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P.E.S. College of Engineering, Mandya - 571 401

(An Autonomous Institution affiliated to VTU, Belagavi)

Fourth Semester, B.E. - Electronics and Communication Engineering

Semester End Examination; May / June - 2018

Digital Signal Processing

Time: 3 hrs *Max. Marks: 100*

***Note**: Answer **FIVE** full questions, selecting **ONE** full question from each unit.*

UNIT - I

1 a. Determine the 4-point DFT of sequence given by,

$$x(n) = \frac{1}{3}; \quad 0 \le n \le 2$$
$$= 0; \quad \text{otherwise}$$

Draw its magnitude and phase spectra — 8

b. Consider a length-12 sequence defined for

$0 \le n \le 11$

$x(n) = \{8, 4, 7, -1, 2, 0, -2, -4, -5, 1, 4, 3\}$

with 12 point DFT x(k); $0 \le n \le 11$ — 6

Evaluate the DFT

$$\sum_{k=0}^{11} e^{-j\frac{4\pi k}{6}} x(k)$$

c. Given that X(k) and Y(k) are the 6-point DFT of sequence x(n) and y(n) respectively

x(k) = {1+j, -2.1+j3.2, -1.2-j2.4, 0, 0.9+j.31, -0.3+j1.1}

The sequence x(n) and y(n) are related by y(n) = $x((n-4))_6$ — 6

Determine Y(k) without computing the DFT.

2 a. Obtain the 4-point DFT of the sequence $x_1(n)$ = {1, 1, 2, 3} and thereby find the 4-point DFT of the sequence $x_2(n)$ = {1, 3, 2, 1} using the appropriate property. — 6

b. Compute the eight point circular convolution of the following sequences :

x(n) = u(n)-u(n-4) — 6

and h(n) = u(n)-u(n-3)

c. Find the output y(n) of a filter whose impure response is h(n) = {1, 1, 1} and input signal is x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}using overlap save method, assuming the length of block as 5. — 8

UNIT - II

3 a. Determine the 8-point DFT the sequence

x (n) $=2^n$; $0 \le n \le 7$ use DIT-FFT algorithm. — 8

b. Compute the four point DFT of the sequence x(n) = {1, 2, 1, 2} using the Goertzed algorithm. — 6

c. Obtain the 4 point IDFT of the sequence X(k) = {10, -2+j2, -2, -2-2j}using DIT-FFT algorithm. — 6

Contd…2

4 a. Determine the DFT of Hamming window

$w(n) = 0.54 - 0.46\cos(2\pi n/(N-1), 0 \le n \le 7$ 8

using the signal flow graph of DIF-FFT.

b. A 512 point DFT of a sequence of length 498 is to be computed.

i) How many Zero valued samples should be appended to x (n) prior to the computation of the DFT radius-2 DIT-FFT algorithms?

ii) What are the total number of complex multiplications and additions needed for the direct evolution of all DFT samples? 6

iii) What are the total number of complex multiplications and addition needed if a radix-2 DIT-FIT algorithm is used?

c. Determine the time domain representation of

X(k) = {7, 1, -j2, -1, 1+j2} using DIF = FFT algorithm. 6

UNIT - III

5a. Design a low-pass FIR filter using rectangular window with cut-off frequency $W_c = \pi/2$ radian. The length of the impulse response should be 11. Also obtain the expression to find the magnitude response. 12

b. If the frequency response of a liner phase FIR filter is given by

$H(e^{iw}) = [0.3+0.5\cos(w)+0.3\cos(2w)]$. Determine the filter coefficients. 4

c. List the advantages and disadvantages of a FIR filter. 4

6a. Design a FIR high pass filter with frequency response

$$H_d(e^{jw}) = \begin{cases} e^{-jwT}; & W_c \le |w| \le \pi \\ 0 \quad ; & 0 \le |w| \le Wc \end{cases}$$ 10

Find the value of h(n) for N = 7 using Hamming window.

b. Design a linear phase FIR filter wire N = 7 are cut-off frequency of $W_c = 0.3\ \pi$ rad using the frequency sampling method. 10

UNIT - IV

7a. Design a digital Butterworth low pass filter with frequency specification given by,

i) Passband $\le$ 3.01 dB ii) Passband edge frequency : 500 Hz

iii) Stop band attenuation $\ge$ 15dB iv) Stop band edge frequency :750 Hz 12

v) Sampling rate f_s = 2 kHz use bilinear transformation method.

b. Using bilinear transformation, convert the low pass filter given by,

$$H_{Lp}(s) = \frac{1}{S^2 + \sqrt{2}\,S + 1}$$ 8

In to a digital high pass filter with pass band edge f_p = 100 Hz and Sampling frequency 1 kHz

8 a. Design a Chebyshev-I low pass filter with ripple of 1 dB in the password $0 \le w \le 0.2\pi$ and attenuation of atleast 15 dB in the stop band $0.3 \le w \le \pi$. Use impulse invariant method. 15

b. Given an analog filter with transfer function

$$Ha(s) = \frac{(s+2)}{(S+1)\,(S+3)}$$ 5

Find the corresponding digital IIR filter H(z) using matched Z-transfer method. Given that Sampling rate is 10Hz.

UNIT - V

9 a. Obtain direct form I, direct from-II, cascade and parallel form realization for the following system Y (n) = 0.75y(n-1) - 0.125y(n-2) + 6x(n) + 7x(n-1) + x(n-2). 14

b. Realize a linear phase filter given by

$h(n) = \left(\frac{1}{2}\right)^n [u(n) - u(n-4)]$ using direct form-I. 6

10 a. Obtain block diagram of the direct form I and direct form II realization for a digital IIR filter described by the system function

$$H(z) = \frac{8Z^3 - 4Z^2 + 11Z - 2}{\left(Z - \frac{1}{4}\right)\left(Z^2 - Z + \frac{1}{2}\right)}$$ 10

b. Explain the ADPCM speech encoding process 5

c. Find the transfer function and difference equation representation for the system in Fig Q.10 5

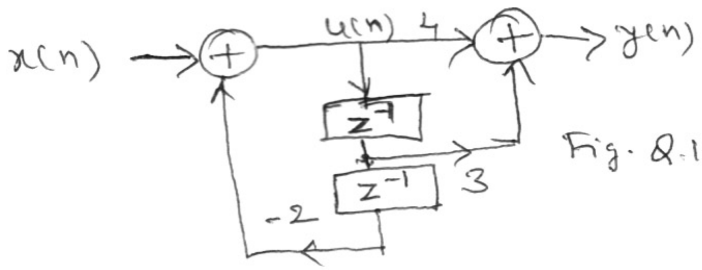

Fig. Q.10

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