EE545
Digital Signal Processing
(DSP)

Prof. Phillip De Leon

Welcome!
…and some things you ought to know

Klipsch School MSEE Requirements

- Department requires at least three "Klipsch-core" courses
  - EE515, EE523, EE528, EE543, EE545, EE551, EE563, EE571, EE577
- Technical groups may have required "area" core courses
  - DSP: EE545, EE555, EE571
  - Communications: EE545, EE555, EE571, EE581
  - Double-dipping encouraged
- Thesis, Technical Report, and Coursework only options
  - Thesis min 24 credits, min 6 credits EE599, and thesis defense
  - Tech report min 27 credits, min 3 credits EE598, and tech presentation
  - Coursework min 30 credits and oral/written exam or pass graduate portion of Ph.D. qualifying exam
- Other
  - No BSEE? Be prepared to take deficiency courses
  - At least 15 credits above 500-level; at least half EE; many "beginning" courses will not count, i.e. CS457; limit of 9 credits EE590 (no more than 6 EE590 non-scheduled classes)

Klipsch School MSEE Requirements

- Watch deadlines
  - File course/program plan with graduate school after 12 credits
  - Complete EE record check before last semester
  - Make arrangements for thesis defense/report presentation/coursework exam, i.e. select committee, set date, thesis/report to advisor, thesis/report to committee, etc.
  - Defend thesis/present report/pass oral exam before semester deadlines (way before semester actually ends)

- My thoughts and advice
  - Plan courseload carefully, get core out of way, and talk to 2nd year students
  - Be prepared to study harder and smarter than ever
  - Be careful of working too close with other students
  - Do not take all courses in one or two areas
  - Thesis vs. technical report vs. coursework
  - Research ideas and publications
  - Assistantships

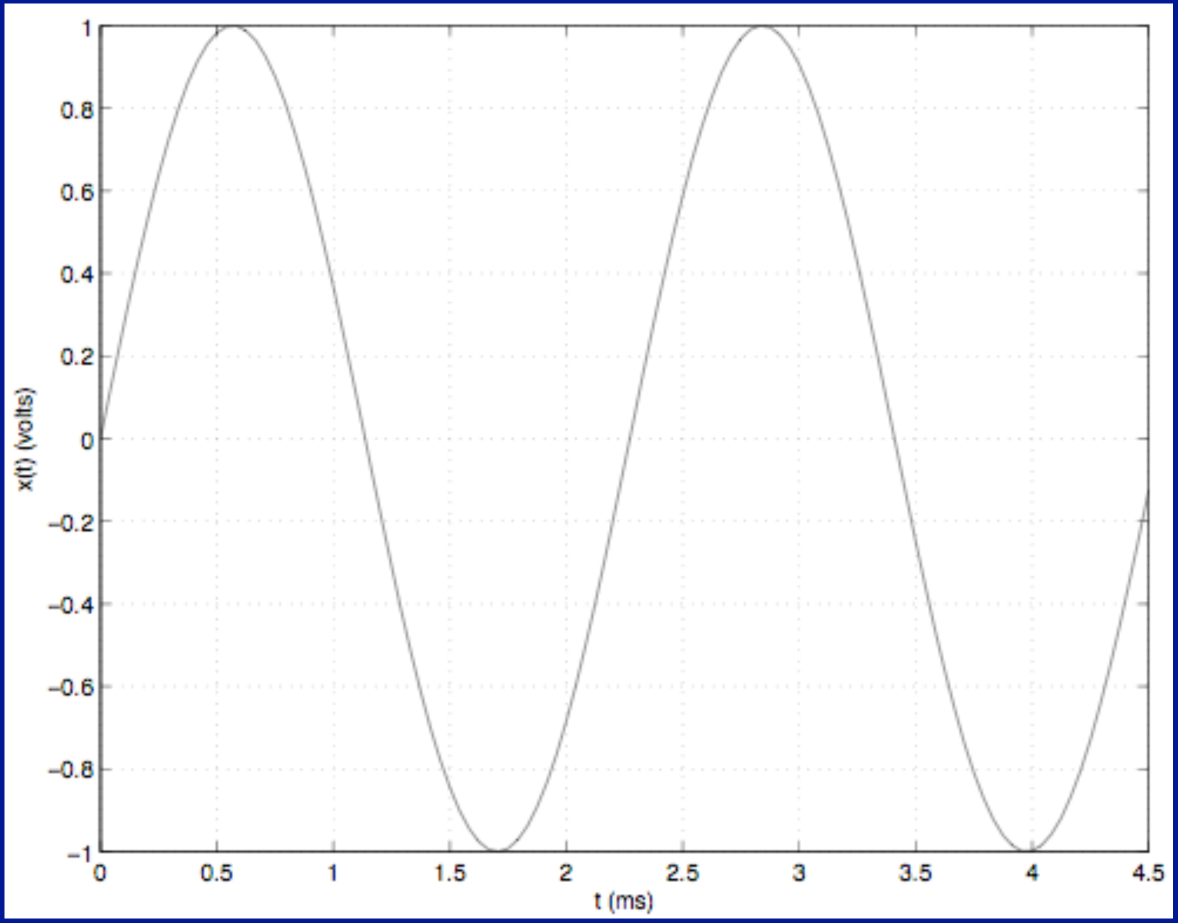
What is DSP?

