A New Class of Filters for Sensor Response Correction

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Abstract: This paper presents a method for dynamic compensation of a load cell response using time-varying continuous-time filter. The proposed approach is used for the improvement of the response of the model of the load cell mounted on an elastic foundation. The paper describes a theoretical implementation of the proposed time-varying filter, and suggests the implementation technique with the aid of which this kind of filter can be implemented in practice. Simulation results verifying the effectiveness of the proposed filter are presented and compared to the traditional time-invariant configuration.

Key–Words: Sensor response correction, analog signal processing, time-varying systems, transients

1 Introduction

Load cells are used in many industrial weighing applications. The settling time of these transducers is long, and lengthen additionally by reason of the settling time of the elastic construction of the scale. Therefore, the weight estimation produced by the load cell is disturbed by a transient process. Since signal processing and control systems cannot operate correctly if they receive inaccurate input data, compensation of the imperfections of sensor response is one of the most important problems in sensor research. Influence of unwanted signals, nonideal frequency response, parameter drift, nonlinearity, and cross sensitivity are the five major defects in primary sensors [1, 2].

Load cells have an oscillatory response which always needs time to settle down. Dynamic measurement refers to the ascertainment of the final value of a sensor signal while its output is still in oscillation. It is, therefore, necessary to determine the value of the measurand in the fastest time possible to speed up the process of measurement, which is of particular importance in some applications. One example of processing that can be done on the sensor output signal is filtering to achieve response correction [2].

In [2], the authors presented a very wide literature review concerning the problem of the load cell response correction. This problem has been solved using a few techniques. Software solutions for sensor compensation are reviewed in [3]. Analog adaptive techniques have been used in [2, 4] and digital adaptive algorithms have been proposed in [5]. In [6, 7], the authors demonstrated that an artificial neural network can also be useful for intelligent weighing systems. Other methods, such as employing a Kalman filter [8] and nonzero initial condition [9] have also been applied for dynamic weighing systems.

In this paper, we present a new competitive method for dynamic compensation of the load cell response using a time-varying continuous-time filter. The outline of the paper is as follows. In Section 2, some aspects of load cell modeling are presented. The analytical synthesis of the time-varying filter are discussed in Section 3. Section 4 then presents the results of simulations carried out with the aid of computer simulations. The conclusions are presented in Section 5. This paper presents the same concept as in [10] but for a more complex model of the load cell.

2 Modeling of a scale based on a load cell

The mathematical model of the scale is a base of a design and tests of the transient time compensation. Previous work [5, 10] consider the scale model assuming that the load cell is fixed in the inertial reference frame. That means the load cell is fixed directly to the rigid foundation or the mount has an infinite stiffness. In practice, the assumption is not always fulfilled. The model of the load cell transducer should take into account the dynamics of the support construction of the load cell.

In practice, a scale with a load cell often composes a mechanical system with more than one degree of freedom. One of the reasons of this property is the

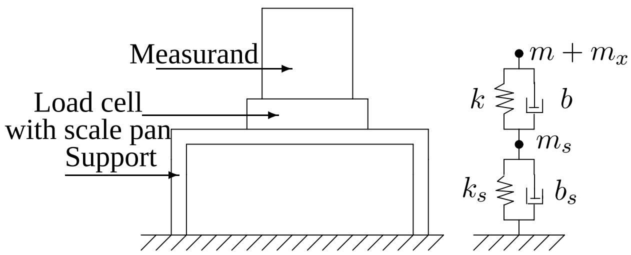

Figure 1: Scale mounted on elastic foundation.

elasticity of fixation of the scale. For example, the scale fixed to the elastic support (Fig. 1) or hung on a gate construction. These systems can be modeled as one-dimensional two-degree of freedom lumped mechanical system with the following parameters: mass of the measurand m_x, mass of the load cell m, mass of the support m_s, elasticity of the load cell k, elasticity of the support k_s, attenuation of the load cell b and attenuation of the support b_s. The load cell signal $\widetilde{m}_x(t)$ fulfills the following system of equations

$$\frac{m+m_x}{k}\ddot{\widetilde{m}}_x(t)+\frac{b}{k}\dot{\widetilde{m}}_x(t)+\widetilde{m}_x(t)=m_x-\frac{m+m_x}{g}\ddot{\delta}(t) \quad (1a)$$

$$m_s\ddot{\delta}(t)+b_s\dot{\delta}(t)+k_s\delta(t)=g\widetilde{m}_x(t)+\frac{bg}{k}\dot{\widetilde{m}}_x \quad (1b)$$

where $\delta(t)$ is related with deflection of the support. The initial conditions of the equation describing the scale (1a) are set on zeros, and the equation of the support (1b) has both the initial deflection and the initial deflection velocity equal zero.

