SIDDHARTH INSTITUTE OF ENGINEERING &TECHNOLOGY:: PUTTUR
ELECTRONICS & COMMUNICATON ENGINEERING
DIGITAL SIGNAL PROCESSING (16EC422)

QUESTION BANK

UNIT-I

Introduction

1 a). Determine the linear convolution for the two sequences x(n)={3,2,1,2},h(n)= {1,2,1,2} [L1][CO1][7M]
b). Explain the power signal and Energy signal [L2][CO1][5M]

2) Find the forced response of the system described by the difference equation: [L2][CO1][12M]
$y(n)+ 2y(n-1)+y(n-2)=x(n)+x(n-1)$ for input $x(n) =(-1)^n u(n)$

3 a). Find impulse response of the system described by the difference equation [L2][CO1][6M]
$y(n)+ y(n-1)-2y(n-2)= x(n-1)+2x(n-2)$.
b). Find 4-point DFT of the sequence x(n)={1,6,4,3} [L2][CO1][6M]

4) State and prove following properties of DFT [L3][CO1][12M]
i) Linearity ii) Circular time shifting iii) Circular frequency shifting iv) Time reversal v) Complex conjugate.

5 a). Determine the circular convolution for the two sequences $x_1(n)$={1,2,3,4}, $x_2(n)$= {1,5,1,3} using concentric circles method. [L1][CO1][7M]
b). Explain the classification of discrete-time signals [L2][CO1][5M]

6 a). Find the natural response of the system described by the difference equation:
$y(n)+ 2y(n-1)+y(n-2)=x(n)+x(n-1)$ with initial conditions y(-1)=y(-2)=1. [L3][CO1][8M]
b) Justify how DFT can used as a linear Transform. [L1][CO1][4M]

7) Find the output y(n) of a filter whose impulse response is h(n)=[1,-1] and input x(n)= [1,-2,2,-1,3,- 4,4,-3] using
i) overlap add method ii)overlap-save method [L1][CO1][12M]

8) For each of the following systems, determine whether or not the system is static/dynamic, linear/non-linear, time variant/invariant, causal/non-causal, stable/unstable. [L1][CO1][12M]
i) y(n)=cos[x(n)] ii) y(n)= x(-n+2)

9a). Explain frequency analysis of discrete-time systems. [L2][CO1][6M]
b). Determine magnitude and phase response for the system described by the difference equation:
$y(n)=\frac{1}{2}x(n)+x(n-1)+\frac{1}{2}x(n-2)$ [L3][CO1][6M]

10a). Find 8 point DFT of the sequence x(n)=[1,2,1,0,2,3,0,1] [L1][CO1][7M]
b). Describe the relation between i) DFT to Z- transform ii) DFT to Fourier Series. [L1][CO1][5M]

UNIT-II

Fast Fourier Transform Algorithm

1) Compute 8-point DFT of the sequence x(n)= {1,2,3,4,4,3,2,1} using radix-2 DIT-FFT Algorithm.[L1][CO2][12M]
2a). Construct Radix-4 DIF FFT algorithm with neat sketch. [L2][CO2][7M]
b). Compare DFT and FFT algorithms. [L4][CO2][5M]

3) Compute 8-point DFT of the sequence x(n)= {1,2,1,2,1,2,2,1} using radix-2 DIF-FFT Algorithm.[L2][CO2][12M]

4 a). Construct the decimation in time FFT algorithm with butterfly diagram. [L2][CO2][7M]
b). Explain use of FFT in linear filtering and correlation. [L1][CO2][5M]

5 a). Explain decimation in frequency FFT algorithm. [L1][CO2][7M]
b). Compare radix-2 DIT-FFT and DIF-FFT algorithms. [L4][CO2][5M]

6) Compute IDFT of the sequence x(n)= { 7,-0.707-j0.707,-j, 0.707-j0.707,1, 0.707+j0.707,j, -0.707+j0.707}. [L2][CO2][12M]

7) Formulate the DFT by divide and conquer approach [L1][CO2][12M]
8) How do you compute DFT using [L1][CO2][12M]
a) The Goertzel Algorithm b) The chrip-z Transform

9 a). Explain Radix-4 FFT algorithm in decimation in time domain. [L2][CO2][7M]
b). Describe Quantization errors in the direct computation of DFT. [L1][CO2][5M]

10 a). With a neat sketch find 4 point DFT of the sequence x(n)=[1,6,7,4] using radix2 DIT-FFT algorithm. [L3][CO2][8M]
b). Interpret the applications of FFT algorithm. [L1][CO2][4M]

UNIT-III

Implementation of Discrete-Time Systems

1 (a). Discuss frequency sampling structure for FIR filter. [L1][CO3][6M]

(b). Realize FIR filter with system function in cascade form [L3][CO3][6M]

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

2.Consider the system $y(n) = y(n-1) + 2y(n-2) + x(n) + 3x(n-1)$ (i) Find H(z) (ii) Realize using direct form-I and direct form-II. [L1][CO3][12M]

3 (a). Obtain direct form-I,direct form-II,cascade,parallel form realization of following system:
$y(n) = 0.75y(n-1) - 0.125y(n-2) + 3x(n) + 7x(n-1) + x(n-2)$ [L2][CO3][12M]

4. A system is represented by a transfer function $H(Z) = 3 + \dfrac{4Z}{Z - \frac{1}{2}} - \dfrac{2}{Z - \frac{1}{4}}$ [L3][CO3][12M]

i) Does this system function H(Z) represent FIR or IIR.Justify?
ii) Give a difference equation for direct form-I structure.
iii) Draw the block diagram for direct form-II and give equations for implementation.

