Techno India NJR Institute of Technology

Course File

Signal & System (3EC4-05)

Session (2022-23)

Dr. Vivek Jain
(Associate Professor)
Department of ECE

3EC4-05: Signals & Systems

3 Credits — **Max. Marks: 150 (IA:30, ETE:120)**
3L:0T:0P — **End Term Exam: 3 Hours**

SN	Contents	Hours
1	Energy and power signals, continuous and discrete time signals, continuous and discrete amplitude signals. System properties: linearity: additivity and homogeneity, shift-invariance, causality, stability, realizability.	6
2	Linear shift-invariant (LSI) systems, impulse response and step response, convolution, input output behavior with aperiodic convergent inputs. Characterization of causality and stability of linear shift-invariant systems. System representation through differential equations and difference equations	7
3	Periodic and semi-periodic inputs to an LSI system, the notion of a frequency response and its relation to the impulse response, Fourier series representation, the Fourier Transform, convolution/multiplication and their effect in the frequency domain, magnitude and phase response, Fourier domain duality. The Discrete-Time Fourier Transform (DTFT) and the Discrete Fourier Transform (DFT). Parseval's Theorem. The idea of signal space and orthogonal bases	8
4	The Laplace Transform, notion ofeigen functions of LSI systems, a basis of eigen functions, region of convergence, poles and zeros of system, Laplace domain analysis, solution to differential equations and system behavior.	6
5	The z-Transform for discrete time signals and systems- eigen functions, region of convergence, z-domain analysis.	5
6	State-space analysis and multi-input, multi-output representation. The state-transition matrix and its role. The Sampling Theorem and its implications- Spectra of sampled signals. Reconstruction: ideal interpolator, zero-order hold, first-order hold, and so on. Aliasing and its effects. Relation between continuous and discrete time systems.	8
	Total	**40**

Course Overview:

Signals and Systems is an introduction to analog and digital signal processing, a topic that forms an integral part of engineering systems in many diverse areas, including seismic data processing, communications, speech processing, image processing, defense electronics, consumer electronics, and consumer products.

The course presents and integrates the basic concepts for both continuous-time and discrete-time signals and systems. Signal and system representations are developed for both time and frequency domains. These representations are related through the Fourier transform and its generalizations, which are explored in detail. Filtering and filter design, modulation, and sampling for both analog and digital systems, as well as exposition and demonstration of the basic concepts of feedback systems for both analog and digital systems, are discussed and illustrated.

Course Outcomes:

CO.NO.	Cognitive Level	Course Outcome
1	**Comprehension**	Classify different types of signals and system properties.
2	**Application**	Demonstrate continuous and discrete systems in time and frequency domain using different transforms.
3	**Analysis**	Analyze whether the system is stable.
4	**Synthesis**	Design and Develop Sampling and reconstruction circuit .
5	**Evaluation**	Evaluate the output of the MIMO systems.

Prerequisites:

1. Fundamentals knowledge of differentiation and integration.
2. Fundamentals knowledge of partial fraction.
3.

Course Outcome Mapping with Program Outcome:

Course Outcome	Program Outcomes (PO's)											
CO. NO.	Domain Specific					Domain Independent						
	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12
CO1	3	3	1	2	2	0	0	1	0	0	0	2
CO2	3	1	0	2	3	0	0	1	0	0	0	2
CO3	3	2	2	3	0	0	0	0	0	0	0	2
CO4	3	2	3	3	1	0	0	0	0	0	0	0
CO5	3	2	2	3	1	0	0	2	0	0	0	1
1: Slight (Low) , 2: Moderate (Medium), 3: Substantial (High)												

Course Coverage Module Wise:

Lecture No.	Unit	Topic
1	**1**	**INTRODUCTION: OBJECTIVE, SCOPE AND OUTCOME OF THE COURSE.**
2	1	Energy signals power signals
3	1	Continuous and discrete time signals.
4	1	Discrete amplitude signals
5	1	Discrete amplitude signals
6	1	System properties: linearity: additivity and homogeneity
7	**2**	**SHIFT-INVARIANCE, CAUSALITY**
8	2	Stability, realizability
9	2	Linear shift-invariant (LSI) systems
10	2	Impulse response
11	2	Step response
12	2	Convolution
13	2	Input output behavior with aperiodic convergent inputs
14	**3**	**CHARACTERIZATION OF CAUSALITY AND STABILITY OF LINEAR SHIFT-INVARIANT SYSTEMS**
15	3	System representation through differential equations and difference equations
16	3	Characterization of causality and stability of linear shift-invariant systems
17	3	System representation through differential equations and difference equations
18	3	Periodic and semi-periodic inputs to an LSI system
19	3	The notion of a frequency response
20	3	Its relation to the impulse response
21	3	Fourier series representation
22	**4**	**FOURIER TRANSFORM**
23	4	Convolution/multiplication and their effect in the frequency

