Question Paper

Exam Date & Time: 26-May-2023 (02:30 PM - 05:30 PM)

MANIPAL ACADEMY OF HIGHER EDUCATION

FOURTH SEMESTER B.TECH. (ELECTRONICS AND COMMUNICATION ENGINEERING) DEGREE EXAMINATIONS - MAY/JUNE 2023
SUBJECT: ECE 2255/ECE_2255 DIGITAL SIGNAL PROCESSING

Marks: 50 | **Duration: 180 mins.**

Answer all the questions.

Missing data may be suitably assumed.

1A) Sketch the frequency sampling realization of M=16 and α=0, linear phase FIR filter which has frequency samples $H\left(\frac{2\pi k}{16}\right) = \begin{cases} 1, & k = 0, 1, 2 \\ 0.5, & k = 3 \\ 0, & k = 4, 5, .\ ., 7 \end{cases}$ (5)

1B) Consider an FIR filter with lattice coefficients K_1 = 0.45, K_2 = -0.61, K_3 =0.7. Obtain the impulse response of the filter and sketch its direct form structure. (3)

1C) Determine the system function of a causal LTI system, with zeros at $z = 0.5$ and $z = 0.8$, and a complex pair of poles at $z = 1.5\, e^{j\frac{\pi}{4}}$. State whether the system is stable and justify your answer, with the help of a pole-zero plot. (2)

2A) Develop radix-2 DIF FFT algorithm. Illustrate with signal flow diagram for N=8. Highlight the computational advantage of this algorithm. (5)

2B) Illustrate with mathematical relations, use of DFT/IDFT in determining the circular convolution between two finite duration sequences. Explain how this is used in determining the response of LTI system to the given input. (3)

2C) Consider the finite duration signal $x[n] = n, 0 \le n \le 7$ and 0 elsewhere with 8-point DFT X[k]. Using suitable properties of DFT, determine sequence y[n] whose 8-point DFT is Y[k] = Real part of |X[k]| (2)

3A) The specifications of the desired low-pass filter are (5)

- Passband edge: 4kHz, Stopband edge: 8 kHz
- Passband ripple: 1 dB, Stopband Attenuation: 40 dB
- Sampling frequency: 24 kHz

Determine the order and poles of Butterworth filter required to meet the above filter specification. Use bilinear transformation.

3B) For the filter specification given in Question 3A, determine analog system function $H_a(s)$ and use bilinear transformation to obtain $H(z)$ of Butterworth digital filter. (3)

3C) Explain how the Geortzel algorithm exploits the periodicity of the complex phase factor and obtain realization of the system to compute the DFT as a linear convolution. (2)

4A) Determine the filter coefficients for a linear phase FIR LPF of length M=7. The approximate desired frequency specifications for the filter is (5)

$$H_d\left(e^{j\omega}\right)=\begin{cases} e^{-j\omega\alpha}, & 0\le|\omega|\le 0.3\pi \\ 0, & elsewhere \end{cases}$$

Use suitable window with a minimum stop band attenuation of 50dB.

4B) Convert the analog filter to its equivalent digital filter whose system function is given by $H(s)=\frac{s+0.4}{s^2+0.8s+25.16}$ using impulse invariance technique. Assume sampling frequency of 10Hz. (3)

4C) Obtain the direct-form II realization for the system $H(z)=\frac{(1-z^{-1}+2z^{-2})}{(1+0.2z^{-1})(1-0.5z^{-1}+0.7z^{-2})}$ (2)

5A) Describe with mathematical expressions the Blackman-Tukey method of power spectrum estimation. Describe the spectral leakage and spectral resolution problems occurring in estimation of spectra from finite duration observation of signals. (5)

5B) Realize an efficient direct form structure of the linear phase FIR filter whose system function is (3)

$$H(z)=0.015-0.145z^{-1}+0.268z^{-2}-0.268z^{-4}+0.145z^{-5}-0.015z^{-6}$$

Determine the corresponding input-output equation.

5C) For the filter given in Question 5B, write the equations for the magnitude response and phase response. Is this filter suitable for the design of a lowpass filter? Justify your answer. (2)

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