Interference Cancelation in Ultrasonic Sensor Arrays by Stochastic Coding and Adaptive Filtering

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Abstract —Interference between sensors is a serious problem when ultrasonic sensor arrays are used to monitor and to explore the environment. In ultrasonic sensor arrays operating in the pulse-echo mode, the transmitters usually are excited serially by time division multiplexing. The repetition rate of the entire system is limited by the time of flight (TOF) of the pulses. For in-room applications, the TOF is in the order of 10 ms up to 100 ms due to multiple reflections. Thus, the repetition rate of an array of only 4 sensors would be in the order of 40 ms to 400 ms. Interference with signals from other ultrasonic sensor systems poses additional problems, e.g., if several robots are operating in the same area. - In this paper a method for interference cancelation is presented and experimentally verified. Ultrasonic sensors are excited simultaneously in order to accelerate the repetition. We use stochastic coding of the transmitted signals and adaptive filtering of the received signal to avoid both, interference of the sensors in the array and interference with other ultrasonic sensor systems.

I. INTRODUCTION

Ultrasonic sensor arrays are becoming increasingly important for a wide range of applications such as service robotics, automotive low-range detection systems like car parking assistants, and other security and alarm systems. Ultrasonic sensor arrays are needed for the observation of a working area, the detection and tracking of objects in this area, collision avoidance, the exploration of the environment by a mobile system, the support of navigation, etc. Ultrasonic sensor arrays can further be used to enhance the detection capabilities of other sensor systems like vision or radar systems by fusion of sensor data.

A single ultrasonic sensor operating in puls-echo mode can be used to measure a distance *d* to a sonic-reflecting object by measuring the time period Δt in which the ultrasonic pulse transmitted by the sensor travels with ultrasonic speed v_{sonic} to the object and from there back to the sensor. The the distance between the sensor and the object is then

$$d = \frac{1}{2} v_{sonic} \Delta t \,, \qquad (1)$$

An array of several sensors allows, additional to distance measurements, to localize an object inside the operating range of at least two sensors of the array. Furthermore, the sensor array can also be used to approximate the geometric dimensions and the shape of the object if two sensors operate in the so-called „cross-echo“ mode [1]. For this operation mode, each sensor has to receive the reflected ultrasonic pulse that it transmitted as well as the reflected pulse from the other sensor. The enhanced sensor information is the input for an algorithm which allows to distinguish between circular and plane objects and to identify object edges [1].

Some of the fundamental problems encountered by using sensor arrays are:

- Mutual disturbances of individual sensors within an array
- Interference of an ultrasonic sensor system with other ultrasonic sensor systems operating in the same working range

Solutions for avoiding signal interference encompasses the following approaches:

- Time division multiplex
- Frequency multiplex
- Coding of the transmitted signals

Time division multiplexing, where transmitters are excited serially, is the most common practice today. To avoid misinterpretations of reflected pulses, each sensor must simply wait for possible reflection echos before a new pulse is transmitted. However, the major drawback of this approach is the resulting long time delay between two subsequent measurements of the sensor array: For example, a time of flight (TOF) of the pulses of 100 ms due to multiple reflections would result in an unacceptable repetition time of 400 ms if the array consists of only 4 sensors. - Applying frequency multiplexing instead would be difficult and expensive from a manufacturing point of view. To overcome the disadvantages associated with the two approaches, the signals can be coded and transmitted simultaneously. This approach is elaborated in the sequel. It enables a high repetition rate and can be implemented with an acceptable effort.

II. THE METHOD OF INTERFERENCE CANCELATION

This novel method uses stochastic coding of the signals transmitted and adaptive filtering to decode the signals received. Stochastic coding means that the transmitter is excited quasi-continuously by random noise $x(t)$. The transmitted ultrasonic signal $x(t)$ is reflected by objects in the sensor's operating range and returned to the receiver, where the received signal $y(t)$ is decoded by comparing it with the original signal $x(t)$ using an adaptive filter. Despite ultrasonic disturbances $n(t)$ the adaptive filter reveals the share of the received signal $y(t)$ that originates from the transmitted signal $x(t)$ (see Fig. I). Furthermore, the adaptive filter allows to continuously measure the time lag between both signals. For this, the following assumptions are made on the disturbance signals:

- The disturbance signals are stationary and their expected values $E[n(t)] = 0$.
- The transmitted signal $x(t)$ is not correlated with the disturbances $n(t)$, i.e., the cross-correlation $E[x(t)\,n(t-\tau)] = E\,[x(t)]\;E\,[n(t-\tau)] = 0$ for all τ.

