VALLIAMMAI ENGINEERING COLLEGE

S.R.M Nagar, Kattankulathur-603203

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

I - M. E (C.S)

AP7101 – ADVANCED DIGITAL SIGNAL PROCESSING

QUESTION BANK

UNIT I

PART A

1. Define statistical variance and covariance.
2. Define wide sense stationary process.
3. Write Yule-Walker equation.
4. What are the three cases of interest to derive $a_p(k)$ of p x p non-symmetric toeplitz matrix?
5. What are the minimum error of all-pole and all-zero modeling.
6. Compute autocorrelation of sequence. $x(n) = (0.5)^n u(n)$.
7. Define ensemble average.
8. Define all modeling using covariance method.
9. Find the power spectrum of wide sense stationary random process for given autocorrelation sequence.
   $r_x(k)=2\delta(k)+j\,\delta(k-1)-j\,\delta(k+1)$.
10. Define periodicity of WSS random process.
11. Write the modified Yule-Walker equation.
12. Define all pole modeling using autocorrelation method.
13. Define linear prediction error and linear prediction coefficient
14. Find the autocorrelation sequence for given power spectral density. $P_x(e^{jw})=3+2\cos w$
15. How do you compute the energy of a discrete signal in time and frequency domain?
16. Check whether given matrix is a valid autocorrelation matrix.
   $$\begin{matrix} 1 & 1+j \\ 1-j & 1 \end{matrix}$$
17. The power density spectrum of AR process [x(n)] is given as
   $$\Gamma_{ww}(w) = \frac{\sigma_w^2}{|A(w)|^2} = \frac{25}{\left|1-e^{-jw}+0.5e^{-2jw}\right|^2}$$
18. State wiener –kitchenane relation.
19. What are the drawbacks of least square modeling method?
20. What are the normal equations of prony method of modeling a signal as unit sample response of a LSI having p poles and q zeros.

PART B

1. State and prove wiener - khinchine relation (16 marks)

2. (i) Obtain the filter to generate a random process with a power spectrum
   $P_x(e^{jw}) = (5+4\cos w) / (10+6\cos w)$ from white noise (10 marks)
   (ii) Find the power spectrum of the wide sense stationary random process that have correlation sequence $r_x(k)=\delta(k)2(0.5)^{|k|}$ (6 marks)

3. Explain and derive the expression for signal modeling using pade approximation. (16 marks)

4. Obtain the expression for prony's method of signal modeling for approximating a signal x(n) as unit sample response of LSI system having p poles and q zeros. (16 marks)

5. Obtain the expression for all –pole modeling using prony's method. (16 marks)

6. Explain all pole modeling using auto correlation and covariance method. (16 marks)

7. Explain the following stochastic models (16 marks)
   Pole-zero model
   Pole model
   Zero model

8. (i) Explain the concept of signal modeling and explain least squares method of signal modeling in detail. (8 marks)
   (ii) Explain the steps in the determination of the autocorrelation and power spectrum of a random process. (8 marks)

9. (i) Explain Iterative prefiltering. (8 marks)
   (ii) Explain FIR least square inverse filter. (8 marks)

10. Explain shank's method for solving normal equations. (16 marks)

11. The input to the linear shift invariant filter with impulse response
    $h(n) = \delta(n) + \frac{1}{2}\delta(n-1) + \frac{1}{4}\delta(n-2)$ is z zero mean wide sense stationary process with autocorrelation $r_x(k) = \left[\frac{1}{2}\right]^{|k|}$
    (i) What is the variance of the output process? (6 marks)
    (ii) Find the autocorrelation of the output process, $r_y(k)$, for all k. (10 marks)

12. (i) Explain spectral factorization (8 marks)
    (ii) Explain filtering random process. (8 marks)

UNIT II

PART A

1. Compare Parametric and Non-Parametric methods of spectral estimation.
2. Define Periodogram? How can it be smoothed?
3. What are the difficulties in FFT based power spectral estimation methods?
4. List the disadvantages of non-parametric methods.
5. Mention the three models used for parametric power estimation.
6. What are the demerits of the periodogram?
7. Define bias and consistency.
8. Find the mean and variance of the periodogram of white noise power spectral density is unity.
9. Compare performance measures for the Parametric and Non-Parametric methods of spectral estimation.
10. Determine the frequency resolution of the Barlett, Welch methods of power spectrum estimates for a quality factor Q=15. Assume that overlap in Welch's method in 50% and length of the sample sequence is 1000.
11. Compare AR, MA, ARMA models with respect to complexity.
12. Write the bias equation of modified periodogram.
13. Define sample autocorrelation function. Give the mean value of this estimate.
14. Define Biasing of periodogram and modified periodogram.
15. What are the variance of Barlett method and Welch's method of periodogram?
16. What are the advantages of parametric methods.
17. Define Autoregressive Spectrum estimation using autocorrelation method
18. Define Autoregressive Spectrum estimation using covariance method
19. Define Autoregressive moving average spectrum estimation.
20. Define moving average spectrum estimation.

