VALLIAMMAI ENGINEERING COLLEGE
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DEPARTMENT OF ELECTRICAL AND ELECTRONICS ENGINEERING

EC2314 –DIGITAL SIGNAL PROCESSING

QUESTION BANK

UNIT I

PART A

1. Define Nyquist rate.
2. Define sampling theorem.
3. What is known as Aliasing?
4. What are even and odd signals?
5. Given a continuous time signal x(t)=2cos500πt. What is the Nyquist rate and fundamental frequency of the signal?
6. Determine whether x[n]=u[n] is a power signal or an energy signal.
7. What is the Nyquist rate for the signal $x_a(t)=3\cos 600\pi t + 2\cos 1800\pi t$?
8. Determine the fundamental period of the signal $\cos\left[\frac{\pi 30n}{105}\right]$
9. Consider the analog signal x(t)=3 cos 50πt + 10 sin 300 πt - cos 100 πt. What is the Nyquist rate for this signal.
10. Mention few applications of Digital Signal processing.
11. What are the different types of representation of discrete time signals?
12. What is Energy and Power signals?
13. What are the classification of system.
14. What are the classification of signals.
15. Define recursive systems.
16. Determine the system described by the equation y (n) = n x (n) is linear or not.
17. Check whether the signal defined by $x(n) = [5\,(1/2)^n + 4\,(1/3)^n]\,u(n)$ is causal.
18. What do you mean by BIBO stable?
19. What is anti aliasing filter? What is the need for it?
20. Define staic and dynamic systems.

PART B

1. (i) What is meant by energy and power signal? Determine whether the following signals are energy or power or neither energy nor power signals.
   (1) $x_1(n)=(1/2)^n u(n)$
   (2) $x_2(n)=\sin(\pi n/6)$
   (3) $x_3(n)=e^{j(\pi n/3+\pi/6)}$
   (4) $x_4(n)=e^{2n}u(n)$ (12)

   (ii) Explain the concept of quantization. (4)

2. Check for following systems are linear, causal, time invariant, stable, static (16)
   (i) y(n)=x(1/2n)
   (ii) y(n)=sin(x(n))
   (iii) y(n)=x(n)cos(x(n))
   (iv) y(n)=x(-n+5)
   (v) y(n)=x(n)+nx(n+2)

3. (i) Check whether the following is linear, time invariant, casual and stable
   y(n) =x(n) +nx (n +1) . (8)
   (ii) Check whether the following are energy or power signals.
   (1) $x(n) = \left[\frac{1}{2}\right]^{n} u(n)$
   (2) $x(n) = Ae^{jw_0 n}$ (8)

4. (i) What do you mean by Nyquist rate? Give its significance. (6)
   (ii) Explain the classification of discrete signal. (10)

5. Check whether following are linear, time invariant, causal and stable.
   (i) y(n) =x(n) +nx (n +1) . (8)
   (ii) y(n) =cos x(n) . (8)

6. Starting from first principles, state and explain sampling theorem both in time domain and in frequency domain. (16)

7. Determine the response of the following systems to the input signal

$$x(n) = \begin{cases} |n|, & -3 \leq n \leq 3 \\ 0, & \text{otherwise} \end{cases}$$

   (i) $x_1(n)=x(n-2)\delta(n-3)$
   (ii) $x_2(n)=x(n+1)u(n-1)$
   (iii) $y(n) = \frac{1}{3}[x(n+1) + x(n) + x(n-1)]$
   (iv) y(n)=max[x(n+1), x(n), x(n-1)]
   (v) Find the even and odd components of given x(n). (16)

8. A discrete time systems can be
   (i) Static or dynamic
   (ii) Linear or non Linear
   (iii) Time invariant or time varying
   (iv) Stable or unstable

   Examine the following systems with respect to the properties above y(n)=x(n)+nx(n+1).
   (16)

9. (i) Check the causality and stability of the systems $y(n)=x(-n)+x(n-2)+x(2n-1)$. (8)
   (ii) Check the system for linearity and time invariance $y(n)=(n-1)x^2(n)+c$. (8)

