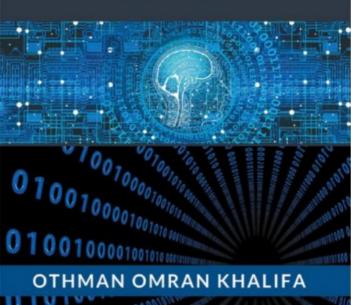
# APPLIED DIGITAL SIGNAL PROCESSING AND APPLICATIONS



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# Preface

Due to the rapid development of technologies, digital information playing a key role in our daily life. In the past signal processing appeared in various concepts in more traditional courses where the analog and discrete components were used to achieve the various objectives. However, in the 21th century, with the rapid growth of computing power in terms of speed and memory capacity and the intervention of artificial intelligent, machine /deep learning algorithms introduces a tremendous growth in signal processing applications. Therefore, digital signal processing has become such a critical component in contemporary science and technology that many tasks would not be attempted without it. It is a truly interdisciplinary subject that draws from synergistic developments involving many disciplines. The developers should be able to solve problems with an innovation, creativity and active initiators of novel ideas. However, the learning and teaching has been changed from conventional and tradition education to outcome based education. Therefore, this book prepared on a Problem-based approach and outcome based education strategies. Where the problems incorporate most of the basic principles and proceeds towards implementation of more complex algorithms. Students required to formulate in a way to achieve a well-defined goals under the guidance of their instructor.

This book follows a holistic approach and presents discrete-time processing as a seamless continuation of continuous-time signals and systems, beginning with a review of continuoustime signals and systems, frequency response, and filtering. The synergistic combination of continuous-time and discrete-time perspectives leads to a deeper appreciation and understanding of DSP concepts and practices.

This book is organized in Ten chapters as follows: Chapter One, introduces the basic terminology of signals in digital signal processing, Classification of signals as well as the elementary signal are explained in detail. Chapter Two describes the concept of systems and characterize and analyze the properties of Discrete systems. Chapter Three covers the sampling process, Quantization, coding and reconstruction of signals. Chapter Four introduces the properties of discrete signals and systems. Chapter Five introduces the z-transform and difference equations and its applications. Chapter Six explains the frequency analysis of Discrete Signals and Systems, Frequency Response of Systems and convolution via frequency domain. Chapter Seven devoted for Discrete Fourier transform. Chapter Eight deals with various methods used in Digital filters design. Chapter Nine introduces the wavelet transforms, Multiresolution Analysis and some applications of discrete wavelet transform. Chapter Ten deals with adaptive signal processing and covers Wiener filter, LMS algorithms, RLS algorithms and ends with applications of adaptive filters.

Author Othman Omran Khalifa

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