Question Paper

Exam Date & Time: 03-May-2024 (02:30 PM - 05:30 PM)

MANIPAL ACADEMY OF HIGHER EDUCATION

FOURTH SEMESTER B.TECH. (ELECTRONICS AND COMMUNICATION ENGINEERING) DEGREE EXAMINATIONS - APRIL / MAY 2024
SUBJECT: ECE 2222/ECE_2222 - DIGITAL SIGNAL PROCESSING

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

Answer all the questions.

1A)	Compute the 6-point DFT of the sequence x(n) = {0, 1, 2, 3, 2, 1} using matrix multiplication method.	(5)
1B)	State and prove the circular convolution property of DFT of two sequences $x_1(n)$ and $x_2(n)$.	(3)
1C)	Describe the Goertzel algorithm with expressions. What is it used for?	(2)
2A)	Compute the 8-point DFT of the sequence x(n) = {1, 0.5, 0, -0.5, -1, -0.5, 0, 0.5} using decimation in frequency FFT algorithm. Illustrate that the computation is faster than the direct computation of DFT.	(5)
2B)	Analyze the FIR lattice structure whose lattice coefficients are: K_1 = 0.65, K_2 = -0.34 & K_3 = 0.8, and obtain its impulse response coefficients.	(3)
2C)	Realize the linear phase FIR filter of length M = 7, whose first four filter coefficients are: 1, 1/3, -1/8 and 1/5.	(2)
3A)	A LPF has the desired frequency response $\lvert H_d(e^{j\omega})\rvert = \begin{cases} 1, & 0 \le \lvert\omega\rvert \le 0.5\pi \\ 0, & elsewhere \end{cases}$ Determine the filter coefficients h(n) using frequency sampling technique. Assume filter length M=9.	(5)
3B)	Determine the unit sample response h(n) of a 4 length linear phase symmetric FIR filter having frequency response $H_r(0) = 1 \ and \ H_r\left(\frac{\pi}{2}\right) = 0.5$	(3)
3C)	From Q3B determine the system function H(z) and the phase $\emptyset(\omega) \ for \ H_r(\omega) > 0.$	(2)
4A)	Certain IIR Butterworth LPF has the following specifications $-1.5dB \le 20log_{10}\left(\lvert H(e^{j\omega})\rvert\right) \le 0dB\,,\ 0 \le \omega \le \pi/3$ $20log_{10}\left(\lvert H(e^{j\omega})\rvert\right) \le \ -10dB, \quad 0.5\pi \le \omega \le \pi$ Assume T=1 second. Obtain the prewarped analog edge frequency specifications , order of filter,3-dB cut-off frequency and poles of the filter.	(5)
4B)	For the filter specification given in Question 4A, determine the analog transfer function H(s).	(3)
4C)	For the filter specification given in Question 4A, determine the system function H(z). Use bilinear transformation.	(2)
5A)	Given the system function $H(z) = \frac{1+z^{-1}+0.5z^{-2}}{1+0.2z^{-1}-0.15z^{-2}}$. Obtain the lattice ladder structure.	(5)

5B) Convert the analog filter into its equivalent digital filter using impulse invariance method whose transfer function is given by $H(s) = \frac{s+1}{s^2+2s+17}$. Assume T=1 second. (3)

5C) Illustrate the concept of spectral leakage and spectral resolution problems occurring in spectral estimation from finite duration signals. (2)

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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)

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

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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FOURTH SEMESTER BTECH. (E & C) DEGREE END SEMESTER EXAMINATION
JUNE 2022
SUBJECT: DIGITAL SIGNAL PROCESSING (ECE - 2255)

TIME: 3 HOURS **MAX. MARKS: 50**

Instructions to candidates
- Answer **ALL** questions.
- Missing data may be suitably assumed.

