EC52

Model Question Paper for V Semester B E
ELECTRONICS AND COMMUNICATION ENGINEERING
DIGITAL SIGNAL PROCESSING

Time : 3 hrs. Max. Marks: 100

Note: 1. Answer any FIVE FULL questions
2. Use of interest tables and normal distribution tables allowed

1. a. Compute the N- point DFT of the Sequence,
$$x(n) = an, \quad 0 \le n \le N-1$$ (6 marks)

b. Compute the 8-point circular convolution for the following sequences.
$$x_1(n) = (1, 1, 1, 1, 0, 0, 0, 0)$$
$$x_2(n) = \sin(3\pi n/8), \quad 0 \le n \le 7$$ (6 marks)

c. For the sequences
$$x_1(n) = \cos(2\pi n/N),$$
$$x_2(n) = \sin(2\pi n/N), \quad 0 \le n \le N-1$$
determine the N-point
(i) circular autocorrelation of $x_1(n)$
(ii) circular autocorrelation of $x_2(n)$. (8 marks)

2. a. Let $x(n)$ be a real sequence of length- N and its N-point DFT is given by $X(k)$. Show that:
(i) $X(N-k) = X^*(k)$
(ii) $X(0)$ is real
(iii) If N is even, $X(N/2)$ is real. (8 marks)

b. The first five points of the eight-point DFT of a real sequence are
$$x(n) = (0.25, 0.125\text{–}j0.3018, 0, 0.125\text{–}j0.0518, 0).$$
Determine the remaining 3 points. (4 marks)

c. Compute the N-point DFT of the Blackman window,
$$w(n) = 0.42\text{–}0.5 \cos(2\pi n/N-1) + 0.08 \cos(4\pi n/N-1), \ 0 \le n \le N-1$$ (8 marks)

3. a. Compute the energy of the N-point sequence
$$x(n) = \cos(2\pi n/N), \quad 0 \le n \le N-1$$ (5 marks)

b. A designer has available a number of eight-point FFT chips. Show explicitly how he could interconnect three such chips in order to compute a 24-point DFT. (5 marks)

c. Derive the signal flow graph for N=8 point, radix-2 decimation-in-time FFT algorithm. (10 marks)

4. a. Compute the 8-point inverse DFT of the sequence:
$$X(k) = (20, -5.828\text{–}j2.414, 0, -0.171\text{–}j0.414, 0, -0.171+j0.414, 0, -5.828+j2.414)$$
using DIF, radix-2, FFT algorithm. (10 marks)

b. Explain how chirp-Z transforms are used for computing DFT using linear filtering. (6 marks)

c. Show that DFT of a real even sequence is purely real. (4 marks)

6. (a) Let $H(s) = \frac{s+\beta}{(s+\beta)^2+\lambda^2}$ be a causal analog transfer function. Determine the causal digital transfer function $G(z)$ from $H_a(s)$ using the impulse invariance method. (6 Marks)

(b) Design a digital bandpass filter from a second-order analog lowpass Butterworth filter $H(s) = \frac{1}{s^2+\sqrt{2}s+1}$ using the bilinear transformation. The cutoff frequencies for the digital filter should lie at $\omega = \frac{5\pi}{12}$ and $\omega = \frac{7\pi}{12}$. (8 Marks)

(c) Let $H(z)$ be an IIR lowpass filter with a zero at $z = z_k$. Let $G(z)$ be the bandpass filter obtained by applying a lowpass-to-bandpass transformation which moves the pole at $z = z_k$ of $H(z)$ to a new location $\hat{z} = \hat{z}_k$. Express $\hat{z}_k$ in terms of z_k. If the lowpass filter has a zero at $z = -1$, where are the corresponding zero(s) of the bandpass filter? (6 Marks)

7. (a) Obtain the direct form I, direct form II, cascade, and parallel structures for the system $H(z) = \frac{2(1-z^{-1})(1+\sqrt{2}z^{-1}+z^{-2})}{(1+0.5z^{-1})(1-0.9z^{-1}+0.81z^{-2})}$. (12 Marks)

(b) Determine the unit sample response $h(n)$ of a linear-phase FIR filter of length $M = 4$ for which the frequency response at $\omega = 0$ and $\omega = \frac{\pi}{2}$ is specified as $H(e^{j0}) = 1$ and $H(e^{j\frac{\pi}{2}}) = 0.5$. (8 Marks)

8. (a) A digital low-pass filter is required to meet the following specifications: Passband ripple: ≤ 1 dB; Passband edge: 4 kHz; Stopband attenuation: ≥ 40 dB; Stopband edge: 6 kHz; Sample rate: 24 kHz. The filter is to be designed using bilinear transformation on an analog system function. Determine the order of Butterworth and Chebyschev filter designs that must be used to meet the above specifications in the digital implementation. (11 Marks)

(b) Let $\mathcal{H}\{x(n)\}$ denote the ideal operation of Hilbert transformation on the sequence $x(n)$. Determine the net result when the sequence $x(n)$ is operated four times; i.e. determine $y(n) = \mathcal{H}\{\mathcal{H}\{\mathcal{H}\{\mathcal{H}\{x(n)\}\}\}\}$. (4 Marks)

(c) Determine all the FIR filters which are specified by the lattice parameters $k_1 = \frac{1}{4}$, $k_2 = \frac{1}{2}$, $k_3 = \frac{1}{3}$, and $k_4 = \frac{1}{6}$. (5 Marks)

Model Question Paper for V Semester B E —143— EC-52

DIGITAL SIGNAL PROCESSING

Time: 3hrs] [Max.Marks: 100]

Note: 1. Answer any FIVE full questions

1. (a) Let $x(n)$ and $h(n)$ be two four-point sequences defined as follows: $x(n) = \cos\left(\frac{\pi n}{2}\right)$, $n = 0, 1, 2, 3$; $h(n) = 2^n$, $n = 0, 1, 2, 3$. Calculate the four-point DFTs X(k) and H(k). Hence evaluate the circular convolution of $x(n)$ and $h(n)$. (12 Marks)

   (b) A finite-duration sequence $x(n)$ of length 8 has the 8-point DFT shown in the figure. A new sequence $y(n)$ of length 16 is defined by $y(n) = \begin{cases} x(n/2), & \text{n even,} \\ 0, & \text{n odd.} \end{cases}$ Sketch the 16-point DFT of $y(n)$. (8 Marks)

Fig. Q1b

2. (a) Let $X(z)$ be the z-transform of $x(n) = u(n) - u(n-6)$. If we sample $X(z)$ at $z = e^{j(2\pi/4)k}$, $k = 0, 1, 2, 3$, we obtain a sequence $X(k)$, $0 \le k \le 3$. Sketch the sequence $x(n)$ obtained as the inverse DFT of $X(k)$. (8 Marks)

   (b) Consider the finite-length sequence $x(n) = \{1, 0.75, 0.5, 0.25\}$. If the 4-point DFT of $x(n)$ is denoted $X(k)$, plot the sequence $y(n)$ whose DFT is $Y(k) = W_4^{3k} X(k)$. (4 Marks)

   (c) A real-valued N-point sequence $x(n)$ is called DFT bandlimited if its DFT $X(k) = 0$ for $k_0 \le k \le N - k_0$. We insert $(L-1)N$ zeros in the middle of a band-limited $X(k)$ to obtain the following LN-point DFT

$$Y(k) = \begin{cases} X(k), & 0 \le k \le k_0 - 1 \\ 0, & k_0 \le k \le LN - k_0 \\ X(k + N - LN), & LN - k_0 + 1 \le k \le LN - 1. \end{cases}$$

   Determine the relationship between $Ly(Ln)$ and $x(n)$, $0 \le n \le N-1$, where $Y(k)$ is the LN-point DFT of $y(n)$. (8 Marks)

3. (a) The linear convolution of 10,000-point sequence with a finite-duration impulse response that is 100 points long is to be implemented. The convolution is to be implemented by using DFTs and inverse DFTs of length 256. (i) If the overlap-add method is used, what is the minimum number of 256-point DFTs and the minimum number of 256-point inverse DFTs needed to implement the convolution for the entire 10,000-point sequence? Justify your answer. (ii) If the overlap-save method is used, what is the minimum number of 256-point DFTs and the

minimum number of 256-point inverse DFTs needed to implement the convolution for the entire 10,000-point sequence? Justify your answer. (8 Marks)

(b) If $x(n)$ is an N-point sequence, and $X(k)$ the corresponding N-point DFT, prove the following equality: $\sum_{n=0}^{N-1} |x(n)|^2 = \frac{1}{N} \sum_{k=0}^{N-1} |X(k)|^2$. (4 Marks)

(c) Let $x(n) = \left\{1, \sqrt{2}, 1, 0, -1, -\sqrt{2}, -1, 0\right\}$. Use the decimation-in-time FFT algorithm with input bit-reversed to compute the DFT of the above sequence. Provide the values at the intermediate stages. (8 Marks)

4. (a) A finite-length sequence $x(n)$ is nonzero in the interval $0 \leq n \leq 19$. This signal is input to the system shown in the figure, where, $h(n) = \begin{cases} e^{j\frac{2\pi}{21}\frac{(n-19)^2}{2}}, & 0 \leq n \leq 28 \\ 0 & \text{otherwise} \end{cases}$ and $W = e^{-j\frac{2\pi}{21}}$. The output $y(n)$ for the interval $19 \leq n \leq 28$ can be expressed in terms of $X(e^{j\omega})$ for appropriate values of ω. Write an expression for $y(n)$ in this interval in terms of $X(e^{j\omega})$. (6 Marks)

