VISVESVARAYA TECHNOLOGICAL UNIVERSITY, BELGAUM

SCHEME OF TEACHING AND EXAMINATION FOR

M.Tech. Signal Processing

I Semester **CREDIT BASED**

Subject Code	Name of the Subject	Teaching hours/week		Duration of Exam in Hours	Marks for		Total Marks	CREDITS
		Lecture	Practical / Field Work / Assignment/ Tutorials		I.A.	Exam		
14ELD11	Advanced Mathematics	4	2	3	50	100	150	4
14ESP12	Statistical signal processing	4	2	3	50	100	150	4
14ESP13	Advances in Digital Signal Processing	4	2	3	50	100	150	4
14ESP14	Adaptive Signal Processing	4	2	3	50	100	150	4
14ESP15X	Elective - 1	4	2	3	50	100	150	4
14ESP16	Signal Processing lab	--	3	3	25	50	75	2
14ESP17	Seminar on Advanced topics from refereed journals	--	3	--	25	--	25	1
Total		**20**	**16**	**18**	**300**	**550**	**850**	**23**

Elective-1:

14ECS151	Wireless and Mobile Networks	14ESP154	Modern Spectral Analysis & Estimation
14ESP152	Multirate Systems and Filter Banks	14 ELD155	Simulation, Modeling, and Analysis
14ESP153	Speech And Audio Processing		

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II Semester **CREDIT BASED**

Subject Code	Name of the Subject	Teaching hours/week		Duration of Exam in Hours	Marks for		Total Marks	CREDITS
		Lecture	Practical / Field Work / Assignment/ Tutorials		I.A.	Exam		
14ESP21	Digital Signal Compression	4	2	3	50	100	150	4
14ESP22	DSP System Design	4	2	3	50	100	150	4
14ESP23	Image and Video Processing	4	2	3	50	100	150	4
14ESP24	Biomedical Signal Processing	4	2	3	50	100	150	4
14ESP25X	Elective-2	4	2	3	50	100	150	4
14ESP26	Image Processing Lab		3	3	25	50	75	2
14ESP27	Seminar on Advanced topics from refereed journals	--	3	--	25	--	25	1
**Project Phase-I(6 week Duration)								
	Total	**20**	**16**	**18**	**300**	**550**	**850**	**23**

Elective - 2:

14ESP251	Detection & Estimation	14ESP254	Advances in Digital Image Processing
14ECS252	Advanced Digital Communication	14ESP255	Error Control Coding
14ESP253	Pattern Recognition		

**** Between the II Semester and III Semester, after availing a vocation of 2 weeks.**

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III Semester: INTERNSHIP # **CREDIT BASED**

Course Code	Subject	Teaching hours/week		Duration of the Exam in Hours	Marks for		Total Marks	CREDITS
		Lecture	Practical / Field Work		I.A.	Exam		
14ESP31	Midterm Presentation on Internship (After 8 weeks from the date of commencement) *	-	-	-	25	-	25	4
14ESP 32	Report on Internship (After 16 weeks from the date of commencement)	-	-	-	75		75	12
14ESP 33	Evaluation and Viva-voce	-	-	3	–	50	50	4
	Total	**-**	**-**	**-**	**100**	**50**	**150**	**20**

* The student shall make a midterm presentation of the activities undertaken during the first 8 weeks of internship to a panel comprising Internship Guide, a senior faculty from the department and Head of the Department.

The College shall facilitate and monitor the student internship program.

The internship report of each student shall be submitted to the University.

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IV Semester **CREDIT BASED**

Subject Code	Subject	Teaching hours/week		Duration of Exam in Hours	Marks for		Total Marks	CREDITS
		Lecture	Practical / Field Work / Assignment/ Tutorials		I.A.	Exam		
14ESP41	DSP Algorithms and Architecture	4	2	3	50	100	150	4
14ESP42X	Elective-3	4	2	3	50	100	150	4
14ESP43	Evaluation of Project Phase-I	-	-	-	25	-	25	1
14ESP44	Phase-II : Midterm evaluation of Project #	-	-	-	25	-	25	1
14ESP45	Evaluation of Project Work and Viva-voce	–	-	3	-	100+100	200	18
	Total	**8**	**04**	**09**	**150**	**400**	**550**	**28**
Grand Total (I to IV Sem.) : 2400 Marks; 94 Credits								

Elective-3:

14ECS421	RF MEMS	14ECS424	Communication System design using DSP algorithm
14ESP422	Mobile Computing	14ESP425	Multirate Signal Processing
14ESP423	Modern DSP		

Note:

1) Project Phase – I**:** 6 weeks duration shall be carried out between II and III Semesters. Candidates in consultation with the guides shall carryout literature survey / visit to Industries to finalize the topic of dissertation.
2) Project Phase – II**:** 16 weeks duration during III Semester. Evaluation shall be taken during the Second week of the IV Semester. Total Marks shall be 25.
3) Project Evaluation: 24 weeks duration in IV Semester. Project Work Evaluation shall be taken up at the end of the IV Semester. Project Work Evaluation and Viva-Voce Examinations shall be conducted. Total Marks shall be 250 (Phase I Evaluation: 25 Marks, Phase –II Evaluation: 25 Marks, Project Evaluation marks by Internal Examiner (guide): 50, Project Evaluation marks by External Examiner: 50, marks for external and 100 for viva-voce).

   <u>Marks of Evaluation of Project</u>:

   - The I.A. Marks of Project Phase – I & II shall be sent to the University along with Project Work report at the end of the Semester.
4) During the final viva, students have to submit all the reports.
5) The Project Valuation and Viva-Voce will be conducted by a committee consisting of the following:

   a) Head of the Department (Chairman)
   b) Guide
   c) Two Examiners appointed by the university. (Out of two external examiners at least one should be present).

Advanced Mathematics

Subject Code	: 14ELD11	IA Marks	: 50
No. of Lecture Hours / Week	: 04	Exam. Hours	: 03
Total No. of Lecture Hours	: 50	Exam. Marks	: 100

Matrix Theory

QR EL Decomposition – Eigen values using shifted QR algorithm- Singular Value EL Decomposition - Pseudo inverse- Least square approximations

Calculus of Variations

Concept of Functionals- Euler’s equation – functional dependent on first and higher order derivatives – Functionals on several dependent variables – Isoperimetric problems- Variational problems with moving boundaries

Transform Methods

Laplace transform methods for one dimensional wave equation – Displacements in a string – Longitudinal vibration of a elastic bar – Fourier transform methods for one dimensional heat conduction problems in infinite and semi infinite rod.

Elliptic Equation

Laplace equation – Properties of harmonic functions – Fourier transform methods for laplace equations. Solution for Poisson equation by Fourier transforms method

Linear and Non Linear Programming

Simplex Algorithm- Two Phase and Big M techniques – Duality theory- Dual Simplex method. Non Linear Programming –Constrained extremal problems- Lagranges multiplier method- Kuhn- Tucker conditions and solutions

Reference Books:

1. Richard Bronson, **"Schaum’s Outlines of Theory and Problems of Matrix Operations"**, McGraw-Hill, 1988.
2. Venkataraman M K, **"Higher Engineering Mathematics"**, National Pub. Co, 1992.

