Part B

Signal Pr

Part B Signal Processing

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This book section deals with signal processing techniques from the perspective of computational systematic musicology. Recent developments in the field have opened up major opportunities for researchers with technical backgrounds to pursue new ways of analyzing music using digital signal processing techniques. Traditionally, the primary work of systematic musicology focused on the manual analysis of music scores. In classical music, where the field of musicology originates from, the concept of work is based on music scores. While the score defines the work of a composer, the performance of these works is considered interpretation and thus of secondary interest. In contrast, the concept of work for many non-Western music genres as well as pop music and jazz is mainly defined through the actual sound of a performance or recording. Different rock bands, for example, use exactly the same chord progression, but the sound of these progressions is so unique that one often can tell the song before the singer starts. One cannot understand the fundamentals of this culture by just analyzing scores, and signal processing techniques are now used as an objective method to analyze the tempo, tonality, and timbre of this music, among many other features.

The underlying music analysis tools, which were introduced to better understand music, have meanwhile found their way into the music industry. Music streaming services use them to classify songs according to user preferences to maintain a competitive edge. The shift from storing music on individual media (e.g., vinyl, CDs) to large databases makes big data analyses possible, as media can now be accessed digitally. Instead of analyzing a few works by hand, general musical trends can be investigated by automatically analyzing a very large corpus of work using batch processes.

Meanwhile, signal processing tools are seeing an increased use in investigations of classical music as well, because there is an emerging focus on understanding the interpretation of works. This trend has to do with the fact that the historic body of classical music works no longer grows, and many major compositions have been analyzed multiple times. In contrast, the practice of interpreting these works is still a developing culture, leading to new opportunities to analyze works.

Moreover, signal processing methods have become important in organology, the science of musical instruments. It is common practice to develop a mathematical model of a musical instrument in order to understand how it operates and understand why certain trends occur. In this book section, we will cover one-dimensional approaches, where the acoustic propagation is simulated along one axis, for example along the resonator of a wind instrument, as well as three-dimensional methods, where the sound can travel across all three spatial dimensions.

Most chapters in this section deal with acoustical analysis methods, but techniques to process or synthesize sounds are also covered. Historically, systematic musicology has solely worked with passive methods, where the researcher merely observes his or her object of study. Meanwhile active research schemes have become more popular; for example, in Simha Arom's approach to experimental ethnomusicology he performs together with indigenous musicians to examine if his theories hold up to a practical test. Digital signal processing and synthesis methods can be instrumental in active research methods, for example in the virtual recreation of a historic music performance in a concert venue that no longer exists. This can be accomplished using wave field synthesis techniques in order to understand why the composer chose a given instrumentation in this venue.

The book section is organized as follows:

Chapter 12 covers the fundamentals of music studio technology. It deals with the whole studio production chain from capturing the signals using different microphone techniques, storing them on various analog and digital media, to different methods of playing back the signals using standard loudspeaker arrangements. In addition, the chapter outlines the complex signal processing techniques in studios that formed the artistic sound concepts of popular music. Practical introductions to digital signals, cables and connectors, and music synthesizers complete this reference on studio technology.

Chapter 13 deals with delay lines and digital waveguides. Both methods are used to understand sound propagation along a one-dimensional axis. This chapter covers the fundamentals including finite impulse response (FIR) filters, linear interpolation, sound reflection, lossy wave propagation, and comb filter effects. After explaining mathematical fundamentals, the chapter focuses on the acoustic simulation of a plucked string instrument by simulating the acoustic behavior of a damped string that is terminated at both ends after being plucked in its initial state.

Chapter 14 gives a mathematical introduction to the fundamental concepts of convolution, Fourier analysis, and cross-correlation. All three concepts are important signal processing methods used to analyze and synthesize sound. Convolution is used to simulate the transformation of a signal by a linear time-invariant acoustical system, Fourier analysis is used to understand how a signal behaves in terms of frequency, and cross-correlation is used to compare the similarity of two systems.

Chapter 15 describes signal processing methods used to separate individual sound sources from a mixture. The techniques described in this chapter are fundamental to analysis of the complex sounds of ensemble music, where more than one musical instrument or voice is involved. Within the chapter, the repeating pattern extraction technique (REPET) is introduced, which detects periodic functions, e.g., in the form of beat patterns, establishes a model of the repeated patterns, and then segregates these pattern from the acoustic background using time/frequency masks. Another focus of this chapter is multipitch streaming to extract pitch contours of overlaid instruments. The contours are used to construct harmonic masks to segregate the individual instruments. The chapter concludes with the description of techniques to align the sound of a recorded performance to the underlying musical score.

Chapter 16 covers optical recognition methods to automatically extract music scores. Optical music recognition (OMR) is an instrumental process for archiving historic scores based on a symbolic notation. OMR provides a basis for automatic music analysis using the converted scores, where the extracted scores can be easily sonified using a music synthesizer. OMR has its roots in optical text recognition, but is affected by additional challenges that mainly result from the circumstance that music symbols, e.g., notes and staff lines, overlap, while text characters are generally isolated. The chapter provides an overview of different techniques to overcome the resulting technical challenges.

Chapter 17 deals with adaptive digital music systems. While it is often easier and therefore advantageous to describe a system mathematically using linear time-invariant systems, there are clear limitations to this approach. In general, musical instruments are nonlinear, time-variant systems, for example the bore of a wind instrument using keys to produces different pitches. In some cases, time-variant systems can be approximated using a sequence of time-invariant systems, but in many other cases only adaptive systems can simulate the behavior of time-variant systems with sufficient complexity. This chapter focuses on a number of techniques including Kalman filters and autoregressive moving average (ARMA) processes to provide expressive control of digital music systems in the joint time/frequency space.

Chapter 18 covers the fundamentals of wave field synthesis (WFS). WFS is an advanced method that simulates how sound propagates in three-dimensional space. It builds on the Huygens–Fresnel principle, which states that any given wave front can be described through the superposition of elementary spherical sound sources, and the Kirchhoff–Helmholtz integral. The latter demonstrates that by controlling the pressure and velocity along the boundaries of an enclosed space any three-dimensional sound field can be synthesized within it. This assumption works as long as the volume inside the enclosure is source-free. The chapter introduces the theoretical framework for WFS and discusses the practical approach and limitations of rendering sound fields using a two-dimensional array of loudspeakers as elementary sound-pressure sources.

Chapter 19 introduces the finite-difference method (FDM) in the context of musical acoustics. The FDM is a popular tool to describe a set of differential equations by discretizing a space and approximating these equations with difference functions. This method can be used for example by discretizing a geometrical model of a room using a constantly spaced grid (e.g., 10 cm along all three Cartesian coordinates). The difference equations are then solved numerically at the grid positions. The method is a useful tool where the environments are too complex to solve analytically and the dimensions for the desired frequencies are too small to use geometrical methods like ray tracing. This is the case for low frequencies in a small room or for the geometry of many musical instruments that are discussed in this chapter.

Chapter 20 discusses the fundamental problems, solutions, and applications of real-time sound processing. Classical applications where real-time processing is needed are digital music keyboards, live sound effect processors, and virtual reality systems. The chapter covers the main challenges for the design of real-time algorithms to minimize latency and cost of the computational operations. Common real-time algorithms will be explained based on software for field programmable gate array (FPGA) hardware of mobile devices. Mobile devices are becoming increasingly important for musicological research in the form of mobile field recorders, digital tuners and loudness meters, among other tools.