Chettinad College of Engineering & Technology
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Academic Year 2024 -2025
Question Bank

Year/Semester:II / IV **Date:19.02.2025**	**Department: ECE** **Subject Code/Title : EC3492 / Digital Signal Processing** **Faculty Name:M.Prabhakaran**	**Unit** : I/~~II/III/IV/V~~ **Section :Part A/B/C**

UNIT I
DISCRETE FOURIER TRANSFORM
PART-A

1. Define Sampling Theorem. [R]

A continuous time signal can be completely represented in its samples and recovered back if the sampling frequency Fs ≥ 2B. Here Fsis the sampling frequency and B is the maximum frequency present in the signal.

2. What is the relationship between z-transform and DFT? [U]

If the z-transform is evaluated on unit circle at evenly spaced points only then it becomes DFT.

$$X(k) = X(z)|_{z_{k=e^{j2\pi k/N}}}$$

3. What is FFT? [R]

Special algorithms are developed to compute DFT quickly. These algorithms exploit the periodicity and symmetry properties of twiddle factors. Hence DFT is computed fast using such algorithms compared to direct computation. These algorithms are collectively called as Fast Fourier Transform (FFT) algorithms. The algorithms are very efficient in terms of computations.

4. What are the advantages of FFT algorithm over direct computation of DFT? [U]

- FFT requires less number of multiplications and addition compared to direct computation of DFT.
- FFT algorithm can be implemented fast on the DSP Processor.
- The calculation of DFT and IDFT both are possible by proper combination of FFT algorithms.

5. Compute the number of multiplications needed in the FFT computation of DFT of a 32 point sequence. [U]

Number of complex multiplications needed in 32 point FFT are

$$\frac{N}{2}log_2N = \frac{32}{2}log_2 32 = 80$$

6. How many multiplications and additions are required to compute the N point DFT using radix-2 FFT? [U]

For N-point DFT using radix-2 DIT or DIF FFT

1. Number of complex multiplications :$\frac{N}{2}log_2N$
2. Number of complex additions : $Nlog_2N$.

7. Compare the number of multiplications required to compute the DFT of a 64 point sequence using direct computation and that using FFT. [U]

Number of complex multiplications needed in 64 point Direct Computation

N^2=4096

Number of complex multiplications needed in 64 point FFT

$$\frac{N}{2}log_2N = 192$$

8.Draw the basic structures of DIT and DIF-FFT flowchart of radix-2. [R]

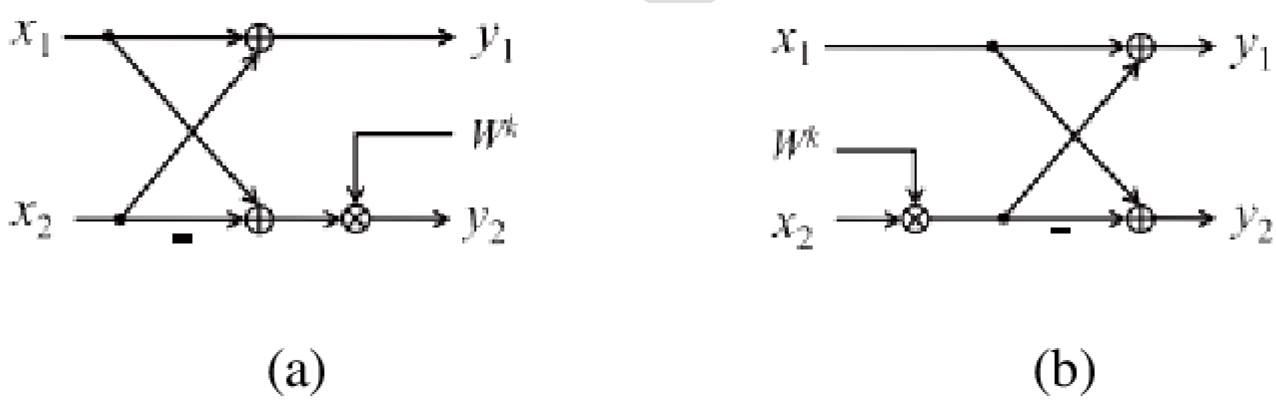

Figure 1: (a) DIF FFT butterfly (b) DIT FFT butterfly

9. Write the equation for N-point DFT and IDFT? [R]

DFT of the of a sequence is x(n)

$$X(k) = \sum_{n=0}^{N-1} x(n)e^{\frac{-j2\pi kn}{N}}, k = 0,1, \dots \dots . . N-1.$$

IDFT of the sequence X(k) is

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{\frac{j2\pi kn}{N}}, n = 0,1, \dots \dots . . N-1.$$

10. Find the DFT of the sequence x(n)={1,1,0,0} [U]

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{\frac{-j2\pi kn}{N}}, k = 0,1, \dots\dots..N-1$$

$$X(k) = \sum_{n=0}^{3} x(n) e^{\frac{-j\pi kn}{2}}, k = 0,1,23$$

$$X(k) = \{2,1-j,0,1+j\}$$

11. State and prove Perceval's relation for DFT. [U]

Perceval's relation for DFT is given as

$$\sum_{n=0}^{N-1} x(n).y^{*}(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)Y^{*}(k)$$

12. Compare radix 2 DIT, DIF FFT algorithm. [U]

S.No	DIT-FFT	DIF-FFT
1	The time domain sequence is decimated	The DFT X(k) is decimated
2	Input sequence is to be given in bit reversed order.	The DFT at the output is in bit reversed order.
3	First calculates 2 point DFT and combines them	Decimates the sequence step by step to 2 point sequence and calculates DFT
4	Suitable for calculating the IDFT	Suitable for calculating DFT.

