SESSION L: POSTER SESSION II - ICASSP'03 PAPERS

Spatial processing, content analysis, and other topics

L-01: Experimental Comparison of Particle Filtering Algorithms for Acoustic Source Localization in a Reverberant Room

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Traditional acoustic source localization techniques attempt to determine the current location of an acoustic source from data obtained at an array of sensors during the current time only. Recently, state-space methods have been proposed that use particle filters to perform recursive estimation of the current source location using all previous data. In this paper we present an overview of these particle filter algorithms, and formulate performance measures for determining their ability to track a moving source. We present results of experiments using reverberant data recorded in a real room, and show that steered beamforming methods have improved performance over GCC-based approaches.

L-02: On the Importance of Exact Synchronization for Distributed Audio Signal Processing

Rainer Lienhart, Igor Kozintsev and Minerva Yeung, *Intel Corporation, Santa Clara, CA, USA*

Stefan Wehr, *University Erlangen-Nürnberg, Erlangen, Germany*

We propose a new paradigm for implementations of audio array processing algorithms on a network of distributed general-purpose computers. In contrast to currently existing DSP processor-based solutions, our approach offers new possibilities for advanced array signal processing by enabling the usage of general-purpose computing platforms with their superior computational and storage resources. We demonstrate that synchronization of sensors is essential for acoustic Blind Source Separation (BSS) algorithms, and we propose a synchronization scheme that enables BSS on distributed, wirelessly networked computers and can easily be implemented on existing hardware.

L-03: Adaptive Esprit Algorithm Based On the Past Subspace Tracker

Roland Badeau, Gaël Richard and Bertrand David, *Ecole Nationale Supérieure des Télécommunications, Paris, France*

Sinusoidal modeling is a powerful tool for audio signal processing, which represents the signal as a sum of sinusoids whose frequencies may vary over time. In most applications, the estimation and tracking of multiple frequencies is achieved by means of the Fourier analysis. Concurrently, we propose a new frequency estimation and tracking method, based on two algorithms derived from the concept of signal subspace. An application to audio is presented at the end of the paper.

L-04: The Plenacoustic Function, Sampling and Reconstruction

Thibaut Ajdler and Martin Vetterli, *Swiss Federal Institute of Technology, Lausanne, Switzerland*

Martin Vetterli, *University of California, Berkeley, CA, USA*

In the present paper, we study the spatialization of the sound field in a room, in particular the evolution of room impulse responses as function of their spatial positions. The presented technique allows us to completely characterize the sound field in any arbitrary location if the sound field is known in a certain finite number of positions. The existing techniques usually make use of room models to recreate the sound field present at some point in the space. Our technique simply starts from the measurements of impulse responses in a finite number of positions and with this information the total sound field can be recreated. An analytical solution of the problem is given for different cases of spaces. Further, we determine the number and the spacing between the microphones needed to perfectly reconstruct the sound field up to a certain temporal frequency. The optimal sampling pattern for the microphone positions is given. Applications are also discussed.

L-05: Fast Convolutive Blind Speech Separation Via Subband Adaptation

François Duplessis-Beaulieu and Benoît Champagne, *McGill University, Montréal, Canada*

In this paper, we consider the problem of blind source separation (BSS) applied to speech signals. Due to reverberation, BSS in the time domain is usually expensive in terms of computations. We propose in this paper a subband BSS system based on the use of adaptive feedback de-mixing networks in an oversampled uniform DFT filter bank structure. We show that the computational cost can be significantly decreased if BSS is carried out in subbands due to the possibility of reducing the sampling rate. Experiments with real speech signals, conducted with two-input twooutput BSS systems using oversampled 32-subband and fullband adaptation, indicate that separation quality and distortion are similar for both systems. However, the proposed subband system is more than 10 times computationally faster than the fullband one.

L-06: Acoustic Beamforming Exploiting Directionality of Human Speech Sources

Terence Betlehem and Robert C. Williamson, *Australian National University, Canberra, Australia*

This paper examines the improvement that can be attained with perfect knowledge of the sound source directivity pattern and orientation in beamformer designs in the problem of speech acquisition. Data-independent beamformers are derived through formulation of a constrained optimization problem with a unity-gain constraint. Using computer simulation, these beamforming schemes