Course Plan

Semester: **6 - Semester**	Year: **2019**
Course Title: **DIGITAL SIGNAL PROCESSING**	Course Code: **EC115**
Semester End Examination : **70**	Continuous Internal Evaluation: **30**
Lesson Plan Author: Ms. P ANURADHA	Last Modified Date: 25-11-2018

Course Outcomes (COs):

At the end of the course the student should be able to:

1. Describe discrete time signals & systems and represent in frequency domain
2. Compute dft using fft algorithms and derive dft properties
3. Design iir digital filters using various techniques
4. Design fir digital filters using various techniques
5. Analyze multirate signal processing techniques

Course Articulation Matrix: Mapping of Course Outcomes (COs) with Program Outcomes (POs)

Course Outcomes (COs) / Program Outcomes (POs)	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**	**9**	**10**	**11**	**12**	**PSO-1**	**PSO-2**
1. Describe discrete time signals & systems and represent in frequency domain	**3**	**3**	**3**	**3**								**3**		
2. Compute dft using fft algorithms and derive dft properties	**3**	**3**		**2**										
3. Design iir digital filters using various techniques	**3**	**3**	**3**	**2**								**3**		
4. Design fir digital filters using various techniques	**3**	**3**	**3**	**3**								**3**		
5. Analyze multirate signal processing techniques	**3**	**3**		**2**								**3**		

Course Content

Content	Hrs
Unit - 1	
Chapter No. 1 - Introduction Discrete time signals and sequences, linear shift invariant systems, stability and causality. Frequency domain representation of discrete time signals and systems	9.00 hrs
Unit - 2	
Chapter No. 2 - Discrete Fourier Transform (DFT) and Fast Fourier Transform Discrete-Time Fourier transform, computation of DFT, properties of DFT, linear convolution, circular convolution of sequences using DFT. Fast Fourier Transform: Fast Fourier transforms (FFT)-Radix2 decimation in time and decimation in frequency FFT algorithms, inverse FFT.	11.00 hrs
Unit - 3	
Chapter No. 3 - IIR Digital Filters Analog filter approximations-Butterworth and Chebyshev, design of IIR digital filters from analog filters, step and impulse invariant techniques. Bilinear transformation method, Realization of IIR Digital Filters – direct, Canonic, Cascade and Parallel forms.	13.00 hrs
Unit - 4	
Chapter No. 4 - FIR Digital Filters Characteristics of FIR digital filters, frequency response, Design of FIR digital filters: Fourier method, window techniques, frequency sampling technique, comparison of IIR and FIR filters. Realization of FIR Digital Filters – transversal structure, linear phase realization.	9.00 hrs
Unit - 5	
Chapter No. 5 - Multirate Digital Signal Processing Introduction, down sampling-decimation, up sampling- interpolation, sampling rate conversion, implementation of narrow band low pass filter, filter banks.	6.00 hrs

Text Books (List of books as mentioned in the approved syllabus)

1. John G. Proakis and Dimitris G. Manolakis, Digital Signal Processing, Principles, Algorithms and Applications, 4th, Pearson Education / PHI, 2007
2. Mithra, Digital Signal Processing, 3rd, McGraw Hill Publications, 2008

References

1. Li Tan, Digital Signal Processing- Fundamentals and Applications, 2nd, Elsevier, 2008
2. Robert J. Schilling and Sandra L. Harris, Fundamentals of Digital Signal Processing Using Matlab, 2nd, Thomson, 2007
3. Nagoorkhani A, Digital Signal Processing, 2nd, TMH, 2012
4. Ramesh Babu P, Digital Signal Processing, 4th, SciTech, 2013

Chapterwise Plan

Course Code and Title: **EC115 / DIGITAL SIGNAL PROCESSING**	
Chapter Number and Title: **1 - Introduction**	

Learning Outcomes:-

At the end of the topic the student should be able to:

	Topic Learning Outcomes
1	**Define** discrete time signals and sequences
2	**Explain** properties of systems
3	**Derive** stability and causality conditions
4	**Find** frequency response of discrete time signals and systems

Lesson Schedule

Lecture No. - Portion covered per hour
1. introduction
2. Basic Elements of Digital Signal Processing: Discrete time signals & sequences, representations
3. Classification of Discrete time signals
4. Classification of Discrete Linear time systems: Linearity ,Time Invariant, Time variant
5. stability, and causality
6. Linear convolution
7. Discrete Time Fourier Transform(DTFT): Definition of DTFT
8. Frequency domain representation of discrete time signals
9. Magnitude and phase transfer function

Review Questions

Sl.No. - Questions
1. What is the causal and non-causal system, time variant and time invariant? $y(n)=x(n-3)+x(n-1)$ check w causality, time in variant and time variant?

