SYLLABUS

ECE 429 / 529 Digital Signal Processing

SPRING 2009

I. Introduction DSP is concerned with the digital representation of signals and the use of microprocessors and computers to analyze, modify, and extract information from signals. The digital signals found in most popular applications of DSP are derived from analog signals that have been sampled at regular intervals and converted into digital form.

DSP is used to:

- remove interference or noise from signals
- detect faint signals in noise, recognize known signals
- enhance recorded music
- obtain and analyze the spectrum of a signal
- compress a signal into fewer bits per sample
- model analog systems
- etc.

Some common applications are speech synthesis and recognition, telephone echo cancellation, and digital audio (CD) technology. DSP (and digital technology in general) is growing in importance with each passing year. Increasingly, DSP is considered to be a core subject in electrical engineering and computer engineering curricula.

The first part of this course covers the fundamentals of discrete-time signals and systems. We will study key DSP operations such as convolution, filtering, and discrete Fourier transforms. Even during this early stage we will practice some applications of the theory covered in class. Then we will progress to digital filter design and spectral analysis, which are the two major branches of DSP. The work involved will be both interesting and demanding.

Instructor, Office: Dr. Robin N. Strickland, ECE

Phone: TEL: 621-6191 FAX: 621-8076

E-mail: **strickland@ece.arizona.edu**

Web: **www.ece.arizona.edu/~429rns**

Office hours: See web page

Text: None required – a comprehensive set of notes will be used. If you wish to purchase a reference text, I recommend *DIGITAL SIGNAL PROCESSING Principles, Algorithms, and Applications*, Fourth Edition, John G. Proakis and Dimitris G. Manolakis, Prentice-Hall, 2007. (ISBN: 0131873741)

Notes: A complete set of (required) course notes is available for purchase in Harvill 137. Bring these to class.

II. Administrative Details and Policies

Prerequisites: ECE 340 *Signals and Linear Systems* (Grade C or higher recommended) and MATH 322 *Math Analysis for Engineers* (or equivalents). It is not acceptable to be *currently* enrolled in either of these courses. ECE 340 covers Fourier Series and Fourier Transforms, which are employed extensively in ECE 429/529. You will be hopelessly lost in this course if you have not acquired a sound understanding of Fourier principles.

Attendance and punctuality: I teach directly to specific course learning outcomes (printed later in this document and in the course notes), so simply skipping class and reading a text is unlikely to lead to success in this course. Excessive or extended absences from class are sufficient reason for me to recommend that a student be administratively dropped from the course. Missing the first class session may be interpreted as excessive absence. If this action is filed in the Registrar's Office by the end of the fourth week of classes, it will result in cancellation of registration in the course. If the student is administratively dropped after the end of the fourth week of classes, it will result in a failing grade being awarded in that course

Participation: Students are expected to take part in general class discussions and any group activities. E-mail communication is essential.

Homework: A few problems will be assigned weekly, and collected the following week at the start of class. (Please staple papers; don't fold.) Some or all homework answers will be posted prior to the due date. All or a subset of each homework set will be graded. One late paper is acceptable, but even then a 50% deduction will be applied. Condensed solutions will be posted a few days after the due date. If you find a grading error, write a note on your paper explaining the problem and I will forward it to the grader. It may take some time to resolve the problem.

Exams: There will be two mid-term exams and one final examination. All exams will be closed book. All necessary tables and major formulae will be provided on the exam paper. The exam topics will be a subset of the course learning outcomes. Any suspected grading errors must be reported no later than the next class meeting after the exam is returned to you. A make-up exam will be given only in the case of illness and medical emergency (doctor's note required), and only if I am notified in advance of the exam by telephone.

Matlab assignments: There will be three Matlab Projects, as well as Matlab problems sprinkled throughout the regular homeworks. As the semester progresses, these will grow in scope in accordance with a realistic Matlab learning curve.

