A Book of Abstracts for the

1999 Underwater Acoustic Signal Processing Workshop*

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Alton Jones Campus

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Contents

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Workshop Schedule

Each talk will be 15 minutes with an additional 5 minutes for questions. The schedule of talks for the morning and afternoon sessions is listed below.

You may check into your room any time after 2 pm. Registration will begin at 4 pm and be available throughout the welcoming reception (5-6 pm) as well as Thursday morning from 8:15-8:35 am.

Lunches are at noon, Dinners at 6:00 pm. The Keynote Talk is at 8:00 pm on Wednesday. The SOB (Science Over Beer) session is Thursday at 8:00 pm.

	Time	Wednesday	Thursday	Friday
morning	8:15-8:35		registration	FM1
	8:40-9:00		TM1	FM1
	9:00-9:20		TM2	FM2
	9:20-9:40		TM3	FM3
	9:40-10:00		TM4	FM4
	10:00-10:20		BREAK	
	10:20-10:40		TM5	FM5
	10:40-11:00		TM6	FM6
	11:00-11:20		TM7	FM7
	11:20-11:40		TM8	FM8
	12:00-1:00		LUNCH	
afternoon	1:20-1:40		TA1	
	1:40-2:00		TA2	
	2:00-2:20		TA3	
	2:20-2:40		BREAK	
	2:40-3:00		TA4	
	3:00-3:20		TA5	
	3:20-3:40		TA6	
	3:40-4:00		TA7	

5:00-6:00	Welcome Reception	
6:00	Dinner	Dinner
8:00	Keynote Talk	Science Over Beer

Workshop Program

The workshop program is as follows (refer to schedule on page 3).

Wednesday Evening

Keynote Talk: Multi-disciplinary, Multiscale Ocean Observation and Forecasting

Henrik Schmidt

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Thursday Morning

TM 1. Range estimates from two beamforming arrays towed in series allows estimation of the source levels of resident killer whale vocalizations

Patrick J.O. Miller and Peter L. Tyack

Biology Department, Woods Hole Oceanographic Institution

TM 2. Locating whale sounds using correlation processing algorithms

Kurt M. Fristrup and Katherine J. Dunsmore

Cornell Lab of Ornithology

TM 3. Three-dimensional localization, geoacoustic inversion, and source signature recovery of blue whale vocalizations using matched-field processing methods.

A.M. Thode, G.L. D'Spain and W.A. Kuperman

MPL, Scripps Institution of Oceanography

TM 4. Automatic Detection and Species Identification of Blue and Fin Whale Calls

David A. Helweg

SPAWARSYSCEN

BREAK

TM 5. Information Entropy of Humpback Whale Songs

Ryuji Suzuki, John R. Buck and Peter L. Tyack

University of Massachusetts Dartmouth

TM 6. Marine mammal signal processing: Accounting for the underwater channel effects
Zoi-Heleni Michalopoulou and Xiaoqun Ma
New Jersey Institute of Technology

TM 7. Biomimetic Models of Dolphin Echolocation
Patrick W.B. Moore and David A. Helweg
SPAWARSYSCEN San Diego

TM 8. Sound Analysis Software and Hardware for Applications in Bioacoustics
G. Pavan
Universita di Pavia

LUNCH

Thursday Afternoon

TA 1. Adaptive noise cancellation in the presence of electronic noise
Trym Eggen
Simrad AS N-3191 Horten, Norway

TA 2. Design of environmentally robust nonuniform arrays for shallow water mode filtering
John R. Buck and Tsung-Jieh Shiao
University of Massachusetts Dartmouth

TA 3. The AWARE System: A Low-cost sonar for anti-grounding and collision avoidance.
David Cousins, James Miller, Angela Tuttle and Thomas Weber
BBN and Univ. of Rhode Island

BREAK

TA 4. Sonar Signal Processing Research at ONR: Status Report and Future Trends
John Tague
ONR

TA 5. Class Specific Classification
Steve Greineder
NUWC

TA 6. Real CFAR: Nonparametric, Data-Adaptive Thresholding as an Example of Tolerance-Region Signal Processing

Don Tufts

University of Rhode Island

TA 7. Cramer-Rao Lower Bounds for use in Towed Array Shape Estimation

D. A. Gray[1,2], B. G. Quinn[3], and J. L. Riley[4]

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2 EEE Dept, University of Adelaide,

3 Dept of Mathematics, University of Manchester Institute of Science and Technology,

4 Defence Science and Technology Organisation, Australia

Friday Morning

FM 1. Some Results about the method "Principal Component Inverse" for the Detection with Reverberation

