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MANIPAL INSTITUTE OF TECHNOLOGY
MANIPAL
A Constituent Institution of Manipal University

VII SEMESTER B.TECH (ELECTRICAL & ELECTRONICS ENGINEERING)

END SEMESTER EXAMINATIONS, NOVEMBER 2017

SUBJECT: ADVANCED DIGITAL SIGNAL PROCESSING [ELE 4012]

REVISED CREDIT SYSTEM

Time: 3 Hours **Date: 23 November 2017** **Max. Marks: 50**

Instructions to Candidates:
- Answer **ALL** the questions.
- Missing data may be suitably assumed.

1A. Consider the multi-rate structure shown in Fig. Q 1A with input transform $X(e^{j\omega})$ and filter response $H_0(e^{j\omega})$, $H_1(e^{j\omega})$ and $H_2(e^{j\omega})$. Sketch the following (i) $X_1(e^{j\omega})$; (ii) $X_2(e^{j\omega})$; (iii) $Y_0(e^{j\omega})$; (iv) $Y_1(e^{j\omega})$ and (iv) $Y_2(e^{j\omega})$ ***(04)***

1B. A sequence $x[n]$ up-sampled by 2, is passed through an LTI system $H_1(z)$ and then it is down-sampled by 2 as shown in Fig. Q1B. Is it possible to replace this process with a single LTI system $H_2(z)$? If possible then determine the system function of this system. ***(03)***

1C. Developed an expression for the output $y[n]$ as a function of input $x[n]$ for multi-rate structure shown in Fig. Q1C. Also mention and prove at least one of the identities used in reducing the structure to obtain $y[n]$. ***(03)***

2A. Develop a computationally efficient realization of a factor of 3 decimator employing a length of 9 linear phase FIR low pass filter. Use the symmetry of the impulse response. ***(04)***

2B. Design an efficient two stages decimator with two suitable pair of decimation factors for the following specification:
Input sampling frequency : 60 kHz; Decimation factor : 20; New output frequency : 3 kHz
The highest frequency of interest after decimation : 1.25 kHz ;
Overall passband ripple $\delta_p = 0.05$ and stopband ripple $\delta_s = 0.01$. Justify the answer with appropriate detailed analysis of computational and storage complexities. ***(04)***

2C. Prove that if the two events A and B are not disjoint then the probability of their union event is defined by:

$$\mathrm{P}(\mathrm{A} \cup \mathrm{B}) = \mathrm{P}(\mathrm{A}) + \mathrm{P}(\mathrm{B}) - \mathrm{P}(\mathrm{A} \cap \mathrm{B})$$ ***(02)***

3A. If X be a continuous random variable with PDF

$$\mathrm{F}_X(x) = \begin{cases} 4x^{-2} & ; \text{ for } 2 < x < 4 \\ 0 & ; \text{ otherwise} \end{cases}$$

Find the Variance of random variable X. ***(02)***

3B. Consider a random process $X(t)$ is described by $X(t) = A\cos(2\pi F_0 t + \phi)$,

where $A \& F_0$ are constants. ϕ is random variable which is uniformly distributed over the interval $(-\pi, \pi)$. Determine whether $X(t)$ is wide-sense stationary (WSS)? ***(03)***

3C. A random process signal $X(t)$ has autocorrelation function $R_{XX}(\tau)$ given as

$\mathrm{R}_{\mathrm{XX}}(\tau) = \frac{1}{2\mathrm{a}} \mathrm{e}^{-\mathrm{a}|\tau|}$ where, a=5 kHz. Obtain the following:

(i) the average power (ii) the power spectral density (PSD) of the random signal (iii) Bandwidth required which contains 90% of the signal power. ***(05)***

4A. Show that when a wide-sense stationary (WSS) random process signal $X(t)$ is applied as an input to an LTI system whose impulse response is $h(t)$, then the random output response $Y(t)$ is also wide-sense stationary (WSS) process. ***(04)***

4B. If the sample sequence of a random process has $N = 1000$ samples.

Determine (i) the frequency resolution of the Bartlett, Welch (for 50% overlap), and Blackman-Tukey methods for a quality factor $Q = 10$.

(ii) the record lengths (M) for the Bartlett, Welch (for 50% overlap), and Blackman-Tukey methods. ***(03)***

4C. Considering the single realization of the random process show that the estimate of the power Spectrum density is given by

$P_{xx}(f) = \frac{1}{N}|X(f)|^2$ where $X(f)$ is the Fourier transform of the sample sequence $x[n]$ ***(03)***

5A. What is STFT? How is it related to Continuous time Wavelet Transform? ***(02)***

5B. Draw 1D analysis and synthesis filter bank for DWT and explain function of each block. ***(04)***

5C. Consider the DSP system used for noise cancellation application as shown in Fig.Q5C in which d(0)=3, d(1)=-2, d(2)=1, x(0)=3, x(1)=-1, x(2)=2, and there is an adaptive filter with two taps y(n)=w(0)x(n)+w(1)x(n-1) with initial values w(0)=0, w(1)=1, and u=0.1. Determine LMS algorithm equations for the adaptive filter. Also, perform adaptive filtering for each n=0, 1, 2. ***(04)***

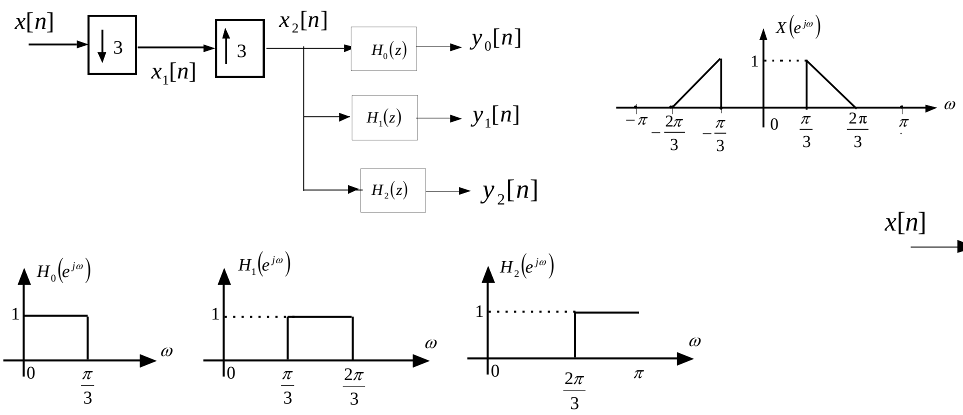

Fig.Q1A

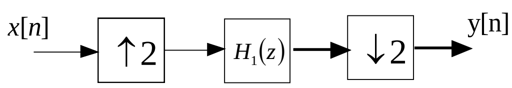

Fig.Q1*B*

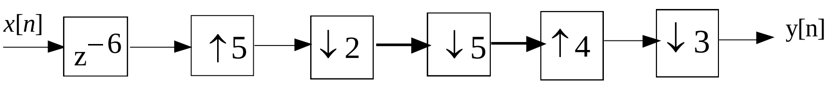

Fig.Q.1*C*

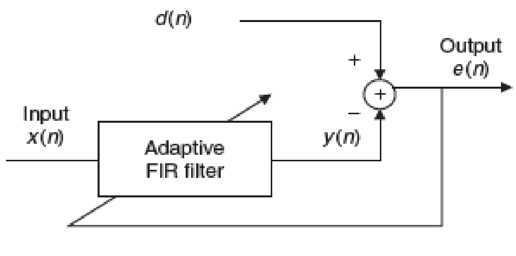

Fig.Q5C