I D NO.

PRESIDENCY UNIVERSITY, BENGALURU

SCHOOL OF ENGINEERING

Weightage: 40 % Max Marks: 80 Max Time: 02 hrs. 09 May Wednesday 2018

ENDTERM FINAL EXAMINATION MAY 2018

Even Semester 2017-18 Course: **ECE/EEE 213** IV,VI Sem.ECE/EEE

DIGITAL SIGNAL PROCESSING

Instructions:

(i) Read the question properly and answer accordingly.

(ii) Question paper consists of 3 parts.

(iii) Scientific and Non-programmable calculators are permitted.

(iv) Butterworth denominator polynomial table is permitted.

Part A

(4 Q x 5 M= 20 Marks)

1. a) Name any two methods for digitizing the transfer function of an analog IIR filter. (1M)

   b) Give the difference between Chebyshev and Butterworth IIR filter . (4M)

2. y(n)= -0.1y(n-1)+0.72y(n-2)+0.7x(n)-0.25x(n-2),

   Determine H(Z) and implement the same using Direct Form-I and Direct Form-II.

3. The Difference Equation of a filter is given by

   $y(n)=x(n)+\frac{2}{5}x(n-1)+\frac{3}{4}x(n-2)+\frac{1}{3}x(n-3)$

   Determine whether it is IIR or FIR filter, and realize the same with any suitable structure.

4. For the digital filter , $H(Z)=\frac{0.041\left(1+Z^{-1}\right)^2}{1-1.4418Z^{-1}+0.6743Z^{-2}}$

   ω_p =0.2π rad and ω_s=0.5π rad. Give the relationship between digital and analog frequencies in Impulse invariance and Bilinear Transformation technique and find the analog frequencies.

Part B

(3 Q x 10 M= 30 Marks)

5. Transform $H(s)=\frac{1}{(S^2+S+1)}$, normalize butterworth LPF to a digital filter using Impulse Invariance transformation technique. The system uses sampling rate of 1Hz.

6. (a) Apply Bilinear Transformation to $H(s)=\frac{2}{(S+1)(S+2)}$, with T=2sec and find H(z). (5M)

   (b)Differentiate Bilinear and Impulse invariance technique. Write short notes on Warping effect and how it can be compensated. (5M).

7. A FIR filter (low pass) is to be designed with a desired frequency response,

$$H_d(e^{j\omega}) = e^{-j3\omega} \ , \frac{-3\pi}{4} \le \omega \le \frac{3\pi}{4}$$

$$= 0 \quad , \frac{3\pi}{4} \le \omega \le \pi$$

   Determine the filter coefficients $h_d(n)$ and h(n). Use Rectangular window with N=5.Also obtain the frequency response.

Part C

(2Q x 15 M= 30 Marks)

8. Design a Low pass Digital filter using Bilinear transformation for the following specifications.
   (a) Monotonic passband and stopband.
   (b) -2 dB passband attenuation at a frequency of 20 rad/sec and atleast -10 dB stopband attenuation at 30 rad/sec.
9. Design a Chebyshev lowpass filter (i.e find H(s)) with the following specifications
   (a) Passband ripple of 1.5 dB ,Ω_p=2 rad/sec.
   (b) Stopband attenuation of 10 dB , Ω_s=30 rad/sec.

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Weightage: 20% Max Marks: 40 Max Time: 1 hr. 27 March Tuesday 2018

TEST – 2 **SET A**

Even Semester 2017-18 Course: **ECE/EEE 213 Digital Signal Processing** IV/VI Sem. **ECE/EEE**

Instruction:

(i) Read the question properly and answer accordingly.
(ii) Question paper consists of 3 parts.
(iii) Scientific and Non-programmable calculators are permitted
(iv) Butterworth Polynomial Tables are permitted

Part A

(4 Q x 4 M = 16 Marks)

1. With respect to filters, Define HPF, BPF, Pass Band, decibel gain Kp with graphs if required.
2. Write equations for N and Ωc for Butterworth filter.
3. Draw both Ideal characteristics and Practical characteristics for **LPF** (graphs of dB gain versus frequency) and indicate Ωp, Ωs, Ωc, Kp, and Ks on the practical characteristics.
4. Write formulae for Number of complex multiplications for (a) direct computation of DFT (b) With Decimation in time FFT (c) Calculate both values for N=8

Part B

(2 Q x 7 M = 14 Marks)

5. For x(n) = [15,16,17,18] draw Decimation in Time FFT for length 4 and find DFT. Again Find IDFT using same method and verify that the result is same as the original sequence.
6. Draw Decimation in Frequency FFT algorithm for both DFT and IDFT for length 8 and find DFT for x(n) =[1,1,1,1, 1,1,1,1] using this.

Part C

(1Q x 10 M = 10 Marks)

7. Clearly write all the formulae and **then** design a Butterworth LPF. **Given** Ωp = 4rad/sec; Ωs = 8 rad/sec, Kp = -1 dB, Ks = -20dB. (Use of Butterworth Table permitted).

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GAIN MORE KNOWLEDGE
REACH GREATER HEIGHTS

Weightage: 20 % Max Marks: 20 Max Time: 1 hr. 22 Feb Thursday 2018

TEST – 1

Even Semester 2017-18 Course: **ECE 213 Digital Signal Processing** IV Sem. **ECE/EEE**

Instruction:

(i) Read the question properly and answer accordingly.
(ii) Question paper consists of 3 parts.
(iii) Scientific and Non-programmable calculators are permitted

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Part A

(3 Q x 2 M = 6 Marks)

1. Define Nyquist Rate. If x(t) = 10 [Sin(2π50t) + Cos(2π100t) + Cos(2π200t)] then find

   a. $f_{sig\ max}$ b. Nyquist rate of sampling

2. State and prove circular convolution property
3. For x(n) = [10,20,30,40,50,60] then find the signals

   a. $x((n-2))_6$ b. $x((n))_8$ c. $x((n-2))_8$ d. $x((n-100))_8$

Part B

(2 Q x 3 M = 6 Marks)

4. Given x1(n) = [1,2,3,4] and x2(n) = [10,20,30,40] find x1 4 x2. Also **calculate** minimum length to be selected for circular convolution to give linear convolution result?
5. x(n) = Sin (2π n / N) using length as 4, write the actual values of x(n) and then find DFT

Part C

(2Q x 4 M = 8 Marks)

6. If x(n) = [1,2,3,4,5,6] and h(n) = [10,11,12] find length and linear convolution by any convenient method. Again find linear convolution using overlap add method using 2 chips only
7. State Parseval's theorem. If x(n) = [1,2,3,4,5,6,7,8] then find

   (a) X(k=0) (b) X(k=4) (c) $\sum_{(k = 0 \text{ to } N-1)} X(k)$ (d) $\sum_{(k = 0 \text{ to } N-1)} | X(k) |^2$