

Signal Processing for Intelligent Sensor Systems

DAVID C. SWANSON

Signal Processing and Communications Series



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DAVID C. SWANSON

*The Pennsylvania State University
University Park, Pennsylvania*



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Series Introduction

Over the past 50 years, digital signal processing has evolved as a major engineering discipline. The fields of signal processing have grown from the origin of fast Fourier transform and digital filter design to statistical spectral analysis and array processing, and image, audio, and multimedia processing, and shaped developments in high-performance VLSI signal processor design. Indeed, there are few fields that enjoy so many applications—signal processing is everywhere in our lives.

When one uses a cellular phone, the voice is compressed, coded, and modulated using signal processing techniques. As a cruise missile winds along hillsides searching for the target, the signal processor is busy processing the images taken along the way. When we are watching a movie in HDTV, millions of audio and video data are being sent to our homes and received with unbelievable fidelity. When scientists compare DNA samples, fast pattern recognition techniques are being used. On and on, one can see the impact of signal processing in almost every engineering and scientific discipline.

Because of the immense importance of signal processing and the fast-growing demands of business and industry, this series on signal processing serves to report up-to-date developments and advances in the field. The topics of interest include but are not limited to the following:

- Signal theory and analysis
- Statistical signal processing
- Speech and audio processing
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- Signal processing for communications
- Signal processing architectures and VLSI design

I hope this series will provide the interested audience with high-quality, state-of-the-art signal processing literature through research monographs, edited books, and rigorously written textbooks by experts in their fields.

K. J. Ray Liu

Preface

Signal Processing for Intelligent Sensor Systems covers a broad range of topics that are essential to the design of intelligent autonomous computing systems. A unified approach is presented linking data acquisition, system modeling, signal filtering in one and two dimensions, adaptive filtering, Kalman filtering, system identification, wavenumber processing, pattern recognition, sensor systems, and noise cancellation techniques. Together these topics form the technical basis for the state of the art in radar, sonar, medical and machinery health diagnosis and prognosis, and “smart” sensor systems in general. Applications are given throughout the book in the areas of passive remote sensing, active radar and sonar, digital image processing, tracking filters, acoustic imaging and diagnostics, and wavenumber filters. Additional references and example problems are provided in each topic area for further research by the reader. This book presents adaptive signal processing from a physical, rather than mathematical, point of view, with emphasis on application to intelligent sensor systems. Engineers and scientists working on the development of advanced sensor and control systems should find this text useful in bringing together the required topics. Unifying these topics in a single book allows the uncertainties from the basic sensor detection elements to be propagated through each adaptive signal processing topic to produce meaningful metrics for overall system performance. Many signal processing applications require knowledge of a wide range of adaptive signal processing topics for developing such systems. This text pays specific attention to the underlying physics behind the signal processing application, and, where appropriate, examines the signal processing system as a physical device with physical laws to be examined and exploited for science.

The text is well suited for senior undergraduate and graduate students in science and engineering as well as professionals with a similar background. Some prior knowledge of digital signal processing, statistics, and acoustics/field theory would be helpful to readers, but an undergraduate level of understanding in complex matrix algebra, field theory, and Fourier-LaPlace transforms should be sufficient background. From this starting point, the book develops basic adaptive signal processing principles and applies them to the problems such as adaptive

beamforming, system identification, and data tracking. The benefit of using this book is that one comes away with a more global view of adaptive signal processing in the context of a “smart” sensor and/or control system as well as a detailed understanding of many state-of-the-art techniques. While adaptive algorithms extract information from input data according to optimization schemes such as least-squared error, they can also be used to “intelligently” adapt the sensor system to the environment. Adaptive systems that optimize themselves based on command inputs and the sensor and/or actuator environment represent the most advanced “sentient” systems under development today. The term “sentient” means “having the five senses” and the associated awareness of the environment. Sensor technology in the 21st century will no doubt achieve sentient processing and this text is aimed at providing an interdisciplinary groundwork toward this end.

