XVI IMEKO World Congress
Measurement - Supports Science - Improves Technology - Protects Environment ... and Provides Employment - Now and in the Future
Vienna, AUSTRIA, 2000, September 25-28

AN ADAPTIVE ALGORITHM FOR THE MEASUREMENT DATA COMPRESSION

W. Gawedzki
Department of Instrumentation and Measurement
University of Mining and Metallurgy, al. Mickiewicza 30, 30-059 Krakow, Poland

Abstract: A proposal for a lossless adaptive compression algorithm to be used in continuous measurement data recording is presented in this paper. The operation implemented by the algorithm can be termed the signal compression before quantization. Although the algorithm itself is inertialess, its efficiency depends on the dynamics of the measurement signal compression. Thanks to the adaptation of the compression algorithm parameters to the variability characteristics (i.e. to the dynamics) of the signals recorded, the compression level can be optimised. The algorithm, method for the adaptation of the algorithm parameters and the compression effect obtained for exemplary measurement signals are presented in this paper.
Keywords: data compression, data acquisition, adaptive compression algorithm

1 DEFINITION OF THE PROBLEM

Seeking the compression algorithms to be applied for the continuous measurement data recording is connected with the need for a solution to the problem of the continuous long-term and simultaneous logging of the measurement signal static (or low-variable) and dynamic components with no loss of information essential from the measurement purpose point of view. There are measured physical quantities of the prevailing static or a low-variable character of their variability and with dynamic components occurring at certain time instants or time periods. For example, the strain state of a rock mass in the mine is of a static or low-variable character under normal operating conditions and of a dynamic character during rock bursts or blasting. In such cases, it is necessary to register both components, the quasi-static and dynamic one, which is in general contradictory to the relation between a relatively short time of signal sampling regarding the dynamic component measurement, and the requirement for the registering device memory capacity, necessary for both continuous and long-term monitoring.

The application of a continuous, „pipeline", data logging with sampling frequency adjusted to the dynamic component measurement, and the simultaneous compression of the measurement data can be a solution to the problem. The existing data compression algorithms are well recognised, analysed and implemented for the needs of the image and sound processing [1,2]. Regarding the conditions for the compression of the „pipeline" registered measurement data and the fact that the nature and properties of the measurement signals of the physical quantities as well as the signal logging quality requirements differ from those for the video and sound signals, this results in the fact that not all existing algorithms can be directly applied and an analysis for their metrological properties is necessary.

The measurement signal compression of a physical quality can result in a substantial reduction of the amount of the information memorised; and the signal compression before quantization can be one of the methods. The problem of signal compression before quantization is known in the field of sound signal transmission, where the emphasising of weak signals with the simultaneous compression of strong signals enables a constant relative quantization error to be obtained. The operation of the above-mentioned algorithms consists in the one-to-one inertialess and nonlinear processing operations where logarithmic functions are used. The application of these algorithms to compression in the continuous measurement signal logging process does not produces the expected results.

The authors have proposed in the paper another version of the inertialess lossless adaptive algorithm for the compression before quantization which leads to a reduction in the absolute and relative quantization errors both for weak and strong signals at the unchanged quantizator resolution (the processing accuracy improvement effect). In other words, which - at the reduced quantizator resolution - allows for the preservation of the assumed input quantization error (the effect of the registered measurement data lossless compression). Although the algorithm itself is inertialess, its efficiency depends on the dynamics of the measurement signal undergoing compression. Thanks to the adaptation of the compression algorithm parameters to the *variability characteristics* (i.e. to the *dynamics*) of the signals recorded, the compression level can be optimised.