Fig. I illustrates the adaptive filter in the context of a block diagram, where it is represented as a systems identifier.

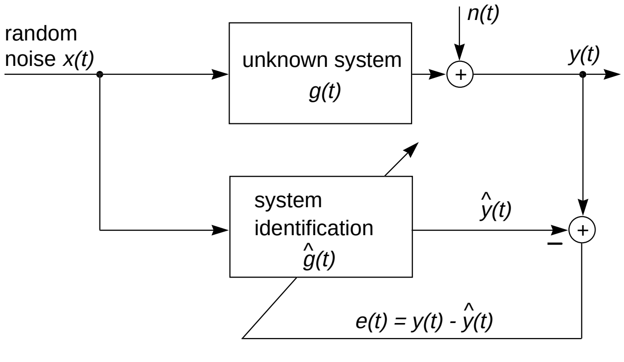

FIG. I. ADAPTIVE SYSTEM FOR ULTRASONIC DISTANCE MEASUREMENT BY STOCHASTIC CODING AND DECODING WITH ADAPTIVE FILTERING.

Considering the simplest case of a single echo returned from a single object, the unknown system is a time-delay system which can be described by the impulse response function $g(t) = \delta\,(t-\Delta t)$. In the general case of multiple echos returned from several objects, several maxima of the impulse response are obtained.

Systems identification is used to estimate the unknown impulse response $\hat{g}(t)$ in the presence of disturbances $n(t)$. For this purpose, an adaptive FIR (finite-duration impulse response) filter or, equivalently, a moving-average filter is suitable. This FIR filter can be described by the following equations, where the * denotes the convolution of two signals, the upper hat ^ estimated signals, and $e(t)$ the estimated error:

$$y(t) \quad = \; g(t) \ast x(t) + n(t) \qquad (2)$$

$$\hat{y}(t) \quad = \; \hat{g}(t) \ast x(t) \qquad (3)$$

$$e(t) \quad = \; y(t) - \hat{y}(t) \qquad (4)$$

There are numerous algorithms to get an estimate of the unknown impulse response $\hat{g}(t)$ [2, 3]. The LMS algorithm is chosen for the sake of simplicity to minimize the mean squared error $e(t)^2$ yielding the recursion

$$\hat{g}^{new}(t') = \hat{g}^{old}(t') + se(t)x(t-t') \qquad (5)$$

to update the filter coefficients using the step-size parameter s. It should be noted that the LMS algorithm applied here is not optimal since it actually uses the quadratic error $e(t)^2$ instead of the expected value $E[e(t)^2]$ of the error function. The deviation from the optimum (Wiener) solution depends on the step-size parameter s [3]. The smaller the values of s the better are the estimates - but at the cost of a slower convergence of the adaption. In our application, large values of s are chosen to obtain a fast adaptation. A fast adaptation is needed for high performance in case of moving objects which imply time-varying impulse responses $g(t)$. The optimum value of s is determined by simulation. The optimum is achieved when local maxima of $g(t)$ are detectable with a signal-to-noise ratio greater than two (SNR > 2).

III. SIMULATION RESULTS

The actual implementation of an ultrasonic distance-measurement system using a digital control unit will be different to the block diagram of Fig. I in some respects. As shown in Fig. II, the analog signals $x(t)$ and $y(t)$ are substituted by $x_k = x(t{=}k\Delta)$ and $y_k = y(t{=}k\Delta)$, which are the values of the time signals at the k-th time instant with sampling period Δ. The random series of numbers x_k that correspond to the random time signal *x(t)* are generated digitally. The system identifier will work with time-discrete signals as well, i.e., $\hat{g}(t)$, $\hat{y}(t)$ and *e(t)* are substituted by $\hat{g}_k$, $\hat{y}_k$, and e_k.

