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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; June - 2017

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. Find the N-Point DFT of the sequence $x(n)=4+\cos^2\left(\dfrac{2\pi n}{N}\right)$, $n=0,1,\ ....\ N-1.$ 6

b. State and prove frequency shifting property of DFT. 4

c. Determine output of a FIR filter with impulse response

$h(n)=2\delta(n)+2\delta(n-1)+\delta(n-2)$

for an input x(n) given below. Using Overlap-add 8-point fast convolution technique. $x(n)=[3,0,-2,0,2,1,0,-2,-1,0]$. 10

2 a. Find the 10-pint DFT of the sequences 6

$(i)\ x(n)=\delta(n)+\delta(n-5)$ $\qquad (ii)\ x(n)=u(n)-u(n-6)$

b. A signal x_a(t) that is bandwidth to 10 kHz is sampled with a sampling frequency of 20 kHz. The DFT of N = 1000 samples *x(n)* is then computed.

(i) To what analog frequencies does the index k = 150 and 800 correspond to ?

(ii) Determine the spacing between spectral samples. 6

c. If x_1(n) and x_2(n) are N-point sequences with N-point DFT's $x_1(k)$ and $x_2(k)$ respectively show that

$$\sum_{n=0}^{N-1} x_1(n)x_2^*(n)=\frac{1}{N}\sum_{n=0}^{N-1} x_1(k)x_2^*(k)$$

$$If\ \ x_1(n)=x_2(n)=\cos\left(\frac{2\pi nk}{N}\right)\ determine\ \sum_{n=0}^{N-1} x_1(n)x_2^*(n).$$ 8

UNIT - II

3 a. Compute 8-point DFT of the sequence *x(n) = n+1* using decimation-in-freq FFT algorithm. 10

b. Use the 4-point inverse FFT to obtain the time-domain sequence given

$x(k)=[6,-2+j2,-2,-2-j2]$. 5

c. Discuss the situation to justify use of chirp-Z transform. 5

4 a. Develop the flowchart of decimation-in-freq FFT algorithm for a 8-point sequence starting from basic definition of DFT. 10

b. Compute the DFT of the sequence $x(n)=[1,2,3,4,4,3,2,1]$ using 4-pont decimation in-time FFT algorithm only once. 10

UNIT - III

5 a. Design a linear phase FIR high-pass filter using hamming window with a cutoff frequency W_c = 0.8π rad/sample and N = 7. Given the desired frequency response as 10

$H_d(e^{jw}) = e^{-jw\alpha};\ -\pi \le \mathrm{w} \le -w \quad \& \quad w_c \le w \le \pi$

$= 0\ ;\ otherwise$

b. Discuss the steps used in design of FIR filler using frequency sampling method. 5

c. Compare FIR & IIR filters for its various performance parameters. 5

6 a. Determine the coefficients of a linear-phase FIR-filter of length N = 15, which has a symmetric unit sample response and frequency response that satisfies the condition

$$H\left(\frac{2\pi k}{15}\right) = 1;\ \text{for } k = 0,1,2,3$$
$$= 0.4;\ \text{for } k = 4$$
$$= 0;\ \text{for } k = 5,6,7$$
10

b. Discuss the effect of windows on the overall frequency response of FIR filter and list the ideal requirements of a window. 6

c. Discuss the importance of linear phase in filters with examples. 4

UNIT - IV

7 a. An analog filter has a transfer function $H(s) = \dfrac{10}{s^2 + 7s + 10}$.

Design a digital filter equivalent to this using impulse invariant method. 6

b. Discuss the frequency axis mapping from analog to digital domain in impulse invariant method. 4

c. Design a digital LPF using BLT to meet the following specifications.

(i) Monotonic pass band and stop band. (ii) –3dB cutoff at 0.5 π rad/s.

(iii) –15dB attenuation at 0.75 π rad/s. 10

8 a. Derive the transformation function used with Bilinear transfer starting from basics and also comment on preservation of frequency selective characteristics in this transformation. 10

b. Design analog Butterworth filter that has a gain of –2 dB at 20 rad/s and attenuation in excess of 10dB beyond 30rad/sec 10

UNIT - V

9 a. Obtain a parallel realization of the transfer function $H(z) = \dfrac{10z^4 - 3.7z^3 - 1.28z^2 + 0.99z}{\left(z^2 + z + 0.34\right)\left(z^2 + 0.9z + 0.2\right)}$. 8

b. Discuss the ADPCM speech encoder which makes use of digital signal processor with a neat block diagram. 8

c. Realize Direct form-II structure for the system given by $H(z) = \dfrac{1 - z^{-1}}{(1 + 0.5z^{-1} + z^{-2})}$. 4

10 a. An impulse response of a FIR filter is given by $h(n) = [1, 2, 3, 4, 3, 2, 1]$, check whether the system has linear phase or not. If it has linear phase then obtain a realization using minimum numbers of multiplications. 8

b. Differentiate between speech analysis and synthesis. Discuss the various process involved in speech synthesis from text. 6

c. Realize the following transfer function in cascade form $H(z) = \dfrac{216z^3 + 96z^2 + 24z}{(2z + 1)(12z^2 + 7z + 1)}$. 6

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