2 AN ADAPTIVE COMPRESSION ALGORITHM

The proposed compression algorithm consists in the inertialess, lossless and nonlinear conversion of the measured signal $u(t)$ before the quantization. This conversion can be carried out in an analogue or digital way to the measured signal $u(n)$ sampled and quantized with much higher resolution according to the following coder function:

$$u_1(n) = \big(u(n) - u(n-1)\big) \cdot m(n) \qquad for \quad n = 1,2\ldots \tag{1}$$

and

$$u_1(0) = u(0) \qquad for \quad n = 0 \tag{2}$$

where the multiplier $m(n)$ is expressed as:

$$m(n) = 2^{M(n)} \qquad for \quad n = 0,1,2\ldots \tag{3}$$

and $M(n)$ is an integer defined as:

$$\left.\begin{aligned} M(n) &= \left\lfloor \log_2 \frac{U_z}{|u(n) - u(n-1)|} \right\rfloor \qquad for \quad n = 1,2\ldots \\ M(0) &= \left\lfloor \log_2 \frac{U_z}{|u(0)|} \right\rfloor \qquad for \quad n = 0 \end{aligned}\right\} \tag{4}$$

and meeting an additional constraint:

$$0 \le M(n) \le N - 1 \tag{5}$$

where: U_z – measuring range of the signal $u(n)$ recording system.
The number N of bits results directly from the assumed limiting quantization error $2 \cdot \delta_q = 1/2^N$ with which the measured signal $u(n)$ is recorded.

The compression algorithm works as follows. For a specified sample of the signal $u(n)$, a difference is determined between its value and the sample value just before (relationships (1) and (2)). The difference is then multiplied by the coefficient $m(n)$ which is determined according to equations (3) through (5). It is easy to see that the value of $m(n)$ depends on the signal dynamics: the larger the difference between the values of the successive samples, the smaller if the value of the multiplier $m(n)$. This results from the fact that the signal $u_1(n)$ obtained from such a conversions has to be contained within the given measuring range U_z. It should be noted that the multiplier $M(n)$ determined according to (4) for a very small (even almost zero) difference between the values of the successive samples will assume a large value exceeding considerably N. The limitation (5) is therefore necessary from the operation correctness and compression algorithm efficiency point of view.
The $m(n)$-fold signal difference amplification causes that the quantization error will be reduced $m(n)$ times if the quantizator of unchanged resolution is applied in the further part of the channel (or, in other words: a quantizator of resolution smaller by $M(n)$ bits may be applied, which will result in the decrease by $M(n)$ of the number of bits necessary for recording values of the signal $u(n)$ while the unchanged quantization error of the compressing system compared to the non-compressing system).

A block diagram for the adaptive compression algorithm under consideration is presented in Fig.1. After being sampled, the measured signal $u(t)$ is compressed following the relationships (1) through (5) and the resulting signal $u_1(n)$ is quantized in a $(N\text{-}M(n))$-bit quantizator. We get this way the compressed signal $u_q(n)$. Based on (5), we now see that for the limit number $M(n)=N$ - 1 bits we get a 1-bit quantizator.

Signal reproduction requires using a decoding procedure according to the decoder function:

$$\left.\begin{aligned} u_d(0) &= \frac{u_q(0)}{m(0)} \qquad for \quad n = 0 \\ u_d(n) &= u_d(n-1) + \frac{u_q(n)}{m(n)} \qquad for \quad n = 1,2\ldots \end{aligned}\right\} \tag{6}$$

where: $u_q(n)$ – signal $u_1(n)$ quantized in a $(N\text{-}M(n))$-bit quantizator
$u_d(n)$ – signal obtained after decoding.

The additional N-bit quantizator used in the coder channel and realising the quantization of the reference signal $u(n\text{-}1)$ is necessary because of the possibility of the correct decoding of the signal

$u_d(n)$. The value of the $u_d(n)$ is determined based on the value of the difference signal $u_q(n)$ by adding the value of an N-bit-quantized signal $u_d(n\text{-}1)$ (6).

