Digital Time-of-Flight Measurement for Ultrasonic Sensors

Daniele Marioli, Claudio Narduzzi, Carlo Offelli, Dario Petri, Emilio Sardini, and Andrea Taroni

Abstract—**Ultrasonic sensor measurements are mostly based on the determination of the time of flight (T.o.F.). The paper presents the development of a digital algorithm for pulse-echo measurement applications, based on the use of a cross-correlation function to determine the T.o.F. Some experimental results are presented, and the possibility to realize a low-cost real-time measurement system is considered.**

I. Introduction

Ultrasonic sensors are widely used in industrial applications to measure both object distance in air and flow velocity. The operating principle is based on the measurement of the time of flight (T.o.F.), that is, the time necessary for an ultrasonic wave to travel from the transmitter to the receiver after being reflected from the target. The object distance from the transducer is $d = [c \times (\text{T.o.F.})]/2$, where c is the velocity of sound.

Different techniques can be used to generate ultrasonic waves. Among them, continuous wave [1] and pulse-echo techniques [2], [3] are widely known. More complex methods involving the modulation of either continuous or pulse waves have been reported, for instance in [4]–[6].

In continuous wave methods, the transmitter generates a continuous output, whose echo is detected by a separate receiver. Accuracy depends on the measurement of phase shift between the transmitted and reflected wave. Although better performances than with pulse-echo measurements can be obtained, complex hardware is required to measure phase and, in most cases, different frequencies need to be used in order to determine the number of integer wavelengths in the phase shift.

Pulse-echo techniques are often used in commercial systems for industrial applications. With this method a short train of waves is generated, enabling the same transducer to be used both as a transmitter and as a receiver. The simplest and most common way to detect the echo signal is the threshold method, where detection occurs when the signal crosses a given amplitude level [7]. Although it has proved to be simple and cheap, the technique may suffer from poor resolution, particularly when the echo pulse has been greatly attenuated. An improvement consists in the adoption of an adjustable amplitude threshold, in which case detection does not depend on the magnitude of the signal but only upon its shape [8]. In the presence of noise due to air turbulence, and/or mechanical vibrations close to the transducer operating frequency, both methods are known to achieve a repeatability of some tenths of the ultrasonic wavelength. This corresponds to different distance values, according to the transducer operating frequency and possible variations in sound velocity.

In this paper the use of digital signal processing methods in a pulse-echo measurement application is presented. The transmitted and echo pulses are transduced into electrical signals, digitized, and their cross-correlation is computed in digital form. The time index of the peak of the correlation function measures the delay between the two signals, that is, the T.o.F.. The reasons for this approach are mainly two: 1) from a conceptual point of view, signal processing algorithms can provide improved resolution and at least comparable accuracy, and 2) a measurement system based on dedicated signal hardware is at present not only feasible, but also competitive in terms of cost and performance.

An analysis of the proposed method will be carried out with particular emphasis on the algorithms that have been selected. Consideration will also be given to the problem of real-time measurement that motivates some of the choices in the implementation. Finally, experimental results will be presented.

II. Analysis of the Problem

Measurement of the time of flight can be considered in the general framework of time delay estimation. The transducer generates a signal, $s(t)$, that propagates to the target and is reflected back, being detected after a delay D. The T.o.F. measurement system has to determine this time interval.

The propagating medium introduces attenuation that increases with frequency, and can distort the reflected wave. However, if the pulse is narrow-band, attenuation by a constant coefficient α can be assumed for the delayed echo, so that one can express it in the form $\alpha s(t - D)$. In practice, these hypotheses are satisfied only approximately. Furthermore, external disturbances such as turbulence and vibrations, as mentioned in the previous section, may affect the signal waveform, and quantization noise is introduced by the conversion process.

Manuscript received May 12, 1991; revised September 26, 1991. This work was supported in part by CNR, Progetto Finalizzato Trasporti.

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IEEE Log Number 9105571.

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The measurement system acquires the two digital sequences $x_T(nT)$ and $x_E(nT)$ representing the transmitted and echo signals, respectively, that can be written in the form

$$x_T(nT) = s(nT) + v(nT);$$
$$x_E(nT) = \alpha s(nT - D) + n(nT) \qquad (1)$$

where T is the sampling interval, while $v(nT)$ and $n(nT)$ take into account the discrepancies from the ideal model and can be considered as zero mean uncorrelated random processes.