13. Distinguish between Overlap save and Overlap add methods. [U]

S.No	Parameter	Overlap Save Method	Overlap Add method
1	Data block size	L+M-1	L
2	Combination of data block	(M-1) data samples of previous sequence and next L samples of x(n)	Next L samples of x(n) and (M-1) zeros are padded at end.
3	Formation of DFT	Initial (M-1) output samples are discarded because of overlap.	Last (M-1) samples of current output sequence are added to initial (M-1) samples of succeeding output sequence.
4	Aliasing	Present	Absent

14. What is zero padding? What are its uses? [R]

When the length of the sequence is to be increased, zeros are inserted as samples. This doesnot change meaning. For example

x(n)={1,1,1} N=3

x(n){1,1,1,0,0,0,0,0} N=8

Thus the length of the sequence is increased from 3 to 8.

Zero padding is used in

1.Calculation of DFT and FFT.

2.Linear filtering

3.Linear convolution using circular convolution.

15. Compute the IDFT of y(k){1,0,1,0} [U]

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} X(k)e^{\frac{j2\pi kn}{N}}, n = 0,1, \ldots\ldots..N-1$$

x(n)={0.5,0,0.5,0}

16. The first five DFT values for N=8 as follows. $\mathbf{X(k) = \{28, -4 + j9.656, -4 + j4,\ -4 + j1.656, -4, \ldots\}}$. Compute the rest of three DFT values. [U]

The first 5 DFT values are given. Remaining three values are

X(5) = -4-j1.656

X(6) = -4-j4

X(7) = -4-j9.656

PART-B & C

1. Compute the 8-Point DFT of the sequence x(n) = {1/2,1/2,1/2,1/2,0,0,0,0} by using the in-place radix-2 DIT FFT algorithm.

2. Compute the 8-Point DFT of the sequence x(n) = {1,2,3,4,4,3,2,1} by using the in- place radix-2 DIT FFT algorithm.

3. Find DFT for {1,1,2,0,1,2,0,1} using FFT DIT butterfly algorithm and plot the spectrum.

4. Determine the 8 point DFT of the sequence X(n)={0,0,1,1,1,0,0,0}

5. Compute the 8-Point DFT of the sequence x(n) = {1,1,1,1,1,1,1,1} by using the in- place radix-2 DIF FFT algorithm.

6. Find the 8-point DFT of the Sequence x(n)= {1,1,1,1,1,1,0,0} using DIF FFT algorithm.

7. Develop 8 point DIT FFT algorithm. Draw the signal flow graph .Determine the DFT of the following sequence x(n)={1,1,1,1,0,0,0,0} using signal flow graph. Show all the intermediate results on the signal flow graph.

8. Calculate the 4 point DFT of the sequence x(n)={0,1,0,-1}

9. 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 (i)Over lap save (ii)Overlap add method

10. Compute the linear convolution of finite duration sequence x(n)={1,-1,2,1,2,- 1,1,3,1} and

h(n)={1,2,1}by overlap save and overlap add methods.

11. Compute the linear convolution of finite duration sequence h(n)={1,2} and x(n)={1,2,-1,2,3,-2,-3,-1,1,1,2,-1}by overlap save and overlap add methods.

12. Find the output response of the given input sequence x(n)={1,1,-1,-1} and h(n)={1,-1,2,1} using DFT and IDFT method.

13. Find linear convolution using overlap–add and overlap-save method of the following sequences $x(n) = 1,2,-1,2,3,-2,-3,-1,1,1,2,-1$ and $h(n) = \{1,2,3\}$. Compare the results and state the usage of each method.

14. Compute IDFT of the sequence
$X(k) = \{20, -5.828 - j2.414, 0, \ -0.1716 - j0.4142, 0, -0.1716 + j0.414, 0, -5.828 + j2.414\}$ using DIT Algorithm.

Subject Incharge **Head of the Department**
(M.Prabhakaran) **(Dr.M.Kumar)**

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UNIT II
INFINITE IMPULSE RESPONSE
PART-A

1. What are the properties of bilinear transformation? [U]

Bilinear transformation is given as

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

- Right hand side of s-plane is mapped outside of the unit circle.
- Left hand side of s-plane is mapped inside of the unit circle
- jΩ axis in s-plane is mapped on unit circle of the s-plane
- Bilinear transformation maps poles as well as zeros

2. State the difference between analog filter and digital filter. [U]

S.No	Analog Filter	Digital Filter
1	Differential Equations are used	Difference Equations are used
2	It is constructed using Passive or active elements	It is constructed based on adder, multiplier and delay elements.
3	Multiple operation is very difficult	Multiple operation using single filter is possible
4	Frequency response of an analog filter can be modified by changing the components.	Frequency response can be modified by changing the filter coefficient.

3. Determine the order N of the following specifications. $\alpha_p = 1$ dB, $\alpha_s = 30$ dB, $\Omega_p = 200$rad/sec, $\Omega_s = 600$rad/sec. [U]

$$N \geq \frac{log\sqrt{\frac{10^{0.1\alpha_s}}{10^{0.1\alpha_p}}}}{log\frac{\Omega_s}{\Omega_p}}$$

N>= 3.7583

N=4

4. List the different types of filter based on frequency response. [U]

Depending upon the frequency response the filters are classified as

(i) Low pass filter

(ii) High pass filter

(iii) Band pass filter

(iv) Band Stop Filter

5. Give any two properties of Butterworth filter and Chebyshev filter. [R]

Butterworth filter

1. Monotonically reducing magnitude squared response.
2. All poles lie on the circle of radius Ω_C

Chebyshev Filters

1. Ripples in the passband or stopband
2. All poles lie on the ellipse

6. What is frequency warping? [U]

The relation between the continuous time frequency and secrete time frequency is given as

$$\omega = 2\tan^{-1}\frac{\Omega T}{2}$$

Note that this relationship is highly non linear. It maps entire range of Ω only in $-\pi \leq \omega \leq \pi$. Such nonlinear relationship is called frequency warping.

