SIDDHARTH GROUP OF INSTITUTIONS :: PUTTUR
Siddharth Nagar, Narayanavanam Road – 517583

QUESTION BANK (DESCRIPTIVE)

Subject with Code : Digital Signal Processing(13A04602) **Course & Branch**: B.Tech - EEE

Year & Sem: IV-B.Tech & I-Sem **Regulation:** R13

UNIT 1

INTRODUCION

1) Explain the classification of discrete time signals and systems. 10M

2) a) Determine if the following systems are time variant or time invariant. 5M

i) $y(n)= x(n)+x(n-1)$ ii) $y(n)=x(-n)$

b) Determine if the system described by the following equation are causal or non-causal 5M

i) $y(n)= x(n)+(1/x(n-1))$ ii) $y(n)=x(n^2)$

3) a) Describe the linear time invariant system. 5M
b) Explain the properties of LTI system. 5M

4) Consider causal and stable LTI system whose I/Ps and O/Ps are related through second order difference equation $y(n)-(1/6)\, y(n-1)-(1/6)y(n-2)=x(n)$ then determine system impulse response h(n) for the system. 10M

5) Find the solution of 2nd order difference equation $y(n)= (5/6)\, y(n-1)-(1/6)y(n-2)+x(n)$, for the input sequence $x(n)=2^n U(n)$ 10M

6) Explain the properties of Discrete Fourier Transform in detail. 10M

7) Explain frequency domine representation of signals and systems in detail 10M

8) Describe the relation between 10M

a) DFT to Z- transform b) DFT to Fourier Series c) DFT to Laplace transform

9) a) Justify DFT can use as a linear Transform. 5M

b) How do you sample and reconstruct a discrete time signal in Frequency domine. 5M

10) a) what is energy of a signal. 5X2=10M

b) what are the differences between time variant and time invariant systems.

c) what are the differences between causal and non-causal systems.

d) What are the multichannel and multi-dimensional signals?

e) what are the differences between DFT and DTFT.

UNIT-II

Fast Fourier Transform Algorithm

1) Compute 8-poin DFT of the sequence x(n)= {1,1,1,1,1,1,0,0} 10M

2)a) Explain about liner convolution of sequence. 5M

b) Compute linear convolution of sequence x(n)={1,2,3,1} and h(n)={1,2,1,-1} 5M

3) Explain about decimation in time FFT algorithm. 10M

4) Explain about decimation in frequency FFT algorithm. 10M

5) Compute DFT of the sequence x(n)={1,2,3,4,4,3,2,1} using DITFFT algorithm. 10M

6) Compute DFT of the sequence x(n)={1,2,3,4,4,3,2,1} using DIFFFT algorithm. 10M

7) Compute IDFT of the sequence x(n)={ 7,-0.707-j0.707,-j, 0.707-j0.707,1, 0.707+j0.707,j, -0.707+j0.707} 10M

8) Compute IDFT of the sequence x(n)={ 7,-0.707-j0.707,-j, 0.707-j0.707,1, 0.707+j0.707,j, -0.707+j0.707} 10M

9) How do you compute DFT using 10M

a) The Goertzel Algorithm b)The chrip-z Transform

10) a) what is twiddle factor 5X2=10M

b) Draw the butterfly diagram for DITFFT algorithm

c) Draw the butterfly diagram for DIFFFT algorithm

d) what are the applications of FFT algorithm

e) what is the bit reversal order of 16 point sequence.

UNIT-III

Implementation of Discrete-Time Systems

1. (a) Discuss the realization of FIR filter structures. 5M

(b) Realize FIR filter with system function in cascade form 5M

H (z) = 1 + (5/2) z-1+2z-2+2z-3.

2. Consider the system $y(n) = y(n - 1) + 2y(n - 2) + x(n)$

(a) Find H(z) . 5M

(b) Realise using direct form-II 5M

3. Realise the discrete system $y(n) = -0.1y(n-1)+0.2y(n-2)+3x(n)+3.6x(n-1)+0.6x(n-2)$

using.(a) Cascade forms 5M

(b) Parallel forms. 5M

4. Consider the discrete system : y(n)=2y(n-1)+2y(n-2)+x(n)+x(n-1).

(a) Find the Z-transform 5M
(b) Realize the system using direct form-I method. 5M

5. (a) Explain the advantages and disadvantages of Direct form-II realization over Direct form-I. 5M
(b) Realize following system with difference equation in cascade form
$y(n)= y(n) = y(n - 1) + 2y(n - 2) + x(n)$ 5M

6. What is the principle of designing FIR filters using windows?

7. Realize system with following difference equation
$y(n) = (3/4)\ y(n-1) - (1/8)\ y(n-2) + x(n) + (1/3)x(n-1)$
a)direct form-I 5M
b)direct form-II 5M

8. Realize system with following difference equation
$y(n) = (3/4)\ y(n-1) - (1/8)\ y(n-2) + x(n) + (1/3)x(n-1)$.
a)cascade form 5M
b)parallel form 5M

9.Expalin briefly about different structures in FIR systems 10M

10.Explain briefly about the following IIR structures

a) a)direct form-I 2M

b)direct form-II 2M

c)cascade form 2M

d)parallel form 2M

e)lattice structure form 2M

UNIT –IV

Design of Digital Filters

1. (a) Explain the FIR filter design using windowing technique. 5M
(b) Compare FIR and IIR filters. 5M