A simple example of an environment-sensitive adaptive algorithm is automatic exposure, focus, and image stabilization on many commercially available video cameras. When coupled to a digital frame-grabber and computer vision system, adaptive image processing algorithms can be implemented for detecting, say, product defects in an assembly line. Adaptive systems designed to detect specific “information patterns” in a wide range of environments are often referred to as automatic target recognition (ATR) systems, especially in defense applications. The ATR problem has proved to be one of the most difficult and intriguing problems in adaptive signal processing, especially in the computer vision area. Acoustic ATRs have enjoyed some degree of success in the areas of machine condition vibration monitoring, sound navigation and ranging (SONAR), and speech recognition. In medical imaging, adaptive image processing systems can produce remarkable measurements of bone density, geometry, or tissue properties, but the actual end recognition is currently done by people, not by machines. It is curious to note that while a toddler can easily find a partially obscured toy in a full toybox, it’s not an easy task for even the most sophisticated computer vision system, due to the complexity of the signal separation problem. It is likely that people will always be “in the loop” in making critical decisions based on information that intelligent sensor systems provide because of the inherent intelligence and personal responsibility human beings can display. But as technology progresses, we should certainly expect many noncritical sensor-controller applications to be fully automated at significant levels.

What we can do today is build adaptive systems in which the accepted laws of physics and mathematics are exploited in computer algorithms that extract information from sensor data. The information is then used to do useful things such as: optimize the sensor configuration for maximum resolution; recognize primitive patterns in the input data and track the patterns and information to determine statistical trends; and logically assemble the information to test and score hypotheses. The results of the adaptive processing can in some instances lead to a completely automatic recognition system. However, since many of the current ATR applications are life-critical (such as military targeting, medicine, and machine or structure prognosis), the idea of eliminating the “man-in-the-loop” is being replaced by the idea of making the man-in-the-loop smarter and more efficient. This text is concerned with the mathematics of the adaptive algorithms and their relationship to the underlying physics of the detection problems at hand. The mechanical control systems on automobile engines (choke, spark-plug timing, etc.), mechanical

audio recordings, and many household appliances changed to digital systems only in the last decade or so. Most industrial process controls and practically all military control and communication systems are digital, at least in part. The reasons for this unprecedented proliferation of basic digital system technology are not just the increased precision and sophistication of digital systems. Digital control systems are now far cheaper to manufacture and have better repeatability and reliability than their analog computers. One could argue that a new industrial revolution is already underway, in which the machines and practices from the previous revolution are being computerized, optimized, and reinvented with “machine intelligence.”