G. Ginolhac and G. Jourdain

LIS-ENSIEG, France

FM 2. Forward-Look Undersampled Synthetic Aperture Sonar

C. Gedney[1], G. Edelson[2], I. Dyer[1], P. Abbot[1] and K. Rolt[3]

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2 Sanders, A Lockheed Martin Co., Nashua, NH

3 Sonetech Corp., Bedford, NH

FM 3. Experimental Results For Matched-Field Localization With a Small-Aperture Mid-Frequency Array

Seth Suppappola, Bruce MacLeod and Brian Harrison

NUWC

FM 4. Coherent Narrowband Multitarget Tracking

Robert Zarnich, Kristine Bell, and Harry Van Trees

George Mason University

BREAK

FM 5. The Class-Specific Classifier and Colored Noise
D. Abraham and and Fang Tu
University of Connecticut

FM 6. Detection and Imaging of Bottom/Buried Mines in Clutter
Ivars Kirsteins, Sanjay Mehta and John Fay
NUWC

FM 7. Nulling Moving Sources, Snapshots and Eigenspread
Harry Cox
ORINCON

FM 8. Adaptive Estimators of Output SNR for Random Channels
Shawn Kraut and Louis Scharf
University of Colorado

Abstracts

TM 1: Range estimates from two beamforming arrays towed in series allows estimation of the source levels of resident killer whale vocalizations

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The stereotyped calls produced by resident killer whales are thought to function in group coordination and cohesion. The source levels (SLs) of these sounds affect the range over which they can be received, so measuring SLs can help elucidate their effective range, which is related to their function. Ranges to vocalizing whales were estimated by triangulating the angles-of-arrival of sounds on two short (2.5m, 8-element) beamforming arrays towed in series with a separation of 100m. The broadband structure of killer whale calls makes it possible to distinguish the mainlobe from grating lobes in a frequency-azimuth (FRAZ) display calculated with a conventional broadband frequency-domain beamforming algorithm. Analysis of separate portions of typical killer whale calls suggest that the precision of the angle-of-arrival estimate is < 3 degrees. SLs (re 1 micro-Pa @ 1m) were calculated using the received level (from 100Hz to 22kHz) at one array, the range estimate, and a transmission-loss equation. In-situ calibration trials showed that range-error effects on the SL estimate are less than 3dB over a 400x200m area on either side of the arrays. To investigate whether different call types were produced at differing SLs, calls were classified to type manually using visual inspection of the spectrogram and aural confirmation.

Using this technique, we obtained SLs of 823 calls and 24 whistles over 5 hours of observation on 2 days in 1998. Overall, call SLs ranged from 130 to 165dB with a broad peak between 145 and 155dB. "Soft" (143-144dB) call types included N3, N8 and N11. "Medium" (145-147dB) call types included N7, N12, and N10. "Loud" (150-153dB) call types included N9, N1, N25, N47, N16, N5, N4 and N2. Standard deviations of SL's ranged from 2.5dB (N8) to 6.8dB (N12). Whistles had a consistently "soft" SL (138dB, sd=4.9dB, n=24). The loudest call type (N2) has previously been shown to be used most often in pod-meeting contexts, suggesting that it may function primarily as a long-range inter-group signal. The softest call (N3) and whistles are noted to occur in resting and beach-rubbing behaviors, suggesting they are shorter-range, intra-group signals. One of the primary advantages of this towable method, compared to fixed-array techniques, is that the SLs of relatively soft sounds can be obtained by positioning the array close to vocalizing animals, allowing a more complete characterization of the vocal repertoire.

TM 2: Locating whale sounds using correlation processing algorithms

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Initial efforts to localize animal sounds utilized cross-correlations and peak picking algorithms. Experience has shown that such techniques are unreliable with narrowband signals in all underwater environments, and with broadband signals in reverberant environments. Alternative methods of utilizing cross-correlation functions have yielded better performance. These performance advantages will be demonstrated with data from large whale species.

TM 3: Three-dimensional localization, geoacoustic inversion, and source signature recovery of blue whale vocalizations using matched-field processing methods