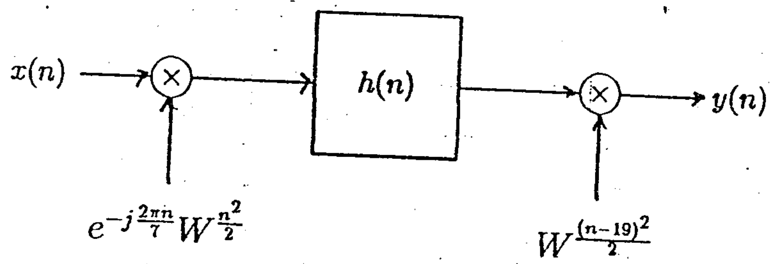

Fig. Q4a

(b) Let $x(n)$ be a 32-point sequence, and suppose that $x_1(n) = x(32-n)$. If $x_1(n)$ be the input to the following system, find an expression for $y_k(32)$ in terms of $X(e^{j\omega})$, the discrete-time Fourier Transform of the original sequence $x(n)$. (6 Marks)

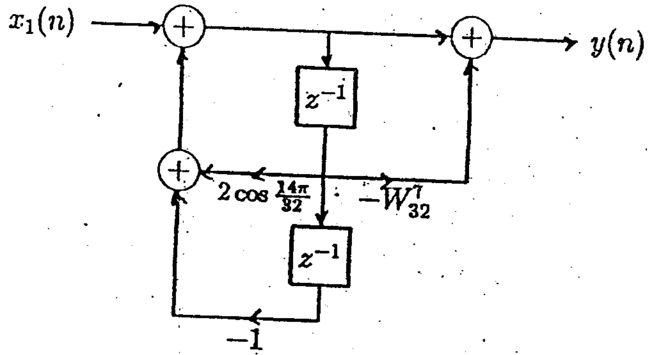

Fig. Q4b

(c) Suppose that $x(n)$ is a finite-duration sequence of $N = 1024$points. It is desired to evaluate the z-transform $X(z)$ of the sequence at the points $z_k = e^{j\frac{2\pi k}{1024}}$, Ec-52

$k = 0, 100, 200, \ldots, 1000$. Choose an algorithm for performing this computation efficiently. Explain how you arrived at your answer. (4 Marks)

(d) Suppose that a computer program implementing the FFT algorithm is available for computing the N-point DFT i.e., the input to the program is the N-point sequence $x(n)$, and the output is the DFT $X(k)$. Let $X(k)$ be the N-point DFT of the sequence $x(n)$, $0 \leq n \leq N-1$. What is the output of the FFT algorithm if the input to the program is $X(k)$, $0 \leq k \leq N-1$? (4 Marks)

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5. (a) Consider the system shown below:

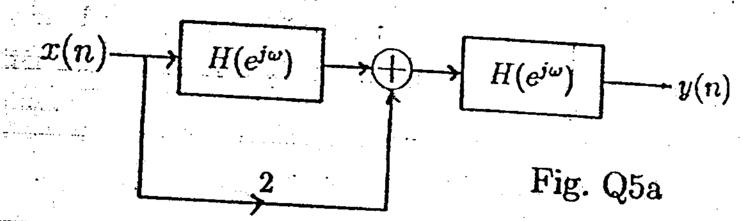

Fig. Q5a

(i) If the impulse response of $H(e^{j\omega})$ is of finite duration and symmetric, i.e., $h(n) = \begin{cases} h(-n) & -L \leq n \leq L \\ 0 & \text{otherwise} \end{cases}$, determine whether the overall impulse response $g(n)$ is FIR and symmetric. (ii) Let $H(e^{j\omega})$ satisfy the following specifications:

$$\begin{aligned} (1-\delta_1) \leq H(e^{j\omega}) \leq (1+\delta_1), &\quad 0 \leq \omega \leq \omega_p \\ -\delta_2 \leq H(e^{j\omega}) \leq \delta_2, &\quad \omega_s \leq \omega \leq \pi \end{aligned}$$

and the overall frequency response $G(e^{j\omega})$ satisfy specifications of the form

$$\begin{aligned} A \leq G(e^{j\omega}) \leq B, &\quad 0 \leq \omega \leq \omega_p \\ C \leq G(e^{j\omega}) D, &\quad \omega_s \leq \omega \leq \pi \end{aligned}$$

Determine A, B, C, and D in terms of δ_1 and δ_2. (12 Marks)

(b) Use the Kaiser window method to design a discrete-time filter with generalised linear phase that meets specifications of the following form:

$$\begin{aligned} |H(e^{j\omega})| \leq 0.01, &\quad 0 \leq \omega \leq 0.25\pi, \\ 0.95 \leq |H(e^{j\omega})| \leq 1.05, &\quad 0.35\pi \leq \omega \leq 0.6\pi, \\ |H(e^{j\omega})| \leq 0.01, &\quad 0.65\pi \leq \omega \leq \pi. \end{aligned}$$

(i) Determine the minimum length of the impulse response and the value of the Kaiser window parameter β for a filter that meets the above specifications. (ii) What is the delay of the filter? (iii) Determine the ideal impulse response $h_d(n)$ to which the Kaiser window should be applied. (8 Marks)

6 a. A third order Butterworth lowpass filter has the transfer function $H(s) = \frac{1}{(s+1)(s^2+s+1)}$. Design H(z) using impulse – invariant technique. **(10 Marks)**

b. Obtain the direct form – I and parallel form realization for a digital filter described by the system function $H(z) = \frac{8z^3 - 4z^2 + 11z - 2}{(z - 1/4)(z^2 - z + 1/2)}$. **(10 Marks)**

7 a. Compare FIR and IIR filters. **(06 Marks)**

b. Design a low-pass filter with a cutoff frequency $\omega_c = \frac{\pi}{4}$, a transition width, $\Delta\omega = 0.02\pi$ and a stop band ripple $\delta_s = 0.01$. Use Kaiser window techniques. **(10 Marks)**

c. Distinguish between Butterworth and Chebyshev filters. **(04 Marks)**

8 a. Design a 17-tap linear-phase FIR filter with a cut-off frequency $\omega_c = \frac{\pi}{2}$, using frequency sampling technique. **(08 Marks)**

b. Develop the lattice-ladder structure for the filter with difference-equation: $y(n) + \frac{3}{4}y(n-1) + \frac{1}{4}y(n-2) = x(n) + 2x(n-1)$. **(09 Marks)**

c. Draw the direct form I structure for the given impulse response of a filter: $h(n) = \left(\frac{1}{2}\right)^n [u(n) - u(n-3)]$. **(03 Marks)**

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NEW SCHEME

EC52

USN

th Semester B.E. Degree Examination, July/August 2005

EC/TE/BM/ML

Digital Signal Processing

3 hrs.] [Max.Marks : 100

Note: *Answer any FIVE full questions.*

a) Define DFT of a finite duration sequence, $x(n)$ say $x(n) = \{1,1,1,0,0\}$. Illustrate and explain the sampling of the Fourier transform of the sequence. **(6 Marks)**

b) Find the DFT of a sequence

$x(n) = 1, \ for \ 0 \le n \le 2$
$\quad = 0, \ otherwise.$

for : i) N=4 ii) N=8

Plot C magnitude of the DFT X(k) and comment on the result obtained. **(14 Marks)**

a) i) If DFT $[x(n)] = X(k)$, then show that

DFT $[x((-n))_N] = X((-k))_N$

ii) If the DFT $[x(n)] = X(k)$, then show that

DFT $[x(n).e^{j2\pi/l_n/N}] = X((k-l))_N$.

iii) If DFT $[x(n)] = X(k)$, then show that

DFT $[x^*(n)] = X^*(N-k)$. **(12 Marks)**

) Make a comparison between circular convolution and linear convolution :

Given $x_1(n) = \{1,-1,-2,3,-1\}$ & $x_2(n) = \{1,2,3\}$.

F d the circular consolation of the $x_n(n)$ & $x_2(n)$. **(2+6=8 Marks)**

) What are the two methods used for the sectional convolution? Write briefly about each one of them. **(8 Marks)**

) Find the output $y(n)$ of a filter whose impulse response is $h(n) = \{1,1,1\}$ and the input signal to the filter is $x(n) = \{3,-1,0,1,3,2,0,1,2,1\}$ using overlap-save method. **(12 Marks)**

Why FFT is needed? What is the speed improvement factor in calculating 64-point DFT of a sequence using direct computation and FFT algorithm? **(8 Marks)**

What are the differences and similarities between DIT and DIF FFT algorithm? **(4 Marks)**

Compute the 8-point DFT of the sequence

$x(n) = \{0.5, 0.5, 0.5, 0.5, 0, 0, 0, 0\}$

Using the inplace radix-2 DIT algorithm. **(8 Marks)**

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5. (a) i) Name the types of filters based on impulse response.
   ii) Name the types of filters based on frequency response.
   iii) What do you understand by linear phase response?
   iv) What conditions on the FIR sequence $h(n)$ are to be imposed in order that the filter can be called a linear phase filter? (8 Marks)

   (b) The frequency response of a linear phase FIR filter is given by

   $H(e^{j\omega}) = e^{j3\omega}\,[2 + 1.8\,\cos\,3\omega + 1.2\,\cos 2\omega + 0.5\,\cos\omega]$

   Find the impulse response sequence of the filter. (12 Marks)

6. (a) What are advantages and disadvantages with the design of FIR filters using window function? (6 Marks)

   (b) Deduce the equation for the frequency spectrum for the rectangular window sequence defined by

   $$W_R(n) = 1\ ;\ for\ \frac{-(N-1)}{2} \leq n \leq \frac{N-1}{2}$$
   $$= 0\ ;\ otherwise$$

   What is the width of the main case of the spectrum? (6 Marks)

   (c) The frequency response of a filter is given by

   $H(e^{j\omega}) = j\omega\ ;\quad -\pi \leq \omega \leq \pi.$

   Design the filter, using a rectangular window function. Take $N = 7$. (8 Marks)

7. (a) Distinguish between Butterworth and Chebysher (Type I) filters. (4 Marks)

   (b) How can one design digital filters from analog filters? (3 Marks)

   (c) Design a Butterworth filter using the bilinear transformation for the following specifications : (13 Marks)

   $$0.8 \leq |H(e^{j\omega})| \leq 1\ ;\ for\ 0 \leq \omega \leq 0.2\pi$$
   $$|H(e^{j\omega})| \leq 0.2\ ;\ for\ 0.6\pi \leq \omega \leq \pi$$

8. (a) Describe the transformation relation used for converting an analog LPF into a HPF. (6 Marks)

   (b) i) Obtain the cascade realisation of the system function :

   $H(z) = (1 + 2z^{-1} - z^{-2})\,(1 + z^{-1} - z^{-2})$

   ii) Determine the direct form realisation of the system function :

   $H(z) = 1 + 2z^{-1} - 3z^{-2} - 4z^{-3} + 5z^{-4}$ (7 Marks)

   (c) Write a note on chirp-Z transform. (7 Marks)

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NEW SCHEME

EC52

USN | | | | | | | | | | |

Fifth Semester B.E. Degree Examination, January/February 2005

EC/TE/BM/ML

Digital Signal Processing

Time: 3 hrs.] [Max.Marks : 100

Note: *1. Answer any FIVE full questions.*
2. Use of normalized chebyshev Butterworth prototype tables NOT allowed.