PART B

1. Explain the following parametric methods to measure be spectrum of long duration signals.
   (i) ARMA model
   (ii) MA model (16 marks)

2. Derive the variance of the periodogram using Blackman-Tukey method. (16 marks)

3. (i) Define periodogram. How DFT is used for its computation. (6 marks)
   (ii) Explain the Blackman Tukey method of power spectrum estimation. (10 marks)

4. (i) Explain how power spectrum can be estimated from the AR model. (8 marks)
   (ii) Discuss the Welch method of periodogram averaging. (8 marks)

5. (i)Explain power spectrum estimation using Barlett method. (8 marks)
   (ii) In the Welch method, calculate the variance of the Welch power spectrum estimate with the Barlett window if there is 50% overlap between successive sequences. (8 marks)

6. (i) Explain parameter estimation using Yule-walker method. (8 marks)
   (ii) Use Levinson recursion to find predictor polynomial corresponding the autocorrelation sequence $R=[2, 1, 1, -2]^T$ (8 marks)

7. (i) Barlett's method is used to estimate the power spectrum of a process from a sequence of N=2000 samples. (8 marks)
   a. What is the minimum Length L that may be used for each sequence if we are to have a resolution of $\Delta f = 0.005$?
   b. Explain why it should not be advantageous to increase beyond L beyond the value found in (a)

   (ii) Explain the modified periodogram. (8 marks)

8. Let $r_x(k)$ be a complex autocorrelation sequence given by $r_x=[2,\ 0.5(1+j), 0.5j]^T$. Use the Levinson-Durbin recursion to solve the autocorrelation normal equations for a second order all-pole model. (16 marks)

9. Explain how the Yule-Walker equations can be solved using Levinson-Durbin Algorithm (16 marks)

10. The estimated autocorrelation sequence of a random process x(n) for lags k=0, 1, 2, 3, 4 are
    $r_x(0)=2$, $r_x(1)=1$, $r_x(2)=1$, $r_x(3)=0.5$, $r_x(4)=0$
    Estimate the power spectrum of x(n) for each of the following cases.
    (a) x(n) is an AR(2) process.
    (b) x(n) is an MA(2) process.
    (c) x(n) is an ARMA(1,1) process. (16 marks)

UNIT III

PART A

1. How will you find the ML estimate?
2. Give the basic principle of Levinson recursion.
3. What are FIR systems?
4. Compare IIR and FIR wiener filters.
5. Write the error criterion for LMS algorithm.
6. Draw the structure of the forward prediction error filters.
7. What is Lattice structure? What is the advantage of such structure?
8. What are the properties of prediction error filters?
9. Mention the advantages of Wiener filter.
10. Name any one application of the AR model.
11. What is a whitening filter?
12. What is meant by linear prediction?
13. How wiener filter can be modified as linear predictor?
14. Define maximum likelihood criterion.
15. Define discrete Wiener Hoff equations.

16. Write the applications of Kalman filter.
17. How FIR wiener filter can be used as noise cancellation?
18. What is the minimum error for a noncausal filter?
19. What is the minimum error for a causal wiener filter?
20. How causal IIR wiener filter can be used as noise cancellation?

PART B

1. (i) Describe the basics of forward linear prediction. Give the schematic of FIR filter and Lattice filter for the first order predictor. (8 marks)
   (ii) Derive the recursive predictor coefficients for optimum lattice predictor by Levinson-Durbin algorithm. (8 marks)

2. Explain how FIR Wiener filter can be used for filtering and prediction. (16 marks)

3. Derive Wiener Hopf equations and the minimum mean square error for the FIR wiener filter. (16 marks).

4. Derive Wiener Hopf equations and the minimum mean square error for a non causal wiener filter. (16 marks).

5. Derive Wiener Hopf equations and the minimum mean square error for a causal wiener filter. (16 marks).

6. Explain how Wiener filter can be used for optimum causal linear predictor. (16 marks)

7. Explain Weiner deconvolution. (16 marks)

8. Briefly explain the estimation of a non stationary process by a Kalman filter. (16 marks)

9. Let us consider linear prediction in noisy environment. Suppose that a signal is corrupted by noise. $x(n)=d(n)+w(n)$, where $r_w(k)=0.5\delta(k)$ and $r_{dw}(k)=0$. The signal d(n) in an AR(1) process that satisfies the difference equation
   $d(n)=0.5d(n-1)+v(n)$, where v(n) is white noise with variance $\sigma_v^2=1$. Assume that w(n) and v(n) are uncorrelated.
   Design a first order FIR linear predictor $W(z)=w(0)+w(1)z^{-1}$ for d(n) and find the mean square prediction error $\varepsilon = E\{[d(n+1) - \hat{d}(n+1)]^2\}$. (16 marks)

