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'BLIND' CORRECTION OF THE MEASUREMENT DYNAMIC ERROR – SIMULATION INVESTIGATIONS OF THE SECOND-ORDER SYSTEM

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Abstract – The paper presents a measuring system, which performs auto-correction of dynamic errors caused by its analogue input circuits. Input signal is used for self-identification of the system parameters. Influence of sampling frequency and A/D converter resolution on the correction efficiency has been investigated.

A range of the system analogue part parameters variation, which allows for the efficient correction, has been determined. A measure, which determines the effectiveness of correction as related to the measurement result without the correction has been introduced. Investigation has been carried out for the second-order inertial plants.

Keywords: dynamic error, correction, simulation

1. INTRODUCTION

The presented method of dynamic correction of a measurement assumes simultaneous identification of dynamic properties of the measuring system and correction of the instantaneous value of dynamic error caused by analogue part of the system.

The foremost advantage of the investigated method is that calibration of the measuring channel is made exclusively by means of constant calibration signals. For the time-varying signals the measuring system performs self-identification of its dynamic properties using solely the measured signal at the system operating site.

As opposed to other methods of the dynamic error correction (e.g. series or parallel ones) there is no need of precise identification of parameters of the measuring channel model related to its dynamic properties. It is sufficient only to estimate the parameters variation range.

Due to the current identification of the measuring channel dynamic properties their changes are immediately detected and taken into account by the correction procedure.

The main goal of this work was to create tools, which by means of simulation methods, allow estimating a permissible range of changes for the parameters which characterise dynamic properties of the analogue part of measuring system, with minimal expected correction effectiveness. There is also a possibility of investigating influence of sampling frequency, A/D converter resolution and the correction algorithm parameters on the correction effectiveness. Investigations have been carried out for the second order inertial input circuits. Simulation results, selected as illustrative examples, are presented in a form of contours.

The chosen algorithm depends on algebraization of differential equations, which describe dynamic behaviour of the analogue part of the measuring system and on selection of algebraic equations set which satisfies criteria of minimal condition number.

2. THE SYSTEM STRUCTURE

Although the task seems infeasible, using the measuring system of the structure presented in Fig. 1 makes it possible [1]. The system employs two channels, measuring the same input quantity u(t). Dynamic properties of the channels are determined by their pulse responses $g_1(t)$ and $g_2(t)$ in the time domain or by the transmittances $G_1(s)$ and $G_2(s)$ in the complex variable domain, while the channels have been assumed linear. The analogue channels' output signals $x_1(t)$ and $x_2(t)$, which contain dynamic errors are transformed to the digital form x_{1i} and x_{2i} at the instants of sampling t_i. Reconstruction y_i of instantaneous values of the measured quantity $u(t_i)$ respectively for instants t_i is carried out in real time by a signal processor executing the investigated correction algorithm.

In order to obtain proper results of the correction two conditions have to be fulfilled. The first one assumes that both channels have the same value of gain k.

The second condition requires dynamic properties of the both channels to be different. It means any pole or zero occurring in the transmittance of one channel must not occur in the second one. Failing to meet this condition causes the correction task has no unique solution.

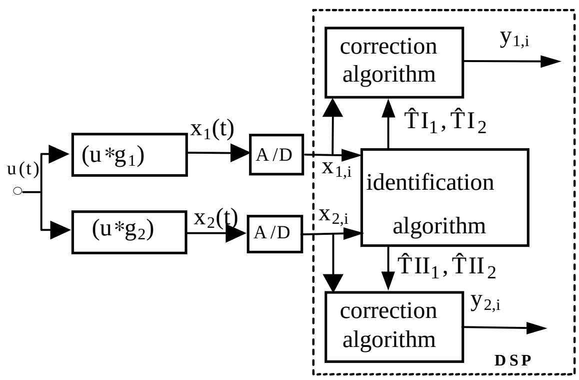

Fig. 1. Block diagram of the system

The correction algorithm contains two tasks hierarchically executed.

The subordinate task performs identification of parameters of the analogue channels dynamic behaviour model. The supervisory task carries out the correction of dynamic errors using results of the subordinate task execution. The correction is not carried out in real time, till the moment of obtaining a satisfactory result of the subordinate task execution. Samples of the recorded signals could be stored until result of the identification is obtained in order to correct dynamic errors in a post-processing mode. The supervisory task is started after obtaining the first satisfactory result of the subordinate task execution. At the same time the subordinate task is running in background, checking currently if the dynamic properties of the channels have changed. In case of detecting such a change the results of a new identification are transferred to the supervisory task. If detected change of the parameters is significantly large and fast, the supervisory task is informed of the system mistuning and waits for the actual result of identification while only recording the acquired signals. The measuring system will correctly operate in real time at slow changes of the dynamic properties of the analogue channels. In case of fast changes of those properties only recording of the measured signal is possible and next the correction of dynamic errors upon completion of the recording.

