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Signal Processing of Speech

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Macmillan New Electronics
Introductions to Advanced Topics

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First published 1993 by
THE MACMILLAN PRESS LTD
Houndmills, Basingstoke, Hampshire RG21 2XS
and London
Companies and representatives
throughout the world

ISBN 978-0-333-51922-6 ISBN 978-1-349-22599-6 (eBook)
DOI 10.1007/978-1-349-22599-6

A catalogue record for this book is available
from the British Library

***To
Mary,
Niall, Gareth, Niamh and Darren***

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Series Editor's Foreword

The rapid development of electronics and its engineering applications ensures that new topics are always competing for a place in university and polytechnic courses. But it is often difficult for lecturers to find suitable books for recommendation to students, particularly when a topic is covered by a short lecture module, or as an 'option'.

Macmillan New Electronics offers introductions to advanced topics. The level is generally that of second and subsequent years of undergraduate courses in electronic and electrical engineering, computer science and physics. Some of the authors will paint with a broad brush; others will concentrate on a narrower topic, and cover it in greater detail. But in all cases the titles in the Series will provide a sound basis for further reading in the specialist literature, and an up-to-date appreciation of practical applications and likely trends.

The level, scope and approach of the Series should also appeal to practising engineers and scientists encountering an area of electronics for the first time, or needing a rapid and authoritative update.

Paul A. Lynn

Preface

The relatively new area of speech processing has undergone significant advances in the three decades that have passed since the advent of digital computing in the early 1960s. The last few years have witnessed the transfer of speech technology from the research laboratory to the market-place. A variety of commercial speech products is now available for speech synthesis, speech recognition and speech coding. This has been made possible by developments in techniques for speech processing, coupled with the significant advances in microprocessor and digital signal processing (d.s.p.) technology.

The aim of this book is to provide an introduction to the techniques and algorithms used in the design of speech systems. The target readership is students studying speech processing in final year undergraduate and postgraduate degree courses in Electrical and Electronic Engineering, Physics and Computing Science. It is hoped that the book will also be of use to anyone beginning to work or carry out research in speech technology. Readers outside these disciplines may find it suitable for general interest reading.

Speech signal processing depends on certain mathematical, systems and communications principles which have to be understood for a proper appreciation of the area. In writing this book, I have tried to minimise, simplify and illustrate the mathematical content. Speech technology is a multi-disciplinary topic, requiring a knowledge of such diverse areas as signal processing, electronics, computing science, linguistics and physiology. While most of these areas are referred to and briefly described in this book, the emphasis is very firmly on the signal processing aspects of speech technology, reflecting the title and my own personal interests. I regret any imbalance that this may cause, but in a short introductory book, it is impossible to cover everything. When writing, I was constantly aware of having to condense important topics or indeed omit them altogether, because of limited space. I hope that this will not detract too much from the value of the book.

The origin of part of this book is courses of lectures, which I have been giving in final year undergraduate and postgraduate courses in Electrical and Electronic Engineering and Computing Science at the University of Ulster in recent years. During that time, I have received a lot of valuable student feedback which I gratefully acknowledge and which has influenced

my treatment of certain topics. I would like to express my gratitude to Bob Linggard who introduced me to the field of speech processing and who has given me valuable support and encouragement. I am indebted to a number of colleagues, postgraduate and undergraduate students, whose research and project work has made a significant contribution to my understanding of speech signal processing; in particular, I would like to thank David Irvine, Sean Murphy, Neville Ramsey, George Wright, Peter Donnelly, Gary Shannon and Kieran Cooney who also generated some of the diagrams. I should like to acknowledge also the authors of numerous research papers and articles, which I have studied and which have provided me with invaluable insight into the challenging area of speech processing. Unfortunately, in an introductory book of this type, I cannot cite all of them individually.

Finally, I would especially like to thank my wife, Mary, for her constant support and encouragement and who, with one finger, typed the entire manuscript from my illegible scrawl.

F.J. Owens