3. Elsgolts, L., **"Differential Equations and Calculus of Variations"**, Mir, 1977.
4. Sneddon,I.N., **"Elements of Partial differential equations"**, Dover Publications, 2006.
5. Sankara Rao, K., **"Introduction to partial differential equations"**, Prentice – Hall of India, 1995
6. Taha H A, **“Operations research - An introduction"**, McMilan Publishing co, 1982.

Statistical Signal Processing

Subject Code	: 14ESP12	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Random Processes: Random variables, random processes, white noise, filtering random processes, spectral factorization, ARMA, AR and MA processes.

Signal Modeling: Least squares method, Padé approximation, Prony's method, finite data records, stochastic models, Levinson-Durbin recursion; Schur recursion; Levinson recursion.

Spectrum Estimation: Nonparametric methods, minimum-variance spectrum estimation, maximum entropy method, parametric methods, frequency estimation, principal components spectrum estimation.

Optimal and Adaptive Filtering: FIR and IIR Wiener filters, Discrete Kalman filter, FIR Adaptive filters: Steepest descent, LMS, LMS-based algorithms, adaptive recursive filters, RLS algorithm.

Array Processing: Array fundamentals, beam-forming, optimum array processing, performance considerations, adaptive beam-forming, linearly constrained minimum-variance beam-formers, side-lobe cancellers, space-time adaptive processing.

Reference Books:

1. Monson H. Hayes, **“Statistical Digital Signal Processing and Modeling”**, John Wiley & Sons (Asia) Pte. Ltd., 2002.
2. Dimitris G. Manolakis, Vinay K. Ingle, and Stephen M. Kogon, **"Statistical and Adaptive Signal Processing: Spectral Estimation, Signal Modeling, Adaptive Filtering and Array Processing”**, McGraw- Hill International Edition, 2000.
3. Bernard Widrow and Samuel D. Stearns**, "Adaptive Signal Processing”**, Pearson Education (Asia) Pte. Ltd., 2001.
4. Simon Haykin, **"Adaptive Filters”**, Pearson Education (Asia) Pte. Ltd, 4th edition, 2002.
5. J.G. Proakis, C.M. Rader, F. Ling, C.L. Nikias, M. Moonen and I.K. Proudler, **"Algorithms for Statistical Signal Processing”,**

Advances in Digital Signal Processing

Subject Code	: 14ESP13	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction and Review: Basic concepts of Digital Signal Processing, Basic digital signal processing examples in block diagram, Overview of typical Digital Signal Processing in real-world applications.

Sampling and Reconstruction of Signals: Sampling band-pass signals, Analog-to-digital and digital-toanalog conversions.

Multirate Digital Signal Processing: Introduction, Decimation by a factor D, Interpolation by a factor I, Sampling rate conversion by a rational factor I/D, Filter design and implementation for sampling rate conversion, Multistage implementation of sampling rate conversion, Sampling rate conversion of band-pass signals, Sampling rate conversion by an arbitrary factor, Applications of multirate signal processing**.**

Linear Prediction and Optimum Linear Filters: Representation of a random process, Forward and backward linear prediction, Solution of normal equations, Properties of the linear error-prediction filters, AR lattice and ARMA lattice-ladder filters, Wiener filters for filtering and prediction.

Power Spectrum Estimation: Estimation of spectra from finite-duration observations of signals, Nonparametric methods for power spectrum estimation, Parametric methods for Power Spectrum Estimation, Minimum variance spectral estimation, Eigen analysis algorithm for spectral estimation.

Hardware and Software for Digital Signal Processors: Digital signal processor architecture, Digital signal processor hardware units, Fixed-point and floating-point formats.

Reference Books:

1. John G. Proakis and Dimitris G. Manolakis, **“Digital Signal Processing”**, 3rd Edition, Pearson, 2003.
2. Li Tan, **“Digital Signal Processing – Fundamentals and applications”**, Elsevier, 2008.
3. Paulo S. R. Diniz, Eduardo A. B. da Silva And Sergio L. Netto, **“Digital Signal Processing: System Analysis and Design”**, Cambridge University Press, 2002.
4. Sanjit K. Mitra, **“Digital Signal Processing”**, A Computer Based Approach, Tata McGraw Hill, 2001.

5. Alan V.Oppenheim and Ronald W.Schafer, **“Digital Signal Processing”**, PHI Learning, 2003.

Adaptive Signal Processing

Subject Code	: 14ESP14	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Adaptive systems : definitions and characteristics - applications - properties-examples - adaptive linear combinerinput 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.

Searching performance surface-stability and rate of convergence : learning curve-gradient search - Newton's method - method of steepest descent - comparison - radient estimation - performance penalty - variance - excess MSE and time constants – mis adjustments.

LMS algorithm convergence of weight vector: LMS/Newton algorithm - properties - sequential regression algorithm - adaptive recursive filters - random-search algorithms - lattice structure - adaptive filters with orthogonal signals

Applications-adaptive modeling and system identification-adaptive modeling: multipath communication channel, geophysical exploration, FIR digital filter synthesis, inverse adaptive modeling, equalization, and deconvolutionadaptive equalization of telephone channels-adapting poles and zeros for IIR digital filter synthesis.

Reference Books:

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.

Wireless and Mobile Networks

Subject Code	: 14ECS151	IA Marks	: 50
No. of Lecture Hours / Week	: 04	Exam. Hours	: 03
Total No. of Lecture Hours	: 50	Exam. Marks	: 100

Review of fundamentals of wireless communication and Networks: Wireless communication channel specifications, Wireless communication systems, Wireless networks, Switching technology, Communication problems, Wireless network issues and standards.

Wireless body area networks: Properties, Network architectures, Components, Technologies, Design issues, Protocols and applications.

Wireless personal area networks: Architectures, Components, Requirements, Technologies and protocols, Bluetooth and Zigbee.

Wireless LANs: Network components, design requirements, Architectures, IEEE-802.11x, WLAN protocols, 802.11p and applications.

WMANs, IEEE-802.16: Architectures, Components, WiMax mobility support, Protocols, Broadband networks and applications, WWANs, cellular networks, Satellite Network, Applications.

Wireless ad-hoc networks: Mobile ad-hoc networks, Sensor network, Mesh networks, VANETs, Research issues in Wireless networks.

Reference books

1. S. S. Manvi, and M. S. Kakkasageri, **"Wireless and Mobile network concepts and Protocols"**, Wiley, 1st edition, 2010.
2. P. Kaveh, Krishnamurthy, **"Principles of Wireless network: A unified approach",** PHI, 2006.
3. Iti Saha Mitra, **"Wireless communication and network: 3G and Beyond"**, McGraw Hill, 2009.
4. Ivan Stojmenovic, **"Handbook of Wireless networks and Mobile Computing"**, Wiley, 2009.
5. P. Nicopolitidis, M. S. Obaidat, et al, **"Wireless Networks"**, Wiley, 2009.
6. Yi-Bing Lin, Imrich Chlamtac, **"Wireless and Mobile Network Architectures"**, Wiley, 2009.
7. Mullet, **"Introduction to Wireless Telecommunication Systems and Networks"**, Cengage, 2009.