1. (a) Let $x_p(n)$ be a periodic sequence with fundamental period N. Let $X_1(k)$ denote the N-point DFT of one period of $x_p(n)$ and $X_3(k)$ be the 3N-point DFT of three periods of $x_p(n)$. What is the relationship between $X_1(k)$ and $X_3(k)$ for $0 \le k \le N-1$. **(10 Marks)**

(b) Let x(n) be any N-point sequence and X(k) be the corresponding N-point DFT. Suppose that $x_1(n) = x((m-n))_N$, where $0 \le m \le N-1$. Starting from the definition of DFT, find $X_1(k)$, the N-point DFT of the sequence $x_1(n)$. Using this result, and any other properties of DFT, determine $X_1(k)$ if $x_1(n) = x((2-n))_8$, if the first five values of the 8-point DFT of a real valued sequence are as follows:

$\{16, -(2\sqrt{2}+2) - j(2\sqrt{2}+2), 0, (2\sqrt{2}-2), -j(2\sqrt{2}-2).0\}$ **(10 Marks)**

2. (a) A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n). If x(n)= {1,1,1,1,1,3,1,1,4,2,1,1,3,1}, h(n) = {1−1}, compute y(n) using the overlap save technique. Use only a 5-point circular convolution in your approach. **(12 Marks)**

(b) Consider a finite-duration sequence x(n)={0,1,2,3,4,5}.

i) Sketch the sequence $s(n)$ with 6-point DFT S(k) $=W_2^k X(k)$.

ii) Determine the sequence y(n) with 6-point DFT Y(k)=ReX(k), the real part of X(k). **(8 Marks)**

3. (a) Starting from the definition, prove the following:

i) $\Sigma_{n=0}^{N-1}$ y(n) $= \Sigma_{n=0}^{N-1} x_1(n) \Sigma_{n=0}^{N-1} x_2(n)$, where $x_1(n)$ and $x_2(n)$ are two N-point sequences and $y(n)$ the N-point circular convolution of $x_1(n)$ and $x_2(n)$.

ii) $\Sigma_{n=0}^{N_1} \mid x(n) \mid^2 = \frac{1}{N} \Sigma_{k=0}^{N_1} \mid X(k) \mid^2$, where x(n) is an N-point sequence, and X(k) the corresponding N-point DFT. Prove any intermediate result that you may use. **(8 Marks)**

(b) Let x(n)= $2\delta(n) + \delta(n-1) + \delta(n-3)$. Obtain the sequence y(n) whose 5-point DFT Y(k)= $(X(k))^2$, where X(k) is the 5-point DFT of x(n). **(4 Marks)**

(c) Let x(n) be the following 8-point sequence:

$$x(n)= \left\{\frac{1}{\sqrt{2}}, 1, \frac{1}{\sqrt{2}}, 0, -\frac{1}{\sqrt{2}}, -1, -\frac{1}{\sqrt{2}}, 0\right\}.$$

Use the decimation-in-time FFT algorithm to compute the DFT of the above sequence. Provide the values at the intermediate stages. **(8 Marks)**

4. (a) Let X(k), $0 \le k \le N-1$, to be discrete Fourier transform of a sequence x(n). How would you use an FFT algorithm to compute the inverse DFT (i.e., obtain x(n) from X(k))? Hence determine the sequence x(n) whose 8-point DFT is given by the sequence X(k)= $\{0, 4-j4, 0, 0, 0, 0, 0, 4+j4\}$. Use the decimation- in-frequency FFT algorithm to arrive at your answer. Provide the values at the intermediate stages. **(12 Marks)**

Contd.... 2

(b) Consider the sequence x(n) = u(n)- u(n-8). Using chirp-z transform, determine the following values $X(z_0)$ and $X(z_1)$ where $Z_0 = e^{j\frac{2\pi}{8}}$ and $z_1 = e^{j\frac{4\pi}{8}}$.

(8 Marks)

5. (a) Design a lowpass digital filter using bilinear transformation. The filter is to be monotonic in both stop-and pass-bands and has all of the following characteristics: (i) an acceptable passband ripple of 1 dB, (ii) a passband edge of 0.3π rad, and (iii) stopband attenuation of 40dB or greater beyond 0.6π rad.

(16 Marks)

(b) Transform the alalog filter H(s) $= \frac{s+3}{(s+1)(s+2)}$ to a digital filter using the matched z-transform. Let T=0.5sec.

(4 Marks)

6. (a) Using the impulse reaponse technique, design a lowpass digital filter that is equiripple in the passband, and monotone in the stopband. The filter passband edge is at 0.1π rad with a ripple of 2.5 dB or less, and the stopband edge at 0.2π rad with attenuation 40 dB or more. Use T=1.

(16 Marks)

(b) Transform the analog filter H(s)= $\frac{1}{s+\alpha}, \alpha > 0$ to a digital filter using the backward- difference mapping. Comment on the stability of the digital filter.

(4 Marks)

7. (a) Let the specifications on a filter be as follows:

$$0.99 \leq |H(e^{jw})| \leq 1.01, \ 0 \leq \omega \leq 0.2\pi$$
$$|H(e^{jw})| \leq 0.05, \ 0.25\pi \leq \omega \leq 0.6\pi,$$
$$0.99 \leq |H(e^{jw})| \leq 1.01, \ 0.7\pi \leq \omega \leq \pi$$

(i) These specifications approximate an ideal filter. Derive the impulse response of this ideal filter.

(ii) Design a linear-phase filter that satisfies the above specifications using the Kaiser window. Give the expression for the impulse response of the designed filter.

(10 Marks)

(b) Starting from a lowpass Butterworth prototype alanog filter, design a fourth-order Butterworth bandpass analog filter with upper and lower band edge frequencies 10rad/sec and 5rad/sec.

(4 Marks)

(c) Consider an FIR lattice filter with coefficients $K_1 = 0.65, K_2 = -0.34, k_3 = 0.8$. Find its impulse response. Draw the equivalent direct-form structure.

(6 Marks)

8. (a) By choosing an appropriate window, design a linear-phase, odd-length, low-pass FIR filter that satisfies the following specifications:

$$0.985 \leq |H(e^{jw})| \leq 1.015, \ 0 \leq \omega \leq 0.2\pi,$$
$$|H(e^{jw})| \leq 0.001, \ 0.3\pi \leq \omega \leq \pi$$

Give the expressions for the impulse response and the corresponding frequency response.

(8 Marks)

(b) Consider the system function H(z) $= \frac{1-\frac{1}{5}z^{-1}}{(1-\frac{1}{2}z^{-1}+\frac{1}{3}z^{-2})(1+\frac{1}{4}z^{-1})}$

i) Realise the system in direct form I.

ii) Realise the system in cascade form using first- and second-order form II sections.

iii) Realise the system in parallel form using first- and second-order direct form II sections.

(12 Marks)

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NEW SCHEME

EC52

Reg. No.

Fifth Semester B.E. Degree Examination, January/February 2006
Electronics & Communication/Telecommunication Engineering
Digital Signal Processing

Time: 3 hrs.) (Max.Marks : 100

Note: ***Answer any FIVE full questions.***

1\. (a) What is the difference between the discrete Fourier series and the discrete Fourier transform? **(3 Marks)**

(b) Consider the finite-length sequence $x(n)$; $x(n) = \left(1, \frac{3}{4}, \frac{1}{2}, \frac{1}{4}\right)$. The 4-point DFT of $x(n)$ is $X(K)$. Plot the sequence $y(n)$ whose DFT is $Y(K) = W_4^{3K}\ X(K)$. **(5 Marks)**

(c) Let $X(K)$ denote the N-point DFT of an N-point sequence $x(n)$. $X(K)$ itself is an N-point sequence. If the DFT of $X(K)$ is computed to obtain a sequence $x_1(n)$, determine $x_1(n)$ in terms of $x(n)$. **(6 Marks)**

(d) $x(n)$ denotes a finite-length sequence of length N. Show that $x\left[(-n)_N\right] = x\left[(N-n)_N\right]$. **(6 Marks)**

2\. (a) Prove Parseval's relation as applied to DFT. **(5 Marks)**

(b) Let $x_1(n) = \left(1, \frac{1}{2}, \frac{1}{4}, \frac{1}{8}\right)$ and $x_2(n) = (1, 1, 1, 1)$. Compute the DFT of $x_1(n)$ by the decimation in time FFT algorithm and that of $x_2(n)$ by the decimation in frequency FFT algorithm. Using the above results, evaluate the circular convolution of $x_1(n)$ and $x_2(n)$. **(3+3+4 Marks)**

(b) How many (real) storage registers are required to evaluate the DFT by an in-place FFT algorithm? **(3 Marks)**

(c) How many twiddles will make a butterfly fast? **(2 Marks)**

3\. (a) It is required to take DFT of a data stream of length 8192. However, the analyzer has only a fixed hardware implementation for a 2048 point DFT. Assuming other storage is available along with ways of adding and multiplying, how could the desired 8192-pt transform be obtained? **(8 Marks)**

(b) Derive the Goertzel algorithm for the computation of DFT. Compare this algorithm with that of the direct method of computing DFT, with respect to number of multiplications and additions. Draw the signal flow graph of the complex recursive computation. Can the said algorithm, at any point of time, be more efficient than the FFT approach? **(6+2+2+2 Marks)**

Contd.... 2

4. (a) Develop a transformation for the solution of a first order linear constant coefficient difference equation by using trapezoidal approximation for the integral approximation. Highlight the features of transformation.

