KINGS

COLLEGE OF ENGINEERING

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

QUESTION BANK

SUBJECT CODE : EC1302 SEM / YEAR : V / III

SUBJECT NAME : DIGITAL SIGNAL PROCESSING

UNIT-I :FAST FOURIER TRANSFORM

PART - A

TWO MARKS:

1. Define about DFT and IDFT?
2. Find the values of W_N^k ,When N=8, k=2 and also for k=3.
3. Compare DIT radix-2 FFT and DIF radix -2 FFT.
4. Draw the radix-2 FFT–DIF butterfly diagram.
5. Draw the radix-2 FFT–DIT butterfly diagram.
6. What is the necessity of sectioned convolution in signal processing?
7. Define Correlation of the sequence.
8. State any two DFT properties.
9. Why impulse invariant transformation is not a one-to-one mapping?

PART - B

1. a) Compute 4- point DFT of casual three sample sequence is given by,

$x(n) = 1/3, \; 0 \le n \le 2$

$= 0, \quad \text{else}$ **(10)**

b) State and prove shifting property of DFT. **(6)**

2. Derive and draw the radix -2 DIT algorithms for FFT of 8 points. **(16)**

3. Compute the DFT for the sequence {1, 2, 0, 0, 0, 2, 1, 1}. Using radix -2 DIF FFT and radix -2 DIT- FFT algorithm. **(16)**
4. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap add overlap save method. **(16)**
5. In an LTI system the input x(n) = {1, 1, 1}and the impulse response h(n) = {-1, -}Determine the response of LTI system by radix -2 DIT FFT **(16)**
6. Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap save method **(16)**

UNIT-II: DIGITAL FILTERS DESIGN

PART - A

TWO MARKS:

1. Differentiate IIR filters and FIR filters.
2. Write the characteristics features of Hanning window
3. Define pre-warping effect? Why it is employed?
4. Give any two properties of Butterworth filter.
5. When a FIR filter is said to be a linear phase FIR filter
6. Write the characteristics features of rectangular window.
7. Write the expression for Kaiser window function..
8. What are the advantages and disadvantages of FIR filters?
9. Write the characteristics features of Hamming window
10. Why mapping is needed in the design of digital filters?

PART - B

1 . With a neat sketch explain the design of IIR filter using impulse invariant transformation. **(16)**

2. Apply impulse invariant transformation to $H(S) = \dfrac{2}{(S+1)(S+2)}$

with T =1sec and find H(Z). **(16)**

3. For a given specifications of the desired low pass filter is
   $0.707 \le |H(\omega)| \le 1.0, \quad 0 \le \omega \le 0.2\pi$
   $|H(\omega)| \le 0.08, \quad 0.4\pi \le \omega \le \pi$
   Design a Butterworth filter using bilinear transformation. **(16)**
4. Explain the procedural steps the design of low pass digital Butterworth filter and list its properties. **(16)**

5. The normalized transfer function of an analog filter is given by,

$$H_a(S_n) = \frac{1}{S_n^2 + 1.414S_n + 1}$$

with a cutoff frequency of 0.4 π, using bilinear transformation. **(16)**

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6. List the three well known methods of design technique for IIR filters and explain any one. **(16)**

7. Design a low pass filter using rectangular window by taking 9 samples of w(n) and with a cutoff frequency of 1.2 radians/sec.
   Using frequency sampling method, design a band pass FIR filter with the following specification. Sampling frequency F_s =8000 Hz, Cutoff frequency fc_1 =1000Hz, fc_2 = 3000Hz.Determine the filter coefficients for N =7. **(16)**

8. Design an ideal high pass filter with $H_d(e^{j\omega}) = 1$; $\pi/4 \le |\omega| \le \pi$
   $= 0$; $|\omega| \le \pi/4$ Using Hamming window with N =11 **(16)**
9. Determine the coefficients of a linear phase FIR filter of length N =15 which has a symmetric unit sample response and a frequency response that satisfies the conditions

$$H(2\pi k/15) = \begin{cases} 1 & ;\ \text{for } k = 0, 1, 2, 3 \\ 0.4 & ;\ \text{for } k = 4 \\ 0 & ;\ \text{for } k = 5, 6, 7 \end{cases}$$

(16)

