ECE 454 and ECE 554 Supplemental reading

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Collection edited by: Thad Welch

Content authors: Don Johnson, Minh Do, C. Burrus, Roy Ha, Michael Haag, Nguyen Huu Phuong, Richard Baraniuk, Melissa Selik, Nasser Kehtarnavaz, Philipos Loizou, Mohammad Rahman, Anders Gjendemsjø, Robert Nowak, Justin Romberg, Stephen Kruzick, Mark Davenport, Tuan Do-Hong, Ricardo Radaelli-Sanchez, Catherine Elder, Ivan Selesnick, Benjamin Fite, Douglas Jones, Swaroop Appadwedula, Matthew Berry, Mark Haun, Jake Janevitz, Michael Kramer, Dima Moussa, Daniel Sachs, and Brian Wade

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ECE 454 and ECE 554 Supplemental reading

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Chapter 1. ECE 454/ECE 554 Supplemental Reading for Chapter 1

1.1. Introduction to Digital Signal Processing*

Not only do we have analog signals --- signals that are real- or complex-valued functions of a continuous variable such as time or space --- we can define **digital** ones as well. Digital signals are **sequences**, functions defined only for the integers. We thus use the notation $s(n)$ to denote a discrete-time one-dimensional signal such as a digital music recording and $s(m, n)$ for a discrete-"time" two-dimensional signal like a photo taken with a digital camera. Sequences are fundamentally different than continuous-time signals. For example, continuity has no meaning for sequences.

Despite such fundamental differences, the theory underlying digital signal processing mirrors that for analog signals: Fourier transforms, linear filtering, and linear systems parallel what previous chapters described. These similarities make it easy to understand the definitions and why we need them, but the similarities should not be construed as "analog wannabes." We will discover that digital signal processing is **not** an approximation to analog processing. We must explicitly worry about the fidelity of converting analog signals into digital ones. The music stored on CDs, the speech sent over digital cellular telephones, and the video carried by digital television all evidence that analog signals can be accurately converted to digital ones and back again.

The key reason why digital signal processing systems have a technological advantage today is the **computer**: computations, like the Fourier transform, can be performed quickly enough to be calculated as the signal is produced, [1] and programmability means that the signal processing system can be easily changed. This flexibility has obvious appeal, and has been widely accepted in the marketplace. Programmability means that we can perform signal processing operations impossible with analog systems (circuits). We will also discover that digital systems enjoy an **algorithmic** advantage that contributes to rapid processing speeds: Computations can be restructured in non-obvious ways to speed the processing. This flexibility comes at a price, a consequence of how computers work. How do computers perform signal processing?

1.2. Introduction to Fundamentals of Signal Processing*

What is Digital Signal Processing?

To understand what is **Digital Signal Processing (DSP)** let's examine what does each of its words mean. "**Signal**" is any physical quantity that carries information. "**Processing**" is a series of steps or operations to achieve a particular end. It is easy to see that **Signal Processing** is used everywhere to extract information from signals or to convert information-carrying signals from one form to another. For example, our brain and ears take input speech signals, and then process and convert them into meaningful words. Finally, the word "**Digital**" in Digital Signal Processing means that the process is done by computers, microprocessors, or logic circuits.

The field DSP has expanded significantly over that last few decades as a result of rapid developments in computer technology and integrated-circuit fabrication. Consequently, DSP has played an increasingly important role in a wide range of disciplines in science and technology. Research and development in DSP are driving advancements in many high-tech areas including telecommunications, multimedia, medical and scientific imaging, and human-computer interaction.

To illustrate the digital revolution and the impact of DSP, consider the development of digital cameras. Traditional film cameras mainly rely on physical properties of the optical lens, where higher quality requires bigger and larger system, to obtain good images. When digital cameras were first introduced, their quality were inferior compared to film cameras. But as microprocessors become more powerful, more sophisticated DSP algorithms have been developed for digital cameras to correct optical defects and improve the final image quality. Thanks to these developments, the quality of consumer-grade digital cameras has now surpassed the equivalence in film cameras. As further developments for digital cameras attached to cell phones (cameraphones), where due to small size requirements of the lenses, these cameras rely on DSP power to provide good images. Essentially, digital camera technology uses computational power to overcome physical limitations. We can find the similar trend happens in many other applications of DSP such as digital communications, digital imaging, digital television, and so on.

