A large-aperture array of nonlinked receivers for acoustic positioning of biological sound sources

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A system of independent recording units that can be used to form an arbitrarily large acoustic array is described. Position of units and timing of signals are obtained from Global Positioning System (GPS) with precisions within 2.5 m and 50 microseconds, respectively. An integrated hardware and software solution is presented and results reported from a four-unit feasibility test in shallow water. Sound sources at a distance of 2 km were located within 2 to 138 m of GPS-derived positions. © *2001 Acoustical Society of America.* [DOI: 10.1121/1.1323462]

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I. INTRODUCTION

A number of large mammals like whales and elephants produce sounds that travel over distances in the order of kilometers. Such sounds can disclose the position of the source when an array of acoustic sensors at known locations is used. The array generates a set of time of arrivals (TOAs) for the sound at the sensors. A position (fix) of the source can then be computed from this set.

There are three fundamental requirements for the method: (1) the position of the sensors must be accurately known; (2) sensor output must be recorded in synchrony, and (3) the aperture (size) of the array must be of the same order of magnitude as the range to be covered.

There are additional requirements such as knowledge of the sound velocity field and the need to identify a time marker in the signal at all sensors. This paper deals with an integrated approach to meet the three fundamental requirements listed above.

Passive, acoustic positioning techniques have largely been applied at relatively short ranges where the sensor positions can be fixed or monitored, and the sensors connected by cables to a central, multitrack recorder. When the aperture exceeds some 10 m, arrays tend to be cumbersome to deploy and operate. This restraint is particularly strong for work at high seas where fixed positions of sensors are not feasible. Watkins and Schevill (1972) have perfected a shipborne, four-sensor array with a 30-m separation of the elements, possibly the limit for a cabled, nonrigid, workable array at sea. A characteristic of such arrays is that references for sensor coordinates and synchronization are local, rather than global. As requirement (3) (aperture equal to range) is not met by such arrays, the potential for long-range positioning and tracking inherent to a number of biological sounds cannot be exploited.

The advent of GPS (Global Positioning System) has made it possible to meet requirement (1) (sensor positioning) by adoption of global coordinates. With this technique, precision of sensor positioning is made independent of their spacing, allowing for arbitrarily large apertures. Radio links can then be used to satisfy requirement (2) (recording synchrony), eliminating the need for cables. Thus, the third requirement can also be met. A number of different implementations of this principle have recently been published (Hayes *et al.*, 2000; Janik *et al.*, 2000; Møhl *et al.*, 2000).

This communication describes an extension of the GPS-based techniques, using global references for position as well as for time. The latter is obtained by synchronization with the atomic clocks onboard the GPS satellites. At each standalone unit of the array, sensor position information together with time information is recorded continuously on one track of a stereo recorder, while the sound signal is recorded on the other. An array built from a number of units satisfies the three fundamental requirements for long-range acoustic positioning of biological sound sources. Test results from such an array are described and evaluated.

II. MATERIAL AND METHODS

A. Overview

The array is constructed from a number of identical units as outlined in Fig. 1. One channel of a stereo recorder is allocated to the signal from the sensor via signal conditioning circuitry. The other channel is effectively turned into a digital channel by a frequency shift keying device (FSK). The latter transforms the stream of ASCII sentences from a GPS unit to a tone signal (much like in a telephone modem) that can be recorded by an audio recorder. The GPS unit (a Garmin GPS25 LV, 12-channel receiver) provides a 20-ms pulse synchronized to the atomic clocks onboard the satellites of the GPS system. The leading front of this pulse coincides with the 1-s increment of UTC time. The actual UTC time is identified in an ASCII sentence following the 20-ms synchronization pulse. The pulse is amplitude-modulating the tone signal from the FSK device, and is identified on the tape track as a sudden drop in amplitude of the FSK signal (see Fig. 2). To increase the positional accuracy beyond what is achievable with standard GPS, a special receiver for cor-

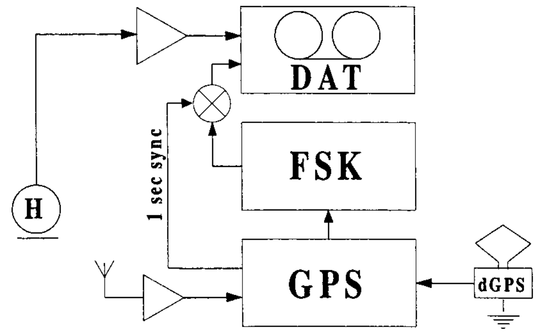

FIG. 1. Schematics of a recording unit. For explanations, see the text.

rective signals from local, ground-based radio beacons is added (dGPS, see Fig. 1.) For a general introduction to the principle of operation of this system, see Kaplan (1996).