(10 Marks)

(b) Using the bilinear transformation $S = \frac{1-Z^{-1}}{1+Z^{-1}}$, What is the image of $S = e^{j\pi/2}$ in the Z-plane.

(5 Marks)

(c) What is the contour in the Z-plane that is the image of $J\Omega$ axis in the S-plane for the mapping $S = \frac{Z-1}{T}$? Are stable system in the S-plane, mapped into stable system in the Z-plane?

(5 Marks)

5. (a) Discuss the frequency sampling method of FIR filter design. Using the above principle, design a FIR filter with $h(n) = (1,\ 2,\ 1)$. Also indicate the signal flow graph.

(4+4+4 Marks)

(b) Design a FIR low pass filter with the frequency response, using rectangular window

$$h_d(\omega) = e^{\frac{-j\omega_c(N-1)}{2}} \qquad \frac{-\pi}{2} \le \omega \le \frac{\pi}{2}$$
$$= 0 \qquad elsewhere$$

(8 Marks)

6. (a) Let $h(n)$ be the unit sample response of a FIR filter so that $h(n) = 0$ for $n < 0,\ n \ge N$. Assume $h(n)$ is real. The frequency response of this filter can be represented in the form

$$H(e^{j\omega}) = \hat{H}\ (e^{j\omega})\ e^{j\Theta(\omega)}$$

i) Find $\Theta(\omega)$ for $0 \le \omega \le \pi$ When $h(n)$ satisfies the condition

$$h(n) = h(N-1-n)$$

(10 Marks)

ii) If N is even, show that

$$h(n) = h(N-1-n)$$

Implies that $H(\frac{N}{2}) = 0$, where $H(K)$ is the N-point DFT of $h(n)$. (4 Marks)

(c) Bring out the comparison between IIR and FIR filters.

(6 Marks)

7. (a) Design a digital LPF with a passband magnitude characteristic that is constant to within 0.75 dB for frequencies below $\omega = 0.2613\pi$ and stop band attenuation of atleast 20 dB for frequencies between $\omega = 0.4018\pi$ and π. Determine the transfer function $H(Z)$ for the lowest order Butterworth design which meets the specifications. Use bilinear transformation.

(12 Marks)

(b) Design an analog Chebyshev filter for which the squared magnitude response $|H_a(J\Omega)|^2$ satisfies the condition.

$$20\ log_{10}\ |H_a(J\Omega)|_{\Omega=0.2\pi} \ \ge -1$$
$$20\ log_{10}\ |H_a(J\Omega)|_{\Omega=0.3\pi} \ \le -15$$

(8 Marks)

Contd.... 3

8. (a) Consider an analog system function

$$H_a(S) = \frac{S+a}{(s+a)^2+b^2}$$

Determine the digital filter from an analog filter by means of impulse invariance. When would it produce good results?

(6 Marks)

(b) Obtain a parallel realization for the following $H(Z)$:

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

Also indicate the governing equations. **(8 Marks)**

(c) Implement a digital network whose unit-sample response is $e^{j\omega_o n}\,u(n)$. While implementing this system with a complex unit-sample response, the real and imaginary parts are to be distinguished as separate outputs.

(6 Marks)

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No...1

EC52

USN

NEW SCHEME

Fifth Semester B.E. Degree Examination, Dec.06/Jan.07

EC / TC

Digital Signal Processing

Time: 3 hrs.] [Max. Marks:100

Note: 1. *Answer any FIVE questions.*
2. *Use of normalized Chebyshev and Butter worth prototype tables NOT allowed.*

1 a. Define DFT. Establish relationship between DTFT and DFT. **(03 Marks)**

b. Compute 8 – point DFT of a sequence $x(n) = (-1)^{n+1}$, $0 \le n \le 7$. Also plot the magnitude of DFT. **(10 Marks)**

c. State and prove the following DFT properties.
i) Time reversal of a sequence ii) Circular frequency-shift of a sequence. **(07 Marks)**

2 a. Compute 8-point circular convolution for the following sequence, using time domain formula $x_1(n) = \{1, 1, 1, 1, 0, 0, 0, 0\}$ and $x_2(n) = \cos\left(\frac{2\pi n}{8}\right), 0 \le n \le 7$. **(08 Marks)**

b. Compute x(n) for the given DFT X(K) = (2, 1+j, 0, 1-j). Use matrix method. **(06 Marks)**

c. Let x(n) be a finite length sequence with X(K) = (10, -2+j2, -2, -2-j2). Using the properties of DFT, find the DFT's of the following sequences.
i) $x_1(n) = x((n+2))_4$ and ii) $x_2(n) = x(4-n)$ **(06 Marks)**

3 a. A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n). If x(n) = { 1, 4, 3, 0, 7, 4, -7, -7, -1, 3, 4, 3 }, h(n) = { 1,2 }. Compute y(n) using overlap add technique. Use only a 5 – point circular convolution in your approach. **(10 Marks)**

b. For sequence x(n) = (2,0,2,0), determine X(2), using Goertzel algorithm. Assume initial conditions are zero. **(06 Marks)**

c. Let x(n) be a real sequence defined by x(n) = (1, 2, 3, -4), without evaluating its DFT, X(k), find i) $\sum_{k=0}^{3} X(K)$ ii) X(0). **(04 Marks)**

4 a. Develop the DIF FFT algorithm for N = 8. Using the resulting signal flow graph compute the 8 – point DFT of the sequence $x(n) = \sin\left(\frac{\pi}{2}n\right), 0 \le n \le 7$. **(11 Marks)**

b. First five points of eight point DFT of a real valued sequence is given by X(k) = { 0, 2 + j2, - j4, 2 - j2, 0 }. Determine the remaining points. Hence find the sequence x(n) using DIT FFT algorithm. **(09 Marks)**

5 a. Let $H(S) = \frac{1}{s^2 + \sqrt{2}S + 1}$ represent the transfer function of a low pass filter with a pass band of 1 rad/sec. use frequency transformation to find the transfer functions of the following analog filters. i) A low pass filter with pass band of 10 rad/sec
ii) A high pass filter with cut off frequency of 10 rad/sec. **(05 Marks)**

Contd... 2

b. The system function of a low pass digital filter is given by, H(Z) = 0.5 [illegible]

Determine i) cut off frequency, w_p and

ii) Use a low pass transformation to obtain another low pass filter with $w_p{}^1 = 1$ rad (06 Marks)

c. Name the types of windows used in the design of FIR filters. Write the ananlytical equations and draw the magnitude response characteristics of each window. (09 Marks)

6 a. A low pass FIR filter is to be designed with the following desired frequency response

$$H_d(w) = \begin{cases} e^{-j2w} & |W| < \pi/4 \\ 0 & \frac{\pi}{4} < |W| < \pi \end{cases}$$

Determine filter coefficient $h_d(n)$ and h(n) if w(n) is a rectangular window defined as

$$W_r(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0 & \text{otherwise} \end{cases}$$

Also, find the frequency response, H(w) of the resulting FIR filter.

b. Compare FIR and IIR digital filters. (… Marks)

c. Determine the transfer functions H(Z) of an FIR filter to implement, $h(n) = \delta(n) + 2\delta(n-1) + \delta(n-2)$, using frequency sampling techniques. (04 Marks) (08 Marks)

7 a. The transfer function of analog filter is given by $H(s) = \dfrac{1}{(S+1)(S+2)}$

Find i) H(Z) using impulse invariance method

ii) H(Z) if $F_s = 5$ samples/sec (06 Marks)

b. Design an IIR low pass Butterworth digital filter to satisfy the following specifications. i) Pass band ripple = 1dB.

ii) Pass band edge frequency = 100 π rad/sec

iii) Stop band attenuation = 35dB

iv) Stop band edge frequency = 1000 π rad/sec

v) Sampling rate of 2000 samples/sec. use Bilinear transformation technique. (14 Marks)

8 a. Determine the order and the poles of a type –I low pass Chebyshev filter that satisfies the following constraints.

$$0.8 \le |H(W)| \le 1, \quad 0 \le W \le 0.2\pi$$

$$|H(W)| \le 0.2, \quad 0.6\Pi \le W \le \pi$$

(0… Marks)

b. Realize the system function

$$H(Z) = \frac{1}{2} + \frac{1}{3}Z^{-1} + Z^{-2} + \frac{1}{4}Z^{-3} + Z^{-4} + \frac{1}{3}Z^{-5} + \frac{1}{2}Z^{-6}$$

(04 Marks)

c. Obtain the cascade form realization for the given difference equation.

$$Y(n) = \frac{3}{4}Y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1).$$

Also, draw the signal flow graph and its transposed graph. (0… Marks)

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(10)

USN

EC52

Fifth Semester B.E. Degree Examination, Dec. 07 / Jan. 08
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note : *1. Answer any FIVE full questions.*
2. Use of normalized Chebyshev, Butterworth prototype tables NOT allowed.

