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ASSESSMENT OF SYNCHRONIC DETECTION AT LOW FREQUENCIES THROUGH DSP-BASED BOARD AND PC SOUND CARD

Mateusz Kotarski, *Janusz Smulko*

Faculty of Electronics, Telecommunications and Informatics, Gdańsk University of Technology, Gdańsk, Poland, jsmulko@eti.pg.gda.pl

Abstract − The lock-in amplifier, working at acoustic frequency range, can be cheaply realized by applying a sound card or a board kit with a digital signal processor, together with a low noise amplifier. Such realization can automatically retune frequency and can perform fast measurements at various frequencies to establish frequency characteristic of the investigated, and often time-varying, object. Quality of these two economical solutions is presented in detail and the possible improvements to limit measurement errors are discussed as well.

Keywords: lock-in amplifier, noise, accuracy

1. INTRODUCTION

Synchronous detection is a well-known technique and widely used for signal detection at presence of intensive noise [1–4]. There are various lock-in amplifiers currently available on a market that apply this technique and use digital solutions based on digital filters performed by digital signal processors (DSP) [5–7]. Unfortunately, the accessible devices are expensive and usually can not retune frequency automatically and sufficiently fast when we need to measure frequency characteristic of a physical object that changes its properties (e.g. impedance) during investigation. A good example of such object is a metal exposed to electrolyte which induces local corrosion events that change metal-electrolyte impedance even within a few seconds [8]. The same problem occurs in electrochromic devices, popularly known as *smart windows*, when we monitor impedance changes between electrolyte and electrochromic nanoparticle layer [9].

In this exploratory study, we applied the lock-in amplifiers that comprise a PC sound card or a DSP-based board. These solutions were presented in detail in the literature [3,4] but there was no clear answer which of them is a better one. Both solutions utilize A/D and D/A converters which can limit measurement accuracy due to their nonlinearities, jitter or quantization errors. Additionally, DSP can generate quantization errors during digital low-pass filtering. This effect depends on number representation that is available in the applied processor and can be probably neglected for the floating-point processors [10,11].

Thus, these issues should be investigated for the example converters to reconsider which of the mentioned reasons determine measurement accuracy. This problem is addressed in the paper and investigated in greater detail for the selected PC sound cards and the Texas Instruments DSP board, type DSK6713.

2. HOW A DIGITAL LOCK-IN AMPLIFIER WORKS

The principle of lock-in amplifier is well known and is explained in other works [1–4]. Here, we give only general introduction to the basic principles and explain the assigned labels.

A lock-in amplifier comprises a reference signal generator, an amplifier, a phase sensitive detector and a low-pass digital filter. The reference harmonic signal is generated digitally by the lock-in amplifier and is available at the output of D/A converter. This reference signal is passed through an investigated system that modifies its phase and amplitude. Moreover, this system, together with arranged in series a low-noise voltage amplifier, introduces noise. The applied amplifier is used to adjust system output signal to input voltage range of the applied A/D converter.

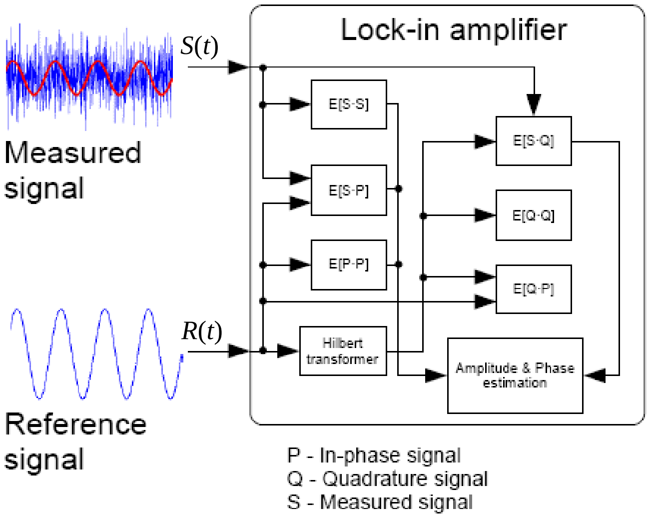

Fig. 1. Block diagram of the applied lock-in amplifier; operator E denotes averaging, $R(t)$ – reference signal, $S(t)$ – measured signal.

