CURRICULUM AND SYLLABUS OF M.TECH. DEGREE PROGRAMME IN SIGNAL PROCESSING

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

NATIONAL INSTITUTE OF TECHNOLOGY CALICUT

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Vision of the Department of Electronics and Communication Engineering

The Department of Electronics and Communication Engineering is envisioned to be a leading centre of higher learning with academic excellence in the field of electronics and communication engineering.

Mission of the Department in pursuance of its vision

The mission of the Department of Electronics and Communication Engineering is to impart high quality technical education by offering undergraduate, graduate and research programs in the domain of Electronics and Communication Engineering with thorough foundation in theory along with strong hands-on design and laboratory components, tools and skills necessary for the students to become successful major contributors to society and profession.

Department of Electronics & Communication Engineering

The Program Educational Objectives (PEOs) of M.Tech. in Signal Processing

Sl. No.	Program Educational Objectives
PEO 1	To provide graduates strong mathematical skills and in depth knowledge in signal theory to analyze and solve complex problems in the domain of signal processing, imparting lateral thought, originality and creativity.
PEO 2	To instill research skills and bring in optimal solutions and novel products to signal processing and allied application areas using modern technology and tools that are technically sound, economically feasible and socially acceptable.
PEO 3	To enable the graduates to engage in lifelong learning in signal processing and its broad range of applications to understand the challenges of the rapidly changing environment and adapt their skills through reflective and continuous learning.
PEO 4	To inculcate professionalism, ethical attitude, communication skills, synergetic and leadership qualities and ability to relate engineering solutions to broader social context.

Department of Electronics & Communication Engineering

The Program Outcomes (POs) of M.Tech. in Signal Processing

At the end of the Program the students will be able to:

Sl. No.	Program Outcome	Graduate Attribute
PO 1	Acquire strong mathematical skills and in-depth knowledge in signal theory that enables to view the signal processing problems in a wider and global perspective, with an ability to discriminate, evaluate, analyze and synthesize the acquired knowledge in the domain of signal processing for enhancement of knowledge.	Scholarship of Knowledge
PO 2	Analyze complex problems in the domain of signal processing critically to make intellectual and creative advances for conducting research in a wider, theoretical, practical and policy context.	Critical Thinking
PO 3	Solve problems in the core area of signal processing, by evaluating a wide range of solutions and arriving at the optimal one considering public health and safety, cultural, societal and environmental factors,	Problem Solving

	imparting lateral thought and originality.	
PO 4	Develop skills to extract information on research problems through literature survey and apply appropriate research methodologies, techniques and tools to design and conduct experiments, analyze and interpret data and demonstrate higher order skills to contribute individually or in groups to the development of scientific or technological knowledge in signal processing and allied disciplines.	Research Skill
PO 5	Model and simulate complex signal processing systems to conduct experiments and analyse the performance, by using modern signal processing tools	Usage of modern tools
PO 6	Possess knowledge and understanding of group dynamics, recognize opportunities and contribute positively to collaborative-multidisciplinary scientific research.	Collaborative and Multidisciplinary work
PO 7	Work as a member and leader in a team, demonstrating the knowledge and understanding of management principles and challenges in signal processing and allied application areas, after consideration of economical and financial factors.	Project Management and Finance
PO 8	Express ideas with clarity and communicate confidently and effectively through reports adhering to appropriate standards and/or oral presentations.	Communication
PO 9	Recognize the need to engage in life-long learning with a high level of enthusiasm and commitment to improve knowledge in the domain of signal processing.	Life-long Learning
PO 10	Acquire professional and intellectual integrity, professional code of conduct and ethics of research, considering the impact of research outcome on professional practices and an understanding of responsibility to contribute to the community for sustainable development of society.	Ethical practices and Social Responsibility
PO 11	Critically analyze one’s own actions and make corrective measures independently.	Independent and Reflective Learning

Department of Electronics & Communication Engineering
Curriculum for M. Tech. in Signal Processing

Semester 1

S.No	Code	Title	L	T	P/S	C
1	EC6401	Linear Systems Theory	4	0	0	4
2	EC6303	Information Theory	4	0	0	4
3	EC6402	Multirate Signal processing	3	0	0	3
4	EC6301	Random Processes	4	0	0	4
5	EC6403	Signal Processing Lab I	0	0	3	2
		Elective 1	3	0	0	3
		Total credits				20

Semester 2

S.No	Code	Title	L	T	P/S	C
1	EC6404	Adaptive Signal Processing	4	0	0	4
2	EC6307	Estimation & Detection Theory	4	0	0	4
3	EC6405	Signal Processing Lab II	0	0	3	2
4	EC6406	Seminar	0	0	2	1
		Elective 1	3	0	0	3
		Elective 2	3	0	0	3
		Elective 3	3	0	0	3
		Total credits				20

Semester 3

S.No	Code	Title	L	T	P/S	C
1	EC7401	Project Work	-	-	-	8

Semester 4

S.No.	Code	Title	L	T	P/S	C
1	EC7402	Project Work	-	-	-	12

<u>Minimum Requirements</u>

1. **Minimum number of credits to be earned by a student is 60**

List of Electives

S.No.	Code	Title	Credit
1	EC6421	Image & Video Processing	3
2	EC6422	Linear & Nonlinear Optimization	3
3	EC6423	Signal Compression - Theory and Methods	3
4	EC6424	Multidimensional Signal Processing	3
5	EC6425	Wavelets: Theory & Construction	3
6	EC6426	Transform Theory	3
7	EC6427	Array Signal Processing	3
8	EC6428	Speech & Audio Processing	3
9	EC6429	Biomedical Signal Processing	3
10	EC6430	Pattern Recognition and Analysis	3
11	EC6431	DSP Algorithms and Architecture	3
12	EC6432	Spectrum Analysis	3
13	EC6433	Compressive Sensing: Theory and Algorithms	3

- Any other subject (core/elective) offered by the Department from time to time shall be taken as elective with the consent of course co-ordinator/faculty.

DETAILED SYLLABI

SEMESTER I

EC6401: Linear System Theory

Pre-requisite: Nil

L	T	P	C
4	0	0	4

Course Outcomes:

- ***CO1:*** *Foundation concepts on Signal Theory and System Theory applicable in in Communication Engineering and Signal Processing*
- ***CO2:*** *Mathematical framework for Signal Theory and System Theory: Analysis and Design*
- ***CO3:*** *Basic concepts that enable designs for environment-friendly direct applications*
- ***CO4:*** *A foundation subject for multi-disciplinary applications*
- ***CO5:*** *Develops ability to think clearly and express precisely, coupled with systematic logical reasoning.*

Total Hours : 56 Hrs

Module 1 : (15 hours) Finite Dimensional Signal Space

Vector Spaces :- Complex Numbers, Definition of Vector Space, Properties of Vector Spaces, Subspaces, Sums and Direct Sums, Span and Linear Independence, Bases, Dimension
Inner-Product Spaces :- Inner Products, Norms, Orthonormal Bases, Orthogonal Projections and Minimization Problems, Linear Functionals and Adjoints

Some Important Bases :- Standard Ordered Bases, DFT Bases, DCT Bases.

Module 2 : (11 hours) Linear Systems

Linear Maps :- Definitions and Examples, Null Spaces and Ranges, The Matrix of a Linear Map, Invertibility.
Eigenvalues and Eigenvectors :- Invariant Subspaces, Polynomials Applied to Operators, Upper-Triangular Matrices, Diagonal Matrices, Invariant Subspaces on Real Vector Spaces

Module 3 : (12 hours) Linear Systems(contd ...)

Operators on Inner-Product Spaces :- Self-Adjoint and Normal Operators, The Spectral Theorem, Normal Operators on Real Inner-Product Spaces, Positive Operators, Isometries, Polar and Singular-Value Decompositions.

Some Important Classes of Linear Systems :- Shift Invariant systems and Toeplitz matrices. Operators and square matrices. Self adjoint operators and Hermitian matrices. Projections and idempotent matrices. Rotations and unitary matrices.

Module 4 : (18 hours) Infinite Dimensional Signal Spaces

Metric Spaces :- Definition, Convergence and Completeness.

Hilbert spaces :- Introduction [Ref 3, Appendix]. l^2 and L^2 spaces. Definition and some properties. Orthogonal Complements, Orthonormal Sets, Fourier Expansion. Conjugate Space, Adjoint of an Operator, Self Adjoint Operators, Normal and Unitary operators, Projections.

References

1. Sheldon Axler, *Linear Algebra Done Right*, Springer
2. G. F. Simmons, *Introduction to Topology and Modern Analysis*, Tata McGraw Hill.
3. Paul R. Halmos, *Finite-Dimensional Vector Spaces*, Springer
4. Todd K. Moon and Wynn C. Stirling, *Mathematical Methods and Algorithms for Signal Processing*, Pearson
5. Arch W. Naylor and George R. Sell, *Linear Operator Theory in Engineering and Science*, Springer
6. Peter D. Lax, *Linear Algebra*, Wiley Students Edition.
7. Michael W. Frazier, *An Introduction to Wavelets Through Linear Algebra*, Springer.