Q. No.	Questions	M*	C*	A*	B*
1A.	Determine the location of poles and all possible impulse response associated with the system function $H(z) = \frac{5z^{-1}}{3-7z^{-1}+2z^{-2}}$. Indicate ROC of H(z) in each case	5	1	1,2	3
1B.	Using DFT-IDFT method determine the response of LTI system with impulse response h(n)= [1,2] to the input x(n)= [1,2,1].	3	2	1,2	3
1C.	Mention the procedure of overlap add method for filtering of long data sequences.	2	2	1,2	2
2A.	Develop DITFFT algorithm for N=8. Compute 8-point DFT of a sequence $x(n) = \{0.5, 0.5, 0.5, 0.5\}$ using DITFFT algorithm.	5	2	1,2	3
2B.	Explain how the Goertzel algorithm exploits the periodicity of the complex phase factor and obtain realization of the system to compute the DFT as a linear convolution.	3	2	1,2	2
2C.	Obtain the parallel structure for the following system $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$.	2	3	1,2	3
3A.	A second order low-pass Butterworth filter is required to meet the following specifications: $\omega_p = 0.3\pi$, $\omega_s = 0.7\pi$, -2dB ripple in the passband and a stopband attenuation of -20dB. Determine the pre-warped analog edge frequencies Ω_p *and* Ω_s, 3-dB cut off frequency Ω_c and transfer function H(s) of the filter, using bilinear transformation at 6Hz sampling.	5	3	1,2	3
3B.	For the above question given in Q3A, Obtain the digital filter system function H(z) using bilinear transformation at 6Hz sampling.	3	3	1,2	3

3C.	Derive the equation for phase response of an even symmetric linear phase FIR filter.	2	4	1,2	2
4A.	A low pass linear phase FIR filter is to be designed with the following desired frequency response: $$H_d(e^{jw}) = \begin{cases} e^{-j3w}, & for -\frac{\pi}{2} \le \omega \le \frac{\pi}{2} \\ 0, & for \frac{\pi}{2} \le \omega \le \pi \end{cases}$$ Determine the filter coefficients for M=7 using Hamming window.	5	4	1,2	3
4B.	For the above question given in Q4A, determine the transfer function H(z) and the frequency response $H(e^{jw})$ of the designed filter.	3	4	1,2	3
4C.	Explain the limitation of rectangular window function for the design of FIR filters.	2	4	1,2	2
5A.	Obtain the lattice ladder structure for $H(z) = \frac{1+2z^{-1}+3z^{-2}+2z^{-3}}{1+0.9z^{-1}-0.8z^{-2}+0.5z^{-3}}$	5	3	1,2	4
5B.	Describe Bartlett method of Power spectrum estimation. Highlight the computation requirement of this method.	3	5	1,2 ,18	2
5C.	List the advantages of Non parametric methods.	2	5	1,2 ,18	2

M*--Marks, C*--CLO, A*--AHEP LO, B* Blooms Taxonomy Level

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FOURTH SEMESTER BTECH. (E & C) DEGREE END SEMESTER EXAMINATION

JULY 2022

SUBJECT: DIGITAL SIGNAL PROCESSING (ECE - 2255)

TIME: 3 HOURS **MAX. MARKS: 50**

Instructions to candidates

- Answer **ALL** questions.
- Missing data may be suitably assumed.

Q. No.	Questions	M*	C*	A*	B*
1A.	Use unilateral z transform to find the response y(n)≥0, for the system described by the difference equation y(n)=x(n-1)+0.5y(n-1) with $x(n) = 0.25^n u(n)$ and $y(-1) = 1$.	5	1	1,2	3
1B.	Given two 8 point sequence $x_1(n)$=[A,C,A,D,E,M,I,C] and $x_2(n)$=[E,M,I,C,A,C,A,D] with 8 point DFT's $X_1(K)$ and $X_2(K)$. Express $X_2(K)$ in terms of $X_1(K)$ in a simplified form.	3	2	1.2	3
1C.	Mention the procedure of overlap save method for filtering of long data sequences.	2	2	1,2	2
2A.	Develop DIFFFT algorithm for N=8. Compute 8-point DFT of a sequence $x(n) = \{3, 2, 1, 0, 1, 2\}$ using DIFFFT algorithm.	5	2	1,2	3
2B.	Explain how the Goertzel algorithm exploits the periodicity of the complex phase factor and obtain realization of the system to compute the DFT as a linear convolution.	3	2	1,2	2
2C.	Obtain the direct form-II structure for the following system $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$.	2	3	1,2	3
3A.	Determine the system function of a second order analog Chebyshev type-1 low pass filter with pass band cut-off frequency 200 Hz at sampling frequency of 1000Hz. The allowable ripple in the pass band is 2dB.	5	3	1,2	3
3B.	Determine the order of a lowpass Butterworth filter that has a -3dB bandwidth of 500Hz and a stopband attenuation of -40dB at 1000Hz.	3	3	1,2	3

