Scheme of Evaluation: Digital Signal Processing

20EC 603 (August-2023)

Prepared By:

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20EC603

Hall Ticket Number:

III/IV B.Tech (Regular) DEGREE EXAMINATION

July/August, 2023 **Electronics & Communication Engineering**

Sixth Semester **Digital Signal Processing**

Time: Three Hours **Maximum:** 70 Marks

Answer question 1 compulsory. **(14X1 = 14Marks)**
Answer one question from each unit.
Butterworth and Chbyshev polynomial tables are allowed. **(4X14=56 Marks)**

			CO	BL	M
1	a)	What are the advantages of DSP?	CO1	L1	1M
	b)	Give the relationship between unit step sequence and unit impulse.	CO1	L2	1M
	c)	Evaluate the summation $$\sum_{n=-\infty}^{\infty} n^2\, \delta(-n+2)$$	CO1	L2	1M
	d)	Define LTI system. Give example.	CO1	L2	1M
	e)	State the Parseval's property of DFS	CO2	L2	1M
	f)	What are the advantages of DFT over DTFT?	CO2	L2	1M
	g)	Determine the number of complex multiplications required to find the 32-point DFT of a sequence using FFT algorithm?	CO2	L3	1M
	h)	What are the advantages of Chebyshev filter over Butterworth filter?	CO3	L2	1M
	i)	What is the drawback of Impulse Invariance technique?	CO3	L2	1M
	j)	Convert analog filter with transfer function H(S) = $\frac{1}{s+1}$ into a digital filter by using bilinear transformation method with T= 1 sec.	CO3	L3	1M
	k)	Write the condition for symmetry with respect to the impulse response of a Linear phase FIR filter.	CO4	L2	1M
	l)	What is the advantage of Linear phase realization?	CO4	L2	1M
	m)	What are the applications of Multirate systems?	CO4	L2	1M
	n)	What is the purpose of a decimation filter?	CO4	L2	1M

Unit-I

2	a)	Derive the conditions for causality and stability of an LTI system.	CO1	L3	7M
	b)	Determine the convolution of following signals by means of Z-transform $x(n) = u(n)$ and $h(n) = 2^n u(n)$	CO1	L4	7M

(OR)

3	a)	State and prove any four properties of Z-transform.	CO1	L3	7M
	b)	Determine the step response of the following causal system described by the difference equation y(n) + 5y(n-1) + 6y(n-2) = x(n-1)	CO1	L3	7M

Unit-II

4	a)	Determine the Fourier Series Coefficients of a signal $x(n) = 1 + 2\cos(\pi n/3) + 3\sin(\pi n/5)$	CO2	L3	7M
	b)	Develop radix-2 Decimation-in-time FFT algorithm for N=8.	CO2	L4	7M

(OR)

5	a)	Determine the linear convolution of the following signals using circular convolution. x(n) = {3, 1, 2, 1} and h(n) = {2, 1, 3}	CO2	L3	7M
	b)	Compute the 8-point DFT of the sequence x(n) = {1, 2, 3, 4, 4, 3, 2, 1} using radix-2 DIF FFT algorithm	CO2	L4	7M

P.T.O

Unit-III

6	a)	Obtain the impulse response of digital filter corresponding to an analog filter with impulse response $h_a(t) = 0.5e^{-2t}u(t)$ and with a sampling rate of 1 kHz using Impulse Invariant method.	CO3	L3	7M
	b)	Derive the relationship between s-plane poles and z-plane poles in Bilinear transformation method.	CO3	L3	7M

(OR)

7	a)	Convert the analog filter with system function H(s) into a digital IIR filter by means of the impulse invariance method. $H(s) = \frac{s^2}{s^2+5s+4}$	CO3	L3	7M
	b)	Design a digital low pass filter is required to meet the following specifications. Pass band attenuation ≤ 1 db Stop band attenuation ≥ 40 db Pass band frequency = 4K Hz Stop band frequency = 8K Hz Sampling frequency = 24K Hz Use Butterworth approximation and bilinear transformation technique.	CO3	L5	7M

Unit-IV

8	a)	Explain the principle and procedure for designing FIR digital filters using window method.	CO4	L3	7M
	b)	Obtain cascade form and Parallel form realizations for the system described by a difference equation $y(n) = \frac{5}{6}\, y(n-1) + \frac{1}{6}\, y(n-2) + x(n) + 3\, x(n-1) + 2\, x(n-2)$	CO4	L3	7M

(OR)

9	a)	The desired response of a high pass filter is $H(e^{j\omega}) = e^{-j5\omega}$; $\pi/4 \le \lvert\omega\rvert \le \pi$ $= 0$; $-\pi/4 \le \omega \le \pi/4$ Determine $H(e^{j\omega})$ for N =11 using Hamming window.	CO4	L4	7M
	b)	Explain the concept of decimation and interpolation with examples.	CO4	L3	7M

Scheme of 20EC 603 (Aug-2023)

