The Journal of the Acoustical Society of America

Vol. 140, No. 4, Pt. 2 of 2, October 2016 www.acousticalsociety.org

Acoustical Society of America

Acoustical Society of Japan

5th Joint Meeting

Hilton Hawaiian Village Waikiki Beach Resort
Honolulu, Hawaii
28 November–2 December 2016

Table of Contents on p. A5

Published by the Acoustical Society of America through AIP Publishing LLC

3:55

1pSPa7. Phase DEMON algorithm for time delay estimation used in small boat tracking. Alexander S. Pollara, Alexander Sutin (Maritime Security Ctr., Stevens Inst. of Technol., 1 Castle Point on Hudson, The Babbio Ctr., Hoboken, NJ 07030, apollara@stevens.edu), Kil W. Chung (Maritime Security Ctr., Stevens Inst. of Technol., Gyeonggi-do, Hwaseong-si, South Korea), and Hady Salloum (Maritime Security Ctr., Stevens Inst. of Technol., Hoboken, NJ)

The Detection of Envelope Modulation on Noise (DEMON) algorithm is a widely used tool in underwater passive acoustics for the detection and classification of vessel sound. The DEMON algorithm extracts the frequencies that modulate the high frequency cavitation noise created by a vessel's propeller. We propose an extension of the DEMON method for time delay estimations of acoustic signals received by two or more hydrophones. This method, based on the phase difference between components in the two DEMON spectra received by different hydrophones, allows the extraction of the Time Difference Of Arrival (TDOA) and direction of arrival of the modulated signals. This method was applied to the acoustic signatures of six small boats collected by Stevens in a large glacial lake in NJ and showed agreement with the traditional cross-correlation method of TDOA estimation and boat GPS tracks. This method allows the separation of several boats TDOA. The DEMON algorithm also provides information of potential use for vessel classification. [This work was supported by DHS's S&T Directorate.]

4:10

1pSPa8. Measuring low-frequency ocean acoustic coherence with an estimator-correlator. Matthew Dzieciuch (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

Acoustic signals in the ocean are scattered by a variety of processes, but the end result is a partially coherent signal. The signal processing solution for estimating the travel-time of a partially coherent signal is to use an estimator-correlator (EC). An assumption of the EC is that the coherence time and the coherent bandwidth are known. When used properly, the EC leads to in increase in the SNR of the detected signal as well as reducing the width of the detected peak. An interesting question is "Can the EC be used to estimate the coherence parameters by maximizing the signal SNR and minimizing its width?" This is in contrast to the standard method of estimating the signal coherence from the sample covariance of the matched filter output. A simple model is constructed to test this proposition and is compared to experimental results obtained in the Philippine Sea.

4:25

1pSPa9. Perceptual thresholds of spatial audio update latency in virtual auditory and audiovisual environments. Narayan Sankaran, James Hillis, Marina Zannoli, and Ravish Mehra (Res., Oculus, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com)

When observers in a virtual sound environment are in motion relative to a source, the virtual-auditory display must rapidly track the users head position and update the location-cueing acoustic filters—known as head-related transfer functions (HRTFs)—in order to accurately reflect the source's location relative to the current head orientation and position. The end-to-end spatial audio system latency (SASL) is the time elapsed between the listener assuming a position and sound being delivered to the ears with an HRTF corresponding to that same position. To maintain the stability and plausibility of a virtual sound "object," the SASL must lie below a perceptual temporal threshold. The current study sought to rigorously probe the threshold of SASL detectability in human observers at three different rotational head velocities and at various source locations. These conditions were repeated in an audiovisual experiment in which the sound stimulus was now accompanied by a zero-latency visual stimulus. Thus, we characterized SASL thresholds in environments where cross-modal interactions occur due to visual information accompanying sound as is often the case in virtual reality (VR) applications. Initial results reveal a rich interaction between the contextual factors of the virtual environment and listeners' sensitivity to SASL.

