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NOVEL ULTRASONIC BEAMFORMING METHOD BASED ON NONLINEAR FILTERING

Risto Suoranta
Machine Automation Laboratory
Technical Research Centre of Finland
P.O. Box 192 SF-33101 Tampere, Finland
Tel. 358-31-163636, Fax. 358-31-163494
Internet: Risto.Suoranta@kau.vtt.fi

Abstract

In this paper a new beamforming method appropriate especially for airborne ultrasonics is presented. The method is based on nonlinear filtering technique called Order Statistic filtering. By exploiting the proposed method we are able to implement ultrasonic range sensing system, for example, for mobile robot applications. The proposed sensing system gives the time-of-flight based distance measure and the angle of incident which is computed by applying the delay difference between sensor array elements. Measures are computed for each detected echo per one transmitted ultrasonic burst. Besides the robust method to obtain the range and the direction information the presented approach offers also an efficient computation algorithm and hardware solution to exploit it in practice.

Introduction

An appropriate environment perception is the base for autonomous operations in the area of robotics. There are several different measurement methods to collect necessary knowledge about surroundings of a mobile robot including mechanical, optical and acoustical sensing. Optical sensing is one of the most promising approaches to environment perception. Methods based on optics include image processing, laser scanners and infrared sensors.

Although optical approach has many outstanding advantages, acoustical sensing has among others one important benefit - the cost of technology. If we are able to perform high quality environment perception with low cost equipments, the range of possible applications will be very wide.

By utilizing the advanced digital signal processing methods and technology we are able to discover new sophisticated possibilities and applications for ultrasonic sensors in air.

Beamforming

The sensor array based beamforming is a widely used method of environment perception in underwater applications. When the echo arrives from a direction which is not perpendicular to the sensor array, there is a time delay between received echoes. Based on this phase shift it is possible to derive the angle of incident.

In traditional beamforming the scale of the monitored phase shift between received echoes corresponds to the wave length of the transmitted sinusoidal burst. In air the velocity of sound is much lower than in water and the wave length of ultrasonic sound is therefore much shorter. Thus it is very difficult to construct an ultrasonic sensor array system corresponding to underwater applications. In airborne ultrasonic beamforming we are forced to observe the phase shift of the envelope of the burst instead of the phase shift inside one cycle.

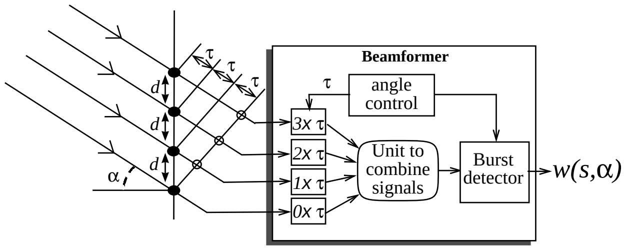

Figure 1. *The principle of beamforming.*

In Figure 1. is illustrated the principle of beamforming approach. The τ is the delay, i.e, the phase shift between channels and it defines the angle of the beam. The incident angle α can be computed using the distance d between elements, the velocity of sound v and the delay τ as follows,

$$\alpha = \operatorname{asin}\left(\frac{v\tau}{d}\right) \tag{1}$$

In the traditional beamforming the signal combining unit produce the linear combination of delayed signals. The delay can vary according to Eq.(2).

$$\tau_{water} = -\frac{1}{2}t_{cycle}\cdots\frac{1}{2}t_{cycle} \quad , \tag{2}$$

Sinusoidal signals in the same phase, i.e., with proper delay settings, will pass through the beamformer unit and sinusoidal signals with phase shift and all uncorrelated noise will be attenuated. The output $w(s,\alpha)$ is two dimensional matrix where s corresponds to the distance and α is the angle of incident. The traditional beamformer with preset angle of incident is depicted in Figure 2.

In air the distance between sensor array elements is several centimeters and thus to obtain the useful range of angles α the delay will grow up so that $\tau_{air} \gg |t_{cycle}|$. Therefore we

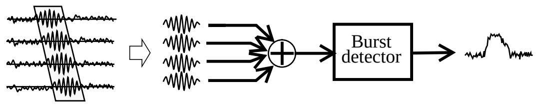

Figure 2. *Traditional beamformer with fixed angle of incident (α).*

can not use the summing to combine signals from different sensor elements.

To overcome this problem a new method to combine signals from different sensor elements is proposed. New method applies nonlinear range-filter to combine signals and to detect bursts with the same time-of-flight. In Figure 3. is shown the idea of new beamformer.

