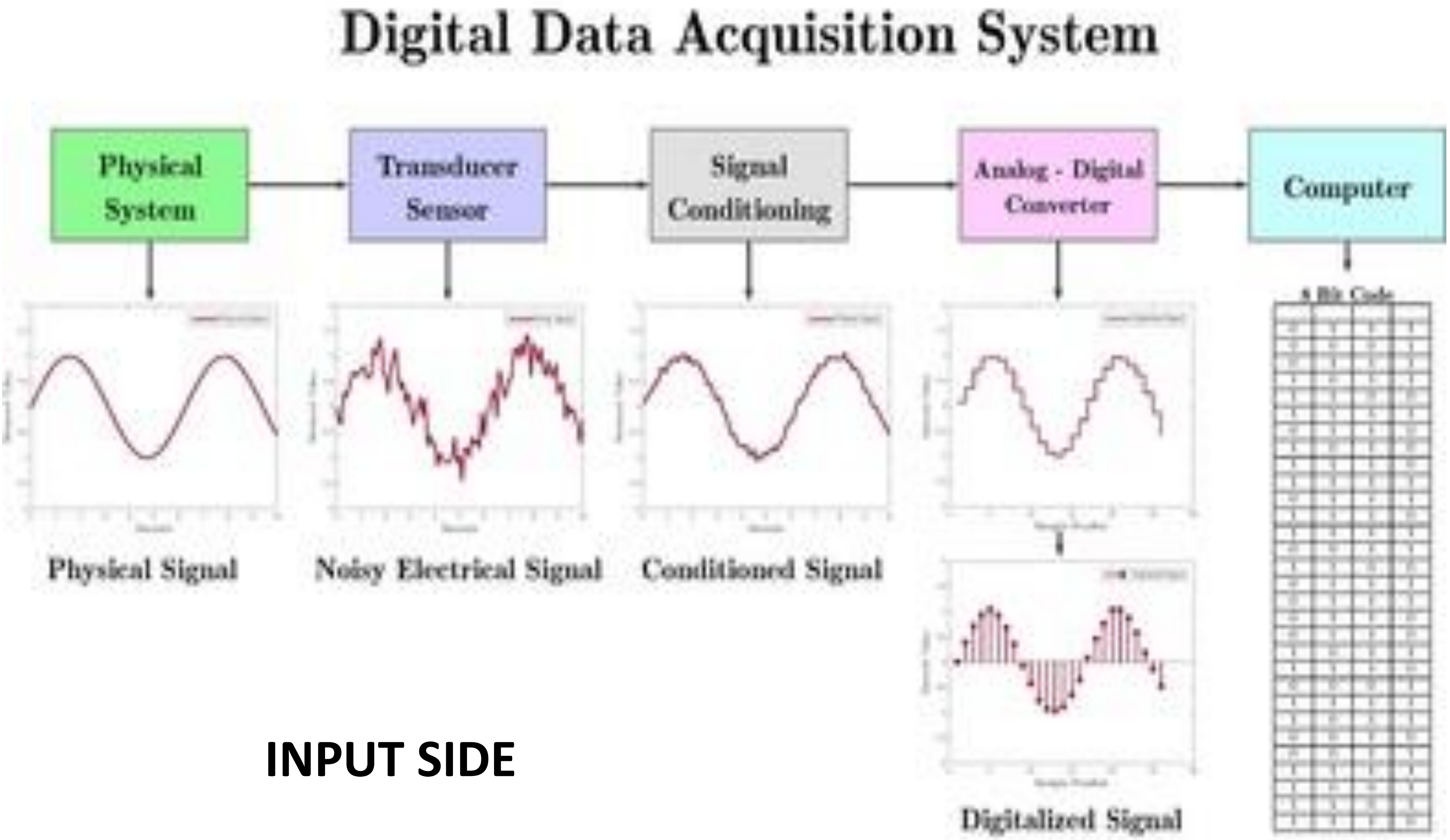
DATA ACQUISITION SYSTEMS

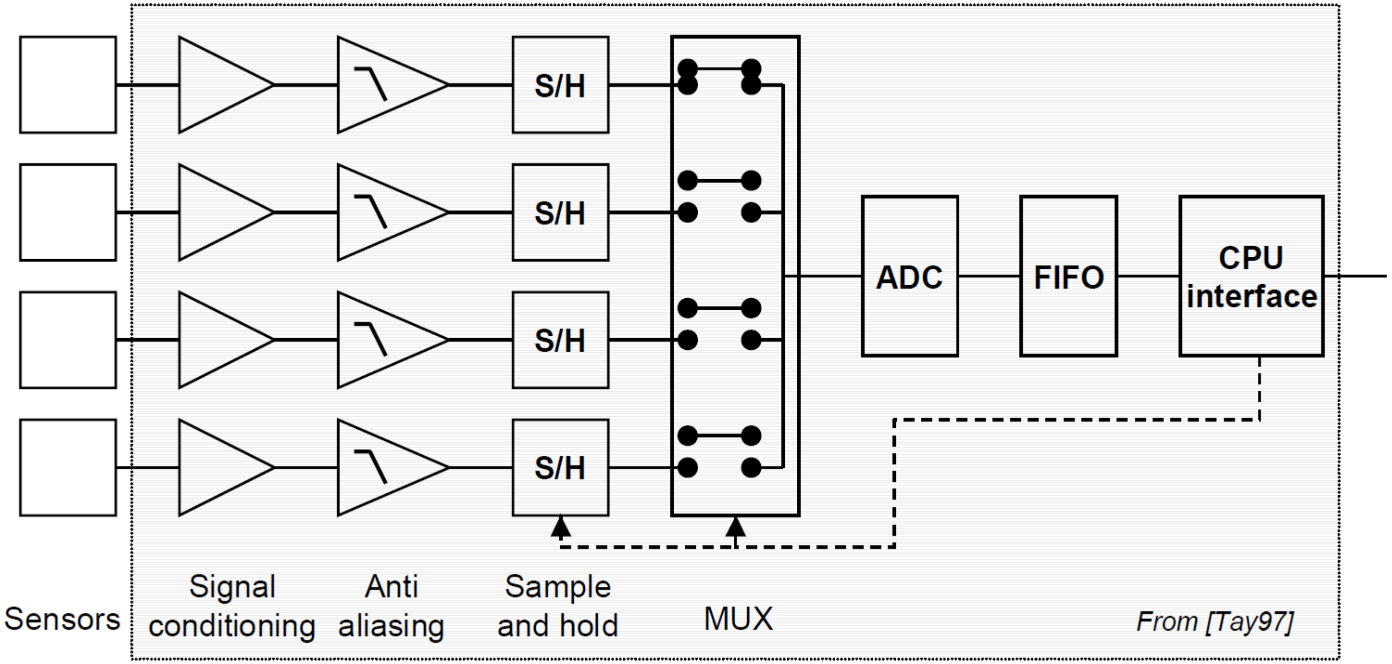
See TLH Chapter 11 for Text

RECORD ON


Digital Data Acquisition System
Physical System
Transducer Sensor
Signal Conditioning
Analog - Digital Converter
Computer
8 Bit Code