The block diagram of the realized lock-in amplifier presents fig. 1. We omitted a low noise amplifier in our experimental studies to concentrate on influence of imperfections introduced by A/D and D/A converters and digital filtering. Therefore, we injected noise into the studied simple low-pass RC system by applying white noise electronic generator, type NRG 201 from MessElektronik - Dresden, to simulate influence of the amplifier inherent noise and the inherent noise of the system under test (fig. 2).

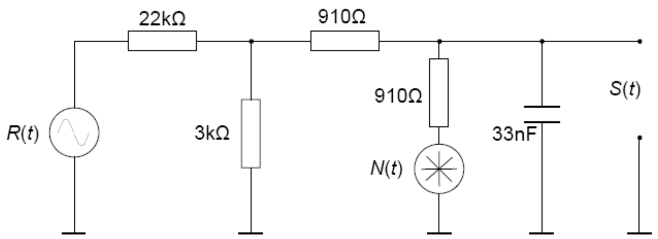

Fig. 2. Electronic scheme of the studied low-pass RC filter; $R(t)$ – reference signal, $N(t)$ – noise signal, $S(t)$ – measured signal.

The investigated system is stimulated by the reference harmonic signal $R(t)$. The system output signal $S(t)$ is sampled and then multiplied by the reference signal. This product is filtered by a low-pass filter to reduce noise which is not correlated with the harmonic component of the signal $S(t)$ induced by the input reference $R(t)$.

In this paper, we assumed the procedure presented in literature [4]. This algorithm assumes that we possess from the reference signal R its in-phase component P and quadrature component Q. The first constituent P is derived directly from R by normalizing its amplitude to unity. The latter Q results from shifting P by discarding first $K/4$ samples, where K means the number of samples per cycle of the reference signal R. In general, the number $K/4$ can differ from the integer value. Therefore, we applied the algorithm that is insensitive to a case when the phase shift between P and Q components differs from $\pi/2$ radians.

The observed signal S can be represented as a linear sum of two components P and Q, together with the additive noise component N:

$$S = aP + bQ + N \tag{1}$$

Then, the following products can be formed:

$$E[S \cdot P] = aE[P \cdot P] + bE[Q \cdot P] + E[N \cdot P] \tag{2}$$

$$E[S \cdot Q] = aE[P \cdot Q] + bE[Q \cdot Q] + E[N \cdot Q] \tag{3}$$

where operator E denotes averaging. The noise component N is not correlated with Q or P and therefore tends to zero as the averaging time grows. This fact is the main advantage of the lock-in amplifier technique which can reduce influence of intensive noise on measurement accuracy by operation of averaging or equally by low-pass filtering.

Equations (2) and (3) give the coefficients a and b which determine the module of the component of signal S induced by the reference signal R and not by noise N that is attenuated due to averaging:

$$|S| = \sqrt{a^2 E[P \cdot P] + b^2 E[Q \cdot Q] + 2abE[P \cdot Q]} \tag{4}$$

The phase shift ϕ introduced by the investigated system is resolved by equation:

$$\cos(\phi) = \frac{E[S \cdot P]}{|S|\sqrt{E[P \cdot P]}} \tag{5}$$

Equation (4), together with the coefficients a and b derived from (2) and (3) allows to estimate transmittance or impedance of the investigated system by retuning frequency of the reference signal $R(t)$ and repeated measurements.

3. EXPERIMENTAL RESULTS

The measurements were preceded by computer simulations of random error that can be expected at the measurement conditions:

- signal-noise ratio equal to -20 dB,
- sampling frequency f_s = 48 kHz,
- signal averaging time of 1 s that corresponds to averaging over 48000 consecutive samples.

