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Question Paper Code: 46401

B.E. / B.Tech. DEGREE EXAMINATION, NOV 2018

Sixth Semester

Electronics and Communication Engineering

14UEC601 - DIGITAL SIGNAL PROCESSING

(Regulation 2014)

Duration: Three hours Maximum: 100 Marks

Answer ALL Questions

PART A - (10 x 1 = 10 Marks)

1. The periodic convolution is

   (a) linear convolution (b) circular convolution

   (c) fast convolution (d) slow convolution

2. How many additions are required to compute N point DFT using radix 2 FFT?

   (a) $\frac{N}{2}\log_2 N$ (b) $N \log_2 N$ (c) $\log_2 N$ (d) N/2

3. What is the order of the normalized low pass Butterworth filter used to design an analog band pass filter with -3.0103dB upper and lower cut-off frequency of 50Hz and 20KHz and a stop band attenuation 20dB at 20Hz and 45KHz?

   (a) 2 (b) 3 (c) 4 (d) 5

4. If N_B and N_C are the orders of the Butterworth and Chebyshev filters respectively to meet the same frequency specifications, then which of the following relation is true?

   (a) $N_C=N_B$ (b) $N_C<N_B$ (c) $N_C>N_B$ (d) Cannot be determined

5. Which region of the frequency specification has to be optimized to reduce side lobes of the FIR filter?

   (a) Stop band (b) Pass band
   (c) Transition band (d) None of these

6. The values of cutoff frequencies in general depend on

   (a) Type of the window (b) Length of the window
   (c) Neither (a) nor (b) (d) Both (a) and (b)

7. Sign magnitude representation of -7/8 is

   (a) 1.001 (b) 1.111 (c) 1.100 (d) 0.111

8. Which of the following is not a quantization error occuring in digital systems?

   (a) Input quantization error (b) Product quantization error
   (c) Coefficient quantization error (d) Output quantization error

9. Which of the following is the disadvantage of sampling rate conversion by converting the signal into analog signal?

   (a) Signal distortion
   (b) Quantization effects
   (c) New sampling rate can be arbitrarily selected
   (d) Both (a) and (b)

10. In subband coding, the input signal is first split into number of non-overlapping frequency by

   (a) Low pass filter (b) High pass filter
   (c) Band pass filter (d) Band stop filter

PART - B (5 x 2 = 10 Marks)

11. What is Zero padding? What is the purpose of it?

12. What is pre-warping?

13. Write the equation of Hamming and Blackman window functions.

14. Define zero input limit cycle oscillations

15. Give the steps in multistage sampling rate converter design.

PART - C (5 x 16 = 80 Marks)

16.(a) Perform circular convolution of the following sequence. X(n)={-1,1,2,-1,1,2 } and

h(n)={2,1,-2}. (16)

Or

(b) Compute the linear convolution of finite duration sequences h(n)={1, 2} and x(n)={1, 2, -1, 2, 3, -2, -3, -1, 1, 1, 2, -1} by overlap add method. (16)

17. (a) Design a digital chebyshev filter that satisfying the following frequency response

$$0.707 \leq |H(e^{j\omega})| \leq 1 \qquad for \quad 0 \leq \omega \leq \frac{\pi}{2}$$

$$|H(e^{j\omega})| \leq 0.2 \qquad for \quad \frac{3\pi}{4} \leq \omega \leq \pi$$

with T=1 sec using impulse Invariant Transformation technique (16)

Or

(b) Design a Digital Butterworth high pass filter with a minimum passband attenuation of 2.5dB at Ωp=20rad/sec and the stop band attenuation of 30 dB at Ωs=50rad/sec use bilinear transformation with the sampling time of 1 sec. (16)

18. (a) Design a FIR Linear phase, Digital filter approximating the ideal high-pass filter

with a frequency response $H_d(e^{j\omega}) = \begin{cases} 1 & for\ \frac{\pi}{4} \leq |\omega| \leq \pi \\ 0 & |\omega| < \frac{\pi}{4} \end{cases}$

(i) Determine the co-efficient of 11 tap filter based on the window method Hanning.

(ii) Determine and plot the magnitude and phase response of the filter. (16)

Or

(b) Using a rectangular window technique, design a low pass filter with a pass band gain of unity, cut-off frequency of 1000 *Hz* and working at a sampling frequency of 5 *KHz*. The length of the impulse response should be 7. (16)

19. (a) A non-recursive system H (z) is designed such a way that, two Linear phase systems $H_1(z)$ and $H_2(z)$ are connected in cascade. Which are given as $H_1(z) = \frac{1}{1-a_1 z^{-1}}$ and $H_2(z) = \frac{1}{1-a_2 z^{-1}}$. Find the output round off noise power? Assume a_1 =0.5 and a_2 =0.6. (16)

Or

(b) Analyze the characteristics of a limit cycle oscillation with respect to the first order recursive system described by the difference equation y (n) = 0.95y (n-1) + x (n). Determine the dead band of the filter. Assume 4-bit sign magnitude representation (excluding sign bit) and the input is given by, (16)

$$x(n) = 0.875 \quad \text{for } n = 0$$
$$= 0 \quad \text{otherwise}$$

20. (a) Implement a two stage decimator for the following specifications:
Sampling rate of the input signal 10 *kHz*, M=100, Pass band= 0 to 50 *Hz*, Pass band ripple = 0.1 and Stop band ripple = 0.001. (16)

Or

(b) (i) Explain the multistage implementation of sampling rate conversion with a block diagram. (8)

(ii) A signal x(n) is given by x(n) = {0, 1, 2, 3, 4, 5, 6, 0, 1, 2, 3…}. Obtain the decimated signal with a factor of 2 and the interpolated signal with a factor of 2. (8)