MUTHAYAMMAL ENGINEERING COLLEGE

(An Autonomous Institution)

(Approved by AICTE, New Delhi, Accredited by NAAC & Affiliated to Anna University)
Rasipuram - 637 408, Namakkal Dist., Tamil Nadu

MUST KNOW CONCEPTS

MKC

ECE

2021-22

Course Code & Course Name : **19ECC09 & DIGITAL SIGNAL PROCESSING**

Year/Sem/Sec : **II/IV/A,B,C**

S.No	Term	Notation (Symbol)	Concept/Definition/Meaning/Units/ Equation/Expression	Units
UNIT I DISCRETE FOURIER TRANSFORM				
1	Signals		A Signal a function that Conveys information about a phenomenon	
2	Systems		It produces an output for a given input signal.	
3	Sinusoidal signals		X(t)=Asinωt X(t)=A cos ωt	
4	Analog signal		A continuous signal that contains time-varying quantities.	
5	Digital signal		It is a signal that is being used to represent data as a sequence of discrete values.	
6	Discrete time signal		A discrete time signal x (n) is a function of an independent variable that is an integer.	
7	Discrete time system		A discrete or an algorithm that performs some prescribed operation on a discrete time signal.	
8	Elementary discrete time signals		Unit Step signal, Unit Ramp, Unit Impulse and Exponential Signal.	
9	Exponential signal		x (n)=a^n where a is real x(n)-Real signal	
10	Classification of discrete time signals		1. Energy and power signals 2. Periodic and A periodic signals 3. Symmetric(even) and Ant symmetric (odd) signals	
11	Energy Signal		$E = \sum \lvert x(n) \rvert^2$	
12	Power Signal		P = Lt (1/2N+1) N-->∞	
13	DFT Equation		$X(k)= \sum_{n=0}^{N-1} x(n)\, e^{-j2\prod nk/N}$, 0<=k<=N-1	

14	IDFT Equation		$x(n)= 1/N \sum_{n=0}^{N-1} X(k)\, e^{j2\pi nk/N}$, 0<=n<=N-	
15	Properties of DFT		Periodicity, Linearity and symmetry, Multiplication of two DFTs, Circular convolution, Time reversal, Circular time shift and frequency shift, Complex conjugate, Circular correlation	
16	Two methods of Circular convolution		1. Concentric Circle Method 2. Matrix multiplication Method	
17	zero padding		To find N point DFT, have to add (N-L) zeros at the sequence x(n).	
18	Methods used for the sectional convolution		1. Overlap-add method and 2. Overlap-save method	
19	Overlap-add method		To each data block we append M-1 zeros and perform N point circular convolution.	
20	Overlap-save method		To each data block we add M-1 zeros at the initial point of input signal.	
21	FFT		The Fast Fourier Transform is an algorithm used to compute the DFT.	
22	Radix-2 FFT		If the number of output points N can be expressed as a power of 2 that is N=2M, where M is an integer, the algorithm is known as radix-2 algorithm.	
23	DIT algorithm		Decimation-In-Time algorithm is used to calculate the DFT of a N point sequence. The sequence x(n) is often splitted into smaller sub-sequences.	
24	DIF algorithm		The output sequence X(k) is divided into smaller and smaller sub-sequences , that is why the name Decimation In Frequency.	
25	Applications of FFT algorithm		1) Linear filtering 2)Correlation 2) Spectrum analysis	
UNIT II IIR FILTER DESIGN				
26	Filter		A **filter** is a circuit capable of passing certain frequencies while attenuating other frequencies.	
27	Types of filters based on impulse response		1. IIR filter 2. FIR filter	
28	IIR filter		IIR filters are easily realized recursively The round off noise in IIR filters is more.	
29	Butterworth filter		• The Magnitude response of Butterworth filter decreases monotonically as the frequency increases. • The Poles of the Butterworth filter lies along the circle.	
30	Low pass signal		A baseband signal is centered around DC (zero) frequency.	
31	High pass signal		A high-pass filter (HPF) is an electronic filter that passes signals with a frequency higher than a certain cutoff frequency.	
32	Band pass signal		A band of frequencies ranging from some non zero value to another non zero value.	

