A DSP based Signal Processing System of Vortex Flowmeters

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Abstract: The IIR wavelet filter banks are adopted to decompose the sensor output signal of vortex flowmeters in order to extract the flow information from the vortex signal with noise in this paper. A digital signal processing system with a digital signal processor (DSP) is developed to realize the wavelet transform and frequency measurement algorithms in real time. The experimental results show that the method and system are effective for vortex flowmeters.

Keywords: vortex flowmeters, wavelet transformation, IIR filter banks, frequency measurement, digital signal processor

1. Introduction

Vortex flowmeters are widely used due to their many advantages, such as good linearity, no moving parts, low pressure-loss and the ability to measure liquid, gas and steam. Their measurement accuracy, however, in fields and range-ability are to be improved. The main cause is that vortex flowmeters has not the ability to eliminate noise from vortex flow signal using the standard signal conditioning electronics that includes a charge amplifier, a Schmitt trigger, the period counting circuitry and a microprocessor. Under ideal conditions the signal sensed by the vortex sensor i.e. piezoelectric elements in the vortex flowmeters would be sinusoidal. In reality the vortex sensor signal comprises a fundamental signal that has a fundamental frequency representing the flow and noise or harmonic wave at various frequencies caused by fluid turbulence and other factors such as pipe vibrations, common mode pressure variation and so on. The noise problem becomes particularly serious when the bluff body and piezoelectric elements are mounted on the end of an elongated steam or probe inserted into the mid-portion of the fluid steam in a cantilever mount.

In order to solve this problem, many modern digital signal processing methods for processing vortex flowmeter signal have been studied. Gerald L. Schlatter used Fourier analysis and cross-correlation techniques to eliminate the noise pulses from the vortex signal [1]. Meng Jianbo et al. proposed an adaptive frequency measurement method (AFM) for vortex flowmeter signals [2]. Masanori Hondoh et al. used the adaptive band pass filter for spectrum analysis of vortex signal [4]. K. J. Xu et al. adopted an adaptive notch filtering approach to estimate the frequency of the flow signal, and to reject noise in the flow measurement and enhance the performances of vortex flowmeters [5]. K. J. Xu et al. utilized wavelet transform to decompose the vortex signal, and calculated the frequency representing the flow [6]. Though many methods are researched for processing vortex flowmeter signal, fewer papers are related to design a digital signal processing system for realizing those algorithms.

In this paper the method based on wavelet transform is approved. A digital signal processing system is developed with a DSP to acquire the vortex signal and realize the algorithm in real time. Finally experiments are carried out to test the method and system proposed by this paper.

2. The processing method based on wavelet transform

The algorithm based on wavelet transform needs two steps to estimate the fundamental frequency representing the flow information. The first step is extracting signal fundamental wave with wavelet decomposition, removing high order harmonics or other noises from the vortex signal; the second is estimating the frequency from the extracted fundamental wave. The process of the method is showed in Fig.1.