2. (a) Explain the features of Chebyshev approximation. 5M
(b) Discuss the location of poles for Chebyshev filter. 5M

3. (a) Discuss the characterization of IIR filter. 5M
(b) Using backward difference method obtain H (z) for following H (s) = 1/(s + 3). 5M

4. (a) Compare features of different windowing functions. 5M
(b) Justify that FIR filter is linear phase filter. 5M

5. Describe the IIR filter design approximation using Bilinear Transformation method. Also sketch the s-plane to z-plane mapping. State its merits and demerits. 10M

6. Convert the following analog filter transfer function using backward difference method, Impulse invariant method and Bilinear Transformation method. 10M
H(s)=1/(s+0.2) Consider T= 1 Sec

7. Give the expression for rectangular window function. Find its frequency response and also sketch its spectrum. Also discuss its features. 10M

8. (a) Discuss about characteristics linear phase FIR filters. 5M
(b) What are the effects of windowing. 5M

9. Design an analog Butterworth filter that has a -2db pass band attenuation at a frequency of 20 rad/sec and at least -10dB stop band attenuation at 30 rad/sec (assume Ωc = 21.3868 rad/sec) 10M

10.Compare
a)rectangular window and Hanning window 2M
b) rectangular window and Hamming window 2M
c) Hamming window and Hanning window 2M
d) Hamming window and Blackman window 2M
e) Hamming window and Kaiser window 2M

UNIT – V

Multirate Digital signal processing

1. (a) Define down sampling and up sampling with suitable example. 5M
(b) What is aliasing ? What is the need for anti- aliasing filter prior to down sampling. 5M

2. (a) Explain the need of multirate signal processing with suitable example. 3M
(b) What is the imaging and aliasing ? How their spectrum differ? 3M
(c) Can fractional sampling implemented directly? Justify your answer with suitable example. 4M

3. (a) What is decimation and interpolation? Explain briefly with suitable sketches. 3M
(b) Find the Z- transform of $a^n u(n)$ upsampled by a factor 2. 3M
(c) What is imaging? 4M

4. (a) What is up-sampling and down sampling ? 3M
(b) What is need for up sampling and down sampling? 3M
(c) Find the Z- transform of $a^n u(-n-1)$ with a down sampling by a factor `2' .Comment on ROC in comparison with original ROC i.e.$a^n u(-n-1)$ ROC. 4M

5. With the help of block diagram explain the sampling rate conversion by a rational factor `I/D'. Obtain necessary expressions. 10M

6. Describe the decimation process with a factor of `M '. Obtain necessary expression, 10M

sketch frequency response. Also discuss aliasing effect.

7. a)What is sub band coding? How is it achieved with the help of multi rate DSP? 5M
   b) Write short note on Decimation 5M

8. (a) Discuss the need for signal compression. 5M
   (b) Explain the concept of dual tone multi frequency signal detection. 5M

9. Discuss the applications digital signal processing. 10M

10. Explain the following:
   a) Decimation 2M
   b) Interpolation 2M
   c) Sampling rate conversion 2M
   d) Up sampling 2M
   e)Down sampling 2M

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UNIT – I

INTRODUCTION

1) Sequence steps for converting analog signal to digital signal [ ]
A) Sampling, coding, quantization B) sampling, quantization, coding,
C) coding, sampling, quantization D) quantization, sampling, coding

2) Analog signal given to the sampler then the output is [ ]
A) Discrete signal B) Digital signal C) Quantized signal D) Analog signal

3) 'A signal that varies continually with time' then the signal is [ ]
A) Digital signal B) Discrete signal C) Quantized signal D) Analog signal

4) 'A signal that has values at particular instant of time' then the signal is [ ]
A) Digital signal B) Discrete signal C) Quantized signal D) Analog signal

5) If X(n) is a signal and X(n+N)=X(n) then X(n) is said to be [ ]
A) Aperiodic signal B) non-periodic signal C) periodic signal D) Stationary signal

6) If X(n) is a signal and X(n+N)≠X(n) then X(n) is said to be [ ]
A) Aperiodic signal B) Stationary signal C) periodic signal D) Stationary signal

7) If X(n) is a periodic signal and X(n+N)=X(n) then N is said to be [ ]
A) Frequency B) Time C) Time period D) Frequency slot

8) If X(n) is a signal and fallow the property X(-n)=X(n) then X(n) is said to be [ ]
A) Symmetric Signal B) Even Signal C) Asymmetric signal D) both a and b

9) If X(n) is a signal and fallow the property X(-n)= -X(n) then X(n) is said to be [ ]
A) Symmetric Signal B) Odd Signal C) Asymmetric signal D) both b and c

10) A signal is defined as X(n)= 1 for n=0; and X(n)= 0 for n≠0; then X(n) is said to be [ ]
A) Unit step B) Unit sample C) ramp D) Exponential

11) A signal is defined as X(n)= 1 for n≥0; and X(n)= 0 for n<0; then X(n) is said to be [ ]
A) Unit step B) Unit sample C) ramp D) Exponential

12) A signal is defined as X(n)= n for n>0; and X(n)= 0 for n<0; then X(n) is said to be [ ]
A) Unit step B) Unit sample C) ramp C) Exponential

13) If the energy of a signal X(n) is finite value then power of that signal is [ ]
A) 1 B) 0 C) not defined D) >1