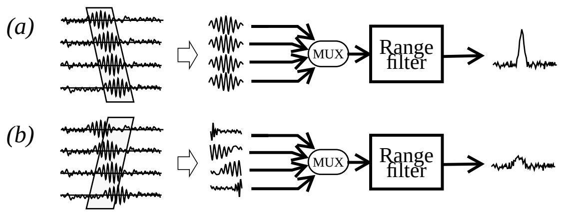

Figure 3. *Proposed new beamformer. (a) Incoming bursts are in the same phase. (b) From different angle of incident there is a phase shift between incoming bursts.*

The new beamformer produces a high output when the preset delays have removed the phase shift between bursts and the centre point of the moving window is in the center of the burst.

Theoretical background

The theoretical basis for the detection of the burst is the fact that the probability density function (*pdf*) of the pure noise and the *pdf* of the sinusoidal signal added with noise (burst) are not equal. By monitoring the selected parameters of the running *pdf* of the signal we can detect the burst. The *pdf* of the Gaussian noise and the *pdf* of the sinusoidal signal added with Gaussian noise with the *signal-to-noise -ratio* (*SNR*) 5 is shown in Figure 4.

During the burst the amplitude deviation or the range of the signal increases compared to the signal with noise only. In proposed method the monitored measure is the distance between selected percentile points in the running *pdf* of the signal. The range measure is given by

$$w_{p\%} = P_{(1-p)} - P_p \,. \tag{3}$$

P_p is the percentile point of *p%* and is defined as

$$\int_{-\infty}^{P_p} f(x)\,dx = p \,. \tag{4}$$

The range measure w_{25} is depicted in Figure 5. and it is called an *interquartile range*.

In practice the range measure w_p is estimated applying

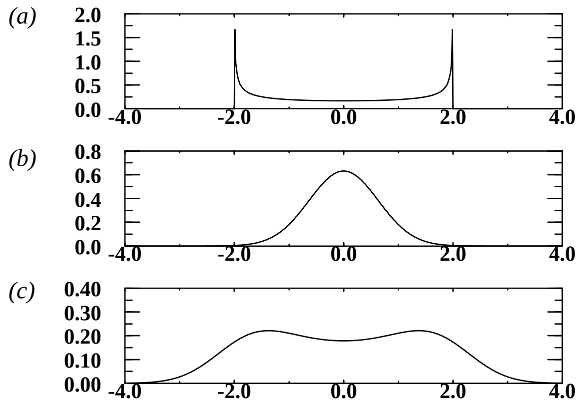

Figure 4. *Probability density functions of (a) sine, (b) Gaussian noise and (c) sine+Gaussian noise (SNR=5).*

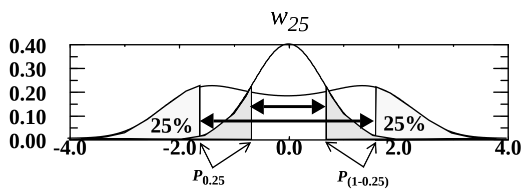

Figure 5. *Interquartile range w_{25} in the case of Gaussian noise and sine added with Gaussian noise.*

some time window to the signal. The running estimate of the range $w_{p\%}(t)$ is obtained by sliding the window over the signal as a function of time.

To consider the effects of such estimate we can make an assumption that the noise does not correlate with sine and also that the noise in time index t_1 is uncorrelated with the noise in time index t_2 when $t_1 \neq t_2$. Using these assumptions we can define the *pdf* of the signal as a function of time.That is

$$f(x,t) = c(t) f_{noise}(x) + (1 - c(t)) f_{burst}(x) \,, \tag{5}$$

where *c(t)* varies between 1 and 0 according to how much sine-signal is inside the time window; if the window contains only noise *c(t)*=1 and if the window is filled with burst signal *c(t)*=0. *c(t)* can be defined if we know the length of both the burst and the time window. $f_{burst}(x)$ can be obtained by applying the convolution to the *pdf* of the noise and the *pdf* of the sine. In our case the most interesting sequence is when the window is sliding over the noisy sine burst. In Figure 6. is shown the running *pdf* of the signal with the sine burst in the middle of the signal.

Let's consider the case with more than one signal, as it is with a sensor array in beamformer. Instead of having one time window we have window in each channel. The actual position of time window depends on besides the time index also on the delay τ which corresponds to the angle of incident α. In other words the coefficient *c* in Eq.(5) is also a function of α, i.e. *c(t,α)*. Now we can define the running *pdf* which is a function of time and beam orientation, that is

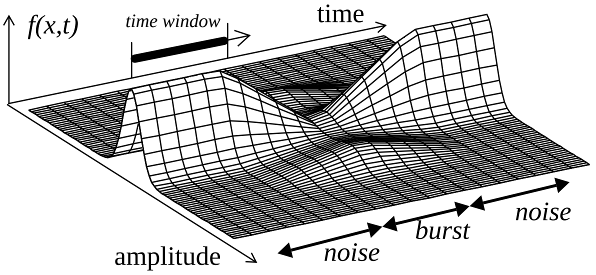

Figure 6. *Running probability density function of the signal (f(x,t)) containing sequences of noise, sine burst and noise. The noise is Gaussian noise and in noisy burst SNR=5.*

$f(x,t,\alpha)$.