7. What is the relationship between analog and digital frequency in impulse invariant transformation? [U]

The relationship between analog and digital frequencies is given as

$$\omega = \Omega T$$

This relationship shows that the mapping of jΩ axis is many to one on unit circle. Therefore there is aliasing in frequency domain.

8. List the various forms of realization of IIR system. [R]

- Direct form 1
- Direct from II
- Cascade realization
- Parallel form realization
- Lattice realization

9. Convert the following analog transfer function into digital using impulse invariant mapping with T=1sec. $H(s) = \frac{3}{(s+3)(s+5)}$ [A]

$$H(s) = \frac{3}{(s+3)(s+5)} = \frac{3/2}{(s+3)} - \frac{3/2}{(s+5)}$$

Here use $\frac{1}{s-p_k} \rightarrow \frac{1}{1-e^{p_k T} z^{-1}}$

$$H(z) = \frac{3}{2}[\frac{1}{1-0.05z^{-1}} - \frac{1}{1-0.0067z^{-1}}]$$

10. What are the limitations of impulse invariant mapping technique? [R]

1. Frequency mapping is many to one. Therefore aliasing takes place in frequency domain.
2. Impulse invariant technique is suitably only for lowpass and narrow bandpass filters.

11. What is impulse invariant mapping? What is its limitation? [U]

The impulse invariant mapping is given as

$$\frac{1}{s-p_k} \rightarrow \frac{1}{1-e^{p_k T} z^{-1}}$$

It maps

(i) LHS of s-plane inside the unit circle

(ii) RHS of s-plane outside the unit circle

(iii) jΩ axis on unit circle

12. What are the parameters that can be obtained from the Chebyshev filter specification? [R]

From the given Chebyshev filter specifications we can obtain the parameters like the order of the filter N, Ɛ, transition ratio k and the poles of the filter.

13. Give the expression for location of poles of normalized Butterworth filter. [R]

The poles of the butterworth filter is given by

$$s_k = e^{j\emptyset_k}, \text{ k=1,2,……N}$$

$$\emptyset_k = \frac{\pi}{2} + \frac{(2k-1)\pi}{2N}$$

N is order of the filter.

14. What is the advantage of direct form II realization when compared to direct form I realization? [R]

- Direct form II realization requires less memory
- Computation time is reduced in direct form II

15. What are the disadvantages of direct form realization? [R]

Direct form structure is sensitive to parameter quantization. For large value of N the location of poles and zeroes is shifted from their actual values in case of direct form realization.

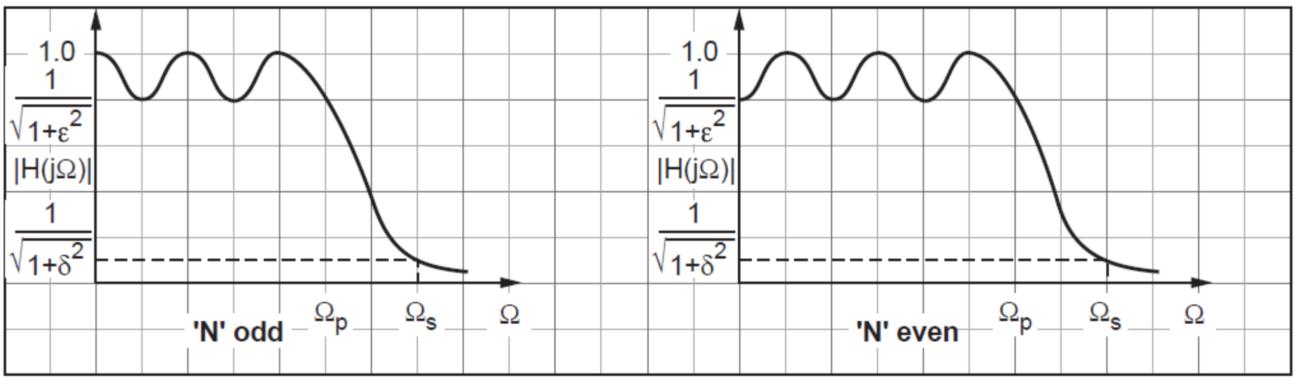
16. Distinguish between Butterworth and Chebyshev filter. [U]

S.No	Parameter	Butterworth Filter	Chebyshev Filter
1	Frequency response	Monotonically decreasing	Ripples in passband and monotonic in stopband
2	Order for given set of specifications	Higher than Chebyshev	Lower than Butterworth
3	Transition band	Transition band is broader than chebyshev for given order	Transition band is narrower than bandwidth for given order
4	Phase response	Fairly linear phase response. It is better than chebyshev.	Relatively nonlinear phase response. It is inferior to butterworth filter.

17. Why impulse invariant method is not preferred in the design of highpass IIR filters? [U]

In impulse invariant method the segments of $\frac{(2k-1)\pi}{T} \leq \Omega \leq \frac{(2k+1)\pi}{T}$ are mapped on the unit circle repeatedly. Hence first set is i.e $-\frac{\pi}{T} \leq \ \Omega \leq \frac{\pi}{T}$ is mapped correctly. Then $\frac{\pi}{T} \leq \ \Omega \leq \frac{3\pi}{T}$ is mapped on the same circle. Thus one point on the circle represents multiple analog frequencies. Hence high frequencies are actually mapped as low frequencies. Therefore all high frequencies are aliased frequencies. Hence impulse invariant technique is not much suitable for high pass filters. But for lowpass filters it is better since actual mapping take place,

18. Sketch the frequency response of even / odd ordered Chebyshev lowpass filter. [U]



19. List the various forms of realizations of IIR system. [R]

1. Direct form I
2. Direct form II
3. Cascade realization
4. Parallel form realization
5. Lattice realization

20. Mention advantages of direct form II and cascade structure. [U]

1. Direct form II structures requires less number of storage locations
2. Cascade structures are easy to implement since second order sections are simply cascaded.