Not shown in Fig. 1. is a palm-top computer (Psion 3MX), serving two purposes. It receives the serial output from the GPS and displays position and quality parameters of the positional fix. Second, it generates digital labels (electronic tape log) describing the recording conditions, including position of the hydrophone relative to the GPS antenna. The labels are transferred via the FSK unit to the recorder under operator control. This facility ensures that log information follows acoustic information through all subsequent copying and analyzing processes.

For data analysis, the content of the DAT tapes is transferred to CD-ROM files in WAV format, preserving the original digitization of the recorder. A standard sound-editing program (Cool Edit, Syntrillium) is used to display the tracks (Fig. 2). The time of occurrence of an event is measured from the onset of the nearest second marker. The identity of that marker, together with the coordinates (latitude, longitude) of the recording platform at that point in time, is displayed using custom software (FSK decoder, see the software section below). The FSK decoder operates on the information in the window of the sound editor (Fig. 2). Records from the other units are treated similarly, and TOAs at each sensor derived for the event. The set of TOAs is subsequently treated by the positioning software (Wahlberg *et al.*, unpublished).

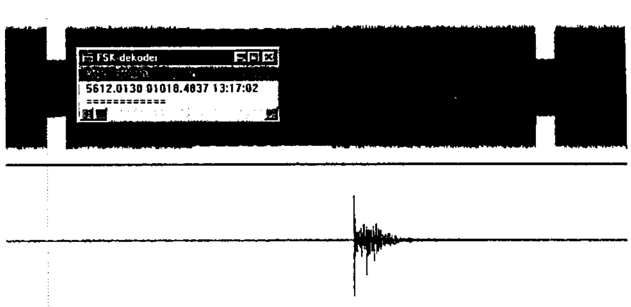

FIG. 2. Snapshot from analyzing screen. Top: FSK trace with 1-pulse-per-second markers; bottom: blasting cap signal. The insert is the decoded GPS information, showing position to be 56°12.0130′ N, 10°18.4837′ E, at 13:17:02 UTC.

B. FSK unit

The FSK unit is build around an XR-2206 chip (Exar). It converts the serial, 4800-baud signal from the GPS into a continuous tone signal consisting of a 20-kHz space tone and a 17-kHz mark tone. A sensing circuitry monitors the transmit line of the palm-top. In case of activity, the input of the FSK modulator is switched from the GPS to the palm-top by a reed relay. In this way, digital label information from the palm-top is recorded on the tape.

C. Decoding of the FSK signal

An FSK demodulator has been implemented as a set of C++ modules. The signal is stored in a WAV file as 16-bit words sampled at 48 kHz. The word stream is copied into two identical word streams. One is sent through an 18-kHz digital finite impulse response (FIR) low-pass filter, and the twin data stream is sent through an 18-kHz digital FIR high-pass filter. Each data stream is then converted to a rms stream in a software filter. The two rms values are compared in a software comparator module. If the rms value from the high-pass filter is larger than the rms value of the low-pass filter, then a 0 (space value) is output, otherwise a 1 (mark value) is output. This stream is then sent through a software state machine, which inputs a stream of bits and outputs a stream of bytes. The stream of bytes is then separated into ASCII lines. From a specific line, position and time information is extracted and output to the screen. The FSK decoding methods are not affected by amplitude modulation caused by the marker pulses.

The program for the palm-top is written in OPL language. It is menu organized and prompts the operator for information about recording and environmental conditions in a standardized way. This ensures that tape log information from all operators is uniform and linked with the sound track at all times. Additionally the program provides the operator with navigational information as well as indicators of the quality of the actual GPS fix.

