Session 2pSP

Signal Processing in Acoustics: Signal Processing Techniques

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Contributed Papers

1:15

2pSP1. Forward-scattered acoustic intensity from prolate spheroids. Brian R. Rapids and Gerald C. Lauchle (Grad. Prog. in Acoust. and Appl. Res. Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804)

In bistatic scattering geometries, the detection of a forward-scattered signal is particularly difficult because the incident and scattered waves combine into a simultaneous mixture. The result is that the amplitude of the scattered wave becomes masked by that of the incident wave. During detection of the scattered signal, conventional space-time processing techniques regard the source signal as interference and attempt to suppress it. While vector sensors alone possess an inherent directivity due to their fundamental nature, intensity vector sensors coherently measure the acoustic pressure and particle velocity components (or related quantity such as acceleration, displacement, or pressure gradient). The coherent measurement of both acoustic field parameters may provide unconventional information regarding the presence of an object because of their known relationship. It is hypothesized that techniques based upon these coherent measurements will be able to process the total acoustic field rather than filter the scattered signal from the incident signal during detection. Theoretical and available experimental results will be presented to describe these hypothesized capabilities. [Work supported by ONR, Code 321SS under Grant No. N00014-01-1-0108, Dr. James F. McEachern, project monitor.]

1:30

2pSP2. Further investigations into performance metrics for underwater communications. Scott L. Whitney, Geoffrey S. Edelson, Ned B. Thammakhoune, and Michael S. Richman (BAE Systems, MER15-2651, P.O. Box 868, Nashua, NH 03061-0868, scott.l.whitney@baesystems.com)

Determining the relationship between the performance of an underwater acoustic data communications system and the operating environmental conditions is a problem that continues to plague researchers. The complexity of the time-varying channel is difficult to measure and model. Therefore an approach that uses metrics measured from data collected at sea to characterize the channel is attractive. As expected, preliminary assessments on limited data have shown that performance depends not only on environmental conditions, but also on system implementation. By extracting a variety of metrics, a better understanding of the subset that discriminate between good and bad performance can be developed. Also, by analyzing the relationship between certain metrics and performance, system limitations can be identified for re-evaluation. For example, a surprising result of the initial assessment of performance using a multichannel decision feedback equalizer on real data showed that sparseness of multipath arrivals may be an arbiter of performance [M. S. Richman *et al.*, J. Acoust. Soc. Am. **110**, 2619 (2001)]. Therefore changes to the algorithm that allows for sparse arrivals may improve performance. In this paper, a larger number of metrics from greater quantities of real-data and system configurations are measured and evaluated against equalizer results.

1:45

2pSP3. Reverberation noise modeling using extreme value theory. Brian La Cour and Robert Luter (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, blacour@arlut.utexas.edu)

Normalized matched filter output forms the basis of target detection in active sonar. In a target-free environment, the central theorem, if valid, predicts that the statistics of the envelope follow a Rayleigh distribution, and, to first approximation, this is indeed observed. However, well-known departures from the Rayleigh model are found in the tail end of observed distributions. Traditional approaches to this problem have focused on constructing a simple, parameterized, non-Rayleigh distribution which more closely models observations. This paper suggests a novel alternative which focuses on a robust method of modeling only the tails of the distribution in favor of the less important body. Results from extreme-value theory are used to fit a generalized Pareto distribution (GPD) to the empirical cumulative distribution function, conditioned on a large threshold value. [A random variable X has a GPD if $P(X \leqslant x) = 1-(1+\gamma x/\sigma)^{-1/\gamma}$ for $x \geqslant 0$, $\sigma > 0$, and γ real; $\gamma = 0$ is the exponential distribution.] Estimates of γ and σ are discussed for a broad range of active sonar data, and the results are compared with fits to other popular non-Rayleigh models. The origins of non-Rayleighness are also considered, including finite-size effects, spatial and temporal correlations, and nonuniformity.

