1 Course Overview

1.1 Course Syllabus

Please see syllabus handout. The syllabus for EE395 Introduction to Digital Signal Processing is also available at:

http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/Syllabus.html

1.2 Course Outline

Please see outline handout. The outline for EE395 Introduction to Digital Signal Processing is also available at:

http://www.ece.nmsu.edu/~pdeleon/Teaching/EE395/Outline.html

2 Lecture Outline

Reading: Chapter 1 Sampling and Reconstruction

- A "lite" introduction to DSP

3 What is Digital Signal Processing (DSP)?

Examples: Some everyday signals

- Speech and audio (air pressure as a function of time or voltage as a function of time if speech is picked up by a microphone)
- Image (brightness function of spatial variables, x and y) and video (brightness function of spatial variables, x and y and time, t)
- HDTV, WiFi signals (wireless digital communications)
- radar signals, CAT/MRI imaging
- temperature data, stock market data

Example: A naturally occurring signal fundamental in engineering, mathematics, physics, and music is the sinusoid or tone

$$x(t) = \sin(\Omega t) = \sin(2\pi f t) \tag{1}$$

where Ω is in radians/s, f is in Hz, t is in seconds, and $x(t)$ is in volts. As an example, a sinusoid with $f = 500$ Hz is shown in Fig. 1. Note this function is continuous in t.

Definition: A signal is a *function that conveys information*, generally about the state or behavior of a physical system. Mathematically a signal is a function of one or more independent variables.

Examples: Signal Processing

- Increasing the bass of your music (low pass filter gain adjustment)

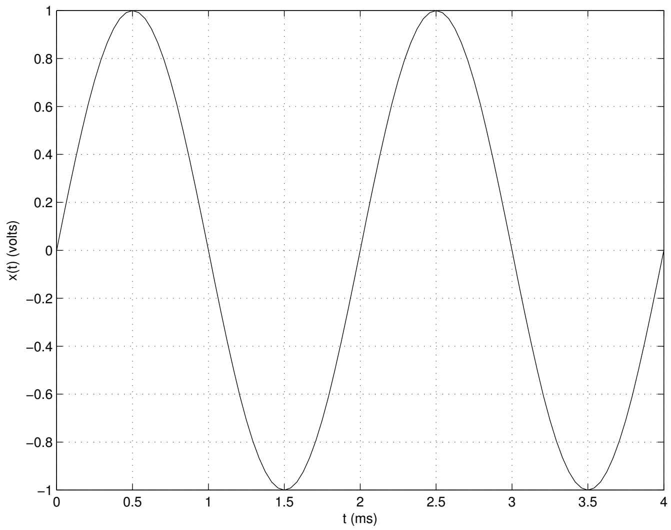

Figure 1: Figure of a 500 Hz sinusoid.

- Removing background noise
- Image enhancement such as color adjustments or red-eye removal
- Automatic tagging (identification) of persons in images
- Equalization of a communications channel

Definition: Signal processing is an *operation on or transformation* of a signal.

Definition: Digital Signal Processing (DSP) is an operation on or transformation of a signal performed *on a computer or other special purpose digital hardware*.

4 How do we do DSP?

In order to process a signal in the digital domain we must convert the continuous-time (CT) or analog signal to a digital signal by doing the first four steps (remaining two are optional depending on the application).

1. Prefilter
2. Sample (convert to discrete-time)
3. Quantize
4. Process
5. Convert to continuous-time
6. Reconstruction filter

Analog-to-Digital Converter (A/D or ADC) for steps 1) - 3)
Digital Signal Processor (DSP) for step 4)
Digital-to-Analog Converter (D/A or DAC) for steps 5) – 6)
Systems may include A/D, DSP, and/or D/A.
A/D and D/A may be on single IC called a codec

4.1 Prefilter

A prefilter or anti-aliasing filter prepares the analog signal for sampling according to Nyquist's sampling theorem. We prefilter to avoid aliasing (much, much more on this later).

