are compared to the delay and sum (DS) beamformer and the best single sensor in a reverberant room environment. Criteria used to measure performance are (1) the direct to reverberant ratio, to assess extent of reverberation suppression, and (2) an objective measure of speech intelligibility called the speech transmission index (STI). For human-speaker source directivity, simulation results show that modest improvements to performance are obtainable.

L-07: Onset-Based Segregation of Stop Consonants

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Speech segregation from acoustic interference is a challenging task. Previous systems have dealt with voiced speech with success, but cannot handle unvoiced speech. We study the segregation of stop consonants, which contain significant unvoiced signals. We propose a novel method that employs onset as a major cue to segregate stop consonants. Our system first detects stops through onset detection and Bayesian classification of acoustic-phonetic features, and then perform grouping based on onset coincidence. The system has been tested and performs well on utterances mixed with various types of interference.

L-08: Multi-Microphone Residual Echo Estimation

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Post-filters are a powerful extension to improve echo attenuation when combined with the well-known echo canceller. In order to guarantee high quality of the transmitted speech signal the primary purpose of a post-filtering system is to estimate the power spectral density (PSD) of the residual echo at the output of the echo canceller as accurately as possible. In this contribution, we introduce a novel technique to estimate the residual echo by using a microphone array. The robustness against double-talk and other additive interferences is reached by means of minimum statistics and further enhanced by exploiting spatial information. Simulation results show that the new methods are able to estimate the residual echo even under adverse conditions.

L-09: Toward Intelligent Sensors Reliability for Time Delay Based Direction of Arrival Estimates

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Increasing demand for automatic large area surveillance has made methods estimating direction of arrival (DOA), especially acoustic methods, an interesting topic. In these systems false alarms and faults are a disturbing factor. This paper presents a reliability criterion and a new method to diminish these errors. In very low signal-to-noise ratio conditions there are seldom any means to improve the performance of a DOA estimator. In these cases it is important to have some information about the estimation reliability. Our focus is on wideband signals propagating as planar waves in three dimensional space. If the estimation method makes no assumptions on the signal propagation speed, it is possible to compute a reliability measure for obtained estimates. With this reliability measure we are able to decide whether the data produced by a time delay based DOA estimation system is usable or not. Indeed, with this criterion an estimation system can state its own reliability.

L-10: A Coherent Multiple Frequency Estimator and Its Applications in Synthesis-Analysis Audio Watermarking

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This paper extends our previous work on single parameter watermarking to the multiple parameter cases. Watermarks are embedded in multiple sinusoidal tracks via frequency quantization index modulation (F-QIM), and extracted by a coherent multiple frequency estimator using Newtons method. Experiments are conducted subject to MP3 attacks and, in an 8-track sinusoidal synthesis-analysis setup, above bits/sec of data hiding is consistently achieved with error correction. Furthermore, the frequency estimators performance is compared against the fundamental limit of the inverse Fisher information.

L-11: Scalable Non-Square Blind Source Separation in the Presence of Noise

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Few source separation and independent component analysis approaches attempt to deal with noisy data. We consider an additive noise mixing model with an arbitrary number of sensors and possibly more sources than sensors (the degenerate separation problem) when sources are disjointly orthogonal. We show how disjoint orthogonality can be viewed as a limit of a stochastic voice modeling assumption. This is the basis for our approach to noisy model estimation by maximum likelihood, under the direct-path far-field assumptions. The implementation of the derived criterion involves iterating two steps: a partitioning of the time-frequency plane for separation followed by an optimization of the mixing parameter estimates. The solution is applicable to an arbitrary number of microphones and sources. Experimentally, we show the capability of the technique to separate four voices from two, four, six and eight channel recordings in the presence of strong noise.