Physical Signal
Noisy Electrical Signal
Conditioned Signal
Digitalized Signal

INPUT SIDE

Architecture of data acquisition systems



SIGNAL CONDITIONING
AMPLIFICATION
ANTI-ALIASING

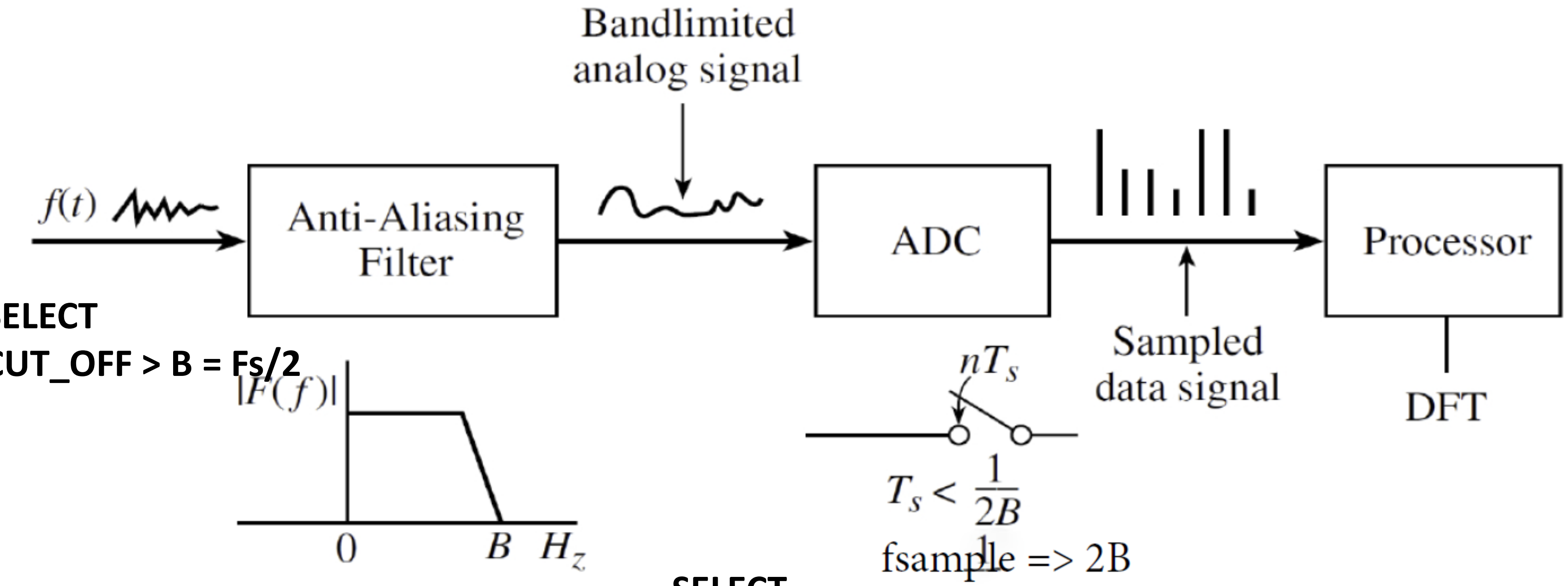
Sampling and the DFT Suppose that an analog signal is sampled at a rate

