SHRI ANGALAMMAN COLLEGE OF ENGINEERING & TECHNOLOGY

(An ISO 9001:2008 Certified Institution)

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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

UNIT –I DFT & Fast Fourier Transform

PART-A

1. What is digital signal?
2. Define DFT of a discrete time sequence
3. Define IDTFT
4. What is the drawback in DTFT?
5. Define circular convolution.
6. Why FFT is needed?
7. Calculate the number of multiplications needed in the calculation of DFT and FFT with 64 point sequence.
8. What is FFT?
9. How many multiplications and additions are required to compute N-point DFT using radix-2 FFT?
10. What is meant by radix-2 FFT?
11. What is DIT radix2 algorithm?
12. What is DIF radix2 algorithm?
13. What are the differences between DIT and DIF algorithms?
14. Draw the flow-graph of a two-point DFT for decimation in time decomposition

15. Draw the basic butterfly diagram for decimation in time algorithm
16. Draw the basic butterfly for decimation in frequency decomposition
17. What are the applications of FFT?
18. What is the drawback in DTFT?
19. What are the differences and similarities between DIF and DIT algorithms?
20. What are the advantages of FFT algorithm over direct computation of DFT?
21. What is decimation-in-time algorithm?
22. What is decimation-in-frequency algorithm?
23. State sampling theorem?
24. What is aliasing?

PART – B

1. Find the DFT of a sequence x(n)={1,2,3,4,4,3,2,1} using DIT algorithm.
2. (a) Find the output y(n) of filter whose impulse response is h(n)={1,2} and input signal x(n)={1,2,-1,2,3,-2,-3,-1,1,1,2,-1}using overlap save method?
3. Proof circular Frequency Shift DFT property?
4. Derive and draw the radix -2 DIT algorithms for FFT of 8 points?
5. Derive and draw the radix -2 DIF algorithms for FFT of 8 points?
6. . Find the output y(n) of a filter whose impulse response is h(n) = {1, 1, 1} and input signal x(n) = {3, -1, 0, 1, 3, 2, 0, 1, 2, 1}. Using Overlap save method

7. Derive and draw the radix -2 DIF algorithms for FFT of 8 points
8. State and prove shifting property of DFT.
9. Compute 4- point DFT of casual three sample sequence is given by,

   $x(n) = 1/3, 0 < n < 2$
   $= 0$, else.

UNIT - II Design of Filter

PART-A

1. What is filter?
2. What are FIR and IIR systems?
3. How phase distortion and delay distortion are introduced?
4. What are FIR filters?
5. What are advantages of FIR filter?
6. What are the disadvantages of FIR FILTER
7. What is the necessary and sufficient condition for the linear phase characteristic of a FIR filter?
8. List the well known design technique for linear phase FIR filter design?
9. Define IIR filter?
10. Distinguish IIR and FIR filters
11. Distinguish analog and digital filters
12. Write the steps in designing chebyshev filter?
13. Write down the steps for designing a Butterworth filter?

14. What is warping effect?
15. Write a note on pre warping.
16. Why impulse invariant method is not preferred in the design of IIR filters other than low pass filter?
17. What is meant by impulse invariant method?
18. List the Butterworth polynomial for various orders.
19. What is Gibbs phenomenon? OR What are Gibbs oscillations?
20. Compare Hamming window with Kaiser window.
21. What are the advantage of Kaiser widow?
22. Give the bilinear transformation.
23. What is linear phase? What is the condition to be satisfied by the impulse response in order to have a linear phase?
24. Give any two properties of Butterworth filter and chebyshev filter.
25. What are the desirable and undesirable features of FIR Filters?

PART-B

1. Design an ideal low pass filter with a frequency response

$$H_d(e^{jw}) = 1 \text{ f or } -\pi/2 \le \omega \le \pi/2$$
$$0 \text{ for } \pi/2 \le \omega \le \pi.$$

Find the values of h(n) for N=11. Find H(Z).Plot the magnitude response.

2. Design an Hamming low pass filter with a frequency response

$$H_d(e^{jw}) = 1 \text{ f or } \pi/4 \le |\omega| \le \pi$$
$$= 0 \text{ for } |\omega| \le \pi.$$

Find the values of h(n) for N=11. Find H(Z).Plot the magnitude response.