The load cell signal $\widetilde{m}_x(t)$ has the following property

$$\lim_{t\to\infty} \widetilde{m}_x(t) = m_x \ , \quad (2)$$

i.e. in the steady-state the load cell signal is equal to the measurand. The deflection of the support in the steady state

$$\lim_{t\to\infty} \widetilde{\delta}(t) = \frac{g}{k_s} m_x \quad (3)$$

is caused by the ”transmitted” load mass.

The equations (1) show that the system consists of two coupled harmonic oscillators: the scale and the support with the natural frequency

$$\omega_{ns} = \sqrt{\frac{k_s}{m_s}} \ , \quad (4)$$

and the damping ratio

$$\zeta_s = \frac{b_s}{2\sqrt{k_s m_x}} \ . \quad (5)$$

The coupling between the oscillators elongates the settling time and causes some resonant effects as strong amplification of the signal $\widetilde{m}_x(t)$.

The load cell model presented in this paper exhibits an oscillatory character. Sensors in contemporary measurement systems should possess a very short settling time. Therefore, it is justified to search a technique for a sensor response improvement.

3 Time-varying filter formulation

The problem of transient improvement was considered in many fields of engineering. For traditional time-invariant filters there are only small possibilities of transient reduction, since the filter parameters are calculated on the basis of the assumed approximation method. This fact guarantees that the frequency requirements are satisfied without taking into consideration the characteristic of the transient state. Previous investigations [11, 12, 13] proved that one can obtain significant changes of the transient duration by the variation of the filter passband. This procedure is related to the change of the value of filter coefficients.

Dynamic properties of the second order lowpass filter (or filter of constant component) are described by the damping ratio ζ and the natural frequency ω_n. The transfer function of this filter can be written as follows:

$$H(s) = \frac{1}{\omega_n^{-2}s^2 + 2\zeta\omega_n^{-1}s + 1}. \quad (6)$$

It is well known that the higher value of the natural frequency ω_n, the shorter transient of the filter. On the other hand, the smaller value of the damping ratio ζ, the smaller rise time of the filter. By changing these parameters in time, we can improve the dynamics of the filter and obtain significant reduction of the transient duration. Hence, we have to do with the time-varying filter.

The analytical synthesis of the time-varying filter is the result of modeling the differential equation which describes the filter in the time domain. For the purpose of the filter response improvement it was assumed that dynamic parameters of the filter will be varied in time. Therefore, the model of the filter has the following form:

$$\frac{1}{\omega_n^2(t)} \cdot y''(t) + \frac{2\zeta(t)}{\omega_n(t)} \cdot y'(t) + y(t) = x(t) \quad (7)$$

where $x(t)$ and $y(t)$ are respectively the input and output of the filter, $\omega_n(t)$ is a function of the natural frequency, and $\zeta(t)$ is a function of the damping ratio.

As it was discussed above, the larger value of ω_n, the shorter transient. On the other hand, the larger

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value of ζ, the smaller overshoot of the filter. On the basis of these rules it is easy to guess that for the improvement of the dynamics of the filter all above mentioned parameters should have larger values in the initial phase of filter run. Therefore, for the purpose of the transient reduction, the functions of the filter parameters have been formulated in the following forms:

$$\omega_n(t) = d_\omega \cdot \overline{\omega_n} \cdot \left[1 - \frac{d_\omega - 1}{d_\omega} \cdot h_s(t)\right], \quad (8)$$

$$\zeta(t) = d_\zeta \cdot \overline{\zeta} \cdot \left[1 - \frac{d_\zeta - 1}{d_\zeta} \cdot h_s(t)\right] \quad (9)$$

where $\overline{\omega}_n$ and $\overline{\zeta}$ are the natural frequency and the damping ratio, which come from the assumed approximation method. The coefficients d_ω and d_ζ are the variation ranges of the functions $\omega_n(t)$ and $\zeta(t)$. These parameters are described by the following ratios:

$$d_\omega = \frac{\omega_n(0)}{\overline{\omega_n}},\ d_\zeta = \frac{\zeta(0)}{\overline{\zeta}}. \quad (10)$$

Functions (8) and (9) can be easily generated in the analog technique, and this time dependency works well [11, 12]. The function $h_s(t)$ in (8) and (9) describes the step response of the second order supportive system $H_s(s)$ which has the following form:

$$H_s(s) = \frac{1}{r^{-2}s^2 + 2\delta r^{-1}s + 1}. \quad (11)$$

Therefore, the step response $h_s(t)$ of $H_s(s)$ can be written as follows:

$$h(t) = \mathcal{L}^{-1}\left[\frac{1}{s} \cdot \frac{1}{r^{-2}s^2 + 2\delta r^{-1}s + 1}\right] \quad (12)$$

where $\mathcal{L}^{-1}$ is the inverse Laplace transform, and r and δ are respectively the natural frequency and damping ratio of the second order supportive system.