14) If the energy of a signal X(n) is infinite then power of that signal is [ ]
A) Finite B) infinite C) finite or infinite D)not able to determine

15) If the system output depends only on present and past inputs,the system is said to be [ ]
A) Causal system B) non causal system C) linear system D)non linear system

16) If the system output depends on present, past and future inputs, the system is said to be [ ]
A) Causal system B) non causal system C) linear system D)non linear system

17) If a system satisfies the superposition theorem then system is said to be---------system [ ]
A) Timevarient B) Time invariant C) non linear D) linear

18) If a relaxed system doesn't satisfy the superposition theorem then system is said to be [ ]
A) Timevarient B) Time invariant C) non linear D) linear

19) A LTI system is said to be stable if=------------------ [ ]
A) Unbounded O/Ps for Unbounded I/Ps B) Unbounded O/Ps for bounded I/Ps
C) bounded O/Ps for bounded I/Ps D) bounded O/Ps for Unbounded I/Ps

20) -------Is example for linear signal [ ]
A) S1 (t) = 5 t B) S2 (t) = 10 t2 C)S3 (t) = 20 t2 D) None

21)------- Is alternate Method for processing analog signals [ ]
A) A to D converter B) D to A converter C) Digital signal processing D) None

22)The Sequence of steps for converting analog signal to digital signal------- [ ]
A) Encoding, Sampling, Quantizing B) Sampling, Quantizing, Encoding
C) Quantizing, Sampling, Encoding D) None

23) Is Operation on Independent Variable [ ]
A) Scalar Multiplication B) Signal Multiplier C) Addition operation D) Time Scaling

24)------- Is Operation on dependent Variable [ ]
A) Scalar Multiplication B) Time Shifting C) Time ReversalD) Time Scaling

25) If x(n) is given signal then x(2n) Indicates ------- [ ]
A) Compressed of x (n) B) Expansion of x (n) C) Multiplication of x (n) D) None

26) If x(n) is given signal then x(n/2) Indicates ------- [ ]
A) Compression of x (n) B) Expansion of x (n) C) Multiplication of x (n) D) None

27) Given is true for unit sample sequence [ ]
A) $\delta(n) = 1 \quad n=0$ B) $\delta(n) = 1 \quad n \neq 0$ C) $\delta(n) = 1 \quad n=1$ D) None

28) Given is true for unit step sequence [ ]
A) $u(n) = 1 \quad n \geq 0$ B) $u(n) = 1 \quad n \neq 0$ C) $\delta(n) = 1 \quad n=1$ D) None

29) -------- is the relation $\delta(n)$ in terms u(n) [ ]
A) $\delta(n) = u(n-1)$ B) $\delta(n) = u(n) - u(n-1)$ C) $\delta(n) = u(n) + u(n-1)$ D) None

30) Given is true for Energy Signal [ ]
A) $P=\infty$ B) $P=0$ C) $E=0$ D) None

31) Given is true for Power Signal [ ]
A) $E=\infty$ B) $E=0$ C) $P=0$ D) None

32) A signal is periodic signal with period 'N' if x(n) = ------ [ ]
A) x (2N) B) x (n+N) C) x(n-1) D) None

33) Is fundamental period of $x(n) = \cos(n\pi/2)$ [ ]
A) 4 B) 8 C) 2 D) None

34) A signal is said to be even signal if ------- [ ]
A) x (-n)=-x(n) B) x (-n)=2x(n) C) x (-n)= x(n) D) None

35) A signal is said to be odd signal if ------- [ ]
A) x (-n)=-x(n) B) x (-n)=2x(n) C) x (-n)= x(n) D) None

36) If x(n) is given signal then even part of x(n) is ------- [ ]
A) $x_e(n)=x(n)+x(-n)$ B) $x_e(n)=x(n)-x(-n)$ C) $x_e(n)=1/2[x(n)+x(-n)]$ D) None

37) If x(n) is given signal then odd part of x(n) is ------- [ ]
A) $x_o(n)=x(n)+x(-n)$ B) $x_o(n)=x(n)-x(-n)$ C) $x_o(n)=1/2[x(n)-x(-n)]$ [D] None

38) A signal is said to be causal signal if ------- [ ]
A) $x(n)=0$ $n<0$ B) $x(n)=0$ $n>0$ C) $x(n)=0$ D) None

39) A System is said to be causal system if present output depends ------- [ ]
A) Present Inputs B) past inputs C) both D) None

40) DFS is a mathematical tool used to analyse ------- [ ]
A) Aperiodic Sequences B) Periodic Sequences C) Both D) None

UNIT – II

FAST FOURIER TRANSFORM ALGORITHM (FFTA)

1) In N-Point DITFFT, number of butterflies per stage is ------- [ ]
[A] 2N **[B]** 3N **[C]** N/2 **[D]** N/3

2) In 16-Point DITFFT, each sample represented by ------- digits [ ]
[A] 2 **[B]** 3 **[C]** 4 **[D]** 8

3) In N-Point DIT-FFT input sequence order is ------- [ ]
[A] Natural **[B]** Bit reversal **[C]** even **[D]** None

4) In N-Point DIT-FFT, number of stages in the flow graph is ------- [ ]
[A] 2N **[B]** 3N **[C]** $\log_2 N$ **[D]** $2\log_2 N$

