

Implementation of the digital phase-sensitive system for low signal measurement

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Abstract. In this paper we present the implementation of a digital lock-in amplifier (LIA) completely based on Labview with a general-purpose data acquisition board (or a high quality sound card) and a high-gain low-noise amplifier. The signal analysis is processed by the software. We describe some characteristics of the LIA including output voltage vs. frequency and output phase vs. noise. The LIA can be used to measure the small signals, even in presence of broadband noise which is several times greater than the signal itself. Since the signal processing takes place on the computer, the ones can display the waveform – as a time series or power spectrum – as it progresses through the instrument, which makes it an excellent tool for the senior-level physics lab.

Keywords: low signal, implementation, digital.

1. Introduction

Low-level signal processing is of practical importance in various aspects but it is usually coupled with difficulties. As the state of the instrument undergoes changes with temperature and time, the measurement results fluctuates. The low-level signal is characterized by the low signal-to-noise ratio (SNR). The common sources of noise include the 50/60 Hz noise from the power network, the $1/f$ noise from the pre-amplifiers, thermal noise and leakage current noise from sensors... or a combination of them. Those kinds of noises are always present and effect the measurement equipment [1].

The lock-in amplifier uses the phase-sensitive detection (PSD) which filters off all signal parts having different frequencies than the nominal frequency so does not effect the measurement. The PSD equipment not only can detect the amplitude of a signal having the same frequency as the reference signal but also is sensitive to the difference in their phases. Therefore, a system involving PSD can be used in detection of both amplitude and phase of a signal in presence of noise. Those systems based on PSD are called the lock-in systems. If the amplifier uses PSD than it will be called the lock-in amplifier [2].

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2. System implementation

The lock-in amplifier is called the digital lock-in amplifier if its PSD is performed digitally. Normally, the lock-in amplifiers are divided into two groups: the first using only a single PSD is called the single-phase lock-in and the second involving two PSD is called the two-phase lock-in [3].

In the single-phase lock-in we have to adjust the phases of reference signal and signal to achieve the maximal output signal. The obtained amplitude is what we need to measure. If the phases are shifted, then the output amplitude decreases. The phase adjustment block is needed here. In the two-phase lock-in the phase shifts do not play any role so we do not need to adjust this shift, furthermore, we can determine this shift [4].

The principle of work:

The principle of a lock-in lies is the phase-sensitive detection technique. The lock-in amplifier needs a reference signal. Normally, the experiment is provided with a fixed frequency (from the frequency generator) and the lock-in amplifier is concerning with the frequency. In Fig.1, the reference signal is a square-pulsed with frequency ω_r . This signal can be retrieved from the synchronous output of a functional generator. If the sinus signal was used than the output signal would be as in the Fig.1 This signal has the following form [5]:

$$V_{sig} \sin(\omega_r t + \theta_{sig})$$

where V_{sig} is an amplitude of signal. The reference signal is of form:

$$V_L \sin(\omega_L t + \theta_{ref})$$

The amplified signal is multiplied by a reference signal by using a PSD or an integration circuit. The output from PSD has a form:

$$\begin{aligned} V_{psd} &= V_{sig} \sin(\omega_r t + \theta_{sig}) * V_L \sin(\omega_L t + \theta_{ref}) = \\ &= 1/2 V_{sig} V_L \cos([\omega_r - \omega_L]t + \theta_{sig} - \theta_{ref}) - 1/2 V_{sig} V_L \cos([\omega_r + \omega_L]t + \theta_{sig} + \theta_{ref}) \end{aligned}$$

