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# Resonant frequency estimation for adaptive notch filters in industrial servo systems☆

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ABSTRACT

This paper proposes a new algorithm for estimating the resonant frequency of adaptive notch filters used in servo systems. Notch filters and adaptive notch filters are widely used in commercial servo systems for suppressing a resonance which is a major obstacle in improving their performance. However, the conventional frequency estimation algorithm gives a dynamic behavior that is proportional to the difference between the square of the estimated frequency and the square of the actual frequency. This can cause the estimation dynamics to be too slow for low-frequency resonances, if both low- and high-frequency resonances are present and if the estimator gain is designed for a high frequency. This paper develops a new algorithm to give a dynamic behavior that is proportional to the difference in the estimated frequency and actual frequency. This allows selecting the estimation parameters independent from the value of resonant frequencies. The developed algorithm is implemented to a production servo controller and applied to a production printed-circuit-board inspection system. The experimental results show that the developed algorithm is much more effective in suppressing the resonances of both low- and high-frequencies, compared to the pervious algorithm. Furthermore, the ability to suppress vibration allows increasing the feedback gain, which in turn allows improving the tack time performance from 190 ms to 100 ms. All experiments reported in this paper were performed in an actual industrial environment using a production system, and the developed algorithms are applied to the production system.



## 1. Introduction

Servo systems are widely used in industrial automations and robotics. The servo systems generally consist of various mechanical components such as motors, movers, gears, belts, and couplings, and are generally applied to mechanical structures which may have flexible modes. If the flexible modes are excited with high controller gains, several resonances can occur, which can cause damage to machines and induce large positioning errors. However, the increasing demand for faster operations in recent years requires increasing controller gains. In order to prevent exciting resonances, detecting and controlling the mechanical resonance, are extremely important.

Several techniques have been developed to overcome the mechanical resonance problem. Using load velocity measurement or acceleration feedback, torsional vibrations can be reduced [1,2]. However, these methods require additional sensors. The notch filter is the most common technique for preventing the resonance in industrial servo systems without using an addition sensor. If the exact characteristics of a servo system are known and the resonant frequency can be modeled, a notch filter can be applied at the resonant frequency. However, obtaining the exact frequency characteristics of the system is very difficult. Even if a very precise initial tuning process is performed, the system configuration may change during operations or as time goes on. Unfortunately, notch filters are very sensitive to the notch frequency. If the notch frequency is different than the actual resonant frequency, the closed loop system could exhibit increased oscillations or even become unstable. Therefore, regular tuning procedures are required with notch filters.

An adaptive notch filter (ANF) has an advantage over a notch filter because an ANF can automatically adjust the notch frequency to the resonant frequency. An ANF can be implemented by adding a frequency estimator to a conventional notch filter. Various ANF-based resonance suppression methods have been proposed and applied to industrial servo systems [3–5].

There are other frequency estimation methods that can be applied to ANF. Fast Fourier transform (FFT) [6–11] methods have been used to calculate resonant frequencies, which are then used

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to tune filters for suppressing resonance vibrations. However, the computational requirements of these methods are generally excessive for industrial servo systems. In addition, the FFT-based methods generally require many data samples for an accurate estimation. Disturbance observers [12–15] or internal model control methods [16,17] have also been used to suppress resonant vibrations, but these techniques require *a prior* knowledge of system parameters, such as a torque constant for implementations. Other advanced frequency estimation algorithms have also been developed, including extended Kalman filter (EKF)-based methods [18–20]. The EKF-based frequency estimation methods can estimate not only the frequency but also the amplitude or damping coefficients of input signals. However, the EKF-based methods require appropriate models of input signals, and generally, computational requirements are excessive for real-time implementations on industrial servo systems. The dissipative control and filtering approach [21] is also an excellent approach; however, implementing it to the current production servo system is difficult.

The ANF-based estimation methods have relatively simple structures for real-time implementation. Regalia proposed one such algorithm for estimating an unknown frequency, and it is very effective in estimating the frequency of periodic signals [22]. Many variations of Regalia's algorithm have been developed [23–25]. Among the variations, Mojiri and Bakhshai's algorithm can be applied to a periodic signal with distortions, such as noise or harmonics [25]. However, these estimators cannot properly estimate low-frequency resonances because their adaptation speed becomes extremely slow when the estimated frequencies are low. To improve the estimation speed at low frequencies, Hsu, Ortega, and Damm proposed a scaled adaptation gain [24]. However, the scaled estimation gain contains additional parameters that require laborious tuning processes. Furthermore, applying the scaled adaptation gain is difficult to apply to servo systems when multiple resonances exist, because scaled gain parameters need to be changed with each resonance.