21. Draw the direct form I structure for the system y(n)=0.5x(n)+0.9y(n-1) [A]

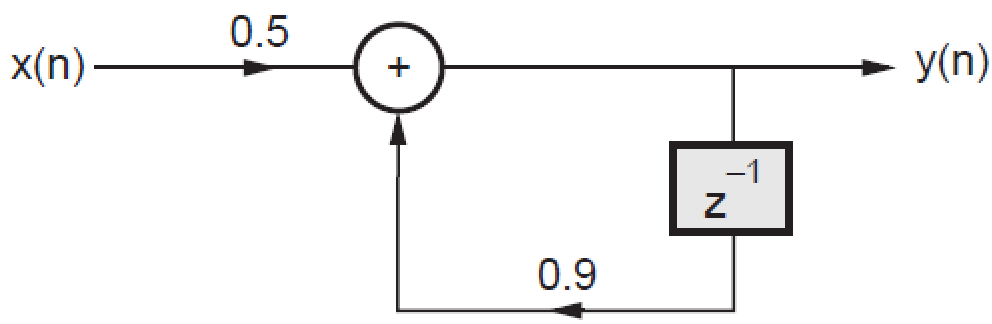


Fig. 2.6.2 Direct form-I structure

22. What are the properties that are maintained same in the transfer of analog filter into digital filter? [R]

1. **Stability:** A stable analog filter is converted to stable digital filter.
2. **Causality:** A causal analog filter is converted to causal digital filter.

23. Sketch the mapping of s-plane and z-plane in bilinear transformation. [R]

Figure shows the mapping of s-plane and z-plane

i) Left hand side of s-plane is mapped inside unit circle

ii) Right hand side of s-plane is mapped outside unit circle

iii) jΩ axis in s-plane is mapped on unit circle.

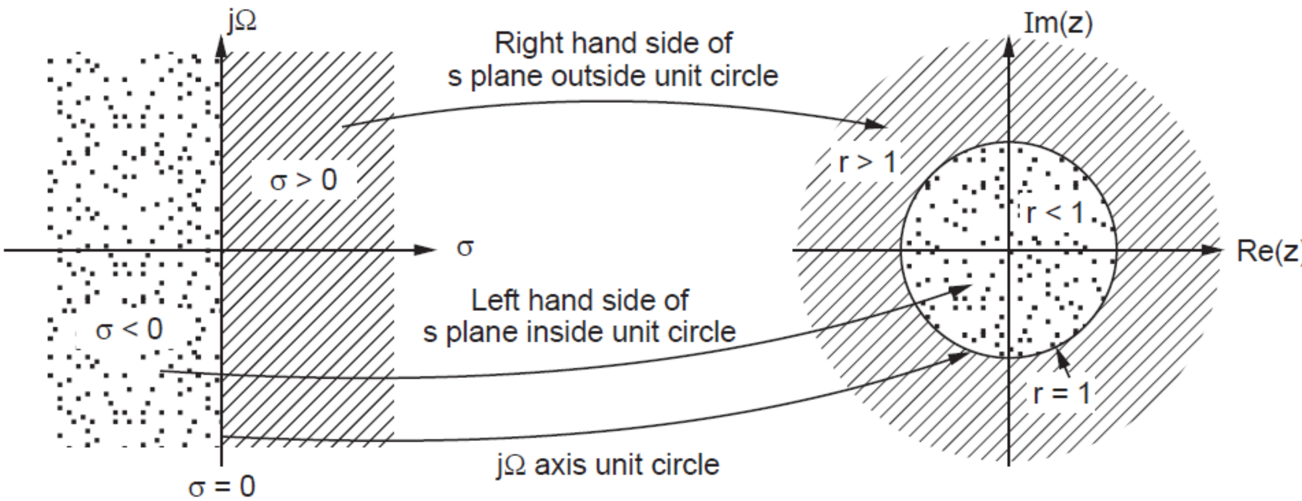


Fig. 2.6.6 Maping of s-plane and z-plane

24. Convert $H(s) = \frac{1}{S^2+1}$ into a digital filter using approximation of derivates with T=0.1 sec. [A]

Approximation of derivates is obtained by putting $s = \frac{1-z^{-1}}{T}$

$$H(z)=H(s)\Big|_{s=\frac{1-z^{-1}}{T}}=\frac{1}{(\frac{1-z^{-1}}{T})^2+1}=\frac{0.01}{1.01-2z^{-1}+z^{-2}}$$

25. What is canonic structure? [U]

If the number of delays in the structure is equal to order of the difference equation or order of the transfer function then it is called canonic form realization.

26. Compare bilinear transformation and impulse invariant method of IIR filter design. [U]

S.No	Impulse Invariant Method	Bilinear Transformation
1	Only poles of H(s) are mapped	Both poles and zeros of H(s) are mapped
2	Aliasing of frequencies takes place	No aliasing since mapping is one to one
3	Linear frequency relationship	Nonlinear frequency relationship