EC6303: Information Theory

Prerequisite: A first course in Probability Theory and Random Processes

L	T	P	C
4	0	0	4

Course Outcomes:

- ***CO1:*** *To learn models of information sources with and without memory and analyse methods for efficiently coding their output ensuring no loss of information with a clear understanding on ultimate limits on performance*
- ***CO2:*** *Develop mathematical models for practical communication channels and analyse their information carrying capacity*
- ***CO3:*** *Evaluate through analysis, the methods for ensuring reliable data transfer and the bounds on their achievable performance by observing modulation-coding trade off given by Shannon's theorems*
- ***CO4:*** *Design and develop methods for distributing energy among parallel channels of communication for maximum information transfer through water filling considerations*
- ***CO5:*** *Mathematically analyse the performance of schemes for compressing the data with trade off on rate and distortion and discuss design considerations*
- ***CO6:*** *Understand research challenges in the design and development of communication systems for real world applications and suggest solutions through innovative design concepts*

Total Hours : 56Hrs

Module-1 - (14 hours) Entropy and Loss less Source coding

Entropy- Memory less sources- Markov sources- Entropy of a discrete Random variable- Joint, conditional and relative entropy- Mutual Information and conditional mutual information- Chain relation for entropy, relative entropy and mutual Information- Lossless source coding- Uniquely decodable codes- Instantaneous codes- Kraft's inequality - Optimal codes- Huffman code- Shannon's Source Coding Theorem.

Module-2 –(15 hours) Channel Capacity and Coding Theorem

Asymptotic Equipartition Property (AEP)- High probability sets and typical sets- Method of typical sequence as a combinatorial approach for bounding error probabilities.

Channel Capacity- Capacity computation for some simple channels- Arimoto-Blahut algorithm- Fano's inequality-Proof of Shannon's Channel Coding Theorem and its converse- Channels with feed back- Joint source channel coding Theorem.

Module-3 -(15 hours) Continuous Sources and Channels Differential Entropy- Joint, relative and conditional differential entropy- Mutual information- Waveform channels- Gaussian channels- Mutual information and Capacity calculation for Band limited Gaussian channels- Shannon limit- Parallel Gaussian Channels-Capacity of channels with colored Gaussian noise- Water filling.

Module-4 -(12 hours) Rate Distortion Theory

Introduction - Rate Distortion Function - Properties - Continuous Sources and Rate Distortion measure- Rate Distortion Theorem - Converse - Information Transmission Theorem - Rate Distortion Optimization.

References

1.Thomas M. Cover and Joy A. Thomas, "Elements of Information Theory", John Wiley & Sons, 2006

2. David J. C. MacKay, "Information Theory, Inference and Learning Algorithms", Cambridge University Press, 2003

3.Robert Gallager, "Information Theory and Reliable Communication", John Wiley & Sons.

4.R. J. McEliece, "The theory of information & coding", Addison Wesley Publishing Co., 1977.

5.T. Bergu, "Rate Distortion Theory a Mathematical Basis for Data Compression" PH Inc. 1971.

6.Special Issue on Rate Distortion Theory, IEEE Signal Processing Magazine, November 1998.

EC6402: Multirate Signal Processing

Pre-requisite: Digital Signal Processing

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Understand the fundamentals of multirate signal processing and its applications.*
- ***CO2:*** *Learn the theory of sampling rate conversion and develop methods for decimating, interpolating and changing the sampling rate of the signal and to develop efficient polyphase implementations of sampling rate converters.*
- ***CO3:*** *Get a solid conceptual background in multirate filter banks an in-depth understanding of both the theoretical and practical aspects of multirate signal processing.*
- ***CO4:*** *Design perfect reconstruction and near perfect reconstruction filter bank system and to learn to assess the computational efficiency of multirate systems.*
- ***CO5:*** *Analyze the quantization effects in filter banks.*
- ***CO6:*** *Understand the use of filter banks in applications such as speech processing and communication*

Total Hours : 42Hrs

Module 1: (12 hours) Fundamentals of Multirate Theory

The sampling theorem - sampling at sub nyquist rate - Basic Formulations and schemes.

Basic Multirate operations- Decimation and Interpolation - Digital Filter Banks- DFT Filter Bank- Identities- Polyphase representation

Maximally decimated filter banks: Polyphase representation - Errors in the QMF bank- Perfect reconstruction (PR) QMF Bank - Design of an alias free QMF Bank

Module 2: (9 hours) M-channel perfect reconstruction filter banks

Uniform band and non uniform filter bank - tree structured filter bank- Errors created by filter bank system- Polyphase representation- perfect reconstruction systems

Module 3: (12 Hours) Perfect reconstruction (PR) filter banks

Paraunitary PR Filter Banks- Filter Bank Properties induced by paraunitarity- Two channel FIR paraunitary QMF Bank- Linear phase PR Filter banks- Necessary conditions for Linear phase property- Quantization Effects: -Types of quantization effects in filter banks. - coefficient sensitivity effects, dynamic range and scaling.

Module 4: (9 Hours) Cosine Modulated filter banks

Cosine Modulated pseudo QMF Bank- Alas cancellation- phase - Phase distortion- Closed form expression- Polyphase structure- PR Systems

References

1. P.P. Vaidyanathan. “Multirate systems and filter banks.” Prentice Hall. PTR. 1993.
2. N.J. Fliege. “Multirate digital signal processing .” John Wiley 1994.
3. Sanjit K. Mitra. “ Digital Signal Processing: A computer based approach.” McGraw Hill. 1998.
4. R.E. Crochiere. L. R. “Multirate Digital Signal Processing”, Prentice Hall. Inc.1983.
5. J.G. Proakis. D.G. Manolakis. “Digital Signal Processing: Principles. Algorithms and Applications”, 3rd Edn. Prentice Hall India, 1999.

EC6301: Random Processes

Prerequisite: Nil

L	T	P	C
4	0	0	4

Course Outcomes

- ***CO1**: Understand the fundamental concepts of probability space and probability axioms*
- ***CO2**: Acquire the knowledge to formulate random variables corresponding to random experiments and to derive their probability distribution functions*
- ***CO3**: Develop an understanding of discrete and continuous random variables and their characterization*
- ***CO4**: Understand the use of Markov and Chebyshev inequalities to obtain bounds on probability of events and the use of central limit theorem to compute probabilities*
- ***CO5**: Learn the use of random vectors to model experiments with multiple simultaneous outcomes and the techniques to determine the distribution function of random vectors*
- ***CO6:**Understand the concept of random processes and techniques to find the correlation, covariance and power spectral density of stationary random processes*
- ***CO7**: Learn how to apply the theory of random processes to analyse linear systems with special focus on telecommunications and signal processing*
- ***CO8**: Develop sound knowledge in topics such as random walks, Markov chains and series representation of random processes and their applications in telecommunications and signal processing*

Total Hours : 56 Hrs

Module 1: (14 hours) Random Variables
Probability axioms, conditional probability, discrete and continuous random variables, cumulative distribution function (CDF), probability mass function (PMF), probability density function (PDF), conditional PMF/PDF, expected value, variance, functions of a random variable, expected value of the derived random variable, multiple random variables, joint CDF/PMF/PDF, functions of multiple random variables, multiple functions of multiple random variables, independent/uncorrelated random variables, sums of random variables, moment generating function, random sums of random variables.

Module 2: (14 hours) Fundamental Theorems and Random Processes
The sample mean, laws of large numbers, central limit theorem, convergence of sequence of random variables. Introduction to random processes, specification of random processes, *n*th order joint PDFs, independent increments, stationary increments, Markov property, Markov process and martingales, Gaussian process, Poisson process and Brownian motion.

Module 3: (14 hours) Response of Processes to LTI Systems
Mean and correlation of random processes, stationary, wide sense stationary and ergodic processes. Random processes as inputs to linear time invariant systems: power spectral density, Gaussian processes as inputs to LTI systems, white Gaussian noise. In-Phase and quadrature representation of random processes.

Module 4: (14 hours) Other Topics
Discrete-time Markov chains: state and n-step transition probabilities, Chapman-Kolmogorov equations, first passage probabilities, classification of states, limiting state probabilities Series representation of random process: Fourier series, Karhunen-Loeve expansion, Mercer's theorem, sampled band-limited processes, filtering using series representation

References

1. Geoffrey Grimmett, "Probability and Random Processes", 3rd edition, Oxford University Press 2001
2. Henry Stark and John W. Woods, "Probability and Random Processes with Applications to Signal Processing", Prentice Hall, 3rd Edition 2001
3. Yannis Viniotis, "Probability and Random Processes for Electrical Engineers" McGraw-Hill College, 1998
4. Albert Leon Garcia: "Probability and Random Processes for Electrical Engineering", Prentice Hall 1993
5. A. Papoulis and S. U. Pillai, "*Probability, Random Variables and Stochastic Processes*", 4th Edition, McGraw Hill 2002
6. V. Krishnan: "Probability and Random Processes", John Wiley & Sons 2006

EC6403: SIGNAL PROCESSING LAB I

Co-requisite : EC 6400, EC 6402, EC6301

L	T	P	C
0	**0**	**3**	**2**

Course Outcomes:

- ***CO1**: Obtain the ability to apply knowledge of linear algebra, random process and multirate signal processing in various signal processing applications.*
- ***CO2**: Develop the student's ability on conducting engineering experiments, analyze experimental observations scientifically*
- ***CO3** : Become familiar to fundamental principles of linear algebra*
- ***CO4**: Familiarize the basic operations of filter banks through simulations*
- ***CO5**: Apply the principles of random process in practical applications*
- ***CO6**: Develop the student's ability on preparing professional report*

Total Hours : 42 Hrs

Course objective :

To experiment the concepts introduced in the courses EC6400 (Linear Systems Theory), EC6402 (Multirate Signal Processing) and EC6301 (Random Processes)

Tools :

Numerical Computing Environments – GNU Octave or MATLAB or any other equivalent tool.