5) In N-Point DITFFT, output sequence order is ------- [ ]
[A] Natural **[B]** Bit reversal **[C]** even **[D]** None

6) Direct DFT requires ------- number of complex multiplications [ ]
[A] N **[B]** N^2 **[C]** $(N/2)\log_2 N$ **[D]** None

7) FFT algorithms requires ------- number of complex multiplications [ ]
[A] N **[B]** N^2 **[C]** $(N/2)\log_2 N$ **[D]** None

8) In DITFFT, Inputs/outputs for each butterfly in stage ‘m’ separated by ------- [ ]
[A] 2^m **[B]** 2^{m-1} **[C]** 2^m-1 **[D]** None

9) In direct computation of DFT the number of real multiplications are [ ]
[A] $2N^2$ **[B]** 2N **[C]** $4N^2$ **[D]** None

10) In direct computation of DFT the number of real additions are ------- [ ]
[A] 2N **[B]** 2(N-1) **[C]** 4N (N-1) **[D]** None

11) In direct computation of DFT the number of complex additions are ------- [ ]
[A] N^2 **[B]** N (N-1) **[C]** (N-1)/2 **[D]** None

12) In direct computation of DFT the number of complex multiplications are [ ]
[A] N^2 **[B]**N (N-1) **[C]** (N-1)/2 **[D]** None

13) In radix 2 FFT, the no of complex multiplications for ‘m’ stages is ------- [ ]
[A] $(N/2)\log_2 N$ **[B]** $(N/2)\log_2(N/2)$ **[C]** $(N+1)\log_2(N/2)$ **[D]** $(N)\log_2(N)$

14) In radix 2 FFT, the no of complex additions for ‘m’ stages is ------- [ ]
[A] $(N/2)\log_2 N$ **[B]** $(N/2)\log_2(N/2)$ **[C]** $(N+1)\log_2(N/2)$ **[D]** $(N)\log_2(N)$

15) For a 32 point DFT using direct method, no of complex additions are ------- [ ]
[A] 992 **[B]** 986 **[C]** 942 **[D]** 936

16) For a 16 point DFT using direct method, no of complex multiplications are ------- [ ]
[A] 240 **[B]** 256 **[C]** 235 **[D]** 128

17) In 128 point FFT, the number of complex additions are ------- [ ]

[A] 992 **[B]** 896 **[C]** 448 **[D]** 16256

18) In 64 point FFT, the number of complex multiplications are ------- [ ]

[A] 1024 **[B]** 896 **[C]** 192 **[D]** 80

19) The value of the twiddle factor at N=4 and n*k=3 is ------- [ ]

[A] j **[B]** -j **[C]** 1 **[D]** 0

20) Complex multiplication takes place before add/sub operations in ------- [ ]

[A] DIT **[B]** DIF **[C]** Both **[D]** None

21) Complex multiplication takes place after add/sub operations in ------- [ ]

[A] DIF **[B]** DIT **[C]** both **[D]** None

22) If X(k) consist of N- no of frequency samples, then its discrete frequency locations are given by the __________ [ ]

[A] $f_k=KF_s/N$ **[B]** $f_k=F_s/N$ **[C]** $f_k=KN/F_s$ **[D]** $f_k=N$

23) Twiddle factor W_N given by --- [ ]

[A] $e^{-j2\pi/n}$ **[B]** $e^{j2\pi/n}$ **[C]** $- e^{-j2\pi/n}$ **[D]** $e^{-j\pi/n}$

24) Symmetry property of twiddle factor is ---------- [ ]

[A] $w_N^{k+n/2}= w_N^k$ **[B]** $w_N^{k+n/2}= -w_N^k$ **[C]** $w_N^{k+n/3}= w_N^k$ **[D]** $w_N^{k+n/4}= w_N^k$

25) Periodicity property of twiddle factor is ---------- [ ]

[A] $w_N^{k+n}= w_N^k$ **[B]** $w_N^{k+n/2}= -w_N^k$ **[C]** $w_N^{k+n}= w_N^k$ **[D]** $w_N^{k+n/2}= w_N^k$

26) By using twiddle factor computational complexicity reduced from N^2 to ------------ [ ]

[A] $N/2 \log_2 N$ **[B]** $N/4 \log_2 N$ **[C]** $N/2 \log_2{}^{2N}$ **[D]** $- N/2 \log_2 N$

27) The number of butterflies per stage is -------- for N-point DFT [ ]

[A] N/2 **[B]** N/4 **[C]** N **[D]** N/6

28) Bit reversal order for I/P of DITFFT algorithm is [ ]

[A] {0,2,4,6,1,3,5,7} **[B]** {0,1,2,3,4,5,6,7} **[C]** {0,4,2,6,1,5,3,7} **[D]** {0,4,6,2,1,5,7,3}

29) Bit reversal order for O/P of DIFFFT algorithm is [ ]

[A] {0,2,4,6,1,3,5,7} **[B]** {0,1,2,3,4,5,6,7} **[C]** {0,4,2,6,1,5,3,7} **[D]** {0,4,6,2,1,5,7,3}

30) The I/Ps and O/Ps for each butterfly in the stage 'm' is separated by [ ]

[A] 2^m **[B]** 2^{m-1} **[C]** 2^{m+1} **[D]** 2^{-m}

31) Computational complexity will__________by using twiddle factors in FFT calculation [ ]