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Matched-field processing (MFP) and global inversion methods have passively recovered the three-dimensional trajectories of six blue whales over periods of up to 90 minutes and ranges of 8 km. The 1996 experiment was conducted in 130 m deep water off the California coast near San Miguel Island, where a 48-element tilted vertical hydrophone array was deployed from the research platform FLIP. A subset of strong signal-to-noise ratio calls from each whale was selected for inversion. Using the genetic algorithm optimization code SAGA, the inversion estimated the local sediment sound speed profile, ocean depth, and array geometry. The fit between the received data and inversion model were excellent. The recovered ocean bottom sound speeds were consistent with those expected from sediment samples obtained from the surrounding area, demonstrating how baleen whale vocalizations can be used as high intensity, low-frequency geophysical tools. These inversion parameters were subsequently used to construct MFP replicas that were applied to the rest of the whale calls, yielding depth and range. Some of the ranges obtained were over 60 times greater than the local water depth, so no direct acoustic path between the whale and the array existed. Whale bearings were obtained through three different methods: DIFAR sonobuoy measurements, inverted array tilt, and range-dependent MFP computations. Finally, propagation effects were removed from selected calls using multichannel deconvolution techniques, allowing the original time series generated by the animal to be recovered. All the whales tracked using these techniques vocalized between 10 to 40 m depth, and three complete dive profiles have been obtained. While most animals swam from the east to the west, one animal remained relatively stationary for 45 minutes while calling. The source signature recoveries allow refined estimates of source level, and suggest the possible presence of an internal resonator inside one animal. These combined results demonstrate how MFP can

recover the ranges, depths, and bearings of low-frequency vocalizations using as few as eight elements on a tilted vertical array, while requiring little knowledge about the surrounding ocean environment.

TM 4: Automatic Detection and Species Identification of Blue and Fin Whale Calls

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Baleen whales live at extended timescales. Study of how these animals use their vocalizations for communication requires massive data sampling over long periods. The volume of data precludes traditional hands-on analysis techniques, at least during the first stages of data reduction. This paper describes a system for automating the sampling and analysis of baleen whale calls. Blue (Balaenoptera musculus) and fin (B. physalus) whale calls are very stereotypical. Blue whale "A" and "B" calls have fundamental frequencies of approximately 17 Hz, narrow bandwidth and well-defined harmonic structure, and typical duration of 15-25 sec. Fin whale "pulses" have fundamental frequencies of approximately 17 Hz, but are broadband in nature and short ($\approx$ 1 sec) duration. The homogeneous call structure lends itself to automated detection. Stable acoustical differences in call structure lend themselves to automated species identification. We have benchmarked a series of bioacoustical call identification algorithms against a set of blue and fin whale calls while systematically manipulating the signal to noise ratio. The results demonstrated a typical tradeoff of speed versus accuracy. The best algorithm was inserted into the underwater sound recording system and its signal-detection theoretic performance was quantified. Results will be discussed with respect to technological and ecological aspects of baleen whale bioacoustics. [Project CS-1082 of the Strategic Environmental Research and Development Program]

TM 5: Information Entropy of Humpback Whale Songs

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Humpback whales produce songs in their winter breeding seasons in warm water. A song consists of a series of more than 12 discrete units, where a unit is the shortest sound element separated by silence. The fundamental frequency ranges from 30 to 3000Hz, and the duration of the longest observed vocalization session was over 24 hours. Several papers have analyzed the song structure with manual classification and visual inspection. Payne and McVay (1971) proposed a hierarchical syntax, which is currently most commonly accepted, but which also has caused a linguistic question: do songs really have a hierarchical syntax, or is it a result of human bias?

Tape-recorded songs were digitized and individual units detected and extracted using an automated detection algorithm. The individual units were encoded into spectrograms with logarithmic frequency axis, and classified in two layers of self-organizing maps (a neural network, Kohonen 1995; Kohonen et. al. 1996), symbolizing the entire song. This process was performed separately for each song. The rest of analysis takes the symbolized song sequences to examine the structure of the songs using information theory.

The entropy of the song generation process was estimated with two parametric models and a non-parametric estimator for comparison, rather than justifying the model selection. We chose independent identical distribution, a.k.a. stationary memoryless model and first order empirical Markov model, the two most popular choices in animal communication studies, and the sliding window match length (SWML) estimator (Kontoyiannis et. al. 1998), derived from Wyner-Ziv result (Wyner and Ziv 1989). This SWML estimator is universal, asymptotically unbiased, robust and fast converging. Its operation somewhat resembles LZ'77 universal compression scheme (Ziv and Lempel 1977).

Non-parametric estimates were so low that the redundancy of the messages were at least 0.75. This indicates that production of the songs are governed by strong structural or syntactic constraints. Also, the estimates from parametric models were significantly higher than that from the non-parametric estimator, indicating that both memoryless and first order Markov models are too simple to model the syntax of humpback whale songs.

The empirical distribution of individual units were observed to be locally but not globally stationary. This implies that empirical Markov models of any order cannot reasonably model the song structure. The autocorrelation of symbolized song sequences exhibited repetitive structure with two different periods, approximately 8 and 200. Such a phenomenon can be simply described by assuming a hierarchy in syntax. Otherwise, without hierarchy, the model has to manage a huge number of parameters. This result is consistent with the Payne and McVay (1971) syntax.

