

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



MKC

2021-22

MUST KNOW CONCEPTS

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
	U	NIT I DISCI	RETE FOURIER TRANSFORM	
1	Signals	7	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	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	DESIG	A discrete time signal x (n) is a function of an independent variable that is an integer.	
7	Discrete time system	Es	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		 Energy and power signals Periodic and A periodic signals Symmetric(even) and Ant symmetric (odd) signals 	
11	Energy Signal		$\mathbf{E} = \sum \mathbf{x}(\mathbf{n}) ^2$	
12	Power Signal		P = Lt (1/2N+1) N> ∞	
13	DFT Equation		$\begin{array}{c} N-1 \\ X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2 \prod nk/N} , 0 <= k <= N-1 \end{array}$	

	1		
14	IDFT Equation	N-1 x(n)= $1/N \sum_{n=0}^{N-1} X(k) e j2 \prod nk/N$, 0<=n<=N-	
		n=0 Periodicity, Linearity and symmetry, Multiplication	
15	Properties of DFT	of two DFTs, Circular convolution, Time reversal, Circular time shift and frequency shift, Complex conjugate, Circular correlation	
	T. (1, 1, 6	conjugate, Circular correlation	
16	Two methods of Circular convolution	1. Concentric Circle Method	
	Circular convolution	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 and2. 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.The Fast Fourier Transform is an algorithm used to	
21	FFT	compute the DFT.	
		If the number of output points N can be expressed	
22	Radix-2 FFT	as a power of 2 that is N=2M, where M is an	
		integer, the algorithm is known as radix-2	
		algorithm.	
		Decimation-In-Time algorithm is used to calculate	
23	DIT algorithm	the DFT of a N point sequence. The sequence $x(n)$	
		is often splitted into smaller sub-sequences.	
24		The output sequence X(k) is divided into smaller	
24	DIF algorithm	and smaller sub-sequences , that is why the name	
		Decimation In Frequency.	
25	Applications of FFT algorithm	 Linear filtering 2)Correlation Spectrum analysis 	
		UNIT II IIR FILTER DESIGN	
26	Filter	DESIG A filter is a circuit capable of passing certain	
		frequencies while attenuating other frequencies.	
27	Types of filters based	2. FIR filter	
21	on impulse response		
	IIR filter	IIR filters are easily realized recursively	
28		The round off noise in IIR filters is more.	
		The round off holds in first fixers is historie. The Magnitude response of Butterworth	
		filter decreases monotonically as the	
29	Butterworth filter	frequency increases.	
		• The Poles of the Butterworth filter lies	
		along the circle.	
30	Low pass signal	A baseband signal is centered around DC (zero)	
50	LOW Pass signal	frequency.	
		A high-pass filter (HPF) is an electronic filter that	
31	High pass signal	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.	

		The design of IIR filter is realizable and stable.	
33	Structure of IIR filter	The impulse response $h(n)$ for a realizable filter is	
55	Structure of fire filter	$h(n)=0$ for $n\leq 0$	
	Advantage of direct	In direct form II structure, the number of memory	
34	form II structure over	locations required is less than that of direct form I	
54	direct form I structure	structure.	
	Design digital filters	1. Map the desired digital filter specifications into	
35	Design digital filters	those for an equivalent analog filter.	
	from analog filters	2. Derive the analog transfer function for the analog	
	Procedures for	prototype.	
26		1. Impulse invariance method.	
36	digitizing the TF of	2. Bilinear transformation method.	
	analog filter		
27	Impulse invariant	The impulse response of resulting digital filter is a	
37	method	sampled version of the impulse response of the	
		analog filter.	
		• The Magnitude response of Chebyshev filter	
		will not decrease monotonically with	
•	Chebyshev Filter	frequency because it exhibits ripples in pass	
38	Chebyshev I hter	band or stop band.	
		• The Transition width is very small	
		• The poles of chebyshev filter lies along the	
		ellipse.	
	וית	The mapping from the S-plane to the Z-plane is in	
39	Bilinear	bilinear transformation is	
	transformation	$S=2(1-z^{-1})/T(1+z^{-1})$	
40	Response of analog	$y_a(t) = L^{-1}[H_a(s)X_a(s)]$	
40	filter	$y_a(l) - L [\Pi_a(3)X_a(3)]$	
41	sampled signal of	$y_a(nT) = [L^{-1}[H_a(s)X_a(s)]]_{t=nT}$	
71	analog filter output		
		When the desired magnitude response is piece-wise	
		constant over frequency, this compression can be	
42	Pre-warping	compensated by introducing a suitable pre-scaling,	
		or pre-warping the critical frequencies by using the	
		formula and a a a a a a a a a a a a a a a a a	
		1. The bilinear transformation provides one-to-one	
	Advantages of bilinear	mapping.	
43	transformation	2. Stable continuous systems can be mapped into	
	transiormation	realizable, stable digital systems.	
		3. There is no aliasing.	
		1. The mapping is highly non-linear producing	
	Dianduantagas	frequency, compression at high frequencies.	
44	Disadvantages of bilinear transformation	2. Neither the impulse response nor the phase	
	onmear transformation	response of the analog filter is preserved in a	
		digital filter obtained by bilinear transformation.	
4 5	Advantage of cascade	Quantization errors can be minimized if we realize	
45	realization	an LTI system in cascade form.	
		It is a graphical representation of the relationships	
46	Signal flow graph	between the variables of a set of linear difference	
		equations.	
		If we reverse the directions of all branch	
47	Transposition theorem		

