinations
AUGUST 2021

DIGITAL SIGNAL PROCESSING
(ELECTRICAL & ELECTRONICS ENGINEERING)

Duration: 3 hours   Max. Marks: 70

PART – A

Answer *all* the questions.   All questions carry equal marks

11x 2 = 22 M

1.

a) Determine the Z-transform of the signal $x[n] = a^n u[n] - b^n u[-n-1]$, and plot the ROC.

b) Determine the value of power and energy of the signal $x[n] = \sin(\frac{\pi}{4})n$.

c) State Circular Time Shift property of DFT.

d) Distinguish DIT FFT and DIF FFT.

e) The first five points of 8-point DFT of a real valued sequence are (0.25, 0.5-j0.5, 0, 0.5-j0.86, 0). Find the remaining three points.

f) What is the disadvantage of impulse invariant method.

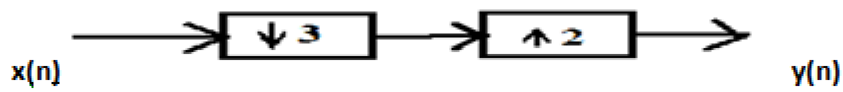
g) What is Gibbs phenomenon?

h) How many number of additions, multiplications and memory locations are required to realize a system H(z) having M zeros and N poles in Direct form-I and Direct form–II realizations.

i) Distinguish IIR and FIR Filter.

j) What are Multirate Systems? What is its importance in real time processing of signals.

k) Given x(n)={1, 2, 3, 4, 5, 6, 7, -1, -2, -3, -4} and x(n) is applied to down sampler and upsampler as shown in below figure. Find y(n).



PART – B

Answer any *THREE* questions. All questions carry equal marks.

3 x 16 = 48 M

2. a) Determine, if the system describes the following input-output equation is linear or nonlinear. 8 M

$$y[n] = x[n] + \frac{1}{x[n-1]}$$

b) If impulse response h(n) = 2^n u [-n]. Test the system for causality and stability. 8 M

3. a) Determine the circular convolution of the sequences x[n]=[1, 0.5 , 1, 0.5, 1 ,0.5, 1, 0.5] and h[n]=[0, 1, 2, 3] and compare the result with linear convolution. 8 M

b) Compute 8 point DFT of the sequence
$x[n]=[1, 1, 1, 1, 0, 0, 0, 0]$ using DIT FFT algorithm. 8 M

4. a) In a speech recording system with a sampling frequency of 10,000 Hz, the speech is corrupted by random noise. To remove the random noise while preserving speech information, the following specifications are given.

Speech frequency range : 0 - 3000 Hz.

Stop band range : 4,000 - 5,000 Hz.

Passband ripple : 3 dB

Stopband attenuation : 25 dB.

Determine the filter order and transfer function using butterworth IIR filter. 8 M

b) For the analog transfer function $H(s)=\frac{2}{(s+1)(s+2)}$ determine $H(Z)$ using Bilinear Transformation method if the sampling frequency is 1Hz. 8 M

5. a) The desired frequency response of a low pass filter is

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega}, & -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$

Determine $H(e^{j\omega})$ for M=7 using a Hanning window. 8 M

b) Realize following digital filter by using direct form - II realization.

$$y(n) = \frac{3}{8} y(n-1) + \frac{3}{32} y(n-2) + \frac{1}{64} y(n-3) + x(n) + 3x(n-1) + 2x(n-2)$$

8 M

6. a) Define up sampling and how do you convert a sampling rate by non integer factor. 8 M

b) Explain the effects of aliasing in decimation with the frequency spectrum and discuss how the aliasing can be eliminated. 8 M