Grading policy: Graded work will include exams, Matlab Projects (with various report formats), and homework problems. Your total number of points compared to an absolute scale will determine your final grade, i.e. no curving of grades. (The number or % of A's, B's, *etc.* is not fixed.) The following weights will be used to determine your point total:

		ECE 429	ECE 529
Exam 1	(50 min.)	15%	15%
Exam 2	(50 min.)	15%	15%
Final Exam	(2 hours)	30%	30%
Homework	(Weekly)	15%	15%
Matlab Projects[1]		25%	25%

A term point total of 90 or above is guaranteed an A, 80 or above at least a B, 70 or above at least a C, 60 or above at least a D.

Withdrawals:[2] You may withdraw without the permission of the instructor up to the end of the 4th week of class. From the end of the 4th week until the end of the 8th week you may withdraw with the permission of the instructor, which will be given **only if you are passing the course at the time**. Withdrawals after the 8th week also require the Dean's approval. Note that students wishing to drop the course AT ANY TIME must take appropriate action. Ceasing attendance

[1] Extra credit – amount to be specified on the assignment - will be awarded for exceptional work.
[2] See http://catalog.arizona.edu/2008-09/0809cal.html

does not automatically drop you from the course. IF YOU ARE STILL ON THE CLASS ROLL AT THE END OF THE SEMESTER, YOU WILL RECEIVE 0'S FOR ANY WORK NOT COMPLETED AND WILL BE GRADED ACCORDINGLY.

Academic Integrity: The University's Code of Academic Integrity (Section 2.1a) states that students shall not "represent the work of others as their own." This policy will be applied to all work submitted for a grade: exams, homework, computer work, and reports. *Any student submitting homework solutions or computer project reports with part(s) copied from solutions provided by any instructor(s) in previous semesters, or from any text solutions manual, or from students who took the course in previous semesters, will automatically receive zero credit for ALL homework/computer work for the entire semester.* In other words, all work must be original. The minimum penalty for cheating on exams is a failing grade.

ECE 529: Students enrolled in the graduate section of the course may be required to complete additional tasks on homework/Matlab Projects.

III. Goals and Learning Outcomes

Overall Educational Goals: Students completing this course are expected to have a good understanding of the fundamentals and applications of discrete-time signals and systems, including sampling, convolution, filtering, and discrete Fourier transforms. They are expected to be able to design digital filters, and perform spectral analysis on real signals using the discrete Fourier transform. They will be practiced in sampling, processing and playing back audio and other signals using Matlab software running on PCs or workstations.

Specific Learning Outcomes: By the end of the course students are expected to be able to:

1. Describe the distinctions between *analog*, *continuous-time*, *discrete-time* and *digital* signals, and describe the basic operations involved in analog-digital (A/D) and digital-analog (D/A) conversion.
2. Define simple non-periodic discrete-time sequences such as the impulse and unit step, and perform time-shifting and time-reversal operations on such sequences.
3. Show that sampling a continuous-time sinusoid $x(t) = A\cos(\Omega t)$ (where $\Omega = 2\pi F$ rad/s) produces the discrete-time sequence $x(n) = A\cos(\omega n)$ whose *digital* frequency is given by $\omega = \Omega T$ rad/sample, and whose *normalized* (or *relative*) frequency is given by $f = \frac{F}{F_s}$.
4. State the condition for a discrete-time sinusoid to be periodic.
5. Explain why the effective, unique range of digital frequency is zero to pi radians.
6. State the sampling theorem.
7. Explain *aliasing* in the sampling of sinusoids and why, for example, sampling two sinusoids of frequency 2200Hz and 1800Hz at a sampling rate of 2kHz produces 200Hz alias sinusoids (of equal phase if cosines and opposite phase if sines).
8. Explain how the digitization process produces additive *quantization* noise; state the relationship between SNR (dB) and the number of A/D converter bits; give typical sampling and bit rates for common types of signal.
9. ~~Describe the time-domain, frequency-domain and autocorrelation properties of discrete white noise (used to model quantization noise).~~
10. ~~Describe the effect of averaging consecutive samples of white noise on the noise power level.~~
11. Explain the operation of a one-sample delay operator and hence write the linear difference equation and draw the system block diagram for simple first-order FIR and IIR filters.
12. Apply simple sequences (impulse, step, and sinusoid) to the input of such filters and hand-calculate the filter output given either the system block diagram or the linear difference equation.