Image: constraint the set of the input and output in the flow graph, the system function remains unchanged.48Transposed structure1. Reverse the directions of all branches in the signal flow graph49Important parameters of band pass2. Interchange the input and outputs. 3. Reverse the roles of all nodes in the flow graph.49Important parameters of band pass \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} \mathcal{O}_{L} 50Sampling frequency $\mathcal{T} = \frac{1}{2}$ \mathcal{O}_{L} \mathcal{O}				transmittance and interesting the ' (1 (]
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Fourier series at $n=\pm(N-1/2)$.	64	r			
				Fourier series at $n=\pm(N-1/2)$.	

	1	
65	Desired frequency response H _d (w)	$h_{d}(n) = \frac{\pi}{1/2\pi \int} H_{d}(w) e^{jwn} dw$ $-\pi$
66	Transfer function of the realizable filter	$\begin{array}{c} (N-1)/2 \\ H(z)=z-(N-1)/2 \left[h(0)+\sum h(n)(zn+z-n)\right] \\ n=0 \end{array}$
67	Types of windowing techniques	 Rectangular window Hamming window Hanning window Bartlett window Kaiser window
68	Equation for Rectangular window	$W(n) = 1 \qquad \text{for } 0 \le n \le M-1$ = 0 \text{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	$WT(n) = 1 - \{2 n - (M-1)/2 \}/(M-1) \text{ for } 0 \le n \le M-1$ = 0 otherwise otherwise
72	Equation for Blackman window	$W_{\rm H}(n) = 0.42-0.5 \cos (2 \pi n / M-1) + 0.08 \cos (4 \pi n / M-1) = 0 \qquad \text{otherwise}$
73	Merits of FIR filters	 FIR Filter is always stable. 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	 The main lobe width is equal to 8π/N. The maximum side lobe magnitude is -31db. 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

			1. Input quantization errors 2. Coefficient	
83	Quantization errors		quantization errors	
00	Qualitization errors		3.Product quantization errors	
	Input quantization		The filter coefficients are computed to infinite	
84	error		precision in theory	
0.5	Product quantization		The product quantization errors arise at the output	
85	error		of the multiplier	
0.6	Input quantization		The input quantization errors arise due to A/D	
86	error		conversion	
87	Quantization methods		Truncation and Rounding	
07	Quantization methods			
88	truncation		A process of discarding all bits less significant than	
			LSB that is retained	
			Rounding a number to b bits is accomplished by	
89	Rounding		choosing a rounded result as the b bit number	
			closest number being unrounded	
00	Limit origina		In recursive system these nonlinearities often cause	
90	Limit cycles		periodic oscillation to occur in the output, even	
			when input sequence is zero or some nonzero valueThe reduction of a continuous-time signal to	
91	Sampling		The reduction of a continuous-time signal to a discrete-time signal	
			The average number of samples obtained in one	
92	sampling rate	-	thus $f_s = 1/T$	
			A bandpass signal is sampled slower than	
	Under		its Nyquist rate, the samples are indistinguishable	
93	sampling		from samples of a low-frequency alias of the high-	
	B		frequency signal.	
		N N	Oversampling is used in most modern analog-to-	
94	Over		digital converters to reduce the distortion	
	sampling		introduced by practical digital-to-analog converters	
95	Sampling Theorem	<u> </u>	fs>=2fm	
95	Sampling Theorem			
96	Nyquist rate		fs=2fm	
	<i>J</i> 1	DESIG		
97	Types of limit cycle	_	1. Zero limit cycle behavior	
9/	behavior of DSP	ES	2. Over flow limit cycle behavior	
			A high-level oscillation that can exist in an	
			otherwise stable filter due to the nonlinearity	
98	Overflow limit cycle		associated with the overflow of internal filter	
			calculations	
			1. Saturation arithmetic	
99	Methods to prevent		2. Scaling	
	overflow			
100	Safa Sacling		v(n) = f(n) * x(n)	
100	Safe Scaling			
		UNIT	V DSP APPLICATIONS	
	Multirate signal		Data communication require more than one	
101	processing		sampling rate for processing data in such a cases	
	processing		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	Y	Mxn=E[xn]=intg xpxn(x,n) dx
116	Define variance		$Zxn2=E[\{xn=mxn\}2]$
117	Cross correlation of random process	DESTG	R xy (n.m) = intxy*pxn, ym(x,n,y,m)dxdy
118	DTFT of cross correlation	ES	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
	wavelonnis	PLAC	CEMENT 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		 The programs can be modified easily for better performance. Better accuracy can be achieved. 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	DESIG	The reduction of a continuous-time signal to a discrete-time signal T
139	Up sampling	Ez	Increasing the sampling rate
140	Down sampling	E3	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		 Zero limit cycle behavior 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	



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