Chapter 7

Sampling, Digital Devices, and Data Acquisition

Material from Theory and Design for Mechanical Measurements; Figliola, Third Edition

Introduction

✶ "Integrating analog electrical transducers with digital data-acquisitions systems is cost effective and commonplace on the factory floor, the testing lab, and even in our homes. There are many advantages to this hybrid arrangement, including the efficient handling and rapid processing of large amounts of data and varying degrees of artificial intelligence by using digital microprocessor systems."

Analog Signal and Discrete Time Series

Common Questions:

- Frequency content of measured signal?
- Size of time increment?
- Total sample period?
- How often should we sample?

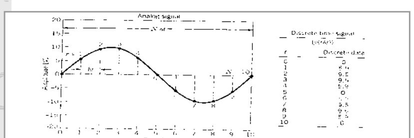

Figure 7.1 Analog and discrete representations of a time-varying signal.

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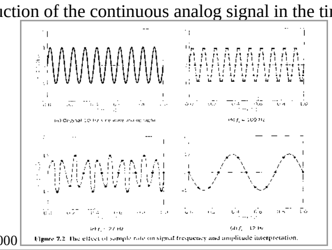
Notes:

- A continuous dynamic signal can be represented by a fourier series.
- The discrete fourier transform can reconstruct a dynamic signal from a discrete set of data.

Sample Rate

- Sample time increment $\tau_f = \delta t$ (seconds)
- Sample Frequency $f_s = 1/\delta t$ (hertz)
- The sample rate has a significant effect on our perception and reconstruction of the continuous analog signal in the time domain.



Figliola, 2000

- Note the decrease in frequency of signal estimated by the slower sampling frequency.
- The sampling theorem states that to reconstruct the frequency content of a measured signal accurately, the sample rate must be more than twice the highest frequency contained in the measured signal.
- Sampling theorem: $f_s > 2f_m$
- Then, $\delta t < 1/(2f_m)$ should always give accurate DFT frequency determination.

f_m is the maximum frequency component in a signal.

Alias Frequency

- If $f_s < 2f_m$, the high frequency content will be falsely represented by a low frequency component. False frequency is called alias frequency and results from discrete sampling of a signal at $f_s < 2f_m$
- The alias phenomenon is an inherent consequence of a discrete sampling process.
- Refer to the discussion of folding frequency for more detail in the text.

Alias Frequency

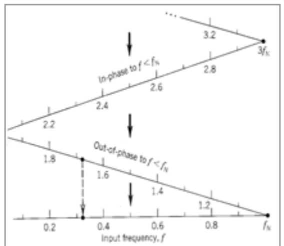
- By following sampling theorem $f_s > 2f_m$, all aliases are eliminated.
- The concepts apply to complex periodic, aperiodic and non-deterministic that are represented by fourier transform

Alias Frequency

- Nyquist frequency: $f_N = f_s/2 = 1/(2\delta t)$
- This represents a folding point for the aliasing phenomenon.
- All actual frequency content in the analog signal that is at frequencies above f_N will appear as alias frequencies of less than f_N; that is, such frequencies will be folded back and superimposed on the signal at lower frequencies.
- An alias frequency, f_a, can be computed from the folding diagram, in which the original frequency axis is folded back over itself at the folding point of f_N and its harmonics, mf_N, where m = 1, 2, …

Alias Frequency

- For example, as noted by solid arrows in this figure, the frequencies of f = $0.5f_N$, $1.5f_N$, $2.5f_N$,… will all appear as $0.5f_N$ in the discrete series $y(r\delta t)$



7.3 The folding diagram for alias frequencies.

Alias Frequency

- How is the alias of an unknown signal avoided?
  - If maximum frequency of interest is known, set $f_s = 2f_{max}$ and use low pass filter with $f_c = f_{max}$
  - Set f_s at max and set $f_c = f_s/2$

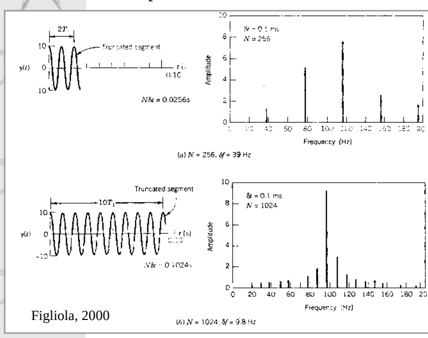
Amplitude Ambiguity

- The DFT (discrete fourier transform) of sampled discrete time signal is unchanged by changes in the total sample period $N\delta t$, provided that:
  1. Total sample period is integer multiple of fundamental period T_1, $mT_1 = N\delta t$
  2. Sample increments meets $\delta t < 1/(2f_m)$
- Thus, the DFT will accurately represent the frequencies and amplitudes of the discrete time series regardless of δt used.