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TM 6: Marine mammal signal processing: Accounting for the underwater channel effects

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Traveling through underwater waveguides, marine mammal signals are highly affected by the properties of the propagation medium. Thus, recorded marine mammal signals differ from originally transmitted vocalizations. Using recordings of marine-mammal signals for species monitoring and classification requires modeling of the propagation effects on the signals, in order to accurately recover the original transmissions. In this work, we study the influence of the underwater channel on the marine-mammal signals and account for it for accurate source localization and source signal deconvolution following two approaches. First, we implement a broadband, incoherent matched-field processing method for marine mammal localization and ocean transfer function estimation for deconvolution of the channel effects from the received sequence. Then, we implement an alternative localization scheme based on specific arrivals identified in correlation patterns of recorded signals. The latter method employs concepts of wave propagation but does not require full-field modeling. The method gives good localization results and is also computationally efficient, requiring fewer field calculations than broadband matched-field processing.

TM 7: Biomimetic Models of Dolphin Echolocation

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The biological sonar system of bottlenose dolphins (Tursiops truncatus) is adapted for cluttered, high noise, and extremely reverberant shallow water environments such as bays, estuaries and near shore waterways. Echolocation behavior and signals are plastic, modified in-stride during encounters with novel targets and in novel surroundings. Moreover, the dolphins' biosonar object discrimination performance provides an existence proof for biological sonar-based recognition of various kinds of targets. For the dolphin this is accomplished in part by auditory neural computations that are not yet fully understood. Although this capability has evolved over the past 60 million years, dolphin echolocation was not discovered until the late 1940's. Since then, dolphins have demonstrated acute ability to judge attributes of size, shape, material composition, whether the target is hollow or solid, and even target thickness, from the target echoes. Thus, research with dolphins provides an excellent paradigm for developing computational models of biological signal categorization. We have pursued this line of investigation by developing a biomimetic model of dolphin active target classification processes. At every stage, we have real data on dolphin discrimination performance that serves as the benchmark against which we gauge our model performance. Experimental behavior and results provide insight into the echolocation information processing strategies used the dolphins during detection and discrimination tasks. These insights motivate information flow within the model, which classifies targets by combining information from successive echoes in a multi-ping fusion neural network architecture. Psychophysical data on auditory filter shapes were combined with sensorineural analyses to design a computational model of the dolphin's ear filter. The spectral filters create principled inputs for the neural net model of the dolphin's decision processes. Currently we are incorporating recent IID and IAD data in a binaural model. Such exercises have been valuable in the long run development of effective biomimetic models for object classification in noisy and reverberant acoustical conditions.

TM 8: Sound Analysis Software and Hardware for Applications in Bioacoustics

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The development of dsp techniques and of low-cost high-speed computer hardware with large hard disks has made the computer analysis of bioacoustical signals an every-day invaluable tool for ethological research and for monitoring the underwater environment. The latest versions of our real-time Digital Signal Processing Workstation (DSPW) allows to cover most of the needs for sound recording and analysing. It is now based on Windows 95/98/NT to provide ease of use and full compatibility with the sound devices and sound analysis software available on the market.

The software package includes analysis, recording and display tools, with real-time spectrogram and cepstrogram, spectral averaging, frequency tracking, event counting, scheduled recording, etc.. Sound files and spectrograms can be saved to allow further processing with other standard software.

A portable version based on a notebook can be easily moved across laboratories or used in on-field applications, for example those requiring real-time visualisation and recording of acoustic events. Depending on the installed sound acquisition devices, analog I/O is allowed in the audio frequency range and/or in the ultrasonic range up to 150kHz. Digital I/O is also possible to provide direct transfer from DAT recorders to the PC.

High resolution real-time capabilities, typically available in much more expensive instruments, are very useful in field experiments to monitor the acoustic activities of the subjects (immediate correlation among observed behaviours and emitted/received signals) and to optimise the instrumental setup (minimisation of noise, transducer placement). These facilities allow to immediately evaluate the results of an experiment instead of waiting for later analyses on the recordings, and they make easier to analyse long periods. Examples will be shown in the technical demonstration. An evaluation version will be available very soon on the CIBRA website.

TA 1: Adaptive noise cancellation in the presence of electronic noise

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An acoustic system for use offshore is employed to keep a floating oil production unit in position relative to a fixed coordinate system. This is implemented by deploying a number of a coustic transponders in known positions at the bottom and measuring their positions with an acoustic array mounted on the hull of the surface unit. The array transducers are mounted on the surface of a sphere. There are strong noise sources mounted close to the array. The noise sources are highly directional. The single transducer signal to noise ratio varies in the range -20...+3 dB. This scenario is well suited for straight forward noise cancellation, forming reference beams in the noise directions, and this is carried out on some of the data.

