(Pages : 2)

C – 2635

Reg. No. : ....

Name : ....

First Semester M.Tech. Degree Examination, June 2017
(2013 Scheme)
Electronics and Communication
Stream : Signal Processing
TSC1003 : Digital Filter Design and Applications

Time : 3 Hours Max. Marks : 60

***Instruction** : Answer **any two** questions from **each** Module.*
***All** questions carry **equal** marks.*

MODULE – I

1. Check the validity of the following statements :
   a) The convolution of two-minimum phase sequences is always a minimum-phase sequence.
   b) The sum of two minimum-phase sequences is always minimum phase.

2. A discrete-time system with input x(n) and output y(n) is described in the frequency domain by the relation $Y(\omega) = e^{-j2\pi\omega} X(\omega) + \frac{d}{d\omega} \times (\omega)$.
   a) Compute the response to the input $x(n) = \delta(n)$.
   b) Check if the system is LTI and stable.

3. Consider a finite-duration sequence :

   $x(n) = \{\underset{\uparrow}{0}, 1, 2, 3, 4\}$

   a) Sketch the sequence S(n) with a 6 – point DFT

      $S(k) = w_2^{*}x(k)$; k = 0, 1, 2, 3, 4, 5.
   b) Determine the sequence y(n) with a 6-point DFT.

      Y(k) = Real part | X(k) |.
   c) Determine the sequence z(n) with a 6-point DFT

      Z(k) = Im part | X(k) |.

P.T.O.

MODULE – II

4. Obtain the transfer function of a low pass digital filter meeting the following specifications :
   Pass band 0 – 60 Hz
   Stop band > 85 Hz
   Stop band attenuation > 15 dB
   Assume a sampling frequency of 256 Hz and a Butterworth characteristics.

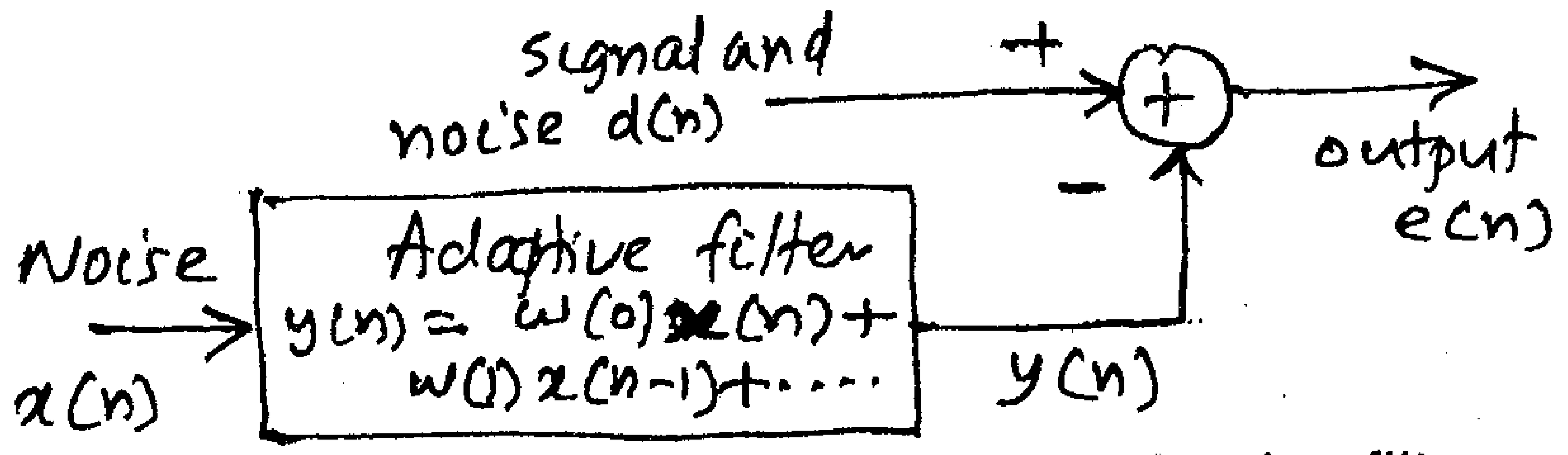
5. A low pass filter is characterized by the following :
   Pass band edge frequency = 1.5 kHz
   Sampling frequency = 10 kHz
   Number of coefficients = 15
   a) Obtain the coefficients of the low pass filter using the Hamming window.
   b) Write down the specifications for an equivalent high pass filter and use these to obtain its coefficients.

6. Sketch a lattice filter implementation of the FIR filter.
   $H(z) = 8 + 4z^{-1} + 2z^{-2} + z^{-3}$.

MODULE – III

7. Explain the basic LMS adaptive algorithm. Discuss in detail its practical limitations.

8. Consider a DSP system for noise cancellation applications with two taps show below.



   a) Set up the LMS algorithm for the adaptive filter.
   b) Assume the inputs x(0) = 1, x(1) = 1, x(2) = – 1, x(3) = 2, d(0) = 0, d(1) = 2, d(2) = – 1 and d(3) = 1; initial weights w(0) = w(1) = 0; and convergence factor μ = 0.1. Perform adaptive filtering to obtain outputs e(n) for n = 0, 1, 2.

9. Suppose there are 1000 samples from a sample sequence of a random process.
   a) Determine the frequency resolution of the Bartlett and Blackman-Tukey methods for a quality factor Q = 10.
   b) Determine the record lengths for the Bartlett and Blackman-Tukey methods.

(6×10=60 Marks)