PART-B & C

1. Describe the concept of impulse invariance method of designing IIR filters
2. Explain in detail the steps involved in the design of IIR filters using bilinear transform method.
3. Use the Impulse invariance method to design a digital filter from an analog prototype that has a system function $H_a(s) = \frac{s+a}{s+a^2+b^2}$
4. Explain the procedure for designing analog filters using the chebyshev approximation.
5. Obtain the direct form –I, direct form-II, cascade and parallel form realization for the following system. y(n) = -0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2).
6. Determine the direct form I ,direct form II ,Cascade and parallel structure for the system y(n)=-0.1y(n-1)+0.72y(n-2)+0.7x(n)-0.25x(n-2).
7. Draw the structure for the IIR filter in direct form-II for the following transfer function. $H(z)=(2+3z^{-1})(4+2z^{-1}+3z^{-2})/(1+0.6z^{-1})(1+z^{-1}+0.5z^{-2}$
8. Obtain the cascade and parallel realizations for the system $H(z)=1+1/4z^{-1}/(1+1/2z^{-1})(1+1/2z^{-1}+1/4z^{-2})$
9. Obtain the cascade form realization of the digital system y(n)=3/4y(n-1)-1/8y(n-2)+1/3x(n-1)+x(n) .
10. Design a digital butterworth filter satisfying the constraints

$$0.707 \leq \left|H\left(e^{j\omega}\right)\right| \leq 1 \quad for\ 0 \leq \omega \leq \frac{\pi}{2}$$

$$\left|H\left(e^{j\omega}\right)\right| \leq 0.2 \quad for\ \frac{3\pi}{4} \leq \omega \leq \pi$$

With T=1 sec using (a) The Bilinear Transformation (b) Impulse invariance. Realize the filter in each case using the most convenient realization form.

11. Design a Butterworth LPF which has the following specifications using impulse invariant transformation

$$0.7 \leq |H(j\Omega)| \leq 1 \qquad \text{for } 0 \leq \Omega \leq 0.2\pi$$

$$|H(j\Omega)| \leq 0.3 \quad \text{for } 0.6\pi \leq \Omega \leq \pi$$

12. Design a Butterworth LPF which has the following specifications using impulse invariant transformation

$$0.8 \leq |H(j\Omega)| \leq 1 \qquad \text{for } 0 \leq \Omega \leq 0.25\pi$$

$$|H(j\Omega)| \leq 0.2 \quad \text{for } 0.75\pi \leq \Omega \leq \pi$$

13. Design a Butterworth LPF which has the following specifications using impulse invariant

transformation

$$0.8 \le |H(j\Omega)| \le 1 \qquad \text{for } 0 \le \Omega \le 0.2\pi$$

$$|H(j\Omega)| \le 0.2 \quad \text{for } 0.6\pi \le \Omega \le \pi$$

14. Design a Chebyshev filter to satisfy the constraints

$$0.8 \le |H(j\Omega)| \le 1 \qquad \text{for } 0 \le \Omega \le 0.25\pi$$

$$|H(j\Omega)| \le 0.2 \quad \text{for } 0.75\pi \le \Omega \le \pi$$

By using Impulse Invariant Technique. Assume T=1sec.

15. Design a butterworth lowpass filter with the specifications α_p=0.5dB ripple in the passband $0 \le \omega \le 0.25\pi$, α_s=20dB ripple in the stopband $0.4\pi \le \omega \le \pi$ using a)Bilinear Transformation b) Impulse Invariance.
16. Obtain the Direct form I, Direct form II, Cascade and Parallel form realization for the system

$$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$$

17. Design a Chebyshev digital filter using IIM for the following constraints

$$0.9 \le |H(j\Omega)| \le 1 \qquad \text{for } 0 \le \Omega \le 0.25\pi$$

$$|H(j\Omega)| \le 0.24 \quad \text{for } 0.5\pi \le \Omega \le \pi$$

18. Convert the analog filter with $H(s) = \frac{s+0.1}{s+0.1^2+9}$ into a digital IIR filter using bilinear transformation. The digital filter should have resonant frequency $\omega_r = \frac{\pi}{4}$
19. Describe a chebyshev filter with specifications α_p=1 dB ripple in passband $0 \le \omega \le 0.2\pi$, α_s=1 dB ripple in stopband α_p=1 dB $0.3\pi \le \omega \le \pi$ using Bilinear transformation.

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UNIT III
FINITE IMPULSE RESPONSE
PART-A

1. What is windowing and why it is necessary? [U]

Unit sample response of the desired filter is obtained from frequency response $H_d(w)$. This unit sample response is normally infinite in length. This truncation creates oscillations in passband and stopband of the filter. This problem can be avoided with windowing. The desired unit sample response is multiplied with suitable window. The length of the window can be selected to desired value. Due to windowing the unit sample response of the filter is reshaped such that ringing (oscillators) are reduced.

2. What are the desirable characteristics of window? [U]

i) The length of the window should be as large as possible.

ii) The width of the main lobe should be as small as possible.

iii) The amplitudes of side lobes should be very small.

3. What is meant by FIR filter and why it is stable? [U]

FIR filter: Its impulse response has finite length

Stability: The output of FIR filter is given as

$y(n) = b_0 x(n) + b_1 x(n-1) + \cdots \ldots \ldots + b_M x(n-M+1)$. Here h(n)={$b_0$,$b_1$, ……$b_{M-1}$}.

Above equation shows that output y(n) is bounded as long as inputs are bounded. This mean FIR filter is inherently stable.

4. What are the disadvantages of FIR filters? [R]

i) FIR filters need higher order compared to IIR filter.

ii) Processing time is more in FIR filter.

iii) FIR filters need more memory
iv) FIR filters are all zero filters.

5. What are the advantages of FIR filters? [R]

i) FIR filter have linear phase characteristic
ii) FIR filters are inherently stable
iii) The design of FIR filters is fairly simple compared to IIR filters.

6. Distinguish between FIR and IIR filters. [U]

S.No	FIR Filter	IIR Filter
1	Unit impulse response has infinite duration	Unit impulse response has finite duration
2	These are all poles or poles-zeros filters	These are all zero filters
3	These filters can be unstable if due care is not taken	These filters are inherently stable.

7. What is the reason that FIR filter is always stable? [U]

FIR filters are inherently stable. They are all zero filters. The poles of FIR filters are located at the origin in z-plane, This mean FIR filter poles are always inside the unit circle. Hence FIR filters are always stable.

