Journal of Southeast University (English Edition) Vol. 21, No. 1, pp. 20 - 23 Mar. 2005 ISSN 1003—7985

Ultrasonic wind velocity measurement based on DSP

Tang Huiqiang[1] Huang Weiyi[2] Li Ping[1] Lü Qinghua[1]

([1] Department of Information Engineering, Nanjing University of Information Science and Technology, Nanjing 210044, China)
([2] Department of Instrument Science and Technology, Southeast University, Nanjing 210096, China)

Abstract: Ultrasonic transmitting, receiving and amplifying circuits are designed. The received signals are sampled with the high speed ADC (analog-to-digital converter), and dealt with the DSP (digital signal processing). A forward-backward IIR (infinitive impulse response) filter with no delay is designed to filter the sampled data, and series A and B are achieved by narrow and wide band filtering, respectively. In series A, the start point of the cycle first exceeding the threshold is calculated accurately by interpolation, and the start cycle is detected by fitting cycles in series A and its inversion A′ to cycles in B with variance analysis. Therefore, the start point of the start cycle is calculated precisely. By deriving the relationships between the travel time in the opposite directions of three axes and the airflow velocities, the wind velocity and direction are calculated. Experiments show that the reliability and the precision are improved, and the circuits are simplified.

Key words: ultrasonics; IIR (infinitive impulse response) filter; DSP (digital signal processing); wind velocity

The sound velocity in the air is related to parameters such as air density, temperature, humidity, air pressure and airflow velocity, etc. Generally instant airflow changes the sound velocity, while the other parameters remain unchanged in the short term and can be ignored. So the airflow velocity in a direction can be calculated by measuring the sound velocity or the travel time to-and-fro. The typical measures are as follows: By overlapping the transmitted and received waves through adjusting transmit frequency, the travel time can be calculated, and the sound velocity can be measured with the accuracy of about 0.02%[1], but it is not fit for automatic measurement. By measuring the phase shifting between the transmitted and the received waves, high accuracy can also be acquired, but correct result cannot be guaranteed with the phase shift more than 180°. By measuring the time from transmitting to the start point of the received wave, or to the peak of the first cycle, the sound velocity can be measured, but high accuracy is difficult to achieve by common methods due to the slightness of the received start cycle. However with DSP, the wind velocity can be measured by processing the ultrasonic travel time, with the precision improved and circuits simplified; moreover, ultrasonic systems with DSP have also been realized in compensating ultrasonic sensors and high accuracy ultrasonic range measurement[2,3].

1 Working Principle

Suppose that the components of airflow velocity v in 3 right-angle axes x, y, z are v_x, v_y, v_z and the ultrasonic velocity in static air is c. A pair of transmitting-receiving ultrasonic transducers T_1, T_2 are in a distance of 331 mm in opposite direction of axis x. The travel time from T_1 to T_2 is t_1, and the time from T_2 to T_1 is t_2. By the equation of wavefront, $(x - v_x t)^2 + (y - v_y t)^2 + (z - v_z t)^2 = (ct)^2$, t_1 and t_2 can be obtained.

$$t_1 = \frac{s\left(\sqrt{c^2 - v^2 + v_x^2} - v_x\right)}{c^2 - v^2} \tag{1}$$

$$t_2 = \frac{s\left(\sqrt{c^2 - v^2 + v_x^2} + v_x\right)}{c^2 - v^2} \tag{2}$$

Therefore the component of the airflow velocity in axis x can be calculated by Eqs. (1) and (2) as

$$v_x = \frac{s(t_2 - t_1)}{2t_1 t_2} \tag{3}$$

Similarly, v_y and v_z can also be calculated. By compound of v_x and v_y, the wind velocity v_{xy} and the wind direction $\theta = \arcsin(v_x / v_{xy})$ can be calculated, and the air flow velocity v can further be synthesized by v_{xy} and the vertical component v_z.