3. (a) Explain design of FIR filter by frequency sampling techniques?
   (b) Derive the frequency response of a linear phase FIR filter when impulse response is symmetric an order N is odd?

4. Derive the frequency response of a linear phase FIR filter when impulse response is symmetric an order N is even?

5. Derive the frequency response of a linear phase FIR filter when impulse response is Asymmetric an order N is odd?

6. Derive the frequency response of a linear phase FIR filter when impulse response is Asymmetric an order N is even?

7.(a) Find the denominator polynomial of a Butterworth filter for N = 4
   (b) Derive the expression for order of the filter N of a Butterworth filter?

8. Design a digital Butterworth filter that satisfies the following constraint using bilinear transformation. Assume T=1 sec.

$$\sqrt{0.5} \le |H(e^{j\omega})| \le 1 ; \quad 0 \le \omega \le \pi/2$$
$$|H(e^{j\omega})| \le 0.2 ; \quad 3\pi/4 \le \omega \le \pi$$

9. Design a digital Chebyshev filter that satisfies the following constraint using Impulse invariance method. Assume T=1 sec.

$$0.707 \le |H(e^{j\omega})| \le 1 ; \quad 0 \le \omega \le 0.2\pi$$
$$|H(e^{j\omega})| \le 0.1 ; \quad 0.5\pi \le \omega \le \pi$$

10.(a) Determine H(Z) using the impulse invariant technique for the analog system function

$$H(s) = 1 / (s+0.5)(s^2+0.5s+2).$$

(b) Determine H(Z) using bilinear transformation for the analog system function

$$H(s) = 4 / (s^2+2\sqrt{2}s+4).$$

11.Design a digital Butterworth filter that satisfies the following constraint using bilinear transformation. Assume T=1 sec.

$$0.9 \leq |H(e^{j\omega})| \leq 1 \; ; \; 0 \leq \omega \leq \pi/2$$

$$|H(e^{j\omega})| \leq 1 \; ; \; 3\pi/4 \leq \omega \leq \pi$$

12. Design a digital Chebyshev filter that satisfies the following constraint using Impulse invariance method. Assume T=1 sec.

$$0.8 \leq |H(e^{j\omega})| \leq 1 \; ; \; 0 \leq \omega \leq 0.2\pi$$

$$|H(e^{j\omega})| \leq 0.2 \; ; \; 0.6\pi \leq \omega \leq \pi$$

13. (a). Determine H(Z) using the impulse invariant technique for the analog system function $H(s) = 1 / (s+0.5)(s^2+0.5s+2)$.

(b).derive on IIR Filter design by the bilinear transformation with mapping in analog and frequency transformation?

14. (a). Draw the structures of cascade and parallel realization of

$$H (Z) = (1-z^{-1})^3 / (1- \quad z^{-1}) (1- z^{-1}).$$

Unit – III FINITE WORD LENGTH EFFECTS

PART-A

1. Express the fraction (-9/32) in sign magnitude, 2's complement notations using 6 bits . (Nov 2008)
2. What are the three types of quantization error occurred in digital systems?

3. What is meant by limit cycle oscillations?
4. Express the fraction(-7/32) in signed magnitude and two's complement notations using 6 bits.
5. Express the fraction 7/8 and -7/8 in sign magnitude, 2's complement and 1's complement.
6. Define Sampling rate conversion.
7. Convert the number 0.21 into equivalent 6-bit fixed point number.
8. Why rounding is preferred to truncation in realizing digital filter?
9. What are the different quantization methods?
10. What is zero padding? Does zero padding improve the frequency resolution in the spectral estimate?
11. List the advantages of floating point arithmetic.
12. Give the expression for signal to quantization noise ratio and calculate the improvement with an increase of 2 bits to the existing bit.
13. Draw the probability density function for rounding.
14. Compare fixed point and floating point representations.
15. What is dead band?
16. How can overflow limit cycles be eliminated?
17. What is zero input limit cycle oscillation?
18. What is steady state noise power at the output of an LTI system due to the quantization at the input to L bits?
19. What is meant by finite word length effects in digital filters?
20. What is round-off noise error?