This paper presents a new algorithm for estimating resonant frequency using an error-proportional adaptation approach. Therefore, its estimation speed for low-frequency resonances can be made fast without sacrificing the estimation speed for high frequencies. The developed algorithm is implemented on a production industrial printed-circuit-board (PCB) inspection machine. This machine is a real, in-use production system that has resonant points at high frequencies as well as low frequencies. Therefore, the conventional ANF cannot be readily applied to this machine. In this paper, two types of ANF approaches were implemented and compared: (1) an ANF using the conventional frequency estimation algorithm and (2) an ANF using the developed frequency estimation algorithm. It is experimentally shown that the resonances are suppressed by using the developed algorithm. The resonance suppression allows using a higher control gain, which in turn results in a faster tack time performance. The achieved experimental results of tack time performance by using the developed algorithm is 100 ms, which is faster than the previous best result of 190 ms. that is obtained by using the conventional algorithm.

## 2. Principles of the adaptive notch filter

### 2.1. Notch filter

A notch filter is a linear time-invariant system with a magnitude response that vanishes at a particular point in the imaginary axis, which is called the notch frequency, However, the magnitude response of this system is nearly constant at other points. Excellent approximations are obtained using a second-order form with a bandwidth parameter that determines the depth of the filter. A typical transfer function of an NF is:

$$H_N(s) = \frac{s^2 + \left(1 - k_{depth}\right)\frac{\omega_{notch}}{Q_{notch}}s + \omega_{notch}^2}{s^2 + \frac{\omega_{notch}}{Q_{notch}}s + \omega_{notch}^2}, \tag{1}$$

where $\omega_{notch}$ is the notch frequency, $k_{depth}$ is the notch depth, and $Q_{notch}$ is the Q factor. The NF can be applied to a servo system to prevent the resonance problems. However, the parameters of the notch filter should be properly determined, with the most important parameter being the notch frequency.

### 2.2. Regalia's adaptive notch filter

The equations of Regalia's ANF were later transposed to continuous-time by Hsu et al. [24], and were modified by Mojiri et al. [25] using an adaptive filter as follow form:

$$\ddot{x} + 2\zeta\hat{\omega}\dot{x} + \hat{\omega}^2 x = 2\zeta\hat{\omega}^2 u, \tag{2}$$

where $x$ is the filter state, $u$ is an input, $\hat{\omega}$ is the estimated frequency, and $\zeta$ is a damping ratio. The Regalia's adaptive frequency estimation equation is:

$$\begin{aligned}\dot{\hat{\omega}} &= -\gamma x(\hat{\omega}^2 u - \hat{\omega}\dot{x}) \\ &= -\gamma x n\end{aligned} \tag{3}$$

where $\gamma$ is an adaptation gain and $n$ represents the parentheses part. Using Eq. (2) and Eq. (3), the transfer function of $u$ to $n$ can be described as a notch filter form. If $\hat{\omega}$ is close to the actual frequency of the input, Eq. (3) becomes minimized and the estimated frequency values will converge.

However, the conventional estimation algorithm, Eq. (3), has some problems if the frequency to be estimated is very low or if the magnitude of the resonance is large. First, the estimation speed depends on the value of the estimated frequency. Rewriting Eq. (3) as a multiplication form of the estimated frequency results in:

$$\dot{\hat{\omega}} = -\hat{\omega} \cdot \gamma x(\hat{\omega}u - \dot{x}). \tag{4}$$

Therefore, if the value of the estimated frequency is very low, then the adaptation speed will be very slow. Furthermore, the adaptation laws of conventional frequency estimation algorithms are proportional to the difference between the square of estimated frequencies and the square of the actual input frequencies.

The behavior of the frequency estimator is analyzed in the steady state. When the input signal is sinusoidal, then the filter states can be described as follows:

$$\begin{aligned}x &= A\sin\omega_o t, \\ \dot{x} &= A\omega_0\cos\omega_o t, \\ \ddot{x} &= -A{\omega_0}^2\sin\omega_o t.\end{aligned} \tag{5}$$

The frequency estimation algorithm, Eq. (3), can be rewritten using Eqs. (2) and (5):

$$\begin{aligned}\dot{\hat{\omega}} &= -\gamma x(\ddot{x} + \hat{\omega}^2 x) \\ &= -\gamma x(-A{\omega_0}^2\sin\omega_o t + \hat{\omega}^2 A\sin\omega_o t) \\ &= -\gamma x A\sin\omega_o t\left(\hat{\omega}^2 - {\omega_0}^2\right) \\ &= -\gamma A^2\sin^2\omega_0 t\left(\hat{\omega}^2 - {\omega_0}^2\right).\end{aligned} \tag{6}$$

Therefore, the relationship between the estimated frequency and the estimation equation is:

$$\dot{\hat{\omega}} \propto \left(\hat{\omega}^2 - {\omega_0}^2\right). \tag{7}$$

This can be problematic when several resonances exist at both low and high frequencies. In addition, the average behavior determined by the estimation algorithm is proportional to the square of the input signal amplitude:

$$\dot{\hat{\omega}} \propto A^2, \tag{8}$$