Examples of “S”

- Speech, Music
  - Air pressure as a function of time, *t*
  - Voltage as a function of time, *t* if audio is picked up by a microphone
- Image
  - Brightness function of spatial variables, *x* and *y*
- Video
  - Brightness function of spatial variables, *x* and *y* and time, *t*

The Tone

- A naturally occurring signal fundamental in engineering, mathematics, physics, and music is the *tone*



A 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 of “SP”

- Equalization of audio
  - Filter gain adjustments
- Audio, video coding (MP3, AAC, MP4)
  - Reduction of data file size
- Signal Enhancement
  - Noise/interference reduction, echo cancellation, signal separation, channel equalization

Another Definition…

"Signal processing is an *operation* or *transformation* on a signal."

Yet Another Definition…

"Digital Signal Processing is an operation on or transformation of a signal performed *on a computer* or other *special purpose digital hardware*."

How do we do DSP?

Roughly Six Steps to DSP Land

- (Roughly) Six steps to processing an analog signal digitally

    1) Prefilter
    2) Analog to digital conversion
    3) Quantize
    4) Digitally process
    5) Digital to analog conversion
    6) Postfilter

- Analog-to-Digital Converter (A/D or ADC) for steps 1 – 3
- Computer, DSP, FPGA for step 4
- Digital-to-Analog Converter (D/A or DAC) for steps 5 – 6
- A/D, D/A, pre-, and post-filters may be on single IC called a codec