8. Write the design steps involved in FIR filter design. [U]

i) From the given frequency responses calculate required order of the filter.
ii) From the order and desired frequency response calculate desired unit sample response $h_d(n)$.
iii) From the attenuation characteristics select the suitable window function w(n).
iv) Calculate $h(n)= h_d(n) . w(n)$

9. What condition on the FIR sequence h(n) is to imposed in order that this filter can be called as linear phase filter? [U]

The phase of FIR filter is linear if unit sample response satisfies following condition

$$h(n) = \pm h(m - 1 - n)$$

In other words unit sample response must be symmetric or antisymmetric.

10. What are the techniques of designing FIR filter? [U]

There are three well known methods for designing FIR filters with linear phase. These are

i) Windows method
ii) Frequency sampling method
iii) Optimal or minimax design

11. State the condition for a digital filter to be causal and stable. [R]

A digital filter is causal if its impulse response h(n)=0 for n < 0.

A digital filter is stable if its impulse response is absolutely summable.

$$\sum_{n=-\infty}^{\infty} |h(n)| < \infty$$

12. What is Gibbs phenomenon? [R]

One possible way to find an FIR filter that approximate $H(e^{j\omega})$ would be to truncate the infinite Fourier series at $n = \pm\left(\frac{N-1}{2}\right)$. Abrupt truncation of the series will lead to oscillation both in passband and in stopband. This phenomenon is known as Gibbs phenomenon.

13. What are the properties of FIR filter? [U]

i) FIR filter is always stable
ii) A realizable filter can always be obtained
iii) FIR filter has a linear phase response

14. Give the equation specifying the Hanning window? [R]

The equation for hanning window is

$$\omega_{Hn}(n) = 0.5 + 0.5\cos\frac{2\pi n}{N-1} \quad for\ \frac{-(N-1)}{2} \le n \le \frac{(N-1)}{2}$$
$$= 0 \quad \text{otherwise}$$

15. Give the equation specifying the Hamming window? [R]

The equation for hanning window is

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

16. For what type of filters frequency sampling method is suitable? [U]

Frequency sampling method is suitable for filter that required the filtering only at particular frequencies. Such filters are narrowband frequency selective filters where only few samples of frequency response are nonzero.

PART-B

1. What is a linear phase filter? What are the conditions to be satisfied by the impulse response of an FIR system in order to have linear phase.
2. Design a FIR linear phase digital filter approximating the ideal frequency response
$$H_d(e^{j\omega}) = e^{-j5\omega}, -\pi/2 \le |\omega| \le \pi/2$$
$$= 0,\ \pi/2 \le |\omega| \le \pi$$
using Hamming window method with N=11
3. Design an FIR lo pass digital filter by using the frequency sampling method for the following specifications cutoff frequency=1500 Hz Sampling frequency=15000Hz, Order of the filter=10, Filter length required L=N+1=11.

4. Design a digital FIR band pass filter with lower cutoff frequency 2000 Hz and upper cut off frequency 3200 Hz using Hamming window of length N=7,Sampling rate is 10000 Hz.
5. Explain the Type-I and Type-II frequency sampling method of designing an FIR filter.
6. The desired frequency response of a LPF is given by

$$H_d(e^{j\omega}) = e^{-j3\omega}, -3\pi/4 \leq |\omega| \leq 3\pi/4$$
$$= 0,\ 3\pi/4 \leq |\omega| \leq \pi$$

   Design a filter for N=7 using Hanning, Hamming window
7. The desired frequency response of a LPF is given by

$$H_d(e^{j\omega}) = e^{-j2\omega}, \qquad \pi/4 \leq |\omega| \leq \pi/4$$
$$= 0,\ \pi/4 \leq |\omega| \leq \pi$$

   Determine the filter coefficeints h(n), if h(n)=h_d(n).w(n) using hamming window. Determine the reponse H(w).
8. Design an ideal highpass filter with a frequency reponse

$$H_d(e^{j\omega}) = 1, \qquad \pi/4 \leq |\omega| \leq \pi$$
$$= 0,\ |\omega| \leq \pi/4$$

   Find the values of h(n) for N=11 using hamming window. Find H(z) and determine magnitude response.
9. Design an ideal bandpass digital FIR filter with desired frequency response

$$H_d(\omega) = 1, \qquad 0.25\pi \leq |\omega| \leq 0.75\pi$$
$$= 0,\ |\omega| \leq 0.25\pi \textit{ and } 0.75\pi \leq |\omega| \leq \pi$$

   By using Hamming window function of length N=11.
10. Determine the coefficients {h(n)} of a linear phase FIR filter of length M=15 which has a symmetric unit sample response and a frequency response that satisfies the condition

$$H\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, \text{for k} = 0,1,2,3 \\ 0, \text{for k} = 4,5,6,7 \end{cases}$$

11. Determine the coefficient of a linear phase FIR filter of length M=15 which has a symmetric unit sample response. The frequency response satisfies the function

$$H\left(\frac{2\pi k}{15}\right) = \begin{cases} 1, \text{for k} = 0,1,2,3 \\ 0.4\ for\ k = 4 \\ 0, \text{for k} = 5,6,7 \end{cases}$$

12. Obtain the linear phase realization of the system function

$$H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$$

Subject Incharge **Head of the Department**
(M.Prabhakaran) **(Dr.M.Kumar)**

HoDRemarks:

Academic Year 2024 -2025		
Question Bank		
Year/Semester: II / IV **Date:**19.02.2025	**Department: ECE** **Subject Code/Title : EC3492 / Digital Signal Processing** **Faculty Name : M.Prabhakaran**	**Unit** ~~: I/II/ III~~ /IV/~~V~~ **Section :Part A/B/C**

UNIT IV

FINITE WORD LENGTH EFFECTS

PART-A

1. What do you understand by a fixed-point number? [U]

In fixed point arithmetic the position of the binary point is fixed. The bit to the right represents the fractional part of the number and those to the left represent the integer part. For example the binary number 01.1100 has the value 1.75 in decimal.