The output signal from the PSD is an AC signal having two frequencies: $(\omega_r - \omega_L)$ and $(\omega_r + \omega_L)$. If this signal is filtered by a low frequency filter than the AC component can be filtered. But if $\omega_r = \omega_L$ than the first component V_{psd} becomes DC. In this case, the output signal is:

$$V_{psd} = \frac{1}{2} V_{sig} V_L \cos(\theta_{sig} - \theta_{ref})$$

Thus the output voltage of a PSD is proportional to $V_{sig} \cos\theta$, where $\theta = \theta_{sig} - \theta_{ref}$. θ is a difference between the phases of the signal and a reference signal. By adjusting the θ_{ref} so that $\theta = 0$, we obtain $V_{sig} (\cos\theta = 1)$. Inversely, if $\theta = 90^\circ$ than the output voltage is 0. The single-phase lock-in provides the output voltage [6]:

$$V_{psd1} = \frac{1}{2} V_{sig} V_L \cos(\theta) \approx V_{sig} \cos(\theta)$$

So with the single-phase lock-in amplifier, the output voltage depends also on the phase shift between signal and the reference level. To omit this dependence the further PSD block must be involved. In the second PSD the signal is multiplied with the reference signal being shifted by $\pi/2$:

$$V_L = V_L \sin(\omega_L t - \theta_{ref} + \pi/2)$$

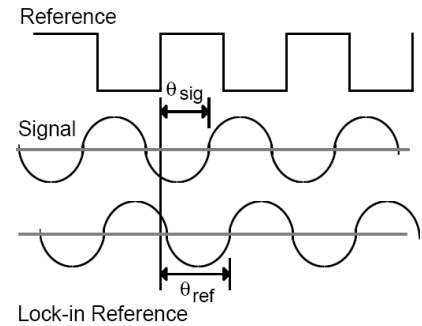


Fig. 1. Wave diagram.

Then the output voltage takes form:

$$\begin{aligned} V_{psd2} &= \frac{1}{2} V_{sig} V_L \cos(\theta_{sig} - \theta_{ref} - \pi/2) \\ &= \frac{1}{2} V_{sig} V_L \sin(\theta_{sig} - \theta_{ref}) \approx V_{sig} \sin \theta \end{aligned}$$

Now the system has two outputs: the first produces voltage proportional to $\cos\theta$ whereas the second to $\sin\theta$. Denote X as the first output and Y as the second output we have:

$$X = V_{sig} \cos \theta$$

$$Y = V_{sig} \sin \theta$$

These two quantities represent the signal and the reference. X is called the synchronous in-phase component and Y is the quadrature. When $\theta = 0$, X reaches maximum and Y = 0. Taking the square of both X and Y then sum up them, we have:

$$R = \sqrt{X^2 + Y^2} = V_{sig}$$

R is an amplitude of a signal. Thus the amplitude can be determined independently on the phase shift between signal and reference [7].

The phase shift is calculated according to:

$$\theta = \arctan\left(\frac{Y}{X}\right)$$

System design:

Based on the theoretical conception given above [8], we designed a digital two phase lock-in amplifier. All the calculation is performed on a computer using Labview [9]. The diagram of the system is feature in Fig. 2 and Fig. 3 gives the interface of the software.

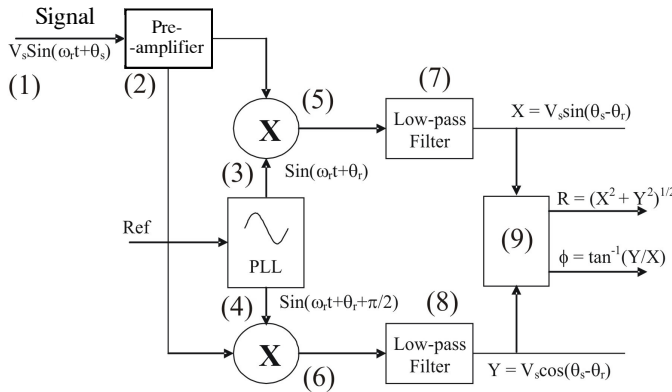


Fig. 2. The schematic diagram of a dual-phase lock-in amplifier.

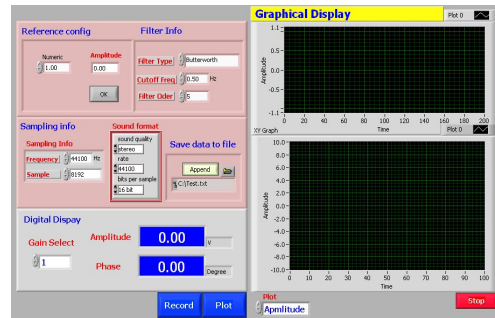


Fig. 3. Program interface.