In summary, DSP has foundations on Mathematics, Physics, and Computer Science, and can provide the key enabling technology in numerous applications.

Overview of Key Concepts in Digital Signal Processing

The two main characters in DSP are **signals** and **systems**. A **signal** is defined as any physical quantity that varies with one or more independent variables such as time (one-dimensional signal), or space (2-D or 3-D signal). Signals exist in several types. In the real-world, most of signals are **continuous-time** or **analog signals** that have values continuously at every value of time. To be processed by a computer, a continuous-time signal has to be first **sampled** in time into a **discrete-time signal** so that its values at a discrete set of time instants can be stored in computer memory locations. Furthermore, in order to be processed by logic circuits, these signal values have to be **quantized** in to a set of discrete values, and the final result is called a **digital signal**. When the quantization effect is ignored, the terms discrete-time signal and digital signal can be used interchangeability.

In signal processing, a **system** is defined as a process whose input and output are signals. An important class of systems is the class of **linear time-invariant** (or **shift-invariant**) **systems**. These systems have a remarkable property is that each of them can be completely characterized by an **impulse response function** (sometimes is also called as **point spread function**), and the system is defined by a **convolution** (also referred to as a **filtering**) operation. Thus, a linear time-invariant system is equivalent to a (linear) **filter**. Linear time-invariant systems are classified into two types, those that have **finite-duration impulse response (FIR)** and those that have an **infinite-duration impulse response (IIR)**.

A signal can be viewed as a **vector** in a **vector space**. Thus, **linear algebra** provides a powerful framework to study signals and linear systems. In particular, given a vector space, each signal can be represented (or expanded) as a **linear combination of elementary signals**. The most important **signal expansions** are provided by the **Fourier transforms**. The Fourier transforms, as with general transforms, are often used effectively to transform a problem from one domain to another domain where it is much easier to solve or analyze. The two domains of a Fourier transform have physical meaning and are called the **time domain** and the **frequency domain**.

Sampling, or the conversion of **continuous-domain real-life signals** to **discrete numbers** that can be processed by computers, is the essential bridge between the analog and the digital worlds. It is important to understand the connections between signals and systems in the real world and inside a computer. These connections are convenient to analyze in the frequency domain. Moreover, many signals and systems are specified by their **frequency characteristics**.

Because any **linear time-invariant system** can be characterized as a **filter**, the design of such systems boils down to the design the associated filters. Typically, in the **filter design** process, we determine the coefficients of an FIR or IIR filter that closely approximates the desired **frequency response** specifications. Together with Fourier transforms, the **z-transform** provides an effective tool to analyze and design digital filters.

In many applications, signals are conveniently described via **statistical models** as **random signals**. It is remarkable that optimum linear filters (in the sense of **minimum mean-square error**), so called **Wiener filters**, can be determined using only **second-order statistics** (**autocorrelation** and **crosscorrelation** functions) of a **stationary process**. When these statistics cannot be specified beforehand or change over time, we can employ **adaptive filters**, where the filter coefficients are adapted to the signal statistics. The most popular algorithm to adaptively adjust the filter coefficients is the **least-mean square (LMS)** algorithm.

1.3. m09 - An Overview of Discrete-Time Signals*

[Discrete-Time Signals]Discrete-Time Signals

Although the discrete-time signal x(n) could be any ordered sequence of numbers, they are usually samples of a continuous-time signal. In this case, the real or imaginary valued mathematical function x(n) of the integer n is not used as an analogy of a physical signal, but as some representation of it (such as samples). In some cases, the term digital signal is used interchangeably with discrete-time signal, or the label digital signal may be use if the function is not real valued but takes values consistent with some hardware system.

Indeed, our very use of the term ``discrete-time" indicates the probable origin of the signals when, in fact, the independent variable could be length or any other variable or simply an ordering index. The term ``digital" indicates the signal is probably going to be created, processed, or stored using digital hardware. As in the continuous-time case, the Fourier transform will again be our primary tool.

Notation has been an important element in mathematics. In some cases, discrete-time signals are best denoted as a sequence of values, in other cases, a vector is created with elements which are the sequence values. In still other cases, a polynomial is formed with the sequence values as coefficients for a complex variable. The vector formulation allows the use of linear algebra and the polynomial formulation allows the use of complex variable theory.

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