10. Design and implement linear phase FIR filter of length N =15 which has following unit sample sequence H(k) = 1 ; for k = 0, 1, 2, 3
    0 ; for k =4, 5, 6, 7 **(16)**

11. Convert the analog filter in to a digital filter whose system function is

$$H(s)=\frac{S+0.2}{(S+0.2)^2+9}$$ .Use Impulse Invariant Transformation .Assume T=1sec **(16)**

12. The Analog Transfer function $H(s)=\frac{1}{(S+1)(S+2)}$.Determine H(Z) .Using Impulse Invariant Transformation .Assume T=1sec **.** **(8)**

13. Apply Bilinear Transformation to $H(s)=\frac{2}{(S+2)(S+3)}$ with T=0.1 sec. **(8)**

<u>UNIT-III:</u> <u>EFFECTS OF FINITE WORD LENGTH</u>

PART - A

<u>TWO MARKS:</u>

1. What are the effects of finite word length in digital filters?
2. List the errors which arise due to quantization process.
3. Discuss the truncation error in quantization process.
4. Write expression for variance of round-off quantization noise.
5. What is sampling?
6. Define limit cycle Oscillations, and list out the types.
7. When zero limit cycle oscillation and Over flow limit cycle oscillation has occur?
8. Why? Scaling is important in Finite word length effect.
9. What are the differences between Fixed and Binary floating point number representation?
10. What is the error range for Truncation and round-off process?

PART - B

1. The output of an A/D is fed through a digital system whose system function is $H(Z)=(1-\beta)z/(z-\beta)$, $0<\beta<1$.Find the output noise power of the digital system**.** **(8)**
2. The output of an A/D is fed through a digital system whose system function is H(Z)=0.6z/z-0.6. Find the output noise power of the digital system=8 bits **(8)**

3. Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR. **(16)**

4 a. Write a short notes on limit cycle oscillation **(8)**
b. Explain in detail about signal scaling **(8)**

5. A digital system is characterized by the difference equation Y(n)=0.95y(n-1)+x(n).determine the dead band of the system when x(n)=0 and y(-1)=13. **(16)**

6. Two first order filters are connected in cascaded whose system functions of the Individual sections are H1(z)=$1/(1-0.8z^{-1})$ and H2(z)=$1/(1-0.9z^{-1})$.Determine the Over all output noise power. **(16)**

UNIT-IV:POWER SPECTRUM ESTIMATION

PART - A

TWO MARKS QUESTIONS:

1. What is the need for spectral estimation?
2. How can the energy density spectrum be determined?
3. What is autocorrelation function?
4. What is the relationship between autocorrelation and spectral density?
5. Give the estimate of autocorrelation function and power density for random signals?
6. Obtain the expression for mean and variance for the autocorrelation function of random signals.
7. Define period gram.

PART - B

1. Explain how DFT and FFT are useful in power spectral estimation.(10)
2. Explain Power spectrum estimation using the Bartlett window.(8)
3. Obtain the mean and variance of the averaging modified period gram estimate.(16)
4. How is the Blackman and Tukey method used in smoothing the periodogram?(10)
5. Derive the mean and variance of the power spectral estimate of the Blackman and Tukey method.(10)
6. What are the limitations of non-parametric methods in spectral estimation?(8)
7. How the parametric methods overcome the limitations of the non-parametric methods?(10)

UNIT-V: DIGITAL SIGNAL PROCESSORS

PART - A

TWO MARKS:

1. What are the factors that influence the selection of DSPs.
2. What are the advantages and disadvantages of VLIW architecture?
3. What is pipelining? and What are the stages of pipelining?
4. What are the different buses of TMS 320C5x processor and list their functions
5. List the various registers used with ARAU.
6. What are the shift instructions in TMS 320 C5x.
7. List the on-chip peripherals of C5x processor.

PART - B

1. Explain in detail about the applications of PDSP **(10)**
2. Explain briefly:
   (i). Von Neumann architecture **(5)**
   (ii). Harvard architecture **(5)**
   (iii).VLIW architecture **(6)**
3. Explain in detail about
   (i). MAC unit **(8)**
   (ii). Pipelining **(8)**

4. Draw and explain the architecture of TMS 320C5x processor **(16)**
5. Explain in detail about the Addressing modes of TMS 320C50 **(16)**

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