4.2 Sample (convert to discrete-time)

A sample and hold circuit periodically measures or samples the voltage or current amplitude of the prefiltered analog signal. The measurement is held until the next measurement occurs. All signal information between samples is lost. This operation discretizes the time axis and yields a discrete-time signal as in Fig. 2.

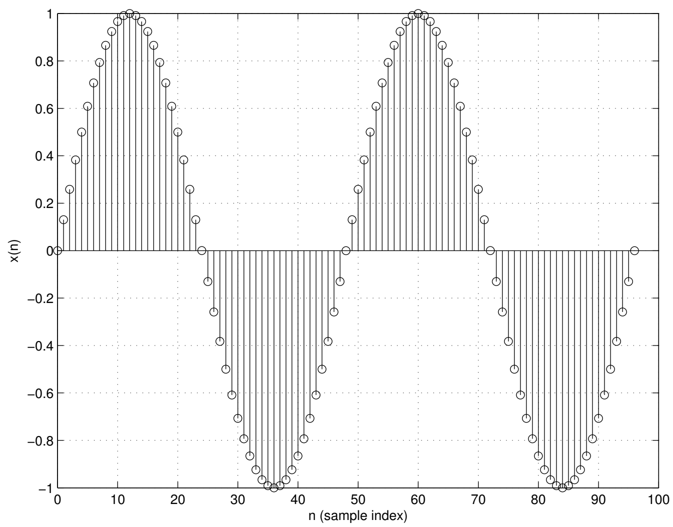

Figure 2: Discrete-time sinusoid signal.

The time between samples is called the sample period, T (units of s) and the number of samples per unit time is called the sample rate, f_s (units of samples per second or Hertz). The relation between T and f_s is

$$f_s = 1/T \tag{2}$$

Example: In CD audio, $T = 22.68\mu$s and $f_s = 44,100$ Hz. Note that the sample rate is distinctly different than MP3/AAC audio coding rate which is about compressing an already digital signal using fewer bits

4.3 Quantize

Before the discrete-time signal can be processed on a computer, the (infinitely precise) sample values must be represented in (finite precision) binary form. A quantizer rounds the sample value [measured with (ideally) infinite precision] to the nearest quantized amplitude value (lower precision) and then converts to a binary code word. This conversion yields a quantized discrete-time or *digital* signal. Quantization can also be thought of as a discretization of the amplitude axis.

Example: Consider a 3-bit quantizer (8 levels) with the following quantizer value and codes

Example: The discrete time signal (500Hz tone sampled @ 22,050Hz) converted to a digital signal using a 3-bit quantizer is shown in Fig. 3.

The more bits in the binary code, the closer the quantized value is to the original measurement for a fixed voltage swing.

Voltage range	Quantized value	Binary code
$0.75 \leq x(n) < 1$	0.875	111
$0.50 \leq x(n) < 0.750$	0.625	110
$0.25 \leq x(n) < 0.50$	0.375	101
$0.0 \leq x(n) < 0.25$	0.125	100
$-0.25 \leq x(n) < 0.0$	-0.125	011
$-0.50 \leq x(n) < -0.25$	-0.375	010
$-0.75 \leq x(n) < -0.50$	-0.625	001
$-1.0 \leq x(n) < -0.75$	-0.875	000