The supervisory algorithm allows for simultaneous correction of both channels. To the instantaneous value of the recorded signal in each channel is added a correction, which is determined as a sum of products of instantaneous values of the recorded signal derivatives in a given channel and the corrector parameters determined on the basis of identification result.

Practical results of the correction using this algorithm always contain an error. Thus the reconstructed instantaneous value of a measured quantity in the first channel y_{1i} will differ slightly from the result of correction at the second channel y_{2i} for the same time instant t_i. Each of them is equally probable. It was then assumed the measurement result with correction y_i is determined as a mean value of the both channel correction results for a given instant of time.

3. IDENTIFICATION METHOD

The method of identification consists in algebraization of differential equations which describe the dynamic properties of analogue channels in each instant t_i [2]. From these equations an algebraic system with square matrix A and of the condition number [3] closest to unity is selected (1).

$$\mathrm{cond}(A) = \|A\|_\infty \cdot \left\|A^{-1}\right\|_\infty \tag{1}$$

A necessary condition of differential equations algebraization is developing such an algorithm of determining n-th order derivative of the signal, represented by a sequence of its discretized samples, which would be of small sensitivity to the quantization noise and to the noise contained in the measured signal. In this investigation a sequence of N subsequent samples of the recorded signal is used in determination of the instantaneous value of a derivative. Such a fragment of the signal is approximated with a polynomial of the degree no lower than the order of the derivative to be found. The instantaneous value of respective derivative of this polynomial, taken for a time instant conforming with midpoint of the time interval determined by N samples, is assumed to be solution sought for the recorded signal derivative.

4. CRITERIA OF EVALUATING THE CORRECTION

Dynamic errors of the measuring system, with and without the correction were preliminary evaluated using standard norms $\|\varepsilon\|_1$, $\|\varepsilon\|_2$ and $\|\varepsilon\|_\infty$ determined for the vector ε of coordinates equal to the instantaneous values of dynamic error at the instances of sampling. Because the norm $\|\varepsilon\|_\infty$ involves more stringent criteria of evaluation than the two preceding ones it has been chosen as the sole criterion of evaluation for further simulation investigations.

There has been introduced an index evaluating effectiveness of correction Q (2) defined as a quotient of the value of dynamic error vector ε_m norm, determined for the faster channel without correction, by the value of dynamic error vector ε norm for the system with correction. The index shows how many times the errors will be reduced using the correction.

$$Q = \min_{m=1,2} \|\varepsilon_m\| / \|\varepsilon\| \quad : m - \text{channel index} \tag{2}$$

where: $\varepsilon_i = y_i - k\,u_i$ coordinates of vector ε

$\varepsilon_{mi} = x_{mi} - k\,u_i$ coordinates of vector ε_m

5. TEST SIGNAL

A ramp signal has been taken as an input test signal for simulation tests. A rate of rise of the ramp could be a measure of a maximum rate of the measured signal changes. The ramp signal could be used as the measured signal majorant for which the dynamic errors would be evaluated. Actuation of the system with a step input signal (usually represented by the square wave signal) could be considered as the boundary case of a trapezoidal signal.

6. SIMULATION INVESTIGATIONS

6.1 Analogue part of the investigated system

Models of each of the analogue channels (3), investigated in the simulation tests have been given in a form of two cascade-connected, first-order inertial plants of unity gain k=1 and time constants TI_1 and TI_2 for the first channel, and TII_1 and TII_2 for the second one.

$$G_I(s) = \frac{1}{1+sTI_1}\,\frac{1}{1+sTI_2}, \quad G_{II}(s) = \frac{1}{1+sTII_1}\,\frac{1}{1+sTII_2} \tag{3}$$

6.2 Parameters of the experiment

L-bit A/D converter of input voltage range ± Um and resolution of L (16, 20 or 24) bits has been modelled as a quantizing operation. Sampling period T_s has been chosen equal to one time unit.

A ramp function rising from 0 to Um in duration of 100 T_s has been used as the test signal.

Derivatives of the recorded signal have been determined using polynomials of the first and second order and a window containing N=7 consecutive samples.

The model time constants have been expressed as values normalized with respect to the sampling period. A variation of the normalized time constants from 1 to 100 with a 2.5 step in the first channel has been assumed for simulation test. Values of time constants for the second channel were set during the experiment in such a manner that their coincidence with those of the first channel will not occur.