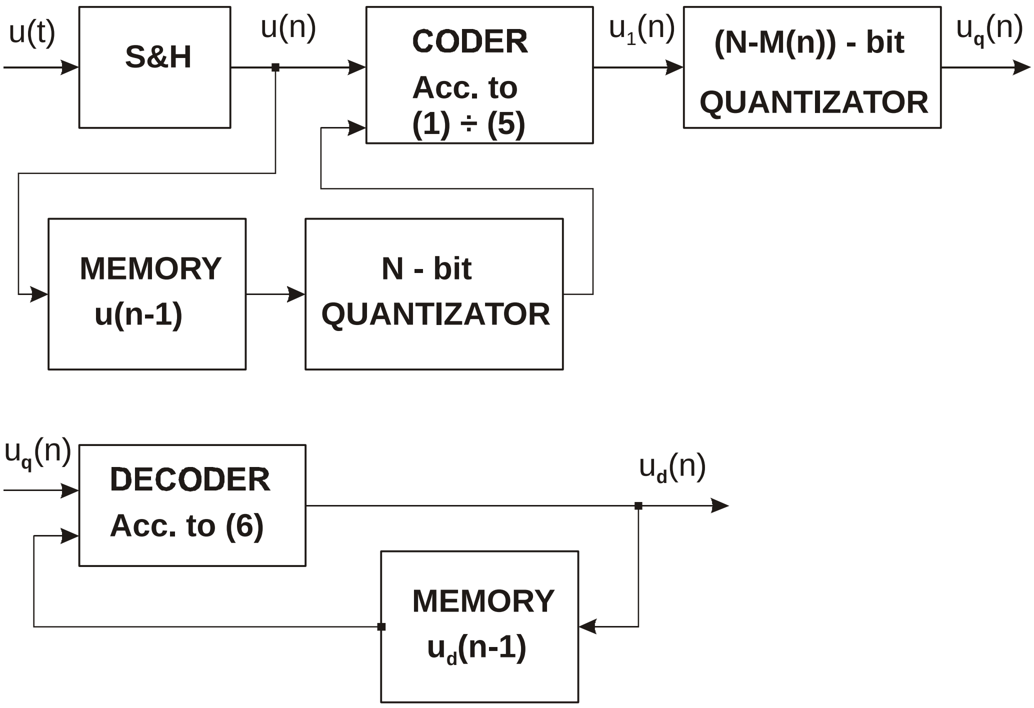

Figure 1. Block diagram for the adaptive compression algorithm

As was mentioned before, the adaptation of the compression parameter $M(n)$ to the variability characteristics of the recorded signals was used, which results in an automatic optimisation of the compression ratio. The differences between the successive samples are small for constant or slow-changing values of the signal $u(n)$; they increase with signal frequency. As can be seen in (4), the optimum compression parameter $M(n)$ is selected depending on a value of the difference.

Let us define the compression ratio G_k for the presented adaptive algorithm, referring the number Ψ of information units of a registered non-compressed signal, $\Psi(u_q^*)$, to the number of information units of a registered compressed signal, $\Psi(u_q)$:

$$G_k = \frac{\Psi(u_q^*)}{\Psi(u_q) + P \cdot 1} \tag{7}$$

Taking into account equations (1) through (6), the compression ratio can be then expressed by the relationship:

$$G_k = \frac{P \cdot N}{P \cdot N - \sum_{n=0}^{P-1} M(n) + P \cdot 1} \tag{8}$$

where: P – number of registered samples of the signal $u(n)$,

N – number of bits results from the assumed limiting quantization error $2 \cdot \delta_q = 1/2^N$

The value of $P \cdot 1$ in the denominator of equations (7) and (8) results from the necessity of using a 1-bit separator isolating the successive values of the binary recorded compressed data $u_q(n)$. This is because the data recording field width is variable and equals to $N\text{-}M(n)$. Thanks to the use of the separator, both the value of the compressed data can be read and the value of the assigned multiplier $M(n)$ can be calculated.

Analysing the adopted criterion of the compression ratio (8) it can be noticed e.g. that the according to (8) and for $M(n)=N-1$ and the quasi-static signal, the compression gain will be the largest and will tend to $N/2$. The analysis of the proposed compression algorithm for signals of other dynamic properties should be carried out by simulation.

