MODIFIED TLH

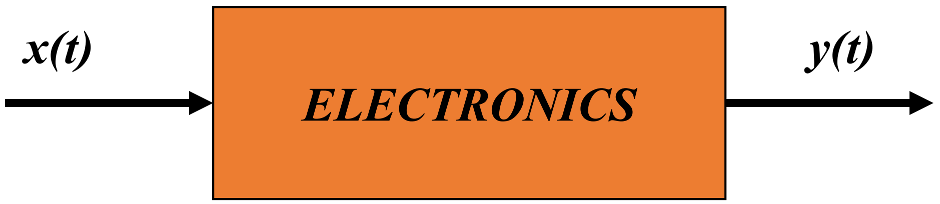
DSP First, 2/e

Sampling & Aliasing

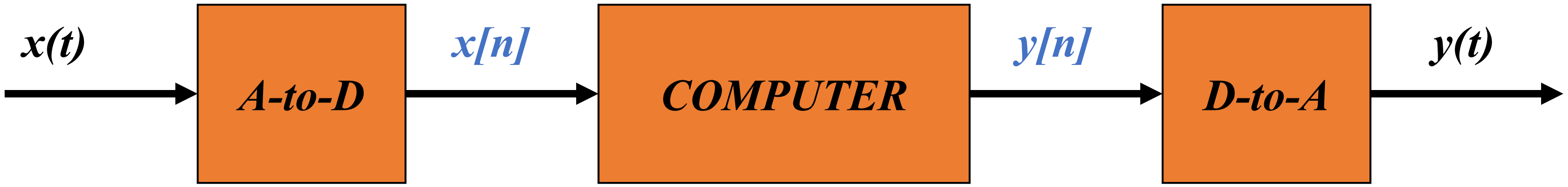
CHAPTER 4 PRESENTATION 2

System IMPLEMENTATION

- ANALOG/ELECTRONIC:
  - Circuits: resistors, capacitors, op-amps

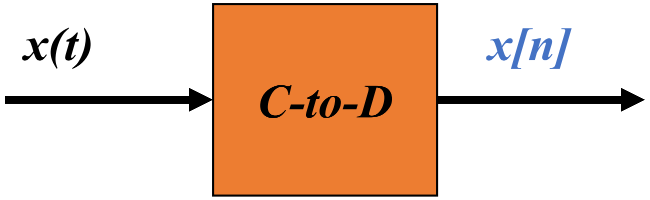


- DIGITAL/MICROPROCESSOR
  - Convert x(t) to numbers stored in memory



SAMPLING x(t)

- SAMPLING PROCESS
  - Convert x(t) to numbers x[n]
  - "n" is an integer index; x[n] is a sequence of values
  - Think of "n" as the storage address in memory
- UNIFORM SAMPLING at $t = nT_s$
  - IDEAL: $x[n] = x(nT_s)$



SAMPLING RATE, f_s

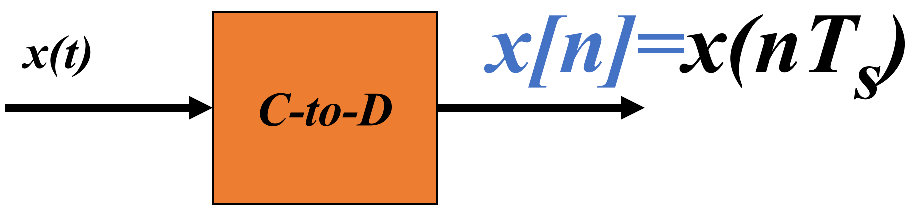
- SAMPLING RATE (f_s)
  - $f_s = 1/T_s$
    - NUMBER of SAMPLES PER SECOND

SOMETIMES GIVEN IN Hz

  - T_s = 125 microsec → f_s = 8000 samples/sec
      - UNITS of f_s ARE HERTZ: 8000 Hz

Thus - Fmax = 4000 Hz

- UNIFORM SAMPLING at $t = nT_s = n/f_s$
  - IDEAL: $x[n] = x(nT_s) = x(n/f_s)$



STORING DIGITAL SOUND

- $x[n]$ is a SAMPLED SISIGNAL
  - A list of numbers stored in memory
- EXAMPLE: audio CD
- CD rate is 44,100 samples per second
  - 16-bit samples
  - Stereo uses 2 channels

THUS – Frequency range of 22,050 Hz is beyond (most) humans hearing range.

- Number of bytes for 1 minute is
  - 2 X (16/8) X 60 X 44100 = 10.584 Mbytes

SAMPLING THEOREM THE BIG DEAL!!

- HOW OFTEN DO WE NEED TO SAMPLE?
  - DEPENDS on FREQUENCY of SINUSOID
  - ANSWERED by SHANNON/NYQUIST Theorem
  - ALSO DEPENDS on "**RECONSTRUCTION**"

Shannon Sampling Theorem

A continuous-time signal $x(t)$ with frequencies no higher than $f_{\max}$ can be reconstructed exactly from its samples $x[n] = x(nT_s)$, if the samples are taken at a rate $f_s = 1/T_s$ that is greater than $2f_{\max}$.