In the perfect gas, Celsius temperature T and ultrasonic velocity c have the following relationship:

$$T = 0.002\,489c^2 - 273.16 \tag{4}$$

By multiplying Eq. (1) and Eq. (2), we get c^2, then put c^2 into Eq. (4), and the instantaneous temperature T can be calculated by

$$T = 0.002\,489\left(\frac{s^2}{t_2 t_1} + v^2\right) - 273.16 \tag{5}$$

Therefore, if we know the distance s between transducers, by measuring the ultrasonic travel time t_1, t_2 in two opposite directions, the wind velocity and the temperature can be solved.

Received 2004-07-10.
Foundation item: The Natural Science Foundation of Higher Education Institutions of Jiangsu Province (No. 02KJD460005).
Biography: Tang Huiqiang (1965—), male, doctor, associate professor, thq@jlonline.com.

2 Circuit Philosophy

The ultrasonic transmitting-receiving circuit is shown in Fig. 1. The 40 kHz transducers are selected because ultrasonic waves can be well coupled in the air. Transducers transmit ultrasonic signals with a high-voltage pulse excitation. Each transducer has a transmitting circuit as shown in Fig. 1 instead of using analog switches that lead to lower performance though simplifying the circuit. In the circuit, with the level of C_1 low, the NMOS FET Q_1 is off, and the voltage U_{GS} of the PMOS FET tube Q_2 is nearly zero leading to Q_2 off, so low level is outputted. With C_1 high, Q_1 is ducted, and the grid-voltage of Q_2 is divided by resistors R_5 and R_9. Q_2 is on with U_{GS} lower than threshold voltage, and high level is outputted. Hence, the positive high-voltage pulses can be generated by control of C_1 to excite the transducer.

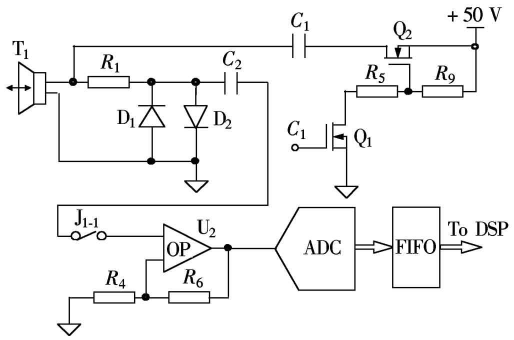

Fig. 1 Ultrasonic transmitting-receiving and ADC circuit

In the receiving circuit, the current limiting resistor and two Schottky diodes confine the input voltage within about 0. 4 V to prevent high voltage from into the receiving circuit but have no effect on the slight received signals. The non-inverting amplifier amplifies the received signals about 200 times to acquire more information. Zero level is adjusted to fit for the input range of the A/D converter. By using the inexpensive 8-bit A/D converter TLC5510, each A/D conversion can be accomplished within a single CLK cycle. For timing precisely and freeing from data lost, the 2 K × 9 bit IDT7203 FIFO is used to buffer the data before connecting to the DSP circuit to lower the real-time requirement of DSP. Read of A/D converter values and control of the circuit and process of data are all performed by the DSP system that is comprised of TI's TMS320C32. It can also be performed by PC104 embedded module, even PCs can be adopted for convenient programming in some ultrasonic detecting systems[4].

3 Calculation of the Travel Time

Fig. 2 shows the output of a receiving transducer varying with time from the beginning of transmitting. In practice, the received signals are amplified about 200 times to accurately locate the start point of the received wave, while the noise is also amplified and clipping distortion occurs near the peak of the envelope. The time from the beginning of transmitting to the start point of the received wave needs to be measured, but the first cycle of received wave is very slight and submerged in noise. The start point of the first cycle is even difficult to locate, while * the amplitude of the successive wave increases gradually. Common methods may delay the start point for even more than one cycle, though the average method[5] can reduce errors. A new processing method is developed in this paper to increase the measurement accuracy.