21. What is meant by fixed point arithmetic? Give example?

PART-B

1. Explain truncations and rounding Errors?
2. $H(z)=H_1(z)H_2(z)$,where $H_1(z)=1/(1-0.5z^{-1})$ and $H_2(z)=1/(1-0.4z^{-1})$ .find out output round off noise power .calculate the value if b=3(excluding sign bit)?
3. Write Short notes on any three Errors?
4. Derive Signal Scaling?
5. Derive on quantization noise power?
6. Explain limit cycle oscillation?
7. Explain input quantization noise error?
8. . The output of an A/D is fed through a digital system whose system function is
9. H(Z)=(1-_)z /(z-_), 0<_<1.Find the output noise power of the digital system.
10. 2. The output of an A/D is fed through a digital system whose system function is
11. H(Z)=0.6z/z-0.6. Find the output noise power of the digital system=8 bits
12. 3. Discuss in detail about quantization effect in ADC of signals. Derive the expression for Pe(n) and SNR.
13. A digital system is characterized by the difference equation
14. Y(n)=0.95y(n-1)+x(n).determine the dead band of the system when x(n)=0 and y(-1)=13.

15.6. Two first order filters are connected in cascaded whose system functions of theIndividual sections are $H1(z)=1/(1-0.8z^{-1})$ and $H2(z)=1/(1-0.9z^{-1})$.Determine theOver all output noise power.

Unit IV POWER SPECTRUM ESTIMATION

1. Define unbiased estimate and consistent estimate.
2. What are the disadvantages of non-parametric methods of power spectral estimation?
3. What is periodogram?
4. Determine the frequency resolution of the bartlett method of power spectrum estimates for a quality factor Q=15. Assume that the length of the sample sequence is 1500.
5. Define the terms autocorrelation sequence and power spectral density
6. Define power spectral density and cross spectral density.
7. Explain deterministic and nondeterministic signals with examples.
8. Explain the use of DFT in power spectrum estimate?
9. Define autocorrelation.
10. List the non-parametric methods for power spectral estimation.
11. What are the steps involved in Bartlett method?
12. What are the steps involved in Welch method?
13. Define Blackman and turkey method?

PART-B

1. Explain how DFT and FFT are useful in power spectral estimation.
2. Explain Power spectrum estimation using the Bartlett window.
3. Obtain the mean and variance of the averaging modified period gram estimate.
4. How is the Blackman and Tukey method used in smoothing the periodogram?
5. Derive the mean and variance of the power spectral estimate of the Blackman and Tukey method.
6. What are the limitations of non-parametric methods in spectral estimation?
7. How the parametric methods overcome the limitations of the non-parametric methods?

UNIT – V DIGITAL SIGNAL PROCESSORS

1. What are the factors that may be consideredd when selecting a DSP processor for an application?
2. State the merit and demerit of multiported memories?
3. What is meant by pipelining?
4. What is the principal features of the harvard Architecture?
5. Differentiate between von Neumann and Harvard architecture?
6. Give the digital signal processing application with the TMS 320 family.
7. What is the advantage of Harvard architecture of TMS 320 series?
8. What are the desirable features of DSP Processors?

9. What are the different types of DSP Architecture?

10. Define MAC unit?

11. Mention the Addressing modes in DSP processors.

12. State the features f TMS3205C5x series of DSP processors.

13. Define Parallel logic unit?

14. Define scaling shifter?

15.Define ARAU in TMS320C5X processor?

16.What are the Interrupts available in TMS320C5X processors?

17.What are the addressing modes available in TMS320C5X processors?

18. Write the syntax of assembly language syntax.

PART-B

1. Explain in detail about the applications of PDSP?

2. Explain briefly :(i). Von Neumann architecture (ii). Harvard architecture

(iii).VLIW architecture

3. Explain in detail about (i). MAC unit (ii). Pipelining

4. Draw and explain the architecture of TMS 320C5x processor?

5. Explain in detail about the Addressing modes of TMS 320C50?

6. Explain in detail about (i). Multiplier (ii). Shifter

7. What is pipelining? Explain various stages of pipelining?