Each transducer has accompanying electronics. There are limitations to the accuracy of the frontend electronics of the spherical array. The electronics inaccuracy limits the effect of the noise cancellation.

TA 2: Design of environmentally robust nonuniform arrays for shallow water mode filtering

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The pressure field in a shallow water channel is often well characterized by a finite, discrete set of propagating normal modes. Mode filtering is the estimation of the amplitudes and phases of these normal modes from the observed pressure samples obtained at a vertical hydrophone array. Many researchers have proposed the use of mode amplitudes, or coefficients, for underwater source localization or remote sensing of oceanographic features. The success of any of these techniques relies on the ability to obtain accurate estimates of the mode amplitudes.

The Cramer-Rao Lower Bound (CRLB) gives the smallest possible variance of any unbiased estimator, and [Buck, et al. JASA, April 1998] derives this bound for the variance of unbiased mode filters. The resulting expression depends on two factors: the shapes of the normal modes at the array location, and the depths of the hydrophones in the array. The former is beyond our control, but is determined by the ocean environment when and where the array is deployed. Consequently, the hydrophone locations are the only factor which we can control that affect the CRLB, the fundamental limit on the performance of the mode filter. We want to choose the array positions to give us the best performance guarantee possible given the uncertainty about the environmental factors. The difficulty lies in designing an appropriate array for the range of frequencies of interest in the notoriously variable environmental conditions of coastal waters.

The goal of our research is to develop algorithms to design hydrophone arrays whose mode filtering performance is robust to environmental variations. To achieve this goal, we use the minimax optimization criteria, which will minimize the maximum (worst case) CRLB over a range of possible deployment environments. This will design nonuniformly spaced vertical arrays whose worst case performance evaluated over a given range of environmental variations is as favorable as possible. This strategy addresses the realistic deployment scenario in which the ocean environment is not known in advance, and may vary substantially over the course of the deployment.

Conventional wisdom holds that uniformly-spaced arrays provide the most flexibility for multiple or extended deployments. We have found that in several simple prototype problems, it is possible to design nonuniform arrays whose worst case CRLB is significantly lower than a uniformly spaced array. In some of these cases, the CRLB for the nonuniform array exceeds the performance of a uniformly spaced array by nearly 8 dB. Intriguingly, all of these prototype problems exhibit convex error surfaces, suggesting that gradient-descent search techniques algorithms may provide an efficient means of designing arrays which optimize the CRLB in a minimax sense. We present results demonstrating such techniques converge to the same optimal array obtained by exhaustive searches for these prototype problems.

TA 3: The AWARE System: A Low-cost sonar for anti-grounding and collision avoidance

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Since in 1997, the Ocean Technology Center of Excellence at the University of Rhode Island funded the AWARE project to investigate the feasibility of applying US Navy torpedo sonar technology to commercial purposes. AWARE demonstrates the viability of adapting US Navy acoustic array design and signal processing concepts to a low cost, forward-looking sonar system for bottom mapping and obstacle avoidance. Prototypes of the AWARE system have been demonstrated aboard the University of Rhode Island Ocean Engineering Department's eighty-foot research vessel, R/V CT. The system has been shown effective for anti-grounding and for collision avoidance against submerged objects and cetaceans. This presentation describes the AWARE systems, and the results of their demonstrations to date.

The initial prototype system proved that it is possible to map the bottom accurately with a forward-looking sonar processor composed of commercial off-the-shelf products (with the exception of the acoustic arrays). An average range of 75 m in front of the sonar (up to four water-depths) was obtained using low cost sampling techniques. The quality and accuracy of the results were confirmed through the use of chart information and radar.

Continuing work on the AWARE project is focusing on the following areas:

1) Building a low cost acoustic transmitter and receive array with enough signal excess to achieve operational ranges of 0.5 to 1 km; 2) Implementing improved bathymetry algorithms recently developed by the US Navy which require two orders of magnitude less computation than the current algorithm. 3) Improving the quality of the bathymetry by fusing multiple sonar ping data with on-ship GPS navigation, and by adaptively estimating the underwater sound speed characteristics.

This project has involved cooperation and contributions in the form of hardware, algorithms, and engineering services from the Naval Undersea Warfare Center, Division Newport (NUWC), BBN Technologies, KVH Industries Inc., Northstar Technologies and Raytheon Electronic Systems.