4:40

1pSPa10. Self-shape estimation algorithm for a flexible ultrasonic array probe. Yoshiaki Nakajima, Kazuhiro Matsui, Takashi Azuma, Etsuko Kobayashi, and Ichiro Sakuma (The Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8654, Japan, nakajima@bmpe.t.u-tokyo.ac.jp)

A self-shape estimation algorithm for a flexible ultrasonic array probe was described in the paper. Position information of each element in the array is essentially required to achieve a focal control in ultrasonic imaging process. The purpose of this study is to develop an algorithm to estimate array shape without other sensors. In our proposed algorithm, beam image (BI) was used as an evaluation function in the estimation of the shape. BI is an image representing a transmitted beam profile in the imaging object. BI was obtained by scanning of receive focal points around the transmit focal point. The quality factor of BI was used as an evaluation function and parameters describing assumed shapes were searched. We conducted simulation and gel experiments with commercially available flexible probe unit. The algorithm was evaluated by the quality of an estimated self-shape and reconstructed images using an estimated self-shape. The theoretical lateral and depth resolutions were 0.5~1mm and 0.3mm in light of transmitted beam profile. The algorithm could have estimated free curve shapes in error by less than 1mm and the sizes of imaged wires were 0.74~1.5 mm and 0.625~1.25 mm in lateral and depth direction.

4:55

1pSPa11. Analysis of a baffled circular array to avoid ambiguity in detection of arrival estimation. Fabricio A. Bozzi, William S. Filho (Signal Processing, Brazilian Navy Res. Inst., Rio de Janeiro, Brazil), Fernando P. Monteiro, Fabio O. Silva (Acoust. Instrumentation, Brazilian Navy Res. Inst., Rio de Janeiro, Brazil), and Leonardo M. Barreira (Signal Processing, Brazilian Navy Res. Inst., Rua Ipiru n04, Jd Guanabara, Rio de Janeiro, Brazil, barreira@ipqm.mar.mil.br)

The number of sensors present in an array imply in cost, length, weight, and computational complexity. So, it is desired that an array satisfies its project purpose using the minimum of elements. In this study, we analyze the characteristics of a uniform circular array (UCA). Previous studies show the ambiguity problem when working with only few sensors. The grating lobes and potentials ambiguities in uniform linear array are generally avoided limiting the space between sensors in half of the wavelength. In UCA, these problems are solved also by limiting the space between sensors using an adequate number of sensors for a given radius or changing the radius for a given number of sensors. This study shows that it is possible to avoid the grating lobes controlling the sensor's directivity. The directional gain model is used to represent a realistic baffled vertical stave, three hydrophone each, and the Multiple Signal Classification (MUSIC) is applied to detect sources. Simulation results show the accuracy when using directives sensors. This array is analyzed here, working with experimental data, acquired in an acoustic tank.

5:10

1pSPa12. Acoustic distance measurement based on the interference between transmitted and reflected waves using cross-spectral method by introducing analytic signal of linear chirp sound. Noboru Nakasako, Shinya Honda (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Nishi-Mitani 930, Kinokawa, Wakayama 649-6493, Japan, nakasako@waka.kindai.ac.jp), Toshihiro Shinohara (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Kinikawa, Nishi-mitani 930, Wakayama, Japan), Masato Nakayama (College of Info. Sci. & Eng., Ritsumeikan Univ., Kusatsu, Nojihigashi, 1-1-1, Shiga, Japan), and Tetsuji Uebo (Faculty of Biology-Oriented Sci. & Tech., Kindai Univ., Kinokawa, Nishi-mitani 930, Wakayama, Japan)

The distance to a target is fundamental information in many engineering applications. Recently, an acoustic distance measurement (ADM) method has been proposed based on the interference between transmitted and reflected waves, but it requires two applications of the Fourier transform. The ADM method in which a linear chirp whose frequency changes linearly