Suggested Experiments :

Linear Systems Theory – Change of Basis, Gram-Schmidt Orthogonalization, Least square solutions, Eigen Value Decomposition, Singular Value Decomposition.

Multirate Signal Processing – Decimation and Interpolation, Filterbank design.

Random Processes – Generation of discrete time i.i.d. random processes with different distributions (Bernoulli, Binomial, Geometric, Poisson, Uniform, Gaussian, Exponential, Laplacian, Rayleigh, Rician) - pmf/pdf estimation, AR, MA and ARMA processes - spectral estimation. Visualization of Central Limit Theorem. Whitening Filter.

SEMESTER II

EC6404: Adaptive Signal Processing

Pre-requisite: Basic Course in Digital Signal Processing

L	T	P	C
4	0	0	4

- ***CO1:*** *Understand the basic concepts of statistical signal processing and analyse the performance requirements in various real life applications.*
- ***CO2:*** *Devise filtering solutions for optimising the cost function indicating error in estimation of parameters and appreciate the need for adaptation in design.*
- ***CO3:*** *Evaluate the performance of various methods for designing adaptive filters through estimation of different parameters of stationary random process clearly considering practical application specifications.*
- ***CO4:*** *Analyse convergence and stability issues associated with adaptive filter design and come up with optimum solutions for real life applications taking care of requirements in terms of complexity and accuracy.*
- ***CO5:****. Study and analyse methods for designing filters to track variations of non-stationary random process and develop algorithms to meet performance requirements.*
- ***CO6:*** *Design and implement filtering solutions for applications such as channel equalisation, interference cancelling and prediction considering present day challenges and recent research development and analyse their performance*

Total Hours: 56 Hrs

Module 1: (11 hours)
Adaptive systems - definitions and characteristics - applications - properties-examples - adaptive linear combiner-input signal and weight vectors - performance function-gradient and minimum mean square error -introduction to filtering-smoothing and prediction - linear optimum filtering-orthogonality - Wiener – Hopf equation-performance surface

Module 2: (13 hours)
Searching performance surface-stability and rate of convergence - learning curve-gradient search - Newton's method - method of steepest descent - comparison - gradient estimation - performance penalty - variance -excess MSE and time constants – misadjustments

Module 3: (18 hours)
LMS algorithm- convergence of weight vector-LMS/Newton algorithm - properties - sequential regression algorithm - adaptive recursive filters - random-search algorithms - lattice structure – Kalman filters-recursive minimum mean square estimation for scalar random variables- statement of Kalman filtering problem-innovation process-estimation of the state-filtering-initial conditions-Kalman filter as the unifying basis for RLS filters

Module 4: (14 hours)
Applications-adaptive modeling and system identification-adaptive modeling for multipath communication channel, geophysical exploration, FIR digital filter synthesis, inverse adaptive modeling, equalization, and deconvolution-adaptive equalization of telephone channels-adapting poles and zeros for IIR digital filter synthesis

References

1.Bernard Widrow and Samuel D. stearns, “Adaptive Signal Processing”, Person Education, 2005.
2.Simon Haykin, “ Adaptive Filter Theory”, Pearson Education, 2003.
3.John R. Treichler, C. Richard Johnson, Michael G. Larimore, “Theory and Design of Adaptive Filters”,Prentice-Hall of India, 2002
4.S. Thomas Alexander, “ Adaptive Signal Processing - Theory and Application”, Springer-Verlag.
5.D. G. Manolokis, V. K. Ingle and S. M. Kogar, “Statistical and Adaptive Signal Processing”, Mc Graw Hill International Edition, 2000.

EC6307: Estimation & Detection Theory

Pre-requisite: Linear algebra, Random Process

L	T	P	C
4	0	0	4

Course Outcomes

- *CO1: Appreciate the need for estimation techniques in Communication and Signal Processing problems and acquire expertise in Classical and Bayesian estimation techniques for parameters and signals, and Detection of signals in the presence of white Gaussian noise*
- *CO2: Conduct in-depth analysis of estimation problems and apply suitable estimation and detection techniques that meet the constraints of the problem such as performance, bandwidth and power overheads and computational complexity*
- *CO3: Judge the scenarios under which signal or parameter estimation techniques are preferred and develop estimation techniques that are suitable for the context from a wider perspective*
- *CO4: Appreciate prior art in some chosen area outside the syllabus through exhaustive literature review, implement an appropriate work, explore the possibility of developing better solution and validate results through appropriate tools and techniques individually as well as member of a team*
- *CO5: Present findings through technical reports and oral presentations to a target group, credit the contributions of others and be responsive to critical feedbacks*
- *CO6: Acknowledge the needs of the society while designing and implementing solutions to problems that are critical to humanity*

Total Hours : 56 Hrs

Module-1 - (13 Hrs) Fundamentals of Estimation Theory

Role of Estimation in Signal Processing, Unbiased Estimation, Minimum variance unbiased(MVU) estimators, Finding MVU Estimators, Cramer-Rao Lower Bound, Linear Modeling-Examples, Sufficient Statistics, Use of Sufficient Statistics to find the MVU Estimator.

Module-2– Estimation Techniques

Deterministic Parameter Estimation: (9 Hrs) Least Squares Estimation-Batch Processing, Recursive Least Squares Estimation, Best Linear Unbiased Estimation, Likelihood and Maximum Likelihood Estimation
Random Parameter Estimation: (6 Hrs) Bayesian Philosophy, Selection of a Prior PDF, Bayesian linear model, Minimum Mean Square Error Estimator, Maximum a Posteriori Estimation
State Estimation: (5 Hrs) Prediction, Single and Multistage Predictors, Filtering, The Kalman Filter

Module-3 - (13 Hrs) Fundamentals of Detection Theory

Hypothesis Testing: Bayes' Detection, MAP Detection, ML Detection, Minimum Probability of Error Criterion, Min-Max Criterion, Neyman-Pearson Criterion, Multiple Hypothesis, Composite Hypothesis Testing: Generalized likelihood ratio test (GLRT), Receiver Operating Characteristic Curves.

Module- 4 - (10 Hrs) Detection of Signals in White Gaussian Noise (WGN)

Binary Detection of Known Signals in WGN, M-ary Detection of Known Signals in WGN, Matched Filter Approach, Detection of signals with Random Parameters

References
1. Jerry M. Mendel, "Lessons in Estimation Theory for Signal Processing, Communication and Control," Prentice Hall Inc., 1995
2 Ralph D. Hippenstiel, "Detection Theory- Applications and Digital Signal Processing", CRC Press, 2002.
3. Steven M. Kay, "Statistical Signal Processing: Vol. 1: Estimation Theory, Vol. 2: Detection Theory," Prentice Hall Inc., 1998.
4 Bernard C. Levy, "Principles of Signal Detection and Parameter Estimation", Springer, New York, 2008.
5. Harry L. Van Trees, "Detection, Estimation and Modulation Theory, Part 1 and 2," John Wiley & Sons Inc. 1968.
6. Neel A. Macmillan and C. Douglas Creelman, "Detection Theory: A User's Guide (Sec. Edn.)" Lawrence Erlbaum Associates Publishers, USA, 2004.
7.Monson H. Hayes, "Statistical Digital Signal Processing and Modelling," John Wiley & Sons Inc., 1996.

EC6405: SIGNAL PROCESSING LAB II

Co-requisite : EC6404, EC6307

L	T	P	C
0	0	3	2

Course Outcomes:

- ***CO1.*** *To experiment the concepts introduced in the courses Adaptive Signal Processing and Estimation and Detection Theory.*
- ***CO2.*** *Ability to program various adaptive filters and optimization of their weight parameters using varying algorithms.*
- ***CO3.*** *Ability to apply algorithms to practical problems involving – system identification, Adaptive Equalization and deconvolution*
- ***CO4.*** *Ability to experiment and identify merits and demerits of various adaptive algorithms.*
- ***CO5.*** *Develop skill to conduct independent research in the area of signal processing*
- ***CO6:*** *Develop programming skills using various tools such as MATLAB or OpenCv and hardware platforms such as DSP processors*

Total Hours: 42 Hrs

Course objective :

To experiment the concepts introduced in the courses EC6404(Adaptive Signal Processing) and EC6307 (Estimation and Detection Theory)

Tools :

Numerical Computing Environments – GNU Octave or MATLAB or any other equivalent tool.

