Seat No.: ________ Enrolment No.______________

GUJARAT TECHNOLOGICAL UNIVERSITY

M. E. - SEMESTER – II • EXAMINATION – WINTER • 2013

Subject code: 1722601 **Date: 24-12-2013**

Subject Name: Advanced Digital Signal Processing and Applications

Time: 10.30 am – 01.00 pm **Total Marks: 70**

Instructions:

1. **Attempt all questions.**
2. **Make suitable assumptions wherever necessary.**
3. **Figures to the right indicate full marks.**

Q.1 **(a)** Explain the forward linear prediction and find the conditions for minimum mean square prediction error. **07**

(b) What is Schure Algorithm? Explain the recursive procedures to describe Schure Algorithm. **07**

Q.2 **(a)** Explain the IIR Wiener filter. Find the system function for the optimum IIR Wiener filter. **07**

(b) Determine the lattice coefficients corresponding to the FIR Filter described by the system function **07**

$$H(z) = A_3(z) = 1 + \frac{13}{24} z^{-1} + \frac{5}{8} z^{-2} + \frac{1}{3} z^{-3}$$

OR

(b) Given a signal x(n) = s(n) +w(n),where s(n) is an AR(1) Process that satisfies the difference equation **07**

s(n)=0.6s(n-1) + v(n),

Where {v(n)} is a white noise sequence with variance σ_v^2 =0.64,and {w(n)} is a white noise sequence with variance σ_w^2 =1. Design a Wiener filter of length M=2 to estimate {s(n)}

Q.3 **(a)** Define multirate digital signal processing .Explain how an increase in a sampling rate by an integer factor of I can be achieved. **07**

(b) Explain and compare AR and ARMA random processes with mathematical expressions for system functions as well as lattice structures. **07**

OR

Q.3 **(a)** Determine the impulse response of the FIR filter that is described by the lattice coefficients. **07**

K_1= 0.6, K_2 = 0.3, K_3 = 0.5 and K_4 = 0.9

(b) Explain the implementation of sampling rate conversion using polyphase filter structure. **07**

Q.4 **(a)** Explain the minimum phase property of the forward prediction error filter and Maximum phase property of the backward prediction error filter. **07**

(b) The polyphase matrix for a three channel perfect reconstruction FIR QMF bank is **07**

$$P(z^3) = \begin{bmatrix} 1 & 1 & 2 \\ 2 & 3 & 1 \\ 1 & 2 & 1 \end{bmatrix}$$

Determine the analysis and the synthesis filters in the QMF bank.

OR

Q.4 **(a)** Define the mean ergodic random process. Derive the condition for a random process to be mean ergodic. **07**

(b) How narrowband interference in wideband signal can be suppressed using adaptive filter? Explain with block diagram and mathematical relations. **07**

Q.5 **(a)** Explain the steps for recursive computation of filter coefficients using RLS adaptive filter algorithm. **07**

(b) Explain about subband coding of speech signal. **07**

OR

Q.5 **(a)** Explain the application of adaptive filter for Echo cancellation in data transmission over telephone lines using Symbol rate echo canceller. **07**

(b) Design a linear phase FIR filter that satisfies the following specifications : **07**

Sampling frequency 8000Hz
Passband: $0 \le F \le 75$
Transition band: $75 \le F \le 80$
Stopband :$80 \le F \le 4000$
Passband ripple: $\delta_1 = 10^{-2}$
Stopband ripple: $\delta_2 = 10^{-4}$

Show that under similar condition, a lowpass linear phase FIR filter may be more efficiently implemented in a multistage decimator interpolator configuration.

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