Multirate Systems and Filter Banks

Subject Code	: 14ESP152	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Fundamentals of Multi-rate Systems: Basic multi-rate operations, interconnection of building blocks, poly-phase representation, multistage implementation, applications of multi-rate systems, special filters and filter banks.

Multirate Filter Banks: (i) Maximally decimated filter banks: Errors created in the QMF bank, alias-free QMF system, power symmetric QMF banks, M-channel filter banks, poly-phase representation, perfect reconstruction systems, alias-free filter banks, tree structured filter banks, trans-multiplexers. (ii) Para-unitary Perfect Reconstruction Filter Banks: Lossless transfer matrices, filter bank properties induced by paraunitariness, two channel Para-unitary lattices, M-channel FIR Para-unitary QMF banks, transform coding. (iii) Linear Phase Perfect Reconstruction QMF Banks: Necessary conditions, lattice structures for linear phase FIR PR QMF banks, formal synthesis of linear phase FIR PR QMF lattice. (iv)Cosine Modulated Filter Banks: Pseudo-QMF bank and its design, efficient poly-phase structures, properties of cosine matrices, cosine modulated perfect reconstruction systems.

Wavelet Transform: Short-time Fourier transform, Wavelet transform, discrete-time Ortho-normal wavelets, continuous time Ortho-normal wavelets.

Reference Books:

1. P.P. Vaidyanathan, **“Multirate Systems and Filter Banks"**, Pearson Education (Asia) Pte. Ltd , 2004.
2. Gilbert Strang and Truong Nguyen, **"Wavelets and Filter Banks"**, Wellesley-Cambridge Press , 1996.
3. N. J. Fliege, **"Multirate Digital Signal Processing”**, John Wiley & Sons, USA, 2000.

Speech and Audio Processing

Subject Code	: 14ESP153	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Digital Models For The Speech Signal: Process of speech production, Acoustic theory of speech production, Lossless tube models, and Digital models for speech signals.

Time Domain Models for Speech Processing: Time dependent processing of speech, Short time energy and average magnitude, Short time average zero crossing rate, Speech vs silence discrimination using energy & zero crossings, Pitch period estimation, Short time autocorrelation function, Short time average magnitude difference function, Pitch period estimation using autocorrelation function, Median smoothing.

Digital Representations of the Speech Waveform: Sampling speech signals, Instantaneous quantization, Adaptive quantization, Differential quantization, Delta Modulation, Differential PCM, Comparison of systems, direct digital code conversion.

Short Time Fourier Analysis: Linear Filtering interpretation, Filter bank summation method, Overlap addition method, Design of digital filter banks, Implementation using FFT, Spectrographic displays, Pitch detection, Analysis by synthesis, Analysis synthesis systems.

Homomorphic Speech Processing: Homomorphic systems for convolution, Complex cepstrum, Pitch detection, Formant estimation, Homomorphic vocoder. Linear Predictive Coding of Speech: Basic principles of linear predictive analysis, Solution of LPC equations, Prediction error signal, Frequency domain interpretation, Relation between the various speech parameters, Synthesis of speech from linear predictive parameters, Applications.

Speech Enhancement: Spectral subtraction & filtering, Harmonic filtering, parametric re-synthesis, Adaptive noise cancellation. Speech Synthesis: Principles of speech synthesis, Synthesizer methods, Synthesis of intonation, Speech synthesis for different speakers, Speech synthesis in other languages, Evaluation, Practical speech synthesis.

Automatic Speech Recognition: Introduction, Speech recognition vs. Speaker recognition, Signal processing and analysis methods, Pattern comparison techniques, Hidden Markov Models, Artificial Neural Networks.

Audio Processing: Auditory perception and psychoacoustics - Masking, frequency and loudness perception, spatial perception, Digital Audio, Audio Coding - High quality, low-bit-rate audio coding standards, MPEG, AC- 3, Multichannel audio - Stereo, 3D binaural and Multichannel surround sound**.**

Reference Books:

1. L. R. Rabiner and R. W. Schafer, **“Digital Processing of Speech Signals"**, Pearson Education (Asia) Pte. Ltd., 2004.
2. D. O’Shaughnessy, **“Speech Communications: Human and Machine”**, Universities Press, 2001.
3. L. R. Rabiner and B. Juang, **“Fundamentals of Speech Recognition”**, Pearson Education (Asia) Pte. Ltd., 2004. Z. Li and M.S. Drew, “Fundamentals of Multimedia”, Pearson Education (Asia) Pte. Ltd., 2004.

Modern Spectral Analysis & Estimation

Subject Code	: 14ESP154	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Basic Concepts: Introduction, Energy Spectral Density of deterministic signals, Power Spectral Density of random signals, properties of Power Spectral Densities, The Spectral Estimatuion problem, Coherence Spectrum.

Spectrum Estimation: Introduction, Correlogram method, Periodogram Computation ia FFT, properties of Periodogram method such as bias analysis, window design considerations. Signals with Rational spectra. ARMA state – space Equation, sub space Parameter Estimation.

Parametric Methods for line Spectra: Models of sinusoidal Signals in Noise, Non-linear least squares method. High Order Yule Walker method, Min – Norm Method, ESPRIT Method, Forward – Backward Estimatuion**.**

Filter Bank Method: Filter bank Interpretation of the period gram, Refined Filter bank Method, Capon Method, Filter Bank Reinterpretation of the periodogram.

Optimum Linear Filter : Optimum Signal Estimation, Linear MSE Estimation, Solution of the normal equations optimum FIR and IIR filters. Inverse filtering and deconvolution.

REFERENCE BOOKS:

1. Stoica and Moses, **"Introduction to Spectral Analysis"**, PHI, 1997.

2. Monalakis, Ingle and Kogen, **"Stastical and Adaptive Signal Procedssing"**, Tata McGraw Hill. 2000.

Simulation, Modelling and Analysis

Subject Code	: 14ELD155	IA Marks	: 50
No. of Lecture Hours / Week	: 04	Exam. Hours	: 03
Total No. of Lecture Hours	: 50	Exam. Marks	: 100

Basic simulation modeling: Nature of simulation, System models, Discrete event simulation, Single server simulation, Alternative approaches, Other types of simulation.

Building valid, credible and detailed simulation models: techniques for increasing model validity and credibility, comparing real world observations.

Selecting input probability distributions: Useful probability distributions, Assessing sample independence, Activity-I, II and III, Model of arrival process.

Random number generators: Linear congruential, Other kinds, Testing number generators, Random variate generation: Approaches, Continuous random variates, Discrete random variates, Correlated random variates.

Output data analysis: Statistical analysis for terminating simulation, Analysis for steady state parameters, Comparing alternative system configuration, Confidence interval, Variance reduction techniques, Arithmetic and control variates.