DSP Kits – TMS320C6X or AD or equivalent

Suggested Experiments :

Numerical Computing Environments – Weiner Filtering, LMS filters, System Identification, Adaptive Equalization, Deconvolution

DSP Kits – LMS filtering, Lattice structures, Adaptive Equalization.

EC6406: SEMINAR

L	T	P	C
0	0	3	2

Course Outcomes:

- ***CO1:*** *Student will get exposure to the recent technical advancements.*
- ***CO2:*** *Student will explore and engage in higher order thinking activities related to a recent topic from their academic area.*
- ***CO3:*** *Student learns to acquire the materials, articulate, create and convey intended meaning of their topics effectively.*
- ***CO4:*** *Student learns to express themselves clearly and persuasively in exposition and in argument.*
- ***CO5:*** *Student will practice oral and written communication skills*

SEMESTER III

EC7401: Project Work

L	T	P	C
0	0	0	8

- ***CO1:*** *Envisaging applications for societal needs*
- ***CO2:*** *Ability to conduct a literature survey, identify a research topic, formulate the problem and conduct its feasibility study*
- ***CO3:*** *Unfolds creative and scientific thinking*
- ***CO4:*** *Learns to use new tools effectively and creatively*
- ***CO5:****Develops ability to write Technical / Project reports, make oral presentation and demonstration of the work done to an audience*
- ***CO6:*** *Develops ability to interpret the results, identify the limitations of the work done and make suggestions to rectify them*
- ***CO7:*** *Ability to communicate innovative work in the form of journal/ conference publication*

Syllabus:

The student will be encouraged to fix the area of the project work and conduct the literature review towards the end of second semester. The project work starts in the third semester. The topic shall be research and development oriented in the emerging areas of Signal Processing/ allied fields, under the supervision of a faculty from the ECE Department. The project can be carried out at the institute or in an industry/research organization. (Students desirous of carrying out project in industry or other organization have to fulfill the requirements as specified in the "Ordinances and Regulations for M. Tech.").

SEMESTER IV

EC7402: Project Work

L	T	P	C
0	0	0	12

- ***CO1:*** *Envisaging applications for societal needs*
- ***CO2:*** *Ability to conduct a literature survey, identify a research topic, formulate the problem and conduct its feasibility study*
- ***CO3:*** *Unfolds creative and scientific thinking*
- ***CO4:*** *Learns to use new tools effectively and creatively*
- ***CO5:****Develops ability to write Technical / Project reports, make oral presentation and demonstration of the work done to an audience*
- ***CO6:*** *Develops ability to interpret the results, identify the limitations of the work done and make suggestions to rectify them*
- ***CO7:*** *Ability to communicate innovative work in the form of journal/ conference publication*

Syllabus:

The students are supposed to complete a good quantum of the work in the third semester. There shall be evaluation of the work carried out in the third semester by the PG evaluation committee constituted by the department for Signal Processing Stream. The project work started in the third semester will be extended to the end of the fourth semester. There shall be evaluations of the project work by the committee and by an external examiner during and at the end of fourth semester.

ELECTIVE COUSESS

EC6421: Image & Video Processing

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Acquire strong mathematical skills in the theory of linear systems for analyzing and solving problems in image and video processing: Basic 2-D signal processing, 2-D Fourier and other transforms, convolution and filtering operations in 2-D.*
- ***CO2:*** *Recognize the needs and challenges of our age, and to assess the global and social impacts of image and video processing solutions: Basic understanding of the widespread use of digital imaging and video acquisition systems; the need for effective use of scarce resources such as storage and bandwidth, and ways to provide effective use of them by data compression; social impacts and applications of object recognition systems, such as in security, entertainment and automation fields.*
- ***CO3:*** *Identify, formulate and solve image processing problems: Modeling of digital images, degradations such as noise and motion blur and video data; derivation of conditions for optimal filtering, thresholding, coding and transmission of images and videos; analyzing and evaluating systems; the performance of image enhancement, restoration and coding algorithms through the use of both subjective and objective metrics; identifying the source of redundancy in images and exploiting this redundancy for developing efficient coding techniques.*
- ***CO4:*** *Design and integrate components of image processing systems to satisfy given requirements: Selecting the design parameters for optimal performance of related image processing systems; designing and integrating enhancement and restoration techniques for different applications; integrating different coding tools and selecting the related coding parameters for efficient lossless and lossy image compression; designing simple object segmentation and recognition algorithms.*
- ***CO5:*** *Use the software based modeling, simulation and design tools necessary for practical image processing applications : Design and implementation of enhancement, restoration, coding, and transformation algorithms for image and video data in MATLAB/ C++.*
- ***CO6:*** *Experience working in teams.*
- ***CO7:*** *Acquire communication skills by conducting seminars on the latest topics relevant to the areas covered in the paper.*
- ***CO8:*** *Explore advanced topics in the current research areas of image and video processing which will enable them to engage in lifelong learning in specific areas of interest*

Total Hours : 42 Hrs.

Module 1: (10 Hrs)

Image representation: Gray scale and colour Images, image sampling and quantization. Two dimensional orthogonal transforms: DFT, WHT, Haar transform, KLT, DCT. Image enhancement - filters in spatial and frequency domains, histogram-based processing, homomorphic filtering. Edge detection - non parametric and model based approaches, LOG filters, localisation problem.

Module 2: (10 Hrs)

Image Restoration: Degradation Models, PSF, circulant and block - circulant matrices, deconvolution, restoration using inverse filtering, Wiener filtering and maximum entropy-based methods.

Image Segmentation: Pixel classification, Bi-level thresholding, Multi-level thresholding, P-tile method, Adaptive thresholding, Spectral & spatial classification, Edge detection, Hough transform, Region growing.

Module 3: (11 Hrs)

Fundamental concepts of image compression - Compression models - Information theoretic perspective - Fundamental coding theorem - Lossless Compression: Huffman Coding- Arithmetic coding - Bit plane coding - Run length coding -

Lossy compression: Transform coding - Image compression standards.

Module 4: (11 Hrs)

Video Processing: Representation of Digital Video, Spatio-temporal sampling; Motion Estimation; Video Filtering; Video Compression, Video coding standards.

References

1. 1. A. K. Jain, Fundamentals of digital image processing, Prentice Hall of India, 1989.
2. R. C. Gonzalez, R. E. Woods, *Digital Image Processing*, Pearson Education. II Ed.,2002
3. W. K. Pratt, Digital image processing, Prentice Hall, 1989
4. A. Rosenfold and A. C. Kak, Digital image processing, Vols. 1 and 2, Prentice Hall, 1986.
5. H. C. Andrew and B. R. Hunt, Digital image restoration, Prentice Hall, 1977
6. R. Jain, R. Kasturi and B.G. Schunck, Machine Vision, McGraw-Hill International Edition, 1995
7. A. M. Tekalp, Digital Video Processing , Prentice-Hall, 1995
8. A. Bovik, Handbook of Image & Video Processing, Academic Press, 2000

EC6422: Linear & Nonlinear Optimization

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Provide an adequate background on linear systems and optimization theory.*
- ***CO2:*** *Equip students with basic mathematical techniques commonly used optimization.*
- ***CO3:*** *Introduce a variety of optimization algorithms, along with relative merits and demerits of each on specific conditions to enable students to make sound decisions on real world problems*

Total Hours: 42 Hrs

Module 1: (8 Hours)

Mathematical Background: Sequences and Subsequences- Mapping and functions- Continuous functions- Infimum and Supremum of functions- Minima and maxima of functions- Differentiable functions.

Vectors and vector spaces- Matrices- Linear transformation- Quadratic forms- Definite quadratic forms- Gradient and Hessian- Linear equations- Solution of a set of linear equations-Basic solution and degeneracy.

Convex sets and Convex cones- Introduction and preliminary definition- Convex sets and properties- Convex Hulls- Extreme point- Separation and support of convex sets- Convex Polytopes and Polyhedra- Convex cones- Convex and concave functions- Basic properties- Differentiable convex functions- Generalization of convex functions.

Module 2: (12 hours)

Linear Programming: Introduction -Optimization model, formulation and applications-Classical optimization techniques: Single and multi variable problems-Types of constraints.

Linear optimization algorithms: The simplex method -Basic solution and extreme point -Degeneracy-The primal simplex method -Dual linear programs - Primal, dual, and duality theory - The dual simplex method -The primal-dual algorithm-Duality applications.

Post optimization problems: Sensitivity analysis and parametric programming-

Module 3: (12 hours)

Nonlinear Programming: Minimization and maximization of convex functions- Local & Global optimum-Convergence-Speed of convergence.

Unconstrained optimization: One dimensional minimization - Elimination methods: Fibonacci & Golden section search - Gradient methods - Steepest descent method.

Constrained optimization: Constrained optimization with equality and inequality constraints.

Kelley's convex cutting plane algorithm - Gradient projection method - Penalty Function methods.

Module 4: (10 Hours)

Constrained optimization: Lagrangian method - Sufficiency conditions - Kuhn-Tucker optimality conditions- Rate of convergence - Engineering applications Quadratic programming problems-Convex programming problems.