2:00

2pSP4. Performance of conventional and fluctuation-based signal detection applied to atmospheric acoustics in the presence of transients. Thomas Null, Chris Clark (Mil Tec, NCPA, Coliseum Dr., University, MS 38677), and R. A. Wagstaff (Univ. of Mississippi, University, MS 38677)

One problem encountered in atmospherics acoustics is the detection of steady signals in the presence of loud transient noise. The ability to discriminate against loud transients is an attractive feature of fluctuation-based beamforming. A fluctuation-based beamformer was developed and there was a need to evaluate its performance for acoustic environments that have frequently occurring transients. Synthetic noise fields, which included loud transient noise, were created. Subsequently, receiver operating characteristic (ROC) curves could be produced in a Monte Carlo fashion. In order to provide a benchmark, the results of a conventional beamformer were similarly tested via ROC curves. These ROC curves allowed for comparison of the two beamformers under specific signal and noise conditions. In this presentation, the effects of transients on the outputs of both the conventional beamformer and the fluctuation-based beamformer are discussed. Particular interest is focused on the amplitude distribution of the outputs.

2:15

2pSP5. Construction of high frame rate images with Fourier transform. Hu Peng and Jian-yu Lu (Ultrasound Lab., Dept. of Bioengineering, The Univ. of Toledo, Toledo, OH 43606, jilu@eng.utoledo.edu)

Traditionally, images are constructed with a delay-and-sum method that adjusts the phases of received signals (echoes) scattered from the same point in space so that they are summed in phase. Recently, the

relationship between the delay-and-sum method and the Fourier transform is investigated [Jian-yu Lu, Anjun Liu, and Hu Peng, "High frame rate and delay-and-sum imaging methods," IEEE Trans. Ultrason. Ferroelectr. Freq. Control (submitted)]. In this study, a generic Fourier transform method is developed. Two-dimensional (2-D) or three-dimensional (3-D) high frame rate images can be constructed using the Fourier transform with a single transmission of an ultrasound pulse from an array as long as the transmission field of the array is known. To verify our theory, computer simulations have been performed with a linear array, a 2-D array, a convex curved array, and a spherical 2-D array. The simulation results are consistent with our theory. [Work supported in part by Grant 5RO1 HL60301 from NIH.]

2:30–2:45 Break

2:45

2pSP6. Using cross-frequency cost functions for broadband source localization and environmental inversion. Ethan P. Honda (Appl. Res. Labs., Univ. of Texas, 10000 Burnet Rd., Austin, TX 78758)

Cost functions that are constructed by coherently summing (model-to-data) correlations over hydrophone pairs and frequency have been used successfully for source localization [E. K. Westwood, J. Acoust. Soc. Am. **91**, 2777–2789 (1992)] as well as source localization and environmental inversion [Neilsen, J. Acoust. Soc. Am. (to be published)]. Although the coherent sum is usually taken over the same frequency for both data and model, it is shown that summing over other regions of the $f_{\mathrm{data}} \otimes f_{\mathrm{model}}$ space is also useful and may facilitate more efficient source localization. It is shown that lines of constant $f_{\mathrm{data}}/f_{\mathrm{model}}$ correspond to different source bearings. Although looking along lines of constant $f_{\mathrm{data}}/f_{\mathrm{model}}$ can be used as a crude form of spatial filtering, a new non-plane-wave spatial filter is also constructed that helps localize sources in the presence of multiple interferers. The spatial filter employed uses the environmental model to construct its set of basis functions and is therefore theoretically capable of spatially filtering in full 3-D as opposed to just bearing, as is done in adaptive beamforming.

3:00

2pSP7. Wigner–Ville representations for acoustic source localization. Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102) and Leon Cohen (Dept. of Phys. and Astron., Hunter College, City Univ. of New York, 695 Park Ave., New York, NY 10021)

Signal dispersion in a waveguide can be linked to source and receiver location as well as physical properties of the propagation medium. Traditionally, the main methods used for source localization in the ocean and in geoacoustic inversion problems have been based on the use of the spectrogram for the extraction of the dispersion information. In this work we explore the possibility of using other time-frequency transforms because other transforms, such as the Wigner–Ville representation, reflect the dispersion properties of the waveguide more accurately than conventional spectrograms. We apply the Wigner–Ville distribution, in conjunction with sound propagation models, for inversion with underwater sound. Results with synthetic data calculated for simplified ocean media indicate the potential of the approach for successful parameter estimation. [Work supported by ONR.]