33	Structure of IIR filter		The design of IIR filter is realizable and stable. The impulse response h(n) for a realizable filter is h(n)=0 for n≤0	
34	Advantage of direct form II structure over direct form I structure		In direct form II structure, the number of memory locations required is less than that of direct form I structure.	
35	Design digital filters from analog filters		1. Map the desired digital filter specifications into those for an equivalent analog filter. 2. Derive the analog transfer function for the analog prototype.	
36	Procedures for digitizing the TF of analog filter		1. Impulse invariance method. 2. Bilinear transformation method.	
37	Impulse invariant method		The impulse response of resulting digital filter is a sampled version of the impulse response of the analog filter.	
38	Chebyshev Filter		• The Magnitude response of Chebyshev filter will not decrease monotonically with frequency because it exhibits ripples in pass band or stop band. • The Transition width is very small • The poles of chebyshev filter lies along the ellipse.	
39	Bilinear transformation		The mapping from the S-plane to the Z-plane is in bilinear transformation is $S=2\,(1-z^{-1})/T\,(1+z^{-1})$	
40	Response of analog filter		$y_a(t)=L^{-1}[H_a(s)X_a(s)]$	
41	sampled signal of analog filter output		$y_a(nT)=[L^{-1}[H_a(s)X_a(s)]\|_{t=nT}$	
42	Pre-warping		When the desired magnitude response is piece-wise constant over frequency, this compression can be compensated by introducing a suitable pre-scaling, or pre-warping the critical frequencies by using the formula	
43	Advantages of bilinear transformation		1. The bilinear transformation provides one-to-one mapping. 2. Stable continuous systems can be mapped into realizable, stable digital systems. 3. There is no aliasing.	
44	Disadvantages of bilinear transformation		1. The mapping is highly non-linear producing frequency, compression at high frequencies. 2. Neither the impulse response nor the phase response of the analog filter is preserved in a digital filter obtained by bilinear transformation.	
45	Advantage of cascade realization		Quantization errors can be minimized if we realize an LTI system in cascade form.	
46	Signal flow graph		It is a graphical representation of the relationships between the variables of a set of linear difference equations.	
47	Transposition theorem		If we reverse the directions of all branch	

			transmittance and interchange the input and output in the flowgraph, the system function remains unchanged.	
48	Transposed structure		1. Reverse the directions of all branches in the signal flow graph 2. Interchange the input and outputs. 3. Reverse the roles of all nodes in the flow graph.	
49	Important parameters of band pass		1.center frequency ω_c 2. ω_u upper critical frequency 3. ω_l low critical frequency	
50	Sampling frequency		T=1/fs $T=\frac{1}{f_s}=\frac{2\pi}{\omega_s}$ $T=\frac{1}{f_s}=\frac{2\pi}{\omega_s}$ $T=\frac{1}{f_s}=\frac{2\pi}{\omega_s}$	
		UNIT III FIR Filter Design		
51	Filter		A filter is a circuit capable of passing certain frequencies while attenuating other frequencies.	
52	Types of filters based on impulse response		1. IIR filter 2. FIR filter	
53	Types of filters based on frequency response		1. Low pass filter 2. High pass filter 3. Band pass filter 4. Band reject filter	
54	Band pass signal		A band of frequencies ranging from some non zero value to another non zero value.	
55	Low pass signal		A baseband signal is centered around DC (zero) frequency.	
56	High pass signal		A high-pass filter (HPF) is an electronic filter that passes signals with a frequency higher than a certain cutoff frequency.	
57	FIR filter		FIR filters can be realized recursively and non-recursively. Errors due to round off noise are less severe in FIR filter	
58	IIR filter		IIR filters are easily realized recursively The round off noise in IIR filters is more.	
59	Linear phase FIR filter		Phase delay, α = (N-1)/2 (i.e., phase delay is constant)Impulse response, h(n) = h(N-1-n)	
60	Anti Symmetric FIR Filters		Impulses occur at the mirror image in the first quadrant and third quadrant or second quadrant and Fourth quadrant or both.	
61	Optimum equiripple design criterion		The Optimum Equiripple design Criterion is used for designing FIR Filters with Equal level filteration throughout the Design.	
62	Design techniques of FIR filters		(1) Window method (2) Frequency sampling method (3) Optimal or minimax design	
63	Reason that FIR filter is always stable		FIR filter is always stable because all its poles are at the origin.	
64	Gibb's phenomenn		One possible way of finding an FIR filter that approximates H(w) would be to truncate the infinite Fourier series at n=±(N-1/2).	