We applied MATLAB® functions: *wgn* to generate white noise and *filter* to simulate low-pass RC filtering. Random error σ_s of the estimated signal |S| was defined as a standard deviation determined within a set of hundred estimators, obtained after 1 s long averaging for each. This limits accuracy of the defined random error to approx. 10% only [10]. The phase accuracy σ_ϕ established using the same computer-generated data was about 3^0. The operations realized by the lock-in amplifier (fig. 1) were also carried out by MATLAB® script.

The measurements of lock-in amplifiers accuracy were performed for:

- the sound card of the IBM ThinkPad T41 notebook and based on AD1981B codec,
- the sound card C-Media 9739A AC'97,
- the PCM3003 codec used in the daughter card that was connected to the DSP-based DSK6713 board.
- the AIC23 codec being part of the DSP-based DSK6713 board.

The lock-in amplifier input signal $S(t)$ is a sum of a white noise component filtered by a low-pass RC filter (fig. 2) and a harmonic signal $R(t)$. The noise component intensity at the input of A/D converter (fig. 1) was equal to 220 mV of rms value. The rms value of the harmonic signal at the same input and at frequencies where the RC filter characteristic is flat (e.g. 100 Hz – fig. 3) was set to 22 mV. These values established the signal-noise ratio as -20 dB.

The signal $S(t)$ was sampled at sampling frequency f_s=44.1 kHz (AD1981B, 9739A AC'97) or at f_s=48 kHz (PCM3003, AIC23) and averaged over 1 s intervals. The established averaging time limited the measurements of a dozen or so frequency points within acoustic bandwidth to the acceptable processing time below 20 s.

We applied the freeware software MPTM to generate harmonic signal at output of the PC sound cards. The signal was sampled and analyzed by the MATLAB® script when A/D converters of the PC cards were investigated. When the DSP-based board was tested we programmed the applied processor by using DDS technique to generate the sinus function at various frequencies. The measurements and necessary analysis was performed by C programme that worked in the Code Composer Studio software.

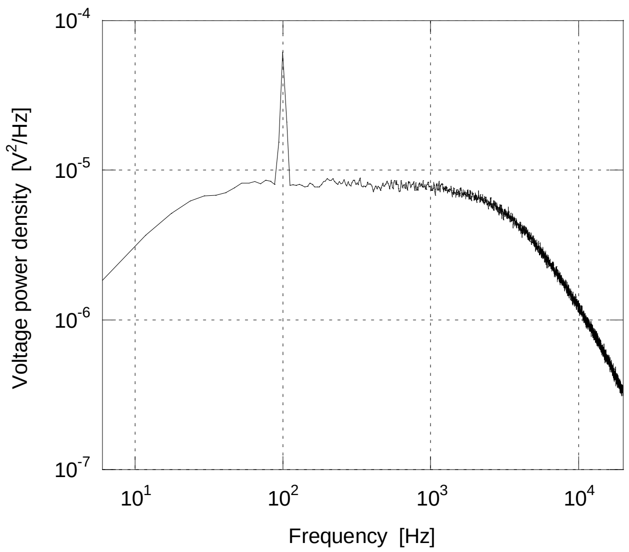

Fig. 3. Voltage power spectrum of the analysed signal $S(t)$ versus frequency at the reference 100 Hz harmonic signal $R(t)$ and sampling frequency 48 kHz when applying the sound card C-media 9739A; signal-noise ratio was -20 dB.