		domain
24	4	Magnitude and phase response
25	4	Fourier domain duality
26	4	The Discrete-Time Fourier Transform (DTFT) and Discrete Fourier Transform (DFT)
27	4	Parseval's Theorem. The idea of signal space and orthogonal bases
28	**5**	**THE LAPLACE TRANSFORM**
29	5	Notion of eigen functions of LSI systems
30	5	A basis of eigen functions, region of convergence
31	5	Poles and zeros of system, Laplace domain analysis
32	5	Solution to differential equations and system behavior
33	**6**	**THE Z-TRANSFORM FOR DISCRETE TIME SIGNALS AND SYSTEMS- EIGEN FUNCTIONS**
34	6	Region of convergence, z-domain analysis
35	6	State-space analysis and multi-input, multi-output representation
36	6	The state-transition matrix and its role
37	6	The Sampling Theorem and its implications- Spectra of sampled signals.
38	6	Reconstruction: ideal interpolator, zero-order hold, first-order hold
39	6	Aliasing and its effects
40	6	Relation between continuous and discrete time systems

TEXT/REFERENCE BOOKS

1. Signals and Systems, A.V. Oppenheim, A.S. Willsky and I.T. Young, Prentice Hall, 1983.
2. Signals and Systems - Continuous and Discrete, R.F. Ziemer, W.H. Tranter and D.R. Fannin ,4th edition, Prentice Hall, 1998.
3. Circuits and Systems: A Modern Approach, Papoulis, HRW, 1980.
4. Signal Processing and Linear Systems, B.P. Lathi, Oxford University Press, 1998.

NPTEL COUSES LINK

1. https://archive.nptel.ac.in/courses/108/104/108104100/
2. https://archive.nptel.ac.in/courses/108/106/108106163/

QUIZ Link

https://www.sanfoundry.com/1000-signals-systems-questions-answers/

Faculty Notes Link

https://drive.google.com/drive/folders/1OzPymFrTTK4fCkrOU-ADluTXYWHok5T_?usp=drive_link

Assessment Methodology:

1. Practical exam using MATALB software.
2. Two Midterm exams where student have to showcase subjective learning.
3. Final Exam (subjective paper) at the end of the semester.

VIVA-VOCE SET OF QUESTIONS

Q.1. What is a signal and system?
Answer: A function of one or more independent variables which contain some information is called signal.
A system is a set of elements or functional blocks that are connected together and produces an output in response to an input signal.

Q.2. How can you differentiate signal and wave?
Answer: A signal is what which contains information while wave does not contain any information.
Q.3. What is the difference between deterministic and random signals?
Answer: A deterministic signal can be completely represented by mathematical equation at any time whereas a signal which cannot be represented by any mathematical equation is called random signal.
Q.4. What will be the signal in the frequency domain when a signal is discrete and periodic in time domain?
Answer: Since periodicity in one domain reveals discrete in other domain, so if the signal is discrete and periodic in one domain then it is periodic nad discrete in other domain.
Q.5. What are analog and digital signals?
Answer: When amplitude of CT signal varies continuously, it is called analog signal.
In other words amplitude and time both are continuous for analog signal. When amplitude of DT signal takes only finite values, it is called digital signal.In other words amplitude and time both are discrete for digital signal.
Q.6. What are even and odd signals?
Answer: A signal is said to be even signal if inversion of time axis does not change the amplitude. i.e, $x(t) = x(-t)$
A signal is said to be odd signal if inversion of time axis also inverts amplitude of the signal. i.e, $x(t) = -x(-t)$
Q.7. What is the significance of even and odd signals?
Answer: Even or odd symmetry of the signal have specific harmony or frequency contents and this even and odd symmetry property is used in designing of filters.
Q.8. Can you able to reconstruct the original signal from sampled signal if it has been sampled at Nyquist rate?
Answer: No original signal cannot be reconstructed because in order to reconstruct the original signal from sampled signal when it is sampled at Nyquist rate, an ideal low pass filter is required which is impossible in real life to construct.
Q.9. What is the difference between power signal and energy signal in terms of energy and power?
Answer: Energy of the power signal is infinite whereas power of the energy signal is zero.
Q.10. What is the significance of unit impulse or unit sample functions?
Answer: Unit impulse or unit sample functions are used to determine impulse response of the system. It also contains all the frequencies from $-\infty$ to ∞.
Q.11. What is the significance of unit ramp function?
Answer: The ramp function indicates linear relationship. It also indicates constant current charging of the capacitor.