The cross-correlation of the two sequences is given by:

$$C(kT) = \sum_{n=-\infty}^{+\infty} x_T(nT)x_E(nT + kT). \qquad (2)$$

According to the hypotheses given above on $v(nT)$ and $n(nT)$, the statistical expectation of this sequence is:

$$E[C(kT)] = \alpha C_{ss}(\tau - D)|_{\tau=kT} \qquad (3)$$

where $C_{ss}(\tau)$ is the auto-correlation function of the continuous signal $s(t)$. For a finite energy signal it can be shown that $|C_{ss}(\tau)| \leq C_{ss}(0)$. Therefore, if k_D is the index of the peak of $E[C(kT)]$, one has $D = k_D T$ when the delay is an integer multiple of the sampling interval, and $D = (k_D + \delta)T$, with $|\delta| \leq 0.5$, otherwise. In practice, the delay can be estimated by finding the maximum of the cross-correlation function (2).

The signal generated by an ultrasonic transducer is a short train of acoustic waves that can be represented in the form:

$$s(t) = a(t) \cdot \sin(2\pi f_o t + \phi_0) \qquad (4)$$

where f_o is the transducer resonant frequency, and the pulse $a(t)$ represents the signal envelope and has finite duration. For this class of signals one may write:

$$C_{ss}(\tau) \cong C_{aa}(\tau) \cdot \tfrac{1}{2} \cos 2\pi f_o \tau \qquad (5)$$

where $C_{ss}(\tau)$ is the auto-correlation of the envelope function $a(t)$. The typical behavior of $C_{ss}(\tau)$ for the transmission pulse of an actual ultrasonic transducer is depicted in Fig. 1. Since $s(t)$ is a narrow-band signal, the envelope of $C_{ss}(\tau)$ (i.e., $C_{aa}(\tau)$) decreases slowly, while the correlation function itself is highly oscillatory, so that it becomes difficult to find the actual maximum among a number of peaks with similar amplitudes. Thus, when the delay D is estimated by searching for the maximum of $C(kT)$, the attainable resolution is often no better than one period $1/f_o$ of the sinusoidal term in (5), even when the sampling interval T is much shorter. This corresponds to a distance resolution of one half of the ultrasonic wavelength λ. Again, it must be noted that the actual value of λ can change with variations in the values of c and f_o.

III. The Measurement Algorithm

The algorithm that is presented in this section has been implemented with the aim of achieving a resolution of the same order of magnitude as the sampling interval T for

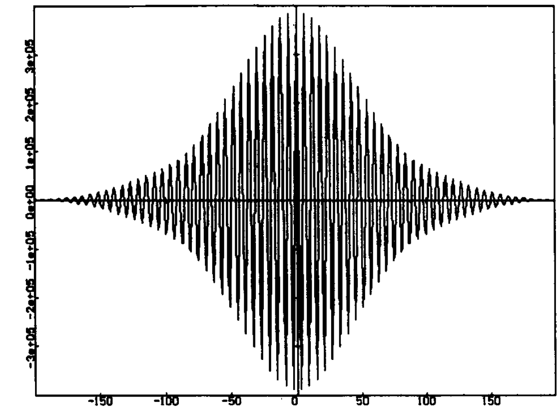

Fig. 1. Auto-correlation of a typical transmitted pulse.

the estimation of the delay D. It is based on the consideration that the term $C_{aa}(\tau)$ in (5) is also an auto-correlation function, having the same properties as $C_{ss}(\tau)$. This suggests the possibility of estimating the delay from the cross-correlation of signal envelopes, with the advantage that improved resolution can be expected, since the sinusoidal term is no longer present [9].

Depending on the distance to be measured, the T.o.F. can be quite large, resulting in very long data sequence if the echo signal is sampled continuously. However, since $s(t)$ in (4) has finite duration, an interval occurs between the transmitted and echo pulses, where the only relevant information is the count of samples falling within.