65	Desired frequency response $H_d(w)$		$h_d(n)=1/2\pi\int_{-\pi}^{\pi} H_d(w)\, e^{jwn}\, dw$	
66	Transfer function of the realizable filter		$H(z)=z^{-(N-1)/2}\, [h(0)+\sum_{n=0}^{(N-1)/2} h(n)(z^n+z^{-n})]$	
67	Types of windowing techniques		1. Rectangular window 2. Hamming window 3. Hanning window 4. Bartlett window 5. Kaiser window	
68	Equation for Rectangular window		$W(n) = 1$ for $0 \le n \le M-1$ $= 0$ otherwise	
69	Equation for Hamming window		$W_H(n)= 0.54-0.46 \cos(2\pi n/M-1)$ for $0 \le n \le M-1$ $= 0$ otherwise	
70	Equation for Hanning window		$W_{Hn}(n)= 0.5[1- \cos (2 \pi n /M-1]$ for $0 \le n \le M-1$ $= 0$ otherwise	
71	Equation for Bartlett window		$WT(n)= 1-\{2\lvert n-(M-1)/2\rvert\}/(M-1)$ for $0 \le n \le M-1$ $= 0$ otherwise otherwise	
72	Equation for Blackman window		$W_H(n)= 0.42-0.5 \cos (2 \pi n /M-1)+0.08 \cos (4 \pi n /M-1)$ $= 0$ otherwise	
73	Merits of FIR filters		1. FIR Filter is always stable. 2. FIR Filter with exactly linear phase can easily be designed.	
74	Demerits of FIR filters		1. High Cost. 2.Require more Memory	
75	Features of hanning window spectrum		1. The main lobe width is equal to $8\pi/N$. 2. The maximum side lobe magnitude is -31db. 3. The side lobe magnitude decreases with increasing.	
UNIT IV FINITE WORD LENGTH EFFECT				
76	Types of arithmetic in digital systems		Fixed point arithmetic, floating point, block floating point arithmetic	
77	Fixed point number		In fixed point number the position of a binary point is fixed. The bit to the right represent the fractional part and those to the left is integer part.	
78	Types of fixed point arithmetic		sign magnitude, 1's complement, 2's complement	
79	Sign magnitude representation		The leading binary digit is used to represent the sign. If it is equal to 1 the number is negative, otherwise it is positive.	
80	1's complement form		Complement all the bits of the positive number	
81	2's complement form		Complement all the bits of the positive number and add 1 to the LSB	
82	Advantages of floating pint representation		1. Large dynamic range 2. Overflow is unlikely	

83	Quantization errors		1. Input quantization errors 2. Coefficient quantization errors 3.Product quantization errors	
84	Input quantization error		The filter coefficients are computed to infinite precision in theory	
85	Product quantization error		The product quantization errors arise at the output of the multiplier	
86	Input quantization error		The input quantization errors arise due to A/D conversion	
87	Quantization methods		Truncation and Rounding	
88	truncation		A process of discarding all bits less significant than LSB that is retained	
89	Rounding		Rounding a number to b bits is accomplished by choosing a rounded result as the b bit number closest number being unrounded	
90	Limit cycles		In recursive system these nonlinearities often cause periodic oscillation to occur in the output, even when input sequence is zero or some nonzero value	
91	Sampling		The reduction of a continuous-time signal to a discrete-time signal	
92	sampling rate		The average number of samples obtained in one thus $f_s = 1/T$	
93	Under sampling		A bandpass signal is sampled slower than its Nyquist rate, the samples are indistinguishable from samples of a low-frequency alias ofthe high-frequency signal.	
94	Over sampling		Oversampling is used in most modern analog-to-digital converters to reduce the distortion introduced by practical digital-to-analog converters	
95	Sampling Theorem		fs>=2fm	
96	Nyquist rate		fs=2fm	
97	Types of limit cycle behavior of DSP		1. Zero limit cycle behavior 2. Over flow limit cycle behavior	
98	Overflow limit cycle		A high-level oscillation that can exist in an otherwise stable filter due to the nonlinearity associated with the overflow of internal filter calculations	
99	Methods to prevent overflow		1. Saturation arithmetic 2. Scaling	
100	Safe Scaling		v (n) = f(n) * x (n)	
UNIT V DSP APPLICATIONS				
101	Multirate signal processing		Data communication require more than one sampling rate for processing data in such a cases increase and/or decrease the sampling rate.	