[A] Increase **[B]** Decrease **[C]** Not effected **[D]** be same

32) How many twiddle factors are required for computing 8-point FFT [ ]

[A] 8 **[B]** 16 **[C]** 2 **[D]** 4

33) How many twiddle factors are required for computing 16-point FFT [ ]

[A] 8 **[B]** 16 **[C]** 2 **[D]** 4

34) How many twiddle factors are required for computing 32-point FFT [ ]

[A] 8 **[B]** 16 **[C]** 2 **[D]** 4

35) W_8^0 value is [ ]

[A] 1 **[B]** 0.707-j0.707 **[C]** -j **[D]** -0.707-j0.707

36) W_8^1 value is [ ]

[A] 1 **[B]** 0.707-j0.707 **[C]** -j **[D]** -0.707-j0.707

37) W_8^2 value is [ ]

[A] 1 **[B]** 0.707-j0.707 **[C]** -j **[D]** -0.707-j0.707

38) W_8^3 value is [ ]

[A] 1 **[B]** 0.707-j0.707 **[C]** -j **[D]** -0.707-j0.707

39) In 16 point DITFFT algorithm number of butterflies per stage is [ ]
[A] 16 **[B]** 8 **[C]** 4 **[D]** 2

40) In 8 point DITFFT algorithm number of butterflies per stage is [ ]
[A] 16 **[B]** 8 **[C]** 4 **[D]** 2

41) In 32 point DITFFT algorithm number of butterflies per stage is [ ]
[A] 16 **[B]** 8 **[C]** 4 **[D]** 2

42) In 4 point DITFFT algorithm number of butterflies per stage is [ ]
[A] 16 **[B]** 8 **[C]** 4 **[D]** 2

UNIT – III
Implimentation of discrete Time Systems

1) The three factors that influence structures are computation complexity, memory and [ ]
[A] Speed **[B]** Accuracy **[C]** finite word length **[D]** None

2**)** The unit sample response of FIR system is identical to [ ]
[A] h(n)=0 **[B]** $h(n)=b_n$ **[C]** h(n)=u(n) **[D]** None

3) The length of FIR filter is [ ]
[A] M-1 **[B]** M **[C]** M-2 **[D]** None

4**)** The direct form structure is equivalent to [ ]
[A] Sampling **[B]** DFT **[C]** convolution **[D]** None

5) The number of memory locations needed to realize direct form structure is [ ]
[A] M-1 **[B]M** **[C]M+N-1** **[D]** None

6**)** The number of additions per output point needed to realize direct form structure is [ ]
[A] M **[B]** M-1 **[C]** M-N-1 **[D]** None

7**)** The number of multiplications per output point in direct form structure is [ ]
[A] M **[B]** M-1 **[C]** M+N-1 **[D]** None

8**)** The tapped delay line filter is also called as [ ]
[A] parallel form **[B]** Direct form **[C]** Cascade form **[D]** None

9) The condition for FIR system to have linear phase is [ ]
[A] h(n)=0 **[B]** h(n)=+/- h(M-N-1) **[C]** h(n)=h(M-N) **[D]** None

10) For a linear phase FIR system if M=even the no of multiplications is [ ]
[A] M **[B]** (M-1)/2 **[C]** M/2 **[D]** (M+1)/2

11**)** For a linear phase FIR system if M=odd the no of multiplications is [ ]
[A] (M-1)/2 **[B]** M/2 **[C]** (M+1)/2 **[D]** (M-N-1)/2

12**)** In frequency sampling structure the value used to characterize the filter is [ ]
[A] impulse response **[B]** step response **[C]** frequency response **[D]** None

13**)** The most efficient form of realization is [ ]
[A] Direct form **[B]** parallel **[C]** frequency sampling **[D]** cascade

14**)** The structure that is mostly used in digital speech processing is [ ]
[A] Cascade **[B]** Parallel **[C]** Lattice **[D]** Direct form

15**)** IIR filter's Direct form is obtained by cascading all zero system with _____ [ ]
[A] Inverse system **[B]** conjugate system **[C]** all pole system **[D]** None

16**)** In IIR direct form I the number of additions is ________ [ ]
[A] (M+N)/2 **[B]** (M-N)/2 **[C]** M-N **[D]** M+N

17**)** The no of memory locations needed to realize IIR direct form I is ________ [ ]
[A] M+N-1 **[B]** M+N+1 **[C]** M+N **[D]** M-N

18**)** In IIR direct form I the number of multiplications is __________ [ ]
[A] M+N-1 **[B]** M+N+1 **[C]** M-N -2 **[D]** None

19) The no of multiplications required to realize IIR direct form II is _______ [ ]
[A] M+N-1 **[B]** M+N+1 **[C]** M-N -2 **[D]** None

20) The Direct form structure is also called as _________ [ ]
[A] sampling **[B]** Canonic **[C]** Parallel **[D]** None

21) The no of additions required to realize IIR direct form II is _______ [ ]
[A] M+N-1 **[B]** M+N+1 **[C]** M+N **[D]** None

22) The structure obtained by changing all branch direction and input & output is [ ]
[A] Canonic **[B]** Cascade **[C]** Transposed **[D]** None

23) The structure that needs lesser memory location is _______ [ ]
[A] Direct form I **[B]** Direct form II **[C]** Cascade **[D]** Parallel

24) The Parallel form realization of IIR system is obtained by _______ [ ]
[A] Differential equation **[B]** Difference equation **[C]** Partial Fraction **[D]** None

