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Adaptive cancellations of systematic disturbances in a synchronous amplitude demodulator for sensors and instrumentation applications

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Abstract

An adaptive technique is presented that performs synchronous amplitude demodulation while adaptively canceling systematic disturbances—frequently encountered in sensors and instrumentation applications. The method is effective in jointly estimating and tracking the signal amplitude along with additive DC bias, interfering sinusoids, or exponentially decaying transients. Lyapunov method is used to study the stability and convergence of the proposed technique. In addition, the extension of the algorithm for multiple sinusoid case is also presented. Computer simulations and the experimental results are included to demonstrate the effectiveness and applicability of the proposed scheme in applications like resolver, synchros, and power system relays. It can easily be implemented in cost effective manner requiring minimal computation effort and hardware.

Keywords: Amplitude demodulation; Amplitude estimation and tracking; Disturbance cancellation; Adaptive algorithms

1. Introduction

Amplitude demodulation finds many practical applications in fields like communication systems [1], sensor instrumentation [2–4], signal processing, and system identification. Different approaches [1] are found in literature for demodulation of amplitude modulated (AM) ordinary, single side band (SSB), and double side band (DSB) signals. The filter method [1] is commonly used when the delay introduced by the filtering is tolerable. On the contrary, when such delay becomes detrimental, synchronous techniques are employed [2,3,8,9]. The technique in [2] is faster in settling time when compared to the filter method, which, however, is at the cost of precise clock timing relations. It is also more susceptible to noise and systematic disturbances at the input signal resulting in substantial distortion of the demodulated signal. The closed loop techniques in [3] is, however, slower than the techniques presented in [2], but it has a better noise suppression capability. In [4] a vision-based navigational sensor (VISNAV) is presented that demodulates the multiple position dependent signals from the position sensitive diodes (PSD) to obtain their magnitudes. The calculated magnitudes are further used in the attitude estimation algorithm for navigation solution. However, it is observed that the complete system was constrained by the computation complexity of amplitude demodulation, which could only be tackled using multirate processing techniques in a high performance DSP. Another interesting example is presented in [5] where a phase-locked loop (PLL) based real-time approach is proposed to measure particle velocity using laser Doppler velocimeter data. The electrical signal from the photo-detector has time varying amplitude modulated over a carrier and DC bias [4]. The analog PLL will cause degraded tracking and estimation performance of the particle velocity, as the DC bias and the time varying amplitude severely affects the stability and the range of operation of the analog PLL [12]. However, with adaptive cancellation

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of such disturbances and using the proposed technique as an automatic gain tuning (AGC) circuit, performance of the laser Doppler velocimetry can be improved. Besides, in application like relays that protects the power systems, the input modulated signal can also have a DC bias [6], or an exponentially decaying transient [7], which severely effects the estimation of the relay parameters for protecting the transmission lines. To solve this, computationally intensive DFT-based techniques are used to remove the additive DC bias and the exponential transient from the modulated input sinusoid signal. Moreover, these methods that are generally offline, do not work when even order harmonics are present at the input signal [6]. Further, it has been observed in many situations, such disturbances are time varying, and the techniques having the capability to adapt to such changes are more effective.

In this paper, a simple computationally efficient, online technique is presented that performs synchronous amplitude demodulation, with the capability to adaptively cancel systematic disturbances like DC bias, decaying transient, and low frequency interfering sinusoids. The method is effective in estimating and tracking the magnitude of an additive DC bias, interfering sinusoid, and exponentially decaying transient signal. The stability and convergence of the proposed algorithm is studied using Lyapunov methods and it is showed that the proposed demodulator is globally exponentially convergent. Further, an extension of the algorithm is also provided for multiple sinusoids. Computer simulation and experimental results are included to illustrate the effectiveness and applicability of the method in many practical scenarios.