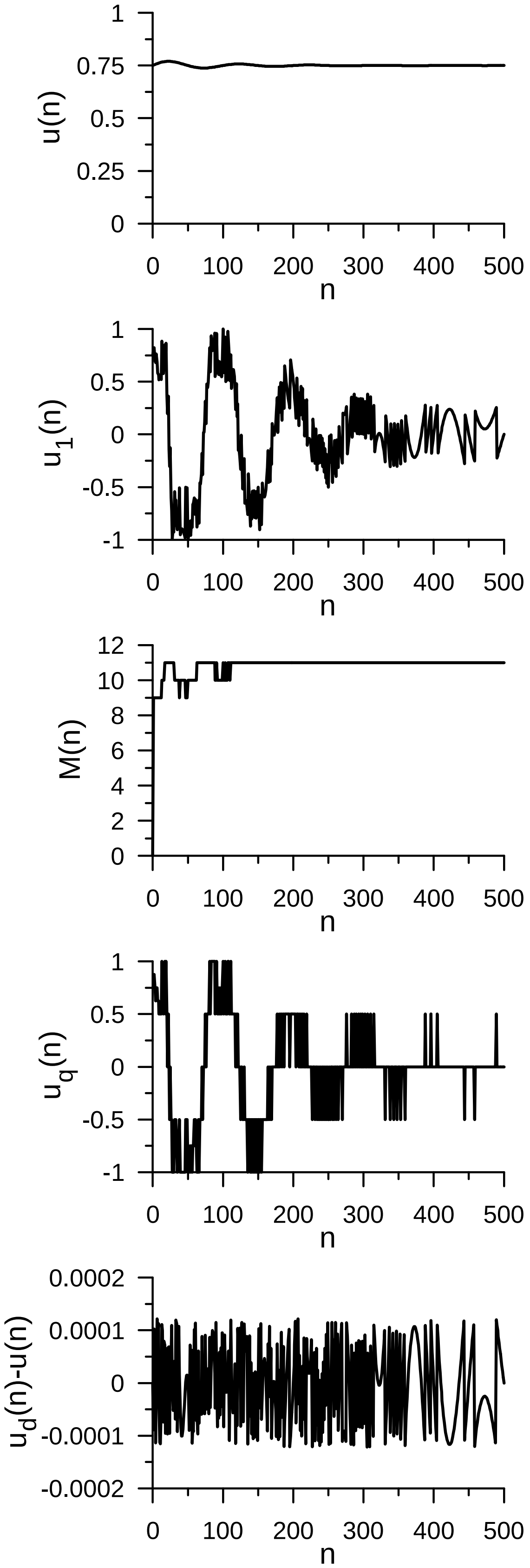

Figure 2. Illustrative discrete-time courses obtained during type ‘a’ signal compression