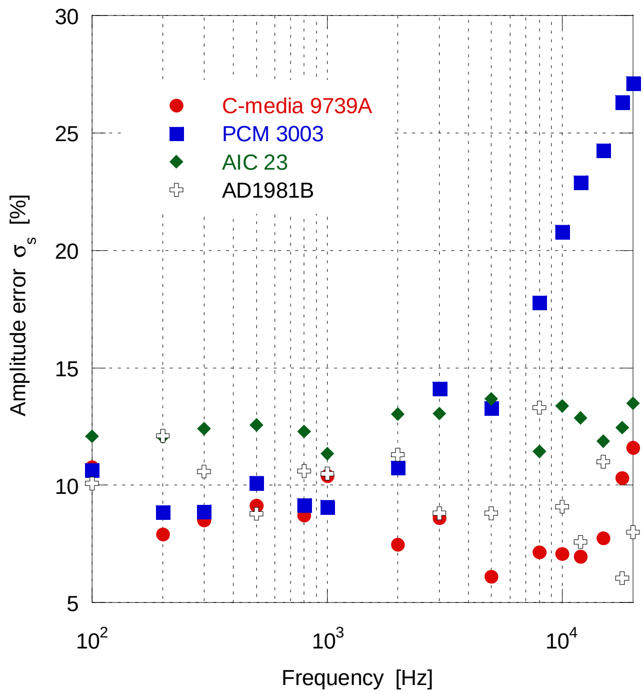

Fig. 4. Error σ_S of the amplitude module $|S|$ estimated after 1 s long signal sampling; signal to noise ratio was -20 dB.

Figure 4 and fig. 5 present the detailed results of amplitude and phase errors versus frequency. We can assess that the best solution within the investigated set is the PC sound card C-Media 9739A. The errors are close to the background level predicted by the computer simulations for the established measurement time and the assumed signal-noise ratio. Thus, eventual measurement error reduction means longer averaging and measurement time. The worse result is observed for the AIC 93 codec due to the built-in programmable amplifier that introduces additional noise, even if its gain is set to one.

The biggest error at the upper part of the acoustic bandwidth was observed for the PCM 3003 codec. We suppose that the reason for this effect is lower oversampling rate than in other tested sigma-delta A/D converters. For example, AIC 23 codec supports the industry-standard oversampling rate of $256{\cdot}f_s$ when the PCM 3003 codec sustains only $64{\cdot}f_s$ rate that deteriorates its parameters as a lock-in amplifier at frequencies higher than 10 kHz according to the detailed specification presented by the producer of both codec's - Texas Instruments.

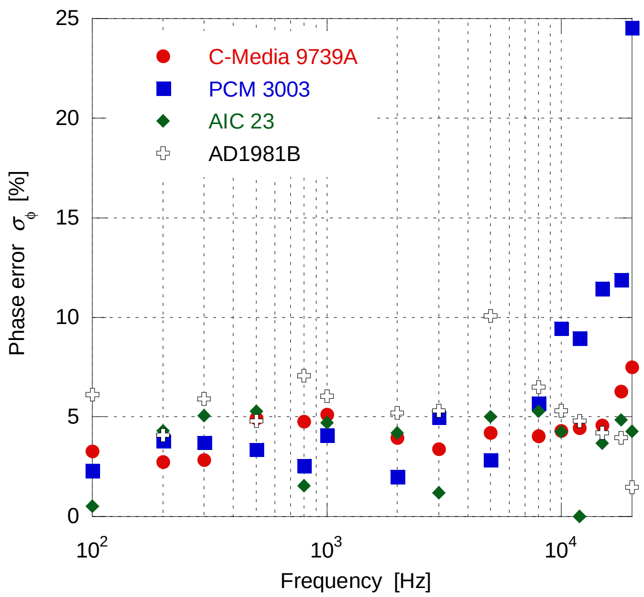

Fig. 5. Error σ_ϕ of the phase ϕ estimated after 1 s long signal sampling; signal to noise ratio was -20 dB.

4. CONCLUSIONS

This experimental study assessed quality of the economical digital lock-in amplifiers, realized by various PC sound cards and the DSP-based DSK6713 board from Texas Instruments. We conclude that all the considered solutions can be applied in practice for measurements within acoustic frequency range but quality of these systems depend strongly on the applied codec. The best solution is the cheap C-Media 9739A sound card. This card assures accuracy of the realized lock-in amplifier close to the theoretical limit established by the carried out computer simulations.

The presented conclusions would probably differ when the measurements were longer than 1 s to assure lower random error. Thus, the more expensive solution would present its advantages. We suppose that in this case the better results should be achieved by applying the high quality measurement board with a 24-bites A/D converter which nonlinearity is corrected by initial procedure of the third order polynomial approximation (e.g.: data acquisition board NI 4474). Unfortunately, this solution is many times more expensive than the considered in this paper and therefore can not be suggested due to economical reasons.

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