10. Explain the digital signal processing system with necessary sketches and give its merits and demerits.(16)

UNIT II

PART A

1. What is ROC of z transform? State its properties.
2. Define discrete time Fourier transform pair for a sequence.
3. Consider the signal $x(n)=|1|$ for $-1\leq n\leq 1$ and 0 for all other values of n, sketch the magnitude and phase spectrum.
4. Find the convolution for $x(n)=\{0,1,0,2\}$ and $h(n)=\{2,0,1\}$.
5. Obtain the Discrete Fourier series coefficients of $x(n) = \cos wn$.
6. Mention the relation between, Z transform and Fourier transform.
7. Give any two properties of linear convolution.
8. Write the commutative and distributive properties of convolution.
9. Given a difference equation $y[n]=x[n]+3x[n-1]+2y[n-1]$. Determine the system function H(z).
10. Find the stability of the system whose impulse response $h(n)=(1/2)^n u(n)$
11. Determine the z-transform and ROC for the signal $x(n)= \delta(n-k)+ \delta(n+k)$.
12. State final and initial value theorem of z-transform.
13. State and prove the time reversal property of Fourier transform.
14. Determine Z-transform of the sequence $x(n)= \{2,1,-1,0, 3\}$
15. Find the z- transform of a digital impulse signal and digital step signal
16. Why the result of circular convolution and linear convolution is not same?
17. Define convolution
18. What is the condition for stability in Z-domain?
19. Obtain the inverse Z – transform of $X(Z) = \log(1-2z)$ for $|Z| < ½$
20. Find Z- transform of $x(n) = a^n u(n)$ and its ROC

PART B

1. (i) Find the Z transform and its ROC of
   $x(n) = \left[\frac{-1}{5}\right]^n u(n) + 5\left[\frac{1}{2}\right]^{-n} u(-n-1)$ (6)

(ii) A system is described by the difference equation $\mathbf{y(n)-\left[\frac{1}{2}\right]^{n}y(n-1)=5x(n)}$. Determine the solution, when the $\mathbf{x(n)=\left[\frac{1}{5}\right]^{n}u(n)}$ and the initial condition is given by y(-1)=1, using z transform. (10)

2. (i) Determine the impulse response of the system described by the difference equation $y(n)=y(n-1)-\left[\frac{1}{2}\right]y(n-2)+x(n)+x(n-1)$ using Z transform and discuss its stability. (10)
   (ii) Find the linear convolution of x(n)={2,4,6,8,10} with h(n)={1,3,5,7,9} (6)

3. (i) Determine the Z transform of
   (1) $x(n)=a^{n}\cos w_{0}n\ u(n)$ (5)
   (2) $x(n)=3^{n}u(n)$ (3)

   (ii)Obtain x(n) for the following
   $X(z)=\frac{1}{1-1.5z^{-1}+0.5z^{-2}}$ for ROC ;|z|>1, |z|<0.5, 05<|z|<1 (8)

4. (i) Explain the properties of Z-transform. (8)
   (ii) Find the impulse response given by difference equation.
   y(n)-3y(n-1)-4y(n-2)=x(n)+2x(n -1) (8)

5. (i) Test the stability of given systems. (8)
   y(n)=cos(x(n))
   y(n)=x(-n-2)
   y(n)=n x(n)
   (ii) Find the convolution. (8)

   $x(n)=\{\underset{\uparrow}{-1},1,2,-2\}$, $h(n)=\{0.5,\underset{\uparrow}{1},-1,2,0.75\}$

6. Obtain the discrete Fourier series coefficients of $x(n)=\cos w_{0}n$.(4)
   Determine x(n) for the given x(2) with ROC
   (1) |z|>2
   (2) |z| <2

   $\boldsymbol{X(z)}=\frac{\mathbf{1+3z^{-1}}}{\mathbf{1+3z^{-1}+2z^{-2}}}$ (12)

7. Find the Z-transform and its associated ROC for the following discrete time signal
   $x[n]=\left[\frac{-1}{5}\right]^{n}u(n)+5\left[\frac{1}{2}\right]^{-n}u(-n-1)$ (8)

   Evaluate the frequency response of the system described by system function

$H(z) = \frac{1}{1-0.5z^{-1}}$ (8)

8. (i) Determine the causal signal x(n) whose z-transform is given by

   $\boldsymbol{X(z)} = \frac{\mathbf{1}}{\mathbf{1-z^{-1}+0.5z^{-2}}}$ (10)

   (ii) Determine the z-transform of the signal $x(n)=(\cos w_0 n)u(n)$. (6)

9. (i) Find the z-transform and ROC of $x(n)=r^n\cos(n\theta)u(n)$ (8)
   (ii) Find Inverse z-transform of $X(z) = z/[3z^2-4z+1]$, ROC $|z|>1$. (8)

10. (i) Determine the DTFT of the given sequence $x[n]=a^n (u(n)-u(n-8))$, $|a|<1$. (8)
    (ii) Prove the linearity and frequency shifting theorems of the DTFT. (8)

