

Preface

This book *Robust Digital Processing of Speech Signals* represents a result of years of cooperation between the Institute for Applied Mathematics and Electronics and the Department for Automatics of the School of Electrical Engineering, University of Belgrade, dedicated to the research of speech signal phenomena. One of the main conclusions of these extensive investigations has been that the accuracy of the speech generation model always plays the key role, regardless of whether the applied procedure for parameter identification and estimation is used for the purposes of coding, analytical-synthetical transmission, recognition, or for some other goal. It is logical that limitations imposed at this lowest level of speech processing can be hardly corrected at higher levels in the mentioned complex systems for digital processing of speech signals. One of the possible directions toward making a more complex speech model is its robustification regarding the presumed types of excitation signals, which is equivalent to the introduction of a class of nonlinear models and the corresponding criterion functions for parameter estimation. Compared to the general class of nonlinear models, such as various types of neural networks, this class of models possesses good properties of controlled complexity, a possibility to work in “online” mode, as well as a low information volume for the needs of efficient speech encoding and transmission.

The material presented in this book dominantly relies upon the authors’ own results, previously verified through publishing in eminent international science journals. In order to arrive at a comprehensive insight into the subject of robust modeling of speech signal, this monograph has been extended by additional texts dedicated to general considerations of speech modeling, linear predictive analysis, and robust parameter estimation. It is our belief that this book’s readability has been thus improved, and that as such, it may serve both as a specialized textbook and as a monograph.

The text of this book is divided into seven chapters. The first six chapters are dedicated to theoretical considerations, synthesis of robust algorithms, and their experimental evaluation, while the seventh chapter unifies the developed robust methods in various practical problems of digital speech processing. The first chapter is dedicated to the general subject of speech modeling as a complex phenomenon

with inherent nonlinearity and non-stationarity. The second chapter comprises a short review of basic procedures of linear speech prediction, from the autocorrelation and covariant method to different versions of predictive lattice structures. The intention of the third chapter is to make the reader acquainted with the basic postulates of the general theory of robust parameter estimation, and especially with the concept of minimax robust estimation. The fourth and fifth chapters, as the central part of this book, represent an overview of the developed robust methods for the estimation of speech signal model parameters in a non-recursive, as well as in a recursive form. The sixth chapter presents the results of one of the alternative approaches to the introduction of a new class of nonlinear algorithms for parameter estimation of speech signal models, based on statistical pattern recognition. Seventh chapter is dedicated to the most important applications of the developed robust procedures, such as the segmentation of speech signal, extraction of formant trajectories, and speech signal coding.

The overall level of the text is suited to the readers with an adequate fore-knowledge in the probability theory and statistics, as well as in identification and estimation of signal model parameters. Graduates from engineering faculties will be able to follow the text without significant difficulties, while an additional effort will be required from the undergraduates at the final years, as is customary for this kind of texts. The methodological approach of this book makes it especially convenient for graduate courses in the fields covered by it, such as modeling and estimation of model parameters of stochastic signals and systems, estimation of time-variable parameters of non-stationary models, digital signal processing, modeling, analysis, and processing of speech signals. It ensures a single place where one can access a number of practical problems to get an insight into the whole procedure of analysis and synthesis of required properties, together with a comprehensive practical evaluation, which is the basis of the research and development in engineering. Because of that, this book is also useful for research institutions whose work is connected with the presented subject.

The authors wish to express their gratitude to the reviewers Prof. Dr. Milan Savić and Prof. Dr. Jovan Golić for their useful suggestions and advices, as well as to all of those who contributed to the publishing of this monograph.

Let us mention at the end that the contributions to this book of all four authors are comparable and that we adopted an ordering of authors according to their academic ranks.

Belgrade, Serbia
2014

Branko Kovačević
Milan Milosavljević
Mladen Veinović
Milan Marković

Robust Digital Processing of Speech Signals

Kovacevic, B.; Milosavljevic, M.M.; Veinović, M.; Marković, M.

2017, XII, 224 p. 54 illus., Hardcover

ISBN: 978-3-319-53611-8