M.E. First Semester (Electrical & Elect.) (New-CGS)

13283 : Advanced Digital Signal Processing : 1 EEEME 3

P. Pages : 2 | AU - 3398

Time : Three Hours | Max. Marks : 80

Notes :
1. Due credit will be given to neatness and adequate dimensions.
2. Assume suitable data wherever necessary.
3. Illustrate your answer necessary with the help of neat sketches.
4. Use of pen Black ink/refill only for writing the answer book.

1. a) State and prove the following properties of DFT. **8**
   i) Time reversal
   ii) Circular time shift
   iii) Periodicity

   b) Show that - **5**
   i) $\delta(n) = u(n) - u(n-1)$
   ii) $u(n) = \sum_{k=-\infty}^{\infty} \delta(k) = \sum_{k=0}^{\infty} \delta(n-k)$

OR

2. a) Show that the energy (power) of a real valued energy (power) signal is equal to the sum of the energies (powers) of its even and odd components. **6**

   b) Determine the eight point DFT of the signal $x(n) = \{1, 1, 1, 1, 1, 1, 0, 0\}$ and sketch its magnitude and phase. **7**

3. a) Determine the variance of the round-off noise at the output of the cascade realization of the filter with system function $H(z) = H_1(z) \cdot H_2(z)$ where, **7**

$$H_1(z) = \frac{1}{1-\frac{1}{2}z^{-1}} \text{ and } H_2(z) = \frac{1}{1-\frac{1}{4}z^{-1}}$$

   b) Determine the cascade and parallel realization of the system described by **6**

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

OR

4. a) Explain the following in brief related to round-off effects in digital filters. **7**
   i) Limit cycle oscillations
   ii) Scaling to prevent overflow.

   b) Determine the parameter $\{k_m\}$ of the lattice filter corresponding to the FIR filter described by the system function : $H(z) = A_2(z) = 1 + 2z^{-1} + z^{-2}$ **6**

5. a) Design a single - pole LPF (digital) with a 3-dB bandwidth of 0.2 π, using BLT applied to analog filter $H(S) = \frac{\Omega_C}{S+\Omega_C}$ where Ω_C is the 3-dB bandwidth of the analog filter. 6

b) Explain the frequency sampling method of designing FIR filter with suitable example. 8

OR

6. a) Explain how stable analog filter can be converted into its equivalent stable digital filter. 7

b) Determine the order and the poles of a lowpass butterworth filter that has a -3dB bandwidth of 500 Hz and an attenuation of 40 dB at 100 Hz. 7

7. a) Explain in detail Levinson - Durbin algorithm. 9

b) Determine the power spectra for the random processes generated by the following difference equation $x(n) = -0.81x(n-2) + \omega(n) - \omega(n-1)$
Also sketch the spectra and determine the autocorrelation $\gamma_{xx}(m)$. 5

OR

8. a) Obtain the relationship between the Autocorrelation and the model parameters $\{a_k\}$ & $\{b_k\}$ for AR, ARMA and MA processes. 6

b) Explain in detail Welch method of averaging the modified periodograms. 8

9. a) Explain polyphase decomposition process. 6

b) Explain in detail LMS adaptive filtering algorithm. 7

OR

10. a) Explain in detail Quadrature Mirror filter bank. 7

b) Explain in detail FIR Wiener filter. 6

11. a) Explain in detail how digital signal processor can be used for digital filtering. Give suitable example. 7

b) Explain in detail how special purpose DSP can be used for FFT implementation. 6

OR

12. a) Explain in detail any one architecture of TMS320C54XX/TMS320C67XX series DSP processor. 7

b) Explain the various issues involved in selection of DSP processor. 6

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