PREFACE

This book is intended for the reader who is interested in learning about *digital signal processing* (DSP) with an emphasis on audio signals. It is an introductory book that covers the fundamentals in DSP, including important theories and applications related to sampling, filtering, sound synthesis algorithms and sound effects, time and frequency-domain analysis, and various topics central to the study of signal processing. The book has been designed to present DSP in a practical way, concentrating on audio and musical signals by addressing the concepts, mathematical foundations, and associated musical and sound examples to provide an appealing and, dare I say, *intuitive* way to become familiar and comfortable with the materials presented. Throughout the chapters, we try to follow the L^3 (look, listen, learn) model as much as possible — a pedagogical approach that my brother has engrained in me over many years. A bulk of the concepts and diagrams presented in this book have accompanying MATLAB® code that can be downloaded from http://music.princeton.edu/∼park/dspBook (details can be found in the appendix). During the course of the book, important theories and ideas are laid out where questions that pertain to the "why" are given special attention, especially when new materials are introduced for the first time. This approach is often rather lacking in standard engineering books, which at times tend to dwell heavily on very abstract ideas with little reference to practical situations and applications. At the same time, the book also tries to fill in some of the holes on the other end of the spectrum — the mathematical side of the coin which often is explained fleetingly without much depth in non-engineering-centric books related to signal processing. In this book, the goal is to focus on the "why" *and* the "how," with the ultimate aim to help the reader learn and understand the art of digital signal processing. As far as the reader's mathematical background is concerned, the majority of the concepts will be manageable having knowledge of algebra. The audience for this book therefore encompasses a wide range of readers, including musicians, composers, engineers, computer scientists, programmers, and undergraduate/graduate students in various disciplines.

Each chapter is structured to include the presentation of concepts, mathematical foundations, practical examples, lots of pictures/plots, and ends with a section introducing compositional and musical examples in relation to the materials covered. The book starts in the time-domain with the familiar sine tone along with the introduction of basics in acoustics and human hearing limitations. This sets up the reader in understanding why and how the computer is used in representing and manipulating analog sound by an inferior digital counterpart without compromising quality. Artifacts of sampling followed by quantization, bit resolution, and the so-called CD-quality standard pave the way to important concepts in digital audio/sound including the ADSR, windowing, RMS envelope detection, wavetable synthesis and sample rate conversion with musical examples such as *Queen's Another One Bites the Dust* and time-domain time-stretching and compression seen in William Schottstaedt's *Leviathan*.

All through the book, especially in the beginning stages, DSP is introduced to the reader in a somewhat *subconscious* manner — via concentration and presentation of fun and interesting sound synthesis and analysis examples (within the context of DSP and the mathematics involved). For example, we start off by introducing sine waves, amplitude modulation, frequency modulation concepts, and the *clapping technique* to explain the impulse response which will help us get ready for the theory and practical use of convolution in audio. By the middle of the book the reader will hopefully be armed with enough confidence to get into the nuts and bolts of signal processing essentials — diving into topics such as difference equations, frequency response, z-transforms, and filters. As usual, DSP pertinent musical examples will follow the materials being presented at the end of each chapter and Matlab® implementations of topics such as comb-filtering and its kinship to physical modeling of plucked strings is also explored. In the final chapters we will fully venture into the frequency-domain, focusing on the Fourier transform with emphasis on the DFT (discrete Fourier transform) and its applications in various areas in computer music. We will at this point also revisit concepts introduced at the beginning of the book, such as up-sampling, down-sampling, decimation, harmonic distortion/dithering, and the Nyquist theorem to reinforce our knowledge of these very fundamental and important signal processing concepts by reviewing and viewing the same theories from different angles.

As a matter of fact, the practice of revisiting previously learned materials is a reoccurring theme throughout the book. This will hopefully further help us strengthen our understanding of principles presented during the course of the book. The final chapter discusses a number of classic vocoder algorithms and finishes off the book with an overview of research topics in the field of digital audio and computer/electro-acoustic music.