The functions $\omega_n(t)$ and $\zeta(t)$ should not possess oscillations in their run, so $\delta = 0.9$ was established. With reference to the functions (8) and (9), the coefficient δ can be named as the oscillation factor, and r as the variation rate of the functions $\omega_n(t)$ and $\zeta(t)$.

The main assumption which must be imposed on the functions $\omega_n(t)$ and $\zeta(t)$ is the necessity of their settling during the transient of the original time-invariant filter. This condition can be written as

$$\forall_{t>t_{s\alpha}}\ \omega_n(t) = \overline{\omega_n} \pm \alpha,\ \forall_{t>t_{s\alpha}}\ \zeta(t) = \overline{\zeta} \pm \alpha \quad (13)$$

where $t_{s\alpha}$ is the settling time (with assumed accuracy of α) of the original time-invariant filter. Therefore, the time-varying filter which is to be designed should possess, after $t_{s\alpha}$, the same spectral properties as the traditional time-invariant filter.

Fig. 2 presents a detailed model of the second order time-varying filter which has been discussed in this paper. A classical implementation of the time–varying approach described in this paper requires the use of multipliers, adders, and two additional integrators. As one can notice, the complexity of the overall system underwent a significant increase. However, in situations, in which the transient should be as short as possible this complexity increase may be profitable. The implementation problems of the described idea will not be analyzed in this paper. Nevertheless, at the end of this paper, the implementation technique will be suggested.

4 Results of simulations

The Bessel filters enjoy the best properties among all analog continuous-time filters when the passing through of rectangular impulses is considered. Therefore, for the purpose of the dynamic correction of the response of the load cell model, the lowpass Bessel filter will be used. The simulations which will be presented are due to show that the time-varying filter will be always faster that the original time-invariant one. Therefore, the cutoff frequency of the filter was arbitrarily chosen at 1 rad/s. Of course, the cutoff frequency of the time-invariant filter can be selected more precisely for a given model.

The transfer function of the second order time-invariant Bessel filter with cutoff frequency $\omega_c = 1$ rad/s has the following form:

$$H(s) = \frac{1}{0.6180s^2 + 1.3616s + 1}. \quad (14)$$

After simple computations we can determine the natural frequency $\omega_n = 1.2720$ and the damping ratio $\zeta = 0.8660$ of this filter.

The optimal values of the functions parameters which vary the filter coefficients are selected on the basis of computer simulations because analytical solutions of differential equations with varying coefficients are impossible to obtain in our case.

For the load cell model, which has been described in the previous section the following variation ranges of the functions which vary the filter parameters were chosen:

$$d_\omega = \frac{\omega_n(0)}{\overline{\omega_n}} = 50,\ d_\zeta = \frac{\zeta(0)}{\overline{\zeta}} = 10 \quad (15)$$

which means that in the initial phase of the filter work the natural frequency is 50–times greater and the