The measuring signal (1) from the settled experiment is taking to the pre-amplifier (2). Here the amplifying coefficient must be chosen appropriately so that the output signal lies in the range of the providing ADC. The digital signal from ADC is divided into 2 multiplying circuits. At the circuit (5) the signal is multiplying with the reference signal of the form $\sin(\omega t + \theta_r)$, and at the circuit (6) the

signal is multiplying with the reference signal of the form $\sin(\omega_r t + \theta_r + \pi/2)$. The outputs from these circuits are put forward to the low pass filter. This filter eliminates the AC component and retain only the DC one. The filter (7) produces the synchronous output and the filter (8) produces the quadrature. These two parts are inputted in the calculation circuit (9), whose outputs are the amplitude and the phase shift. The amplitude may be the maximal amplitude V_0 or the effective voltage V_{rms} , the phase shift can be given in *radian* or *degree*.

3. System testing

The dependence of the dc-output voltage from a PSD on the phase shift between the signal and the reference ($\theta_s - \theta_r$)

The development of the output voltage $X = V_s \sin(\theta_s - \theta_r)$ according to the phase shift between the signal and the reference ($\theta_s - \theta_r$) in the first PSD is given in Fig.4. The same is for the output voltage from the second PSD (Fig.5).

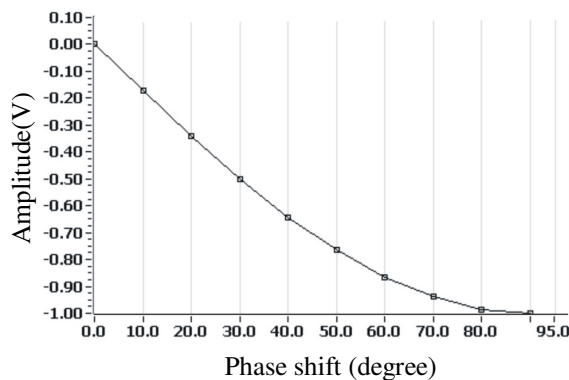


Fig. 4. Output voltage X vs. phase shift.

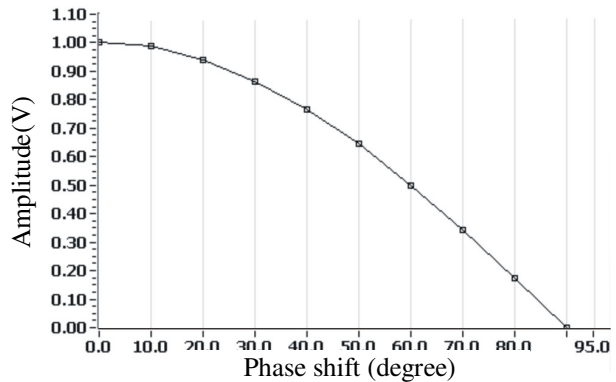


Fig. 5. Output voltage Y vs. phase shift.

The dependence of the dc-output voltage on frequency

Let have the reference at frequency 1kHz and the effective signal $V_0 = 1V$. The frequency of the signal is modulating from 980 Hz to 1020 Hz. The development of the dc-output voltage (R) of a PSD according to the frequency is given in Fig.6

This figure shows the maximal output voltage at the frequency exactly equal to the reference frequency (1000 Hz). There are also three different smaller amplitudes. This shows the improvement of the lock-in amplifier: it amplifies only the signal having the same frequency as the reference signal whereas the other signals are filtered off. In brief, the lock-in amplifier has an ability to suppress the noise belonging to different frequencies than the reference.