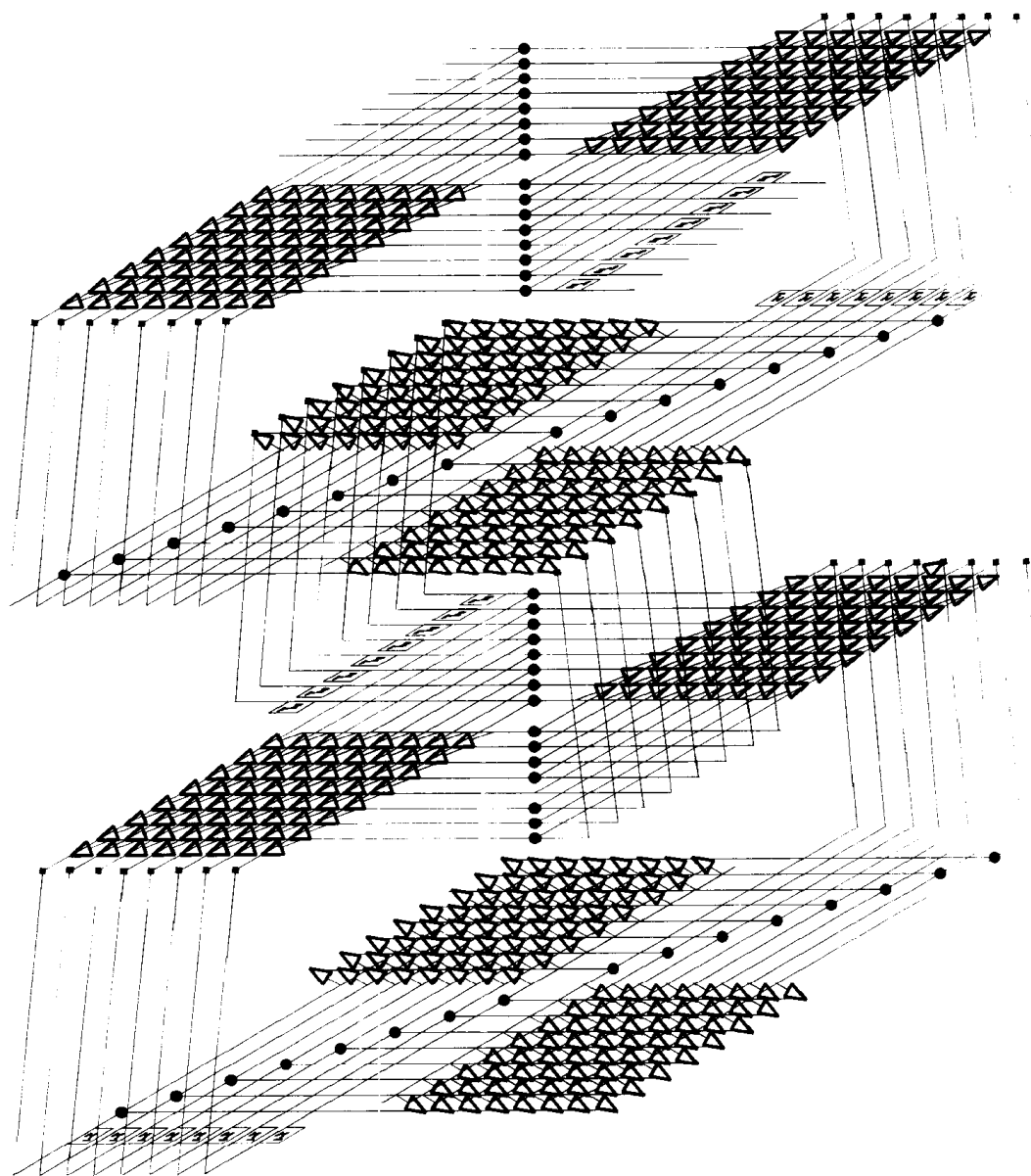
The text is organized into five main parts. Fundamentals, frequency domain processing, adaptive filtering, wavenumber systems, and signal processing applications. Together, these five parts cover the major system operations of an intelligent sensor system. However, the emphasis is on applied use, physics, and data confidence rather than just applied mathematical theory. There are many available texts listed in each chapter’s bibliography that can supply the reader with sufficient detail beyond that offered here. By focusing on the essential elements of measurement, filtering, detection, and information confidence metrics, this book provides a good foundation for an intelligent sensor system. In the future, sensor system engineering will need to be more accessible to the nonelectrical engineering discipline, while electrical engineers will need a stronger applied background in physics and information processing. Bridging this gap in an interdisciplinary approach has been the main challenge of preparing this text. The student with a strong signal processing background should be advised that even in the fundamentals chapter there are subtle physical application points to be learned. Furthermore, students relatively new to signal processing should not be intimidated by the advanced signal processing topics such as lattice filters and adaptive beamforming. These advanced topics are explained in a very straightforward and practical manner for the essential techniques, leaving many of the more narrow techniques to presentations given in other texts.