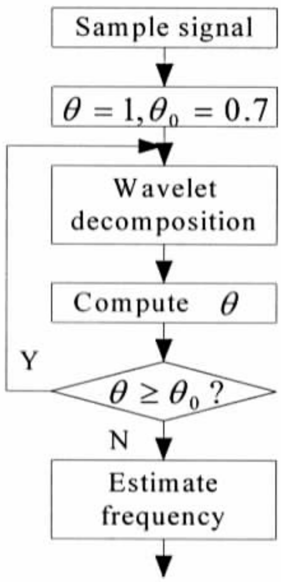

Fig. 1 Process of fundamental frequency wave extraction

Discrete wavelet transform decomposes the sampled signal into signal approximation and signal detail, which are obtained by filtering the signal with a low-pass filter, a high pass filter respectively, and called the scale part, wavelet part of signal wavelet decomposition. In Fig.1, θ is a ratio of the energy of the scale part to that of signal and θ_0 is the threshold of θ, which ranges between $0 \sim 1$, in this paper, $\theta_0 = 0.7$. When $\theta \geq \theta_0$, the fundamental frequency is still included in the scale part of wavelet transform, and the scale part is to be decomposed by wavelet transform; as $\theta < \theta_0$, the fundamental frequency is included in the wavelet part, and the extraction process ends. The characteristic of filters is an important factor influencing the accuracy of extracting fundamental frequency. All realized filters have transition band, which brings two side effects on fundamental frequency extraction. The first is the overlay between the high-pass filter and the low-pass filter, which can cause that the wavelet part including the fundamental frequency may comprise higher order harmonics or other noises. The other effect is that the fundamental frequency lying in the transition band included in wavelet part would be attenuated. The two kinds of effects both can reduce the algorithm precision, and so the filters with short transition band should be adopted. In this paper, Butterworth filter is selected, having comparatively better characteristics with Db5 filter, showed in Fig.2. In Fig.3, the curve 1 is the magnitude frequency characteristic of the Butterworth high pass filter, and the curve 2 is that of Db5 high pass filter.

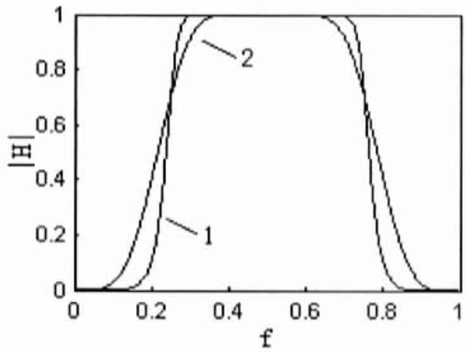

Fig. 2 Comparison between Butterworth and Db5 filter

After the fundamental wave is extracted with wavelet decomposition, to improve the accuracy of frequency estimation, the following steps are adopted to estimate the frequency: ($\{x(n), n = 1, \cdots, Ns\}$ is the discrete fundamental signal having been extracted)

(a) Pick out all maximum value points from the extracted fundamental wave, and place their corresponding sampling positions sequentially in a sequence T_2. The judging rule is:

$$\{x(i) \geq x(i-1)\} and \{x(i) > x(i+1)\}$$

For high order harmonics or other noises may be remained in the wave, the local maximum value points of the wave can also be picked out.

(b) Get rid of the local maximum value points that haven't maximum value, and the remained real maximum value points are placed in the sequence T_1.

(c) For non-integer sample, the real maximum value points may not be sampled. So a method that the sampled data are interpolated by parabola is adopted to correct the sampled maximum value points. The correction formula is Equation (1) (the corrected value is place in the sequence T_0).

(d) With the corrected values, the fundamental frequency is estimated with the least square method, the estimation formulas is (2), where L_0 is the length of T_0 、 f_s is sample frequency and the estimated frequency is $\hat{f}_0$.

$$T_0(i) = \frac{4 \times T_1(i) \times x(T_1(i)) - (2 \times T_1(i) - 1) \times x(T_1(i)+1) - (2 \times T_1(i)+1) \times x(T_1(i)-1)}{2 \times (2 \times x(T_1(i)) - x(T_1(i)+1) - x(T_1(i)-1))} \tag{1}$$

$$\hat{f} = \frac{f_s(L_0 \sum_{i=1}^{L_0} i^2 - (\sum_{i=1}^{L_0} i)^2)}{L_0 \sum_{i=1}^{L_0} i \cdot T_0(i) - \sum_{i=0}^{L_0} i \cdot \sum_{i=1}^{L_0} T_0(i)} \tag{2}$$

