

<b>COURSE CODE</b>	<b>COURSE NAME</b>	<b>L-T-P-C</b>	<b>YEAR OF INTRODUCTION</b>
EC333	Digital Signal Processing Lab	0-0-3-1	2015
<b>Prerequisite:</b> EC 213 Electronics Design Automation Lab, EC 202 Signals & Systems			
<b>Course objectives:</b> <ul style="list-style-type: none"><li>• Enable the students to explore the concepts of design, simulation and implementation of various systems using MATLAB/SciLab/OCTAVE and DSP kit.</li></ul>			
<b>List of Experiments:</b>  <b>Part A: Experiments on Digital Signal Processor/ DSP kits: (All experiments are mandatory)</b> <ol style="list-style-type: none"><li>1. Generation of sine wave and standard test signals.</li><li>2. Convolution : Linear and Circular</li><li>3. Real Time FIR Filter implementation (Low-pass, High-pass and Band-pass) by inputting a signal from the signal generator</li><li>4. Real Time IIR Filter implementation ( Low-pass, High-pass and Band-pass) by inputting a signal from the signal generator</li><li>5. Sampling of analog signal and study of aliasing.</li></ol> <b>Part B: Experiments based on MATLAB/SciLab/OCTAVE (7 experiments are mandatory)</b> <ol style="list-style-type: none"><li>1. Generation of Waveforms (Continuous and Discrete)</li><li>2. Verification of Sampling Theorem.</li><li>3. Time and Frequency Response of LTI systems (First and second order).</li><li>4. Linear Convolution, Circular Convolution and Linear Convolution using Circular Convolution.</li><li>5. To find the DFT and IDFT for the given input sequence.</li><li>6. Linear convolution using DFT (Overlap-add and Overlap-Save methods).</li><li>7. To find the DCT and IDCT for the given input sequence.</li><li>8. To find FFT and IFFT for the given input sequence.</li><li>9. FIR and IIR filter design using Filter Design Toolbox.</li><li>10. FIR Filter (Low-pass, High-pass and Band-pass)design (Window method).</li><li>11. IIR Filter (Low-pass, High-pass and Band-pass)design (Butterworth and Chebychev).</li><li>12. Generation of AM, FM &amp; PWM waveforms and their spectrum.</li><li>13. Generation of DTMF signal.</li><li>14. Study of sampling rate conversion (Decimation, Interpolation, Rational factor).</li><li>15. Filtering of noisy signals</li><li>16. Implementation of simple algorithms in audio processing (delay, reverb, flange etc.).</li><li>17. Implementation of simple algorithms in image processing (detection, de-noising, filtering etc.)</li></ol>			
<b>Expected outcome:</b>			
The student should able to:			
Design, simulate and realize various systems related to DSP.			