5 (a). Differentiate the different structures for IIR systems [L1][CO3][5M]
(b) Realize following system with difference equation in cascade form [L2][CO3][7M]
$y(n) = y(n-1) + 2y(n-2) + x(n) - x(n-1)$

6 (a). Explain lattice & lattice-ladder structure for IIR digital filter. [L2][CO3][6M]
(b). Discuss transposed structures. [L1][CO3][6M]

7. The transfer function of a discrete causal system is given as $H(Z) = \dfrac{1 - Z^{-1}}{1 - 0.2Z^{-1} - 0.15Z^{-2}}$ [L1][CO3][12M]

i) Find difference equation ii) Draw cascade & parallel realizations
iii)Calculate impulse response of the system.

8. Realize system with following difference equation [L3][CO3][12M]
y(n) = (3/4) y(n-1) – (1/8) y(n-2) + x(n) + (1/3) x(n-1).
a) Cascade form
b) Parallel form

9. a). Illustrate the realization of the IIR filter in cascade form [L1][CO3][6M]

(b). Explain representation of structures using signal flow graphs. [L2][CO3][6M]

10(a). Explain conversion from lattice structure to direct form. [L2][CO3][6M]

(b). Determine the direct form realization of FIR with system function [L1][CO3][6M]

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

UNIT –IV

Design of IIR Filters

1. (a).The analog transfer function H(s)= 2/(s+1)(s+2) Determine H(z) using impulse invariance method [L1][CO4][7M]

(b) Compare FIR and IIR filters. [L4][CO4][5M]

2. (a) Explain the features of Chebyshev approximation. [L2][CO4][6M]
(b) Discuss the location of poles for Chebyshev filter. [L1][CO4][6M]

3. (a) Discuss the characterization of IIR filter. [L1][CO4][5M]
(b) Given specifications αp=1 dB; αs= 30dB; Ωp= 200rad/sec; Ωs=600 rad/sec. Determine the order the filter. [L2][CO4][7M]

4. (a) Compare features of different windowing functions. [L4][CO4][5M]
(b) Determine the order and the pole of the low pass filter that has a 3-dB attenuation at 500 Hz and an attenuation of 40 dB at 1000 Hz. [L1][CO4][7M]

5. Describe the IIR filter design approximation using Bilinear Transformation method. [L2][CO4][12M]
Also sketch the s-plane to z-plane mapping. State its merits and demerits.

6. Using the bilinear transform, design a high pass filter, monotonic in pass band with cut off frequency of 100Hz and down 10dB at 350 H. the sampling frequency is 5000Hz. [L3][CO4][12M]

7. a). Discuss the frequency selective filters with plot. [L1][CO4][6M]
b). Give the advantages and disadvantages of the digital filters [L1][CO4][6M]

8.Design a Chebyshev filter for the following specifications using [L3][CO4][12M]
a) Bilinear transformation b) Impulse invariant method

$0.8 \leq |H(e^{jw})| \leq 1 \quad 0 \leq w \leq 0.2\pi$

$|H(e^{jw})| \leq 0.2 \quad 0.6\pi \leq w \leq \pi$

9. Describe the frequency transformation in digital filters [L1][CO4][12M]

10. a). Explain the frequency transformation in analog filters [L2][CO4][8M]
b). Distinguish the Butterworth and Chebyshev filters [L4][CO4][4M]

UNIT – V
Design of FIR Filters

1.Design an ideal HPF with desired frequency response $H_d(e^{jw})=$ 1, $\pi/4 \leq|w|\leq\pi$ [L3][CO5][12M]

0, $|w|\leq \pi/4$

Find the values of h(n) for N=11 and also find H(Z) using Hanning window technique.

2. a). Determine the frequency response of the FIR filter defined by y(n)= 0.25x(n)+ x(n-1) + 0.25x(n-2). [L1][CO5][6M]

b). Explain about the Rectangular window of the FIR filter. [L2][CO5][6M]

3. Design a ideal band pass filter with a frequency response $H_d(e^{jw})=1$, $\pi/4 \leq|w|\leq 3\pi/4$

= 0 otherwise

Find the values of h(n)=11 and plot frequency response [L3][CO5][12M]

4 a) Design FIR filter using symmetric filter [L2][CO5][6M]

b) Design a linear phase FIR filter using frequency sampling method. [L1][CO5][6M]

5. Design a filter with $H_d(e^{j\omega})= e^{-j3\omega}$ $-\pi/4\leq\omega\leq\pi/4$ [L3][CO5][12M]

= 0 $\pi/4\leq\omega\leq\pi$

Using Hamming window with N = 7

6. a) Discuss about characteristics linear phase FIR filters [L1][CO5][6M]

b) Compare features of different windowing functions ? [L4][CO5][6M]

7.(a) Discuss about Asymmetric FIR filters. [L1][CO5][5M]

(b) What are the effects of windowing? [L1][CO5][7M]

8. Design a FIR low pass filter satisfying the following specifications $\alpha p\leq 0.1$ dB; $\alpha s\geq 44.0$ dB; ωp= 20rad/sec;ωs=600 rad/sec and ωsf=100rad /sec. [L3][CO5][12M]

9. A band pass FIR filter of length 7 is require. It is to have lower and upper cut off frequencies of 3kHz and is intended to be used with a sampling frequency of 24kHz. Determine the filter coefficients using hamming window. Consider the filter to be causal. [L3][CO5][12M]

10. Illustrates the followings

a) Rectangular window [L1][CO5][4M]

b) Hamming window [L1][CO5][4M]

c) Hanning window [L1][CO5][4M]