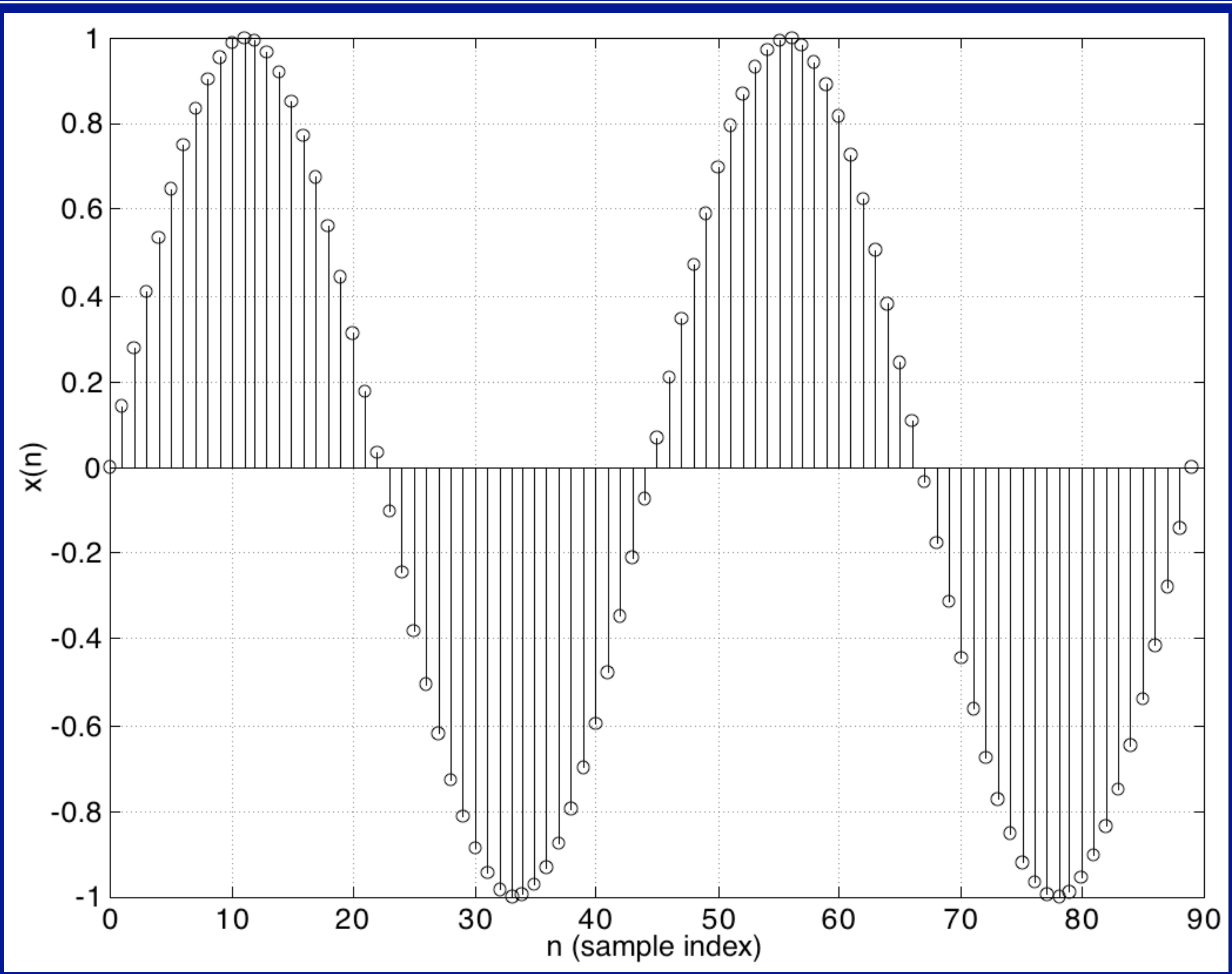
Step 1 Prefilter

- 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 soon…)

Step 2 Sample

- A sample and hold circuit periodically measures or *samples* the voltage of the prefiltered analog signal
  - Measurement is *held* until the next measurement occurs
  - Original signal waveform between samples is lost
- Sampling yields a *discrete-time signal* ("sample #1, sample #2, etc…)
  - Time between samples is called the *sample period*, T.
  - Number of samples per second is called the sample rate, f_s and is measured in units of samples per second or Hertz
  - We relate the sample period and rate as $f_s = 1 / T$
- In CD audio, f_s = 44,100 Hz and T = 22.68μs

Step 2 Sample (cont.)