If the onset of the echo can be detected by some simple experiment, the sequence of echo samples, $x_E(nT)$, can be much shorter. In fact, it suffices that its length N is equal to that of the sampled transmission sequence $x_T(nT)$, the only requirements being that the time interval NT is longer than the duration of $s(t)$ and that the echo is entirely contained in the sequence. In this case the T.o.F. is determined by algebraically adding the delay estimate provided by the algorithm to the number of samples that were counted before the acquisition of the echo sequence. The main advantage lies in the reduced computations involved in processing shorter sequences.

The first step of the algorithm is to recover from the sequences $x_T(nT)$ and $x_E(nT)$ their sampled envelopes, that will be indicated as $y_T(nT)$ and $y_E(nT)$, respectively. It is apparent from (4) that the transducer output can be interpreted as an amplitude modulated signal having carrier frequency f_o; this suggests the use of demodulation. After analyzing and trying different methods, the choice was to realize in digital form a square law envelope detector. This is a simple kind of demodulator that does not require accurately determining the carrier frequency f_o, thus avoiding the need for a specific setup of the system if different ultrasonic transducers must be used.

A low-pass filter is included in the detector to eliminate the spectral components that are generated around the frequency $2f_o$ as a consequence of squaring, and also to reduce the noise content in the signal. The measured am-

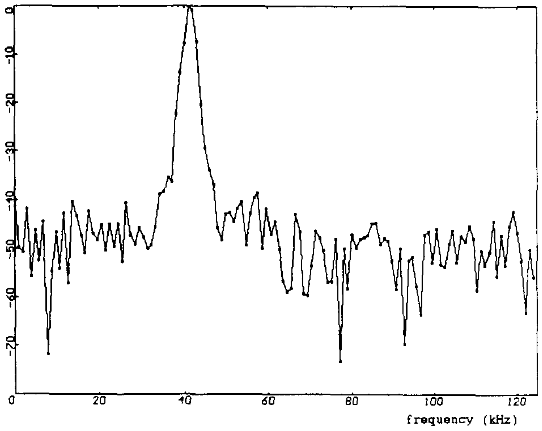

Fig. 2. Amplitude spectrum of the transmitted signal.

plitude spectrum of the transmitted signal is shown in Fig. 2 for one of the sensors that has been used in the experiments. The signal can be seen to have quite a narrow bandwidth; therefore, the low-pass digital filter can also have a low cutoff frequency. Since the same filter must be used both for the transmitted and the echo sequence, the signal-to-noise ratio (SNR) of the latter signal dictates such features as transition bandwidth and stopband attenuation, which ultimately determine the order of the filter.

If the bandwidth of the envelope $a(t)$ is B_a, the highest frequency in the signal $s(t)$ is $f_o + B_a$, and the sampling interval T must be chosen such that $T < 1/2(f_o + B_a)$. Therefore, the baseband signals that are obtained by demodulation are in general much oversampled, so that the sequences $y_T(nT)$ and $y_E(nT)$ can be decimated without any loss in information. From a computational point of view decimation by a factor k (with K an integer) has the advantage that a lower number of data samples is processed in the operations that follow, since a sequence with N samples has its length reduced to N/K. However, it should be remembered that the sampling interval is increased as a consequence.

A possible solution to avoid any loss in the resolution of the delay estimate lies in the interpolation of the samples of the obtained correlation sequence. The correlation of the demodulated and decimated sequences $y_T(nKT)$ and $y_E(nKT)$ can be written as:

$$C(kKT) = \sum_{n=0}^{(N/K)-1} y_T(nKT) y_E(nKT + kKT),$$

$$k = -\frac{N}{K} + 1, \cdots, \frac{N}{K} - 1. \quad (6)$$

With k_D the index of the peak sample in the sequence, $D = (k_D + \delta)KT$, with $|\delta| \leq 0.5$. Thus, if the delay D has a fractional component (i.e., $\delta \neq 0$), the peak of the corresponding continuous correlation function does not coincide with the position of the sample of index k_D.

