APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIRST SEMESTER M.TECH DEGREE EXAMINATION, DECEMBER 2015

Electronics and Communication Engineering

(Applied Electronics And Instrumentation)

01EC6105 Advanced Digital Signal Processing

Max. Marks : 60 Duration: 3 hrs

Answer any two questions from each part

PART A

1. a) A discrete time low pass filter is to be designed with the following specifications.
   Maximally flat pass band and stop band
   Passband ripple : $-3\text{ dB} \le |H(e^{j\omega})| \le 0\text{dB}$, $|\omega| \le 0.15\pi$
   Stop band attenuation : $|H(e^{j\omega})| \le -20\text{dB}$, $|\omega| \le 0.35\pi$
   i) Clearly show the design steps, using impulse invariance method (4)
   ii) Show the location of poles in s-plane (1)
   ii) Give the expression for H(z) (2)

   b) Explain how can you convert the above filter to a high pass filter with pass band edge frequency 0.25π (2)

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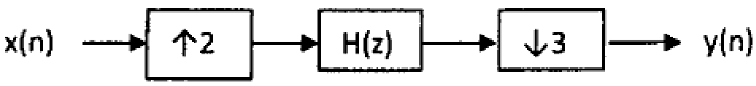
2. a) For the following multirate system

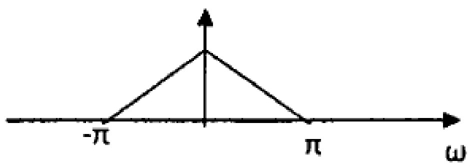
   i) Express y(n) in terms of x(n) (2)
   ii) Express $Y(e^{j\omega})$ in terms of $X(e^{j\omega})$ (2)
   iii) If $x(n)=a^n u(n)$. Find y(n) for $-3 \le n \le 4$ (a=0.5) (1)

   b) For the sampling rate converter shown sketch $Y(e^{j\omega})$



   Given $H(e^{j\omega}) = 2$, $0 \le |\omega| \le \pi/3$

   $X(e^{j\omega})$ =



(4)

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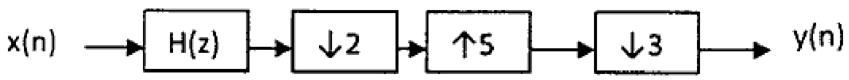
3. a) Given that the ideal frequency response of a low pass filter as
$H_d(e^{j\omega}) = 2e^{-j4\omega}$, $|\omega| \le \pi/4$

i) Find the impulse response to get this response (3)

ii) Find the size of the Hanning window to design a linear phase FIR filter with this response (3)

b)Simplify the multirate system shown below and develop an expression for y(n) in terms of x(n). (Use Vaidyanathan identities wherever applicable). Given $H(z) = z^{-6}$



(3)

PART B

4. a) A decimator with M=3 is implemented with a FIR filter of length 12 .
How can we reduce computational requirement using polyphase representation? Compare the computational requirements of direct implementation and polyphase implementation. (4)

b) A 4 channel analysis uniform DFT filter bank has a set of filter transfer functions $H_k(z)$, k=0,1,2,3 . H0(z) has polyphase components given as

$E_0(z)=1+3z^{-1}-0.8z^{-2}$ $E_1(z)=2-1.5z^{-1}-3.1z^{-2}$

$E_2(z)=4-0.9z^{-1}+2.3^{-2}$ $E_3(z)=1+3.7z^{-1}+1.7z^{-2}$

i) Sketch the analysis section of the filter bank (2)

ii) Determine $H_0(z)$, $H_1(z)$, $H_2(z)$, $H_3(z)$ (3)

5. a) Discuss how STFT overcome Heisenberg Uncertainity problem in time-frequency analysis. Give illustrations of the windows used in finding $X_r(K)$ (3)

b) Consider the filter response in one channel of analysis section $H_0(z) = \frac{1+z^{-1}}{2}$

i) Design a perfect reconstruction 2 channel QMF filter bank (2)
(Find $H_1(z)$, $G_0(z)$,$G_1(z)$)

ii) Sketch the complete analysis synthesis section (2)

iii) Modify the above using polyphase decomposition and noble identities to achieve computational simplicity (2)

6. a) Find the wavelet coefficients W(a,b) for the signal f(t) as a function of b for different values of a such that 0<a<1. Use Haar wavelet

$$f(t) = \begin{cases} 1, & 0 \leq t \leq 1 \\ 0 & otherwise \end{cases}$$ (6)

b) Obtain the 2 band polyphase decomposition of the filter with transfer function

$H(z)=\frac{1-3z^{-1}}{1+4z^{-1}}$ (3)

PART C

7. a) Decompose the signal x(n)= 4, 8, 2 ,6 -2, 4, 2, 6, 2,2 in V1 space to its coarser components in V0 space using Haar Wavelet function (4)

b) Sketch the filter bank implementation of DWT with respect to signal decomposition(4)

c) How can a linear predictor implemented using FIR filter with lattice structure? (4)

8. a) Let $x(n)= A\cos(\omega_0 n+\theta) +e(n)$ where θ is a random variable that is uniformly distributed on the interval from 0 to 2π and e(n) is a sequence of zero-mean random variables that are uncorrelated with each other and also uncorrelated with θ.
   - i) Find the ACF for x(n) if σ_e^2 is variance of e(n) . (2)
   - ii) The power spectrum of x(n) over one period (2)
   - iii) Explain periodogram analysis of the signal using Barlett method (3)

b) Determine the frequency resolution of the Barlett method of power spectrum estimation for a quality factor 10 . The length of the sample sequence is 2000. Comment on the result. (5)

9. Explain how Levinson Durbin algorithm can be used in

i) Design of forward linear predictor (6)

ii) Design of power spectrum estimator using Yule Walker method (6)

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