Q.No: 1

Answer all questions

Each question carries one mark **1x14=14M**

(a) Advantages of DSP:

- Greater accuracy,
- Inexpensive
- Ease of storage
- Flexibility
- Time sharing

(b) Impulse sequence:

$$\delta(n) = u(n) - u(n-1)$$

(c) Evaluation of Summation:

$$\sum_{n=-\infty}^{\infty} n^2 \delta(-n+2) = n^2 \big| n = 2 = 4$$

(d) LTI system

- A system that satisfies both linearity and time invariance properties. Example:

$$y(n) = x(n) + \frac{1}{x(n-1)}$$

(e) Parseval's property of DFS

$$DFS[x_1(n)x_2^*(n) = \frac{1}{N}\sum_{k=0}^{N-1} X_1(k)X_2^*(k)$$

(f) DFT Vs DTFT

- DFT is discrete in both time and frequency
- Sampled version of DTFT
- DTFT is discrete in time and Continuous in frequency domain

(g) Number of Complex multiplications:

- 80 for DIFFFT
- 160 for DITFFT

(h) Advantage of Chebyshev filter over Butterworth filter:

- For the same specifications the order of the filter is less than Butterworth filter.

(i) Disadvantage of Impulse Invariance method :

- Not suitable for the design of IIR filters other than LPF
- Many to One mapping
- Spectrum aliasing

(j) Digital filter for the given analog filter:

$$H(s) = \frac{1}{s+1}$$

$$H(z) = H(s)\Big|_{s = \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right)}$$

(k) Symmetry Condition:

(l) Less number of multiplications are sufficient to realize the filter

$$h(n) = h(N-1-n)$$

(m) Applications:

- High quality DAS
- Audio signal Processing
- Speech processing
- Narrow band filtering of Fetal ECG, EEG

(n) Decimation filter:

- To maintain good SNR

UNIT-I

Q.No:2.

(a)

- Causality condition derivation: 3M;
  $h(n) = 0 \;\; for\; n < 0$
- Stability Condition derivation: 4M;
  $$\sum_{n=-\infty}^{\infty} |h(n)| < \infty$$

(b)

- Determination of *X(z)*: 1M
- Determination of *H(z)*: 1M
- Determination of *Y(z)*: 1M
- Inverse Z-transform of *Y(z)* to obtain *y(n)*
- *y(n)* is the convolution between *x(n)* and *h(n)*

(OR)

Q.No:3.

(a)

- Statements and Proofs of any four properties of Z-transform such as Linearity, Time shifting, Time reversal, Differentiation…etc. (7M)

(b)

- For the given LCCDE
- Determination of *H(z)*: 2M
- Determination of *Y(z)* for the step input: 1M
- Inverse Z-transform of *Y(z)* to obtain *y(n)*: 4M

UNIT-II

Q.No:4.

(a)

- Determination of the period of the given sequence: 1M
- Determination of DFS co-efficients using the following relation

  $\sum_{n=0}^{N-1} x_p(n) e^{-i2\pi kn/N}$: 6M

(b)

- Decimating the 8 point DFT in to two 4-point DFTs : 2M
- Decimating each 4 point DFTs into 2 point DFTs: 2M
- Combining smaller DFTs to using combining algebra to form 8-point DFT: 3M

(OR)

Q.No:5.

(a)

- Determination of number of zeros to be padded to the given sequence: 1M
- Computation of circular convolution of zero-padded sequences using any method: 6M

(b)

- DIFFFT flow graph: 3M
- First stage values: 1M

  $\{5,5,5,5,-3,-0.707+j0.707,-j,-2.121-j2.121\}$

- Second stage values: 1M

  $\{10,10,0,0,-3-j,-2.828-j1.414,-3+j,2.828-j1.414\}$

- Final stage values: 1M

  $\{20,0,0,0,-5.828-j2.414,-0.172+j0.414,-0.172-j0.414,-5.828+j2.414\}$

- Final DFT values: 1M

$$X(k)=\{20,-5.828-j2.414,0,-0.172-j0.414,0,-0.172+j0.414,0,-5.828+j2.414\}$$

UNIT-III

Q.No:6.

(a)

- Sampling of given analog filter impulse response to obtain impulse response of digital filter: 2M
- Determination of *H(z)* for the determined *h(n)*: 5M

(b)

- Obtain differential equation corresponding to the Analog transfer function: 1M
- Convert the differential equation into difference equation:1M
- Determine the z-transform of the obtained difference equation: 2M
- Correlate this z-transform with the given transfer function to obtain the relation between S and Z

(OR)

Q.No:7.

