B.E./B.TECH. DEGREE EXAMINATION, DECEMBER 2020

Fifth Semester

EC18502-PRINCIPLES OF DIGITAL SIGNAL PROCESSING

(Regulation 2018)

Time: Three hours Maximum: 80 Marks

Answer **ALL** questions

PART A - (8 X 2 = 16 marks)

1. The 4-point DFT of the sequence x[n]= {1,0,2,3}is
   a) 0, -2-3j,0, -2+3j
   b) 2, 2+2j,0, 2-2j
   c) 6, -1+6j,2, -1-6j
   d) 6, -1+3j,0, -1-3j

2. A signal $x[n]=2\cos(\frac{\pi}{4}n+\frac{\pi}{3})+\sin(\frac{3\pi}{4}n)$ is passed through a linear phase filter with frequency response $H(e^{j\omega})$, given as

$$H(e^{j\omega}) = \begin{cases} A\,e^{-j2\omega} & \text{for } \frac{-\pi}{2} \le \omega \le \frac{\pi}{2}, \\ 0 & \text{for } -\pi \le \omega \le -\frac{\pi}{2} \text{ and } \frac{\pi}{2} \le \omega \le \pi. \end{cases}$$

   The filter output will be given by

   a) $2A\cos(\frac{\pi}{4}(n+2)+\frac{\pi}{3})+A\sin(\frac{3\pi}{4}(n+2))$
   b) $2A\cos(\frac{\pi}{4}(n-2)+\frac{\pi}{3})$
   c) $2A\cos(\frac{\pi}{4}(n-2)+\frac{\pi}{3})+A\sin(\frac{3\pi}{4}(n-2))$
   d) None of the above.

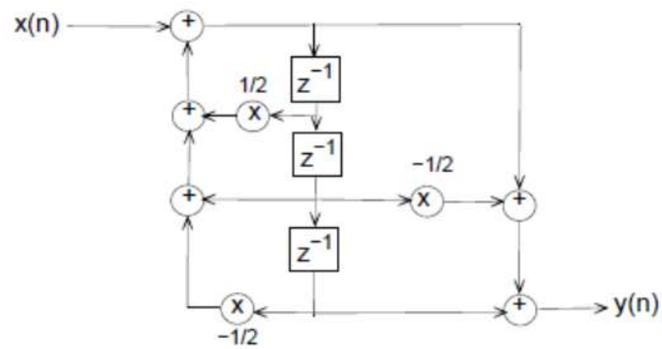
3. The digital transfer function H(z) by using impulse invariant method for the analog transfer function $H(s)=\left(\frac{1}{s+1}\right)$ and T=1 sec is given by

   a) $H(z)=\frac{1}{1-e^{-2}z^{-2}}$
   b) $H(z)=\frac{1}{1-e^{-1}z^{-1}}$
   c) $H(z)=\frac{1}{1-e^{-2}z^{-1}}$
   d) $H(z)=\frac{1}{1-e^{1}z^{-1}}$

4. Consider the assertions (steps) given below. Which among the following is a correct sequence of designing steps for the sampling rate converters?
   A. Computation of decimation/interpolation factor for each stage
   B. Clarification of anti-aliasing / anti-imaging filter requirements.
   C. Designing of filter at each stage.
   D. Calculation of optimum stages of decimation/ interpolation yielding maximum efficient implementation.

   a) A, B, C, D
   b) C, A, D, B
   c) D, A, B, C
   d) B, D, A, C

5. Given $x_1[n]=\{1,2,3,0\}$ and $x_2[n]=\{1,3,2,1\}$. Determine $x_3[2]$ if $X_3[k]=X_1[k]X_2[k]$. where $X(k)=DFT[x(n)]$

6. 

For the given IIR filter, determine the difference equation.