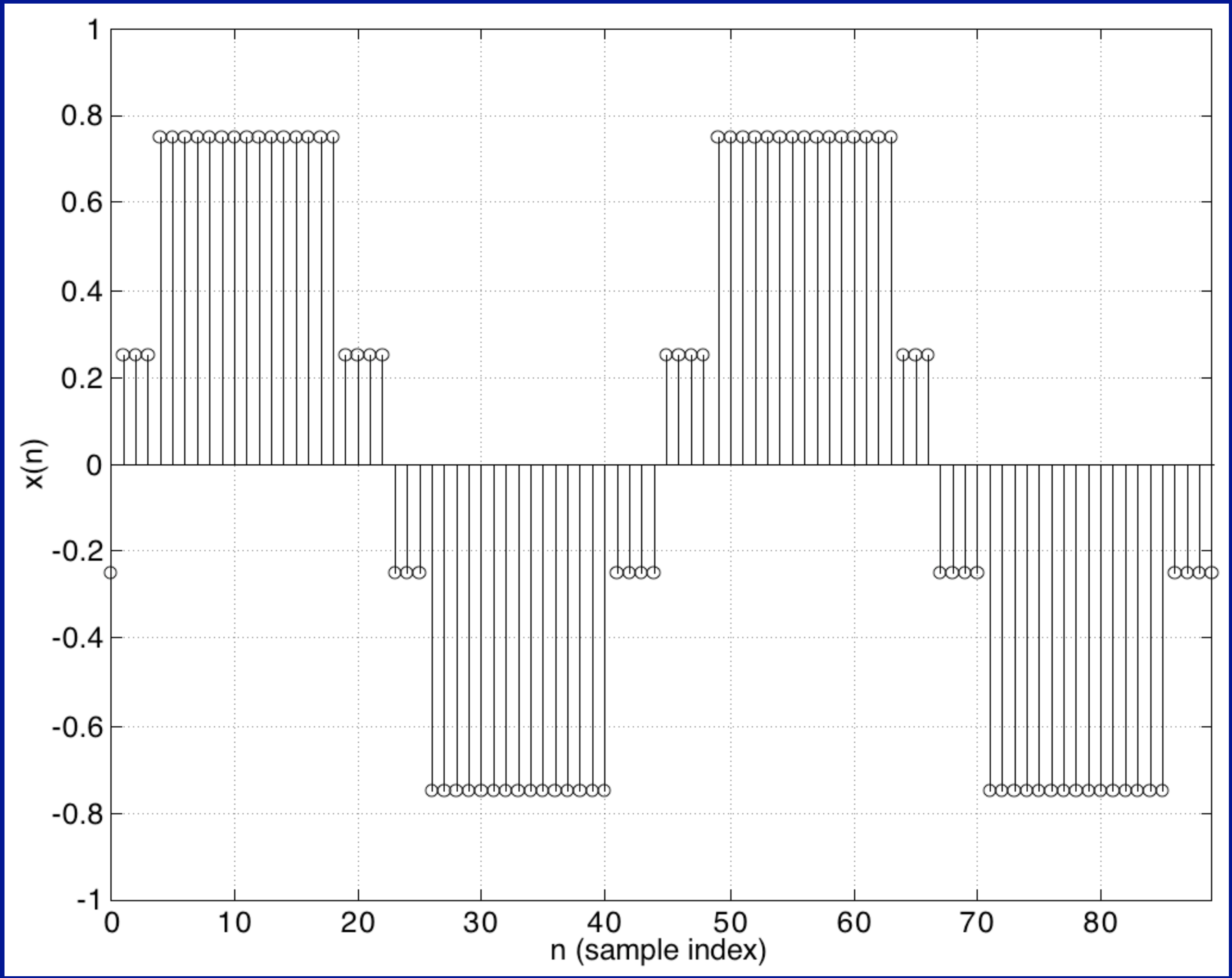
Step 3 Quantize

- A *quantizer* rounds the sample value (infinite precision) to the nearest quantized amplitude value (lower precision). Quantized value then converted to a binary code word. This conversion yields a *digital signal.*

- Example: Consider a 2-bit quantizer (4 levels) with the following quantizer value and codes

Voltage Range	Quantized Value	Binary Code
$0.5 \leq x(n) < 1.0$	0.75	11
$0.0 \leq x(n) < 0.5$	0.25	10
$-0.5 \leq x(n) < 0.0$	–0.25	01
$-1.0 \leq x(n) < -0.5$	–0.75	00

Step 3 Quantize (cont.)