25) The Lattice coefficients are also called as _______ [ ]
[A] Constants coefficient **[B]** parallel coefficient **[C]** reflection coefficient **[D]** None

26) The Polar form of Z can be expressed as ------- [ ]
[A] $-re^{jw}$ **[B]** re^{jw} **[C]**) e^{jw} **[D]** None

27) Z transform of sequence x(n)={1,0,3} is ------- [ ]
[A] $1+Z+3Z^{-2}$ **[B]** $1+3Z^{-2}$ **[C]** $1+Z^{-2}$ **[D]** None

28) Z transform of sequence x(n)={1,1,3}is --------(take origin at second sample) [ ]
[A] $Z^{-1}+1+Z$ **[B]** $Z^{-1}+1+2Z$ **[C]** $1+Z^{-2}$ **[D]** None

29) ROC for Left hand finite sequence is -------- ------- [ ]
[A] Entire Z except Z=0 **[B]** Entire Z except Z=∞ **[C]** Entire Z except Z=1 **[D]** None

30) ROC for Right hand finite duration sequence is ------- [ ]
[A] Entire Z except Z=0 **[B]** Entire Z except Z=∞ **[C]** Entire Z except Z=1 **[D]** None

31) ROC for Left hand infinite duration sequence is ------- [ ]
[A] Inside circle **[B]** Outside circle **[C]** Entire Z **[D]** None

32) ROC for Right hand infinite duration sequence is ------- [ ]
[A] Inside circle **[B]** Outside circle **[C]** Entire Z **[D]** None

33) The range of values of Z for which z-Transform converges called as ------- [ ]
[A] Region of complex **[B]** Region of covariance **[C]** Region of convergence **[D]** None

34) ROC for Two sided finite duration sequence is ------- [ ]
[A] Inside circle **[B]** Outside circle **[C]** Entire Z except Z=0 & Z=∞ **[D]** None

35) Z-transform of unit sample sequence is -------- [ ]
[A] 1 **[B]** 0 **[C]** u(n) **[D]** None

36) Z-transform of δ (n-m) is ------- [ ]
[A] Z^{m} **[B]** Z^{m-1} **[C]** Z^{-m} **[D]** None

37) Z-transform of unit step sequence is -------- [ ]
[A] z **[B]** z/z-1 **[C]** z-1 **[D]** None

38) Z-transform of a^{n} u(n) is -------- [ ]
[A] z **[B]** z/z-a **[C]** z-a **[D]** None

39) ROC for unit sample sequence is ------- [ ]
[A] Entire Z **[B]** Entire Z except Z=0 **[C]** Entire Z except Z=∞ **[D]** None

40) ROC for unit step sequence is ------- [ ]
[A] Entire Z **[B]** Entire Z except Z=0 **[C]** |Z| > 1 **[D]** None

UNIT – IV

Design of Digital Filters

1.IIR Filters are [ ]

A)Recursive type B)Non-Recursive type

C)Neither Recursive nor non-recursive D)None

2.In the Impulse Invariance Transformation, relationship between Ω and ω is [ ]

A) $\Omega = \omega T$ B) $\Omega = \omega/T$

C) $\omega = \Omega/T$ D) $\omega = T/\Omega$

3.Non-linearity in the relationship between Ω and ω is known as [ ]

A)Aliasing B)Frequency Warping

C)Unwarping D)Frequency Mixing

4. In The Bilinear Transformation, the Relationship between Ω And Ω Is [ ]

A) $\Omega = 2\tan(\omega/2)$ B) $\Omega = 2/T \tan(\omega/2)$

C) $\Omega = 1/T\tan(\omega/2)$ D) $\Omega = \tan(\omega T/2)$

5.Butterworth filters have [ ]

A)Wideband Transition Region B) Sharp Transition Region

C) Oscillation in Transition Region D) None

6. Chebyshev filters have [ ]

A)Wideband Transition Region B) Sharp Transition Region

C) Oscillation in Transition Region D) None

7. Type-1 Chebyshev filters contains [ ]

A)oscillations in the passband B) oscillations in the passband

C) oscillations in the stop and pass banda D) Oscillation in Transition band

8. Type-2 Chebyshev filter is also called [ ]

A)inverse chebyshev filter B)elliptic filter

C) reverse chebyshev filter D)None

9. The physically realizable IIR filters do not have -------------- phase [ ]

A)linear B)Non-linear

C)magnitude D)None

10.In ---------------------transformation,the impulse response of digital filter is the Sampled version of the impulse of analog filter. [ ]

A)impulse invariant B)bilinear

C)magnitude D)phase

11.Alaising occurs only in ---------------------transformation. [ ]

A) impulse invariant B)bilinear

C)magnitude D)phase

12.In -----------------approximation,the magnitude response is equiripple in the Passband and monotonic in the stopband [ ]

A) Type-1 Chebyshev B) Type-2 Chebyshev

C)butterworth D)None

13.In -----------------approximation,the magnitude response is monotonic in the Passband and equiripple in the stopband [ ]

A) Type-1 Chebyshev B) Type-2 Chebyshev

C)butterworth D)None

14. In -----------------approximation,the magnitude response is maximally flat at the origin and monotonically decreaseswith increasing frequency [ ]

A) Type-1 Chebyshev B) Type-2 Chebyshev

C)butterworth D)None

15.At the cutoff frequency ,the magnitude of the butterworth filter is----------- times the maximum value [ ]