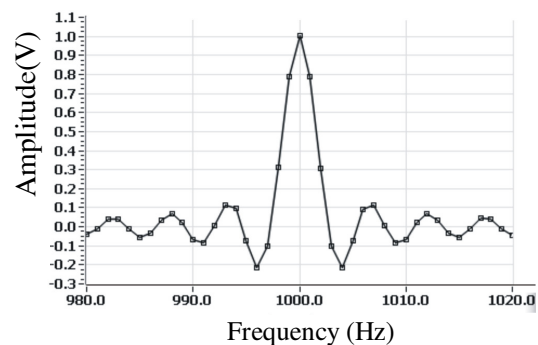


Fig. 6. Output voltage vs. frequency.

The development of the amplitude and the phase shift according to the noise level

Let the reference frequency equal 1 kHz, the maximal amplitude $V_r = 1$ V. The measured signal has the same frequency, and shifted in phase ϕ in comparison to the reference. Let the amplitude of the signal without noise is $V_s = 0.495$ V. After the incorporation of noise (white noise), the maximal amplitude of the noise is 2.5 V, we received the signal of the form showed in Fig.7. To continue this test, we mix the signal with the increasing noise and record the output voltage.

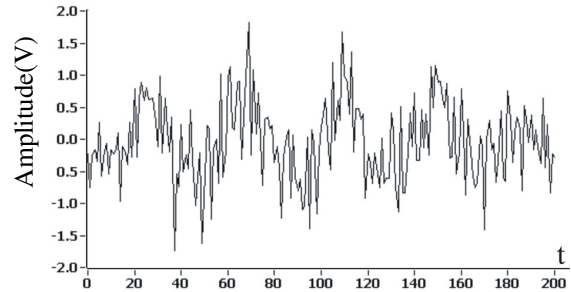


Fig. 7. Signal plus noise (maximal noise level 2.5V).

The result is showed in Fig.8. The Fig. 8 shows the constancy of the output dc-voltage ($V_s = 0.495$ V) when the amplitude of noise reaches 5V. Similarly, we record the dependence of the phase shift on the amplitude of noise in Fig.9. If we take the average of the measured data, we can obtain the more accurate result.

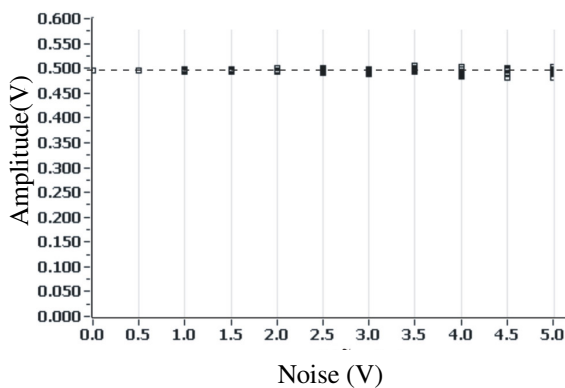


Fig. 8. The dependence of the output dc-voltage on the noise level.

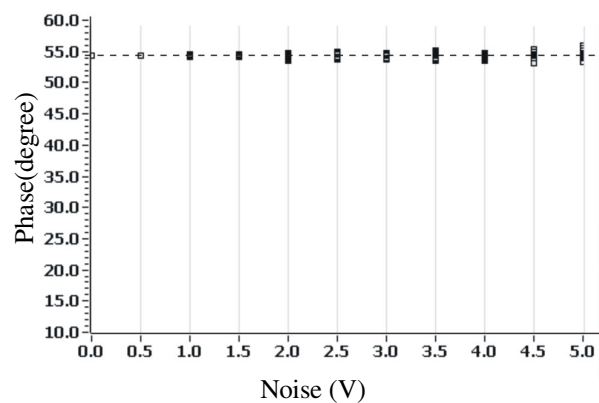


Fig. 9. The dependence of the phase shift on the noise level.

4. Conclusion

The implemented two-phase lock-in amplifier is low-cost because it uses only one PC with network card and a sound card with the pre-amplifier. It is easy to implement this device in the laboratories. Its functionality is adaptable for very low level signal. The design characteristics can be easily modified for various kinds of experiments. The data processing is automatic and convenient and can be saved directly to disks. The further important application is that it can be used as a teaching tool. Since the system uses Labview, it is easy to control and to evaluate the signals at every its segments so provides clarity for users to understand the working of the digital lock-in amplifier.

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