(a)

- Resolve the given function into partial fractions: 1M
- Find the inverse Laplace transform to get *h(t)*: 1M
- Sample *h(t)* with the given sampling rate to get *h(n)*: 1M
- Find the z-transform of *h(n)* to get *H(z)*

(b)

- *Prewarp* the given digital frequencies to get equivalent analog frequencies: 1M
- Find the order of the filter using these frequencies and pass band and stop band attenuations:1M
- Write the transfer function of analog Butterworth filter: 1M
- Determine the cut-off frequency and Obtain the Analog filter transfer function:1M
- Convert this into digital filter using

$$s = \frac{2}{T}\left(\frac{1-z^{-1}}{1+z^{-1}}\right) : 3\text{M}$$

UNIT-IV

Q.No:8.

(a)

Principle: Truncate the impulse response of IIR filter using windows to get the impulse response of FIR: 1M
Design steps:

- For the desired frequency response $H_d(e^{jw})$ find the impulse response $h_d(n)$ using

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{jw})e^{jwn}dw : 2\text{M}$$

- Multiply the infinite impulse response with a chosen window sequence *w(n)* of length *N* to obtain filter co-efficients *h(n)*

$$h(n) = h_d(n)w(n) \;\; for\; |n| \le \frac{N-1}{2} \quad : 2\text{M}$$
$$= 0 \; otherwise$$

- Find the transfer function of realizable filter:

$$H(z) = z^{-\left(\frac{N-1}{2}\right)}\left[h(0) + \sum_{n=1}^{\frac{N-1}{2}} h(n)(z^n + z^{-n})\right] : 2\text{M}$$

(b)

- Determine *H(z)* :1M
- Resolve *H(z)* into partial fractions for parallel realization: 1M
- Parallel realization using hard ware elements: 2M
- Cascade realization using hardware elements: 3M

(OR)

Q.No:9.

(a)

- For the desired frequency response $H_d(e^{jw})$ find the impulse response $h_d(n)$ using

$$h_d(n) = \frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{jw})e^{jwn}dw : 2\text{M}$$

- Multiply the infinite impulse response with a chosen window sequence $w(n)$ of length N to obtain filter co-efficients $h(n)$

$$h(n) = h_d(n)w(n) \;\; for\; |n| \le \frac{N-1}{2} \quad : 2\text{M}$$
$$= 0\; otherwise$$

- Find the transfer function of realizable filter:

$$H(z) = z^{-\left(\frac{N-1}{2}\right)}\left[h(0) + \sum_{n=1}^{\frac{N-1}{2}} h(n)(z^n + z^{-n})\right] : 3\text{M}$$

- Here $w(n)$

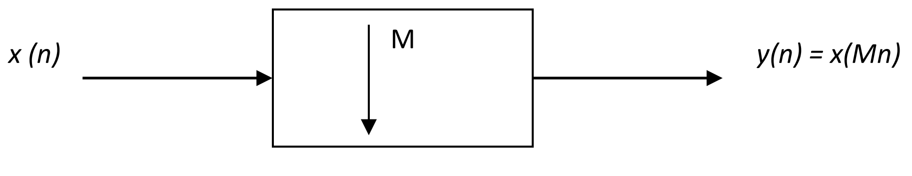
$$w(n) = 0.54 + 0.46\cos\frac{2\pi n}{N-1} \;\; for\; \frac{-(N-1)}{2} \le n \le \frac{(N-1)}{2}$$
$$= 0 \quad otherwise$$

(b)

- Down sampling process (Decimation) with examples: 3M
- Interpolation process with examples: 4M

Decimation (or) Down sampling:

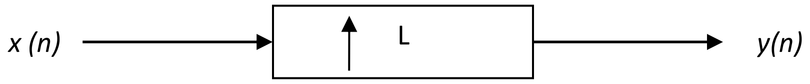
- The sampling rate of a discrete-time signal $x(n)$ can be reduced by a factor 'M' by taking every M[th] value of the signal.
- The following figure shows the block diagram representation of down sampler



- The above symbol with arrow pointing downwards is called a down sampler.
- The output signal $y(n)$ is a down sampled signal of the input signal $x(n)$ and can be represented by $y(n) = x(Mn)$.
- This process leads to potential loss of information.

Interpolation:

- The sampling rate of a discrete-time signal $x(n)$ can be increased by a factor '*L*' by placing *L-1* new values after each sample.
- The new sample values may equal to zero, which is known as Zero interpolation or Up sampler
- The new sample values may equal to previous values (Step interpolation), or linearly interpolated values.
- The following figure shows the block diagram representation of Up sampler



- The above symbol with arrow pointing upwards is called an up sampler.
- The output signal $y(n)$ is a up sampled signal of the input signal $x(n)$ and can be represented by

$$y(n) = \begin{cases} x\left(\frac{n}{L}\right) & n = 0, \pm L, \pm 2L, \cdots\cdots \\ 0 & Otherwise \end{cases}$$

- Decimation is the inverse interpolation but not the other way around
- If a signal $x(n)$ is interpolated by '*N*' and then decimated by '*N*', we recover the original signal $x(n)$
- If a signal $x(n)$ is decimated by '*N*' and then interpolated by '*N*', we may not recover the original signal $x(n)$
- If both interpolation and decimation are required, it is better to interpolate first.