2. What is periodic and aperiodic signal and determine the periodicity for multiple sig $x(n)=cos(\pi n/4)+ cos(\pi n/6)+sin(\pi n/8-\pi/3)$	
3. What is energy signal and check $y(n)=(1/4)^n u(n)$ is energy signal or not? What is power signal and check is	
4. $y(n)=x(n)^2$ check linearity, causality, time invariant, stability?	
5. find the frequency response and sketch the magnitude and phase response? $y(n)=x(n)+0.81x(n-1)+0.81x$	
6. Determine the range of values of the parameter 'a' for which the linear time invariant system with impulse res	
7. Consider the special case of a finite duration sequence given as x(n)={2,4,0,3}. Resolve the sequence x(n) int	
8. Decompose the following signal into even and an odd components. Is the decomposition unique? x(n)={2,3,4,	

Course Code and Title: **EC115 / DIGITAL SIGNAL PROCESSING**	
Chapter Number and Title: **2 - Discrete Fourier Transform (DFT) and Fast Fourier Transform**	

Learning Outcomes:-

At the end of the topic the student should be able to:

	Topic Learning Outcomes
1	**Explain** Discrete-Time Fourier transform
2	**Compute** Discrete Fourier Transform
3	**List out** properties of DFT
4	**Compare** linear convolution and circular convolution
5	**Derive** radix 2 FFT algorithms
6	**Find** IFFT from FFT algorithms

Lesson Schedule

Lecture No. - Portion covered per hour
1. Discrete Fourier transforms
2. Inverse Discrete Fourier Transform (IDFT)
3. Properties of DFT
4. Properties of DFT1
5. comparison between convolution and linear convolution , linear convolution of sequences using DFT
6. Fast Fourier Transform (FFT): Computational Complexity of DFT Introduction to FFT
7. Radix-2 Decimation in Time Algorithm

8. DIT FFT Algorithm
9. DIT FFT Algorithm1
10. DIT FFT Algorithm2
11. Inverse FFT

Review Questions

Sl.No. - Questions
1. Find the circular convolution of the sequence (a) $x(n)=\{1,2,1\}; h(n)=\{1,-2,2\}$ (b) $x(n)=\{1,2,3,4\}; h(n)$
2. Compute 8-point DFT of a given x(n) by radix 2 DIT-FFT method also sketch the magnitude and phase. $x(n$
3. Find the 8-point DFT of the given sequence $x(n)=\begin{cases}1 & 0\le n\le 7;\\ 0 & otherwise.\end{cases}$
4. Measure the linear convolution of the sequence $x(n)=\{1,-1,1,-1\}$ and $h(n)=\{1,2,3,4\}$ using DFT method?
5. Find the inverse of Fourier transform (or) IDFT of $X(k)=\{6,-2+2j,-2,-2-2j\}$
6. Consider a causal LTI system with system function $H(z)=\frac{1}{1-0.5Z^{-1}}$. The output y(n) of the system is know available, can you develop a 64-point DFT method to recover the sequence x(n), 0<n<65? Can you recover all va
7. Consider the eight-point decimation-in-time (DIT) flow graph How many paths lead from the input to a given output sample?
8. The first five points of the 8-point DFT of a real valued sequence are {0.25,0.125-j0.3018,0,0.125-j0.051 points.