6.3 Simulation results

Subsequent contours in Fig. 2 represent the correction effectiveness Q and the corresponding condition number using the investigated method for a linear input signal. These contours represent two-dimensional sections of a 5-dimensional space. The axes display values of the first time constant TI_1 and the second one TI_2 of the first measuring channel. Value of the index Q is represented by intensity of shading as defined in the appended legend. Permissible range of variation of the time constants values TI_1 and TI_2 at the set values TII_1 =12 and TII_2 = 67 of the second channel, could be determined for assumed correction effectiveness using the presented drawings. For instance, if the assumed correction effectiveness shall be not worse than 100 then the point determined by these time constants should belong to the most intensely shaded area. If the estimated range of changes of the first measuring channel time constants overlaps with the given shaded area, the assumed correction effectiveness will be achieved. Identical analysis should be made for the estimated range of changes of the second measuring channel time constants. In the course of this analysis it should be checked if the considered range of changes of time constants overlaps the area where the condition number attains small values. Consequently, the area of satisfactory correction effectiveness could be significantly shrunk. Operating the algorithm within an area of large condition number may cause measurement result containing significant numerical errors. These errors result from propagation of inaccuracy in determining values of the elements of matrix A. The intensity of propagation is directly determined by the condition number.

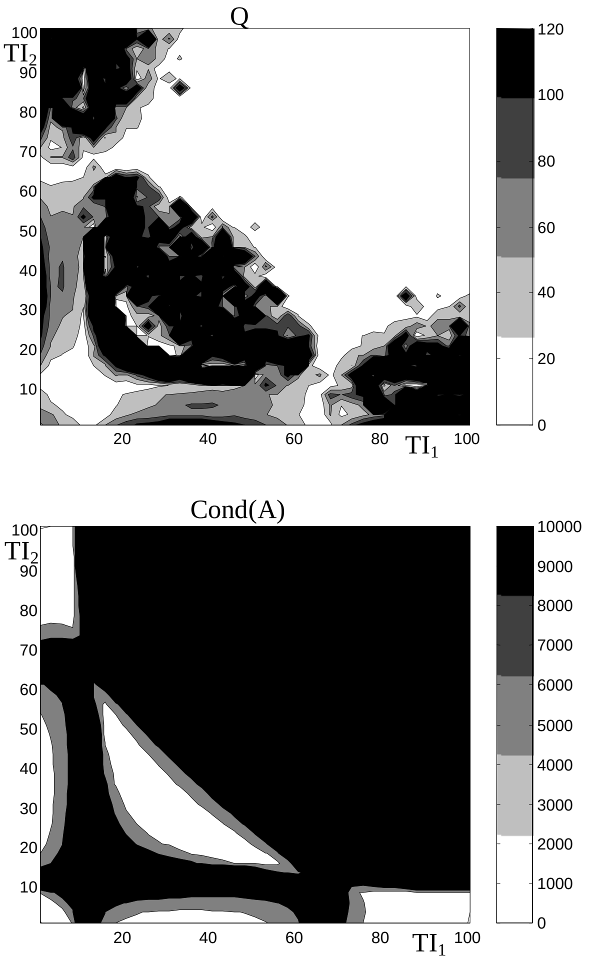

Fig. 2. Correction effectiveness Q and condition number cond(A) vs. time constants of the first channel for fixed time constants (12, 67) of the second one