Q.12. Can you able to construct original signal from the quantized signal?
Answer: No, since quantizer is a non invertible system so we cannot construct original signal from quantized signal.
Q.13. What is the basic difference between amplitude and magnitude?
Answer: Amplitude is a vector quantity having both value and direction whereas magnitude is a scalar quantity having only value but not the direction.
Q.14. What are the limitations of Fourier transform and use of Laplace transform?
Answer: They are:

- Fourier transform can be calculated only for the signals which are absolutely integrable. But Laplace transform exists for signals which are not absolutely integrable.
- Fourier transform is calculated only on the imaginary axis, but Laplace transform can be calculated over complete s-plane. Hence Laplace transform is more broader compared to Fourier transform.

Q.15. What are the applications of initial and final value theorems?
Answer: They are:

- The initial voltage on the capacitor or current through an inductor can be evaluated with the help of initial value theorem.
- The final charging voltage on capacitor or saturating currents through an inductor can be evaluated with the help of final value theorem.

Q.16. Can we interchange the sampling and quantization operations, means instead of sampling the signal first and then quantized, can we do quantization first and then sampling?
Answer: Yes we can interchange the sampling and quantization operations but the drawback is that it results in increased quantization noise.
Q.17. What is the significance of region of convergence (ROC) of Z transform?
Answer: The significance of region of convergence (ROC) of Z transform are:

- ROC gives an idea about values of z for which Z-transform can be calculated.
- ROC can be used to determine causality of the system.
- ROC can be used to determine stability of the system.

Q.18. What is the relationship between z-transform and DTFT?
Answer: When z-transform is evaluated on unit circle, then it becomes Fourier transform or in other words we can say that DTFT is a special case of z-transform on unit circle.
Q.19. What is the similarity between Laplace transform and z-transform?
Answer: Z-transform is the discrete time counter part of Laplace transform with negative real axis mapped within unit circle, $j\omega$ axis mapped on unit circle and right half mapped on outside a unit circle.
Q.20. What is the relationship between Laplace transform and CTFT?
Answer: When Laplace transform is evaluated on $j\omega$ axis , then it becomes Fourier transform or in other words CTFT is a special case of Laplace transform evaluated on $j\omega$ axis.
Q.21. What is the difference between DTFT and DFT?
Answer: In DTFT the discrete signal is assumed to be aperiodic so the frequency domain signal is periodic and continuous whereas in DFT, the discrete signal is assumed to be periodic so frequency domain signal is periodic and discrete.

Q.22. What do you mean by Gibbs phenomenon?
Answer: Gibbs phenomenon says that whenever there is abrupt discontinuity in the signal which is being sampled, the reconstructed signal will always have high frequency oscillations and as the number of samples increases the oscillations compress towards discontinuity but their maximum value remains the same.
Q.23. Define invertible system?
Answer: A system is said to be invertible if there is unique output for every unique input.
Q.24. What is the difference between convolution and correlation?
Answer: In convolution one of the two signals is folded and shifted while in correlation none of the signal is folded but one signal is shifted to right or left.
Q.25. What are the applications of convolution?
Answer: The applications of convolution are:
- It is used for system analysis such as causality, stability, step response, impulse response, invertibility etc.
- It is used to determine output of the system if input and impulse response is given.
- It relates input output and impulse response.
- Convolution helps to represent system in frequency domain using Fourier, Laplace and z-transform.
- This is used to study pole-zero plots, stability, filtering etc.

Q.26. What is autocorrelation?
Answer: When we calculate correlation function of the signal with itself, then it is called autocorrelation. Thus if $x1(t) = x2(t)$, then correlation becomes autocorrelation.
Q.27. What is the importance of unit impulse function?
Answer: One of the important characteristics of unit impulse is that very general signals can be represented as linear combination of delayed impulses.
Q.28. Define Parseval's Theorem.
Answer: It states that the power of the signal is equal to the sum of the square of the magnitudes of various harmonics present in the spectrum.
Q.29. What are Dirichlet's condition?
Answer: Following are the Dirichlet's conditions:
- The function f(t) is the single valued function of the variable t within the interval (t1, t2).
- The function f(t) has a finite number of discontinuities in the interval (t1, t2).
- The function f(t) has a finite number of maxima and minima in the interval (t1, t2).
- The function f(t) is absolutely integrable.