2. Write an account on floating point arithmetic? [R]

In floating point representation a positive number is represented as F = 2c.M where M, called mantissa is a function such that $\frac{1}{2} \leq M < 1$ and C exponent can be either positive or negative. The decimal number 2.25, 0.75 have floating point representation as

$$2.25 = 2^{2} \text{ x } 0.5625 = 2^{010} \text{ X } 0.1001$$

And

$$0.75 = 2^{0} \text{ X } 0.75 = 2^{000} \text{ X } 0.1100$$

Respectively.

Negative floating point numbers are generally represented by considering the mantissa as a fixed point number the sign of the floating point number is obtained from the first by bit of mantissa.

3. Express the fraction 7/8 and -7/8 in sign magnitude, 2s complement and 1s complement. [A]

Fraction 7/8 = (0.111)2 in sign magnitude , 1s complement 2s complement

Fraction (-7/8)= (1.111) 2 in sign magnitude

=(1.000)2in 1s complement

=(1.001)2in 2s complement

4. Compare the fixed point and floating point arithmetic. [U]

S.No	Fixed point arithmetic	Floating point arithmetic
1.	Fast operation	Slow operation
2.	Relatively economical.	More expensive because of costlier hard ware
3.	Small dynamic range	Increased dynamic range
4.	Round off errors occur only for addition	Round off errors can occur with both addition and multiplication
5.	Overflow occur in addition	Over flow does not arise
6	Used in small computers	Used in larger general purpose computers

5. What are the three quantization errors due to finite word length registers in digital filters? [R]

1) Input quantization error
2) Coefficient quantization error
3) Product quantization error.

6. What is product quantization error (or) what is product roundoff error in digital signal processing? [U]

Product quantization errors arise at the output of a multiplier multiplication of a B bit data with a B bit coefficient result a product having 2 B bits since a B bit register is used the multiplier output must be rounded or truncated to B bits which produces an error this error is known as product quantization error.

7. What do you understand by input quantization error? [U]

In digital signal processing, the continuous signals are converted into digital using B – bit ADC. The representation of continuous signal amplitude by fixing digit produces an error. This is known as input quantization error.

8. What are the different quantization methods? [R]

The common methods of quantization's are
1. Truncation 2. Rounding

9. What is meant by (zero-input) limit cycle oscillation? [U]

For an IIR filter implemented with infinite precision arithmetic the output should approach zero in the steady state if the input is zero and it should approach a constant value if the input is a constant. However with an implementation using finite length register an output can occur even with zero input if there is a non zero initial condition on one of the registers. The output may be a fixed value or it may oscillate between finite positive and negative values. This effect is referred to as (zero input) limit cycle oscillations and is due to the nonlinear nature of the arithmetic quantization.

10. What is overflow oscillations? [R]

The addition of two fixed point arithmetic numbers cause overflow when the sum exceeds the word size available to store the sum. This overflow caused by adder make the filter output to oscillate between maximum amplitude limits Such limit cycles have been referred to as overflow oscillations.

11. What is meant by saturation arithmetic? What is its disadvantage? [R]

When the sum of two fixed point numbers exceeds the dynamic range, over- flow occurs, which causes the output of adder to oscillate between maximum amplitude limits. Such limit cycle has been referred to as overflow oscillations One way to avoid the overflow is to modify the adder characteristics so that it performs saturation arithmetic. Thus when an overflow is sensed, the sum of the adder is set equal to the maximum value. But saturation arithmetic causes undesirable signal distortion due to non-linearity in the adder.

12. Determine "dead band" of the filter. [R]

The limit cycle occurs as a result of quantization effect in multiplication. The amplitude of the output during a limit cycle are confined to a range of values called the deadband of the filter.

13. Explain briefly the need for scaling in the digital filter implementation. [R]

To prevent overflow, the signal level at certain points in the digital filters must be scaled so that no overflow occurs in the adder.

14. What are the methods used to prevent overflow? [R]

There are two methods used to prevent overflow

1) Saturation arithmetic
2) Scaling

15. Why rounding is preferred to truncation in realizing digital filter? [U]

1) The quantization error due to rounding is independent of the type arithmetic
2) The mean of rounding error is zero
3) The variance of the rounding error signal is low.

PART-B

1. Explain the various formats of the fixed point representation of binary numbers

2. What is meant by finite word length effects on digital filters? List them.

3. Represent the following number in floating point format with five bits mantissa and three bits for exponent:(1)7_{10} (2)0.25_{10} (3) -7_{10} (4) -0.25_{10}

4. Explain fixed and floating point representation in detail.

5. Discuss in detail the errors resulting from rounding and truncation.

6. Explain the quantization process and errors introduced due to quantization.

7. Derive the signal to quantization noise ratio of A/D converter.

8. Derive the expression for quantization noise power.

9. A cascaded realization of two first order digital filters are given. Draw the product quantization noise model of the system & determine the overall noise power.
$H_1(z) = \frac{1}{1-0.9\,z^{-1}}$ and $H_2(z) = \frac{1}{1-0.8\,z^{-1}}$

10. Consider a first order IIR filter with difference equation y(n)=x(n)+ay(n-1) ; a=1/2 & Input

$$x(n) = \begin{cases} 0.875; n = 0 \\ 0 \,; otherwise \end{cases}$$

Round to 3 bits plus sign bit. Calculate the limit cycle oscillation.

11. Consider a second order system function $H(z) = \frac{1}{(1-0.5\,z^{-1})(1-0.45\,z^{-1})}$. Study the effect of shift of pole location with 3bit coefficient quantization in both direct form and cascade form.