Reference books:

1. Averill Law, **"Simulation modeling and analysis"**, McGraw Hill 4th edition, 2007.
2. Jerry Banks, **"Discrete event system Simulation"**, Pearson, 2009.
3. Seila Ceric and Tadikamalla, **"Applied simulation modeling"**, Cengage, 2009.
4. George. S. Fishman, **"Discrete event simulation"**, Springer, 2001.
5. Frank L. Severance, **"System modeling and simulation"**, Wiley, 2009.

Signal Processing Lab

Subject Code	: 14ESP16	IA Marks	: 25
No. of Lecture Hours /week	: 03	Exam Hours	: 03
Total no. of Lecture Hours	: 42	Exam Marks	: 50

1. Generate various fundamental discrete time signals.
2. Basic operations on signals (Multiplication, Folding, Scaling).
3. Find out the DFT & IDFT of a given sequence without using inbuilt instructions.
4. Interpolation & decimation of a given sequence.
5. Generation of DTMF (Dual Tone Multiple Frequency) signals.
6. Estimate the PSD of a noisy signal using periodogram and modified periodogram.
7. Estimation of power spectrum using bartlett and welch methods.
8. Estimation of power spectrum using blackman-tukey method.
9. Estimation of power spectrum using parametric methods (yule-walker & burg).
10. Design of LPC filter using levinson-durbin algorithm.

Digital Signal Compression

Subject Code	: 14ESP21	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: Compression techniques, Modeling & coding, Distortion criteria, Differential Entropy, Rate Distortion Theory, Vector Spaces, Information theory, Models for sources, Coding – uniquely decodable codes, Prefix codes, Kraft McMillan Inequality

Quantization: Quantization problem, Uniform Quantizer, Adaptive Quantization, Non-uniform Quantization; Entropy coded Quantization, Vector Quantization, LBG algorithm, Tree structured VQ, Structured VQ, Variations of VQ – Gain shape VQ, Mean removed VQ, Classified VQ, Multistage VQ, Adaptive VQ, Trellis coded quantization

Differential Encoding: Basic algorithm, Prediction in DPCM, Adaptive DPCM, Delta Modulation, Speech coding – G.726, Image coding.

Transform Coding: Transforms – KLT, DCT, DST, DWHT; Quantization and coding of transform coefficients, Application to Image compression – JPEG, Application to audio compression.

Sub-band Coding: Filters, Sub-band coding algorithm, Design of filter banks, Perfect reconstruction using two channel filter banks, M-band QMF filter banks, Poly-phase decomposition, Bit allocation, Speech coding – G.722, Audio coding – MPEG audio, Image compression.

Wavelet Based Compression: Wavelets, Multiresolution analysis & scaling function, Implementation using filters, Image compression – EZW, SPIHT, JPEG 2000.

Analysis/Synthesis Schemes: Speech compression – LPC-10, CELP, MELP, Image Compression – Fractal compression**.**

Video Compression: Motion compensation, Video signal representation, Algorithms for video conferencing & videophones – H.261, H. 263, Asymmetric applications – MPEG 1, MPEG 2, MPEG 4, MPEG 7, Packet video.

Lossless Coding: Huffman coding, Adaptive Huffman coding, Golomb codes, Rice codes, Tunstall codes, Applications of Huffman coding, Arithmetic coding, Algorithm implementation, Applications of Arithmetic coding, Dictionary techniques – LZ77, LZ78, Applications of LZ78 – JBIG, JBIG2, Predictive coding – Prediction with partial match, Burrows Wheeler Transform, Applications – CALIC, JPEG-LS, Facsimile coding – T.4, T.6.

Reference Books:

1. K. Sayood, **“Introduction to Data Compression”**, Harcourt India Pvt. Ltd. & Morgan Kaufmann Publishers, 1996.

2. N. Jayant and P. Noll, **“Digital Coding of Waveforms: Principles and Applications to Speech and Video”**, Prentice Hall, USA, 1984.

3. D. Salomon, **“Data Compression: The Complete Reference”**, Springer, 2000.

4. Z. Li and M.S. Drew, **“Fundamentals of Multimedia”**, Pearson Education (Asia) Pte. Ltd., 2004.

DSP System Design

Subject Code	: 14ESP22	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction to a popular DSP from Texas Instruments: CPU Architecture - CPU Data Paths and Control - Timers - Internal Data/Program Memory - External Memory Interface - Programming - Instructions Set and Addressing Modes - Code Composer Studio - Code Generation Tools - Code Composer Studio Debug tools – Simulator

Sharc Digital Signal Processor: A popular DSP from Analog Devices - Sharc/ Tiger Sharc/ Blackfin (one of them) - Architecture - IOP Registers - Peripherals - Synchronous Serial Port - Interrupts - Internal/External/Multiprocessor Memory Space - Multiprocessing - Host Interface - Link Ports.

Digital Signal Processing Applications: FIR and IIR Digital Filter Design, Filter Design Programs using MATLAB - Fourier Transform: DFT, FFT programs using MATLAB - Real Time Implementation : Implementation of Real Time Digital Filters using DSP - Implementation of FFT Applications using DSP – DTMF Tone Generation and Detection

Current trends: Current trend in Digital Signal Processor or DSP Controller- Architecture and their applications.

References Books:

1. Naim Dahnoun, **“Digital Signal Processing Implementation using the TMS320C6000 DSP Platform”**, 1st Edition.
2. T.J. Terrel and Lik-Kwan Shark, **“Digital Signal Processing - A Student Guide**”,1st Edition; Macmillan Press Ltd.
3. David J Defatta J, Lucas Joseph G & Hodkiss William S, **“Digital Signal Processing: A System Design Approach**”, 1st Edition, John Wiley
4. Rulf Chassaing, **“Digital Signal Processing and Application with C6713 and C6416 DSK”**, Wiley-Interscience Publication
5. Steven K Smith, Newnes, **“Digital Signal Processing-A Practical Guide for Engineers and Scientists”**, Elsevier Science
6. Rulph Chassaing, **“DSP Applications using 'C' and the TMS320C6X DSK”**, 1st EditionAndrew Bateman, Warren Yates, **“Digital Signal Processing Design”**, 1st Edition

7. John G Proakis, Dimitris G Manolakis, **"Introduction to Digital Signal Processing"**, 2nd Ed.
8. Kreig Marven & Gillian Ewers, **"A Simple approach to Digital Signal processing"**, 1st Edition, Wiely Interscience
9. James H. McClellan, Ronald, Schaffer and Mark A. Yoder, "DSP FIRST" - A Multimedia App

Image and Video Processing

Subject Code	: 14ESP23	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: 2D systems, Mathematical preliminaries – Fourier Transform, Z Transform, Optical & Modulation transfer function, Matrix theory, Random signals, Discrete Random fields, Spectral density function.

Image Perception: Light, Luminance, Brightness, Contrast, MTF of the visual system, Visibility function, Monochrome vision models, Fidelity criteria, Color representation, Chromaticity diagram, Color coordinate systems, Color difference measures, Color vision model, Temporal properties of vision.