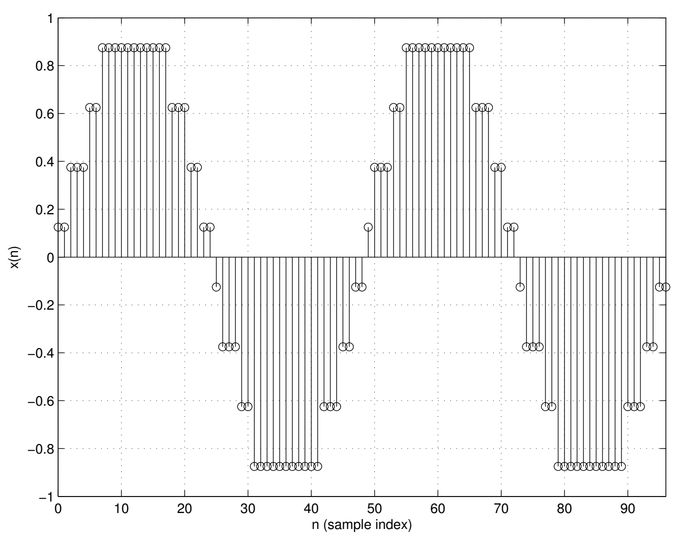

Figure 3: Digital sinusoid signal.

Example: In CD audio, each sample is quantized with 16 bits which yields 216 = 65,536 levels. Assuming a peak-to-peak voltage swing of 1V, the resolution is 30.5mV!

Note: The data rate for CD audio is 1.4 Mbps [2 channels (L/R) $\times$ 44,100 samples/s $\times$ 16 bits/sample]. High-quality MP3 or AAC is compressed (lossy) to 256 kbps (5.5$\times$ compression).

4.4 Process

In order to digitally process the signal, we design digital hardware or write software to transform the quantized values into other values. Digital hardware or software implements a digital signal processing system.

Example: Suppose we take the first four samples in the digital input signal and compute, as the output signal, the average. We then move over one sample and take the average of the four. We repeat this process for the entire signal, i.e.

$$y(n) = \frac{1}{4}\left[x(n) + x(n-1) + x(n-2) + x(n-3)\right] \tag{3}$$

Suppose our signal is a low frequency sinusoid. This process yields pretty much the same sinusoid.

Figure 4: Output from signal averaging (low frequency sinusoid input)

Suppose our signal is a high frequency sinusoid. This process quite possibly yields a signal with zero values (or very small values).

Figure 5: Output from signal averaging (high frequency sinusoid input)

We consider a device which “passes” low frequency sinusoids but “rejects” high frequency sinusoids a “low pass filter.”

4.5 Convert to Continuous-Time

A D/A takes a binary code word and produces a continuous-time output by holding the voltage constant at the quantized value over the sample period. This leads to a staircase form of the CT output as shown in Fig. 6.

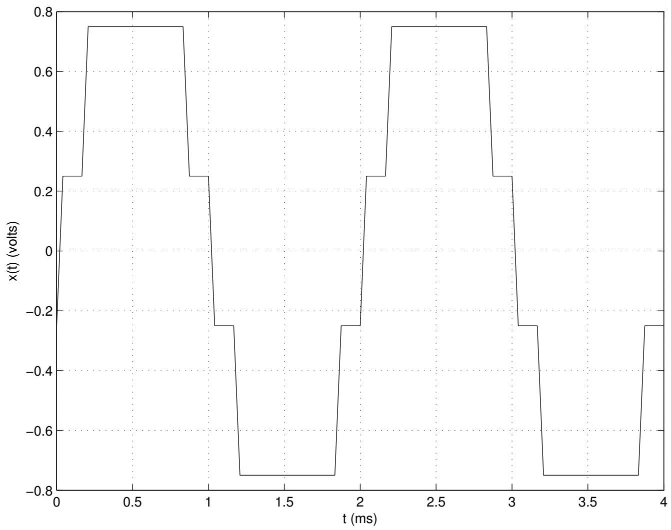

Figure 6: D/A (staircase) sinusoid signal.

4.6 Reconstruction filter

A reconstruction filter "smoothes" the staircase form after D/A conversion (much, much more on this later...) as shown in Fig. 7. Under ideal conditions, we can sample an analog signal and reconstruct the signal exactly. Under non-ideal conditions, our reconstruction can be very close to the original signal.

