Chapter I

Introduction to Audio and Speech Signal Processing

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Abstract

The development of very efficient digital signal processors has allowed the implementation of high performance signal processing algorithms to solve an important amount of practical problems in several engineering fields, such as telecommunications, in which very efficient algorithms have been developed to storage, transmission, and interference reductions; in the audio field, where signal processing algorithms have been developed to enhancement, restoration, copy right protection of audio materials; in the medical field, where signal processing algorithms have been efficiently used to develop hearing aids systems and speech restoration systems for alaryngeal speech signals. This chapter presents an overview of some successful audio and speech signal processing algorithms, providing to the reader an overview of this important technology, some of which will be analyzed with more detail in the accompanying chapters of this book.

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Introduction

The advances of the VLSI technology have allowed the development of high performance digital signal processing (DSP) devices, enabling the implementation of very efficient and sophisticated algorithms, which have been successfully used in the solution of a large amount of practical problems in several fields of science and engineering. Thus, signal processing techniques have been used with great success in telecommunications to solve the echo problem in telecommunications and teleconference systems (Amano, Perez-Meana, De Luca, & Duchen, 1995), to solve the inter-symbol interference in high speed data communications systems (Proakis, 1985), as well as to develop efficient coders that allow the storage and transmission of speech and audio signals with a low bit rate keeping at the same time a high sound quality (Bosi & Golberg, 2002; Kondoz, 1994). Signal processing algorithms have also been used for speech and audio signal enhancement and restoration (Childers, 2000; Davis, 2002) to reduce the noise produced by air conditioning equipment and motors (Kuo & Morgan, 1996), and so forth, and to develop electronic mufflers (Kuo & Morgan, 1996) and headsets with active noise control (Davis, 2002). In the educational field, signal processing algorithms that allow the time scale modification of speech signals have been used to assist the foreign language students during their learning process (Childers, 2000). These systems have also been used to improve the hearing capability of elder people (Davis, 2002).

The digital technology allows an easy and error free reproduction of any digital material, allowing the illegal reproduction of audio and video material. Because this fact represents a huge economical loss for the entertainment industry, many efforts have been carried out to solve this problem. Among the several possible solutions, the watermarking technology appears to be a desirable alternative for copyright protection (Bassia, Pitas, & Nikoladis, 2001; Bender, Gruhl, Marimoto, & Lu, 1996). As a result, several audio and speech watermarking algorithms have been proposed during the last decade, and this has been a subject of active research during the last several years. Some of these applications are analyzed in the remaining chapters of this book.

This chapter presents an overview of signal processing systems to storage, transmission, enhancement, protection, and reproduction of speech and audio signals that have been successfully used in telecommunications, audio, access control, and so forth.

Adaptive Echo Cancellation

A very successful speech signal processing application is the adaptive echo cancellation used to reduce a common but undesirable phenomenon in most telecommunications systems, called echo. Here, when mismatch impedance is present in any telecommunications system, a portion of the transmitted signal is reflected to the transmitter as an echo, which represents an impairment that degrades the system quality (Messerschmitt, 1984). In most telecommunications systems, such as a telephone circuit, the echo is generated when the long distant portion consisting of two one-directional channels (four wires) is connected with a bidirectional channel (two wires) by means of a hybrid transformer, as shown in Figure 1. If the hybrid impedance is perfectly balanced, the two one-directional channels are...
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