1 a. If X(K) is the DFT of the sequence x(n), determine the N-point DFTs of the sequences $x_c(n) = x(n)\cos\frac{2\pi K_0 n}{N};\ 0 \le n \le N-1$ and $x_s(n) = x(n)\sin\frac{2\pi K_0 n}{N};\ 0 \le n \le N-1$. Use appropriate properties. **(06 Marks)**

b. State and prove circular convolution property and Parseval's theorem. **(08 Marks)**

c. Find the Z-transform of the sequence x(n) = {0.5, 0, 0.5, 0}. Using Z-transform result find its DFT. **(06 Marks)**

2 a. Distinguish between linear convolution and circular convolution. **(02 Marks)**

b. Using overlap-save method, compute y(n), of a FIR filter with impulse response h(n) = {3, 2, 1} and input x(n) = {2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1}. Use only 8-point circular convolution in your approach. **(10 Marks)**

c. Find the 4-point DFT of sequence $x(n) = 6 + \sin\left(\frac{2\pi n}{N}\right)$, n = 0, 1, ... N-1. **(08 Marks)**

3 a. Tabulate the number of complex multiplication and complex additions required for the direct computation of DFT and FFT algorithm for N = 16, 32, 128. **(04 Marks)**

b. Compute circular – convolution using DFT and IDFT formulae for the following sequences, $x_1(n) = n$ and $x_2(n) = \cos\frac{n\pi}{2}$ for $0 \le n \le 3$. **(10 Marks)**

c. Write a note on chirp-Z transform algorithm. **(06 Marks)**

4 a. i) Using DIF-FFT algorithm compute DFT of the sequence $x_1(n) = \{1, 2, -1, 2, 4, 2, -1, 2\}$.
ii) If $x_2(n) = x_1(-n)$ without performing FFT find $X_2(k)$ using $X_1(k)$. **(10 Marks)**

b. Derive the signal flow graph for 8-point Radix-2 DIT-FFT algorithm. **(10 Marks)**

5 a. Design a digital IIR low-pass Butterworth filter that has a 2 db passband attenuation at a frequency of 300π rad/sec and atleast 60 db stop band attenuation at 4500π rad/sec. Use backward difference transformation. **(10 Marks)**

b. Design a single pole lowpass digital filter with 3 db bandwidth of 0.3π, using the bilinear transformation applied to the analog filter $H(s) = \frac{\Omega_c}{s + \Omega_c}$, where Ω_c is the 3 db bandwidth of the analog filter. **(05 Marks)**

c. The system function of the first order normalized lowpass filter is $H(s) = \frac{3}{s+5}$. Obtain the system function of second order bandpass filter having passband from 1 kHz to 3.5 kHz. **(05 Marks)**

5. a. A low-pass analog filter is defined by the following transfer function and the corresponding impulse response:

$$H_a(s) = \alpha/(s+\alpha) \leftrightarrow h_a(t) = \alpha e^{-\alpha t}$$

(i) What is the gain at dc? At what radian frequency is the analog frequency response zero? At what time has the analog frequency response decayed to $1/e$ of the initial value?

(ii) Prewarp the parameter α and perform bilinear transformation to obtain the digital transfer function $H(z)$ from the analog design. What is the gain at dc? At what frequency is the response zero? Given an expression for the 3-dB radian frequency. Also find $h(n)$.

(10 marks)

b. Find $H(z)$ for the following analog transfer functions using impulse invariant transformation.

(i) $H_a(s) = b/\{(s+a)^2+b^2\}$

(ii) $H_a(s) = 1/(s+a)^2$

(10 marks)

6. a. Design an IIR digital filter that when used in the pre-filter A/D-H(z)-D/A structure will satisfy the following specifications (use Chebyshev prototype).

(i) LPF with 2-dB cutoff at 100 Hz.

(ii) Stop band attenuation of 20dB or greater at 500 Hz, and

(iii) Sampling rate of 4000 samples/second.

Verify the design.

(14 marks)

b. Show that an FIR filter will have linear phase if and only if

$$h(n) = \pm h(N-1-n),\ n = 0,1 \ldots N-1$$

Take only N equal to odd in the analysis.

(8 marks)

7. a. The desired frequency response of a low pass filter is given by

$$H_d(\omega) = \begin{cases} e^{-j3\omega}, & |\omega| < 3\pi/4 \\ 0, & 3\pi/4 < |\omega| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if Hamming window is used with N=7.

(8 marks)

b. Use the window method with a Hamming window to design a 7-tap FIR differentiator. The frequency response of an ideal differentiator is defined as below.

$$H_d(\omega) = j\omega,\ -\pi \le \omega \le \pi$$

Compute and plot the magnitude response of the resulting FIR differentiator. (12 marks)

8. a. A low-pass filter has the desired frequency response,

$$H_d(\omega) = \begin{cases} e^{-j3\omega}, & 0 < \omega < \pi/2 \\ 0, & \pi/2 < \omega < \pi \end{cases}$$

Determine $h(n)$ based on frequency-sampling technique. Take N=7. (10 marks)

b. Sketch the direct form – I, direct form – II and transposed realization for the system function given below.

$$H(z) = (2z^2+z-2)/(z^2-2)$$

(10 marks)

143 (1) / (3)

USN

EC52

Fifth Semester B.E. Degree Examination, June / July 08
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note : *1. Answer any FIVE full questions.*
2. Use of normalized Chebyshev and Butterworth prototype tables are not allowed.

1 a. Find DFT of the sequence $x(n) = \begin{cases} 1 & 0 \le n \le 2 \\ 0 & \text{otherwise} \end{cases}$ for N = 8. Plot $|X(k)|$ and $\angle X(k)$. **(10 Marks)**

b. State and prove the following DFT properties : i) Time reversal of a sequence. ii) Circular Time shift of a sequence iii) Parseval's theorem. **(10 Marks)**

2 a. Compare linear and circular convolution. **(04 Marks)**

b. Compute circular convolution using DFT and IDFT for the following sequences. $x_1(n) = \{2, 3, 1, 1\}$ $x_2(n) = \{1, 3, 5, 3\}$. **(12 Marks)**

c. The even samples of the 11 – point DFT of a length-11 real sequence are given by X(0) = 2, X(2) = -1 –j3, X(4) = 1 + j4, X(6) = 9 + j3, X(8) = 5, X(10) = 2 + j2. Determine the missing odd samples of the DFT. **(04 Marks)**

3 a. Let x(n)be a finite length sequence with X[k] = {0, 1+j, 1, 1 – j} using the properties of DFT, find DFT's of the following sequences. i) $x_1(n) = e^{j\frac{\pi}{2}n} x(n)$ ii) $x_2(n) = \cos\left(\frac{\pi}{2}n\right) x(n)$ iii) $x_3(n) = x((n-1))_4$ iv) $x_4(n) = (0,0,1,0) \otimes_4 x(n)$. **(08 Marks)**

b. Find x [2] , given x(n) = { 1, 0, 1, 0} using Goertzel algorithm. Assume initial conditions as zero. **(04 Marks)**

c. Write a note on Chirp Z – Transform algorithm. **(08 Marks)**

4 a. What are the generic differences and similarities between DIT and DIF FFT algorithm? Explain. **(05 Marks)**

b. Find the DFT of the sequence x(n) = {1, 2, 3, 4, 4, 3, 2, 1} using Radix – 2 DIT FFT algorithm. **(10 Marks)**

c. Compute the DFT's of the sequence $x(n) = \cos n\frac{\pi}{2}$, where N = 4, using DIF FFT algorithm. **(05 Marks)**

5 a. How the frequency transformation helps in design of filters? Explain. **(06 Marks)**

b. The system function of a Low pass digital filter is given by $H[z] = \frac{1}{2} \frac{(1+z^{-1})}{(2-z^{-1})}$. Determine i) cut – off frequency, w_p and ii) use a low pass transformation to obtain another Lowpass filter with $w'_p = 1$ rad. **(06 Marks)**

c. Design an FIR Low pass filter of length 7 using rectangular window with pass band gain of unity, cut off frequency of 200 Hz and sampling frequency of 1kHz. **(08 Marks)**

6 a. If h[n] is impulse response of an FIR filter, so that h[n] = 0 for n < 0 and n ≥ N. Assume h[n] is real, symmetric with respect to midpoint for odd N. The frequency response of this filter is represented as $H(e^{jw}) = Hr(e^{jw}) \cdot e^{j\theta(w)}$. Find $Hr(e^{jw})$ and $\theta(w)$ for $0 \le w \le \pi$, when h[n] satisfies the condition h[n] = h[N – 1 – n]. (06 Marks)

b. Design an ideal Hilbert transformer having frequency response $H_d(e^{jw}) = \begin{cases} -j \text{ for } 0 \le w \le \pi \\ j \text{ for } -\pi \le w \le 0. \end{cases}$ using rectangular window for N = 11. (10 Marks)

c. Compare the FIR and IIR filters. (04 Marks)

7 a. Determine the order and the poles of a type – 1 low pass Chebyshev filter for the following specifications:
PassBand ripple : - 3 d B ; Stop Band attenuation : - -20dB ; PassBand edge : - 2 rad/sec
StopBand edge : - 4 rad/sec. (08 Marks)

b. Design a IIR Low pass Butterworth digital filter to satisfy the following analog specifications :
i) PassBand ripple : ≤ 3.01 dB ii) PassBand edge : 500 Hz iii) StopBand attenuation : ≥ 15dB iv) StopBand edge : 750 Hz v) Sample rate : 2kHz. Use Bilinear transformation technique. Also obtain the difference equation realization. (12 Marks)

8 a. Convert the analog filter into a digital filter whose system function is $H(s) = \frac{[illegible]}{(s+0.2[illegible]}$
Assume T = 1 sec. Use Impulse invariant technique. (04 Marks)

b. Obtain a parallel realization for the system described by

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

(08 Marks)

c. Consider a three stage FIR Lattice structure having the co-efficients : $K_1 = 0.65$, $K_2 = -0.34$, and $K_3 = 0.8$. Evaluate its impulse response by tracing a unit impulse δ[n] at its i/p through the Lattice structure. Also draw its direct form – I structure. (08 Marks)

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06EC52

Fifth Semester B.E. Degree Examination, Dec.08/Jan.09
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note : *1. Answer any FIVE full questions choosing at least Two questions from each part A and B.*
2. Standard notations are used.
3. Missing data if any, may be suitabley assumed.
4. Draw neat diagrams wherever necessary.

