EE538 Digital Signal Processing I — Fall 2014

Topic:	*Lectures*	*Reading: P&M Text*
I. Review: Discrete-Time Signals, Systems, & Transforms	21	
A. Basic Sampling Theory and D/A Conversion		1.1-1.4
B. Discrete-Time Linear Systems (DT Convolution)		2.1-2.3
1. Autocorrelation, Cross-Correlation (VIP)		2.4.1, 2.4.2, 2.5, 2.6
C. Z Transform		3.1-3.5
		3.5
D. Discrete-Time Fourier Transform		4.1-4.5
E. Frequency Selective Linear Filtering		5.1-5.4
F. Sampling and Reconstruction		6.1-6.6
G. Multirate DSP ****most emphasis* ***		11.2-11.4
1. Efficient Up-sampling/Down-sampling		11.5
2. Multi-Stage Interpolation		11.6
3. Digital Subbanding		11.9
4. Two-Channel Perfect Reconstruction Filter Bank		11.11
5. M-Channel Perfect Reconstruction Filter Bank		11.11
I. Apps: CD Players, Radar, Wireless Comm: CDMA, OFDM		11.9
II. Digital Filter Design	6	
A. FIR Filters – Equiripple Designs		10.2.4-10.2.6
B. IIR Filters		
1. Common analog filters		10.3.5
2. Bilinear transformation		10.3.3
3. Frequency transformations		10.4
III. Discrete Fourier Transform	6	
A. Definition and Properties		7.1-7.4
B. Fast Fourier Transform Algorithms		
1. Divide and Conquer Approach		8.1.1, 8.1.2
2. Radix-2 FFT		8.1.3
C. Sectioned Convolution: Overlap Add/Overlap Save		7.3, 8.2-8.3
IV. Nonparametric methods of power spectrum estimation	2	
A. Discrete random processes		12.1-12.2
B. Estimation of autocorrelation sequence		14.1.2
C. Periodogram; Smoothed periodograms		14.2
V. Model-Based Spectrum Estimation	6	
A. Autoregressive (AR) Modelling		14.3
B. Forward/Backward Linear Prediction		12.3
C. Levinson-Durbin Algorithm		12.4
D. Minimum Variance Method		14.4
E. Eigenstructure Methods I: MUSIC		14.5.2,14.5.3
F. Eigenstructure Methods II: ESPRIT		14.5.1,14.5.4
G. Applications in Speech Processing, Communications, and Acoustics		