13. Given the difference equation of a discrete-time system, be able to apply tests (or examples and counter examples) to demonstrate linearity, time-invariance, causality and stability, and hence show whether or not a given system belongs to the important class of causal, LTI (linear time-invariant) systems.
14. Given (or calculate by hand from the difference equation) the impulse response of a causal LTI system, show whether or not the system is bounded-input/bounded-output (BIBO) stable.
15. From a linear difference equation of a causal LTI system, draw the Direct Form I and Direct Form II filter realizations.
16. Find the impulse response of a digital LTI filter from its linear difference equation: from the sequence of coefficients in the case of FIR systems, and using the homogeneous solution method for IIR systems.
17. Find the step response a(n) by summing the impulse response: $a(n) = \sum_{k=-\infty}^{n} h(k)$
18. Perform tabular calculations to convolve two finite-length sequences.
19. Use the series summation $\sum_{n=N_1}^{N_2} \alpha^n = \frac{\alpha^{N_1} - \alpha^{N_2+1}}{1-\alpha}$ to compute the closed-form result of convolution between two sequences where one or both sequences is of infinite length.
20. Perform tabular calculations to find the cross-correlation between two sequences (or the autocorrelation of a single sequence) and be able to estimate the delay from the cross-correlation in a simple radar application.
21. Calculate the z-transform X(z) of a simple sequence x(n) (such as exponentials and sinusoids): specify the region of convergence (ROC) and the bounding poles of X(z).
22. Derive the time-shifting property of the z-transform, i.e. $x(n-m) \Leftrightarrow z^{-m}X(z)$.
23. Given a z-transform X(z) and its ROC, state whether or not the DtFt of x(n) exists, and predict whether the sequence x(n) is left-sided, right-sided, two-sided, and/or of finite duration.
24. Find the time sequence x(n) given X(z) and the ROC, using power series expansion and partial fraction expansion.
25. Apply z-transform properties and theorems, notably convolution, time reversal, and multiplication by an exponential sequence (plus time-shifting property listed earlier).
26. Given the transfer function H(z) and ROC of an LTI system, find the system poles (and zeros) and state whether or not the system is BIBO stable.
27. Apply z-transforms to find the (output) response of a digital filter when the input sequence and the system impulse response or transfer function are given. (Assuming initial conditions are zero.)
28. Compute the Discrete-time Fourier transform (DtFt) of a simple sequence such as the impulse response of an FIR filter, and explain how the DtFt and z-transform are linked by the unit circle whose locus is $z = e^{j\omega}$.
29. Derive and apply the time-shifting property of the DtFt, i.e. $x(n-m) \Leftrightarrow e^{-jm\omega}X(\omega)$.
30. Explain the form of symmetry of the DtFt (magnitude & phase components) for real and complex time sequences.
31. Derive the frequency-shifting property of the DtFt, i.e. $x(n)e^{j\omega_o n} \Leftrightarrow X(\omega - \omega_o)$, and apply the special case $x(n)(-1)^n \Leftrightarrow X(\omega - \pi)$.
32. Derive the modulation property of the DtFt, i.e. $x(n)\cos(\omega_o n) \Leftrightarrow 0.5X(\omega + \omega_o) + 0.5X(\omega - \omega_o)$.
33. Apply the inverse DtFt to design ideal lowpass, highpass, bandpass and bandstop filters, and discuss how to implement approximations of these using digital FIR filters.
34. Apply prototype FIR filters in combination with multiplying sequences by $(-1)^n$ to synthesize other types of filter.
35. Apply DtFt theorems to compute the spectra of sampled, compressed, and expanded sequences.
36. Show that $e^{j\omega n}$ is an eigenfunction of a discrete-time LTI system, which when applied to the system input, produces an output equal to $e^{j\omega n}$ scaled by the quantity $H(\omega) = \sum_{k=-\infty}^{\infty} h(k)e^{-j\omega k}$.