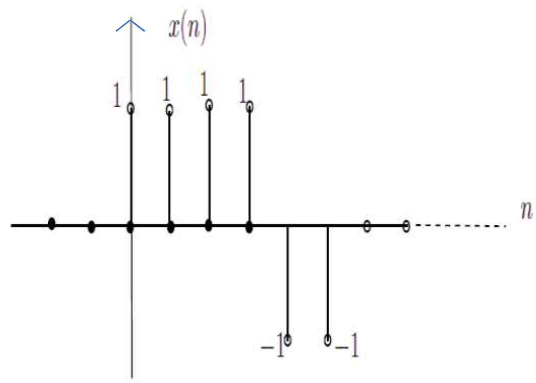
7. Determine the frequency response of FIR filter given by difference equation $y(n)=0.25x(n)+x(n-1)+0.25x(n-2)$ and check its linear phase characteristics.

8. Determine the dead band of the filter with pole at 0.5 and the number of bits used for quantization is 4(including sign bit).

PART B - (4 X16 = 64 marks)

09. (a) (i) Determine the response of a LTI filter using filtering of long duration sequences. **(10)**
$x[n]=\{1,-1,2,1,2,-1,1,3,1\}$ and $h[n]=(1,2,1\}$.

(ii) Find y(7) through circular convolution if $y(n)=x(n)*x(n)$. **(6)**



(OR)

(b) Given $x[n]=\{1,2,3,4,5,6,7,8\}$, compute X(k) using DIF FFT algorithm. Also determine Y[k] if $y[n]=\{5,6,7,8,1,2,3,4\}$ using circular time shifting property. **(16)**

10. (a) A high pass FIR filter of length 9 is required. It has to have a cut off frequency of 3 KHz and is intended to be used with a sampling frequency of 24 KHz. Determine the filter coefficients using Hamming window. Consider the filter to be causal. **(16)**

(OR)

(b) Design a linear phase FIR low pass filter with cut off frequency of $\pi/2$ radians per second using frequency sampling method with N=9.Also draw its linear phase structure. **(16)**

11. (a) Design a Butterworth digital IIR LPF using Impulse Invariant Transformation technique by taking T=1 second satisfying the following specifications, **(16)**

$$0.707 \leq |H(e^{j\omega})| \leq 1, \qquad 0 \leq \omega \leq 0.3\pi$$
$$|H(e^{j\omega})| \leq 0.2, \quad 0.75\pi \leq \omega \leq \pi$$

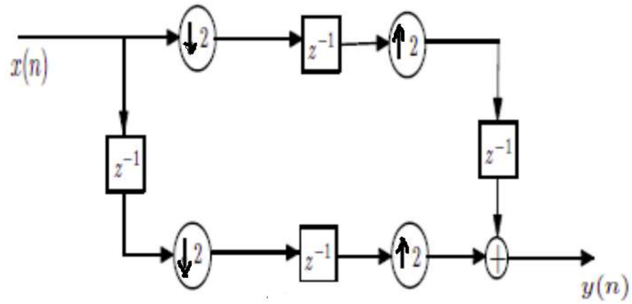
(OR)

(b) (i) Convert the analog filter with system function **(8)**
$H(s)=\frac{s+0.1}{(s+0.1)^2+0.9}$ into a digital IIR filter using Bilinear transformation with T=0.2 seconds.

(ii) Obtain the direct form, cascade and parallel realization structures for the signal represented by the difference equation as given below. **(8)**
$y[n] = \frac{1}{2}\, y[n-1] + \frac{1}{4}\, y[n-2] + x[n] + x[n-1]$

12. (a) (i) Briefly discuss about the techniques which will overcome the overflow in addition. **(8)**

(ii) **(8)**



In the given multirate system, determine the output y(n) as a function of x(n) with the relevant equations.

(OR)

(b) (i) Explain how the Quadrature Mirror Filter will be designed using multirate signal processing. **(8)**

(ii) Consider a second order IIR filter with **(8)**
$H(z)=\frac{1}{(1-0.5z^{-1})(1-0.45\ ^{-1})}$. Analyze the effect of quantization on pole locations of the given system function in direct form and in cascade form. Assume b = 3 bits.