USN | | | | | | | | | | 14ELD21

Second Semester M.Tech. Degree Examination, Dec.2015/Jan.2016
Modern DSP

Time: 3 hrs. Max. Marks:100

Note: ***Answer any FIVE full questions.***

1 a. Derive an expression for SQNR of sinusoidal signals. **(08 Marks)**

b. Consider the following analog sinusoidal signal: $X_\alpha(t) = 3\sin(100\pi t)$.

i) Sketch the signal $X_a(t)$ for $0 \le t \le 30$ ms.

ii) The signal $X_a(t)$ is sampled at Fs = 300 samples/sec. determine the frequency of the discrete-time signal $x(n) = X_a(nT)$. T = 1/Fs and show that it is periodic.

iii) Compute the sample values in one period of x(n). Sketch x(n) and mention the period of discrete time signal in MS.

iv) Find a sampling rate F, such that the signal x(n) reaches its peak value of 3. What is the minimum Fs, suitable for this task? **(12 Marks)**

2 a. A digital communication link carries binary-coded words representing samples of an input signal $X_a(t) = 3\cos 600t - 2\cos/800t$. The link is operated at 10,000 bits/s and each input sample is quantized into 1024 different voltage levels.

i) What is the sampling frequency and folding frequency?

ii) What is the Nyquist rate for the signal $X_a(t)$?

iii) What are the frequencies in the resulting discrete-time signal x[n]?

iv) What is the resolution Δ? **(10 Marks)**

b. State and prove the following properties of DFT:

i) Circular time-shift property.

ii) Symmetry of real-valued sequences. **(10 Marks)**

3 a. Consider a FIR filter with impulse response h(n) = {3, 2, 1, 1}. If input x(n) sequence is x(n) = {1, 2, 3, 3, 2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1}, find the output using overlap add method, assuming a block length of 7. **(10 Marks)**

b. i) Give the differences between FIR and IIR filters. **(04 Marks)**

ii) Give the conditions for physically realizable FIR filters. Explain Paley-Wiener theorem. **(06 Marks)**

4 a. The desired frequency response of an LPF is given by

$$H_d(e^{j\omega}) = \begin{cases} e^{-j3\omega} & |\omega| < 3\pi/4 \\ 0 & 3\pi/4 < |\omega| < \pi \end{cases}.$$

Determine the frequency response of an FIR filter if hamming window is used with M = 7. **(10 Marks)**

b. Design a 17 tap linear-phase FIR filter with a cut-off frequency $\omega_c = \pi/2$. The design is to be done using frequency sampling technique. **(10 Marks)**

5 a. Design a single-pole low pass digital filter with a 3-dB bandwidth of 0.2π, using the bilinear transformation applied to the analog filter

$$H(s) = \frac{\Omega_c}{s + \Omega_c}$$

where Ωc is the 3dB bandwidth of the analog filter. **(10 Marks)**

b. Define up sampling and down sampling with the help of an example. Derive an expression for the spectrum of decimation process down sampled by an integer factor D. **(10 Marks)**

6 a. With a neat block diagram, explain the application of multirate DSP in subband coding of speech signals. **(08 Marks)**
b. Explain the analysis and synthesis structure of UDFT filter bank with efficient realization structure. **(06 Marks)**
c. Explain the polyphase decomposition of a linear filter for down sampling and up sampling. **(06 Marks)**

7 a. Explain two channel quadrature minor filter bank and alias elimination method. Also, explain the perfect reconstruction of a 2 channel QMF bank. **(10 Marks)**
b. Explain the application of adaptive filtering by channel equalization technique. **(10 Marks)**

8 a. Explain the LMS algorithm based on the minimum mean squared error criterion. **(10 Marks)**
b. Explain the RLS algorithms and mention their properties and advantages over LMS algorithm. **(10 Marks)**

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