Amplitude Ambiguity

- Sample period defines the frequency resolution of the DFT:
  - $\delta f = 1 / (N\delta t) = f_s/N$
- Problems occur when $N\delta t \neq mT_1$.
  - problems due to truncation of complete cycle.
- Error decreases as $N\delta t$ more closely approaches mT_1 or as $f_s >> f_m$

Observe noise on the amplitude spectrum plots, known as leakage, where amplitude is leaked to adjacent frequencies, which prevents accurate reconstruction of signal.



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y(t) = 10 cos 628t
Note in a) and b) the original signal can not be reconstructed.

"The reconstruction of a measured waveform from a discrete signal is controlled by the sampling rate and DFT resolution."

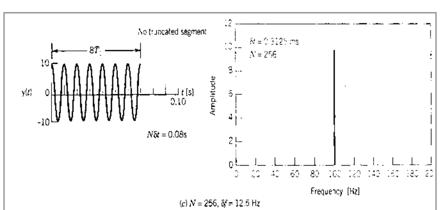

Figure 7.4 Amplitude spectra for $y(t)$.

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Amplitude Ambiguity

- By varying the sample period or its equivalent, the DFT resolution, leakage can be minimized, and the accuracy of the spectral amplitudes can be controlled.
- If y(t) is an aperiodic or nondeterministic waveform, there may not be a fundamental period.
- In such a situation, one controls the accuracy of the spectral amplitudes by varying the DFT resolution, δf, to minimize leakage.

Amplitude Ambiguity

- In summary, the reconstruction of a measured waveform from a discrete signal is controlled by the sampling rate and the DFT resolution.
- By adherence to the sampling theorem, one controls the frequency content of both the measured signal and the resulting spectrum.
- By variation of δf, one can control the accuracy of the spectral amplitude representation.

Selecting Sample Rate and Data Number

- Use $\delta t < 1/2f_m$ (eq 1)
- For an exact discrete representation in both frequency and amplitude of any periodic, analog waveform, both the number of data points and the sample rate should be chosen based on the preceding discussion.
- This equation sets the maximum value for δt, or the minimum sample rate f_s, and the next equation sets the total sampling time $N\delta t$, from which the data number N is estimated.

Selecting Sample Rate and Data Number

- For most real signals, exact discrete representations of the input analog signal frequency and amplitude content are not possible or practical.
- Setting the sample rate f_s at five time the maximum signal frequency together with large values of $N\delta t$ is recommended to minimize spectral leakage and provide a good approximation of the original signal.

Selecting Sample Rate and Data Number

- An antialias filter should be used to ensure that no frequency above a desired maximum frequency is encountered.
- Still, the maximum sample rate available will be limited by the data-acquisition system and the maximum data number by the memory size available.

Selecting Sample Rate and Data Number

- $\delta f = 1/N\delta t = f_s/N$ (eq 2)
  - Eq 1 sets max value of δt and Eq 2 sets N
- Exact representation is not possible
- Set sampling rate $f_s = 5 * f_m$ and set $N\delta t$ to large values to reduce/minimize spectral leakage and get good approximation of signal.
- Recommend use of anti-aliasing filter.

Selecting Sample Rate and Data Number

- Estimation of y(t)

$$y^*(t) = 1/\pi \sum_{r=1}^{N} y(r\delta t) \frac{\sin[\pi(t/\delta t) - r]}{(t/\delta t) - r}$$

  - Where y*(t) is a reconstructed signal

Digital Devices

- Digital signals are discrete in time and amplitude. Almost all systems use some variation of the binary numbering system, base 2.
  - Bit = on/off ~ 0/1
  - Byte = 8 bits
  - Word = 4 bits to 64 bits depending on microprocessor
  - 2 bit: 00, 01, 10, 11 is 0 to 3

Representation

N Bits	Data Range		
1 bit	0 to 1	0	range (2^n-1)
2 bit	0 to 3	00	
3 bit	0 to 7	000	
4 bit	0 to 15	0000	
8 bit	0 to 255		
16 bit	0 to 65535		

Representation

- $b_8b_7b_6b_5b_4b_3b_2b_1b_0$ where b_n = 0 or 1
  ↑ MSB ↑ LSB
- 10010011
  ↑ MSB ↑ LSB

LSB = least significant bit
MSB = most significant bit

Straight Binary

- Equivalent decimal value
  - Value= $\sum_{i=0}^{n-1} b_i 2^i$
  - Value= $b_0 2^0 + b_1 2^1 + b_3 2^3 + \ldots + b_{n-1} 2^{n-1}$

= 1(1) + 1(2) + 0*4 + 0*8 + 1*16 + 0*32 + 0*64 + 1*128

= 1 + 2 + 16 + 128

= 147

Examples of other binary is given in Table 7.1

Binary Coded Decimal (BCD)

- 532_{10} – 0101 0011 0010
  5 3 2
  - Each four bits represent the decimal number between 0 to 9 used in digital readouts.