3. Processing system

3.1 Hardware of the processing system

Fig. 3 is the hardware block diagram of the digital signal processing system. It consists of a piezoelectric sensor, a charge amplifier, a programmable control amplifier, a anti-alias filter, an analog to digital converter, a LCD, a keyboard input part, a CPLD chip, a TMS320vc5409 DSP chip, an analog output circuit, a flash EEPROM chip and a system guard circuit. The piezoelectric sensor of a vortex flowmeter transforms the flow information into the electrical signal. The signal is amplified with the charge amplifier and then with the program-control amplifier. The amplified signal is filtered with the anti-alias filter to avoid the frequency alias. After the filtered analog signal is sampled and quantified into a digital signal, the digital signal, instead of being transmitted to the DSP directly, must be converted from the voltage level 5V to 3.3V with the CPLD integral chip, and then transmitted to the DSP, which is because the DSP chip selected is TMS3205409, a type of low cost chips, which I/O voltage level is 3.3V, but the A/D output voltage level is 5V. With the DSP computation ability at high speed, the mentioned above algorithm based on the wavelet transform can be used to compute the signal frequency. And then besides the computed signal frequency can be displayed with the LCD, the computed result can also be converted into an analog quantity with a D/A converter as input for other control system. In the system the keyboard input part can be used to adjust the display content and input instrument parameters of vortex flowmeters. The LCD can also display other information of the vortex flowmeter besides displaying the signal frequency, and the CPLD can also be used to provide controlling signals to other chips in the system. The FLASH is applied to store the system program that had been tested, and in charge of transmitting the program to the DSP as soon as

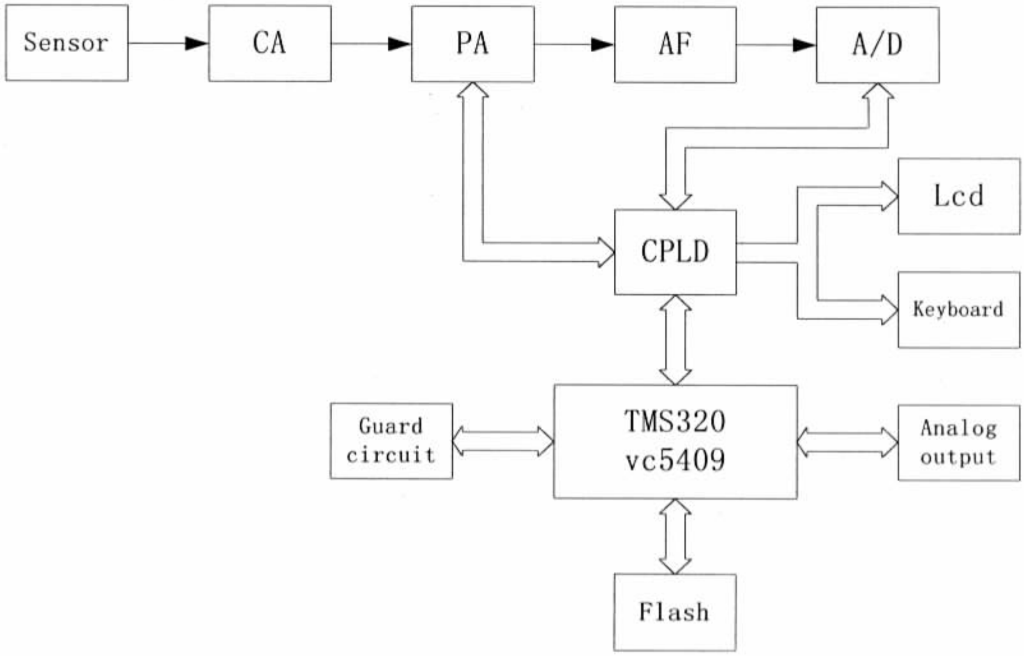

Fig. 3 Hardware block diagram of the signal processing system based on DSPs
CA---charge amplifier PA---programmable-control amplifier AF---anti-alias filter

the system is powered. At last a guard circuit is needed assuring the system running rightly and reset the system when system is in software error.

3.2 Software of the processing system

Fig.4 is the software block diagram of the system. The software of DSP based signal processing system is designed based on modules, it includes: a main monitoring module, a external interrupt service module, a timer interrupt service module, a initialization module, a computation sub-program, a LCD displaying module, a keyboard scanning and monitoring module, a watchdog module.