The fractional index δ can be determined from the samples of $C(kKT)$ through an interpolation formula. Although its analytical expression can be given, in most practical cases the relationship is unlikely to be satisfied exactly. Therefore, it was decided to avoid this complication and rely instead on an approximate interpolation formula. Given the samples with index k_D, $k_D - 1$ and $k_D + 1$, a simple second-order polynomial approximation of $C(\tau)$ yields a parabola, and the fractional index δ is obtained as the abscyssa of its vertex. Deleting the term kT from the argument of $C(kKT)$ for simplicity:

$$\delta = \frac{C(k_D + 1) - C(k_D - 1)}{2 \cdot [2C(k_D) - C(k_D - 1) - C(k_D + 1)]}. \quad (7)$$

Simulation analysis of this approximate relationship showed that it introduces an error which in most cases is only marginally larger than with exact interpolation formulas.

IV. Experimental Results

The method discussed in section III was verified by means of the experimental setup shown in Fig. 3. T_1 is a 40-kHz ultrasonic transducer that acts both as a transmitter and as a receiver, converting an electrical signal into an acoustical one and vice versa. Signals from T_1 are filtered by a band pass amplifier, whose center frequency is synchronous with the transducer operating frequency, and acquired by a digitizer for which a 250-kHz sampling rate was adopted. Both the transducer and the digitizer are controlled by the processing unit, where the data sequences are stored and processed.

This system arrangement enables both $x_T(nT)$ and $x_E(nT)$ to be acquired. However, it should be noted that the signal corresponding to the echo is the result of a double conversion process. In fact, the electrical signal applied to the transmitter is converted into an acoustic wave, and the acoustic echo is converted back into an electrical signal. Differences can occur, due to the transducer bandwidth and to the electric impedances of the transmitter and receiver circuits. In order to avoid differences between $x_T(nT)$ and $x_E(nT)$ due to mismatches of the above parameters, it has been chosen to digitize $x_T(nT)$ through a separate receiver, according to the arrangement of Fig. 4.

In this scheme T_1 and T_2 are two matched 40-kHz ultrasonic transducers; T_1 acts as the transmitter, while the other is the receiver. Transmission parameters (amplitude and length of the pulse train) have been rigorously standardized. The processing unit simultaneously drives the transmitter and the digitizer, whose input is connected to the receiver and accepts the acquired data from the digitizer for storage. Strictly speaking, the resulting sequence $x_T(nT)$ is delayed by the time of flight between the transducers T_1 and T_2. However, the difference can be corrected as a result of calibration procedures.

With the described system a set of experiments has been carried out for different transducer-to-target distances, with reference distance measurements being obtained by laser interferometry. The T.o.F. has been determined according to the algorithm described in the previous section,

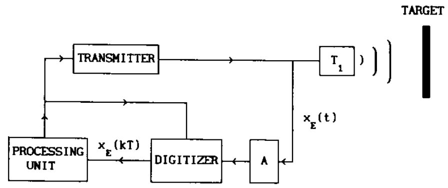

Fig. 3. Block diagram of the experimental set-up.

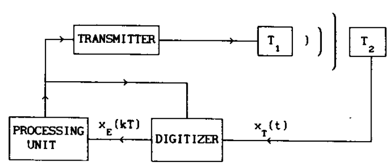

Fig. 4. Acquisition of the reference transmitted pulse.

using data sequences whose lengths are 256 points. Different decimation factors have been tried; eventually, decimation by 4 was settled for, but factors up to 8 still give similar results. To turn T.o.F. into a distance measurement, the velocity of sound under the actual experimental conditions must also be known.

Experimental data are reported in Table I. All distances are measured with respect to a common reference point; that represents zero distance; the T.o.F. related to this point is always subtracted from subsequent measurements. This enables compensation for a number of possible factors that introduce offsets, such as time delays between the digitizer trigger and the actual start of the transmit pulse, constant biases in the evaluation of the origin for measuring distance, or the T.o.F. difference affecting $x_T(nT)$ in the measurement scheme described above.

Once the offset is corrected for, a second reference measurement can be taken on a target placed at a known distance to determine a proportionality constant between the T.o.F., expressed in number of sampling intervals, and the distance. Although not strictly necessary, the use of a second reference point avoids the need to determine by other means the sound velocity and then, rely on the long-term stability of the digitizer timebase. With the employed digitizing rate, one sampling interval approximately corresponds to a distance variation of 0.7 mm.