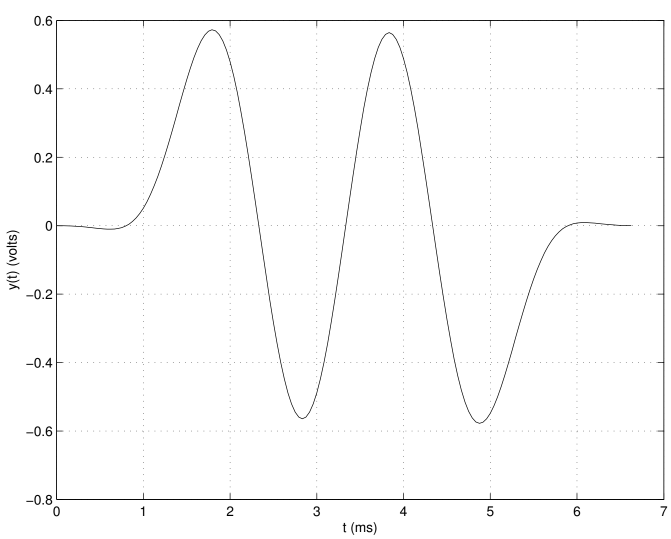

Figure 7: Reconstructed sinusoid.

A complete DSP system is shown in Fig. **??**.

Figure 8: DSP System Orfanidis p. 1

5 What are the advantages and disadvantages of DSP?

Advantages

- Flexibility: processing done in software
- Adaptability: possible time-varying (adaptive) systems - systems that "adjust" to their environment
- Accuracy: typical 16 bit precision can be used to specify very accurate system parameters
- Cost: digital signal processors continue to increase performance/price (for fixed cost, performance has doubled every 18 months for past 15 years)
- Reduced Power Consumption: digital implementation of analog functions allows for component integration, e.g. "system-on-a-chip" or SOC
- New Possibilities: complicated or impossible analog SP may now be simplified or made possible with DSP

Disadvantages

- Requires a powerful computer: computational horsepower proportional to sample rate and complexity of processing

Example: In a one minute stereo signal we have 60sec $\times$ 44,100samples/sec $\times$ 2 channels = 5,292,000 samples. Samples 100,000 100,010 for an audio signal

6 What is the history of DSP?

(1805) Gauss discovered the fundamental principle of the Fast Fourier Transform (FFT)

(1921) Nyquist of AT&T Bell Laboratories proves the sampling theorem which states that: "A signal $x(t)$ can be reconstructed from its samples $x(n)$ if the sampling rate, f_s is greater than or equal to twice the highest frequency in $x(t)$."

Example: The human ear can hear frequencies up to 20kHz. Therefore if we wish to include these frequencies in music for CD, we must sample at 40kHz or higher.

(1948) This is considered the annus [(Latin) year] mirabilis [mirus: (Latin) wonderful, astonishing, extraordinary] of signal processing. In this year:

- Claude Shannon published "A mathematical theory of communication" where he analyzed communication as the transmission of a message from a source through a channel to a receiver.
- Oliver, Pierce, and Shannon published "The philosophy of PCM" which is the classic argument for the use of pulse code modulation (PCM) or the transmission of information in the form of on-or-off pulses.
- Hamming invented error control codes
- Demonstration of the first stored-program computer
- Announcement of the invention of the transistor
- Goldmark presents the paper "The Columbia long-playing microgroove recording system", which advanced the technology behind 33 1/3 rpm LPs which provided longer play time and better sound quality.

(1965) Cooley and Tukey developed efficient algorithms for computation of Fourier transforms (FFTs) that decreased processing time by orders of magnitude. Many SP algorithms began to appear to have practical implementations.

(1980's) Rapid development in the field of microelectronics paved the way for very powerful microprocessors with special architectures designed for implementing DSP algorithms in real-time. This technology allowed the widespread application of DSP.

(1990's) Digital signal processors are applied in a wide range of consumer products

- Cordless telephones, cellphones, wireless networking devices
- Electronics such as CD, DVD/BluRay, Home Theater Receivers, gigital cameras
- Sound cards, modems, network hubs/routers