Table 7.1 Binary Codes (Example: 4-Bit Words)

Bits	Straight	Offset	Twos Complement	Ones Complement	AVPS
0000	0	−7	+0	+0	−0
0001	1	−6	+1	+1	−1
0010	2	−5	+2	+2	−2
0011	3	−4	+3	+3	−3
0100	4	−3	+4	+4	−4
0101	5	−2	+5	+5	−5
0110	6	−1	+6	+6	−6
0111	7	−0	+7	+7	−7
1000	8	+0	−8	−7	+0
1001	9	+1	−7	−6	+1
1010	10	+2	−6	−5	+2
1011	11	+3	−5	−4	+3
1100	12	+4	−4	−3	+4
1101	13	+5	−3	−2	+5
1110	14	+6	−2	−1	+6
1111	15	+7	−1	−0	+7

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Parallel vs. Serial Communication

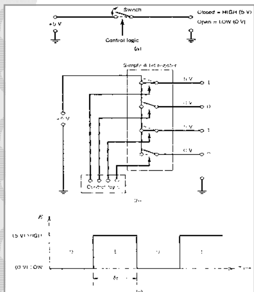

Figure 7.5 Methods for transmitting digital information. (a) Simple on/off (1 or 0) switch. (b) 4-bit register transmitting 1 0 1 0 in parallel. (c) Serial transmission of 1 0 1 0 as a TTL pulse signal.

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- A simple on/off switch
- Parallel: by setting control logic a single +5v source is used to set a binary register
- Serial: by turning on/off a single output for duration δt, a serial representation of a bit string can be communicated.

Digital to Analog Converter

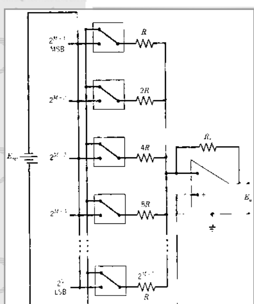

Figure 7.6 D/A converter.

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- A digital-to-analog (D/A) converter is an M-bit digital device that converts a digital binary word into an analog voltage.
- Network of M binary weighted resistors with a common summing point.
- MSB has R, while LSB has 2_{m-1}*R resistance.
- Network output of $$\mathbf{I} = \mathbf{E}_{\mathbf{ref}} \sum_{\mathbf{m}=1}^{\mathbf{M}} \mathbf{cm} \;/ (2_{m-1} * \mathbf{R})$$
- Output voltage: $E_o = IR_r$
- The full-scale output range E_{FSR} is 0 to 10 V or ±5V for m of 8, 12, 16, 18 bits.

Analog to Digital Converter

- <u>Quantization</u> – the conversion of analog voltage input to binary numbers.
- A/D converter is rated by its full-scale voltage range E_{FSR} and the number of bits “M” in the output resister.
- Ex: if E_{FSR} = 10 V or ±5 V, M = 8 bits or 256 values

Analog to Digital Converter

- The resolution of an A/D converter is defined in terms of the smallest voltage increment that will cause a bit change.
- Resolution is specified in volts and determined by:
- $Q = E_{FSR}/2^{M}$ = 10 V/ 256 = 39.1 mV, for m=8.

Analog to Digital Converter

- <u>Quantization Error</u> – due to limited resolution, results in error between analog and digital voltage
- For A/D converter with E_{FSR} = 0 to 4V, M = 2 bit

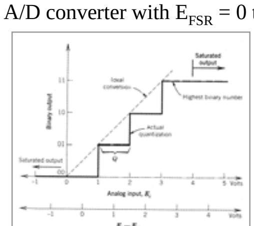

Figure 7.7 Binary quantization and saturation.

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A/O converter resolution in terms of signal to noise ratio

- Ratio of power of signal (E^2/R) to power that can be resolved by quantization (resolution of power $P/2^m$) expressed in decibels. SNR[dB] = 20 log 2^m

Table 7.2 Conversion Resolution

Bits M	Q^a [V]	SNR [dB]
2	2.5	12
4	0.625	24
8	0.039	48
12	0.0024	72
16	0.15 (10^{-3})	96
18	0.0381 (10^{-3})	108

[a] Assumes E_{FSR} = 10 V.

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Saturation vs. Conversion Error

- Saturation Error – A/D converter is limited to minimum and maximum voltage. If E_i is outside range of A/D converter, E_o is saturated at the level of A/D range. The difference between A/D limit and E_i is considered the saturation error.
- Conversion Error – A/D converter experiences bias and precision error based on method of conversion. They fall into categories we have seen previously: hystersis, linearity, sensitivity, zero, and repeatability.