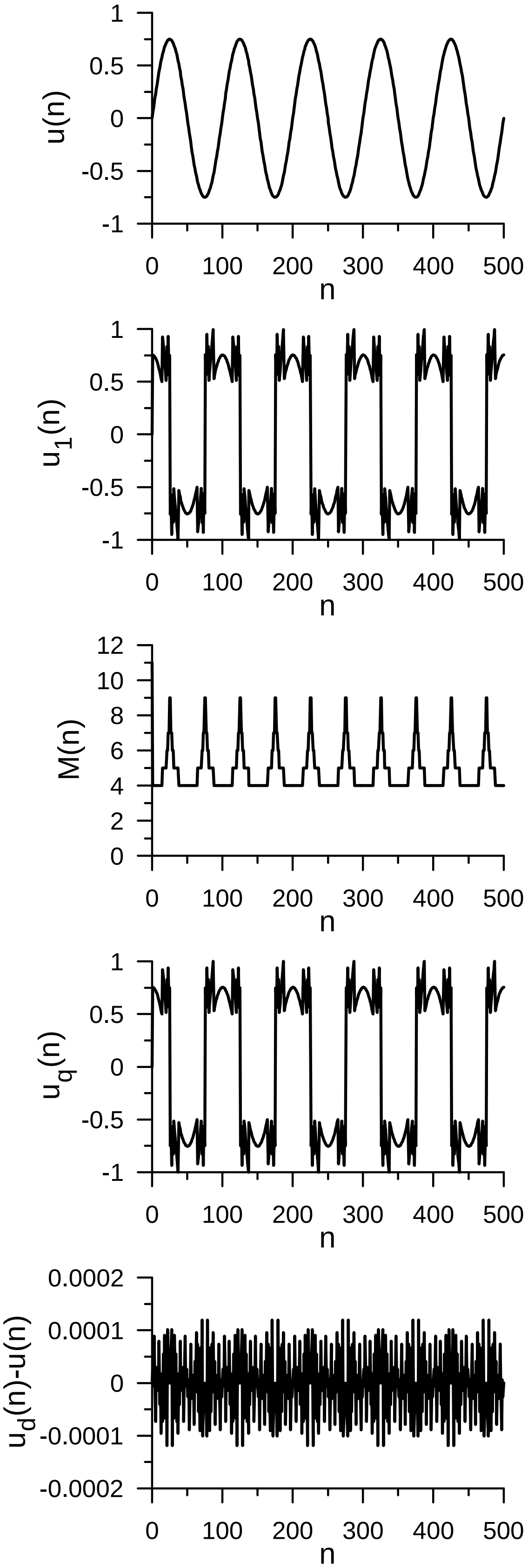

Figure 3. Illustrative discrete-time courses obtained during type ‘b’ signal compression

3 EXAMPLE OF ALGORITHM OPERATION

To illustrate the way the adaptive algorithm operates, the compression of two selected signals of various dynamic properties was made. The signals were described with the following functions:

signal ‘a’: $u(n) = 0.75 + 0.025 \cdot \exp(-n \cdot T) \cdot \sin(2 \cdot \pi \cdot f \cdot n \cdot T)$
signal ‘b’: $u(n) = 0.75 \cdot \sin(2 \cdot \pi \cdot f \cdot n \cdot T)$

where: T=0.01s is a sampling period,

$n \in [0, 500]$

The following parameters of the simulation experiment were assumed:

- $U_z = 1$, $N = 12$, $P = 501$, f=1 Hz,

The research was carried out with the Matlab simulation software [5] according to the scheme presented in Fig. 1 determining the compression ratio in accordance with (8).

Illustrative discrete-time courses characterising the compression algorithm operating way are presented in Figs. 2 and 3. Denotations in both figures are consistent with those in Fig. 1.
The values of the compression ratio determined according to (8) for investigated signals are respectively:

$G_k = 5.50$ for signal ‘a’,
$G_k = 1.48$ for signal ‘b’.

Based on the obtained illustrative values of the compression ratio, it should be stated that better effects are obtained for slower signals and also that the predicted application area for the algorithm referred just to this type signals.

It should be stressed that in the paper an idea of a inertialess and lossless adaptive compression algorithm was presented, which can be of special importance in applications of continuous and accurate measurement data recording.

REFERENCES

[1] V. Bhaskaran, K. Konstantinides, *Image and Video Compression Standards. Algorithms and Architectures*. Kluwer Academic Publishers 1997
[2] N.S. Jayant, P. Noll, *Digital Coding of Waveforms. Principles and Applications to Speech and Video*. Prentice-Hall, Inc. Englewood Cliffs, New Jersey 1984
[3] Gawê dzki W.: New Compression Algorithm for Continuous Recording of Measurement Data. *IX Sympozjum Modelowanie i Symulacja Systemów Pomiarowych*, Krynica, Poland 1999 (in polish).
[4] W. Gawedzki, J. Jurkiewicz, Investigation on the Limit Quantization Error for a General Case of a Three-Voltage Measurement Method for Ratiometric Output Sensors. *IEEE Conference IMTC’99*, Venice, Italy, May 24-26, 1999
[5] MATLAB & SIMULINK for Windows - User’s Guide The MathWorks, Inc.

AUTHOR: Waclaw GAWEDZKI, University of Mining and Metallurgy, Department of Instrumentation and Measurement, al. Mickiewicza 30, 30-059 Krakow, Poland, Phone: +0048 12 617 28 28, Fax: +0048 12 633 85 65, E-mail: waga@galaxy.uci.agh.edu.pl