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    $d(n)=0.5d(n-1)+v(n)$, where v(n) is white noise with variance $\sigma_v^2=1$. Assume that w(n) and v(n) are uncorrelated.
    Design a causal Wiener predictor and compute mean square error. (16 marks)

UNIT IV

PART A

1. Why are FIR filters used in adaptive filter application?
2. What is adaptive noise cancellation?
3. Define misadjustment of adaptive filter
4. What is need for adaptivity?
5. How will you avoid echos in long distance telephonic circuits?
6. Express the LMS adaptive algorithm. State its properties.
7. What is the need for adaptive filters?
8. What is meant by channel equalization?
9. State the properties of Widrow-Hopf LMS adaptive algorithm.
10. List some applications of Adaptive filters.
11. What is the principle used in LMS algorithm?
12. What are the advantages of FIR adaptive filters?
13. Why LMS is normally preferred over RLS?
14. What is the relationship between the order of the filter with the step size in LMS adaptive filter?
15. Write the difference between LMS algorithm and RLS algorithm.
16. What is the principle of steepest descent adaptive FIR filter?
17. What is the advantage of normalized LMS over LMS adaptive filter?
18. Define error function of exponentially weighted RLS
19. Define error function of sliding window RLS
20. Define time constant for the steepest descent FIR adaptive filter

PART B

1. (i) Explain direct form FIR adaptive filter. (10 marks)
   (ii) Derive the weight vector update equation of the LMS algorithm. (6 marks)

2. Discuss adaptive noise cancellation using LMS algorithm. (16 marks)

3. (i)Explain adaptive echo cancellation. (8 marks)
   (ii) Explain adaptive channel equalization. (8 marks)

4. Obtain Widrow-Hoff LMS adaptation algorithm. (16 marks)

5. Explain steepest descent algorithm for FIR adaptive filter. (16 marks)

6. Explain the sliding window RLS algorithm. (16 marks)

7. Explain the RLS algorithm with the exponentially weighted factor. (16 marks)

8. Explain normalized LMS algorithm. (16 marks)

9. Discuss the convergence of the LMS algorithm in detail. (16 marks)

10. The first three autocorrelation of a process x(n) are

$$r_x(0)=1,\ r_x(1)=0.5,\ r_x(2)=0.5$$

Design a two-coefficient LMS adaptive linear predictor for x(n) that has a misadjustment M=0.05 and find the steady state error. (16 marks)

UNIT V

PART A

1. State the need for multirate signal processing
2. Define wavelet.
3. What is interpolation? Give one example.
4. Write down the application of Wavelet transform in Data Compression.
5. What is discrete wavelet transform?
6. What are the advantages of multistage implementation in multirate signal processing?
7. What is purpose of low pass filter before the down sampler?
8. What is the function of the downsampler?
9. List the applications of wavelet transform.
10. What is sub-band coding?
11. Determine the polyphase decomposition for the following FIR filter
12. Give the two channel wavelet filter banks to decompose the input signal into frequency bands.
13. What is meant by image smoothing and image sharpening?
14. What is the effect on power spectrum due to up sampling and down sampling?
15. What is the need for anti-imaging filter in multirate digital signal processing?
16. Name two applications of multirate sampling
17. Why Polyphase filters are named so?
18. Differentiate decimation and interpolation.
19. When can a digital system be called as multirate system?
20. What are the advantages of wavelet transform?

PART B

1. Describe the mathematical equations how sampling rate can be increased by a factor of L. (16 marks)

2. Explain the need for multistage implementation of sampling rate conversion. Describe the implementation for a factor of I/D. (16 marks)

3. Enumerate in detail about the continuous and discrete wavelet transform (16 marks)

4. Explain the concept of multirate signal processing with spectral interpretation of decimation of a signal from 6 KHz to 2KHz and spectral interpretation of interpolation of a signal from 2 KHz to 6 KHz. (16 marks)

5. (i) Explain the realization of an FIR filter based on Type I and Type II poly phase decomposition (8 marks)
(ii) Explain the Encoder and decoder-operation of sub-band coding technique. (8 marks)

6. Consider a Decimator with down sampling factor 3 and a 12^{th} order filter. After deriving necessary equations draw the structure of the Decimator with the derived poly phase filters. (16 marks)

7. With necessary equations and diagrams, discuss about the interpolation and decimation in multirate signal processing. (16 marks)

8. Explain the application of multirate signal processing in adaptive sub-band coding system. (16 marks)

9. (i) Explain the concept of polyphase decomposition in implementing multirate systems. (8 marks)
(ii) Describe about sub band coding. (8 marks)

10. Design a two stage decimator for the following specifications
D = 100, Passband $0 \leq F \leq 50$Hz, Transition band $50 \leq F \leq 55$Hz and Input sampling rate 10KHz
Ripple S1 = 1/10 and S2 = 1/1000. (16 marks)