Data Acquisition Systems

- Data Acquisition Systems – quantifies and stores data
- Types:
  - Manual Dedicated
  - Processor and A/D PC
  - Data Acquisition
  - Storage

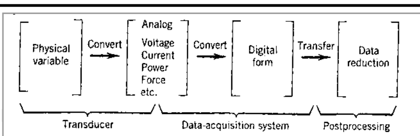

Figure 7.13 Typical analog signal and measurement scheme.

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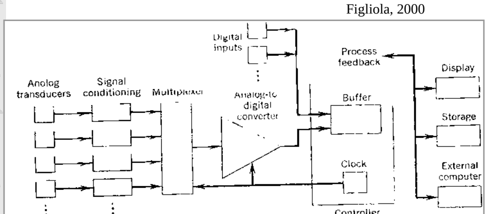

Figure 7.14 Signal flow scheme for an automated data-acquisition system.

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Filters

- Filters:
  - Analog- control frequency content of signal being sampled, remove aliases.
  - Digital- forward or backward moving average, remove unwanted components.
    - Take DFT, modify signal in frequency domain, inverse DFT.

Voltage Divider Circuit

- Amplifiers:
  - most signals need amplification or attenuation before A/D conversion to prevent saturation.
- Voltage divider to attenuate input
  - $E_o = E_i (R_2 / (R_1 + R_2))$

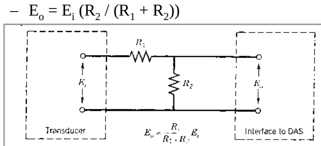

Figure 7.15 Voltage divider circuit for signal amplitude attenuation.

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- Note: Digital filters can not remove aliases.

Amplifiers

- When only the dynamic content of signal is important. Ex: mean value of signal is large with respect to the dynamic content.
  - $y(t) = 2 + 0.05 \sin 2\pi ft$ [volts]
  - f = 5 hz
- Solution:
  1) remove mean component before amplification by adding on an equal but opposite sign, mean voltage.
  2) pass signal through a high pass filter to remove DC component, setting $f_{cutoff} = 1$ hz

Shunt Resistors

- Many common transducers produce a current as their output; however, an A/D converter requires a voltage signal at its input.
- It is straightforward to convert current signals into voltage signals by using a shunt resistor.

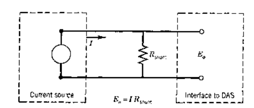

Figure 7.16 Simple shunt resistor circuit.

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- Current output sensor
  - $E_o = I\,R_{shunt}$ for I = 4 – 20 M_A
- A 500Ω shunt provides 2-10V signal for 10V analog input.

Data Acquisition Boards

- Common name DAS I/O
- Available as expansion plug-in-boards or as PCMCIA cards
  - A/D – analog in
  - D/A – analog out
  - Digital I/O
  - Counter-timer ports
- A/D amplifier gain is set either externally by jumpers or internally in software.
- Gains range from 0.5 to 1000
- Resolution = $E_{FSR}/G(2^m)$ = $10V/(1000*2^{12})$ = 2.44 MV

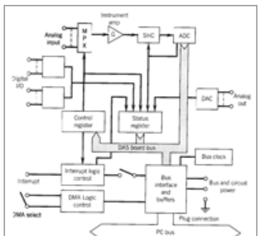

Figure 7.20 Typical layout for a data-acquisition plug-in board.

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Data Acquisition Boards

- PCMCIA – mostly for laptops, while expansion boards are for desktops
- The board shown has a 12-bit successive approximation A/D converter with 8-9 ms conversion
- Sampling rates up to 100,000 hz and higher
  - (0 to 10V or ± 5V)

Analog Inputs

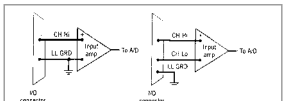
- <u>Single-ended connections</u> – use one signal line (+-high) that is measured relative to ground (gnd). No local ground and short wires due to EMI noises.
- <u>Differential-ended connections</u> – allows voltage difference between two distinct input signals. High (+) and low (-) signals are isolated from ground. It is called a floating input because the difference is between + and -, not ground. Good for low voltage.

Analog Inputs

- When signals from various instruments are used, differential-ended connections are usually required.
- DATA Acquisition is triggered by software, external pulse, or on-board clock.

Analog Inputs

- Single-ended connections are suitable only when all analog signals in the system can be made relative to a common ground
- EMI – Electro-magnetic interference
- For low level measurements a $10^k – 100^k$ Ω resistor should be connected between low (-) and ground at the DAS board.



(a) Single-ended connection (b) Differential-ended connection

Figure 7.23 Single- and differential-ended connections.

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