Q.30. Why do we do Fourier Transform?
Answer: By Fourier Transform we can represent the signal from time domain to frequency domain, thus we can find the various frequency components contained in the given signal. Helping us to find the total bandwidth required for the transmission of the given signal.
Q.31. Why for signal analysis we use only sinusoidal waves and not other signals?
Answer: We use only sinusoidal waves and not other signals because:
- The response of sine wave to a LTI system is also sinusoidal.
- The sinusoidal analysis of electric network is more simple and convenient.

Q.32. What do you mean by sinusoidal fidelity?
Answer: sinusoidal fidelity is an important characteristic of linear system. If input to a linear system is a sine wave, the output will also be a sine wave at exactly the same frequency. Only the amplitude and phase can be different.

Q.33. Define Aliasing effect.
Answer: Aliasing is an effect that causes different signals to become indistinguishable (or aliases of one another) when sampled. It also refers to the distortion or artifact that results when the signal reconstructed from samples is different from the original continuous signal.
Q.34. What is the double curse effect of Aliasing?
Answer: Due to Aliasing high frequency contents are recovered at low frequency so both high frequency and low frequency contents are lost.
Q.35. What are mutually orthogonal functions?
Answer: Two vectors are said to be orthogonal if their product is zero. i.e, the two vectors have nothing in common. Example, Trignometric and exponential functions.
Q.36. Define linearity or linear system.
Answer: A linear system is a system that possesses the property of superposition, i.e, additive property and scaling or homogeneity property.
Q.37. Define fundamental frequency.
Answer: It is the smallest frequency with which a signal repeats itself.
Q.38. What are passive and active filters?
Answer: A passive filter is a kind of electronic filter that is made only from passive elements – in contrast to an active filter, it does not require an external power source (beyond the signal). An active filter is a type of analog electronic filter, distinguished by the use of one or more active components and require an external power source.
Q.39. Is it possible to design a filter which can give good results both in time domain and frequency domain?
Answer: No it is not possible to design such kind of filter.
Q.40. What are the advantages of digital filter over analog filters?
Answer: Digital filters have the following advantages compared to analog filters:

- Digital filters are software programmable, which makes them esay to build and test.
- Digital filters require only the arithmetic operations of addition, subtraction and multiplication.
- Digital filters do not drift with temperature or humidity or require precision components.
- Digital filters have a superior performance to cost ratio.
- Digital filters do not suffer from manufacturing variations or aging.

Q.41. Which property of Fourier transform is used in Analog modulation?
Answer: Time shifting property of Fourier transform.
Q.42. What are the classification of the system based on unit sample response?
Answer: FIR (Finite impulse response) system and IIR (Infinite impulse response) system.
Q.43. Define FIR system.
Answer: If the system has finite duration impulse response then the system is said to be finite impulse response (FIR) system.
Q.44. Define IIR system.
Answer: If the system has infinite duration impulse response then the system is said to be infinite impulse response (IIR) system.
Q.45. Give one example of FIR and IIR filters?
Answer: Windowed sinc filters are the examples of FIR filters and moving average filters are the example of IIR filters.
Q.46. Which filter (FIR or IIR filters) is always stable and why?
Answer: FIR filters are always stable because they do not contain poles.

Q.47. Define Fourier series?
Answer: Fourier series is the representation of a function f(t) by the linear combination of elements of a closed set of infinite mutually orthogonal functions.
Q.48. What is Hilbert Transform?
Answer: Hilbert transform of a signal x(t) is defined as the transform in which phase angle of all components of the signal is shifted by ±90∘.
Q.49. What do you mean by linear phase filter?
Answer: If the system is having even symmetry around some frequency other than zero then the filter is said to have linear phase.
Q.50. Can we make non linear phase of IIR filter as linear phase?
Answer: Yes linear phase can be obtained by bidirectional filtering at the expense of double the execution time and program complexity.

Assignment and Quiz Solution Link

https://ocw.mit.edu/courses/res-6-007-signals-and-systems-spring-2011/pages/assignments/