12. With a block diagram, explain the quantization noise model. Derive the expressions for steady state noise power.

13. Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation y(n)=0.95y(n-1)+x(n) ,x(n)=0 and y(-1)=13. Determine the dead band range of the system.

14. Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation y(n)=0.95y(n-1)+x(n). Determine the dead band range of the system.

Subject Incharge **Head of the Department**
(M.Prabhakaran) **(Dr.M.Kumar)**

HoDRemarks:

Chettinad College of Engineering & Technology
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Academic Year 2024 -2025

Question Bank

Year/Semester: II / IV **Date:19.02.2025**	**Department: ECE** **Subject Code/Title : EC3492 / Digital Signal Processing** **Faculty Name : M.Prabhakaran**	**Unit** ~~: I/II/ III /IV~~/V **Section :Part A/B/C**

UNIT V
DSP APPLICATIONS
PART-A

1. What is multirate signal processing? [U]

The theory of processing signals at different sampling rates is called multirate signal processing.

2. Define Down sampling. [R]

Down sampling a sequence x(n) by a factor M is the process of picking every Mth sample and discarding the rest

3. What is meant by Up sampling? [U]

Up sampling by a factor L is the process of inserting L - 1 zeros between two consecutive samples.

4. What are the classification digital signal processors? [R]

The digital signal processors are classified as

1. General purpose digital signal processors.
2. Special purpose digital signal processors.

5. What are the applications of PDSPs? [R]

Digital cell phones, automated inspection, voicemail, motor control, video conferencing, noise cancellation, medical imaging, speech synthesis, satellite communication etc.

6. Give some examples for fixed point DSPs. [R]

TM320C50, TMS320C54, TMS320C55, ADSP-219x, ADSP-219xx..

7. Give some example for floating point DSPs? [R]

TMS320C3x, TMS320C67x, ADSP-21xxx

8. What is pipelining? [U]

Pipelining a processor means breaking down its instruction into a series of discrete pipeline stages which can be completed in sequence by specialized hardware.

9. What is pipeline depth? [U]

The number of pipeline stages is referred to as the pipeline depth.

10. What are the different buses of TMS320C5x? [R]

The C5x architecture has four buses

1) Program bus (PB)

2) Program address bus (PAB)

3) Data read bus (DB)

4) Data read address bus (DAB)

11. What are the different stages in pipelining? [R]

1) The fetch phase

2) The decode phase

3) Memory read phase

4) The execute phase

12. List the various registers used with ARAU. [R]

- Eight auxiliary registers (AR0 – AR7)
- Auxiliary register pointer (ARP)
- Unsigned 16-bit ALU

13. What is the operation blocks involved in C5x processors? [R]

The central processing unit consists of the following elements:

1) Central arithmetic logic unit (CALU)

2) Parallel logic unit (PLU)

3) Auxiliary register arithmetic unit (ARAU)

4) Memory mapped registers

5) Program controller

14. What is the function of parallel logic unit? [U]

The parallel logic unit is a second logic unit that executes logic operations on data without affecting the contents of accumulator.

15. List the on chip peripherals in ‘C5x? [R]

The on-chip peripherals interfaces connected to the „C5x CPU include

- Clock generator
- Hardware timer
- Software programmable wait state generators
- General purpose I/O pins
    - Parallel I/O ports
    - Serial port interface
    - Buffered serial port
    - Time-division multiplexed (TDM)serial port
    - Host port interface
    - User unmask able interrupts

16. Mention the addressing modes available in TMS320C5X processor? [R]

1) Direct addressing mode
2) Indirect addressing mode
3) Circular addressing mode
4) Immediate addressing
5) Register addressing
6) Memory mapped register addressing

17. Give the applications of DSP Processors? [U]

Digital cell phones, automated inspection, voicemail, motor control, video conferencing, noise cancellation, medical imaging, speech synthesis, satellite communication etc.

18. What is use of ADD instruction? [R]

ADD – Add to accumulator with shift.

Add the content of addressed data memory location or an immediate value of accumulator, if a shift is specified, left-shift the data before the add. During shifting, low- order bits are Zero-filled, and high-order bits are sign extended if SXM=1.

19. Give the features of DSPs? [U]

1) Architectural features

2) Execution speed
3) Type of arithmetic
4) Word length

20. What are load/store instructions? [R]

LACB, LACC, LACL, LAMM, LAR, SACB, SACH, SACL, SAR, SAMM.

PART-B

1. Explain the poly phase structure of decimator and interpolator?

2. Discuss the procedure to implement digital filter bank using multi rate signal processing?

3. (i) Explain how various sound effects can be generated with the help of DSP?

(ii) State the applications of multirate signal processing?

4. With neat sketch explain the functional block diagram of TMS320C5X.

5. List and explain the various types of addressing modes of digital signal processor with suitable example.

6. (i) Explain how DSP can be used for speech processing?

(ii) Explain in detail about decimation and interpolation?

7.A signal x(n) is given by x(n)={0,1,2,3,4,5,6,0,1,2,3,…….)

(i) Obtain the decimated signal signal with a factor of 2

(ii) Obtain the interpolated signal with a factor of 2.

8. Explain the efficient transversal structure for decimator and interpolator?

9. Explain sub band coding in detail.

10. Explain sampling rate conversion by a rational factor and derive input and output relation in both time and frequency domain.

11. Explain the design of narrow band filter using sampling rate conversion.

12. Explain the design steps involved in the implementation of multistage sampling rate Converter.

13. Explain the implementation steps in speech coding using transform coding?

14. Discuss about the principle of operation of floating point architecture with necessary diagram.

15. Explain in detail about adaptive filter

Subject Incharge **Head of the Department**
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