Image Sampling and Quantization: Introduction, 2D sampling theory, Limitations in sampling & reconstruction, Quantization, Optimal quantizer, Compander, Visual quantization.

Image Transforms: Introduction, 2D orthogonal & unitary transforms, Properties of unitary transforms, DFT, DCT, DST, Hadamard, Haar, Slant, KLT, SVD transform.

Image Representation by Stochastic Models: Introduction, one-dimensional Causal models, AR models, Non-causal representations, linear prediction in two dimensions.

Image Enhancement: Point operations, Histogram modeling, spatial operations, Transform operations, Multispectral image enhancement, false color and Pseudo-color, Color Image enhancement.

Image Filtering & Restoration: Image observation models, Inverse & Wiener filtering, Fourier Domain filters, Smoothing splines and interpolation, Least squares filters, generalized inverse, SVD and Iterative methods, Maximum entropy restoration, Bayesian methods, Coordinate transformation & geometric correction, Blind de-convolution.

Image Analysis & Computer Vision: Spatial feature extraction, Transform features, Edge detection, Boundary Extraction, Boundary representation, Region representation, Moment representation, Structure, Shape features, Texture, Scene matching & detection, Image segmentation, Classification Techniques.

Image Reconstruction from Projections: Introduction, Radon Transform, Back projection operator, Projection theorem, Inverse Radon transform, Fourier reconstruction, Fan beam reconstruction, 3D tomography.

Image Data Compression: Introduction, Pixel coding, Predictive techniques, Transform coding, Inter-frame coding, coding of two tone images, Image compression standards**.**

Video Processing: Fundamental Concepts in Video – Types of video signals, Analog video, Digital video, Color models in video, Video Compression Techniques – Motion compensation, Search for motion vectors, H.261, H.263, MPEG I, MPEG 2, MPEG 4, MPEG 7 and beyond, Content based video indexing.

Reference Books:

1. K. Jain, **"Fundamentals of Digital Image Processing"**, Pearson Education (Asia) Pte. Ltd./Prentice Hall of India, 2004.
2. Z. Li and M.S. Drew, **"Fundamentals of Multimedia"**, Pearson Education (Asia) Pte. Ltd., 2004.
3. R. C. Gonzalez and R. E. Woods, **"Digital Image Processing"**, 2nd edition, Pearson Education (Asia) Pte. Ltd/Prentice Hall of India, 2004.
4. M. Tekalp, **"Digital Video Processing"**, Prentice Hall, USA, 1995.

Biomedical Signal Processing

Subject Code	: 14ESP24	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: Genesis and significance of bioelectric potentials, ECG, EOG, EMG and their monitoring and measurement, Spectral analysis, digital and analog filtering, correlation and estimation techniques, AR / ARMA models, Adaptive Filters**.**

ECG: Pre-processing, Measurements of amplitude and time intervals, Classification, QRS detection, ST segment analysis, Baseline wander removal, wave form recognition, morphological studies and rhythm analysis, automated diagnosis based on decision theory ECT compression, Evoked potential estimation.

EEG: evoked responses, Epilepsy detection, Spike detection, Hjorth parameters, averaging techniques, removal of Artifacts by averaging and adaptive algorithms, pattern recognition of alpha, beta, theta and delta waves in EEG waves, sleep stages,

EMG: wave pattern studies, biofeedback, Zero crossings, Integrated EMG. Time frequency methods and Wavelets in Biomedical Signal Processing

Reference Books:

1. Willis J Tompkins, ED, **"Biomedical Digital Signal Processing"**, Prentice-Hall of India, 1996.
2. R E Chellis and R I Kitney, **"Biomedical Signal Processing"**, in IV parts, Medical and Biological Engg. And current computing, 1990-91.
3. Special issue on **"Biological Signal Processing"**, Proc. IEEE 1972
4. Arnon Kohen, **"Biomedical Signal Processing"**, Volumes I & II, CRC Press.
5. Metin Aray, **"Time frequency and Wavelets in Biomedical Signal Processing"**, IEEE Press, 1999.Current Published literature.

Detection and Estimation

Subject Code	: 14ESP251	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 52	Exam Marks	: 100

Classical Detection and Estimation Theory: Introduction, simple binary hypothesis tests, M Hypotheses, estimation theory, composite hypotheses, general Gaussian problem, performance bounds and approximations.

Representations of Random Processes: Introduction, orthogonal representations, random process characterization, homogenous integral equations and eigen functions, periodic processes, spectral decomposition, vector random processes.

Detection of Signals – Estimation of Signal Parameters: Introduction, detection and estimation in white Gaussian noise, detection and estimation in nonwhite Gaussian noise, signals with unwanted parameters, multiple channels and multiple parameter estimation.

Estimation of Continuous Waveforms: Introduction, derivation of estimator equations, lower bound on the meansquare estimation error, multidimensional waveform estimation, nonrandom waveform estimation.

Linear Estimation: Properties of optimum processors, realizable linear filters, Kalman-Bucy filters, fundamental role of optimum linear filters.

Reference Books:

1. Harry L. Van Trees, **“Detection, Estimation, and Modulation Theory”**, Part I, John Wiley & Sons, USA, 2001.
2. M.D. Srinath, P.K. Rajasekaran and R. Viswanathan, **"Introduction to Statistical Signal Processing with Applications”**, Pearson Education (Asia) Pte. Ltd. /Prentice Hall of India, 2003.
3. Steven M. Kay, **"Fundamentals of Statistical Signal Processing," Volume I: "Estimation Theory"**, Prentice Hall, USA, 1998;
4. Steven M. Kay, **"Fundamentals of Statistical Signal Processing", Volume II: "Detection Theory,"** Prentice Hall, USA, 1998.
5. K Sam Shanmugam, Arthur M Breipohl, **“Random Signals: Detection, Estimation and Data Analysis”**, John Wiley & Sons, 1998

Advanced Digital Communication

Subject Code	: 14ECS252	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Digital Modulation Techniques: QPSK, DPSK, FQPSK, QAM, M-QAM, OFDM, Optimum Receiver for Signals Corrupted by AWGN, Performance of the Optimum Receiver for Memory-less Modulation, Optimum Receiver for CPM Signals, Optimum Receiver for Signals with Random Phase in AWGN Channel.

Coding Techniques: Convolutional Codes, Hamming Distance Measures for Convolutional Codes; Various Good Codes, Maximum Likelihood Decoding of Convolutional codes, Error Probability with Maximum Likelihood Decoding of Convolutional Codes, Sequential Decoding and Feedback Decoding, Trellis Coding with Expanded Signal Sets for Band-limited Channels, Viterbi decoding**.**

Communication through band limited linear filter channels: Optimum receiver for channels with ISI and AWGN, Linear equalization, Decision-feedback equalization, reduced complexity ML detectors, Iterative equalization and decoding-Turbo equalization.

Adaptive Equalization: Adaptive linear equalizer, adaptive decision feedback equalizer, adaptive equalization of Trellis- coded signals, Recursive least squares algorithms for adaptive equalization, self recovering (blind) equalization.