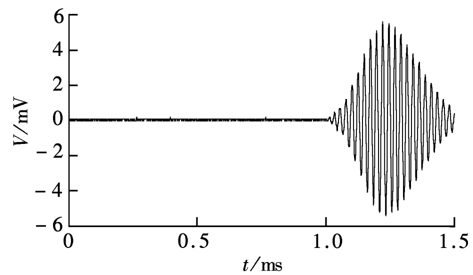

Fig. 2 Received ultrasonic waveform

3.1 Filtering the signals

The received signal is in the narrow frequency band with a slight frequency shifting for the Doppler effect, and the beginning signal frequency is a little higher than the transmitting signal. Due to the existence of background and circuit noises, power line frequency interference and offset voltage, the received signals should be filtered to decrease noise and diminish DC component. A band pass IIR elliptic filter is suitable, with its low order to satisfy the same requirement of the FIR filter. Its transfer function based on Z transform is

$$H(z) = \frac{b(1) + b(2)z^{-1} + \ldots + b(n_b + 1)z^{-n_b}}{a(1) + a(2)z^{-1} + \ldots + a(n_a + 1)z^{-n_a}} \quad (6)$$

The nonlinear phase delay cased by IIR filter can be diminished in the following ways[6] shown in Fig. 3. First, filtering the received series according to Eq. (6), then reversing the filtered series order and filtering it again, finally reversing the resulting series to its original order. The transfer function based on Z transform is

$$H'(z) = H\left(\frac{1}{z}\right)H(z) \quad (7)$$

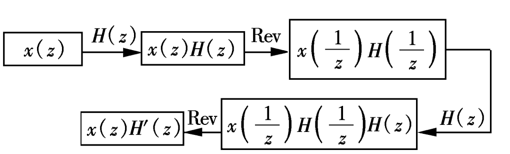

Fig. 3 Filtering method without delay

Set the sampling frequency to be 1 MHz, the pass band [35, 50] kHz, stopband attenuation 60 dB, and the ripple 0.01 dB, the IIR filter can be designed with 3 orders for both high pass and low pass. The coefficients of this IIR filter are $b=(0.0018, -0.0052, 0.0049, 0.0000, -0.0049, 0.0052, -0.0018)$, $a=(1.0000, -5.5002, 12.7975, -16.1164, 11.5844, -4.5072, 0.7420)$. All the poles of the filter are within the unit circle, and the filter is stable. To save process time, only signals near the received front are processed. The series filtered by this filter is called series A shown in Fig. 4. After filtering, the noise is greatly attenuated, and the clipping distortion generated by over amplification is recovered. But damped vibration caused by filtering interferes with the discrimination of the first cycle received. Series A is used to accurately locate zero crossing time.

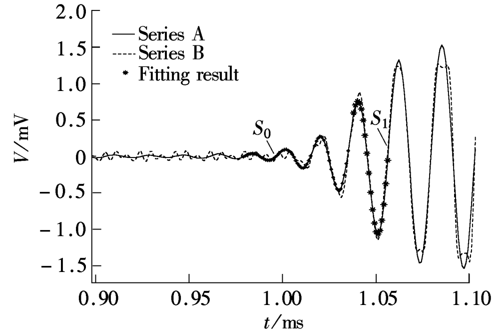

Fig. 4 Series after filtering and the fitting result

Increase of bandwidth can diminish the damped vibration. Set the pass band to be [20, 160] kHz, stopband attenuation 60 dB, passband ripple 0.01 dB, a 3-order passband filter can be designed. The coefficients of the filter are $b=(0.1298, -0.0134, -0.3624, 0.0000, 0.3624, 0.0134, -0.1298)$, $a=(1.0000, -3.1412, 3.9924, -2.9962, 1.6497, -0.5604, 0.0562)$.

The filtering result of the signals is shown in Fig. 4. Due to the large bandwidth, the filtering effect gets worse, and the clipping distortion does not disappear, but DC component and the damped vibration are diminished. This series is called series B.