D. Field test

Four recording platforms were instrumented with units as described above. The platforms were anchored in the Bight of Aarhus in an L-shaped configuration at water depths between 12 and 23 m with a separation of 1 km, on 6 May 2000. Hydrophones were lowered to a depth of 5 m. A fifth platform generating test signals was anchored 2 km away from platform 1 in a direction perpendicular to a line between recording platforms 2 and 3 (Fig. 3). The signal platform also had a stand-alone recording unit. Sensors were B&K 8101 hydrophones, supplied with power from B&K 2804 power supplies. Sensor at the source platform was a B&K 8100 hydrophone. Five Sony DAT recorders of types TCD-D3, TCD-D7, and TCD-D8 served as recorders. Anti-alias filters (see Møhl *et al.*, 2000) were used. The passband of each chain was from 0.1 to 22 kHz and deviations from flat response compensated for during analysis. Passive attenuators were used between the hydrophones and the signal conditioning circuitry when appropriate. All recording systems were calibrated with a B&K 4223 hydrophone calibra-

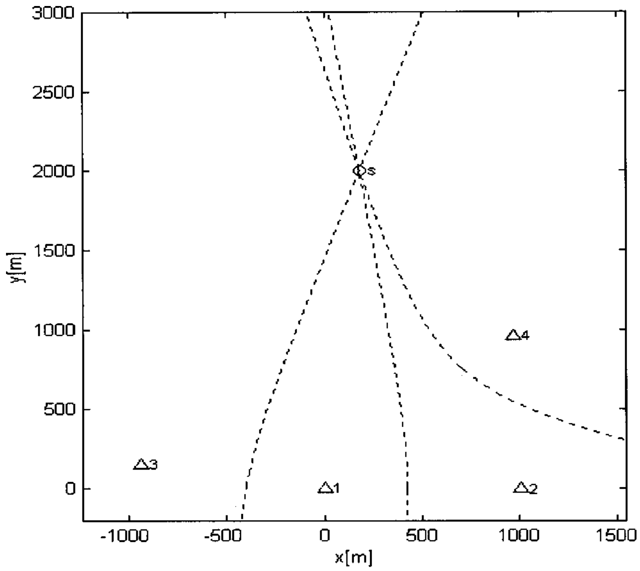

FIG. 3. Hyperbola plot for event no. 5. Receiver dGPS positions are plotted as triangles, source position as a circle.

tor. Water temperature was measured to 13.7 °C, salinity to 16.4 ppm by a Grant/YSI type 3800 water quality meter. Velocity of sound was derived from the Leroy equation (Urick, 1983), as well as from direct observation of distance and travel time of the acoustic events. Sea state was 0 to 1. Traffic from ferries and leisure crafts dominated the background noise.

A preliminary test was carried out on 12 March at the same location but with only three units. Water temperature and salinity at that occasion were 3.5 °C and 24 ppm, respectively.

E. Test signals

Five blasting caps were fired, giving rise to the acoustical events analyzed below. Source levels (SLs) for the five events were 233 dB$\pm$4 dB, as derived from measurements at the source platform at a distance of 3 m.

F. Analysis

Time of arrival of the events at each hydrophone was measured using CoolEdit to display the number of milliseconds since the preceding UTC second mark. The latter, as well as receiver position, was read from the FSK data with custom-built software described above. Source positioning and error analysis were derived from algorithms for 2D arrays given in Wahlberg *et al.* (unpublished).

For linear error propagation analysis, sound velocity was assumed to be known within $\pm$10 m/s, hydrophone positions within 2.5 m for dGPS data (data from 3600 measurements with fixed antenna), 25 m for plain GPS data, and time measurements to be accurate within 1 ms, all errors given as rms errors (2$\times$standard deviation, or 95% of the error distribution).

TABLE I. Differences (*m*) between acoustic and GPS-derived positions for five blasting cap events. Results from the linear error propagation analysis (LEP) are added.