3C.	Derive the equation for phase response of an odd antisymmetric linear phase FIR filter.	2	4	1,2	2
4A.	A low pass filter is to be designed with the following desired frequency response: $$H_d(e^{jw}) = \begin{cases} e^{-j3w}, & for -\frac{\pi}{2} \le \omega \le \frac{\pi}{2} \\ 0, & for\ \frac{\pi}{2} \le \omega \le \pi \end{cases}$$ Determine the filter coefficients for M=7 using Hanning window.	5	4	1,2	3
4B.	For the above question given in Q4A, determine the transfer function H(z) and the frequency response $H(e^{jw})$ of the designed filter.	3	4	1,2	3
4C.	Explain the zero-location symmetry property of linear-phase FIR filter.	2	4	1,2	2
5A.	Obtain the lattice ladder structure for $y(n) = -0.1y(n-1) + 0.72y(n-2) + x(n) - 0.8x(n-1) + 0.15x(n-2)$	5	3	1,2	4
5B.	Describe Welch method of PSD estimation. Highlight the computation requirement of this method.	3	5	1,2,18	2
5C.	List the AR model estimation methods.	2	5	1,2,18	2

M*--Marks, C*--CLO, A*--AHEP LO, B* Blooms Taxonomy Level

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FOURTH SEMESTER BTECH. (E & C) DEGREE END SEMESTER EXAMINATION AUGUST 2021

SUBJECT: DIGITAL SIGNAL PROCESSING (ECE - 2255)

TIME: 2 HOURS **MAX. MARKS: 40**

Instructions to candidates

- Answer **any four full** questions.
- Missing data may be suitably assumed.

1A. It is required to design a causal discrete LTI system such that it produces an output $y[n] = \left(\frac{1}{3}\right)^n u[n]$ when the input $x[n] = \left(\frac{1}{2}\right)^n \left\{u[n] - \frac{1}{2}u[n-1]\right\}$
   i. Determine impulse response h(n) and system function H(z)
   ii. Determine difference equation.
   iii. Is the system stable? If so, identify and sketch the ROC of H(z)

1B. With example explain the concept of circular shift of a sequence. Determine 8-point circular convolution between the signals $x_1[n]$ = [1 2 3 1] and $x_2[n]$ = [4 3 2 1] using DFT/IDFT calculations.

(5+5)

2A. Using radix-2 DIF algorithm, determine 8-point inverse DFT of X[k]= [0 4 0 0 0 0 0 4]. Clearly indicate the values at every node of the flow diagram.

2B. Derive Goertzel algorithm for the computation of N-point DFT of a signal. Determine the system function H(z) and difference equation for the system that uses Geortzel algorithm to compute DFT value $X(-k)_N$ for the real valued signal x(n).

(5+5)

3A. Obtain and sketch the direct form I, direct form II and cascade structures for the IIR system $H(z) = \frac{2(z-1)(z^2+\sqrt{2}\,z+1)}{(z+0.5)(z^2-0.9z+0.81)}$
Second order sections are allowed in cascade structure.

3B. Consider an FIR system having impulse response h(n) = δ(n)+ δ(n-1)+ 0.5δ(n-2)+ δ(n-3)+ δ(n-4). Realize the system using frequency sampling structure.

(5+5)

4A. With relevant mathematical analysis explain the design of IIR filters by the bilinear transformation. Describe why pre-warping is necessary when using bilinear transformation.

4B. Determine the poles and transfer function H(s) for the third order analog Butterworth prototype (cut-off frequency=1 rad/sec) filter. Digitize this using impulse invariance transformation with T=0.1sec and obtain the system function H(z).

(5+5)

5A. Illustrate with diagram the frequency response of Chebyshev type-I LPF. Compute the minimum order and analog transfer function H(s) of such filter, where the maximum allowable ripple is 1dB in the pass-band extending from 0 to 0.1π radians/sec. The minimum attenuation should be 40dB at the stop-band edge frequency of 0.3π radians/sec.

5B. Design digital IIR notch filter to suppress 50 Hz interference. The filter should work at a sampling frequency of 500 Hz. The notch bandwidth should be 5Hz. Assume $b_0=1$. Write down the corresponding frequency response.

(5+5)

6A. Describe the time domain and frequency domain characteristics of linear phase FIR filters. Obtain an expression for system function H(z) of even length symmetric linear phase FIR filter in terms of (M-1)/2 coefficients where M is the length of the filter. Sketch the tapped delay line realization for the same.

6B. Determine coefficients of 9 length symmetric digital FIR high pass filter using causal hamming window. The filter has desirable pass band extending from 400 Hz to 1000 Hz at a sampling frequency of 2000 Hz. The filter should have precisely linear phase response in the pass band. Obtain the frequency response of this filter.

(5+5)