UNIT III

PART A

1. Find the 4-point DFT of the sequence x(n)={1,1}
2. What is FFT? What is it advantage?
3. Differentiate IIR and FIR filter.
4. What is the relation between DFT and Z transform?
5. Draw the butterfly diagram for DITFFT.
6. Calculate DFT of x(n)={1,1,-2,-2}.
7. Differentiate between DIF and DIT.
8. Draw the basic butterfly diagram for Radix 2 DITFFT.
9. Write the DTFT for (a) $x(n) = a^n u(n)$ (b) $x(n) = 4\delta(n) - 3\delta(n-1)$.
10. Find the discrete Fourier Transform for $\delta(n)$.
11. Draw the basic butterfly diagram for DIF algorithm.
12. Draw the butterfly diagram for decimation in time FFT algorithm.
13. State circular frequency shift property of DFT.
14. Compare DIT radix-2 FFT and DIF radix-2 FFT.
15. Define Twiddle factor and what are its properties?
16. List the properties of DFT
17. What is zero padding? What are its uses?
18. How many multiplications and additions are required to compute N-point DFT using radix – 2 FFT?
19. State and prove Parseval's relation for DFT.
20. What do you mean by the term bit reversal as applied to DFT?

PART B

1. (i) State and prove convolution property of DFT. (6)
   (ii) Find the inverse DFT of
   $X(K) = \{7, -\sqrt{2} - j\sqrt{2}, -j, \sqrt{2} - j\sqrt{2}, 1, \sqrt{2} + j\sqrt{2}, j, -\sqrt{2} + j\sqrt{2}$ (10)

2. (i) Derive the decimation-in time radix-2 FFT algorithm and draw signal flow graph for 8-point sequence. (8)
   (ii) Using FFT algorithm, compute the DFT of x(n)={2,2,2,2,1,1,1,1}. (8)

3. (i) Explain the following properties of DFT.
   (1) Convolution.
   (2) Time shifting
   (3) Conjugate Symmetry. (10)
   (ii) Compute the 4 point DFT of x(n) ={0,1, 2,3}. (6)

4. (i) Explain the Radix 2 DIFFFT algorithm for 8 point DFT. (8)
   (ii) Obtain the 8 point DFT using DITFFT algorithm for
   $$x(n) = \{\underset{\uparrow}{1},1,1,1,1,1,1,1\}$$
   (8)

5. An 8-point sequence is given by x(n)={2, 2, 2, 2, 1,1,1,1}. Compute 8-point DFT of x(n) by radix DIT-FFT method also sketch the magnitude and phase. (16)

6. Determine the response of LTI system when the input sequence is x(n)={-1,1,2,1,-1} using radix 2 DIF FFT. The impulse response is h(n)={-1,1,-1,1}. (16)

7. (i) Describe the following properties of DFT.
   (1) Time reversal
   (2) Circular convolution. (10)
   (ii) Obtain the circular convolution of
   x_1(n)= {1, 2, 2, 1}
   x_2(n) ={1, 2, 3, 1} (6)

8. 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. (16)

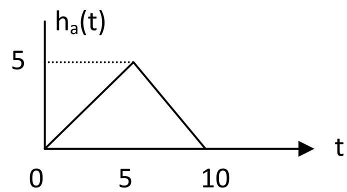
9. (i) The first five points of the eight point DFT of a real valued sequence are {0.25, 0.125 – j0.3018, 0 , 0.125 - j0.0518 , 0 }. Determine the remaining three points. (4)
   (ii) Compute the eight point DFT of the sequence x=[1,1,1,1,1,1,1,1], using Decimation-in-Frequency FFT algorithm. (12)

10. (i) Determine 8 point DFT of the sequence x(n)={1,1,1,1,1,1,0,0,0}. (12)
    (ii) Find circular convolution of the sequence using concentric circle method x_1={1,1,2,1} and x_1={1,2,3,4} (4)

UNIT IV

PART A

1. What is need for employing window for designing FIR filter?
2. What is warping effect? What is its effect on frequency response?
3. Is the given transfer function $H(z) = \frac{1+0.8z^{-1}}{1-0.9z^{-1}}$ represents low pass filter or high pass filter?
4. The impulse response of an analog filter is given in figure. Let h(n)= $h_a(nT)$ where T=1. Determine the system function.



5. Define pre-warping effect. Why it is employed?
6. Give Hamming window function.
7. State warping and give the necessity of prewarping.
8. Define the condition for stability of digital filters.
9. What is prewarping?
10. What are the special features of FIR filters?
11. What are the advantages of FIR filter?
12. Mention the significance of Chebyshev's approximation.
13. What is meant by linear phase response of a filter?
14. Compare bilinear transformation and Impulse invariant method of IIR filter design.
15. Name two methods for digitizing the transfer function of an analog filter.
16. List the properties of chebyshev filter.
17. What is bilinear transformation? What are its advantages?
18. Draw a causal FIR filter structure for length M= 5.
19. Draw the direct form – II structure of IIR filter.
20. Draw the magnitude response of 4th order Chebyshev filter.