PART – A

1 a. Let X (k), $0 \le k \le N-1$ be the N point DFT of the sequence x (n), $0 \le n \le N-1$. We define

$$\hat{X}(k) = \begin{cases} X(k) & 0 \le k \le k_c, N-k_c \le k \le N-1 \\ 0 & k_c < k < N-k_c \end{cases}$$

and we compute the inverse N point DFT of $\hat{x}(k)$, $0 \le k \le N-1$. What is the effect of this process on the sequence x (n)? Explain. **(04 Marks)**

b. Determine N point DFT of $x(n) = \cos\frac{2\pi k_o n}{N}$, $0 \le n \le N-1$. **(06 Marks)**

c. State and prove the relationship between Fourier series coefficient of a continuous time signal and DFT. **(10 Marks)**

2 a. State and prove : i) Ciseular convolution property of DFT; ii) DFT of real and even sequence. **(10 Marks)**

b. Determine the response of an LTI system with h (n) = { 1, -1, 2} for an input x (n) = { 1, 0, 1, -2, 1, 2, 3, -1, 0, 2 }. Employ over lap add method with block length L = 4. **(10 Marks)**

3 a. How many complex multiplications are required for direct computation of 64 point DFT? What is its value if FFT is used? **(04 Marks)**

b. Determine 8 point DFT of x (n) = { 1, 0, -1, 2, 1, 1, 0, 2 } using radix -2 DIT FFT algorithm. Show clearly all the intermediate results. **(12 Marks)**

c. What are the two properties of phase factor W_N that are exploited in fast Fourier Transform algorithms? Prove them. **(04 Marks)**

4 a. Determine 4 point IDFT of :
X (k) = { 2.5, -0.25 + j 0.75, 0, -0.25 – j 0.75 } using DIFFFT algorithm. **(04 Marks)**

b. Consider a finite duration sequence x (n), $0 \le n \le 7$ with Z Transform, X (Z). It is desired to compute X (Z) at the following sets of values :

$$Z_k = 0.8\, e^{j\left(\frac{2\pi k}{8} + \frac{\pi}{8}\right)} ;\ 0 \le k \le 7.$$

Sketch the points Z_k in the complex plane. Determine a sequence s(n) such that its DFT provides the desired samples of X(Z). **(06 Marks)**

c. Explain Goertzal algorithm and draw the DF – II structure for the same. **(10 Marks)**

PART – B

5 a. Determine the order of Butterworth and Chebyshev approximation analog filters used to meet the following specifications : Pass band attenuation of 1 d B at 4 kHz and stop band attenuation of 40dB at 6 kHz. (06 Marks)

b. Design a Chebyshev type 1 analog filter to meet the following specifications : Pass band attenuation 2 d B at 4rad / sec and stop band attenuation of 10 d B at 7 rad / sec. (14 Marks)

6 a. Determine the FIR filter coefficients, h (n), which is symmetric low pass filter with linear phase. The desired frequency response is :

$$H_d(w) = \begin{cases} e^{-j\left(\frac{M-1}{2}\right)w} & ; \quad 0 \le |w| \le \pi/4 \\ 0 & ; \quad \text{otherwise.} \end{cases}$$

Employ rectangular window with M = 7. (08 Marks)

b. What is Gibbs phenomenon? How it can be reduced? (04 Marks)

c. Show that the roots of H(Z) occur in reciprocal pair for a linear phase FIR filter. (08 Marks)

7 a. Explain, how an analog filter is mapped on to a digital filter using impulse invariance method. What are the limitations of the method? (08 Marks)

b. Design a digital band pass filter from a 2nd order analog Low pass Butterworth prototype filter using bilinear transformation. The lower and upper cutoff frequencies for band pass filter are 5π/12 and 7π/12. Assume T = 2sec. (12 Marks)

8 a. Consider a FIR filter with system function :
$H(z) = 1 + 2.82z^{-1} + 3.4048z^{-2} + 1.74z^{-3}$. Sketch the direct form and lattice realizations of the filter. (08 Marks)

b. For y (n) = -0.1y (n-1) + 0.2y (n-2) + 3x (n) + 3.6x (n-1) + 0.6x (n-2), obtain direct form I and II, cascade form and parallel form with single pole – zero subsystems. (12 Marks)

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143

2002 SCHEME

EC52

USN

Fifth Semester B.E. Degree Examination, Dec.08/Jan.09
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: *Answer any FIVE full questions*

1 a. Consider the finite duration signal x(n)={1,2,3,1}
(i) Compute its 4 point DFT by solving explicitly the 4-by-4 system of linear equations defined by the IDFT formula.
(ii) Check the answer in part(i) by computing the 4 point DFT using its definition. (12 Marks)
b. Establish the relationship between DFT & DFS. (04 Marks)
c. If DFT of x(n) is x(k). What are DFTs of $x((-n))_N$ and IDFT of $X^*(N-k)$ (04 Marks)

2 a. Perform x(n)*h(n) using overlap-add technique. Given h(n)={1,1,1}, x(n)={1,2,3,4,-2,1,0,6,2,8,9,6,2,2,1} Calculate N from the formula (06 Marks)
b. Compute circular convolution using DFT & IDFT for
$X_1(n)=2\delta(n)+3\delta(n-1)+\delta(n-2)+\delta(n-3)$
$X_2(n)=\delta(n)+3\delta(n-1)+5\delta(n-2)+3\delta(n-3)$ (06 Marks)
c. Let x(k) denote a 6 point DFT of real sequence x(n)={1,-1,2,3,0,0}. Without computing the IDFT, determine y(n) whose 6 point DFT is given by $y(k)=W_3^{2k}x(k)$ (04 Marks)

3 a. Explain how windowing a signal leads to spectral leakage of the signal. (05 Marks)
b. Compare Direct computation of DFT & FFT with respect to number of complex additions and multiplications for 8 point sequence. Also explain how the inplace computations and Bit reversal order helps to save memory in FFT algorithm. (05 Marks)
c. What are the differences and similarities between DIT and DIF-FFT algorithms? Find the 4 point real sequence x(n) if its 4 point DFT samples are x(o)=6, x(1)=-2+j2, x(2)= -2 Use DIF-FFT algorithm. (10 Marks)

4 a. Explain Linear filtering approach to computation of the DFT using Goertzel algorithm. (06 Marks)
b. Draw signal flow diagram for a 16 point radix-2 DITFFT algorithm. (06 Marks)
c. The system function of the first order normalized low pass filter is given as $H_{an}(S)=\frac{1}{S+1}$ Obtain the system function of second order bandpass filter having passband from 1 KHz to 2 KHz. (04 Marks)
d. Convert the following low pass digital filter of cutoff frequency 0.2π into highpass filter of cutoff frequency 0.3π radians $H(z)=\frac{0.245(1+z^{-1})}{1-0.509z^{-1}}$ (04 Marks)

5 a. Compare design methods for Linear phase FIR filters. (04 Marks)
b. A linear phase FIR filter has response given by e^{-j2w}. What is the order of the filter? Assume filter is symmetric. (04 Marks)
c. Design a low pass FIR filter using frequency sampling technique having cutoff frequency of $\pi/2$ rad/sample. The filter should have linear phase and length of 17. (12 Marks)

6 a. Design the band pass linear phase FIR filter having cutoff frequencies of w_{c1}=1 rad/sample and w_{c2}=2 rad/sample. Obtain the unit sample response through following window:

$$w(n)=\begin{cases}1 & \text{for } 0 \le n \le 6 \\ 0 & \text{otherwise}\end{cases}$$

Also obtain magnitude/frequency response. (12 Marks)

b. Compare IIR and FIR digital filters with respect to design methods, complexity of implementation, applications & memory requirement. (04 Marks)

c. Explain the window where a designer can adjust the tradeoff between main spectral lobe width and side lobe levels. (04 Marks)

7 a. Consider the analog filter having the transfer function $H_a(S) = \frac{1}{S+2}$

(i) Transform Ha(S) to a digital filter H(z) using impulse invariance technique. Sampling rate=2 samples/second.

(ii) Will the impulse response h(n) match the impulse response h(t) of the analog filter at the sampling instants. Assume system to be causal.

(iii) Will the step response s(c) match the step response s(t) of the analog filter at the sampling instants. (12 Marks)

b. A second-order Butterworth low pass analog filter with a half power frequency of 1 rad/sec is converted to a digital filter h(z) using Bilinear Transformation at sampling rate 1/T =1 Hz.

(i) What is the transfer function H(S) of analog filter.