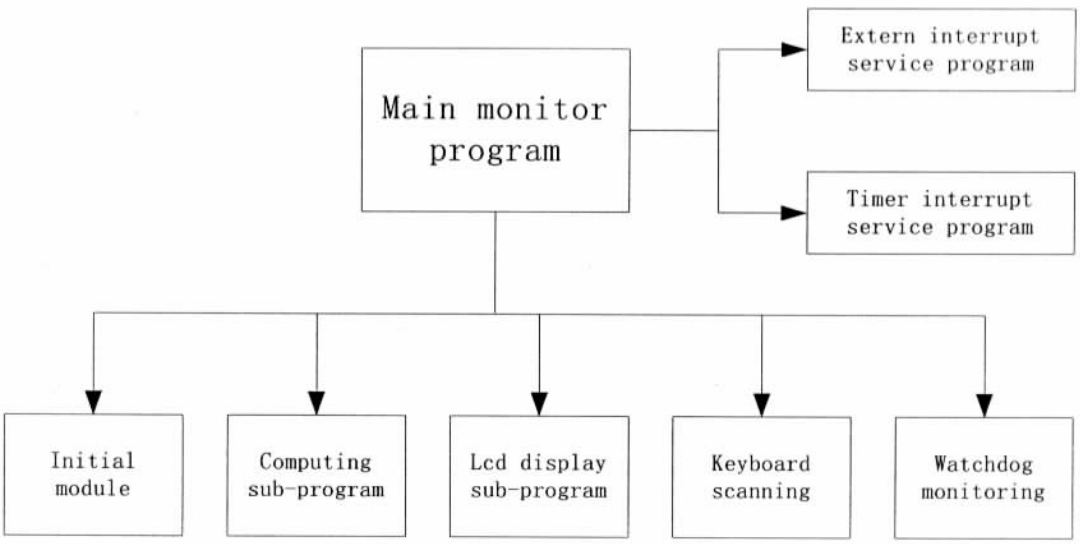

Fig. 4 System software block diagram

The main monitoring module takes charge of the detailed arrangement of the whole system running, its flowchart is shown in Fig. 5. There are two software timers to decide when the computation sub-program and the LCD display sub-program begin to run respectively. Before all programs begin to run, the initialization module must be executed. The initialization module includes initializing the registers of DSP, the LCD display part, the parameters of the vortex

