CBCS Scheme

USN | | | | | | | | | |

16/17ECS/EIE/ELD21

Second Semester M.Tech. Degree Examination, June/July 2018
Advanced DSP

Time: 3 hrs. Max. Marks: 80

Note: Answer any FIVE full questions, choosing one full question from each module.

Module-1

1 a. Define: (i) Down sampling, (ii) Up sampling. With neat diagrams, explain down sampling and up sampling effects in the frequency domain. **(08 Marks)**

b. Consider a signal x[n] sampled at a frequency F_S = 10 kHz. Determine and draw the spectrum of the output when x[n] is re-sampled at a new sampling frequency of (i) F_y = 22 kHz (ii) F_y = 8 kHz. **(04 Marks)**

c. For each of the systems shown in Fig.Q1(c), show a more efficient realization in terms of polyphase decompositions given that $H(z)=1+z^{-1}+2z^{-2}-z^{-3}+z^{-4}-z^{-5}+z^{-6}$.

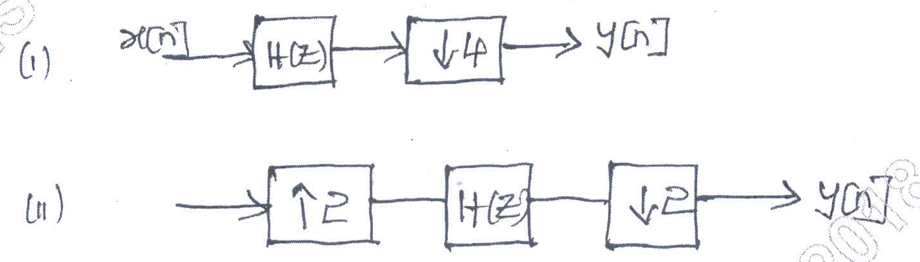

Fig.Q1(c) **(04 Marks)**

OR

2 a. With a neat block diagram and equations, explain two channel QMF bank. Explain how it elements aliasing. **(10 Marks)**

b. Suppose the polyphase matrix for a three channel perfect reconstruction FIR QMF bank is

$$P(z^3)=\begin{bmatrix}1 & 1 & 2\\ 2 & 3 & 1\\ 1 & 2 & 1\end{bmatrix}$$

Draw the analysis and synthesis filters in the QMF bank. **(06 Marks)**

Module-2

3 a. Define random process. Explain: (i) Ergodic process, (ii) Autocorrelation function and (iii) Power density spectrum of a random process. **(08 Marks)**

b. Show that a stationary random process x(n) with mean E[x(n)] = μ_x is a mean ergodic process. Derive the corresponding sufficient conditions. **(08 Marks)**

OR

4 a. What is forward prediction? Derive an expression for the Minimum Mean Square Error (MMSE) of a forward prediction process. **(08 Marks)**

b. Using the Levinson-Durbin algorithm, obtain the solution for the normal equations for the prediction coefficients and also MMSE ε_p^f. **(08 Marks)**

Module-3

5 a. With a neat block diagram, explain the application of an adaptive filter to adaptive channel equalization. Also indicate the necessary equations. **(08 Marks)**

b. With the help necessary block diagrams and relevant equations, explain the linear predictive coding of speech signals. **(08 Marks)**

OR

6 a. Explain the LMS algorithm based on the minimum mean squared error criterion. **(08 Marks)**

b. Explain the RLS algorithm. Mention its properties and advantages over LMS algorithm. **(08 Marks)**

Module-4

7 a. What is spectral estimation? What are the basic methods available? Briefly explain its classification. **(08 Marks)**

b. Explain the Bartlett method for reducing the variance in the periodograms and hence obtain an expression for the variance of the Bartlett power spectrum estimate. **(08 Marks)**

OR

8 a. Explain the ARMA model for power spectrum estimation. **(08 Marks)**

b. An AR(2) process is described by the difference equation $x[n] = 0.81x[n-2] + w[n]$ where $w[n]$ is a white noise process with variance σ_w^2. Determine the parameters of MA(2), MA(4) and MA(8) models which provide the minimum mean squared error fit to the data $x[n]$. **(08 Marks)**

Module-5

9 a. What is wavelet? Explain wavelet transform in brief. **(04 Marks)**

b. Define continuous wavelet transform. Explain in detail, the steps involved in the computation of CWT. **(08 Marks)**

c. Define short time Fourier transform (STFT). Explain how it overcomes the limitations of the Fourier transform. **(04 Marks)**

OR

10 a. What is discrete wavelet transform? Explain the Haar wavelet function and scaled Haar wavelet functions. **(08 Marks)**

b. Write a note on Daubechies wavelet transforms. **(04 Marks)**

c. List out different applications of wavelet transforms. **(04 Marks)**

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