Past, Prediction Error, Prediction with Finite Past, Solution of the Normal Equation, The Triplet a,c,k, Lattice Filters for Prediction, Positivity and Stability, S-Step Prediction, The Maximum Entropy Method, Wold Decomposition and Related Problems, Appendix on Stability, Problem. It deals with the prediction theory providing one of the most illuminating applications of the results given in the previous two chapters. The first reason is that the calculations, which are rather complex in the general case, are greatly simplified in prediction, mainly due to the fact that the signal being estimated is only a shifted version of the observed signal. The second reason is that prediction theory of stationary signals is a very powerful method for the analysis and classification of these signals. Finally, the calculation of filters introduced in prediction theory, also called predictors, introduces a very powerful class of filters known as lattice filters.

The thirteenth chapter, entitled "Time Recursive Methods," discusses: Time Recursions for Normal Equation, Recursion Least Square Methods, Introduction to Adaptive Filtering, Kalman Filtering. This chapter deals with the time recursive methods, which are very interesting in signal processing problems because they are easy to implement on computers or on digital signal processors. Therefore, there is a strong incentive to give recursive form to many signal processing methods. The fourteenth chapter, entitled "Matched Filters," has sections entitled The Classical Matched Filter, Amplitude Estimation, and Generalized Matched Filter. The purpose of this chapter is to show that several problems usually presented independently have in reality the same nature, which is why the term matched filtering has been used, although this expression is commonly restricted to signal-to-noise ratio maximization.

Finally, it is worth noting that suitable problems are set out at the end of each chapter. According to the French tradition: a problem is like a story requiring various steps, the first elementary and the last often complex. It is possible to leave the story at any step and to return to finalize it later.

Dynamic Speech Models—Theory, Algorithms, and Applications—L. Deng (New York: Morgan & Claypool, 2006, Series on Synthesis Lectures on Speech and Audio Processing, series editor: B. H. Juang, pp. 106, ISSN: 1932-121X). *Reviewed by Francesco Gianfelici*

The speech signals are, on the one hand, the essential carriers of human intelligence and knowledge, and on the other hand, the most natural form of human communications that start with the formations of a message in a speaker's brain and ends with the arrival of the message in a listener's brain, in other words, a pervasive form of rich communication, which is characterized by causal style nature.

The mathematical nature of these signals is a highly dynamic process, relying on the coordination of linguistic, acoustic, and perceptual mechanisms that are individually dynamic as well. Therefore, the natural modeling of speech dynamics requires to define computational models with necessary simplifications and approximations aimed at mathematical and computational tractability, which is on the one hand, a key factor to understand why humans speak as they

The reviewer is with the Dipartimento di Elettronica, Intelligenza Artificiale e Telecomunicazioni, Università Politecnica delle Marche, Ancona 60121, Italy (e-mail: f.gianfelici@deit.univpm.it).

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do and how humans exploit redundancy and variability, and on the other hand, a key factor to enhance the efficiency and effectiveness of human speech communications for practical algorithm implementations in a broad range of modern electronic devices. This promising approach opens new research directions that represent the starting point to enhance the current state-of-the-art in human language technologies, especially in automatic speech recognition that is expected to benefit from comprehensive computational modeling of speech dynamics and should enable machines to communicate with humans in a natural and unconstrained way. To achieve these challenging goals, many researchers believe that the severe limitations of the hidden Markov models (HMMs) should be overcome, and novel approaches to represent key aspects of human speech process are highly desirable, especially in conversational and spontaneous speech recognition. These aspects, many of which are of dynamic nature, have been largely missing in the conventional HMM-based framework. Towards this objective, one specific strategy is the introduction of appropriate dynamic structure on the speech model that allows for the kinds of variations observed in conventional speech. Enhanced computational methods, including learning and inference techniques, will also be needed based on new or extended models beyond the HMM.

The major components of dynamic speech modeling are the following (to mention just a few): 1) the target-based dynamic modeling that interfaces between phonology and articulation-based phonetics, 2) the switching dynamic system modeling that represents the continuous, target-directed movement in the hidden articulations and in the vocal track resonances being closely related to the articulatory structure, and 3) the relations between the hidden articulatory or vocal tract resonance parameters to the measurable acoustic parameters, enabling the hidden speech dynamics to be mapped stochastically to the acoustic dynamics that are directly accessible to any machine processor.

The excellent book written by Li Deng deals with the theory, algorithms, and applications of dynamic speech models. It contains a survey done in a holistic manner of the relevant research and the current best practices in this area spanning over the last two decades. This well-written monograph is intended to be advanced materials of speech and signal processing for graduate level teaching, for professionals and engineering practitioners as well for researchers and engineers specialized in speech processing, and also for researchers in applied neural networks. In order to better appreciate the aforementioned contents let us briefly introduce the four book chapters.

The second chapter, entitled "A General Modeling and Computational Framework," presents a framework for modeling and computing. This chapter provides an excellent design philosophy for dynamic speech models and outlines five major model components, including phonological construct, articulatory targets, articulatory dynamics, acoustic dynamics, and acoustic distortions. It includes discussion of overlapping models for multitiered phonological construct, segmental target models, articulatory dynamic models, functional nonlinear model for articulatory-to-acoustic mapping, weakly nonlinear model for acoustic distortion, and piecewise linearized approximation for articulatory-to-acoustic mapping. For each of these components, good references are cited and discussed, and the approximations of the mathematical descriptions are introduced and justified. Dynamic Bayesian networks are exploited to provide a consistent probabilistic language for quantifying the statistical relationships among all the random variables in the dynamic models.

The third chapter, entitled "Modeling: From Acoustic Dynamics to Hidden Dynamics," is devoted to a comprehensive survey of many types of statistical models. It contains discussions of statistical models for acoustic speech dynamics, and statistical models for hidden speech dynamics. This chapter classifies the existing models into two main categories, acoustic and hidden dynamic models, and provides a unified perspective viewing these models as having different degrees of approximations to the realistic speech chain. Nonstationary-state HMMs, multiregion recursive models, multiregion nonlinear dynamic system models, and hidden trajectory models are described in some details.

The fourth chapter, "Models with Discrete-Valued Hidden Speech Dynamics," presents hidden dynamical models, application to automatic tracking of hidden dynamics, and the intractable algorithms of parameter estimation for decoding the phonological states. Modeling accuracy is inherently limited by the discretization precision, and the difficulty arising from the large discretization levels due to high dimensionality of the hidden dynamic variables is addressed with a new optimization technique. Except for these two approximations, the parameters estimation and the decoding algorithms developed and described in this chapter are based on rigorous expectation—maximization (EM) and dynamic programming techniques. Additionally, applications of this model and related algorithms are presented.

The fifth chapter, entitled "Models with Continuous-Valued Hidden Trajectories," presents a dynamic speech model with continuous hidden dynamic values, and uses explicit temporal function to represent the hidden dynamics or trajectories. Generation of stochastic hidden vocal tract resonance trajectories and acoustic observation data are discussed. Technical issues related to linearizing cepstral prediction function, computing acoustic likelihood, and model prediction of vocal tract resonance trajectories for real speech utterances are included, and simulation results on model prediction for cepstral trajectories are presented. Approximations introduced to overcame the original intractability problems are made by interactively refining the boundaries fixed when carrying out parameter estimation. Additionally, applications to phonetic recognition are also presented and analyzed.

Finally, it is worth noting the ambitious goal for the dynamic speech models to contribute to the foundation of the next generation speech technology by means of the integration of this unique style of research along with other powerful pattern recognition and machine learning approach.