Reg. No. :

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M.E./M.Tech. DEGREE EXAMINATIONS, NOVEMBER/DECEMBER 2022.

First Semester

Applied Electronics

AP 4151 – ADVANCED DIGITAL SIGNAL PROCESSING

(Common to: Electronics and Communication Engineering)

(Regulations 2021)

Time : Three hours | Maximum : 100 marks

Answer ALL questions.

PART A — (10 × 2 = 20 marks)

1. Write the equation for finding DCT of a signal $x(t)$ and mention any two applications of DCT.

2. If $u(t) = 5\cos(2000\pi t)$ and $v(t) = 2\cos(5000\pi t)$, determine the Nyquist sampling rate for the signal $y(t) = u(t) \times v(t)$.

3. Differentiate between FIR and IIR filters.

4. Write the equations used to generate the coefficients of Hamming window and Hanning window.

5. What is the basic principle of Welch method to estimate power spectrum?

6. Differentiate between Parametric and Non-Parametric methods of power spectrum estimation.

7. State any two Nobel identities used in multirate signal processing.

8. For the system show in the Figure.1, determine $y[n]$ if $x[n] = \{1, 2, 3, 4, 5, 6\}$.

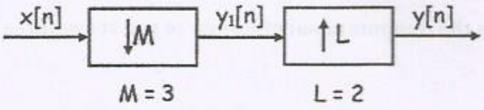

Figure.1

9. What is phase shifter? Give its applications.

10. Give the working principle of over sampling A/D converter.

PART B — (5 × 13 = 65 marks)

11. (a) Obtain the 8 point DFT of the following sequence using redix-2 DIF-FFT algorithm. Show all the results along the signal flow graph. $x(n) = \{2,1,2,1,1,2,1,2\}$.

Or

(b) State and prove Nyquist Sampling Theorem with suitable illustrations.

12. (a) Describe the basics of linear prediction filters. Derive the recursive predictor coefficients for optimum Lattice predictor by Levinson-Durbin algorithm.

Or

(b) Explain with neat sketches the implementation of FIR filters in direct form and Lattice form. Design a digital FIR band pass filter with lower cut off frequency 2000 Hz and upper cutoff frequency 3200 Hz using hamming window of length N = 7. Sampling rate is 10000 Hz.

13. (a) Derive the Yule-Walker equation for ARMA, AR and MA model in detail.

Or

(b) Explain the Bartlett and Blackman-Tukey methods of power spectrum estimation and compare their performance.

14. (a) Implement a decimator $x[3n]$ given the input $x[n] = \{1,2,1,2,1,2,1,2,1\}$ and anti-aliasing filter coefficients $h[n] = \{1,1,1\}$ using

(i) The original structure with an anti-aliasing filter followed by downsampler

(ii) An efficient structure

(iii) Polyphase filters

Compare the computational efficiency of the above three structures

Or

(b) Implement a rational sampling rate convertor to change the sampling rate by the rational sampling factor 5/3 given the input $x[n] = [1, 2, 3, 4, 5]$, anti-imaging filter coefficients $g[n] = [1, 2, 3, 4]$ and anti-aliasing filter coefficients $h[n] = \{1, 3\}$ using polyphase filters. Compare the computational efficiency with that of direct implementation.

15. (a) Illustrate the application of multirate signal processing in interfacing of Digital systems with different sampling rates with suitable examples.

Or

(b) (i) How do you design a phase shifter? Explain with an example. (7)

(ii) How do you use subband coding in speech processing? (6)

PART C — (1 × 15 = 15 marks)

16. (a) Design a two stage decimator (16 × 4) for high quality data acquisition system with the following overall specification for the decimation filter:

Audio band – 0 to 20 kHz

Input sampling frequency – 3.072 MHz

Output sampling frequency – 48 kHz

Passband ripple < 0.001 dB

Stopband attenuation >80 dB

Also specify the computations and storage requirements.

Or

(b) (i) Show that two channel QMF filter banks will exhibit complimentary frequency response, and derive the Perfect Reconstruction (PR) conditions. (7.5)

(ii) Prove that QMF bank is a perfect reconstruction system if the filters are chosen to be (7.5)

$H_0(z) = 4z^{-2}, H_1(z) = z^{-1}, G_0(z) = 0.5z^{-1}$ and $G_1(z) = z^{-2}$