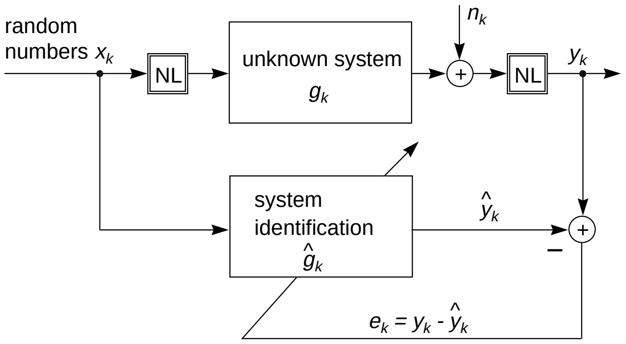

FIG. II. DIGITAL ADAPTIVE SYSTEM FOR ULTRASONIC DISTANCE MEASUREMENT BY STOCHASTIC CODING AND DECODING WITH ADAPTIVE FILTERING.

Due to the physical realization of the transmitter/receiver unit, the real transmitted and received signals will be some nonlinear function of x_k and y_k. To take this into account, two nonlinear blocks NL are introduced in Fig. II.

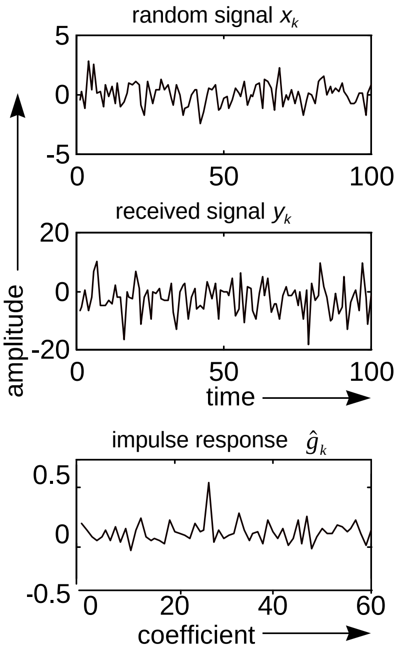

FIG. III. SIMULATION OF THE TIME-OF-FLIGHT MEASUREMENT WITHOUT THE NONLINEARITIES “NL“ OF FIG. II.

For simulation purposes, the unknown system *g(t)* in Fig. II was modeled by a delay of 27 sampling periods ($d=27\Delta$), and the disturbance *n(t)* was modeled by normally distributed random numbers n_k. Values for the random input-signal x_k were chosen normally distributed as well. The mean amplitude of the disturbance signal n_k was 5 times the mean of the signal values x_k transmitted. For systems identification, a FIR filter with 60 filter coefficients was used, and the step size s = 0.0005 was chosen.

A simulation result of the system in Fig. II is shown in Fig. III without the two nonlinear functions NL. It can be seen from the impulse-response plot that the system identification to estimate $\hat{g}_k$ reveals a distinct maximum at the 27-th filter coefficient despite of the disturbance. Thus, the modeled delay could correctly be identified.

A second simulation result is shown in Fig. IV with the same model and filter but with the two nonlinearities NL shown in Fig. II. The nonlinearities under consideration are saturation functions that impose upper and lower bounds on the values x_k and y_k, i.e., they limit the amplitude of the transmitted and received signals. It can be seen from the impulse-response plot of Fig. IV that compared to Fig. III the nonlinearities did not affect the position of the maximum value of $\hat{g}_k$ although they had impact on the course of the signal. Thus, the unknown system model of *g(t)* could again be correctly identified.

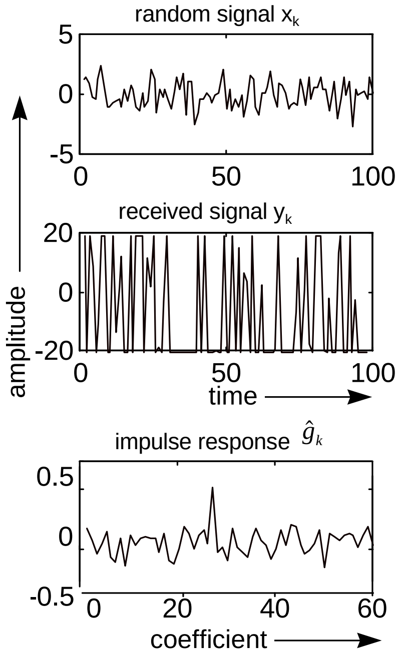

FIG. IV. SIMULATION OF THE TIME-OF-FLIGHT MEASUREMENT WITH THE NONLINEARITIES “NL“ OF FIG. II.