TA 5: Class Specific Classification

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The performance of traditional automatic classification approaches are dominated by dimensionality issues. Specifically, probabilistic classifiers require the estimation of the joint density function of all the features given each class. For most real problems, as the number of classes increases so does the dimension of the feature set. This increase in dimension, requires an exponential and unachievable increase in the requirement for training data. This talk presents an alternative paradigm called "Class Specifics". In this paradigm low dimensional class specific models are developed and compared. The new method affords rapid in-situ training, low false alarm rate performance, accurate synthetic model generation, and a method of efficiently mining data. Examples using real data are presented.

TA 6: Real CFAR: Nonparametric, Data-Adaptive Thresholding as an Example of Tolerance-Region Signal Processing

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The broad goals of this research are to develop methods for using nonparametric, statistical tolerance intervals (1) to measure the performance of signal processing algorithms on real data and (2) to provide data-adaptive procedures for using such measures to adaptively control the performances of communication, detection, classification, localization, and tracking systems. The presentation is concentrated on the details of a specific example, the design of a nonparametric, constant false alarm rate (CFAR) , data-adaptive detection threshold.

The major objective is to develop methods for designing robust radar, sonar and communication systems using real measured data. Our approach is to provide the means, through application of tolerance regions, for such systems to quickly recognize the presence of a new, statistically different environment and quickly adapt to preserve or improve performance. Another objective is to develop methods that enable one to assess the performance of signal processing algorithms on real data so that the performances of different methods can be fairly compared using real data sets.

In summary, application of tolerance-region concepts in the design of adaptive systems can provide performance which can be reliably controlled and predicted, based on real, measured data.

TA 7: Cramer-Rao Lower Bounds for use in Towed Array Shape Estimation

D. A. Gray[1,2], B. G. Quinn[3] and J. L. Riley[4]

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It has been shown both theoretically [1] and practically [2], that provided there is a single, strong, far-field acoustic source present, the shape of a towed sonar array may be estimated from the Fourier transformed hydrophone outputs. The proposed approach uses phase information, derived from the hydrophones, to provide a ruler against which transverse displacements of the array may be measured. From an averaged estimate of the whitened cross-spectral matrix of the receiver outputs, the eigen-vector corresponding to the maximum eigen-value is first estimated. It will be shown that this can be interpreted as an unconstrained maximum likelihood estimate of each component of the steering vector. The "chord approach" in which a straight line is fitted between adjacent hydrophones is then used. This method can indeed be interpreted as a rank one approximation to the data matrix followed by a smoothing approach parameterised by the slopes of the straight lines which are derived from the rank one approximation.

In practice dynamic variations of the array shape limit the averaging time that can be used in the estimation. A useful measure of array shape estimation errors, particularly as a function of averaging time and SNR, are the Cramer-Rao lower bounds (CRLBs). These are derived for both random and deterministic signals. A two stage approach to deriving the CRLBs is considered. The first step is to derive the Fisher Information Matrix (FIM) for the each component of the steering vector without imposing the full set of constraints on the steering vector, i.e, it is not specifically assumed that the steering vector is of the form $v_j = exp[(2\pi/\lambda) * (x_j \cos(\theta) + y_j \sin(\theta)]$. Constraints to handle obvious rank one ambiguities are however imposed and having derived the FIM for the unconstrained v_j's the FIM for the parameters of the chord approximation [1], i.e., the slopes of straight lines joining adjacent hydrophones, are derived by pre and post multiplication. After some extensive algebra CRLBs are derived and it is shown how the asymptotic variances of the proposed estimators approach the derived CRLB's.

The shape of a towed array may also be estimated from compasses mounted along the array and a Kalman filter approach to this was proposed in [3,4] and experimentally verified in [5]. Fusion of the above acoustic technique was proposed in [6] and predictions on the performance of the fused approach requires a knowledge of the estimation noise power associated with the acoustic measurements. The use of the above CRLBs in this context will be discussed.

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FM 1: Some Results about the method Principal Component Inverse for the Detection with Reverberation

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In underwater acoustics, the reverberation phenomenon is considered as a source of noise. Its presence in active sonar detection makes the detection very difficult. For example, the classical matched filter is not powerful in this case. In this article, we study a method proposed by Tufts and Kirsteins to delete reverberation. The name of this method is "Principal Component Inverse" (PCI). It consists in calculating reverberation, then deleting it to have to use the classical treatments.

After a description of the method which introduces a "forward" matrix, we present some results about the rank of this matrix for two kinds of signals (monochromatic and wideband signals). The link of rank with the frequency contents of the received signal is demonstrated. The method is tested with numerical simulations and in active sonar detection data. The target is located in real reverberation noise. These results show that PCI is very efficient in some particular conditions of signal to reverberation noise ratio.