$$f_{\text{sample}} = \frac{1}{T_s} > 2B \quad \text{samples/second.}$$

The highest possible frequency in the DFT spectrum would be $F_{\max} = 1/(2T_s)$ hertz. If the signal is bandlimited to B hertz and sampled properly, the component at the DFT maximum frequency should be zero since $F_{\max} > B$ hertz.

11.4 PRACTICAL SIGNAL ANALYSIS

Assume that a physical signal is to be analyzed using the DFT to determine the spectral components. Figure 11.7 presents a simplified diagram of the input stage of a data acquisition system.



SELECT
CUT_OFF > B = Fs/2

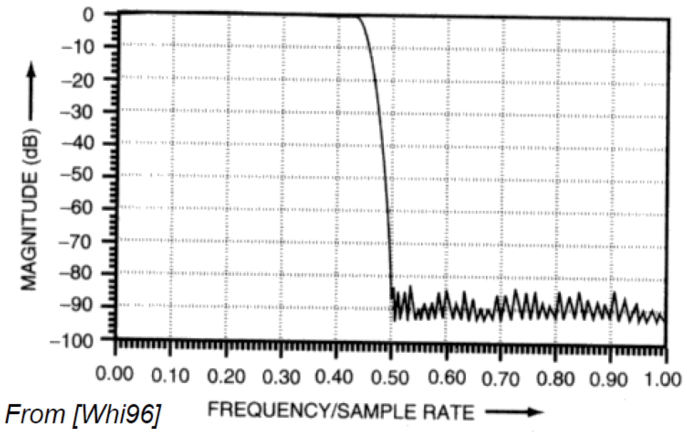
SELECT
fsample > 2x Highest Frequency of Interest

FIGURE 11.7 *Data acquisition system*

Anti-aliasing filters

ANALOG FILTER

- **An anti-aliasing filter is a low-pass filter designed to filter out frequencies higher than the sampling frequency**
  - An anti-aliasing filter should have
    - Steep cut-off and
    - Flat response in the frequency band



From [Whi96]

- **Typical filters are:**
  - **Butterworth**: flattest response in the frequency band but phase shifts well below the break frequency
  - **Bessel**: phase shift proportional to frequency, so the signal is not distorted by the filter
    - Recommended for anti-aliasing if it is important to preserve the waveform
  - **Chebyshev**: steepest cut-off but it has ripples in the band-pass

SIGNAL CONDITIONING
AMPLIFICATION
ANTI-ALIASING

Anti-aliasing

- **The effects of aliasing can also be observed on the frequency spectrum of the signal**
- **In the figures below**
  - F_1 appears correctly since $F_1 \leq F_S/2$
  - F_2, F_3 and F_4 have aliases at 30, 40 and 10Hz, respectively
  - You can compute these aliased frequencies by folding the spectrum around $F_S/2$ or with the expression

$$\text{Alias frequency } \hat{F} = \min\left|kF_S - F\right|_{\forall k \text{ integer}}$$