Size limitations in the interferometer set-up unfortunately precluded the measurement of distances above 1.2 m. To account for this limitation and to extrapolate, partly at least, to longer distances, the transducer driver has been operated at reduced power to simulate a greater attenuation.

It would appear from Table I that by using the interpolation equation (7), the T.o.F. can be determined to within fractions of the sampling interval. Comparison of the obtained results with the reference measurements partly supports this claim, since discrepancies lower than 0.5 mm have been obtained in some cases. However, further work is needed to fully assess the performance of the measuring system.

TABLE I
EXPERIMENTAL RESULTS IN DISTANCE MEASUREMENTS

Reference (mm)	Measured Distances (points)	Measured Distances (mm)	Difference (mm)
0	0	0	—
100.0	146.0	100.7	+0.7
203.5	296.8	204.8	+1.3
306.0	443.7	306.1	+0.1
407.2	589.1	406.5	−0.7
503.8	729.5	503.4	−0.4
600.8	869.8	600.2	−0.6
698.8	1013.6	699.4	+0.6
799.4	1159.1	799.8	+0.4
898.9	1303.5	899.4	+0.5
998.7	1448.7	999.6	+0.9
1101.1	1598.1	1102.7	+1.6
1200.2	1736.9	1198.5	−1.7

V. Final Remarks

The approach presented in the paper provides interesting performances in the measurement of the time of flight with cheap ultrasonic transducers. In the experiments carried out so far, a good agreement with reference data has been obtained; without external noise the attainable resolution is of the order of a few tenths of the ultrasonic wavelength. A preliminary analysis suggests that, with noise added, similar results can be obtained up to a SNR of about 10 dB, if care is taken to process the sampled data sequences by appropriate digital filters.

The measurement algorithm can be implemented on state-of-the-art digital signal processing systems at moderate cost. A preliminary assessment of the computational effort required by the algorithm to process 256-point data sequences, with decimation by 4 after demodulation, shows that a measurement can be obtained in less than

1 ms on a DSP chip. This would enable the realization of a low-cost real-time measuring instrument that can be easily integrated, for instance, into a more complex automation system.

References

[1] T. Ono, M. Kohata, and T. Niyamoto, "Ultrasonic phase-sensitive rangefinder with double modulation doppler-free method for shallow seafloor survey," in *Proc. IEEE 1984 Ultrasonic Symp.*, Dallas, TX, Nov. 14-16, 1984.

[2] P. Kleinschmidt and V. Magori, "*Ultrasonic remote sensors for non-contact object detection*," *Siemens Forsch.-u. Entwickl.-Ber.*, vol. 10, Nr. 2, 1981, pp. 110-118.

[3] D. Marioli, E. Sardini, and A. Taroni, "Ultrasonic distance measurement for linear and angular position control," *IEEE Trans. Instrum. Meas.*, vol. IM-37, pp. 578-581, Dec. 1988.

[4] W. K. Kim, S. B. Park, and J. A. Johnson, "Signal-to-noise ratio and bandwidth for pseudo-random codes in an ultrasonic imaging system," *Ultrasonic Imaging*, vol. 6, pp. 113-123, 1984.

[5] A. K. T. Lee, J. Lucas, and L. E. Vin, "Microcomputer-controlled acoustic rangèfinding technique," *J. Phys. E: Sci. Instrum.*, vol. 22, pp. 52-58, 1989.

[6] L. Foulloy and G. Mauns, "An ultrasonic fuzzy sensor," in *Proc. Int. Conf. Robot Vision and Sensory Controls*, Zurich, Switzerland. Feb. 2-4, 1988.

[7] T. M. Frederiksen and W. H. Howard, "A single-chip monolithic sonar system," *IEEE J. Solid-State Circuits*, vol. SC-9, pp. 394-403, Dec. 1974.

[8] J. D. Fox, B. T. Khuri-Yakub, and G. S. Kino, "High frequency acoustic wave mesaurement in air," in *Proc. IEEE 1983 Ultrasonic Symp.*, Atlanta, GA, Oct. 31-Nov. 2, 1983.

[9] S. Chow and P. M. Schultheiss, "Delay estimation using narrow-band processes," *IEEE Trans. Acoust. Speech Signal Proc.*, vol. ASSP-29, pp. 478-484, June 1981.