37. With the input $x(n) = e^{j\omega n}$, show that the forced response $y(n) = H(\omega)e^{j\omega n}$ is a solution of the linear difference equation, and hence derive: $H(\omega) = \dfrac{\sum_{k=0}^{M} b_k e^{-j\omega k}}{1 + \sum_{k=1}^{N} a_k e^{-j\omega k}}$.
38. Show that the output produced by the input $x(n) = \cos(\omega n)\ \forall n$ is the sinusoid given by: $y(n) = |H(\omega)| \cos[\omega n + \angle H(\omega)]\ \forall n$.
39. Describe the symmetry of the frequency response $H(e^{j\omega})$ for LTI filters with real coefficients.
40. Explain the relationship between linear phase $\angle H(e^{j\omega}) = -\omega n_d$ and time delay of the output sequence.
41. ~~Explain how *envelope (or group) delay* relates to~~ $\angle H(e^{j\omega})$ ~~, and how this determines the time delay as a narrowband signal is passed through an LTI system.~~
42. Knowing the poles and zeros of a transfer function, make a rough sketch of the gain response $|H(e^{j\omega})|$.
43. Design simple digital filters such as comb and notch filters to remove unwanted interference and noise.
44. Convert digital filters into minimum-phase filters and explain why this is advantageous.
45. Define the Discrete Fourier Transform (DFT) and the inverse DFT (IDFT) of length N, and show that both transforms are periodic with period N.
46. Show that the DFT computes a discretely sampled version of the DtFt; that the frequency resolution of the DFT depends on N; and that the region of support of the DFT spans exactly one cycle of the DtFt, i.e. once around the unit circle in the z-plane.
47. Apply DFT properties (especially time- and frequency-shifting properties, duality theorem) and symmetry relationships to compute the DFT and IDFT of simple sequences.
48. Apply the DFT to compute both circular convolution and (using zero padding) linear convolution between two finite length sequences of arbitrary length.
49. Apply the DFT to compute the correlation between two sequences.
50. Describe the symmetry of the DFT coefficients {X(k); $0 \le k \le N-1$) computed from N samples of real data.
51. Describe the form of the DFT computed for a single sinusoid, when the data record captures an integer number of cycles of the sinusoid. Repeat for the case of a non-integer number of cycles.
52. Design the parameters associated with DFT implementation (sampling rate and record length) to provide an accurate analysis of the frequency and strength of dominant frequency components present in some given, unknown signal. (Known as *spectral analysis* of a signal.)
53. Explain the need for zero padding and tapered windows when doing spectral analysis of real world signals. (Hence explain the terms *picket fence effect* and *spectral leakage*.)
54. Compensate for the reduction in spectral magnitudes when using tapered windows.
55. From a given signal flow graph, use Mason's Gain Rule to derive the transfer function H(z).
56. Design simple filter architectures (signal flow graphs) to realize given digital filter transfer functions, using *Direct Form II* structures connected in cascade or in parallel.
57. Explain the cause of *limit cycles* in the implementation of IIR filters.
58. Compare the characteristics (advantages & disadvantages) of IIR and FIR filters.
59. Discuss the four types of symmetry for the coefficients of a digital FIR filter and how this symmetry produces linear phase in the frequency response.
60. Design FIR filters that approximate an ideal differentiator system.
61. Use the windowing method to design digital lowpass, highpass and bandpass FIR filters to meet specific filtering criteria (passband width, transition band width, stopband attenuation, and linear phase).
62. Use the bilinear transform to design digital lowpass and highpass Butterworth IIR filters to satisfy given cutoff frequencies and attenuation factors.
63. Derive this expression for the spectrum of a bandlimited signal that has been ideally sampled: $X(\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} X_c\left[j\left(\frac{\omega}{T} - \frac{2\pi k}{T}\right)\right]$.
64. Draw the spectrum of an ideally sampled sinusoid, showing aliasing when appropriate.

65. Explain the stages in the D/A process from ideal samples and explain the need for an ideal analog reconstruction filter.
66. Draw block diagram systems for signal interpolation and decimation, and explain their operation.
67. Explain (using frequency domain sketches) the application of oversampling and subsequent decimation for A/D recording in digital audio systems.
68. Explain (using frequency domain sketches) the application of interpolation to D/A playback in digital audio systems.