FM 2: Forward-Look Undersampled Synthetic Aperture Sonar

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It is well known from classical array theory that Nyquist spatial sampling requires an element spacing less than or equal to half the wavelength ($\lambda/2$) of the highest frequency in order to completely avoid grating lobes. Assuming free-field acoustics, this guarantees that the inter-element phase difference never exceeds p radians. In most applications of aperture synthesis design, the transmit/receive (i.e. element) spacing is half the physical aperture of the transmitter (D/2). This value is greater than $\lambda/2$ so that the resulting images have unwanted aliases or false targets. The theoretical benefit of the D/2 spatial sampling comes from the alignment of the nulls of the real aperture with the angular positions of the aliases, thereby causing the aliases to be suppressed. Even if the D/2 spacing can be achieved, aliases are still present. Inherent in the derivation of the D/2 spacing is a far-field assumption that is, more often than not, violated in synthetic aperture sonar (SAS) operation. In SAS, the need to use practical sonar platform speeds means that the spatial aperture is most often even more undersampled, i.e. the inter-ping distance is greater than D/2. Under these constraints, the resulting SAS images are practically guaranteed to contain unsuppressed aliases or false targets.

The processing presented herein is based on understanding the underwater acoustics environment and signal adaptive, ping-to-ping correlation autofocusing. This corrects for the effects of platform and target motion, and ocean instabilities. The two-way multipath group arrival structure of the channel is used to set the cross-correlation window size for the autofocusing procedure (i.e. the processing is tuned to the specific environment). Aliasing that results from the undersampled data is reduced to acceptable levels by using appropriate cross-range matched field processing. The undersampled SAS processing methodology and issues, including target coherence, autofocusing, and ping stacking, will be discussed.

Active surface ship sonar data from a recent shallow water sea trial have been processed using synthetic aperture algorithms. The test was not designed specifically for SAS application and the data are spatially undersampled. However, because the environment was stable and accompanying measured one-way transmissions from the test provide a physical understanding of the shallow water channel response, useful SAS images were formed. The acoustic environment was characterized by shallow water (depth about 100 m), downward refracting sound speed conditions, over a sand/mud bottom, during calm seas. The target of opportunity closed toward the surface ship at a relative speed of 13 knots. The test scenario not only represents an endfire condition but also a tactically significant case. In this forward-look (90 degree squint angle) condition, the SAS array resolution is considerably degraded relative to the broadside case. Nevertheless, the results show that forward-look, undersampled aperture synthesis does provide certain benefits.

FM 3: Experimental Results for Matched-Field Localization with a Small-Aperture Mid-Frequency Array

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A data set was acquired from the ACOMMS project which proved to be highly applicable to the analysis of matched-field localization performance using a small-aperture, mid-frequency array. The data set is in the mid-frequency range (3 kHz - 4 kHz), was obtained from an array with approximately a 2-meter aperture, and data acquisition was highly controlled. We have applied a subset of this data to the L-infinity Norm Estimator for robust matched-field localization, which is an extension to the well known Bartlett and maximum a posteriori estimators. Working in the small-aperture, mid-frequency case has proven quite challenging due to sensitivity and intense computational requirement issues. Localization performance using both ray theoretic and normal mode propagation models will be discussed.

FM 4: Coherent Narrowband Multitarget Tracking

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This presentation describes a method in which the contact tracking and narrow band direction finding processes are treated as a joint estimation problem, as opposed to the conventional approach which treats them as two isolated processes. A joint probability density function for the array snap shot batch with moving contacts is formulated. The constraints for the maximum a-posteriori (MAP) estimator for contact states are derived for temporally uncorrelated signals and uncorrelated contact tracks. A nested pair of Expectation Maximization (EM) algorithms are used simplify the optimization problem. The hidden data of the outer EM loop are direction finding estimates conditioned on each track distribution. This eliminates the data association step of traditional multitarget tracking approaches. The hidden data of the inner loop are the synthetic signals for each contact. This approach results in a process similar to the EM algorithm for direction finding by Miller and Furhmann, with an additional penalty term imposed by the track distribution.

Observations and interpretations of the resulting formulations are given. Simulation results for two relevant submarine towed array scenarios are presented and discussed.

FM 5: The Class-Specific Classifier and Colored Noise

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Baggenstoss [1] describes a new classification algorithm, the class-specific classifier (CSC), that exploits statistically based sufficient statistics rather than heuristically chosen features. The CSC was developed under the assumption that the underlying additive noise corrupting the signal was white and Gaussian.