A)$1/\sqrt{2}$ B)1/2

C)1 D)-1/2

16.The ideal filters are [ ]

A)causal B)Non causal

C)symmetric D)none

17.In fourier series method to get transfer function of realizable filter,H(z) is to be Multiped by [ ]

A)$z^{-(N-1)/2}$ B) $z^{(N-1)/2}$

C) $z^{-(N-1)}$ D) $z^{(N-1)}$

18.The abrupt truncation of fourier series results in oscillations in [ ]

A)stopband B)passband

C)both A and B D)none

19.The frequency of a digital filter is [ ]

A)periodic B)Non periodic

C)may be periodic or Non periodic D)none

20.For rectangular window ,the main lobe width is equal to [ ]

A)$2\pi/N$ B) $4\pi/N$

C) $8\pi/N$ D) $12\pi/N$

21.For Hanning window ,the main lobe width is equal to [ ]

A)2π/N B) 4π/N

C) 8π/N D) 12π/N

22.For Hamming window ,the main lobe width is equal to [ ]

A)2π/N B) 4π/N

C) 8π/N D) 12π/N

23.For blackman window ,the main lobe width is equal to [ ]

A)2π/N B) 4π/N

C) 8π/N D) 12π/N

24.For kaiser window ,the main lobe width is equal to [ ]

A)Adjustable B) 4π/N

C) 8π/N D) 12π/N

25.For Rectangular window ,the peak side lobe magnitude in dB is [ ]

A)-13 B)-31

C)-41 D)-58

26.For Hanning window ,the peak side lobe magnitude in dB is [ ]

A)-13 B)-31

C)-41 D)-58

27.For Hamming window ,the peak side lobe magnitude in dB is [ ]

A)-13 B)-31

C)-41 D)-58

28.For Blackman window ,the peak side lobe magnitude in dB is [ ]

A)-13 B)-31

C)-41 D)-58

29.for a linear phase filter the delay is [ ]

A)variable B)constant

C)function D)sequence

30.In FIR filters ,--------------- is a linear function of ω [ ]

A) phase B)width

C)oscillations D)None

31.In--------------------- window spectrum the higher side lobe attenuation is [ ]
Achieved at the expense of increased main lobe width

A)Blackman B)Hamming

C)Kaiser D)Hanning

32.In--------------------- window spectrum the increase in side lobe attenuation is [ ]
Achieved at expense of constant attenuation at high frequencies

A)Blackman B)Hamming

C)Kaiser D)Hanning

33. In--------------------- window spectrum has the highest attenuation for side lobes [ ]

A)Blackman B)Hamming

C)Kaiser D)Hanning

34. In--------------------- window spectrum,the side lobe magnitude is variable [ ]

A)Blackman B)Hamming

C)Kaiser D)Hanning

35. In--------------------- window spectrum, the width of the main lobe is triple that of Rectangular window for same value of N [ ]

A)Blackman B)Hamming

C)Kaiser D)Hanning

36.In--------------------- window spectrum, the width of the main lobe is double that of Rectangular window for same value of N [ ]

A)Blackman B)Hamming

C)Kaiser D)Hanning

37.The -------------------- response of the filter is fourier transform of impulse response Of the filter. [ ]

A)magnitude B)phase

C)frequency D)natural

38.The ideal filters are -----------------------,and hence physically unrealizable [ ]

A)causal B)Non causal

C)symmetric D)none

39.In FIR filters with constant phase delay,the impulse response is [ ]

A)causal B)Non causal

C)symmetric D)none

40.The generation of oscillations due to slow convergence of the fourier series near the Points of discontinuity is called -------------------------- phenomenon [ ]

A)Gibbs B)Guassian

C)Poission D)Rayleigh

UNIT – V

Multirate Digital signal processing

1.Decimation results in [ ]

A)decrease in sampling rate B) increase in sampling rate

C)no change in sampling rate D)random change in sampling rate

2.Interpolation results in

A)decrease in sampling rate B) increase in sampling rate [ ]

C)no change in sampling rate D)random change in sampling rate

3.The down-sampled signal is obtained by multiplying the signal x(n) with [ ]

A)impulse function B)unit step function

C)unit ramp function D)train of impulses

4.Anti-aliasing filter is to be kept [ ]

A)before down sampler B)after the down sampler

C)after up sampler D)before up sampler

5. Anti-imaging filter is to be kept [ ]

A)before down sampler B)after the down sampler

C)after up sampler D)before up sampler

6.Up sampler and down sampler are [ ]

A)time varying systems B) time invarying systems

C)both A and B D)unpredictable

7.Up sampling by a factor I introduces [ ]

A)I zeros between samples B) I-1 zeros between samples

C)no zeros D) I/2 zeros between samples

8. Down sampling by a factor D skips [ ]

A)D samples B)D-1 samples

C)no samples D)D/2 samples

9. Down sampling by a factor D introduces how many additional images? [ ]

A)D images B) D-1 images

C)no images D)D/2 images

10. Up sampling by a factor I introduces how many additional images? [ ]

A)I images B) I-1 images

C)no images D)I/2 images

11.A delay of D sample periods before a down sampler is the same as a delay of [ ] how many sample periods after the down sampler.

A)D B)1

C)D/2 D)D-1

12.A delay of one sample period before up sampling leads to a delay of how many [ ]
sample periods after the up sampling.