The output of the new beamformer is the range measure w_p, i.e., $w(s,\alpha)=w_{p\%}(t,\alpha)$ where t is a time-of-flight and thus corresponds to the distance. The relation between $w_{p\%}(t,\alpha)$ $f(x,t,\alpha)$ can now be expressed as

$$w_{p\%}(t, \alpha) = P_{(1-p)}(t, \alpha) - P_p(t, \alpha) \ , \qquad (6)$$

where

$$\int_{-\infty}^{P_p(t, \alpha)} f(x, t, \alpha)\, dx = p. \qquad (7)$$

By applying Eq.(5), Eq.(6) and Eq.(7) the theoretical beam pattern of a new beamformer can be computed. In computation we can select the size of the sensor array, the noise characteristics, the length of used time window, the length of used sine burst or other wave forms, the signal-to-noise ratio and the $p\%$-value to be used in $w(s,\alpha)$ computation. The theoretical beam pattern of four element sensor array with $SNR=1$ is shown in Figure 7.

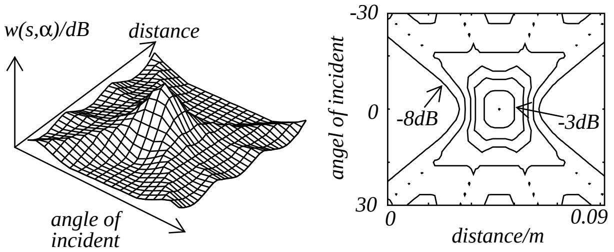

Figure 7. *The theoretical beam pattern of the proposed new beamformer. Four element sensor array with d = 12.5 cm, SNR=1, p%=25, time window = burst length = 105µs.*

Order statistic based range filter

Order statistic filters (OS-filters), also called *L*-filters, form a general class of nonlinear filters. OS-filters are widely used in the area of digital signal and image processing due to their robust behavior in the presence of outlying,i.e., erroneous observations. OS-filter is defined as

$$y(i) = \sum_{j=1}^{n} a(j)\, x_{(j)} \ , \qquad (8)$$

where $x_{(j)}$, $j=1\ldots n$ are the ordered samples of $X(i) = \{x(i-m),...,x(i),...,x(i+m)\}$ and $a(j), j=1...n$ are the filter coefficients. Filter length n equals $2m+1$. By selecting the appropriate set of coefficients $a(j)$ we can realize various different filter types like median filter, α-trimmed mean filter and rank-filter which can be considered as *percentile filter.*

To estimate the percentile point $p\%$ using OS-filter we use the coefficient set in which the coefficient corresponding to the $p\%$ is one and all other coefficients are zeros. Possible percentile points depends on the length of the filter. By applying filter with $n=3$ we can estimate percentile points of *0%* (min-filter), *50%* (median-filter) and *100%* (max-filter). If the length of the filter is n we can choose percentile points in steps of *100%/(n-1)* from *0%* up to *100%*.

To implement range filter we choose the length of the filter so that we can express the selected percentile points of $p\%$ and $100\%-p\%$. In the case of interquartile range filter appropriate values for n are $4k+1$, $k=1,2,...$ and ordered samples which correspond to the $P_{0.25}$ and $P_{(1-0.25)}$ percentiles are $x_{(k+1)}$ and $x_{(n-k)}$. By using the coefficient set in which $a(k+1) = -1$, $a(n-k) = 1$ and all other coefficients are zeros we obtain a OS-filter which equals the interquartile range filter. The structure of OS-filter based interquartile range filter is depicted in Figure 8.

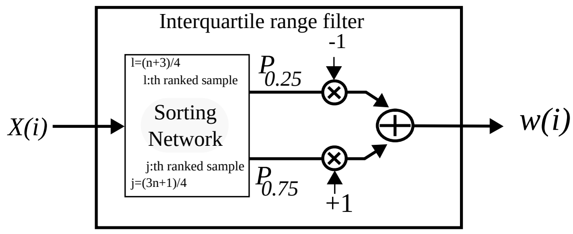

Figure 8. *The structure of interquartile range filter, n is the length of the filter and can get values of 4k+1, k=1....*

The proposed range filter is based on the theory of robust estimation. Robust statistics offers us a some safeguard against errors and outliers. Moreover nonvalid assumptions about data do not cause total invalidation of our estimates and data analysis.

In the deep sense we rely on the distributional robustness which is very important in the case of ultrasonic signals. In practical applications signals are originated from low cost sensors with significant deviation on their properties and ultrasonic signals themselves contains components having distinct distributional behavior. Above facts give a strong justification for chosen signal processing approach.

