The spatial amplitude mapping method for estimation of time delay using adaptive filtering

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convergence of the simulated annealing process. Tests using simulated data constructed by convolving a known signal with a theoretical Green's function and then adding Gaussian white noise have been conducted. Excellent reconstructions of the source time series have been achieved for cases of high and low signal-to-noise ratio. This methodology is being tested on deep ocean data obtained in a multipath environment. Results from simulations and experimental data will be discussed. [Work supported by ONR and NRL.]

10:00–10:15

Break

10:15

4aUW9. The spatial amplitude mapping method for estimation of time delay using adaptive filtering. Mohammad K. Nehal, Juan A. Henriquez, Terry E. Riemer, and Russell E. Trahan (Dept. of Elec. Eng., Univ. of New Orleans, New Orleans, LA 70148)

Uniform and multiple delays/advances were estimated under heavy noisy conditions [signal-to-noise-ratio (SNR$\triangleq$signal energy/noise energy) below 0 dB]. The technique introduced, called the *spatial amplitude mapping* (SAM) method, isolates a data segment from each of the channels in a multichannel system by using a suitable window. When two matched segments are plotted on an x–y plane, the distribution of every pair of windowed coordinates will remain near a straight 45° line that passes through the origin. The distribution of two nonmatched segments or two matched segments containing noise will scatter around this line; hence, a pair of matching windowed segments can be found by searching for the distribution closest to the 45° line. This information is then used to estimate the segment delays. Under very noisy conditions, however, multiple delays can be detected. A recursive least-squares (RLS) filter is then used to adaptively estimate the correct delay. The technique was implemented in the delay estimation of synthetic (stationary) data and neurophysiological (nonstationary) data with satisfying results. Compared to the window correlation technique (WCT) [Callison *et al.*, J. Acoust. Soc. Am. **81**, 1000–1006 (1987)], SAM can estimate delays down to a SNR of -17.1 dB while the lower bound for the WCT is -2 dB.

10:30

4aUW10. Developments in phase-matching filter techniques. G. Orris, B. E. McDonald, and W. Kuperman (Naval Res. Lab., Washington, DC 20375-5000)

For low signal-to-noise (S/N) data, an algorithm has been reported that is very effective when the shape of the noise spectrum is known and smooth as a function of frequency. The method uses the phases of the spectral components as free parameters in a variational problem. Improvements on the basic method to help relax the idealistic assumptions made have been investigated. Methods employing empirical orthogonal functions (EOFs) to exploit further degrees of freedom in the solution are presented. For cases of moderate S/N, an algorithm is presented that attempts to utilize signal-free intervals in improving S/N.

10:45

4aUW11. LS multi-line curved array signal localization. Homer Bucker (Code 541, NRaD, NCCOSC, San Diego, CA 92152)

Let F_j be the analytic signal received at sensor j of a hydrophone array of unknown shape and let $g_{jk}=\langle F_j F_k^* \rangle$ be an element of the covariance matrix. Above, the $*$ means complex conjugate and $\langle\ \rangle$ indicates a time average. If there are N plane wave signals incident upon the array and $\hat{g}_{jk}$ is the expected covariance element after "sufficient" time averaging then $\hat{g}_{jk}$ is equal to $\Sigma_n A_n^2 \exp\{i(2\pi/\lambda)[(x_j-x_k)\cos\phi_n+(y_j-y_k)\sin\phi_n]\}$. In the above equation, A_n and ϕ_n are the amplitude and bearing of signal n, x_j, and y_j are the horizontal coordinates of sensor j, and λ is the wavelength. An LS (least-squares) iteration takes place to reduce the error function $E=\Sigma_j\Sigma_k|g_{jk}-\hat{g}_{jk}|^2$ to a minimum. Values of $\{A_n\}$, $\{\phi_n\}$, and several harmonic coefficients that define the sensor coordinates relative to straight lines are adjusted to minimize the error function. Several examples will be presented to illustrate the method.

11:00

4aUW12. A Bayesian approach to passive acoustic signal processing. Richard Pitre and Nolan R. Davis (Code 5160, Naval Res. Lab., Washington, DC 20375-5000)

This work considers a systematic approach to data inversion and data fusion for stationary passive sonar in low signal-to-noise situations. Bayesian inversion is applied to the probability distributions that are implicit in conventional signal processing methods. The resulting source location probability distributions are multimodal, reflecting the sidelobe structure of conventional ambiguity functions. Using the probability interpretation of these distributions the secondary sidelobe peaks can be compared quantitatively with the mainlobe. Results of model calculations are presented for an ocean waveguide in order to demonstrate the method, provide a comparison with conventional approaches, and assess the performance under low signal-to-noise conditions. A preliminary discussion of data fusion is given for probability distributions derived from inversion of independent data sets. An application to frequency fusion is made, and a performance improvement is demonstrated with the computational model.

11:15

4aUW13. Fourth-order cumulants and spectra of acoustic data. Roger F. Dwyer (Naval Undersea Warfare Ctr., Code 3331, New London, CT 06320)

In a recent paper [Dwyer, J. Acoust. Soc. Am. **90**, 918–926 (1991)] properties of the Fourier transform of a special case of the fourth-order cumulant were discussed and the results of a transmit–receive experiment presented. The data from this experiment have now been analyzed for more general cases of the fourth-order cumulant and its corresponding spectrum. The paper reviews theoretical properties of fourth-order cumulants and spectra, discusses applications in underwater acoustics, and presents results from acoustical experiments and simulations.

11:30

4aUW14. Acoustic classification using fuzzy sets. Marc C. Leonetti and Edward A. Hand (Computer Sci. Corp., P.O. Box N, Moorestown, NJ 08057)