Course Code and Title: **EC115 / DIGITAL SIGNAL PROCESSING**	
Chapter Number and Title: **3 - IIR Digital Filters**	

Learning Outcomes:-

At the end of the topic the student should be able to:

	Topic Learning Outcomes
1	**Design** of IIR Digital filters from analog filters
2	**Discuss** various transformations techniques
3	**Explain** analog filter designing methods
4	**Compare** Butterworth and Chebyshev filters
5	**Construct** IIR digital filters with all realization techniques

Lesson Schedule

Lecture No. - Portion covered per hour
1. Infinite Impulse Response (IIR) Filters: Introduction
2. Design of IIR Digital filters from analog filters Transformations techniques
3. Impulse Invariant Transformation method
4. Impulse Invariant Transformation method1
5. Bilinear transformation
6. Bilinear transformation1
7. Necessity of Filter Approximation, Analog filter design: Butterworth approximations
8. Butter worth approximations: Order of the filter, Design steps
9. Butter worth approximations
10. Frequency transformations in analog domain
11. comparison of butterworth and chebyshev filters
12. realization of digital filters – direct
13. Canonic, Cascade, parallel forms

Review Questions

Sl.No. - Questions
1. Design a digital chebyshev filter to meet the constraints $\begin{matrix} 1/\sqrt{2} \le \lvert H(\omega) \rvert \le 1, & 0 \le \omega \le 0.2\pi \\ 0 \le \lvert H(\omega) \rvert \le 0.1, & 0.5\pi \le \omega \le \pi \end{matrix}$ by using bilinear tra
2. Design a butter worth filer using the impulse invariance method for the following specification $\begin{matrix} 0.8 \le \lvert H(e^{j\omega}) \rvert \\ \lvert H(e^{j\omega}) \rvert \le 0.2 \end{matrix}$
3. Explain the procedure for impulse invariant method of designing IIR filter.
4. Determine cascade and parallel realization for the system having difference equation $y(n) + 0.1y(n-1) - 0.2y$
5. For the analog transformation function $H(s) = 2/(s+1)(s+3)$ .determine H(z) using bilinear transformation
6. Design a single-pole low pass digital filter with a 3-dB bandwidth of 0.2π , using the bilinear $H(s) = \frac{\Omega_c}{s+\Omega_c}$
7. Convert the analog filter with system function $H_a(s) = \frac{s+0.1}{(s+0.1)^2+16}$ into a digital IIR filter by means of the

to have a resonant frequency of $w_r=\pi/2$.
8. Determine the system function H(z) of the lowest order chebyshev digital filter that meets the following $0 \le \lvert\omega\rvert \le 0.24\pi$. At least 50dB attenuation in the stop band $0.35 \le \lvert\omega\rvert \le \pi$. Use the bilinear transformation

Course Code and Title: **EC115 / DIGITAL SIGNAL PROCESSING**	
Chapter Number and Title: **4 - FIR Digital Filters**	

Learning Outcomes:-

At the end of the topic the student should be able to:

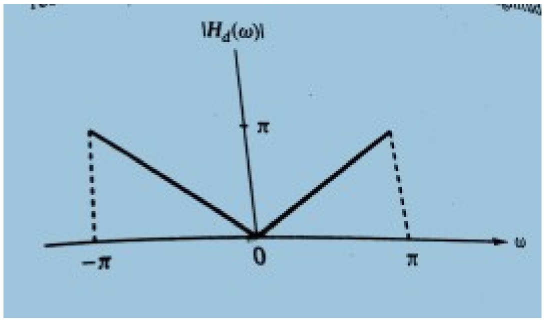
	Topic Learning Outcomes
1	**Discuss** Characteristics of FIR Digital Filters
2	**Derive** frequency response of Linear phase FIR filters
3	**Design** of FIR Digital Filter using Fourier series method and window techniques
4	**Compare** IIR and FIR filters
5	**Construct** FIR digital filters using transversal and linear phase realization technique

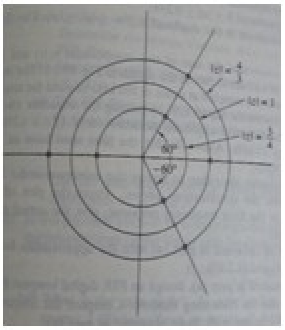
Lesson Schedule

Lecture No. - Portion covered per hour
1. FIR DIGITAL FILTERS : Introduction, Characteristics of FIR Digital Filters
2. frequency response of Linear phase FIR filters
3. Design of FIR Digital Filter Fourier series method: LPF
4. Design of FIR Digital Filter Fourier method: HPF
5. Design of Design of Linear phase FIR filters using windows Techniques: Rectangular window, Triangular win
6. Hamming window, Hanning window
7. Frequency Sampling technique
8. Comparison of IIR and FIR filters Realization of FIR Digital Filters – transversal structure
9. Linear phase realization.

