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broadband signal in an underwater acoustic waveguide. The processor estimates the modal spectrum of a broadband signal for source localization; as a consequence, the source spectrum and signature (time domain waveform) are estimated. (Broadband modal decomposition is the equivalent of time domain deconvolution for source signature extraction.) In addition to the usual lofargram, a frequency-depth plot is proposed to display a broadband signal. Source localization is conducted in the frequency domain by summing the (range and depth) ambiguity functions over the signal bandwidth. Sidelobes are suppressed in broadband processing if the signal has sufficient bandwidth. Coherent summation of the ambiguity functions yields the expected time-bandwidth gain if source signature is known (the extended matched filter processing). Numerical simulations are shown for an Arctic waveguide. The processor works for transient as well as continuous broadband signals.

9:00

**4aUW5. A normal-mode based approach for localization of transient signals.** S. M. Jesus (UCEH—University of Algarve, PT-8000 Faro, Portugal)

Normal-mode modeling is a well-accepted representation for acoustic signals propagating in a number of environmental conditions. Detection of the spatial normal-mode structure makes possible signal localization enhancement against the noise that has no spatial structure. A simple algebraic argument allows one to separate the vector subspace that contains the signal from the vector subspace that contains the noise and to obtain a narrow-band estimate of the source location [S. M. Jesus, *Signal Process.* **28**, 117–122 (1992)]. This subspace splitting algorithm has been extended for localizing broadband transient signals assuming that the signal only has a normal-mode structure. It is shown with synthetic data that the proposed broadband algorithm outperforms both the generalized minimum variance and the conventional processors. As an example, this processor has been used to localize short transient pulses collected in a 120-m depth shallow water area with a 62-m aperture vertical array. The experimental results show that stable and accurate localizations could be obtained during long time intervals. This shows that the sound field, received over a given frequency band, is relatively stable over time and is in agreement with the predictions given by a standard normal-mode propagation model.

9:15

**4aUW6. Detection of transients using the nonstationary bispectrum.** Martin L. Barlett, Kevin W. Baugh, and Gary R. Wilson (Appl. Res. Lab., Univ. of Texas at Austin, P. O. Box 8029, Austin, TX 78713-8029)

An algorithm based on the nonstationary bispectrum is proposed for detection of sampled finite duration signals. The detection algorithm uses a combination of coherent and incoherent smoothing in the frequency domain to produce a test statistic which is suitable for display as a spectrogram. A model transient waveform imbedded in Gaussian noise is used to evaluate the detection performance of the proposed detector. Detectors based on the spectral correlation and power spectrum are used as metrics to evaluate the relative effectiveness of the proposed detector. Performance is investigated both as a function of signal-to-noise ratio for a fixed transient duration and processing length and as a function of transient duration/processing length mismatch at a fixed signal-to-noise ratio. The benefits and limitations of the proposed test statistic based on the nonstationary bispectrum relative to the other detectors investigated will be noted. [Work supported under contract with Space and Naval Warfare Systems Command.]

9:30

**4aUW7. Model-based matched-filter processing of Doppler-shifted signals in a time dispersive ocean environment.** J.-P. Hermand (SACLANT Undersea Res. Ctr., Viale Bartolomeo, 400, I-19138 La

Spezia, Italy) and W. I. Roderick (Naval Undersea Warfare Ctr. Div., Newport, RI 02841-5047)

Dispersive multipath propagation in an ocean medium distorts wide-band linear frequency-modulated (LFM) transmitted signals. As a result, the performance of correlation receivers is degraded if the receiver does not account for the Doppler of the incoming signal or the multipath (energy splitting) in the medium. In this study, results are presented that demonstrate that the performance of a conventional correlation can be improved if the reference (replica) channel is compensated in both Doppler and time dispersion. The model-based matched filter is generated by correlating the received signal with a reference channel that consists of the transmitted signal convolved with the impulse response of the medium and Doppler compensated. The channel impulse responses were predicted with a broadband propagation model using environmental (sound velocity) data. The data were collected during a February 1990 experiment conducted in deep water in an area west of Sardinia. The acoustic data set consisted of linear frequency-modulated signals, with a time-bandwidth product of 4000, transmitted from a moving source and received on a towed array. Comparison with conventional processing shows improvement (about 3 dB) in peak output signal-to-noise ratio for the propagation conditions encountered in the experiment. [Work supported by ONR.]

9:45

**4aUW8. Target tracking using matched-field processing.** Michael J. Wilmut (Royal Roads Military College, FMO Victoria, BC V0S 1B0, Canada), John M. Ozard, and Bryan Woods (Defence Res. Establishment Pacific, FMO, Victoria, BC V0S 1B0, Canada)

The objective of this paper is to illustrate the use of matched-field processing (MFP) for tracking low signal-to-noise ratio targets moving linearly and at constant speed. The input to the tracker consists of the positions and power of the largest peaks on the MFP ambiguity surface. These largest peaks usually include the match at or near the source position even at low signal-to-noise ratio. An exhaustive search for the best matching track over all possible target tracks (that is allowing varying speed and heading) is beyond the scope of today's computers for any realistic search region. In this paper, an efficient algorithm is described based on examining the average Bartlett statistic along a set of linear tracks that connect only the largest peaks. This set was restricted to the physically possible tracks to further reduce the number to be examined. Examples of the ambiguity surfaces and the probability of examining the true track are given. The algorithm performance is a function of the scenario, signal-to-noise ratio, number of ambiguity surfaces, and number of peaks examined on each surface. It is shown that if the true target track is one of those examined its Bartlett statistic is almost certainly maximum. This efficient tracking requires only modest computing beyond that required to generate the ambiguity surfaces.

10:00

**4aUW9. Synthetic aperture processing of a moving cw sound source in a range-dependent underwater environment.** Thomas N. Lawrence and Nancy R. Bedford (Appl. Res. Lab., Univ. of Texas at Austin, P. O. Box 8029, Austin, TX 78713)

Synthetic aperture processing is a method of extracting horizontal wave numbers from recordings on one hydrophone. Such an approach has been previously suggested; most recently by Collins *et al.* [*J. Acoust. Soc. Am.* **92**, 2366(A) (1992)], who proposed single hydrophone matched-field processing. A similar method was used by Frisk *et al.* [*J. Acoust. Soc. Am.* **86**, 1928 (1989)] to derive bottom properties from data. The experimental method requires that a source (or receiver) move at a constant depth and a constant velocity with respect to a fixed receiver (or source). The moving source thus sweeps out a synthetic aperture with range, and array element recordings are processed by means of a Fourier transform of complex pressure with range, yielding the wave-number spectrum (or modal eigenvalues) over the chosen aperture. A series of overlapping apertures over the source track shows the evolution of the wave-number spectrum with the change in