References

1. David G Luenberger, .Linear and Non Linear Programming., 2nd Ed, Addison-Wesley.
2. S.S.Rao, .Engineering Optimization.; Theory and Practice; Revised 3rd Edition, New Age International Publishers, New Delhi
3. S.M. Sinha, Mathematical programming: Theory and Methods, Elsevier, 2006.
4. Hillier and Lieberman *Introduction to Operations Research*, McGraw-Hill, 8th edition, 2005.
5. Saul I Gass, Linear programming, McGraw-Hill, 5th edition, 2005.
6. Bazarra M.S., Sherali H.D. & Shetty C.M., Nonlinear Programming Theory and Algorithms, John Wiley, New York, 1979.
7. Kalyanmoy Deb, Optimization for Engineering: Design-Algorithms and Examples, Prentice Hall (India), 1998.

EC6423: Signal Compression - Theory and Methods

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Acquire strong mathematical skills in the theory behind various data compression techniques: Concept of Modeling and Coding, Mathematical Preliminaries for Lossy and Lossless Compression, Rate distortion theory.*
- ***CO2:*** *Recognize the needs and challenges of our age, and to assess the global and social impacts of data compression problems and its solutions: Basic understanding of the widespread use of multimedia systems; the need for effective use of scarce resources such as storage and bandwidth, and ways to use effective data compression algorithms to facilitate the current needs.*
- ***CO3:*** *Identify and formulate signal compression problems and design algorithms to solve them effectively by finding optimal solutions for both lossy and lossless data transmission/ storage: Huffman Coding, Arithmetic Coding, Adaptive coding teqchniques, Run Length Coding, Dictionary based Encoding Techniques, Predictive Coding Techniques, Various Quantization schemes and their application in multimedia data compression, Multimedia compression standards.*
- ***CO4:*** *Choose appropriate model for the data to exploit redundancy so as to meet the required rate constraints for a given application and design systems by integrating relevant components: Integrating different coding tools and selecting the related coding parameters for efficient lossless and lossy image compression.*
- ***CO5:****. Use the software based modeling, simulation and design tools necessary for practical image processing applications : Ability to implement various signal compression algorithms in MATLAB/ in a high-level language such as C++*
- ***CO6:*** *Experience working in teams*
- ***CO7:*** *Acquire communication skills by conducting seminars on the current and upcoming multimedia compression technologies and standards.*
- ***CO8:*** *Explore advanced topics in the latest research areas of multimedia data compression technology and techniques which will enable them to engage in lifelong learning in specific areas of interest*

Total Hours: 42 Hrs

Module 1: (12 hours)

Review of Information Theory: The discrete memoryless information source - Kraft inequality; optimal codes Source coding theorem.

Compression Techniques - Lossless and Lossy Compression - Mathematical Preliminaries for Lossless Compression -Huffman Coding - Optimality of Huffman codes - Extended Huffman Coding - Adaptive Huffman Coding - Arithmetic Coding - Adaptive Arithmetic coding, Run Length Coding, Dictionary Techniques - Lempel-Ziv coding, Applications - Predictive Coding - Prediction with Partial Match - Burrows Wheeler Transform, Dynamic Markov Compression.

Module 2: (10 hours)

Rate distortion theory: Rate distortion function R(D),Properties of R(D); Calculation of R(D) for the binary source and the Gaussian source, Rate distortion theorem, Converse of the Rate distortion theorem, Quantization - Uniform & Non-uniform - optimal and adaptive quantization, vector quantization and structures for VQ, Optimality conditions for VQ, Predictive Coding - Differential Encoding Schemes

Module 3: (10 hours)

Mathematical Preliminaries for Transforms, Karhunen Loeve Transform, Discrete Cosine and Sine Transforms, Discrete Walsh Hadamard Transform, Lapped transforms - Transform coding - Subband coding - Wavelet Based Compression - Analysis/Synthesis Schemes

Module 4: (10 hours)

Data Compression standards: Zip and Gzip, Speech Compression Standards: PCM-G.711, ADPCM G.726, SBC G.722, LD-CELP G.728, CS-ACELP (-A) G.729, MPC-MLQ , G.723.1, GSM HR VSELP, IS-54 VSELP, IS-96 QCELP, Immarsat - B APC, MELP, FS 1015, LPC10, FS1016, CELP, G721. Audio Compression standards: MPEG, Philips PASC, Sony ATRAC, Dolby AC-3, Image Compression standards: JBIG, GIF, JPEG & JPEG derived industry standards, CALIC, SPIHT, EZW, JPEG 2000. Video Compression Standards: MPEG, H.261, H.263 & H264.

References

1. Khalid Sayood, “Introduction to Data Compression”, Morgan Kaufmann Publishers., Second Edn., 2005.
2. David Salomon, “Data Compression: The Complete Reference”, Springer Publications, 4th Edn., 2006.
3. Thomas M. Cover, Joy A. Thomas, “Elements of Information Theory," John Wiley & Sons, Inc., 1991.
4. Toby Berger, “Rate Distortion Theory: A Mathematical Basis for Data Compression”, Prentice Hall, Inc., 1971
5. K.R.Rao, P.C.Yip, “The Transform and Data Compression Handbook”, CRC Press., 2001.
6. R.G.Gallager, “Information Theory and Reliable Communication”, John Wiley & Sons, Inc., 1968.
7. Ali N. Akansu, Richard A. Haddad, “Multiresolution Signal Decomposition: Transforms, Subbands and Wavelets”, Academic Press., 1992
8. Martin Vetterli, Jelena Kovacevic, “Wavelets and Subband Coding”, Prentice Hall Inc., 1988

EC6424: MULTIDIMENSIONAL SIGNAL PROCESSING

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1***:*Understand the concepts of multi-dimensional signals and systems and learn the difference between one dimensional and multi dimensional signal processing systems.*
- ***CO2***:*Get a deep understanding on how to apply these concepts i.e. how to acquire, process and display two or multi-dimensional signals.*
- ***CO3***:*Learn the concept of 2D sampling theorem and sampling with different sampling geometries.*
- ***CO4***:*Understand what representation of multi dimensional signal is most suitable for manipulation and resolution adaptation.*
- ***CO5***:*Design and implement various types of two dimensional digital filters given a set of specifications.*
- ***CO6***: *Explore various applications and research areas related to the topics covered in the paper*

Total Hours: 42 Hrs

Module 1: (11 hours) Multidimensional Systems

Fundamental operations on Multidimensional signals, Linear Shift - Invariant systems-cascade and parallel connection of systems- separable systems, stable systems- Frequency responses of 2D LTI Systems- Impulse response- Multidimensional Fourier transforms- z transform, properties of the Fourier and z transform.

Module 2: (10 hours) Sampling continuous 2D signals

Periodic sampling with rectangular geometry- sampling density, Aliasing effects created by sampling - Periodic sampling with different sampling geometrics-hexagonal- Quincunx etc.- comparison

Module 3: (11 hours) Multidimensional Discrete Fourier Transform

Multidimensional discrete Fourier transform- Properties of DFT, Circular convolution- Calculation of DFT- DFT for periodically sampled signals - Fast Fourier transform for periodically sampled signals- The Discrete Cosine Transform.

Module 4: (10 hours) Multidimensional Digital Filter Design

Separable Filters- Linear phase filters- FIR Filters- Implementation of FIR filters - design of FIR filters using windows- Two dimensional window functions, IIR Filters

References

1. Dudgeon Dan E. , Multidimensional Digital Signal Processing, Prentice Hall, Englewood Cliffs, New Jersey
2. P.P. Vaidyanathan. “Multirate systems and filter banks.” Prentice Hall. PTR. 1993.
3. Two- Dimensional Signal and Image Processing , JAE S. LIM - Prentice Hall Englewood Cliffs, New Jersey, 1990.

EC6425: Wavelets: Theory & Construction

Pre-requisites: Linear system theory, Multirate systems and filter banks

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Understand the mathematical basis of the wavelet transform and its performance in the analysis of non – stationary signals.*
- ***CO2:*** *Understand the concepts and properties of Continuous Wavelet Transform, Multi-Resolution Analysis, Discrete Wavelet Transform and Wavelet Packets.*
- ***CO3:*** *Learn to implement Discrete Wavelet transform using Filter banks and Fast Lifting Scheme.*
- ***CO4:*** *Learn the time domain and frequency domain approaches for the construction of wavelets.*
- ***CO5:****. Learn to design and implement wavelet packet transform & best basis algorithm for a desired application*
- ***CO6:*** *Implement Discrete Wavelet Transform and Wavelet Packet Transform for various applications like Signal compression, de-noising, detection of anomalies in ECG, EEG etc*

Total Hours: 42 Hrs

(1. a) Fourier and Sampling Theory: (5 hours)
Generalized Fourier theory, Fourier transform, Short-time(windowed) Fourier transform, Time-frequency analysis, Fundamental notions of the theory of sampling.
(1. b) Theory of Frames: (6 hours)
Bases, Resolution of unity, Definition of frames, Geometrical considerations and the general notion of a frame, Frame projector, Example – windowed Fourier frames.

