Introduction to the Issue on Statistical Parametric Speech Synthesis

STATISTICAL parametric speech synthesis has been a hot topic for some time. Classic statistical parametric speech synthesizers are able to produce fairly natural-sounding and flexible voices, needing only a relatively small training database, and can be more easily adapted to a new voice or speaking style than concatenative systems.

Although this approach has proved very successful in recent decades, there are still several open problems in this rich field. For instance, (a) how to build a statistical parametric speech synthesizer that can generate emotional or expressive speech, and how to use it in natural dialog applications; (b) whether, and how, to integrate physical speech production models into statistical parametric speech synthesis to improve the speech quality; (c) how to build high quality statistical parametric speech synthesizers using only a *very* small training database; (d) how to build parametric synthesizers or apply speaker adaptation for cross-language or multilingual applications. Motivated by these open problems, this special issue includes papers on expressive speech synthesis, multilingual techniques, excitation/generation models, adaptation and also introduces some new applications of statistical parametric speech synthesis.

The first group of papers suggests innovations in the area of statistical modeling. Koriyama *et al.* propose a statistical parametric speech synthesis technique based on Gaussian process regression (GRP) instead of the Hidden Markov Model (HMM). There is no question that the HMM is the most successful statistical models in spoken language technology and has been applied to almost every imaginable application, from speech and speaker recognition to speech synthesis. However, there are other statistical models which may be suitable for speech technology, including speech synthesis, such as GRP.

The second group of papers considers the parametric representation of speech for statistical parametric speech synthesis. Erro *et al.* explore the potential of the harmonic-plus-noise model of speech in the development of a high-quality vocoder applicable to statistical parametric speech synthesis. Cabral *et al.* propose an analysis method to separate the glottal source and vocal tract components of speech, which can produce high-quality synthetic speech using an acoustic glottal source model. Csapó *et al.* propose two alternative extensions of excitation models in order to model irregular phonation, which typically occurs phrase-finally. From these papers, we conclude that to continue to improve the speech quality of statistical parametric speech synthesis, further attention needs to be paid to the parametric representation of speech. These papers make the following contributions to that effort:

1) Improve the existing parametric representation of speech, Erro, *et al.*;
2) Propose a new parametric representation of speech, Cabral, *et al.*;
3) Parametric representation of irregular phonation, Csapó, *et al.*

and together they suggest a variety of future directions in this area.

The third group of papers focuses on the speech parameter generation stage of statistical parametric speech synthesis. The speech generated from statistical parametric speech synthesis is often said to suffer from over-smoothing caused by statistical modeling. The standard parameter generation algorithm, through which the speech parameters are generated from trained HMMs, produces a sequence of observation vectors according to a maximum likelihood optimization criterion. In the papers in this issue, two ways to overcome the supposed over-smoothing problem are presented. One is to adjust the optimization criterion of the parameter generation algorithm. Nose *et al.* introduce a parameter generation algorithm using a local variance (LV) model and provide experimental results that indicated better results than using global variance (GV). The other way is to improve the accuracy of HMMs themselves. Takaki *et al.* examine the use of a spectral modeling technique based on an additive structure of context dependencies.

Although improvements to the parametric representation of speech and the parameter generation algorithm are both shown to improve synthetic speech quality, still the speech signal generated from the vocoder is of lower quality than the original speech waveform. Is it therefore natural to think about ways to directly use original speech waveforms in statistical parametric synthesis. So, the fourth group of papers in this issue concern a hybrid method in which the parameters generated from a statistical model are used to guide concatenative speech synthesis using a stored speech corpus. Takamichi *et al.* use rich context models to construct a hybrid speech synthesis system that combines HMM-based speech synthesis and unit selection waveform generation. This hybrid method combines some of the advantages of both techniques, such as flexibility, rapid system construction and high-quality speech. A key problem in this hybrid method is how to define the target and join costs for the unit selection synthesis, based on the HMM-generated speech parameters. It seems that further work on this aspect of hybrid speech synthesis system is still needed.

The fifth group of papers is about adaptation. In these papers, two new adaptation methods are proposed. Sung *et al.* introduce factored maximum penalized likelihood kernel regression (FMLKR) and Saheer *et al.* combine vocal tract length normalization (VTLN) with linear transforms in a hierarchical Bayesian framework. Urbain *et al.* exploit the adaptation of HMM-based speech synthesis to generate laughter. Karhila *et al.* analyze the robustness to noise of HMM adaptation. The

Digital Object Identifier 10.1109/JSTSP.2014.2309416

flexibility of speech adaptation in statistical parametric speech synthesis is also covered: Wan *et al.* describe the application of average voice models (AVMs) and a novel application of cluster adaptive training (CAT) with multiple context-dependent decision trees to create HMM-TTS voices using diverse data. Together, this group of papers investigate a number of important issues in constructing adaptation system, namely the adaptation method, corpus construction, and the application of adaptation during synthesis.

The final group of papers is about applications of statistical parametric speech synthesis. Picart *et al.* focus on the automatic modification of the degree of articulation of an existing standard neutral voice in the framework of HMM-based speech synthesis. Chen *et al.* explore an integrated approach in order to construct a synthesizer with more expressions. Schabus *et al.* use a joint audiovisual hidden semi-Markov model for speech synthesis. These papers all showcase the effectiveness of statistical parametric speech synthesis in expressive speech synthesis. What we don't see in the papers in this issue are cross-language or multi-lingual applications. So, it would seem that more attention should be paid to how to use the technology of statistical parametric speech synthesis in those applications.

Together, the papers included in this special issue cover a broad range of current issues in statistical parametric speech synthesis, each of them offering novel ideas and improvements to the state of the art. Nevertheless, the speech generated from statistical parametric speech synthesis still has inadequate quality and naturalness, and it is likely this is a combined effect of the oversimplified parametric representation of speech used by the vocoder and the subsequent statistical modeling. The first four groups of papers each focus on different ways to improve the quality of synthesized speech by addressing one or other of those components. The last two groups tell us about the flexibility of statistical parametric speech synthesis, which is its key 'selling point' and the reason that continued work to improve the speech quality is warranted.

The editors would like to sincerely thank both the EiC for recommending this special issue be published and for his patience in guiding us—the guest editors—through the process, the publication assistants who provided invaluable assistance in navigating both the process and the Manuscript Central website. We hope that you—the reader—will be stimulated by the interesting work represented in this special issue and that it helps spur additional research within this exciting and growing field.

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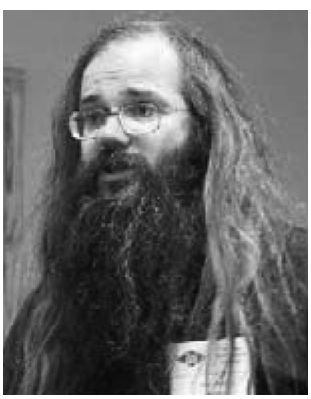

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