Spread Spectrum Signals for Digital Communication: Model of Spread Spectrum Digital Communication System, Direct Sequence Spread Spectrum Signals, Frequency-Hopped Spread Spectrum Signals, CDMA, time-hopping SS, Synchronization of SS systems.

Digital Communication Through Fading Multi-Path Channels: Characterization of fading multi-path channels, the effect of signal characteristics on the choice of a channel model, frequency-Nonselective, slowly fading channel, diversity techniques for fading multi-path channels, Digital signal over a frequency-selective, slowly fading channel, coded wave forms for fading channels, multiple antenna systems.

Reference Books:

1. John G. Proakis, **“Digital Communications”**, 4th edition, McGraw Hill, 2001.
2. Stephen G. Wilson, **“Digital Modulation and Coding”**, Pearson Education (Asia) Pte. Ltd, 2003.
3. Kamilo Feher, **“Wireless Digital Communications: Modulation and Spread Spectrum Applications”,** Prentice-Hall of India, 2004.
4. Andrew J. Viterbi, **“CDMA: Principles of Spread Spectrum Communications”,** Prentice Hall, USA, 1995.

Pattern Recognition

Subject Code	: 14ESP253	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: Applications of pattern recognition, statistical decision theory, image processing and analysis.

Probability: Introduction, probability of events, random variables, Joint distributions and densities, moments of random variables, estimation of parameters from samples, minimum risk estimators

Statistical Decision Making: Introduction, Baye's Theorem, multiple features, conditionally independent features, decision boundaries, unequal costs of error, estimation of error rates, the leaving-one—out technique. Characteristic curves, estimating the composition of populations.

Nonparametric Decision Making: Introduction, histograms, Kernel and window estimators, nearest neighbor classification techniques, adaptive decision boundaries, adaptive discriminate Functions, minimum squared error discriminate functions, choosing a decision making technique.

Clustering: Introduction, hierarchical clustering, partitional clustering

Artificial Neural Networks: Introduction, nets without hidden layers. nets with hidden layers, the back Propagation algorithms, Hopfield nets, an application.

Processing of Waveforms and Images: Introduction, gray level sealing transfoniiations, equalization, geometric image and interpolation, Smoothing, transformations, edge detection, Laplacian and sharpening operators, line detection and template matching, logarithmic gray level sealing, the statistical significance of image features.

Reference Books:

1. Eart Gose, Richard Johnsonburg and Steve Joust, **"Pattern Recognition and Image Analysis"**, Prentice-Hall of India-2003.
2. Duda and Hart, **"Pattern recognition (Pattern recognition a scene analysis)".**
3. Robert J Schalkoff, **"Pattern recognition: Statistical, Structural and neural approaches"**, John Wiley.

Advances in Digital Image Processing

Subject Code	: 14ESP254	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: Origins of Digital Image Processing, examples, Fundamental Steps in Digital Image Processing, Components of an Image Processing System, Image analysis and computer vision, spatial feature extraction, transform features, Edge detection, gradient operators, compass operators, stochastic gradients, line and spot detection.

Digital Image Fundamentals: Elements of Visual Perception, A Simple Image Formation Model, Basic Concepts in Sampling and Quantization, Representing Digital Images, Zooming and Shrinking Digital Images, Some Basic Relationships Between Pixels, Linear and Nonlinear Operations.

Image Enhancement in the Spatial Domain: Some Basic Gray Level Transformations, Histogram Processing, Enhancement Using Arithmetic/Logic Operations, Basics of Spatial Filtering, Smoothing Spatial Filters, Sharpening Spatial Filters, Combining Spatial Enhancement Methods.

Image Enhancement in the Frequency Domain: Background, Image Enhancement in the Frequency Domain, Introduction to the Fourier Transform and the Frequency, Domain, Smoothing Frequency-Domain Filters, Sharpening Frequency Domain Filters, Homomorphic Filtering.

Image Restoration: A Model of the Image degradation/Restoration process, Noise Models, Restoration in the Presence of Noise Only–Spatial Filtering, Periodic Noise Reduction by Frequency Domain Filtering, Linear, Position-Invariant Degradations , Estimating the Degradation Function, Inverse Filtering ,Minimum Mean Square Error (Wiener) Filtering.

Color Fundamentals: Color Models, Pseudo color Image Processing, Basics of Full-Color Image Processing, Color Transformations, Smoothing and Sharpening, Color Segmentation, Noise in Color Images, Color Image Compression.

Image Transformation: Discrete Cosine Transforms, Walsh Hadmard Transforms, Wavelet Transforms and Multiprocessing, Background, Multiresolution Expansions, Wavelet Transforms in one Dimension, Wavelet Transforms in Two Dimensions, Wavelet Packets, an overview of Second Generation Wavelet Transforms.

Image and Video Compression: Fundamentals, Image Compression Models, Lossless compression Methods: Huffman coding, run length coding, LZ coding, Arithmetic coding, Lossy Compression: Gray level Run length coding, Block truncation coding, vector quantization, Differential predictive coding, Transform coding , Hybrid coding, Video Compression Techniques – Motion compensation, Search for motion vectors, H.261, H.263, MPEG I, MPEG 2, MPEG 4, MPEG 7 .

Morphological Image Processing: Preliminaries, Dilation and Erosion, Opening and Closing, The Hit-or- Miss Transformation, Some Basic Morphological Algorithms. Image Segmentation and Object Recognition: Detection of Discontinuities, Edge Linking and Boundary Detection, Thresholding, Region-Based Segmentation, Patterns and Pattern Classes, Recognition Based on Decision-Theoretic Methods, Structural Methods.

Reference Books:

1. Rafel C Gonzalez and Richard E. Woods, **“Digital Image Processing”**, 3rd Edition, Pearson Education, 2003.
2. Scott.E.Umbaugh, **“Computer Vision and Image Processing”**, Prentice Hall, 1997.

A. K. Jain, **“Fundamentals of Digital Image Processing”**, Pearson, 2004.

3. Z. Li and M.S. Drew, **“Fundamentals of Multimedia”**, Pearson, 2004.
4. S.Jayaraman, S.Esakkirajan, T.Veerakumar, **“Digital Image Procesing”**, TataMcGraw Hill, 2004.

Error Control Coding

Subject Code	: 14ESP255	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction to Algebra: Groups, Fields, Binary Field Arithmetic, Construction of Galois Field GF (2m) and its basic properties, Computation using Galois Field GF (2m) Arithmetic, Vector spaces and Matrices**.**

Linear Block Codes: Generator and Parity check Matrices, Encoding circuits, Syndrome and Error Detection, Minimum Distance Considerations, Error detecting and Error correcting capabilities, Standard array and Syndrome decoding, Decoding circuits, Hamming Codes, Reed – Muller codes, The (24, 12) Golay code, Product codes and Interleaved codes.

Cyclic Codes: Introduction, Generator and Parity check Polynomials, Encoding using Multiplication circuits, Systematic Cyclic codes – Encoding using Feed back shift register circuits, Generator matrix for Cyclic codes, Syndrome computation and Error detection, Meggitt decoder, Error trapping decoding, Cyclic Hamming codes, The (23, 12) Golay code, Shortened cyclic codes.