The prime objective of this book is to organize the broad scope of adaptive signal processing into a practical theory for the technical components of smart machines. The longer a normal human being works in the area of developing a computing system’s eyes, ears, motor control, and brains, the more incredible biological life appears to be. It’s almost funny how our most powerful supercomputers, with a throughput of over billions of operations per second, have the real-time neural network capacity of a slug (okay, a smart slug). Computer programs and algorithms teach us a great deal about our own thought processes, as well as how our imaginations lead us to both inventiveness and human error. Perhaps the most important thing building machine intelligence can teach us is what an amazing gift we all have: to be human with the ability to learn, create, and communicate.

David C. Swanson

Acknowledgments

Preparation of this text began in earnest in the fall of 1993 and took more than five years to complete. There are many reasons that projects such as creating a textbook take longer than originally envisioned, not the least of which is the fact that it is far more interesting to create a textbook than to read and use one. I can acknowledge three main factors that contributed enormously to the content and the required time spent preparing this text. In order of importance, they are my family, my students and colleagues, and my research sponsors. At Penn State's Applied Research Laboratory I have been extremely fortunate to have very challenging applied research projects funded by sponsors willing to pay me to learn. Teaching graduate students with backgrounds in electrical and mechanical engineering, physics, and mathematics in Penn State's Graduate Program in Acoustics not only has been a privilege but has shaped the interdisciplinary approach of this text. As a result of the time demands of applied research and teaching, this text was written in its entirety on a home PC during evenings and weekends. That time reasonably belongs to a very patient wife, Nadine, and three children, Drew, age 6, Anya, age 3, and Erik, age 3 months at the time of this writing. They made things far easier on me than my absences were on them. In particular, I owe Nadine enormous love, gratitude, and respect for tolerating the encumbrance the creation of this book placed on our home life. If you find this text interesting and useful, don't just consider the author's efforts—consider the role of a strong and loving family, outstanding students and colleagues, and challenging applied research projects. Without any one of these three important factors, this text would not have been possible. Finally, the creators of Matlab, WordPerfect, and Corel Draw also deserve some acknowledgment for producing some really outstanding software that I used exclusively to produce this book.



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Part I

Fundamentals of Digital Signal Processing

Sampled Data Systems

Figure 1 shows a basic general architecture which can be seen to depict most adaptive signal processing systems. The number of inputs to the system can be very large, especially for image processing sensor systems. Since adaptive signal processing system is constructed using a computer, the inputs generally fall into the categories of analog “sensor” inputs from the physical world and digital inputs from other computers or human communication. The outputs also can be categorized into digital information such as identified patterns, and analog outputs which may drive actuators (active electrical, mechanical, and/or acoustical sources) to instigate physical *control* over some part of the outside world. In this chapter we examine the basic constructs of signal input, processing using digital filters, and output. While these very basic operations may seem rather simple compared to the algorithms presented later in the text, careful consideration is needed to insure a high fidelity adaptive processing system. For example, Figure 1 shows the adaptive process controlling the analog input and output gains. This technique is relatively straightforward to implement and allows high fidelity signal acquisition and output over a wide dynamic range. With a programmed knowledge-base of rules for acceptable input and output gains, the adaptive system can also decide if a transducer channel is broken, distorted, or operating normally. Therefore, we will need to pay close attention to the fundamentals of sampling analog signals and digital filtering. The next chapter will focus on fundamental techniques for extracting information from the signals.

Consider a transducer system which produces a voltage in response to some electromagnetic or mechanical wave. In the case of a microphone, the transducer sensitivity would have units of volts/Pascal. For the case of a video camera pixel sensor, it would be volts per lumen/m², while for an infrared imaging system the sensitivity might be given as volts per °K. In any case, the transducer voltage is conditioned by filtering and amplification in order to make the best use of the A/D convertor system. While most adaptive signal processing systems use floating-point numbers for computation, the A/D convertors generally produce fixed-point (integer) digital samples. The integer samples from the A/D are con-