3:15

2pSP8. Multichannel active noise control and acoustic equalization using fast affine projection algorithms. Martin Bouchard (School of Information Technol. and Eng., Univ. of Ottawa, 161 Louis Pasteur, Ottawa, ON K1N 6N5, Canada)

In the field of adaptive signal processing, it is well known that affine projection algorithms or their low-computational implementations, fast affine projection algorithms, can produce a good trade-off between convergence speed and computational complexity. Although these algorithms typically do not provide the same convergence speed as recursive-least-squares algorithms, they can provide a much improved convergence speed compared to stochastic gradient descent algorithms, without the high increase of the computational load or the instability often found in recursive-least-squares algorithms. In this presentation, multichannel fast affine projection algorithms are introduced for active noise control or acoustic equalization. Multichannel fast affine projection algorithms have been previously published for acoustic echo cancellation, but the problem of active noise control or acoustic equalization is a very different one, leading to different structures. The computational complexity of the new proposed algorithm is evaluated, and it is shown through simulations that not only can the new algorithm provide the expected trade-off between convergence performance and computational complexity, it can also provide the best convergence performance (even over recursive-least-squares algorithms) when non-ideal noisy acoustic plant models are used in the adaptive systems.

3:30

2pSP9. A comparison of algorithms and the development of a new fast convergence and reduced computational load algorithm for multichannel active noise control. Martin Bouchard and Scott Norcross (School of Information Technol. and Eng., Univ. of Ottawa, ON K1N 6N5, Canada)

In this presentation, the three main factors that affect the convergence speed of learning algorithms for adaptive FIR filters used in multichannel active noise control are described. Based on these three factors, a comparison of several adaptive FIR filter algorithms for multichannel active noise control is done, including several existing algorithms and a few unpublished algorithms. Of the unpublished algorithms, one algorithm has the potential for optimal convergence speed, and this algorithm is described in more detail in the presentation. The algorithm combines the use of recursive-least-squares algorithms with the use of an inverse model of the multichannel acoustic plant between the actuators and the error sensors. The resulting algorithm is called the multichannel inverse delay-compensated filtered-x RLS algorithm for active noise control. This algorithm can not only provide fast convergence, but for multichannel systems it also provides a significant reduction of the computational load compared to the previously published algorithm with the fastest convergence speed. Simulation results are presented to validate the convergence behavior of the new proposed algorithm.

3:45

2pSP10. Inverse source problem by convex optimization with constraints over the object space and signal field. Kenbu Teramoto (Dept. of Mech. Eng., Saga Univ., Saga-shi 8408502, Japan, tera@me.saga-u.ac.jp)

In the acoustical endscopy, due to the physical limitations, the transducer array is composed of a small number of elements and each interspacing is larger than the acoustical wavelength that is called a sparse array system. In such cases, avoiding the ill-posed problems, projection onto convex sets (POCS) methods are used with incorporating constraints about both the signal field and the object space. POCS, however, is based on the alternating projections paradigm, which has a slow-convergence property in general. Furthermore if inconsistency exists in the set of constraints, this POCS algorithm cannot guarantee the convergence to the optimal estimate. The proposed algorithm is based on convex optimization over the direct product of the object space and the observed signal field. By acoustical experiments, it is proved that the proposed algorithm has the following improvements: (1) Targets can be identified when unknown components exist in the transfer function. (2) Transient behavior of the convergence becomes more stable than that of POCS algorithm. (3) Instability caused by the inconsistency in the constraints can be reduced. (4) Artifacts caused by the spurious lobes can be reduced under the condition that the interspacing of transducer elements is larger than the wavelength.