Review Questions

Sl.No. - Questions

1. Design an ideal high pass filter with a frequency response. $H_d(e^{j\omega}) = \begin{cases} 1 & for\ \frac{\pi}{4} \le \lvert\omega\rvert \le \pi \\ 0 & for\ \lvert\omega\rvert \le \frac{\pi}{4} \end{cases}$ Find the valu window. Find H(z) and determine the magnitude response.
2. Design an ideal Low pass filter with a frequency response $H_d(e^{j\omega}) = \begin{cases} 1 & for\ \frac{-\pi}{2} \le \omega \le \frac{\pi}{2} \\ 0 & for\ \frac{\pi}{2} \le \omega \le \pi \end{cases}$ Find the valu the magnitude response using Blackman window for N=11 and plot the frequency response
3. How is the design of linear phase FIR filter done by frequency sampling method? Explain. Compare IIR and
4. Determine the condition for linear phase FIR filters Determine the cascade realization of system function $H($
5. Evaluate the given system function $H(z) = \frac{1}{2} + \frac{1}{3}z^{-1} + z^{-2} + \frac{1}{4}z^{-3} + z^{-4} + \frac{1}{3}z^{-5} + \frac{1}{2}z^{-6}$
6. The ideal analog differentiator is described by $y_a(t) = \frac{dx_a(t)}{dt}$,where xa(t)is the input, $x_a(t) = e^{j2\pi Ft}$ an and phase response of an ideal analog differentiator band-limited to B hertz. The ideal digital differentiator is d this definition by comparing the frequency response, with that in part 1.
7. Use the window method with hamming window to design a 21-tap differentiator as shown in figure. Compute of the resulting filter 

8. Consider the pole-zero plot shown in figure Does it represent an FIR filter? 

Course Code and Title: **EC115 / DIGITAL SIGNAL PROCESSING**
Chapter Number and Title: **5 - Multirate Digital Signal Processing**

Learning Outcomes:-

At the end of the topic the student should be able to:

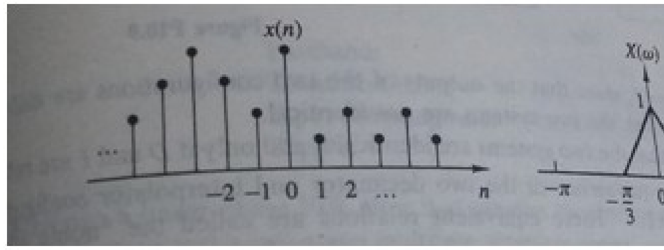
	Topic Learning Outcomes
1	**Derive** Spectrum of the down sampled signal
2	**Explain** interpolation concepts
3	**Discuss** sampling rate conversion
4	**Explain** narrow band low pass filter and filter banks

Lesson Schedule

Lecture No. - Portion covered per hour
1. Introduction, Down sampling ,Spectrum of the down sampled signal
2. Decimation
3. Up sampling i.e. interpolation
4. Implementation of sampling rate conversion.
5. Applications of multi rare signal processing
6. implementation of narrow band low pass filter , filter banks

Review Questions

Sl.No. - Questions
1. Explain spectrum of Down sampled signal with diagrams
2. Design a two stage decimator for the following specifications: Sampling rate of the input sampling rate = 10

band = 0 to 50 Hz, Transition band = 50 to 75 Hz, pass band ripple = 10-1, stop band ripple=10-3
3. Explain spectrum of Up sampled-signal with diagrams
4. Explain sampling rate conversion. What is the use of Anti imaging filter?
5. List out the applications of multi-rate signal processing
6. Sampling x(n) with a sampling period D=2 results to the signal $x_s(n)=\begin{cases} x(n), & n=0,\pm2,\pm4,\_\_ \\ 0 & n=\pm1,\pm3,\pm5,\_\_ \end{cases}$ Compute  transform Xs(w).Can we reconstruct x(n) from xs(n)? How?
7. Design a two stage interpolator for the following specifications: Sampling rate of the input sampling rate = 10 band = 0 to 50 Hz, Transition band = 50 to 75 Hz, pass band ripple = 10-1, stop band ripple=10-3