Step 3 Quantize (cont.)

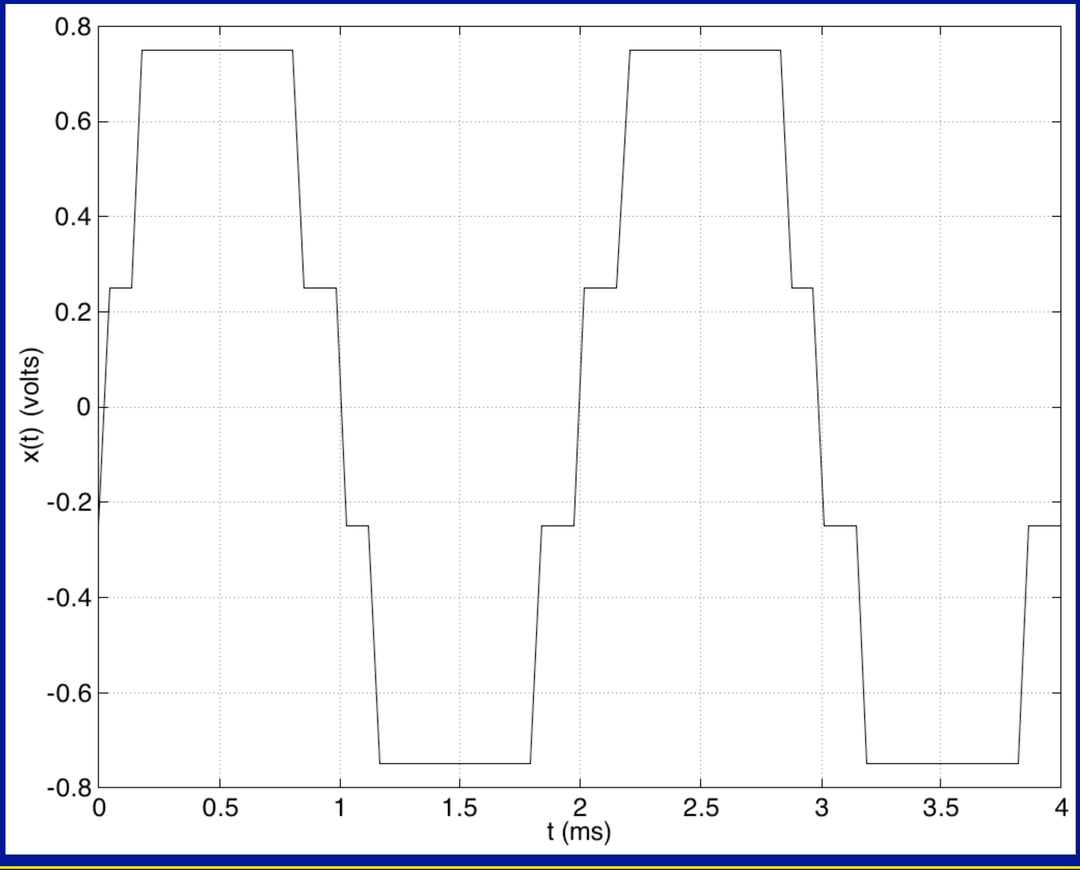
- The more bits in the binary code, the closer the quantized value is to the original measurement for a fixed voltage swing
- Example: In CD audio, each sample is quantized with 16 bits which yields $2^{16} = 65{,}536$ levels.
  - Assuming a peak voltage of +/– 1V, the resolution is 30.5 μV (which is quite high for audio signals)