IV. Course Schedule

See next page.

Please refer to the course web site (www.ece.arizona.edu/~429rns) where an updated version with assignments and due dates will be maintained.

SPRING 2009 All dates are tentative.

Lesson	Day	Date	Learning outcomes to be covered	Topics	HW/PROJECT (approximate dates)
1	W	Jan 14	1	Introduction to DSP	
2	F	Jan 16	2-5	Sequences, digital frequency	
3	W	Jan 21	6-7	Sampling, aliasing	HW1
4	F	Jan 23	8-10	Quantization noise	
5	M	Jan 26	11-12	DT system components	
6	W	Jan 28	13-14	System properties	HW2
7	F	Jan 30	15-17	Filter realizations, impulse response	
8	M	Feb 2	18-19	Convolution	HW3
9	W	Feb 4	20	Correlation	
10	F	Feb 6	21	(Forward) z-transform	
11	M	Feb 9	22-23	Time-shifting, DtFt existence, sequence type from ROC	HW4
12	W	Feb 11	24	(Inverse) z-transform	
13	F	Feb 13	25	Applying z-transform properties	
14	M	Feb 16	26-27	Poles & stability, system analysis using z-transform	REVIEW
15	W	Feb 18	REVIEW		REVIEW
	F	Feb 20	**EXAM 1**		REVIEW
16	M	Feb 23	28-30	(Forward) Discrete-time Fourier transform (DtFt), symmetry	PROJ 1
17	W	Feb 25	31-33	Frequency-shifting, modulation, filter design from lowpass prototypes	
18	F	Feb 27	33-34	Synthesis of filters using DtFt properties	
19	M	Mar 2	35-37	DtFt analysis of downsampling, expansion, compression operations	HW5
20	W	Mar 4	38-39	DtFt systems analysis	
21	F	Mar 6	40-41	Phase and group delay of filters	
22	M	Mar 9	42	Frequency response from poles & zeros	HW6
23	W	Mar 11	43	Comb and notch filters	
24	F	Mar 13	44	Minimum-phase filters	
25	M	Mar 23	45-46	Forward DFT and Inverse DFT, relationship to DtFt	PROJ 2
26	W	Mar 25	47	Applying DFT properties	
27	F	Mar 27	48-49	Convolution and correlation using DFT	
28	M	Mar 30	50-52	DFT symmetry, sinusoidal analysis and freq resolution	HW7
29	W	April 1	53	Zero-padding and windowing	
30	F	April 3	54	Spectral analysis	
31	M	April 6	55	Mason's gain rule	REVIEW
32	W	April 8	REVIEW		REVIEW
	F	April 10	**EXAM 2**		REVIEW
33	M	April 13	56-58	Filter architecture, filter comparisons, limit cycles	PROJ 3
34	W	April 15	59-60	Linear phase FIR filter types	
35	F	April 17	61	FIR design by windowing	HW8
36	M	April 20	61	FIR design by windowing	
37	W	April 22	62	IIR filter design using bilinear transforms	
38	F	April 24	62	IIR filter design using bilinear transforms	HW9
39	M	April 27	63-64	DtFt analysis of sampling and aliasing	
40	W	April 29	65-66	Analog signal reconstruction, decimation and interpolation	
41	F	May 1	67-68	Digital audio applications of multirate DSP	REVIEW
42	M	May 4	REVIEW		REVIEW
42	W	May 6	REVIEW		
Final Exam	Wednesday, May 13, 11:00-1:00 p.m.				

Personal Information Questionnaire

Please complete and give to Dr. Strickland in class today. Write clearly, as this data will be used to verify the class list as well as update the e-mail distribution list.

Course (circle one): **429** **529**

Last Name: ___________________________________ **First Name: ______________________**

E-mail address: __________________________________

Phone (optional): __________________

How much "Matlab experience" do you have?____________________________

(Optional) Any relevant personal information, e.g. current job, electronics experience, interests/hobbies…