(2. a) Wavelets: (4 hours)
The basic functions, Specifications, Admissibility conditions, Continuous wavelet transform (CWT), Discrete wavelet transform (DWT).
(2. b) The multiresolution analysis (MRA) of $L^2(R)$: (5 hours)
The MRA axioms, Construction of an MRA from scaling functions - The dilation equation and the wavelet equation, Compactly supported orthonormal wavelet bases - Necessary and sufficient conditions for orthonormality.

(3) Regularity and selection of wavelets: (5 hours)
Smoothness and approximation order - Analysis in Soboleve space, Criteria for wavelet selection with examples.

(4. a) Construction of wavelets (1): (5 hours)
Splines, Cardinal B-spline MRA, Subband filtering schemes, Compactly supported orthonormal wavelet bases.
(4. b) Wavelet transform: (6 hours)
Wavelet decomposition and reconstruction of functions in $L^2(R)$. Fast wavelet transform algorithms - Relation to filter banks, Wavelet packets – Representation of functions, Selection of basis.

(5) Construction of wavelets (2): (6 hours)
Biorthogonality and biorthogonal basis, Biorthogonal system of wavelets - construction, The Lifting scheme.

References

1. Stephen G. Mallat, “A wavelet tour of signal processing” 2nd Edition Academic Press, 2000.
2. M. Vetterli, J. Kovacevic, “Wavelets and subband coding” Prentice Hall Inc, 1995
3 Gilbert Strang and Truong Q. Nguyen, “Wavelets and filter banks” 2nd Edition Wellesley-Cambridge Press, 1998.
4 Gerald Kaiser, “A friendly guide to wavelets” Birkhauser/Springer International Edition, 1994, Indian reprint 2005.
5 L. Prasad and S. S. Iyengar, “Wavelet analysis with applications to image processing” CRC Press, 1997.
6 J. C. Goswami and A. K. Chan, “Fundamentals of wavelets: Theory, Algorithms and Applications” Wiley-Interscience Publication, John Wiley & Sons Inc., 1999.
7 Mark A. Pinsky, “Introduction to Fourier Analysis andWavelets” Brooks/Cole Series in Advanced Mathematics, 2002
8 Christian Blatter, “Wavelets: A primer” A. K. Peters, Massachusetts,1998.
9 M. Holschneider, “Wavelets: An analysis tool” Oxford Science Publications, 1998.
10 R. M. Rao and A. Bopardikar, “Wavelet transforms: Introduction to theory and applications” Addison-Wesley, 1998.
11 Ingrid Daubechies, “Ten lectures on wavelets” SIAM, 1990.
12 H. L. Resnikoff and R. O. Wells, Jr., “Wavelet analysis: The scalable structure of information” Springer, 1998.

13 P. P. Vaidyanathan, “Multirate systems and filter banks” Prentice Hall P T R, 1993.
14 P. Wojtaszczyk, “A mathematical introduction to wavelets” Cambridge University Press 1997.
15 Michael W. Frazier, “An introduction to wavelets through linear algebra” Springer-Verlag, 1999.
16 Anthony N. Michel and Charles J. Herget, “Applied algebra and functional analysis” Dover Publications Inc., 1993.

EC6426:   Transform Theory

Pre-requisite: Introductory understanding of Linear Algebra.

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Learns foundation concepts of Transformations on vector spaces.*
- ***CO2:*** *Learns representations of signals in various domains amenable for further processing.*
- ***CO3:*** *Develops analytical skills and ability to synthesize starting from first principles.*
- ***CO4:*** *Unfolds creative, scientific thinking and ability to visualize subtle concepts leading to cost effective and environment-friendly designs .*
- ***CO5:****. Develops clarity of thoughts, logical reasoning and ability to express thoughts with precision*

Total Hours : 42Hrs

(1) Linear Operators on Finite-dimensional Vector Spaces (9 hours)
Eigenvalue problems, eigenvalues, eigenvectors and eigenspace of a linear operator, Linear operators with an eigenbasis, decomposition of vector spaces, Similarity transformations - Diagonalization, Primary decomposition theorem, Jordan Canonical form/decomposition; Fredholm alternative theorem, Least squares solutions and pseudo-inverses, LU decomposition, Orthogonal transformations, Singular value decomposition, Householder transformation.

(2. a) Normed Linear Spaces (4 hours)
Functionals - Norm, Convergence - Cauchy sequence, Completeness of vector spaces; Infinite dimensional vector spaces - Normed linear spaces; Banach Spaces, Inner product spaces, Hilbert spaces; Continuous **linear operators.**
(2. b) Bounded Linear Operators and Spectral Theory (9 hours)
Bounded linear operators in finite dimensional inner product spaces - Adjoint of an operator, Norm of an operator; Self-adjoint operators - Spectral analysis of self-adjoint operators; Bessel's inequality, Parseval's identity; Reisz Representation Theorem, Compact linear operators.

(3. a) Theory of Distributions (5 hours)
Generalized functions and the Dirac's delta; Differential operators - Green's function and the inverse linear operators.
(3. b) The Making of Integral Transforms (5 hours)
The making of Laplace transform and Fourier transform, Self-reciprocal functions and operators under Fourier transform - The construction of Fractional Fourier transform; Construction of z-transform - Discrete-time Fourier transform and discrete Fourier transform.
(3. c) Lapped Transforms (6 hours)
Karhunen-Loeve transform - Lapped orthogonal transforms and biorthogonal transforms – Construction of discrete cosine and sine transforms.

(4) The Making of Continuous Wavelet Transform (4 hours)
Reisz basis, Resolution of unity, Definition of frames, Geometrical considerations and the general notion of a frame, Frame projector, Example - windowed Fourier frames; Continuous wavelet transform.

References

1.Arch W. Naylor and George R. Sell, "Linear Operator Theory in Engineering and Science," 2nd Edition, Springer-Verlag, New York, 1982.
2. Larry Smith, "Linear Algebra," 2nd Edition, Springer-Verlag, New York 1982
3. Lokenath Debnath and Piotr Mikusinski, "Hilbert Spaces with Applications," 3rd Edition, Academic Press, Indian reprint 2006.
4. A. David Wunsch, "Complex Variables with Applications," 2nd Edition, Addison-Wesley Publishing Company, New York, 1994.
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11. Stephen G. Mallat, "A Wavelet Tour of Signal Processing," 2nd Edition, Academic Press, 2000.
12. Gerald Kaiser, "A Friendly Guide to Wavelets," Birkhauser/Springer International Edition, 1994, Indian reprint 2005.
13 Ingrid Daubechies, "Ten Lectures on Wavelets," SIAM, 1990.

EC6427: Array Signal Processing

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Obtain basic understanding of spatial signals and its analysis.*
- ***CO2:*** *Familiarize with the basic knowledge of the sensor arrays.*
- ***CO3:*** *Perform the spatial frequency analysis of array signals.*
- ***CO4:*** *Become familiar to various methods and algorithm the estimation of direction of arrival.*
- ***CO5:*** *Explore various application areas related to the topic of study through literature review and improve the communication skills through oral presentation*

Total Hours: 42 Hrs

Module 1 : (10 hours) Spatial Signals

Signals in space and time. Spatial frequency, Direction vs. frequency. Wave fields. Far field and Near field signals.

Module 2 : (10 hours) Sensor Arrays

Spatial sampling, Nyquist criterion. Sensor arrays. Uniform linear arrays, planar and random arrays. Array transfer (steering) vector. Array steering vector for ULA. Broadband arrays.

Module 3 : (10 hours) Spatial Frequency

Aliasing in spatial frequency domain. Spatial Frequency Transform, Spatial spectrum. Spatial Domain Filtering. Beam Forming. Spatially white signal.

Module 4 : (12 hours) Direction of Arrival Estimation

Non parametric methods - Beam forming and Capon methods. Resolution of Beam forming method. Subspace methods - MUSIC, Minimum Norm and ESPRIT techniques. Spatial Smoothing.

References

1. Dan E. Dugeon and Don H. Johnson. (1993). Array Signal Processing: Concepts and Techniques. Prentice Hall.
2. Petre Stoica and Randolph L. Moses. (2005, 1997) Spectral Analysis of Signals. Prentice Hall.
3. Bass J, McPheeters C, Finnigan J, Rodriguez E. Array Signal Processing [Connexions Web site]. February 8, 2005. Available at: http://cnx.rice.edu/content/col10255/1.3/

EC6428: Speech & Audio Processing

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Derive the digital model of speech production.*
- ***CO2:*** *Understand the Algorithms for speech analysis and synthesis*
- ***CO3:*** *Analyze Speech coding techniques*
- ***CO4:*** *Speech and speaker recognition systems.*
- ***CO5:***. *Concepts of Audio Processing and learn modeling*
- ***CO6:*** *Implement Applications-New audiogram matching techniques*

Total Hours: 42 Hrs

Module I (10 hrs)

Digital models for the speech signal - mechanism of speech production - acoustic theory - lossless tube models - digital models - linear prediction of speech - auto correlation - formulation of LPC equation - solution of LPC equations - Levinson Durbin algorithm - Levinson recursion - Schur algorithm - lattice formulations and solutions - PARCOR coefficients - Spectral analysis of speech - Short Time Fourier analysis - filter bank design. Auditory Perception : Psychoacoustics- Frequency Analysis and Critical Bands - Masking properties of human ear.

