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14ELD21

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

Time: 3 hrs. Max. Marks:100

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

1 a. Classify the following signals according to whether they are :
   i) One–or multidimensional
   ii) single or multichannel
   iii) continuous time or discrete time
   iv) analog or digital (in amplitude).
      i) A color movie
      ii) Position of the stearing wheel of a car in motion relative to ground reference frame
      iii) Weight and height measurements of child taken every month. **(06 Marks)**

b. Determine whether or not the following signals are periodic. If periodic, determine the fundamental period.
   i) $x(n) = ne^{j\pi n}$ ii) $x(n) = 2\exp(j(n/6 - \pi))$ iii) $x_a(t) = 3\cos(5t + \pi/6)$. **(06 Marks)**

c. For the signal $s(t) = \cos(2\pi t)$, plot its waveform, and then illustrate the resulting samples with the following sampling intervals : i) $T_S = 0.5$sec ii) $T_S = 1$ sec iii) $T_S = 0.75$ sec. For each case also sketch the reconstructed continuous time signal from the samples using linear interpolation. Based on your sketch determine the fundamental frequency of the reconstructed signal. In which case you had aliasing distortion? What is the minimal sampling frequency and the corresponding sampling interval needed to avoid aliasing? **(08 Marks)**

2 a. An analog signal $x_a(t) = \sin(480\pi t) + 3\sin(720\pi t)$ is sampled 600 times per second.
   i) Determine the Nyquest sampling rate for $x_a(t)$
   ii) Determine the folding frequency
   iii) What are the frequencies, in radians, in the resulting discrete time signal x(n)?
   iv) If x(n) is passed through an ideal D/A converter, what is het reconstructed signal $y_a(t)$? **(06 Marks)**

b. A compensated quantize has an input range from – 8V to + 8V. Assuming the signal amplitudes are distributed uniformly over any quantization interval what is the minimum number of bits required to guarantee that the maximum quantization error is less than 0.01V? What are the fewest bits required to guarantee that the root mean squared quantization error is less than 0.01V? **(04 Marks)**

c. The first five points of the eight point DFT of a real value sequence are – {0.25, 0.125–j0.3018, 0, 0.125–j 0.0518, 0}. Determine the remaining three points. **(04 Marks)**

d. Let x(n) and y(n) be two three point sequence :

$$x(n) = \begin{cases} 1 & \text{for} \quad n = 0 \\ 2 & \text{for} \quad n = 1; \\ 1 & \text{for} \quad n = 2 \end{cases} \qquad y(n) = \begin{cases} -1 & \text{for} \quad n = 0 \\ 2 & \text{for} \quad n = 1 \\ 1 & \text{for} \quad n = 2 \end{cases}$$

Compute the 3 – point circular convolution z(n) between x(n) and y(n). **(06 Marks)**

3 a. Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = h(N-1-n)$. Also discuss the symmetric and antisymmetric cases of FIR filter when N is even. **(10 Marks)**

b. Design on FIR linear phase, digital filter approximating the ideal frequency response :

$$H_d(w) = \begin{cases} 1 & \text{for} \quad |w| \le \pi/6 \\ 0 & \text{for} \quad \pi/6 < |w| \le \pi \end{cases}$$

i) Determine the coefficients of a 25 tap filter based on the window method with a rectangular window

ii) Determine and plot the magnitude and phase response of the filter. **(10 Marks)**

4 a. Explain both advantages and disadvantages of FIR filters over IIR filters. **(04 Marks)**

b. Consider a Butterworth analog prototype $H_L(s) = \dfrac{1}{1+\frac{s}{\Omega_c}}$ with $\Omega_C = 1$. Find as IIR discrete time filter H(z)with cut off frequency $w_C = 0.3\pi$ via bilinear translation. **(06 Marks)**

c. Design a digital lowpass Butterworth filter where transfer function is given by:

$$0.7 \le |H(e^{jw})| \le 1; \quad 0 \le w \le 0.2\pi$$
$$0 \le |H(e^{jw})| \le 0.3; \quad 0.6\pi \le w \le \pi$$

Use impulse invariance method. **(10 Marks)**

5 a. Derive an expression for s to z phase transformation based on bilinear transformation method. Also explain what is frequency warping and pre warping? **(10 Marks)**

b. Determine the order and the poles of a Type-I lowpass Chebyshev filter that has a 1 – dB ripple in the passband, a cutoff frequency $\Omega_P = 1000\pi$, a stopband frequency of $2000\pi$and attenuation of 40dB or more for $\Omega \ge \Omega_S$. **(10 Marks)**

6 a. Explain sampling rate convention by a factor : i) D and ii) I. **(10 Marks)**

b. Consider the two different ways of cascading a decimator with a interpolator shown in Fig.Q6(b).

i) If D = I show that the output of the two configurations are different. Hence in general the two systems are not identical.

ii) Show that the two systems on identical if and only if D and I are relatively prime. **(10 Marks)**

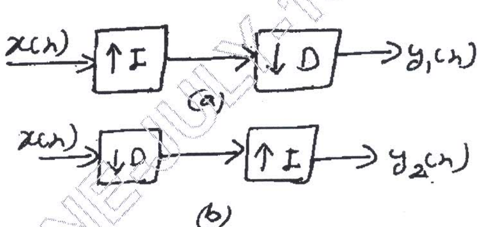

Fig.Q6(b)

7 a. Plot the signals and their corresponding spectra for rational sampling rate conversion by i) I/D = 5/3 dn ii) I/D = 3/5. Assume that the spectrum of the input signal x(n) occupies the entire range $-\pi \le w_x \le \pi$. **(10 Marks)**

b. Explain two channel quandrature minor filter bank in detail. **(10 Marks)**

8 a. Explain adaptive noise cancelation with a neat diagram. **(10 Marks)**

b. Explain the RLS algorithm and mention their properties and advantages over LMS algorithm. **(10 Marks)**

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