102	Examples of multirate digital systems		Decimator and interpolator	
103	Input output relationship for a decimator		Fy = Fx/D	
104	Input output relationship for an interpolator		Fy = IFx	
105	Aliasing		The original shape of the signal is lost due to under sampling. This is called aliasing	
106	Avoid Aliasing		Placing a LPF before down sampling	
107	How sampling rate be converted by a factor I/D		Cascade connection of interpolator and decimator	
108	Sub-band coding		It is an efficient coding technique by allocating lesser bits for high frequency signals and more bits for low frequency signals.	
109	Up sampling		Increasing the sampling rate	
110	Down sampling		Decreasing the sampling rate	
111	Decimator		own sampling and a anti-aliasing filter	
112	Interpolator		An anti-imaging filters and Up sampling	
113	Sampling rate conversion		Changing one sampling rate to other sampling rate is called sampling rate conversion	
114	Sections of QMF		Quadrature Mirror Filters-Analysis section and synthesis section	
115	Define mean		Mxn=E[xn]=intg xpxn(x,n) dx	
116	Define variance		Zxn2=E[{xn=mxn}2]	
117	Cross correlation of random process		R xy (n.m)= intxy*pxn,ym(x,n,y,m)dxdy	
118	DTFT of cross correlation		Txy(e jw) = x rxy(l) e jwl	
119	Cutoff frequency of Decimator		Pi/M where M is the down sampling factor	
120	Cutoff frequency of Interpolator		Pi/L where L is the UP sampling factor.	
121	Difference in efficient transversal structure		Number of delayed multiplications are reduced	
122	Shape of the white noise spectrum		Flat frequency spectrum.	
123	Sub-band coding		transform coding that breaks a signal into a number of different frequency bands, typically by using a fast Fourier transform, and encodes each one independently	
124	Channel vocoder		A bank of filters that breaks two incoming sound sources into compatible frequency regions.	

125	Encoding of Waveforms		Data, Speech, Image	
PLACEMENT QUESTIONS				
126	Define DSP		Digital signal processing improves the sensitivity of a receiving unit.	
127	Signals		A signal is a function that conveys information about a phenomenon.	
128	Systems		It produces an output for a given input signal.	
129	continuous-time signal		A signal of continuous amplitude and time is known as a continuous-time signal or an analog signal	
130	continuous-time system		The signals at input and output are continuous-time signal	
131	Applications of DSP		Speech processing, Communication, Biomedical signal processing, Image processing, Radar signal processing, Sonar signal processing etc.	
132	Advances of DSP		1. The programs can be modified easily for better performance. 2. Better accuracy can be achieved. 3. The digital signal can be easily stored and transported.	
133	Analog signal		A continuous signal that contains time-varying quantities.	
134	Digital signal		It is a signal that is being used to represent data as a sequence of discrete values.	
135	Discrete time signal		A discrete time signal x (n) is a function of an independent variable that is an integer.	
136	Discrete time system		A discrete or an algorithm that performs some prescribed operation on a discrete time signal.	
137	Methods to prevent overflow		1. Saturation arithmetic 2. Scaling	
138	Sampling		The reduction of a continuous-time signal to a discrete-time signal	
139	Up sampling		Increasing the sampling rate	
140	Down sampling		Decreasing the sampling rate	
141	Quantization		Transforming a continuously valued input into a representation that assumes one out of a finite set of values	
142	Types of limit cycle behavior of DSP		1. Zero limit cycle behavior 2. Over flow limit cycle behavior	
143	truncation		Truncation is a process of discarding all bits less significant than LSB that is retained	
144	Rounding		Rounding a number to b bits is accomplished by choosing a rounded result as the b bit number closest number being unrounded.	
145	zero padding		To find N point DFT, have to add (N-L) zeros at the sequence x(n).	

146	FFT		The Fast Fourier Transform is an algorithm used to compute the DFT.	
147	Pre-warping		When the desired magnitude response is piece-wise constant over frequency, this compression can be compensated by introducing a suitable pre-scaling, or pre-warping the critical frequencies	
148	Signal flow graph		It is a graphical representation of the relationships between the variables of a set of linear difference equations.	
149	Region Of Convergence		The region of convergence (ROC) of X(Z) the set of all values of Z for which X(Z) attain final value.	
150	Applications of FFT algorithm		Linear filtering, Correlation Spectrum analysis	

Faculty Team Prepared **Signatures**

1. **Dr.T.R. Ganeshbabu, Prof/ECE**
2. **Mr.M.Eswaramoorthy,AP/ECE**
3. **Ms.K.Shenbagadevi,AP/ECE**

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