Module 2 (12 hrs)

Speech coding -subband coding of speech - transform coding - channel vocoder - formant vocoder - cepstral vocoder - vector quantizer coder- Linear predictive Coder. Speech synthesis - pitch extraction algorithms - gold rabiner pitch trackers - autocorrelation pitch trackers - voice/unvoiced detection - homomorphic speech processing - homomorphic systems for convolution - complex cepstrums - pitch extraction using homomorphic speech processing. Sound Mixtures and Separation - CASA, ICA & Model based separation.

Module 3 (10 hrs)

Speech Transformations - Time Scale Modification - Voice Morphing. Automatic speech recognition systems - isolated word recognition - connected word recognition -large vocabulary word recognition systems - pattern classification - DTW, HMM - speaker recognition systems - speaker verification systems - speaker identification Systems.

Module 4 (10 hrs)

Audio Processing : Non speech and Music Signals - Modeling -Differential, transform and subband coding of audio signals & standards - High Quality Audio coding using Psychoacoustic models - MPEG Audio coding standard. Music Production - sequence of steps in a bowed string instrument - Frequency response measurement of the bridge of a violin. Audio Data bases and applications - Content based retrieval.

References

1. Rabiner L.R. & Schafer R.W., “Digital Processing of Speech Signals”, Prentice Hall Inc.
2. O'Shaughnessy, D. “Speech Communication, Human and Machine”. Addison-Wesley.
3. Thomas F. Quatieri , “Discrete-time Speech Signal Processing: Principles and Practice” Prentice Hall, Signal Processing Series.
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5. Ben Gold & Nelson Morgan , “ Speech and Audio Signal Processing”, John Wiley & Sons, Inc.
6. Owens F.J., “Signal Processing of Speech”, Macmillan New Electronics
7. Saito S. & Nakata K., “Fundamentals of Speech Signal Processing”, Academic Press, Inc.
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10. Jayant, N. S. and P. Noll. “Digital Coding of Waveforms: Principles and Applications to Speech and Video. Signal Processing Series”, Englewood Cliffs: Prentice-Hall
11. Thomas Parsons, “Voice and Speech Processing”, McGraw Hill Series
12. Chris Rowden, “Speech Processing”, McGraw-Hill International Limited
13. Moore. B, “An Introduction to Psychology of hearing”Academic Press, London, 1997
14. E. Zwicker and L. Fastl, “Psychoacoustics-facts and models”, Springer-Verlag., 1990

EC6429: Biomedical Signal Processing

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Understand the fundamental tools that are used to model, analyze and process biomedical signals like ECG, EEG, EMG etc.*
- ***CO2:*** *Understand the techniques to eliminate background noise, enhance signal detection, analyze computer data, make results easy to comprehend and apply.*
- ***CO3:*** *Understand the principles in the analysis and design of adaptive and optimal filters, power spectral density estimation and non stationary signal processing techniques with applications to biomedical signals.*
- ***CO4:*** *Understand electrical activity of the heart, Multi scale analysis for the estimation of ECG parameters, ECG Data Compression and Spectral Analysis of Heart rate Variability.*
- ***CO5:****. Understand the techniques for EEG Analysis & Modeling, apply those techniques in Epilepsy Detection, Study of Sleep Disorders and Development of Brain Computer Interface*

Total Hours: 42 Hrs

Module 1 (10 hrs)

Introduction to Biomedical Signals - Examples of Biomedical signals - ECG, EEG, EMG etc - Tasks in Biomedical Signal Processing - Computer Aided Diagnosis. Origin of bio potentials - Review of linear systems - Fourier Transform and Time Frequency Analysis (Wavelet) of biomedical signals- Processing of Random & Stochastic signals - spectral estimation - Properties and effects of noise in biomedical instruments - Filtering in biomedical instruments

Module 2 (10 hrs)

Concurrent, coupled and correlated processes - illustration with case studies - Adaptive and optimal filtering - Modeling of Biomedical signals - Detection of biomedical signals in noise - removal of artifacts of one signal embedded in another -Maternal-Fetal ECG - Muscle-contraction interference. Event detection - case studies with ECG & EEG - Independent component Analysis - Cocktail party problem applied to EEG signals - Classification of biomedical signals.

Module 3 (11 hrs)

Cardio vascular applications : Basic ECG - Electrical Activity of the heart- ECG data acquisition - ECG parameters & their estimation - Use of multiscale analysis for ECG parameters estimation - Noise & Artifacts- ECG Signal Processing: Baseline Wandering, Power line interference, Muscle noise filtering - QRS detection - Arrhythmia analysis - Data Compression: Lossless & Lossy- Heart Rate Variability - Time Domain measures - Heart Rhythm representation - Spectral analysis of heart rate variability - interaction with other physiological signals.

Module 4 (11 hrs)

Neurological Applications : The electroencephalogram - EEG rhythms & waveform - categorization of EEG activity - recording techniques - EEG applications- Epilepsy, sleep disorders, brain computer interface. Modeling EEG- linear, stochastic models - Non linear modeling of EEG - artifacts in EEG & their characteristics and processing - Model based spectral analysis - EEG segmentation - Joint Time-Frequency analysis - correlation analysis of EEG channels - coherence analysis of EEG channels.

References

1. Bruce, “Biomedical Signal Processing & Signal Modeling,” Wiley, 2001
2. Sörnmo, “Bioelectrical Signal Processing in Cardiac & Neurological Applications”, Elsevier
3. Rangayyan, “Biomedical Signal Analysis”, Wiley 2002.
4. Semmlow, Marcel Dekker “Biosignal and Biomedical Image Processing”, 2004
5. Enderle, “Introduction to Biomedical Engineering,” 2/e, Elsevier, 2005
6. D.C.Reddy , “ Biomedical Signal Processing: Principles and techniques” , Tata McGraw Hill, New Delhi, 2005

EC6430: Pattern Recognition and Analysis

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Provide an adequate background on probability theory, statistics and optimization theory.*
- ***CO2:*** *Equip students with basic mathematical and statistical techniques commonly used in pattern recognition.*
- ***CO3:*** *Introduce a variety of pattern recognition algorithms, along with relative merits and demerits of each on specific conditions to enable students to make sound decisions on real world problems.*
- ***CO4:*** *Develop skill to conduct independent research in the area of pattern recognition*

Total Hours: 42 Hrs

Module 1: (12 hrs)

Introduction - features, feature vectors and classifiers, Supervised versus unsupervised pattern recognition. Classifiers based on Bayes Decision theory- introduction, discriminant functions and decision surfaces, Bayesian classification for normal distributions, Estimation of unknown probability density functions, the nearest neighbour rule. Linear classifiers,- Linear discriminant functions and decision hyper planes, The perceptron algorithm, MSE estimation, Logistic determination, Support Vector machines.

Module 2: (10 hrs)

Non-Linear classifiers- Two layer and three layer perceptrons, Back propagation algorithm, Networks with Weight sharing, Polynomial classifiers, Radial Basis function networks, Support Vector machines-nonlinear case, Decision trees, combining classifiers, Feature selection, Receiver Operating Characteristics (ROC) curve, Class separability measures, Optimal feature generation, The Bayesian information criterion.

Module 3: (10 hrs)

Feature Generation 1- Linear transforms-KLT, SVD, ICA, DFT, DCT, DST, Hadamard Transform, Wavelet Transform, Wavelet Packets etc- Two dimensional generalizations - Applications. Feature Generation 2- regional features, features for shape and characterization, Fractals, typical features for speech and audio classification, Template Matching, Context dependent classification-Bayes classification, Markov chain models, HMM, Viterbi Algorithm. System evaluation - Error counting approach, Exploiting the finite size of the data.

Module 4 (10 hrs)

Clustering- Cluster analysis, Proximity measures, Clustering Algorithms - Sequential algorithms, Neural Network implementation. Hierarchical algorithms - Agglomerative algorithms, Divisive algorithms. Schemes based on function optimization - Fuzzy clustering algorithms, Probabilistic clustering, K - means algorithm. Clustering algorithms based on graph theory , Competitive learning algorithms, Binary Morphology Clustering Algorithms Boundary detection methods, Valley seeking clustering, Kernel clustering methods. Clustering validity.