The paper is organized in six sections. Section 2 describes the structure of the algorithm and the basic assumptions. The stability and convergence study is included in Section 3; whereas extension of the algorithm for multiple sinusoidal cases is included in Section 4. Section 5 includes the simulation result and different case studies. Finally, the conclusion is drawn in Section 6.

2. Adaptive synchronous amplitude demodulator

The proposed algorithm is similar to least mean square (LMS) adaptive algorithm which takes the advantage of gradient techniques. Adaptive techniques have been classically used in applications where stationary and time invariant systems are assumed and the convergence properties have been verified for a wide range of applications. However, they have also been used for non-stationary and time varying systems with appreciable performance. The proposed demodulator has these advantages making it widely applicable.

The block diagram representation of the proposed demodulator is shown in Fig. 1. Let us consider a sinusoid signal

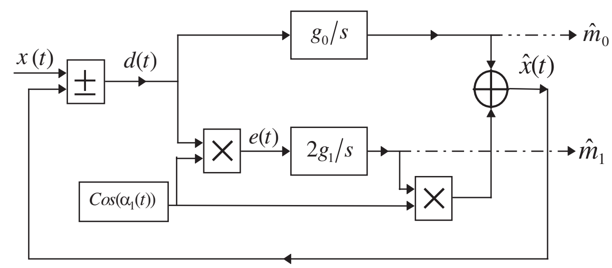

Fig. 1. Block diagram representation of the algorithm.

with a DC bias component given as:

$$x(t) = m_0 + m_1 \cos(\alpha_1(t)), \tag{1}$$

where m_0 is the DC bias/disturbance, m_1 and $\alpha_1(t)$ is the magnitude and the phase of the sinusoid respectively. It is assumed that the phase (or the frequency) $\alpha_1(t) = \omega_1 t$ is known and the disturbances and the magnitude of the sinusoid are constants. The problem considered here is to find the estimates of the amplitude and the disturbance in the sinusoid, which is represented by $\hat{m}_0$ and $\hat{m}_1$, respectively.

The block diagram of the algorithm is shown in Fig. 1 where an estimate of the incoming signal $\hat{x}(t)$ given by

$$\hat{x}(t) = \hat{m}_0(t) + \hat{m}_1(t)\cos(\alpha_1(t)) \tag{2}$$

is produced and subtracted to obtain a difference signal $d(t) = x(t) - \hat{x}(t)$. The difference signal is multiplied with a sinusoid of the same phase and integrated to obtain magnitude of the sinusoid. Similarly, the integration of the difference signal gives the estimate of the DC bias. Thus the equations describing the algorithm are as follows:

$$\frac{\mathrm{d}}{\mathrm{d}t}\hat{m}_0(t) = g_0 d(t), \tag{3}$$

$$\frac{\mathrm{d}}{\mathrm{d}t}\hat{m}_1(t) = 2g_1 d(t)\cos(\alpha_1(t)). \tag{4}$$

3. Analysis

The analysis of the proposed demodulator can be carried out using similar approach as that of the LMS algorithm. It can be shown to have unbiased global convergence by using Lyapunov's method [11]. For this we consider the error dynamics representation of the algorithm as described in (8). When difference signal $d(t)$ is expanded, we get

$$d(t) = (m_0 - \hat{m}_0(t)) + (m_1 - \hat{m}_1(t))\cos(\alpha_1(t)), \tag{5}$$

$$\frac{\mathrm{d}}{\mathrm{d}t}\hat{m}_0(t) = g_0(m_0 - \hat{m}_0(t)) + g_0(m_1 - \hat{m}_1(t))\cos(\alpha_1(t)), \tag{6}$$

$$\frac{\mathrm{d}}{\mathrm{d}t}\hat{m}_1(t) = 2g_1(m_0 - \hat{m}_0(t))\cos(\alpha_1(t)) + 2g_1(m_1 - \hat{m}_1(t))\cos^2(\alpha_1(t)). \tag{7}$$