BCH Codes: Binary primitive BCH codes, Decoding procedures, Implementation of Galois field Arithmetic, Implementation of Error correction. Non – binary BCH codes: q – ary Linear Block Codes, Primitive BCH codes over GF (q), Reed – Solomon Codes, Decoding of Non – Binary BCH and RS codes: The Berlekamp - Massey Algorithm.

Majority Logic Decodable Codes: One – Step Majority logic decoding, one – step Majority logic decodable Codes, Two – step Majority logic decoding, Multiple – step Majority logic decoding.

Convolutional Codes: Encoding of Convolutional codes, Structural properties, Distance properties, Viterbi Decoding Algorithm for decoding, Soft – output Viterbi Algorithm, Stack and Fano sequential decoding Algorithms, Majority logic decoding

Concatenated Codes & Turbo Codes: Single level Concatenated codes, Multilevel Concatenated codes, Soft decision Multistage decoding, Concatenated coding schemes with Convolutional Inner codes, Introduction to Turbo coding and their distance properties, Design of Turbo codes. Burst – Error – Correcting Codes: Burst and Random error correcting codes, Concept of Inter – leaving, cyclic codes for Burst Error correction – Fire codes, Convolutional codes for Burst Error correction.

Reference Books:

1. Shu Lin & Daniel J. Costello, Jr. **“Error Control Coding”**, Pearson / Prentice Hall, Second Edition, 2004. (Major Reference).
2. Blahut, R.E. **“Theory and Practice of Error Control Codes”**, Addison Wesley, 1984.

Image Processing Lab

Subject Code	: 14ESP26	IA Marks	: 25
No. of Lecture Hours /week	: 03	Exam Hours	: 03
Total no. of Lecture Hours	: 42	Exam Marks	: 50

1. Basics of an Image Processing (reading an image to mat lab, display pixel operations, flipping and cropping).
2. Write a program for image enhancement.
3. Write a program for image compression
4. Write a program for color image processing
5. Write a program for image segmentation
6. Write a program for image morphology
7. Image Restoration
8. Edge detection
9. Blurring 8 bit color versus monochrome
10. Object Reorganization like circles and triangles.

DSP Algorithms and Architecture

Subject Code	: 14ESP41	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

DSP Algorithm Design: 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: Fast filtering algorithms (Winograd's, FFT, short- length FIR), re-timing 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 non ideal characteristics of analog functional blocks on the system performance**.**

DSP Module Synthesis: 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: 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.

Reference Books:

1. Sen M.Kuo , Woon-Seng S. Gan, **"Digal Signal Processors: Architectures, Implementations, and Applications"**, Prentice Hall 2004.
2. Keshab K. Parhi, **"VLSI Signal Processing Systems, Design and Implementation",** John Wiley & Sons,1999.
3. Uwe Meyer-Baese, **"Digital Signal Processing with Field Programmable Gate Array"**, Springer- Verlag 2001
4. John G. Proakis , Dimitris Manolakis K, **"DSP Principles, Algorithms and Applications"**, Prentice Hall 1995
5. Pirsch, **"Architectures for Digital Signal Processing"**, John Wiley and Sons, 1998.

6. Lars Wanhammar, **“DSP Integrated Circuits”**, Academic Press, 1999
7. Parhami, Behrooz, Computer Arithmetic, **“Algorithms and Hardware Designs”**, Oxford University Press, 2000
8. Israel Koren, A. K. Peters, Natick, **“Computer Arithmetic Algorithms”**, MA, 2002,Internal continuous

RF MEMS

Subject Code	: 14ECS421	IA Marks	: 50
No. of Lecture Hours / Week	: 04	Exam. Hours	: 03
Total No. of Lecture Hours	: 50	Exam. Marks	: 100

Review: Introduction to MEMS: Fabrication for MEMS transducers and actuators, Microsensing for MEMS, Materials for MEMS.

MEMES materials and fabrication techniques: Metals, Semiconductors, Thin films, Material s for polymer MEMS, Bulk machining for Silicon based MEMS, Surface machining for Silicon based MEMS, Micro stereo-lithography for polymer MEMS.

RF MEMS Switches and micro-relays: Switch parameters, Basics of switching, Switches for RF and Microwave applications, Actuation mechanisms, Micro-relays and micro-actuators, Dynamic of switch operations, MEMS switch design and design consideration, MEMS inductors and capacitors.

Micro machined RF filters and phase shifters: RF filters, Modelling of mechanical filters, Micro-mechanical filters, SAW filters - Basic, Design consideration. Bulk acoustic wave filters, Micro-machined filters for millimetre wave frequencies. Micro-machined phase shifters, Types and limitations, MEMS and Ferroelectric phase shifters, Application.

Micromachined transmission line and components: Micromachined transmission line: Losses in transmission line, coplanar lines, MicroshiECS and membrane supported lines, MicroshiECS components, Micromachined waveguides, Directional couplers and Mixers, Resonators and Filters.

Micromachined antennas: design, Fabrication and measurements, Integration and packaging for RF MEMS, Roles and types of packages, Flipchip techniques, Multichip module packaging and Wafer bonding, Reliability issues and thermal issues.

Reference books:

1. V. K. Varadan, A. Laktakia, and K. J. Vinoy, **"RF MEMS",** John Wiley, 2003 reprint.

2. J De Los Santos, **"RF MEMS circuit design",** Artech House, 2002.

3. Frank Ghenassia, **"Transaction Level Modelling with System C: TLM concepts and applications for Embedded Systems"**, Springer, 2005.

4. Luca Beninid, **"Networks on chips: Technology and Tools",** Morgan Kaufmann Publishers, 2006.

Mobile Computing

Subject Code	: 14ESP422	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Wireless and Mobile Network Architecture: Principle of Cellular Communication, Overview 1G, 2G, 2.5G and 3G and 4G technologies. GSM Architecture and Mobility management hand off management, Network signaling. Mobile Computing fundamental challenges, Mobile Devices –PDA and mobile OS, PalmOs, Win CE and Symbian.

Mobile IP Protocol Architecture: Mobile IP and IP v 6 and its application in mobile computing, Cellular Digital Packet Data CDPD, VOIP, GPRS Services, Wireless Local Loop-WLL system. Wireless Application Protocol (WAP): The Wireless Application Protocol application environment, wireless application protocol client software, hardware and websites, wireless application protocol gateways, implementing enterprise wireless application protocol strategy,

Wireless Markup Language: An Introduction to Wireless Technologies, Markup Languages , An Introduction to XML, Fundamentals of WML., Writing and Formatting Text , Navigating Between Cards and Decks, Displaying Images, Tables, Using Variables, Acquiring User Input

Wireless Markup Language Script: An Introduction to WML Script, WML Script Control Structures, Events, Phone.com Extensions, Usability

Application of Mobile Computing: ASP and Dynamic WAP Sites, XML and XSLT, Dynamic WML Generation with ASP and XSLT, Developing WAP Applications using Emulators. Distributed Mobile Computing: Distributed OS and file systems, Mobile Computing Software (Pervasive Computing) Development Strategies and tools, Data Management for Mobile Computing.