Real life trial

To develop and test proposed beamforming method we have made tests using computer simulations and prototype testing. A 3-D ultrasonic simulator using Matlab software package was programmed and the prototype ultrasonic scanning system was build for the test mobile robot ’Akseli’.

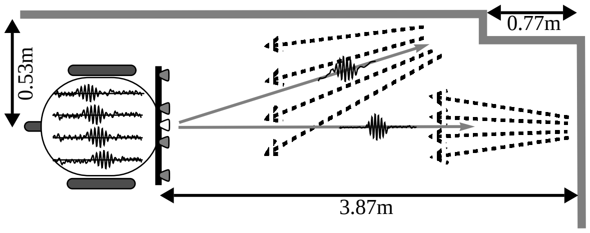

Figure 9. *Prototype testing environment used to test proposed beamforming method in real mobile robot.*

The prototype system consists of sensor array with four receiving 1" Polaroid elements and one similar transmitting element. Distance between elements is 10 cm. All necessary electronics has been built in our laboratory. Ultrasonic data is measured applying commercial A/D-board with sampling rate of 200 kHz per channel. Transmitted signal is 50 kHz burst with 10 cycles.

The algorithm for computation contains only simple integer operations like additions and comparisons and thus it is easy to implement in many different kinds of DSP architectures. In our first prototype the computation is done applying TMS320C30 DSP-board in PC 386SX. The test environment is depicted in Figure 9.

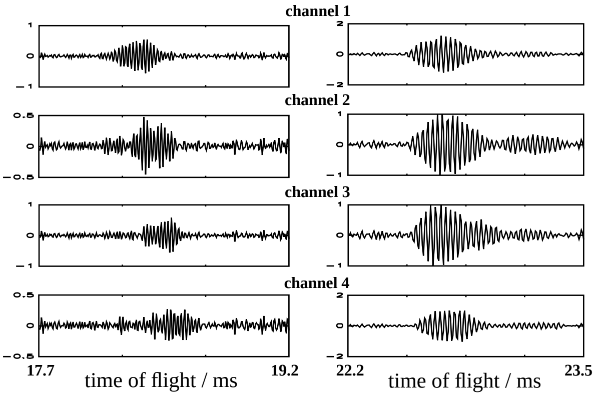

Figure 10. *Ultrasonic echoes measured applying prototype system. Left: echo from corner in sidewall, right: echo from backwall.*

Short sequences of measured ultrasonic signals are shown in Figure 10. In the left hand side are echoes coming from the corner on a sidewall (see Figure 9.) and in the right hand side are echoes bouncing from the backwall. Because we have used low cost elements there is obvious variation in sensitivity between elements. Also the different positions of receiving elements cause severe fluctuation in the envelopes of the received bursts.

However, the proposed method gives very good estimates for both the distance and the angle of incident. Measurements propose that the distance to the backwall is 3.86 m and the echo is coming from 2 degrees left. The sidewall echo is from the distance of 3.13 m with incoming angle of 12 degrees left. By utilizing the trigonometry we can compute the distance between robot and the sidewall, the result is 0.54 m and, furthermore, the distance of the small corner from the backwall is according to our measurements 0.77 m. True measures are shown in Figure 9.

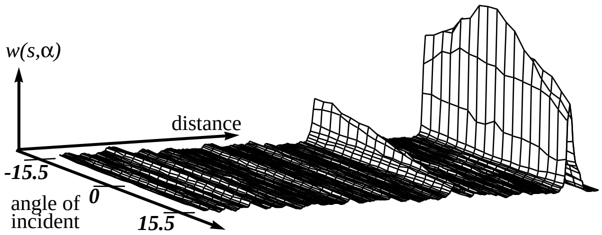

Figure 11. *The output $w(s,\alpha)$ of proposed new beamformer. In meshplot echoes from sidewall corner and from backwall can be seen.*

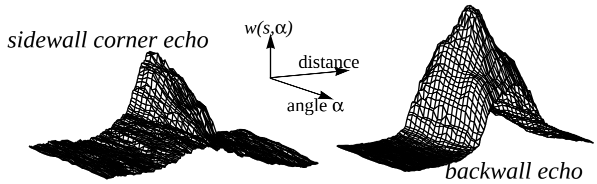

Figure 12. *Magnified full resolution meshplots of echoes (see Figure 11.)*

Summary

In this paper a new beamforming method appropriate for airborne ultrasonics is presented. The robustness of the method together with efficient low cost implementation makes it very attractive concerning various applications in robotics.

In an ongoing research project we are building a second prototype which exploits FPGA technology in computation (algorithm throughput over 100 k samples/s) and uses a new more appropriate sensor array construction which utilizes sensor elements with wide beam angle and wide frequency response.

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