Further simulations of sensors operating in the pulse-echo mode confirmed these results. A robust systems identification could be achieved even under severe constraints imposed by nonlinearities.

IV. EXPERIMENTAL VERIFICATION

For an experimental setup, Bosch ultrasonic sensors of large horizontal and vertical spread-angles of up to 120° were used, which are primarily manufactured for use in automotive applications like parking assistance systems (see Fig. VIII). In the puls-echo operation mode, the sensor uses the same piezo-ceramic disc to transmit pulses and to receive echos. The received echo signals are preprocessed, comprising band pass filtering and threshold-logic algorithms, by an integrated circuit inside the sensor box. The sensor's output is binary signals that reflect the time instants of arriving echos. An example of the sensor's output signal is given in Fig. V.

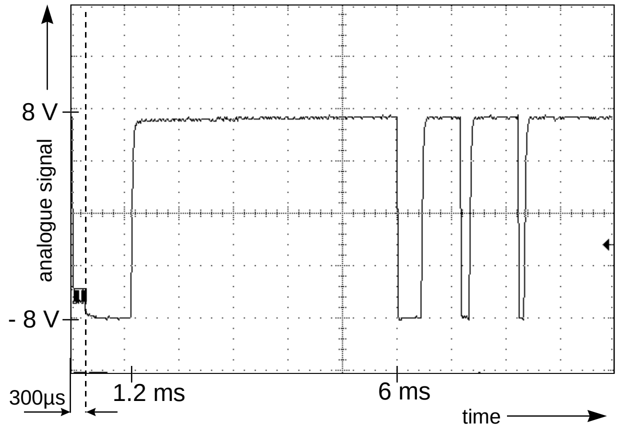

FIG. V. OUTPUT OF A SINGLE BOSCH ULTRASONIC SENSOR WITHIN THE SENSOR ARRAY UNDER INVESTIGATION.

As it can be seen in Fig. V, each sensor transmits ultrasonic pulses of a duration of approximately 300 µs. After additional 900 µs, the ultrasonic resonator is ready to receive echo signals, which allow the sensor to measure distances bigger than 20 cm. In Fig. V, a first echo signal is detected after 6 ms which corresponds to a distance of approximately 1m. The further two echos recveived are multiple reflection echos.

A simple experiment with two identical sensors operating simultaneously is described in the following. One sensor is used to measure distances and the other one to generate ultrasonic disturbances. In front of the two sensors, an object is manually moved back and forth, which describes the behaviour of a time-variant system.

In Fig. VI, the estimated amplitudes of the impulse response are plotted versus time, where each amplitude value of $\hat{g}_k$ is graphically represented as an amplitude-proportional brightness. Thus, the sampling instant of the k*-th filter coefficient, where $\hat{g}_{k^*} = \max_k(\hat{g}_k)$, corresponds to the white "trace" that can be seen in Fig. VI. Since these values represent the TOF of the ultrasonic pulses transmitted, they are proportional to the time-varying distance as indicated at the horizontal axis of Fig. VI.

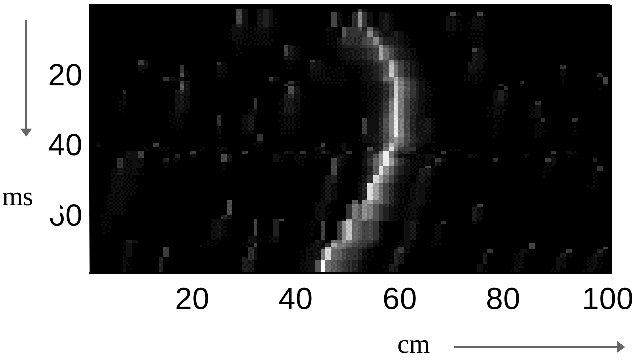

FIG VI. INTERFERENCE CANCELATION BY STOCHASTIC CODING USING BOSCH ULTRASONIC SENSORS OPERATING SIMULTANEOUSLY.

An extension of this experiment is to move the sensor through a small corridor towards an object. This allows us to investigate the detection capability for echos reflected from the left- and right-hand walls as shown in Fig. VII (a).