In passive sonar classification, ambient noise is not expected to be white and may at times even be non-Gaussian. This presentation investigates the effects of utilizing the CSC in the presence of colored noise for the following classifier designs:

(1) Application of the CSC under the assumption that the noise is white

(2) Pre-whitening of the noise followed by the CSC under the assumption that the noise is white (i.e., ignore the effect of whitening on the signal characterization)

(3) The optimal classifier assuming knowledge of the noise spectrum

For various signal types, the loss in using (1) and (2) as opposed to (3) is determined.

The optimality of the CSC relies on the sufficiency of the features used to represent the various models. Obviously, a loss results when the features are not sufficient. A measure of sufficiency is proposed that helps to illuminate where a particular feature set lies between sufficiency and a complete lack of sufficiency, ancillarity. An ancillary feature contains no information useful for classification. Use of this measure is illustrated through examples.

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[1] P. Baggenstoss, "Class-specific feature sets in classification," accepted for publication in IEEE Trans. on Signal Processing, 1999.

FM 6: Detection and Imaging of Bottom/Buried Mines in Clutter

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The ability to detect, localize and classify (DCL) buried or partially-buried mines in littoral waters is an important problem. However, present day high frequency mine hunting sonars perform poorly against buried mines. This is because sediment penetration at traditional mine hunting frequencies (e.g., 10s of khz) is minimal or non-existent. To overcome this limitation, low frequency pencil-beams generated by a parametric sonar have been proposed for probing the sediment. Unfortunately, actual application is often difficult. The main hindrance of low frequency use is that the receive aperture is usually small relative to the transmit wavelengths due to physical constraints. The resultant large beamwidth (Rayleigh limit) makes separation of the mine echo from nearby false targets such as rocks and debris difficult.

It is well known that the Rayleigh limit is not a physical limit on array resolution, but rather the resolution limit of the delay-sum beamformer. Model-based high resolution bearing estimation techniques such as linear prediction, MUSIC, and ESPRIT can resolve scatterers more closely spaced in bearing than the Rayleigh limit and are only limited by available signal-to-noise ratio. A drawback of these methods however, is that they assume the scatterers (target and interference) to be composed of simple discrete point-like scatterers. Actual scatterers such as mines, rocks, debris, wrecks, etc. are often extended and amorphous-like, i.e., consisting of a continuum of scatterers which have an extended spatial scattering response. Traditional high resolution bearing estimators applied to this problem would either fail to resolve the target or yield highly biased estimates.

In examining the mine imaging problem we propose the use of reduced-rank representations to model spatially extended scatterers. We investigate the application of GLRT / FML and PCI high resolution detection and interference removal approaches (Kirsteins [1], MacLeod [2]) for clutter suppression to obtain high resolution cross-range imaging capability for mine hunting sonars. A key issue which we will examine is the validity of the underlying models which provide the basis for processing by principle components or maximum likelihood. By extending these models to more robust and realistic representations of the scatterers present in a reverberant field, improved reverberation suppression and imaging capability will be possible. Using actual SACLANTCEN parametric sonar data, we show that the new methods provides significantly improved resolution over the delay-sum beamformer in resolving mines from nearby false targets.

1) Kirsteins, I.P, Improved Detection of a Signal in Reverberation and Interference using the GLRT and PCI Methods, SACLANTCEN Report, Jan. 1996.

2) MacLeod, R. B. and Tufts, D. W., High Resolution Feature Extraction from Reverberation Data, In High Frequency Acoustics in Shallow Water, SACLANTCEN conference proceedings, 1997.

FM 7: Nulling Moving Sources, Snapshots and Eigenspread

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The critical problem in achieving high gains for adaptive beamforming in low frequency underwater acoustics systems is that surface ships are moving interferers that make the background non-stationary. The motion is usually significant during the interval required to obtain sufficient snapshots for covariance matrix estimation for large arrays. The movement of an interferer during an integration period causes a spread of energy across multiple eigenvalues in turn requiring more degrees-of-freedom for effective mulling. The author has introduced multirate adaptive beamforming to deal with this problem. In this paper, the effectiveness of using low numbers of snapshots and the impact of source motion are discussed and rules-of-thumb are developed and illustrated with simulation results.

FM 8: Adaptive Estimators of Output SNR for Random Channels

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When cast in the right form, adaptive detector statistics may be interpreted as maximum likelihood estimators of output signal-to-noise ratio. This means the detector statistics bring information about the quality of random channels, and this information may be used to adaptively adjust power, bandwidth, and diversity for reliable communication or detection.

In this paper we interpret the performance of these estimators of output SNR in terms of measurement dimension, signal subspace dimension, and channel stability. We derive exact stochastic representations for the estimators and use these representations to find integral formulas for their probability distributions.

Our results are applicable to adaptive radar and to adaptive multiuser communication.

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