(ii)What is the transfer function h(z) of digital filter. (08 Marks)

8 a. Consider the signal flow graph given in Fig.8(a). Write difference wquation & system function. (06 Marks)

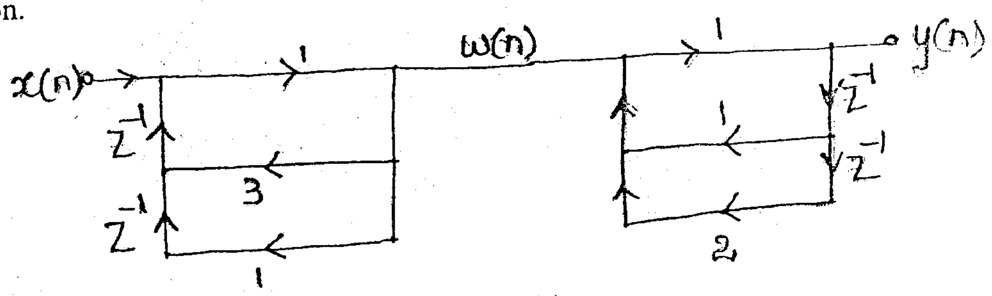

Fig.8(a)

b. Consider the 3stage FIR lattice structure having the coefficients k1=0.1, k2=0.2, k3=0.3. Evaluate its impulse response by tracing a unit impulse δ(n) at its input through the lattice structure. (06 Marks)

c. Write short notes on any two:

(i) Frequency prewarping

(ii) Matched z transformation

(iii)Hilbert Transform (08 Marks)

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06EC52

Fifth Semester B.E. Degree Examination, June-July 2009

Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: 1. ***Answer any FIVE full questions choosing at least two from each part.***
2. ***Missing data if any may be suitably assumed.***
3. ***Use of normalized chebyshev, Butterworth prototype tables not allowed.***

Part A

1 a. Define DFT. Establish a relationship between the Fourier series coefficients of a continuous-time signal and DFT. **(07 Marks)**

b. The N-point DFT of the N-point sequence, $x(n)=e^{-j\left(\frac{\pi}{N}\right)n^2}$, for N even is $X(k)=\sqrt{N}e^{-j\frac{\pi}{4}}e^{j\left(\frac{\pi}{N}\right)k^2}$. Determine the 2N-point DFT of the 2N-point sequence $y(n)=e^{-j\left(\frac{\pi}{N}\right)n^2}$, assuming that N is even. **(05 Marks)**

c. i) Let N = 4M, where M is an integer. Let

$$X(k)=\begin{cases}0.5, & k=M\\ 0.5, & k=3M\\ 0, & \text{otherwise}\end{cases}.$$

Compute x(n). Let $y(n)=(-1)^n x(2n),\quad 0\le n\le 2M-1$. Compute V(k).

ii) Evaluate the sum $s=\sum_{n=0}^{15} x_1(n)x_2^*(n)$ when $x_1(n)=\cos\left(\frac{3\pi n}{8}\right)$, $x_2(k)=3,\ 0\le k\le 15$. **(08 Marks)**

2 a. Determine 4-point circular convolution of the following pair of length-4 sequences using [illegible] IDFT computations: x(n)=(3, 2, 1, 4), h(n) = (2, 1, 1, 3) **(08 Marks)**

b. [illegible](2, 1), ω(n)=x(n)*y(n) and ω(n)=(6,−1,7,−4) compute the sequence y(n) **(07 Marks)**

c. [illegible] the output y(n) of a filter whose impulse response is h(n)=(1, -2) and input signal x(n)= (3, -2, 4, 1, 5, 7, 2, -9) using overlap-add method. Use only 5-point circular convolution in your approach. **(05 Marks)**

3 a. For sequence x(k)=(5, 3-j2, -3, 3+j2), determine x(2) using Goertzel algorithm. Assume that the initial conditions are zero. **(06 Marks)**

b. Develop the DIF-FFT algorithm to compute IDFT. Write the signal flow graph for N = 8. **(10 Marks)**

c. What are the orders of Butterworth and Chebyshev filters necessary to meet the following design specifications?

$\delta_P=\delta_S=0.01$, Ω_P = 0.6682 rad/sec, Ω_S = 1 rad/sec. **(04 Marks)**

4 a. A sequence x(n) is filtered using a first-order LTI system with an impulse response h(n) giving output y(n). A 4 point DFT and IDFTs via radix-2 FFT algorithms are used to compute y(n). Select these algorithms appropriately and write the signal flow graph between normal ordered x(n) and y(n). What is the length of a non zero padded sequences x(n) and h(n) where y(n) represents alias-free output of the circular convolution? **(12 Marks)**

b. Show that the product of two complex numbers (a+jb) and (c+jd) can be performed with three real multiplications and five additions. **(03 Marks)**

4 c. Consider the system in figure Q4 (c), where x(n), h(n) and y(n) are finite-length sequences. Use DFT and IDFTs to compute y(n) in terms of x(n) and h(n). (05 Marks)

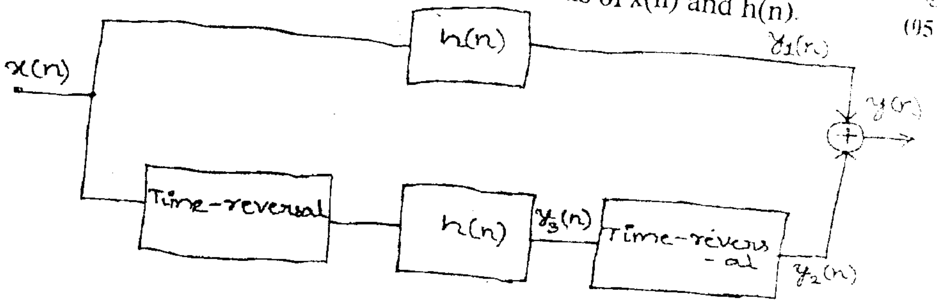

Fig. Q4 (c)

Part B

5 a. Show that the bilinear transformation maps.

i) The $j\Omega$ axis in s-plane onto the unit circle, $|z|=1$.

ii) The left half s-plane, $Re(s)<0$ inside the unit circle, $|z<1|$. (05 Marks)

b. Figure Q5 (b) shows the frequency response of an infinite-length ideal multi-band real filter. Find h(n), impulse response of this filter. Present the sketch of implementation of $\omega(n)h(n)$ (Truncated impulse response of this filter) via block diagram. Where $\omega(n)$ is a finite length window sequence? (12 Marks)

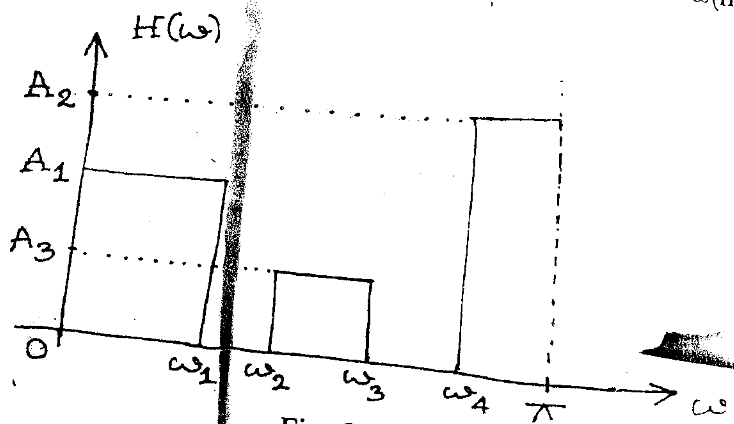

Fig. Q5 (b)

c. We are interested to design an FIR filter with a stopband attenuation of 64 dB and $\Delta\omega=0.05\pi$ using windows. Provide the means to achieve precisely this attenuation using suitable window function. (03 Marks)

6 a. The transfer function of analog low pass filter is given by $H(s)=\dfrac{(s-1)}{(s^2-1)(s^2+s+1)}$. Find H(z) using impulse invariance method. Take T = 1 sec. (06 Marks)

b. Design a linear phase highpass filter using the Hamming window for the following desired frequency response.

$$H_d(\omega)=\begin{cases} e^{-j3\omega} & \frac{\pi}{6}\le|\omega|\le\pi \\ 0 & |\omega|<\frac{\pi}{6} \end{cases}$$

$\omega(n)=0.54-0.46\cos\left(\dfrac{2\pi n}{N-1}\right)$, where N is the length of the Hamming window. (08 Marks)

6 c. Let $H(s)=\frac{1}{s^2+\sqrt{2}s+1}$, a second-order low pass Butterworth filter prototype having the half-power point at $\Omega = 1$. Determine the system function for the digital bandpass filter using bilinear transformation. The cutoff frequencies for the digital filter should lie at $\omega_L = \frac{5\pi}{12}$ and $\omega_u = \frac{7\pi}{12}$. Take T = 2. **(06 Marks)**

7 a. Obtain the direct form II, cascade and parallel structures for the following difference equation. $y(n)=\frac{3}{4}y(n-1)-\frac{1}{8}y(n-2)+x(n)+\frac{1}{3}x(n-1)$. **(08 Marks)**

b. Design a digital lowpass Butterworth filter using Bilinear transformation method to meet the following specifications. Take T = 2 sec.
Passband ripple ≤ 1.25 dB
Passband edge = 200 Hz
Stopband attenuation ≥ 15 dB
Stopband edge = 400 Hz
Sampling frequency = 2 kHz **(12 Marks)**

8 a. Design a linear phase lowpass FIR filter with 7 taps and a cutoff frequency of $\omega_C = 0.3\pi$ using the frequency sampling method. **(06 Marks)**

b. A z-plane pole-zero plot for a certain digital filter is shown in figure Q8 (b). The filter has unity gain at DC. Determine the system function in the form,

$$H(z)=A\left[\frac{(1+a_1z^{-1})(1+b_1z^{-2}+b_2z^{-2})}{(1+c_1z^{-1})(1+d_1z^{-1}+d_2z^{-2})}\right]$$

giving the numerical values for parameters A, a_1, b_1, b_2, c_1, d_1 and d_2. Sketch the direct form II and cascade realizations of the system. **(09 Marks)**

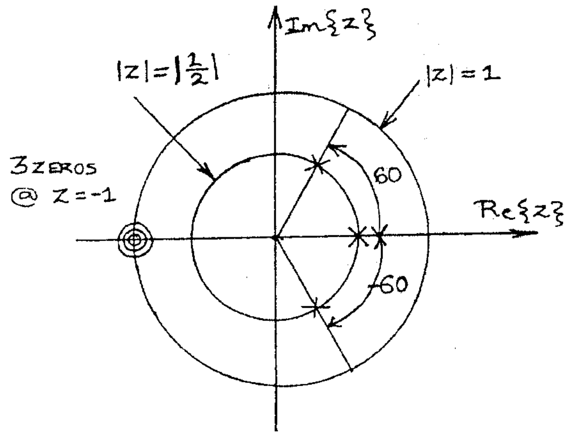

Fig. Q8 (b)

c. Determine the parameters K_m of the Lattice filter corresponding to the FIR filter described by the system function,