3.2 Locating the start point of the wave

In this system, by calculating the SSD (sum of squares of deviations) between series B and series A, and the SSD between series B and the reversed series A′, the ratio of the two SSDs is calculated to judge the start cycle. Analysis of the data of the received signal in Fig. 4 is as follows:

1) In series A, compare the data with a threshold voltage along the time axis from left to right and confirm a point which first exceeds the threshold within several cycles of the wave received. Find the zero crossing point 1/4 cycle ahead of the threshold point to settle the start point named S_1. Locating of S_1 in this cycle is accurate for the perfect noise reduction to series A.

2) On the left of S_1 of series A, take the data of a full cycle and the data of the corresponding cycle in series B, and judge if the cycle of series A could fit to cycle of series B. Generally, we only test if the amplitude V_B of B cycle is equal to zero by t-test or F-test with null hypothesis H0: $V_B=0$. For increasing the sensitivity, the ratio of the SSD between the cycles in A and B to the SSD between the cycles in reversed A′ and B is calculated. Suppose the distribution of noise is normal, the SSD between $\hat{y}_i$ (the data of a cycle in A) and y_i (the data of a cycle in B) is $S_{E1}=\sum_{i=1}^{n}(y_i-\hat{y}_i)^2$, and the SSD between $\hat{y}_i'=-\hat{y}_i$ and y_i is $S_{E2}=\sum_{i=1}^{n}(y_i+\hat{y}_i)^2$. The two variances are equal, i.e. $\sigma_1=\sigma_2$, if the amplitude $V_A=0$. There are no constraint conditions between y_i and $\hat{y}_i$, and between y_i and $\hat{y}_i'$, therefore the degree of freedom is the sample number n, so $S_{E1}/\sigma_1^2 \sim \chi^2(n)$, $S_{E2}/\sigma_2^2 \sim \chi^2(n)$, then

$$S_{E2}/S_{E1} \sim F(n,n) \tag{8}$$

By formula (8), the hypothesis testing can be performed through variance analysis. The value of F with 99% degree of confidence is relative to the sample number only. If sampling frequency is 1 MHz, i.e. 25 samples per cycle, then $F_{25,25}(0.01)=2.61$. Hence, when $S_{E2}/S_{E1}<2.61$, the hypothesis is true, it is judged only as received noise, otherwise, it is judged as a received cycle. If the sampling frequency increases to 5 MHz, i.e. 125 samples per cycle, $F_{125,125}(0.01)=1.52$. Therefore the judgment effect of the start cycle improves notably compared with that of an ordinary curve fitting.

3) Test the cycles of series A one by one from right to left with step 2) until the fitting effect is no longer significant. The fitting results are shown in Fig. 4 and Tab. 1. The SSD ratio of the two cycles on the right side is greater than 2.61, it is significant. The ratio of the third cycle on the left side is only 1.785, so it is not significant, and it is not a received cycle. Only 3 cycle's testing is needed to locate the start cycle in the example. Subtracting two wave periods from the position of S_1, we can get $t_1=995$ in 1 MHz sample rate in Fig. 4, so the travel time from transducer T_1 to T_2 is 0.995 ms. Obviously, this method can save data processing time if an appropriate threshold point is selected in step 1).

Tab. 1 Results of variance analysis

Cycle	S_{E2}	S_{E1}	S_{E2}/S_{E1}
1	6.4511	0.1377	46.860
2	0.4731	0.0749	6.314
3	0.0701	0.0393	1.785

Similarly, the ultrasonic travel time t_2 in opposite directions along axis x and those along the other two axes can be solved.

By the sound speed formula $c = s/t$, the error of wind velocity Δv caused by time measurement error Δt is deduced as

$$\Delta v = c\Delta t/(s/c + \Delta t) \tag{9}$$

With the transducer frequency of 40 kHz and the distance of 331 mm, ultrasonic travel time is about 1 ms, a cycle delay $T_s = 25$ μs may result in an error of wind velocity about 8. 1 m/s. Even a 1 μs measurement error may make an error about 0. 33 m/s. Therefore the locating precision of the start point can be improved by increasing the sample frequency.