Reference Books:

1. Yi Bing Lin, **“Wireless and Mobile Networks Architecture”**, John Wiley.
2. Wrox **“The Beginning WML and WML Script”**, Wrox Publication.
3. Tomasz Imielinski et.al, **“Mobile Computing”**, Kluwer Academic Press 1996.
4. Uwe Hansmann, **“Pervasive Computing Handbook. The Mobile World”**, IEE publication 2002.

5. Jochen Burkhardt, et.al. **“Pervasive Computing, Technology and Architecture of Mobile Internet Applications”**, Addison Wesley, 2002.

Modern DSP

Subject Code	: 14ESP423	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

Introduction: Advances in Digital Signal Processing involve variable sampling rates and thus the multirate signal processing and hence their applications in communication systems and signal processing. It is intended to introduce a basic course in multirate signal processing especially meantfor students of branches eligible for M Tech courses in EC related disciplines.

Introduction and Discrete Fourier Transforms: Signals, Systems and Processing, Classification of Signals, The Concept of Frequency inContinuous-Time and Discrete-Time Signals, Analog-to-Digital and Digital-to-Analog Conversion, Frequency-Domain Sampling: The Discrete FourierTransform, Properties of the DFT, Linear Filtering Methods Based on the DFT (Ref.1 Chap. 1 & 7)

Design of Digital Filters: General Considerations, Design of FIR Filters, Design of IIR Filters from Analog Filters, Frequency Transformations. (Ref.1Chap.10)

Multirate Digital Signal Processing: Introduction, Decimation by a factor‘D’, Interpolation by a factor ‘I’, Sampling rate Conversion by a factor ‘I/D’, implementation of Sampling rate conversion, Multistage implementation of Sampling rate conversion, Sampling rate conversion of Band Pass Signals, Sampling rate conversion by an arbitrary factor, Applications of Multirate Signal Processing, Digital Filter banks, Two Channel Quadrature Mirror Filter banks, M-Channel QMF bank. (Ref.1 Chap.11)

Adaptive Filters: Applications of Adaptive Filters, Adaptive Direct Form FIR Filters- The LMS Algorithm, Adaptive Direct Form Filters-RLS Algorithm. (Ref.1 Chap.13)

REFERENCE BOOKS:

1. Proakis and Manolakis, **“Digital Signal Processing”**, Prentice Hall 1996.(fourth edition).
2. Roberto Cristi, **“Modern Digital Signal Processing”**, CengagePublishers, India, (erstwhile Thompson Publications), 2003.
3. S.K. Mitra, **“Digital Signal Processing: A Computer Based Approach”**,III Ed, Tata McGraw Hill, India, 2007.

4. E.C. Ifeachor and B W Jarvis, **“Digital Signal Processing, a practitioners approach”,** II Edition, Pearson Education, India, 2002 Reprint**.**

Communication System design using DSP algorithm

Subject Code	: 14ESP424	IA Marks	: 50
No. of Lecture Hours / Week	: 04	Exam. Hours	: 03
Total No. of Lecture Hours	: 50	Exam. Marks	: 100

Introduction to the course: Digital filters, Discrete time convolution and frequency responses, FIR filters - Using circular buffers to implement FIR filters in C and using DSP hardware, Interfacing C and assembly functions, Linear assembly code and the assembly optimizer. IIR filters - realization and implementation, FFT and power spectrum estimation: DTFT window function, DFT and IDFT, FFT, Using FFT to implement power spectrum.

Analog modulation scheme: Amplitude Modulation - Theory, generation and demodulation of AM, Spectrum of AM signal. Envelope detection and square law detection. Hilbert transform and complex envelope, DSP implementation of amplitude modulation and demodulation.

DSBSC: Theory generation of DSBSC, Demodulation, and demodulation using coherent detection and Costas loop. Implementation of DSBSC using DSP hardware.

SSB: Theory, SSB modulators, Coherent demodulator, Frequency translation, Implementation using DSP hardware.

Frequency modulation: Theory, Single tone FM, Narrow band FM, FM bandwidth, FM demodulation, Discrimination and PLL methods, Implementation using DSP hardware.

Digital Modulation scheme: PRBS, and data scramblers: Generation of PRBS, Self synchronizing data scramblers, Implementation of PRBS and data scramblers. RS-232C protocol and BER tester: The protocol, error rate for binary signalling on the Gaussian noise channels, Three bit error rate tester and implementation.

PAM and QAM: PAM theory, baseband pulse shaping and ISI, Implementation of transmit filter and interpolation filter bank. Simulation and theoretical exercises for PAM, Hardware exercises for PAM.

QAM fundamentals: Basic QAM transmitter, 2 constellation examples, QAM structures using passband shaping filters, Ideal QAM demodulation, QAM experiment. QAM receivers-Clock recovery and other frontend sub-systems. Equalizers and carrier recovery systems.

Experiment for QAM receiver frontend. Adaptive equalizer, Phase splitting, Fractionally spaced equalizer. Decision directed carrier tracking, Blind equalization, Complex cross coupled equalizer and carrier tracking experiment.

Echo cancellation for full duplex modems: Multicarrier modulation, ADSL architecture, Components of simplified ADSL transmitter, A simplified ADSL receiver, Implementing simple ADSL Transmitter and Receiver.

Reference Books:

1. Robert. O. Cristi, **"Modern Digital signal processing",** Cengage Publishers, India, 2003.
2. S. K. Mitra, **"Digital signal processing: A computer based approach",** 3rd edition, TMH, India, 2007.
3. E.C. Ifeachor, and B. W. Jarvis,**"Digital signal processing: A Practitioner's approach",** Second Edition, Pearson Education, India, 2002,
4. Proakis, and Manolakis, **"Digital signal processing"**, 3rd edition, Prentice Hall, 1996.

Multirate Signal Processing

Subject Code	: 14ESP425	IA Marks	: 50
No. of Lecture Hours /week	: 04	Exam Hours	: 03
Total no. of Lecture Hours	: 50	Exam Marks	: 100

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.

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.

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.

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

Reference Books:

1. P.P. Vaidyanathan, **"Multirate systems and filter banks"**, Prentice Hall. PTR. 1993.
2. N.J. Fliege, **"Multirate digital signal processing"** John Wiley.
3. Fredric J. Harris, **"Multirate Signal Processing for Communication Systems"**, Prentice Hall, 2004
4. Ljiljana Milic, **"Multirate Filtering for Digital Signal Processing: MATLAB Applications"**, Information Science Reference; 1/e, 2008
5. Sanjit K. Mitra, **"Digital Signal Processing: A computer based approach"**, McGraw Hill. 1998.
6. R.E. Crochiere. L. R. Rabiner, **"Multirate Digital Signal Processing"**, Prentice Hall. Inc.1983.
7. J.G. Proakis. D.G. Manolakis, **"Digital Signal Processing: Principles. Algorithms and Applications"**, 3rd Edn. Prentice Hall India, 1999