$H(z)=1+1.38z^{-1}+1.311z^{-2}+1.337z^{-3}+0.9z^{-4}$ **(05 Marks)**

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06EC52

Fifth Semester B.E. Degree Examination, Dec.09/Jan.10
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions, selecting at least TWO questions from each part.***

PART – A

1 a. Consider the sequence $x_1(n) = \{\underset{\uparrow}{0}, 1, 2, 3, 4\}$, $x_2(n) = \{\underset{\uparrow}{0}, 1, 0, 0, 0\}$, $s(n) = \{\underset{\uparrow}{1}, 0, 0, 0, 0\}$ and their 5-point DFTs
i) Determine a sequence y(n) so that $y(k) = x_1(k)x_2(k)$
ii) Is there a sequence $x_3(n)$ such that $s(k) = x_1(k)x_3(k)$? **(10 Marks)**

b. Suppose that we are given a program to find the DFT of a complex-valued sequence x(n). How can we use this program to find the inverse DFT of x(k)? **(04 Marks)**

c. Consider the sequence $x(n) = 4\delta(n) + 3\delta(n-1) + 2\delta(n-2) + \delta(n-3)$. Let x(k) be the six point DFT of x(n). Find the finite length sequence y(n) that has a six point DFT $y(k) = W_6^{4k}x(k)$. **(06 Marks)**

2 a. State and prove time shifting property of DFT. **(08 Marks)**

b. Explain how the DFT can be used to compute N equispaced samples of the Z transform, of an N-point sequence, on a circle of radius r. **(04 Marks)**

c. A long sequence x(n) is filtered through a filter with impulse response h(n) to yield the output y(n), if $x(n) = \{1, 1, 1, 1, 1, 3, 1, 1, 4, 2, 1, 1, 3, 1\}$, $h(n) = \{1, -1\}$. Compute y(n) using overlap save techniques. [Use only 5 point circular convolution]. **(08 Marks)**

3 a. What is FFT? Explain Radix-2 DIT-FFT algorithm. **(06 Marks)**

b. Develop decimation-in-frequency (DIF) FFT algorithm with all necessary steps and neat signal flow diagram used in computing N-point DFT, x(k) of a N-point sequence x(n). Using the same, compute the DFT of sequence $x(n) = \{1,1,1,1,1,1,1,1\}$. **(14 Marks)**

4 a. Let, $x(n) = \{1, 2, 0, 3, -2, 4, 7, 5\}$ with a 8-point DFT x(k). Evaluate the following without explicitly computing x(k):

i) x(0) ii) x(4) iii) $\sum_{k=0}^{7} x(k)$ iv) $\sum_{k=0}^{7} |x(k)|^2$ **(08 Marks)**

b. Let x(n) be a given sequence with N points with x(k) the corresponding DFT. Denote the operation of finding DFT as follows: $x(k) = F\{x(n)\}$. What is the resulting sequence x(n) operated upon four times. i.e., determine y(k) where $y(k) = F\{F\{F\{F\{x(n)\}\}\}\}$ **(06 Marks)**

c. What is linear filtering? Explain how DFT is used in linear filtering. **(06 Marks)**

PART – B

5 a. A designer is having a no. of 8-point FFT chips. Show explicitly how he should interconnect three chips in order to compute a 24-point DFT. **(10 Marks)**

b. Explain analog to analog frequency transformation. **(10 Marks)**

6 a. Determine the order of a Chebyshev digital filter that meets the following specifications:
i) 1 dB ripple in the passband $0 \le |w| \le 0.3\pi$
ii) At least 60 dB attenuation in the stopband $0.35\pi \le |w| \le \pi$. (06 Mark

b. Use the bilinear transformation to design a discrete time Chebyshev high pass filter with equiripple passband with $0 \le |H(e^{jw})| \le 0.1,\ 0 \le |w| \le 0.1\pi$ and $0.9 \le |H(e^{jw})| \le 1.0,\ 0.3\pi \le |w| \le \pi$ (14 Mar

7 a. Consider the pole zero plot, as shown in Fig.7(a).

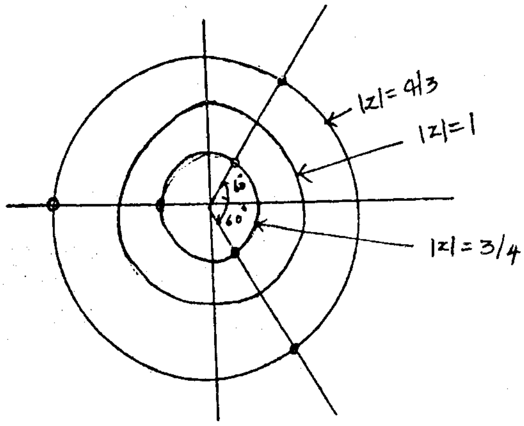

Fig.7(a)

i) Does it represent an FIR filter? ii) Is it a linear phase system? (04 Ma

b. Compare FIR versus IIR. (06 Ma

c. A filter is be designed with the following desired frequency response.

$$H_d(w) = \begin{cases} 0, & -\frac{\pi}{4} < w < \frac{\pi}{4} \\ e^{-j2w}, & \frac{\pi}{4} < |w| < \pi \end{cases}$$

Find the frequency response of the FIR filter designed using a rectangular window de: as $w_R(n) = \begin{cases} 1, & 0 \le n < 4 \\ 0, & \text{otherwise} \end{cases}$. (10 M

8 a. Explain the structures used for realizing FIR filters by illustrations. (10 M

b. Consider the causal linear shift invariant filter, with system function

$$H(z) = \frac{1 + 0.875z^{-1}}{(1 + 0.2z^{-1} + 0.9z^{-2})(1 - 0.7z^{-1})}$$

Draw a signal flow graph for this system using
i) Direct form-I
ii) Direct form-II
iii) A cascade of first and second order systems realized in direct form-II
iv) A cascade of first and second-order systems realized in transposed direct form-II. (10 M

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06EC52

Fifth Semester B.E. Degree Examination, May/June 2010
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions, selecting at least TWO questions from each part.***

Important Note: 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages. 2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

PART – A

1 a. For the following sequences, find: **(12 Marks)**

i) N-point DFT of $x(n) = \cos\frac{2\pi}{N}K_0 n$ ii) 5-point DFT of $x(n) = \{1, 1, 1\}$

b. Find IDFT for the sequence : $x(k) = \{5, 0, (1-j), 0, 1, 0, (1+j), 0\}$ **(08 Marks)**

2 a. State and prove circular frequency shift property of DFT. **(04 Marks)**

b. Compute the circular convolution of the sequences $x_1(n) = \{2, 1, 2, 1)$ and $x_2(n) = \{1, 2, 3, 4)$ using DFT and IDFT method. **(08 Marks)**

c. Find the output y(n) of a filter whose impulse response is $h(n) = \{1, 2\}$ and the input signal to the filter is $x(n) = \{1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1\}$ using overlap-save method. **(08 Marks)**

3 a. Determine the number of complex multiplications, complex additions and trigonometric functions, required for direct computation of N-point DFT, **(10 Marks)**

b. How many complex multiplications and additions are required for 64-point DFT in FFT? **(04 Marks)**

c. Prove : i) Symmetry and ii) Periodicity property of a twiddle factor. **(06 Marks)**

4 a. Develop Radix-2 N-point DIT-FFT algorithm and draw the signal flow graph. **(12 Marks)**

b. Obtain 8-point DFT of the sequence, $x(n) = \{2, 1, 2, 1, 0, 0, 0, 0\}$. Using Radix-2 DIF-FFT algorithm. Show clearly all the intermediate results. **(08 Marks)**

PART – B

5 a. Given $|Ha(j\Omega)|^2 = \frac{1}{1+16\Omega^4}$, determine the analog filter system function Ha(S). **(08 Marks)**

b. Derive an expression for 'N' and Ω_{cp} of Butterworth filter if passband and stopband attenuations are in dB. **(08 Marks)**

c. Let $H(s) = \frac{1}{s^2+s+1}$ represents transfer function of a low pass filter with a passband of 1 rad/sec. Use frequency transformation to find the transfer function of the following analog filters:

i) A LPF with $\Omega'p = 10$ rad/sec ii) A HPF with $\Omega'p = 100$ rad/sec. **(04 Marks)**

6 a. Derive an expression for frequency response of a symmetric impulse response for N-odd. **(08 Marks)**

6 b. A lowpass filter is to be designed with the following desired frequency response:

$$H_d(e^{j\omega}) = H_d(\omega) = \begin{cases} e^{-ijw}, & |\omega| < \pi/4 \\ 0, & \pi/4 < |\omega| < \pi \end{cases}$$

Determine the filter coefficients $h_d(n)$ and $h(n)$ if $\omega(n)$ is a rectangular window defined as follows:

$$W_R(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & \text{otherwise} \end{cases}$$

Also, find the frequency response, $H(\omega)$ of the resulting FIR filter. (12 Marks)

7 a. Derive the expression for the bilinear transformation, to transform an analog filter to a digital filter, by trapezoidal rule and explain the mapping from s-plane to z-plane. (08 Marks)

b. Convert the analog filter with system function $Ha(s) = \frac{(s+0.1)}{(s+0.1)^2+9}$ into a digital filter (IIR) by means of impulse invariance method. (08 Marks)

c. Given the analog transfer function, $H(s) = \frac{(s+2)}{(s+1)+(s+3)}$. Find $H(z)$, using matched z-transform design. The system uses sampling rate of 10Hz (T = 0.1 sec). (04 Marks)

8 a. Obtain direct form I, direct form II, cascade and parallel structure for the system described by $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$. (16 Marks)

b. Obtain the direct form realization of linear phase FIR system given by $H(z) = 1 + \frac{2}{3}z^{-1} + \frac{15}{8}z^{-2} + \frac{2}{3}z^{-3} + z^{-4}$. (04 Marks)

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