Step 4 Digitally Process

- To do actual digital signal processing, we design digital hardware or write software codes to transform the quantized values into other values
- Example: Suppose we compute the average the first four samples in the signal and output the average. We then move over one sample and compute and output the average of the next four….
  - Low frequency tone: The output is pretty much the same tone
  - High frequency tone: The output quite possibly has all zero values (or very small values)
  - We consider a device which “passes” low frequency sinusoids but “rejects” high frequency sinusoids a “low pass filter”

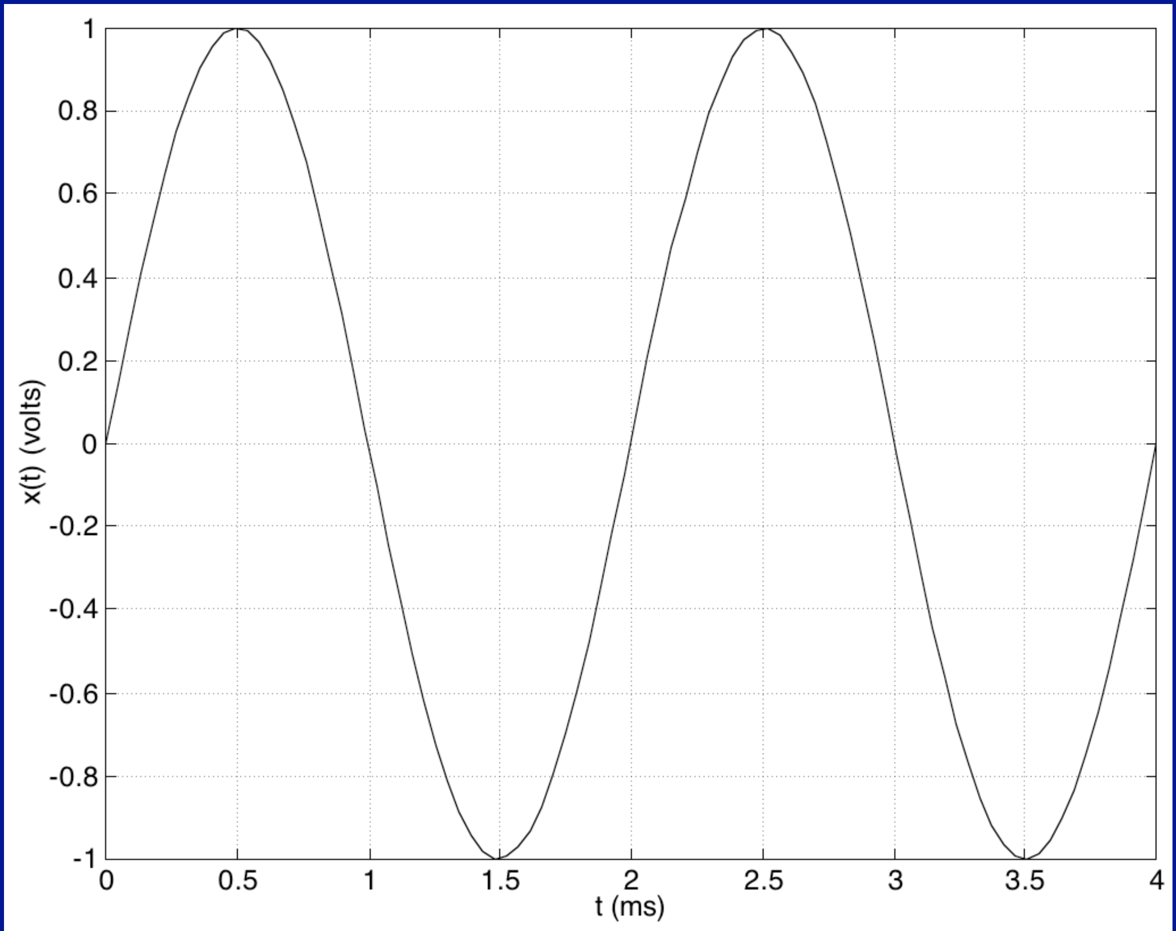
Step 5 Convert Output to Analog

- DAC takes a code word and produces a CT output by holding voltage constant at quantized value over sample period. This leads to a staircase form of the CT output.