References

1. Sergios Theodoridis, Konstantinos Koutroumbas, “Pattern Recognition”, Academic Press, 2006.
2. Duda and Hart P.E, Pattern classification and scene analysis, John Wiley and sons, NY, 1973.
3. Earl Gose, Richard Johnsonbaugh, and Steve Jost; Pattern Recognition and Image Analysis, PHI Pvte. Ltd., NewDelhi-1, 1999.
4. Fu K.S., Syntactic Pattern recognition and applications, Prentice Hall, Eaglewood cliffs, N.J., 1982
5. Rochard O. Duda and Hart P.E, and David G Stork, Pattern classification , 2nd Edn., John Wiley & Sons Inc., 2001
6. Andrew R. Webb, “ Statistical Pattern Recognition”, John Wiley & Sons, 2002

EC6431: DSP Algorithms and Architectures

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- *CO1: To familiarize various VLSI algorithms and architectures for DSP: VLSI algorithm transforms including retiming, pipelining and parallel processing ,folding/unfolding, algebraic transforms, relaxed look-ahead transforms, and dynamic algorithm transforms, systolic architectures for DSP, asynchronous and wave pipelines.*
- *CO2: Analyze and design the DSP systems by using the appropriate DSP algorithms and architectures: Various fast convolution algorithms, Filter techniques and structures, Design using systolic architectures.*
- *CO3: Understand the practically and economically feasible design: scaling and round-off noise issues and their impact on performance of the digital system, performance evaluation of various digital signal processors in terms of low power dissipation.*
- *CO4: Use software based modeling, simulation and design tools necessary for practical implementation of DSP architectures : Ability to implement various algorithms such that it span entire design hierarchy from algorithm design to integrated circuit layout using a mix of tools such as MATLAB (algorithm design), Verilog or VHDL (architecture), SYNOPSYS (logic), and CADENCE (circuit).*
- *CO5: Explore advanced topics in the latest research areas of DSP system design at the algorithmic and architectural levels through group works and seminars which will enable them to engage in lifelong learning in specific areas of interest*

Total Hours: 42 Hrs

DSP Algorithm Design (12 hours)

DSP representations (data-flow, control-flow, and signal-flow graphs, block diagrams),fixed-point DSP design (A/D precision, coefficient quantization, round-off and scaling),filter structures (recursive, non-recursive and lattice), algorithmic simulations of DSP systems in C , behavioral modeling in HDL. System modeling and performance measures.

Circuits and DSP Architecture Design (10 hours)

Fast filtering algorithms (Winograd's, FFT, short- length FIR), retiming and pipelining, block processing, folding, distributed arithmetic architectures, VLSI performance measures (area, power, and speed), structural modeling in VHDL. Analog signal processing for fast operation. Impact of nonideal characteristics of analog functional blocks on the system performance.

DSP Module Synthesis (10 hours)

Distributed arithmetic (DA). Advantageous of using DA? Size reduction of look-up tables. Canonic signed digit arithmetic. Implementation of elementary functions Table-oriented methods. Polynomial approximation Random number generators. Linear feedback shift register. High performance arithmetic unit architectures (adders, multipliers, dividers), bit-parallel, bit-serial, digit-serial, carry-save architectures, redundant number system, modeling for synthesis in HDL, synthesis place-and-route.

Parallel algorithms and their dependence (10 hours)

Applications to some common DSP algorithms. System timing using the scheduling vector. Projection of the dependence graph using a projection direction. The delay operator and z-transform techniques for mapping DSP algorithms onto processor arrays. Algebraic technique for mapping algorithms. The computation domain. The dependence matrix of a variable. The scheduling and projection functions.Data broadcast and pipelining.
Applications using common DSP algorithms.

References

1. Digital Signal Processors: Architectures, Implementations, and Applications Sen M.Kuo , Woon-Seng S. Gan Prentice Hall 2004
2. VLSI Signal Processing Systems, Design and Implementation.Keshab K. Parhi, John Wiley & Sons,1999.
3. Digital Signal Processing with Field Programmable Gate Array, Uwe Meyer-Baese, Springer- Verlag 2001
4. DSP Principles, Algorithms and Applications, John G. Proakis , Dimitris Manolakis K - Prentice Hall 1995
5. Architectures for Digital Signal Processing, Pirsch, John Wiley and Sons, 1998.
6. DSP Integrated Circuits, Lars Wanhammar, Academic Press, 1999
7. Computer Arithmetic: Algorithms and Hardware Designs, Parhami, Behrooz, Oxford University Press, 2000

8. Computer Arithmetic Algorithms, Israel Koren, A. K. Peters, Natick, MA, 2002.

EC6432: Spectrum Analysis

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- *CO1: Obtain basic understanding of energy spectral density and power spectral density.*
- *CO2: Familiarize with various parametric and non-parametric techniques for estimating the power spectral density*
- *CO3: Become familiar to different filterbank methods for spectral analysis*
- *CO4: Explore various application areas related to the topic of study through literature to engage them in lifelong learning*

Total Hours: 42 Hrs

Module 1: (5 hours) Power Spectral Density

Energy spectral density of deterministic signals, Power spectral density of random signals, Properties of PSD.

Module 2 : (10 hours) PSD Estimation - Non-parametric methods

Estimation of PSD from finite data, Non-parametric methods : Periodogram properties, bias and variance analysis, Blackman-Tuckey method, Window design considerations, time-bandwidth product and resolution - variance trade-offs in window design, Refined periodogram methods : Bartlet method, Welch method.

Module 3 : (15 hours) PSD Estimation - Parametric methods

Parametric method for rational spectra :- Covariance structure of ARMA process, AR signals, Yule-Walker method, Least square method, Levinson-Durbin Algorithm, MA signals, Modified Yule-Walker method, Two-stage least square method, Burg method for AR parameter estimation.

Parametric method for line spectra :- Models of sinusoidal signals in noise, Non-linear least squares method, Higher order Yule-Walker method, MUSIC and Pisayenko methods, Min-norm method, ESPIRIT method.

Module IV: (12 hours) Filterbank methods

Filterbank interpretation of periodogram, Slepia base-band filters, refined filterbank method for higher resolution spectral analysis, Capon method, Introduction to higher order spectra.

References

1. Introduction to Spectral Analysis, Stoica , R.L. Moses, Prentice Hall
2. Modern Spectral Estimation Theory & Applications, Kay SM, Prentice Hall

EC6433: Compressive Sensing: Theory and Algorithms

Pre-requisite: Nil

L	T	P	C
3	0	0	3

Course Outcomes:

- ***CO1:*** *Foundations on signal representation and the sampling theorem*
- ***CO2:*** *Mathematical framework for multi-rate systems, and multi-band and multi-resolution representation of signals*
- ***CO3:*** *Motivation for compressed sampling for energy-efficient designs*
- ***CO4:*** *A foundation subject for multi-disciplinary applications*
- ***CO5:****. Basic concepts that enable designs for environment-friendly direct applications*
- ***CO6:*** *Develops ability to apply mathematical principles for cost-effective practical designs.*
- ***CO7:*** *Develops ability to think clearly with sound logical reasoning to unfold new theory and design methods.*

Total Hours: 42 Hrs

Module 1: (6 hours) Fundamentals of Sampling Analog Signals

Classical sampling theorem for band-limited signals; Bandpass sampling theorem; Sample-rate reduction and multi-channel sampling; Sampling of random signals; Sampling of duration-limited signals and motivation for compressed sampling.

Module 2 : (10 hours) Signal Models - Mathematical Preliminaries

Sampling as a signal representation problem; Signal spaces: normed linear spaces - topology, Convergence, completeness and stable signal synthesis; Finite and infinite dimensional signal spaces; Hamel basis, Schauder basis and Riesz basis; Orthogonality and bi-orthogonality; Frames; Linear transformations and change of basis; Sampling as an isomorphism; Separable signal spaces, Quotient spaces and, Decomposition of signals; Under-determined system of equations - methods of solution, sparse solution.

Module 3 : (12 hours) Compressed Sensing

Sparse representation of signals - Sparsity and compressibility; Construction of measurement basis - Sensing matrix; Null-space conditions and the spark; Johnson-Lindenstrauss (JL) lemma; The Restricted Isometry Property (RIP); relation between JL lemma and the RIP; RIP and null-space property; Measurement bounds and condition for stable recovery; Coherence of measurement basis; mutual coherence between sensing and representation bases.

Parametric method for line spectra :- Models of sinusoidal signals in noise, Non-linear least squares method, Higher order Yule-Walker method, MUSIC and Pisayenko methods, Min-norm method, ESPIRIT method.

Module IV: (14 hours) Sparse Signal Recovery

Recovery through l1-norm minimization; Recovery under noiseless and noisy conditions; Algorithms for sparse recovery - Design requirements; Convex optimization based methods: linear programming, fixed-point continuation, Bergman iteration; Greedy algorithms: Matching pursuit, Orthogonal matching pursuit, Stage-wise orthogonal matching pursuit, Regularized orthogonal matching pursuit; Compressive sampling matching pursuit; Performance analysisFilterbank interpretation of periodogram, Slepia base-band filters, refined filterbank method for higher resolution spectral analysis, Capon method, Introduction to higher order spectra.

References

1. Yonina C. Eldar and Gitta Kutyniok, \Compressed Sensing: Theory and Applications," Cambridge University Press, 2012.
2. Richard G. Baraniuk, Mark A. Davenport, Marco F. Duarte, Chinmay Hegde (Collection Editors), An Introduction to Compressive Sensing," CONNEXIONS (Publishing) Rice University, Houston, Texas, 2012.
3. Michael Elad, Sparse and Redundant Representations," Springer, New York, 2010.
4. S. G. Mallat, A Wavelet Tour of Signal Processing: The Sparse Way," Academic Press/Elsevier, 2009.