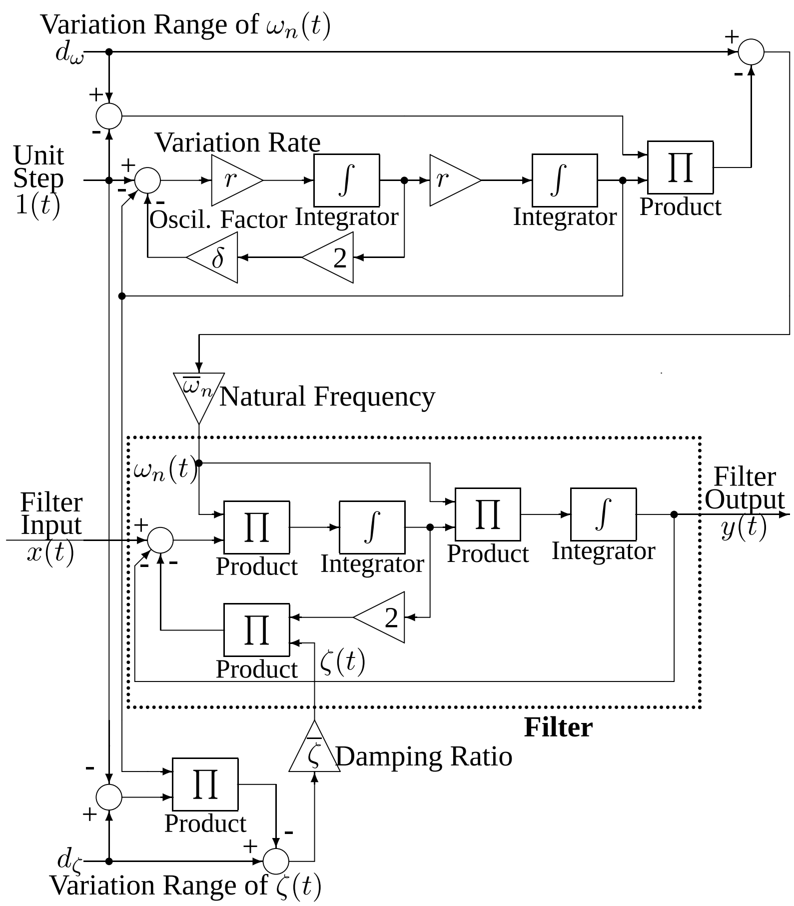

Figure 2: Detailed model of the second order time-varying filter.

damping factor 10–times greater than the ones following from the approximation, i.e. when the parameters are settled.

Figs. 3 and 4 show the load cell outputs and the compensation filter outputs for the normalized mass $m_x = 10$ and $m_x = 15$. From these figures we can notice that the time-varying filter is considerably faster than the traditional time-invariant one. Adaptive techniques [2, 4, 5] are usually useless in the case of a complex model.

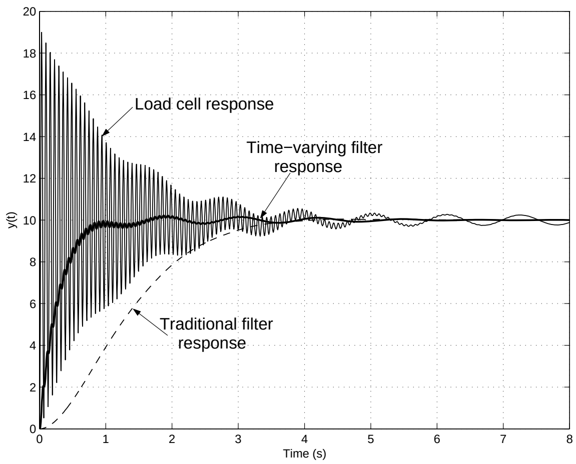

Figure 3: Simulation result of load cell compensation using time-varying filter for $m_x = 10$.

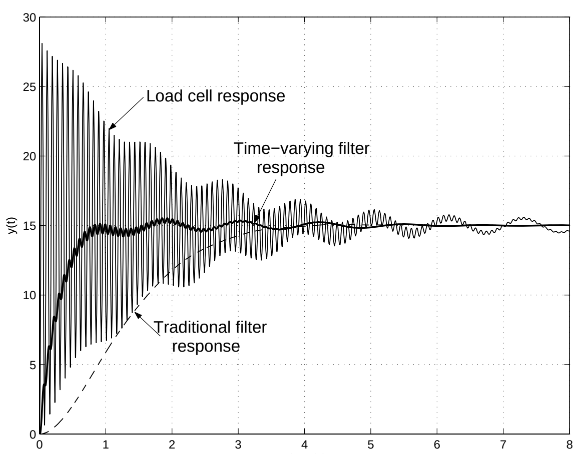

Figure 4: Simulation result of load cell compensation using time-varying filter for $m_x = 15$.

5 Conclusions

As it has been proven, the introduction of time-varying coefficients to the filter yields good results. The proposed filter is able to improve the load cell response irrespective of the complication degree of the model. It seems that further examinations of time-varying filters with application to the dynamic correction of sensor response are needed.

In the future, the proposed filter configuration will be implemented with the aid of the dynamic translinear technique [14]. By using the dynamic translinear principle, it is possible to implement linear and nonlinear differential equations, using transistors and capacitors only. Dynamic translinear circuits are excellently tunable across a wide range of several parameters, such as cutoff frequency, quality factor and gain, which increases their designability and makes them attractive to be used as standard cells or programmable building blocks. In fact, the dynamic translinear principle facilitates a direct mapping of any function, described by differential equations, onto silicon.

At the end of this paper, it is worth to add that the proposed filter structures can be easily transformed to digital filters. For that purpose, the continuous-time integrators from Fig. 2 should be transformed to their digital equivalents with the aid of the well known bilinear transform.

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