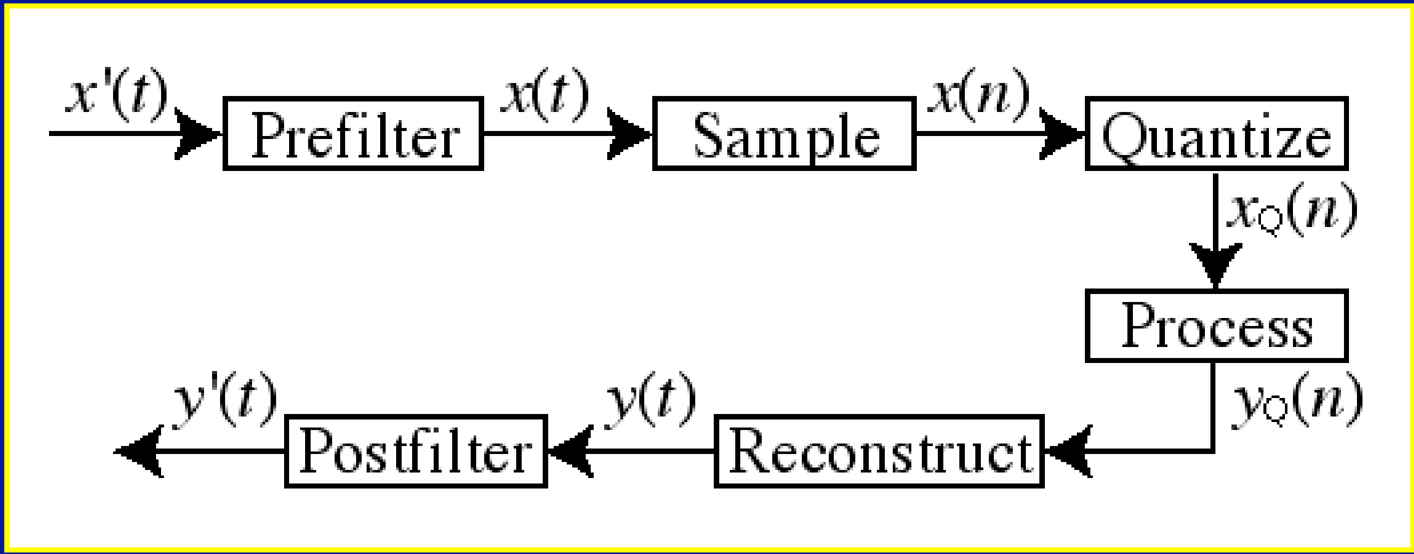


Step 6 Postfilter

- A postfilter or reconstruction filter "smoothes" the staircase form after D/A conversion



A Complete DSP System



Seems like a lot of trouble…

Advantages/Disadvantages of DSP

- Advantages
  - Flexibility: processing done in software
  - Adaptability: possible time-varying (adaptive) systems - systems that learn and "adjust" to their environment
  - Accuracy: typical 16 bit precision can be used
  - Cost: DSPs continue to increase performance/price (for fixed cost, performance has doubled every 18 months for past 20 years)
  - 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 significant processing: computational horsepower proportional to sample rate and complexity of processing

History of DSP

- (1805) *Gauss* discovers 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 $x(t)$."

History of DSP (cont.)

- (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.

History of DSP (cont.)

- (1965) *Cooley and Tukey* developed fast Fourier transforms (FFTs) that decreased processing time by orders of magnitude.
- (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 - Present) Role of DSP is accelerating

You're going to learn a lot …good luck!


NEW MEXICO STATE UNIVERSITY
New Mexico
State University