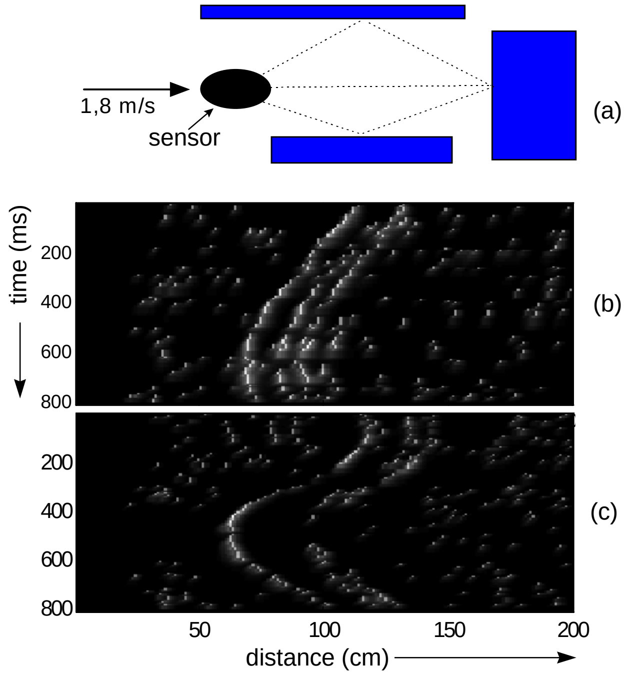

FIG VII. INTERFERENCE CANCELATION BY STOCHASTIC CODING USING BOSCH ULTRASONIC SENSORS: (a) EXPERIMENTAL SET-UP, (b) SYSTEM IDENTIFICATION RESULT FOR LOWER, AND (c) FOR HIGHER SPEED.

The estimated impulse response coefficients are shown in Fig. VII (b) for a speed of 1.8 m/s and stochastically generated pulses transmitted in the time span of 1.5 to 10 ms. As above, the first white “trace“ corresponds to the distance to the object. The next two traces are caused by the two reflection echos. The estimated distances of the moving sensor corresponded to the real situation observed. Fig. VII (c) shows the results of this experiment for higher speed.

As the experimental results show, the stochastic coding and the adaptive-filter decoding gives information about the distance and the directional motion of objects despite the simultaneously generated disturbances. Thus, a robust object tracking can be achieved by the ultrasonic sensor array under consideration.

FIG. VIII. SINGLE ULTRASONIC SENSOR USED WITHIN AN ARRAY COMPRISING A MAXIMUM OF 8 SENSORS.

To control the ultrasonic sensors and to decode the received signals by adaptive filtering, a digital control unit was developed (see Fig. IX), which is based on the digital signal processor ADSP2181/16MHZ from Analog Devices. This control unit is used to monitor experimental results by either directly analyzing d.c. voltages on a screen or connecting its serial RS232 interface to another computer.

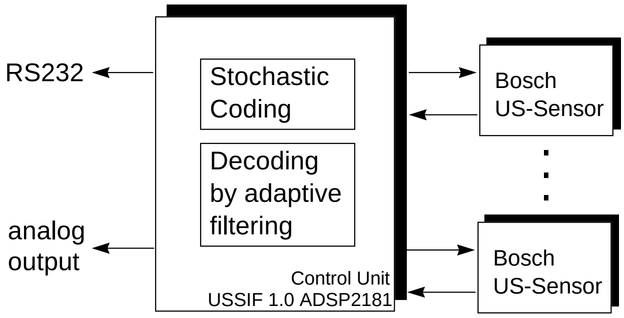

FIG. IX. BLOCK DIAGRAM OF THE CONTROL UNIT FOR STOCHASTIC CODING AND DECODING BY ADAPTIVE FILTERING.

V. CONCLUSIONS AND FUTURE WORK:

It could be shown by simulations and experimental verifications that the proposed method for interference cancelation for ultrasonic sensor arrays allows to localize obstacles and to track objects despite of disturbances caused by simultaneously operating sensors. The method is based on stochastic coding of signals transmitted and adaptive filtering of echo signals received.

Further work related to the results presented so far focuses on the following issues:

- Investigation of the SNR for multiple-sensor operation, and comparison of the SNR of time multiplexing with that of stochastic coding.
- Optimization of the adaptation mechanism for moving objects, where the speed of adaptation is a function of the estimated error.
- Combination of the proposed method with Kalman filtering.

ACKNOWLEDGEMENT:

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