PART B

1. (i) Obtain cascade and parallel realization for the system having difference equation y(n)+0.1y(n-1)-0.2y(n-2)=3x(n)+3.6x(n-1)+0.6x(n-2) (8)
   (ii) Design a length-5 FIR band reject filter with a lower cut-off frequency of 2KHz, an upper cut-off frequency of 2.4KHz, and a sampling rate of 8000Hz using Hamming window. (8)

2. (i) Explain impulse invariant method of designing IIR filter. (6)
   (ii) Design a second order digital low pass Butterworth filter with a cut-off frequency 3.4 KHz at a sampling rate of 8 KHz using bilinear transformation. (10)

3. Design an FIR linear phase, digital filter approximating the ideal frequency response

$$H_d(w) = \begin{cases} 1, & |w| \le \dfrac{\pi}{6} \\ 0, & \dfrac{\pi}{6} < |w| \le \pi \end{cases}$$

Determine the coefficients of a 25 tap filter based on the window method with a rectangular window. (16)

4. (i) Convert the analog filter with system function $H_a(s) = \frac{s+0.1}{(s+0.1)^2+9}$ into a digital IIR filter by means of the impulse invariance method. (8)
   (ii) Draw the direct form I and direct form II structures for the given difference equation y(n)=y(n-1)-0.5y(n-2)+x(n)-x(n-1)+x(n+2). (8)

5. Design a Chebyshev filter for the following specification using bilinear transformation.
   $0.8 \le |He^{jw}| \le 1 \qquad 0 \le w \le 0.2\pi$

   $|He^{jw}| \le 0.2 \qquad 0.6\,\pi \le w \le \pi$ (16)

6. For the analog transfer function H(s)=2/(s+1)(s+3) determine H(z) using bilinear transformation with T=0.1 sec. (16)

7. Design an ideal high pass filter with $H_d(e^{jw}) = \begin{cases} 1 & \frac{\pi}{4} \le |w| < \pi \\ 0 & |w| \le \frac{\pi}{4} \end{cases}$ using Hamming window with N=11. (16)

8. (i)Realize the following using cascade and parallel form (12)
   $$H(z) = \frac{3 + 3.6z^{-1} + 0.6z^{-2}}{1 + 0.1z^{-1} - 0.2z^{-2}}$$
   (ii) Explain how an analog filter maps into a digital filter in Impulse Invariant transformation. (4)

9. Design a butterworth filter using the Impulse invariance method for the following specifications.
   $0.8 \le |He^{jw}| \le 1 \qquad 0 \le w \le 0.2\pi$

   $|He^{jw}| \le 0.2 \qquad 0.6\,\pi \le w \le \pi$ (16)

10. Design and realize a digital filter using bilinear transformation for the following specifications.
    Monotonic pass band and stop band -3.01 dB cut off at 0.5 π rad magnitude down atleast 15dB at w=0.75 π rad. (16)

UNIT V

PART A

1. What is pipelining? What are the different stages in pipelining?
2. What is the function of parallel logic unit in DSP processor?
3. What is meant by bit reversed addressing mode? What is the application for which this addressing mode is preferred?
4. Compare the RISC and CISC processors.
5. List the various registers used with ARAU.
6. What are the different buses of TMS 320C54x processor and list their functions?
7. Define periodogram.
8. Define Gibbs phenomena.
9. Mention the features of DSP processor.
10. What is the condition for linear phase in FIR filters?
11. Define warping.
12. What is BSAR instruction? Give an example.
13. Give the special features of DSP processors.
14. What is pipelining?
15. Mention one important feature of Harvard architecture.
16. What is the advantage of pipelining?
17. Compare fixed point arithmetic and floating point arithmetic.
18. What is meant by rounding? Discuss its effects?
19. List out the features of TMS 320 C54 processors.
20. What are the 3 quantization errors due to finite word length registers in digital filters?

PART B

1. (i) Draw the block diagram of Harvard architecture and explain. (8)
   (ii) Explain the advantages and disadvantages of VLIW architecture. (8)

2. Write short notes on
   i. Memory mapped register addressing
   ii. Circular addressing mode
   iii. Auxiliary registers (6+6+4)

3. Explain various addressing modes of a digital signal processor. (16)

4. Draw the functional block diagram of a digital signal processor and explain. (16)

5. Explain Von Neumann, Harvard architecture and modified Harvard architecture for the computer. (16)

6. (i) Explain how convolution is performed using a single MAC unit. (8)
   (ii) What is MAC unit? Explain its functions. (8)

7. Explain in detail about MAC unit and pipelining. (16)
8. Explain the addressing formats and functional modes of a DSP processor. (16)
9. Explain the architecture of TMS320C50 with a neat diagram. (16)
10. Describe the Architectural details and features of a DSP processor. (16)