Event	Array configuration:	Linear	Triangular	ODA[b]
1		61	138	67$\pm$16
2		37	15	34$\pm$30
3		2	3	2$\pm$1
4		6	30	3$\pm$8
5		3	21	6$\pm$4
	LEP dGPS	20[a]	100	7
	LEP plain GPS[c]	30[a]	500	70

[a]LEP analysis of a strict linear array.
[b]$\pm$rms errors calculated from residual analysis.
[c]GPS without differential correction.

III. RESULTS

Sound velocity from the Leroy equation was 1481 m/s. Direct observations yielded a mean value of 1476$\pm$8 m/s. The positional results of the acoustic events, including error analyses, are listed in Table I for three different after-the-fact configurations: a quasilinear array, composed of receivers 1 +2+3, a triangular array from receivers 1+2+4, and an overdetermined array (ODA) with all four receivers. The term overdetermined is used to signify the use of receivers above the minimum number required, which for a 2D array is three. Figure 3 shows a hyperbola plot for event 5.

IV. DISCUSSION

The difference between the acoustic source positioning and the dGPS positions ranges from 2 to 138 m, depending on event and array configuration (Table I). The larger differences coincide with cases where it was difficult to identify the onset of the event at some of the receivers (events 1 and 2). The consequences of the differences are quite small in situations where the calculated range is used to estimate transmission losses since if a spherical transmission situation could be assumed, the largest difference in Table I will result in an error of 0.6 dB. For the remaining events, the difference between ODA and GPS positions is of the same magnitude as the rms error of the dGPS itself.

The difference in performance of the three configurations is illustrative. The linear array is performing well for sources in broad side positions. Outside the triangular array there are areas with poor resolution. For further discussion of array geometry, see Wahlberg *et al.* (unpublished). Finally, an overdetermined array is advantageous both with regard to precision, and because it allows for a measured value for the magnitude of localization error. Addition of more units will improve the ODA properties.

The linear error propagation model (LEP), which gets its input from data known in advance of the actual experiment, predicts resolution reasonably well, though on the conservative side, for events 3 through 5. This indicates that one or more of the parameters entered in the computations is too pessimistic. TOA timing (set to 1 ms) is a likely candidate as resolution allowed by recorder bandwidth and signal waveform is somewhat better, at least for events 3 through 5.

The large LEP values for the triangular array are in part explained by problems of predicting the magnitude of positioning error in a source–receiver geometry close to an endfire situation (as discussed by Wahlberg *et al.*, unpublished). Still, as a tool for choosing the best configuration of receivers before deployment, LEP is clearly useful. The last row in Table I for a situation with plain GPS without differential correction is derived from LEP (receiver position rms errors set to 25 m). It illustrates the importance of precise knowledge of the position of the hydrophones.

The acoustic signals received at the test day of 6 May were characterized by a prolonged reverberation and at times by a poor definition of onset. This is contrary to the situation from 12 March where reverberation was low and onset well defined. Figure 2 is from a record of that day. Further, the source levels (SL) derived from measurements at the remote receivers from 12 March were within 11 dB $(n=8)$ of that expected from spherical spreading losses and absorption (1 dB/1000 m), while the SLs from the 6 May test were 20 to 40 dB below expectations from such mechanisms.

Compared with a traditional, cabled array, the system has some advantages and some disadvantages. Among the advantages is that the range is restricted only by the signal-to-noise ratio of the sounds, their directionality, and their timing properties, not by the array itself. With regard to analysis, we find the present system simpler/faster to operate than conventional, multitrack data when TOAs differ by more than some 10 ms. Further, the implementation of a rigidly formatted, electronic log for each platform is found to be a time-saving asset during copying and analysis. On the negative side is the impossibility of using this system for real-time tracking. Also, operating a fair number of platforms at sea presents logistic problems, even though the platforms may not necessarily have to be manned.

The principle of a globally referenced array of independent receivers was conceived as a tool for obtaining source levels and radiation patterns of clicking sperm whales, a task where it has satisfied expectations (Møhl, unpublished). It may have potentials in fields outside bioacoustics. However, the modest costs (a unit is about $2000, the recorder being the most expensive item) make it particularly interesting for bioacousticians. With this tool it should be possible not only to obtain source levels at large distances, but also to make acoustic inventories of certain vociferate populations, to make acoustic tracking, and to study long-range acoustic interactions.

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