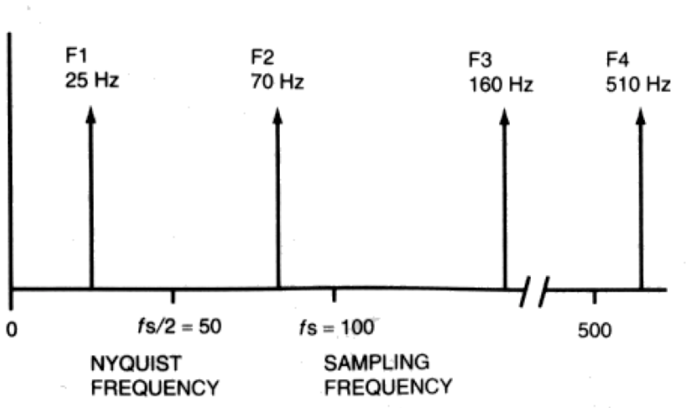
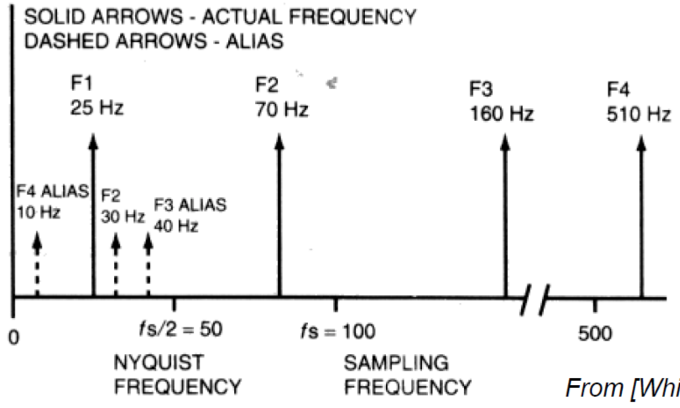


FIGURE 117.5 Spectral of signal with multiple frequencies.



From [Whi96]

FIGURE 117.6 Spectral of signal with multiple frequencies after sampled at $fs = 100$ Hz.

SAMPLING

Two of the most important questions in the specification of a data acquisition system such as that shown in Figure 11.7 are the following:

1. How often should the analog signal be sampled?

2. How long should the signal be sampled?

If the highest frequency of interest in the signal is B hertz and the frequency spectrum of the signal is limited to B hertz by the anti-aliasing filter, the *sampling theorem* answers the question in Part 1. The theorem is the cornerstone of practical and theoretical studies in electronic communication.

TABLE 11.4 *DFT parameters*

Parameter	*Notation*
Time domain:	
Sample interval	T_s (s)
Sample size	N points
Length	$(N-1)T_s$ (s)
Period (from IDFT)	$T = NT_s$ (s)
Frequency domain:	
Frequency Spacing	$f_s = \frac{1}{T} = \frac{1}{NT_s}$ (Hz)
Spectrum size	N components
Maximum frequency	$\frac{N}{2} f_s = F_{\max}$ (Hz)
Frequency period	$F_p = N f_s = \frac{1}{T_s}$ (Hz)

Fmax = fsample/2

Sampling Example

This example defines the relationship between sampling interval, frequency resolution, and number of samples for the DFT. In terms of previous notation, T_s is the sampling interval in seconds, f_s is the frequency resolution, and N is the number of sample points in time and in frequency.

Consider an analog signal with frequencies of interest up to 1200 hertz. The desired frequency resolution is 0.5 hertz. Thus, the signal should be filtered so that $B = 1200$ hertz. This filtering removes frequencies in the signal above 1200 hertz and *noise* above B hertz. The noise consists of unwanted signals added to the desired signal that are the result of environmental effects as the signal is transmitted to the data acquisition system.

By the sampling theorem, the sampling interval in time must be

$$T_s < \frac{1}{2B} = \frac{1}{2400} \quad \text{seconds},$$

so that at least 2400 samples per second are needed. For a resolution of 0.5 hertz, $T = 1/0.5 = 2$ seconds. The total number of points required is thus

$$N = \frac{T}{T_s} = \frac{2}{(2400)^{-1}} = 4800.$$

If N is to be a power of 2 for the FFT algorithm, $2^{13} = 8192$ samples would be taken. The sampling rate could be increased to 4096 samples per second, which is sampling at a rate corresponding to about 3.4 times the highest frequency of interest.

□

GO TO TLH CHAPTER 11

11.3 MATLAB FOURIER COMMANDS

MATLAB contains a number of commands to compute, manipulate and plot the DFT of a function. These are listed in Table 11.5. Except for the command **fourier**, the commands are used for numerical computation. The symbolic command **fourier** is part of the *Symbolic Math Toolbox*. If a function can be defined symbolically, **fourier** computes the Fourier integral transform. The *Signal Processing Toolbox* has additional commands for more advanced signal processing.

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