With interpolation, the start point can also be located accurately with a relatively low sample rate. Suppose the voltage of the rough start point S is y_1 in series A. If $y_1 < 0$, the zero cross point should be on the right of S, and take the right side data y_2; if $y_1 > 0$, the zero cross point should be on the left of S, and take the left side data y_2. With linear interpolation, the time error Δt can further be calculated as

$$\Delta t = |y_1|/(y_2 - y_1) \tag{10}$$

Hence, the actual ultrasonic travel time is $t + \Delta t$. The whole cycle measurement error can be avoided because the wind velocity change between twice continuous measures does not exceed about 8. 1 m/s.

4 Conclusion

Ultrasonic wind velocity measurement based on DSP simplifies processing hardness compared with the methods of sensitivity time control, signal envelope detection, and extremum detection with differential processing. The measurement precision is close to the pulse echo superposition method. Due to the measuring process completed in several milliseconds, the instantaneous airflow can be calculated. The average or the maximum wind velocity, and the vertical airflow can also be calculated. Compared with a wind vane, hot wire anemometer, and wind pressure set, there is no start up wind speed, no distance constant. The flexibility of numerical processing methods makes it convenient to be improved and applied to other measurements such as the flow rate of gas or liquid, and the level of liquid or material.

References

[1] Wang Yinguan. The precision measurement theory of ultrasonic wave speed and its application[J]. *Chinese Sound Technology*, 1989, **8**(2): 18 − 23. (in Chinese)

[2] Sabatini A M. A digital signal-processing technique for compensating ultrasonic sensors[J]. *IEEE Transactions on Instrument and Measurement*, 1995, **44**(4): 869 − 874.

[3] Parilla M, Anaya J J, Fritsch C. Digital signal processing for high accuracy ultrasonic range measurements[J]. *IEEE Transactions on Instrument and Measurement*, 1991, **40**(4): 759 − 763.

[4] Zhang Yu, Wang Yuanyuan, Liu Bin. Development of a Doppler ultrasonic spectrum analysis system based on a personal computer[J]. *Chinese Journal of Scientific Instrument*, 2001, **22**(4): 365 − 368. (in Chinese)

[5] Zhu Shiming, Liu Zhengqing, Wei Moai. The zero crossing digital average measurement method to increase measurement accuracy [J]. *Chinese Sound Technology*, 1990, **9**(3): 36 − 39. (in Chinese)

[6] Gustafsson F. Determining the initial states in forward-backward filtering [J]. *IEEE Transactions on Signal Processing*, 1996, **44**(4): 988 − 992.

基于 DSP 的超声风速测量

唐慧强[1]　黄惟一[2]　李　萍[1]　吕清华[1]

([1] 南京信息工程大学信息工程系, 南京 210044)
([2] 东南大学仪器科学与工程系, 南京 210096)

摘要: 设计了超声波发射、接收、放大等电路, 采用高速 A/D 转换器采集超声波接收信号, 利用 DSP 进行数据处理. 设计 IIR 带通滤波器对采集信号进行前后向滤波以消除延迟, 通过窄带及宽带滤波分别取得序列 A 与 B. 利用序列 A 通过插值取得首先超过阈值的某一周期的精确起始时间, 通过序列 A 及反相序列 A′分别与序列 B 的对应周期进行方差分析, 利用假设检验来识别接收波形的起始周期, 从而求出超声波的传输时间. 通过超声波在三坐标轴向来回的传输时间与空气流速间的关系, 求出三维风速与风向. 结果表明, 测量可靠性及精度得到提高, 电路设计也得到简化.

关键词: 超声波; IIR 滤波器; DSP; 风速

中图分类号: TP274$^+$.5