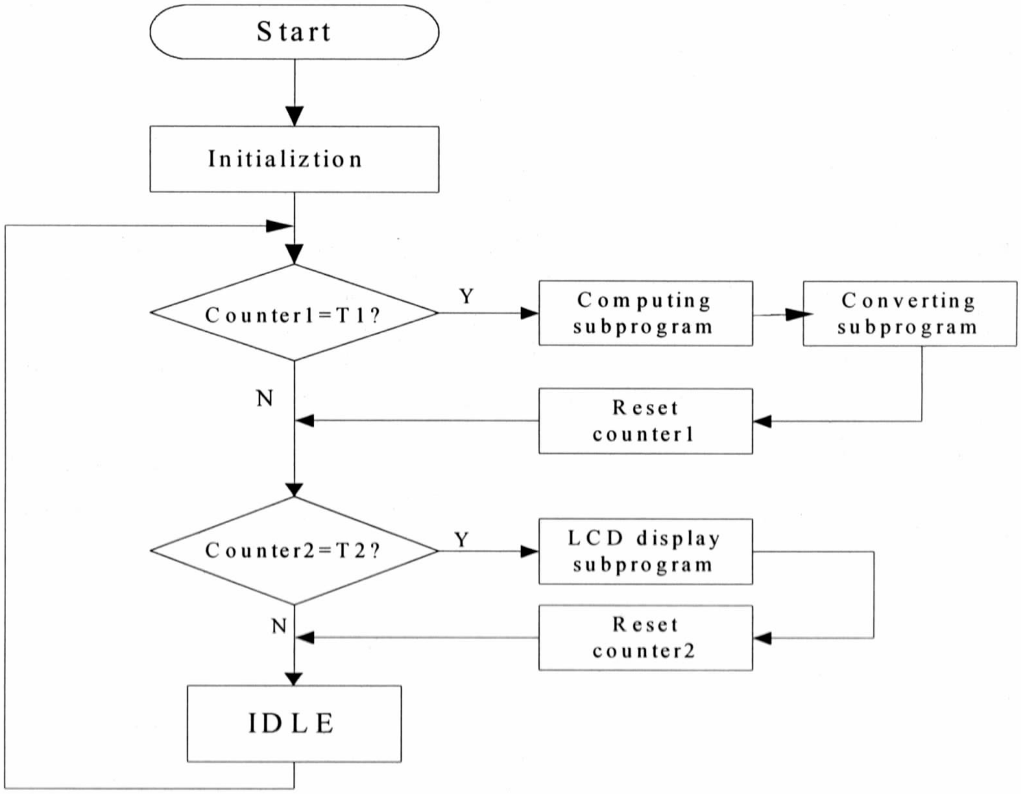

Fig.5 System monitor program flowchart

flowmeter, some quantities of the computation module, and system sampling frequency. The computation process is mentioned above, its flowchart is showed in Fig. 1.In the system, there are two interrupts. One is the external hardware interrupt, and the other is the timer interrupt. The timer interrupt is in charge of deciding system sample frequency with interrupt interval and sends out control signals starting up the A/D sampling through the timer interrupt service sub-program. And the external hardware interrupt answers the inquiry from A/D and transmit the converter result to the DSP through the external interrupt service sub-program.

4. Results of test

A HP3325B Synthesizer/Function Generator produces sine wave and modulated signal as shown in equation (3), which simulates the output signal of piezoelectric sensor in the vortex flowmeter and harmonic disturbances.

$$y(t) = V_{cm} \cdot \sin\omega_c t + \frac{1}{4} \cdot V_{cm} \cdot \cos(\omega_c t - \Omega t) - \frac{1}{4} \cdot V_{cm} \cdot \cos(\omega_c t + \Omega t) \tag{3}$$

In (3), ω_c is the frequency of main signal, Ω is the modulation frequency, and V_{cm} is the magnitude of main frequency. The real time processing system is tested and the results are shown in Table I. The first row of Table I is the main frequency, which values selected for test are the maximum error points of the non-integral period sampling in order to exam the algorithms and system strictly. The modulation frequencies are shown the first column of Table I. It is thus clear that our algorithms are of high measurement accuracy under the condition of disturbance, and the measurement errors of the signal processing system are not greater than 0.15%.

Table I Results of test

Main Modulation Frequencies Hz	0Hz	5Hz	10Hz
51.2695	51.2727	51.2828	51.3505
Errors(‰)	0.06	0.26	1.5
75.6836	75.6868	75.6671	75.6017
Errors (‰)	0.04	0.22	1.08
100.0976	100.0962	100.1565	100.0287
Errors (‰)	0.01	0.59	0.69
197.7539	197.7473	197.8492	197.7071
Errors (‰)	0.03	0.48	0.24
295.4102	295.4130	295.3744	295.4210
Errors (‰)	0.01	0.12	0.04
393.0664	393.0653	393.0793	393.0873
Errors (‰)	0.00	0.03	0.05
490.7227	490.7275	490.7399	490.7461
Errors (‰)	0.01	0.04	0.05
732.4218	732.4142	732.4376	732.4089
Errors (‰)	0.01	0.02	0.02
976.5625	976.5613	976.4995	976.5146
Errors (‰)	0.00	0.06	0.05
1464.8436	1464.8502	1464.8406	1464.9037
Errors (‰)	0.00	0.00	0.04

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