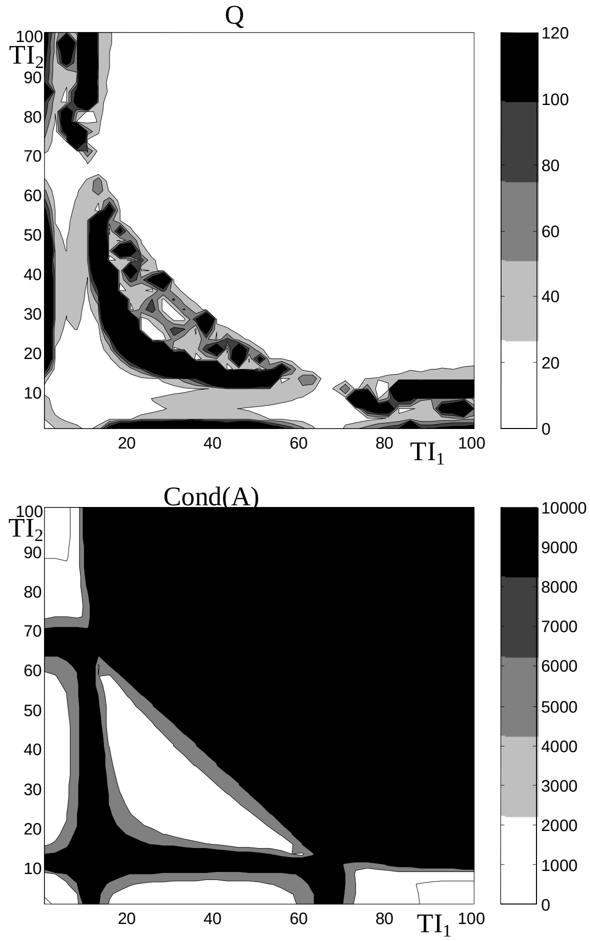

Fig. 3. The case like in Fig. 2, but the sampling frequency reduced in half

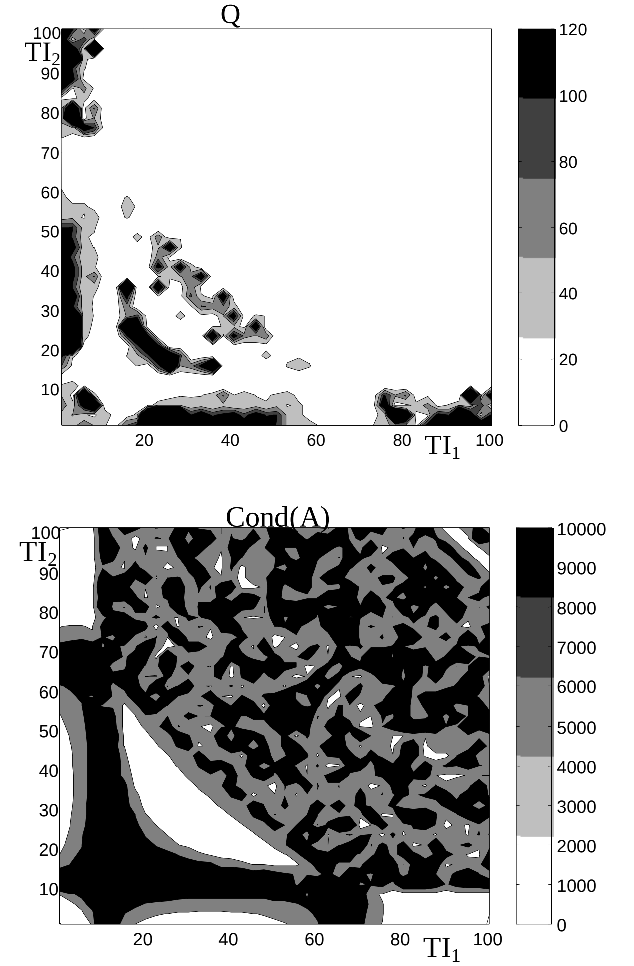

Fig. 4. The case like in Fig. 2, but the A/D resolution reduced to 20-bits

Reduction in half of the sampling frequency is shown in Fig.3. Two times lower sampling frequency does not cause significant changes of the condition number for the investigated case. However the high effectiveness area becomes significantly reduced, still remains the area, similar to that from the former case, which overlaps with small values of the condition number.

It should be noted that investigating this algorithm with respect to the first-order plants the condition number values of tens, or several tens at the most, have been obtained.

Correction effectiveness strongly depends on the converter resolution. Former investigations of the first-order plant had shown that a measuring system employing 12-bit converters achieves correction effectiveness above 100 in wide range of time constants changes [2]. A comparable correction effectiveness for the second-order plants can be achieved using 24-bit A/D converters. Effect a 20-bit converter A/D being used without reducing the sampling frequency is shown in Fig. 4. The area of high and medium effectiveness has been dramatically reduced. The condition. number values are noticeably decreasing but outside the correction area.

7. CONCLUSIONS

The simulation investigations have shown that the tested correction algorithm significantly reduces the dynamic error of time-varying signals measurement. In the consequence of this works has been created the tool, which, by means of simulation, allows verifying the effectiveness of investigated method for a selected set of parameters describing dynamic properties of measuring channel. It is also possible estimating influence of other parameters of the measuring system on the correction effectiveness. Results of the simulations have confirmed usefulness of the tested algorithms for correction of dynamic errors. Existence of the areas of the analogue part dynamic parameters variation, for which very good effectiveness of correction could be obtained, has been demonstrated. This feature enables correctors self-tuning to a priori unknown dynamic properties of the channels and continuous following-up their fluctuations.

The investigated method show different sensitivity to noise influence. Noise added to the measured signal has no effect on the accuracy of identification of measuring channels' time constants. However noise interference in one or both of the twin channels causes significant identification errors and, as a consequence, deteriorates correction quality.

Theoretical analysis of the investigated system leads to the conclusion that the analogue part of the system need not contain channels of the same order. Cooperation of channels of the first- and second-order also allows for advantageous correction effectiveness.

The correction task could also be solved when the gain of one of the channels remains unknown. In this case however the dimension of equation set increases and that worsens its properties in terms of numerical calculation.

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OTHER REMARKS:

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