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Editorial

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Interest in digital processing of audio signals has been reinvigorated by the introduction of multimedia communication via the Internet and digital audio broadcasting systems. These new applications demand high bandwidth and require innovative solutions to an old problem: how to achieve high quality at low bit rates. Often this problem is addressed by transmission schemes in which only part of the original audio data is transmitted. Other sources, voices or channels. The output must be reconstructed at the receiver from purely synthetic or incomplete data. Additionally, the global networked audio community must solve a new class of problems concerning protection of audio streams and documents. Accordingly, robust methods are sought for enforcing security, privacy, ownership, and authentication of audio data. Furthermore, the maintenance of audio archives—our cultural heritage—requires the development of efficient techniques for the restoration of corrupted audio documents.

This special issue provides a sample of the new directions of digital audio research.

In audio synthesis, real-time computation of physical models of acoustic instruments is now possible due to the steady progress of Moore's law. In the paper by B. Bank et al., a review of piano synthesis is given. The synthesis is described in terms of structured audio and the structured audio orchestral language (SAOL) which is included in MPEG-4. Through the use of filtering and interpolation, P. A. A. Esquef et al. describe the use of the frequency-zooming analysis method to derive an ARMA model for synthesizing stringed instruments. Model-based computation of string sounds can be used to create more expressive synthesis of string sounds by offering a wide space of controllable parameters.

Multichannel audio promises to bring more realistic reproduction to the listener. In the paper by A. Mouchtaris et al., a small number of microphone signals are resynthesized into a larger number of "virtual microphones," thereby reducing the transmission bandwidth while enhancing the final rendering. In the paper by D. Yang et al., a high-performance scheme based on the MPEG advanced audio coding system that allows for the efficient transmission of multiple audio channels at scalable bit rates is proposed.

Watermarking and data-hiding techniques try to prevent unauthorized use of audio resources and additionally make it possible to include additional metadata in the audio stream. In their paper, M. F. Mansour and A. H. Tewfik introduce a new method for robust scale and shift invariant data-hiding based on wavelet transforms. The paper by M. Steinebach and J. Dittmann addresses the problem of authenticating audio streams by embedding content related data that allow the decoder to check for integrity.

Quality networked speech communication poses not only bandwidth but also privacy concerns. In their paper, C. R. N. Athaudage et al. propose a new method for efficiently encoding the spectral information in a low-rate speech coder. The authors exploit the possibility of increasing the coding gain at the cost of introducing a substantially higher coding delay. Real-time software applications designed for securing speech transmission over the Internet are reviewed in the paper by A. Aldini et al.

In denoising or noise-reduction problems, a time varying filter can be applied to the corrupted audio signal. Earlier work on a minimum mean square error (MMSE) estimator by Ephraim and Malah is quite expensive to compute. In P. J. Wolfe and S. J. Godsill's paper, a Bayesian estimator that is easier to compute and easier to understand is derived.

The guest editors would like to thank the authors and the reviewers of the papers for their contributions in maintaining clarity, coherence, and consistency in this special issue.

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Gianpaolo Evangelista received the Laurea in physics (summa cum laude) from the University "Federico II" of Naples, Napoli, Italy in 1984 and the M.S. and Ph.D. degrees in electrical engineering from the University of California, Irvine, in 1987 and 1990, respectively. Since 1995, he is Assistant Professor in the Department of Physical Sciences, University "Federico II" of Naples. From 1998 to 2002 he was Scientific Adjunct in the Laboratory for Audiovisual Communications, Swiss Federal Institute of Technology, Lausanne, Switzerland. From 1985 to 1986, he worked at the Centre d'Etudes de Mathématique et Acoustique Musicale (CEMAMu/CNET), Paris, France, where he contributed to the development of a DSP-based sound synthesis system, and from 1991 to 1994, he was a Research Engineer at the Microgravity Advanced Research and Support (MARS) Center, Napoli, where he was engaged in research in image processing applied to fluid motion analysis and material science. His interests include digital audio; music, speech, and image processing; synthesis and coding; wavelets; and multirate signal processing. Dr. Evangelista was a recipient of the Fulbright Fellowship.

Mark Kahrs received an A.B. degree in applied physics and information science (with high honors) from Revelle College, University of California, San Diego in 1974. He received his Ph.D. degree in computer science from the University of Rochester in 1984. He has held positions at Stanford University, Xerox PARC, Institut de Recherche et Coordination Acoustique/Musique (IRCAM) in Paris, Bell Laboratories, and Rutgers University. In the Spring of 2001, he was a Fulbright Scholar at the Acoustics Laboratory, Helsinki University of Technology. He is currently a visiting Associate Professor in the Department of Electrical Engineering at the University of Pittsburgh. His audio specific interests include DSP for electroacoustic transducers, multichannel DSP hardware and new analysis and synthesis methods for computer music.

Emmanuel Bacry graduated from École Normale Supérieure, Ulm, Paris, France in 1990. He received the Ph.D. degree in applied mathematics from the University of Paris VII, Paris, France in 1992 and obtained the "habilitation à diriger des recherches" from the same university in 1996. Since 1992, he is a Researcher at the Centre Nationale de Recherche Scientifique (CNRS). After spending four years in the Applied Mathematics Department of Jussieu (Paris VII), he moved, in 1996, to the Centre de Mathématiques Appliquées (CMAP) at École Polytechnique, Palaiseau, France. During the same year, he became a part-time Assistant Professor at École Polytechnique. His research interests include signal processing, wavelet transform, and fractal and multifractal theory with applications to very various domains such as sound processing and finance.