A)I B)I-1

C)I/2 D)1

13.Cascading a factor of I interpolator and a factor of D decimator results in a sampling [ ]
Rate conversion by a factor of

A)I/D B)ID

C)D/I D)1/ID

14. if x(n)={1,2,3,4,5,6,7,..} then x(n/2)= [ ]

A){1,0,2,0,3,0,4,0,5,0,6,0,..} B){1/2,2/2,3/2,4/2,/2,5/2,6/2,7/2,…}

C){1,3,5,7,..} D){2,4,6,8,10,..}

15. if x(n)={1,2,3,4,5,6,7,..} then x(2n)= [ ]

A) {2,4,6,8,10,..} B) {1,0,2,0,3,0,4,0,5,0,6,0,..}

C){1,3,5,7,..} D){1,0,0,2,0,0,3,0,0,4,0,0,5,0,0..}

16. sampling rate conversion ,first--------------to be performed and then --------------- [ ]
is to be performed.

A)decimation ,interpolation B) interpolation, decimation

C)up sampling,down sampling D) down sampling, up sampling

17.The reciprocal of nyquist rate is called the------------- [ ]

A)decimation B) interpolation

C)nyquist period D)up sampling

18.In ----------------systems,single sample rate is used. [ ]

A)multirate B)single

C)continuous time D)non causal

19.The systems that process data at more than one sampling rate are called--------- systems[ ]

A) multirate B)single

C)narrow band D)wide band

20.The basic two operations in multi rate signal processing are --------and -------- [ ]

A) interpolation, decimation B) up sampling,down sampling

C)scaling,shifting D)none

21.The complete process of -------------and then------------is referred to as a [ ]
Decimation

A)filtering,down sampling B) filtering,up sampling
C)delay,down sampling D)delay,up sampling

22.The filter used to band limit the signal prior to down sampling is called------ [ ]
A)Anti-aliasing B)aliasing
C)multirate filter D)single rate filter

23.The sampling rate of a discrete time signal can be ------------- by a factor I by Placing I-1 equally spaced zeros between each pair of samples. [ ]
A)decreased B)increased
C)equal D)none

24.Interpolation is the complete process of ---------and ------------ to remove Image spectra [ ]
A) delay,up sampling B)up sampling,filtering
C) filtering,down sampling D)none

25.The low pass filter which is used after the upsampler to remove the image spectra is called the ------------filter. [ ]
A)Anti-aliasing B)Anti-imaging
C)aliasing D)single rate

26.A sampling rate conversion by a factor I/D can be achieved by --------- a factor of I interpolator and a factor of D decimator. [ ]
A)cascading B)connecting parallel
C)converting D)none

27.The --------------- of a decimator is an interpolator and vice versa [ ]
A)inverse B)transpose
C)aliasing D)none

28.In digital audio ,the different sampling rates used are ----------kHz for broadcasting, ----------kHz for compact disc and -------kHz for audio tape [ ]
A)48,44.1,32 B)32,44.1,48
C)48,32,44.1 D)32,48,44.1

29.how many types of filter banks in multirate digital signal processing? [ ]
A)four types B)two types
C)three types D)five types

30.The phenomenon of getting image spectra in the output of up sampler in addition to the scaled input spectra is called-------- [ ]
A)imaging B)aliasing
C)Anti-aliasing D)none

31.The overlapping of the spectra at the output of the down sampler due to the [ ]
Lack of band limiting of the signal fed to the down sampler is called

A) imaging B)aliasing

C)Anti-aliasing D)none

32.The sampling rate of $2f_h$ samples per second where f_h is the highest frequency [ ]
Component in the signal is called the ------------------

A)nyquist rate B)multi rate

C)single rate D)none

33.By performing sampling rate conversion for either D>>1 and/or I>>1 [ ]
We go for ------------------------implementation

A)multistage B)single stage

C)two stage D)none

34.Up sampling means [ ]

A)decrease the sampling rate B) increase the sampling rate

C)no change in sampling rate D)random change in sampling rate

35.Down sampling means [ ]

A)decrease the sampling rate B) increase the sampling rate

C)no change in sampling rate D)random change in sampling rate

36.If x(n)={1,-1,3,4,0,2,5,1,6,9,…} then x(3n)--- [ ]

A) {2,4,6,8,10,..} B) {1,0,2,0,3,0,4,0,5,0,6,0,..}

C){1,4,5,9,..} D){1,0,0,2,0,0,3,0,0,4,0,0,5,0,0..}

37.If x(n)={1,2,3,7,4,-1,5,..} then x(n/3)----- [ ]

A) {2,4,6,8,10,..} B) {1,0,2,0,3,0,4,0,5,0,6,0,..}

C){1,3,5,7,..} D){1,0,0,2,0,0,3,0,0,7,0,0,4,0,0..}

38.A cascade of a factor of D down sampler and a factor of I upsampler is [ ]
Interchangeable with no change in input and output relation if

A)D and I are integers B) D and I are co-prime

C) D and I are rational D) D and I are finite

39.The D-channel synthesis filter bank is the ---------------- of D-channel [ ]
analysis filter bank

A)dual B)single

C)triple D)none

40.filter banks may be ----------------filter banks or ---------------- filter banks [ ]

A)analysis, synthesis B)sampled, sequence

C)time, frequency D)none