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ELIMINATING PULSES FROM MEASUREMENT SIGNALS - CONTEMPORARY APPROACH

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Abstract - Some new methods of elimination of impulsive disturbances are discussed in the paper. The elimination procedures must differ substantially from other filtering methods because of the specific character of disturbance; a wide frequency spectrum results from its shape. The elimination procedure consists of several main steps: prewhitening algorithm, nonlinear transform, inverse filtering. Also some adaptive approaches are mentioned.

Keywords: digital signal filtering, nonlinear transforms, pulse disturbances.

1. INTRODUCTION

The elimination of disturbances is one of the most important goals of signal filtering in a great variety of measurement systems. Different optimal approaches (Wiener, Kalman, adaptive procedures, etc.) are well known and applied since many years. A very specific situation appears if the disturbances are of impulsive type - i.e. certain isolated samples of the received signal take enormous values - because of the very broad frequency spectrum of such pulses. On the other hand it must be noted that impulsive disturbances can be of different origin; they can be generated by technical devices as well as by living organisms.

An overview of possible sources of such disturbing signals can be found in [2].

The influence of impulsive disturbances on the final result of signal processing algorithms can be actually reduced in two different ways: either by using algorithms which are insensitive with respect to these disturbances or by eliminating them from the signal before processing.

A good example of the first way are all methods based on the median calculations e.g. determining particular harmonics as medians of quotients of signal samples and harmonic functions.

The second approach can be derived from analog circuitry (amplifier with limited output) and includes a lot of nonlinear algorithms, together with some filtering and prediction procedures. It has been shown that the following circumstances must be taken into consideration - signal correlation properties, shape of pulses, variations of signal and disturbance power, as well as possible adaptation - if impulsive disturbances are to be effectively eliminated.

A very specific problem is the shape of disturbing pulses, resulting from inevitable "smearing out" of nonobservable primary disturbances. Usually, the pulses after having passed a channel extend over several signal samples.

One of possible solutions of the problem will be discussed below.

2. OUTLINE OF THE ALGORITHM

The removal of disturbing pulses requires generally certain signal samples to be modified, normally their values should be substantially reduced. We assume that this intervention should concern only separate samples, that is the disturbed signal should be prewhitened already before the nonlinear operation of "cutting off" sample values. Pulses extending over several sample periods will be reduced then to separate Dirac pulses. The design procedure of the pulse elimination system includes therefore [1]:

* identification of impulsive disturbance (p.d.f.'s of amplitude and time instants when the pulses occur, shape of a particular pulse) [3],
* prewhitening procedure resulting from the above model,
* nonlinear processing,
* recovery of the disturbance free signal.

The first point (modelling of disturbance) depends on the source of impulsive disturbances which can be of natural origin or result from human activity. It was stated after many experiments that the p.d.f.'s of impulsive disturbances are mostly similar to gaussian distributions, except for their "tails" which are more elongated. One possible approximation of such distribution is the following sum of two gaussian p.d.f.'s having different variances and zero mean values [1]

$$f(n) = (1 - e) f_N(n) + e f_I(n) \tag{1}$$

where $f_N(n)$ is the "nominal" component with variance σ_N^2, whereas $f_I(n)$ is the "impulsive" component having variance σ_I^2 . It is normally required that $\sigma_I^2 >> \sigma_N^2$ and $0 < e < 1$, $e << 1$.

A uniform distribution of time instants when the pulses apear was assumed.

An important problem is the mathematical model of signal shaping procedure which must be determined in order to implement the "whitening" procedure. In most cases an autoregressive model of a particular order can be applied, which corresponds to a typical IIR filtering procedure.

In our case the following autoregressive formula is assumed

$$y(n) = x(n) + b_1 y(n-1) + b_2 y(n-2) \qquad (2)$$

where x*(n)* = *s(n)* + *n(n)* is the sum of original information (measurement) signal and disturbance. It corresponds usually to inevitable linear distortion of signals by transducers, transmission lines etc. In order to minimize the effect of correlated disturbing pulses and to improve the process of their elimination an initial filtering circuit ("prewhitening") should be used. Exactly, this filter should be characterized by a transfer function, inverse with respect to the transfer function of the above model. Nevertheless a simplified approach is possible where the prewhitening filter is of first order. The quality of filtering was evaluated for both (precise and approximate prewhitening processes) cases. The primarily "extended" pulses will be thus compressed to single (separate) pulses which can be eliminated by the use of an amplifier limiter, more generally by the use of a corresponding nonlinear circuit or algorithm. A certain delay of the output signal depending on the computational complexity of the procedures and on the throughput of the system can be then expected.

Generally, the above mentioned nonlinear circuit "cuts off" samples taking values substantially greater than all other ones. There are two possible methods enabling us to discover samples of this kind:

- by comparing the absolute value of input sample with an assumed threshold,
- by comparing the absolute value of difference of two subsequent samples with another given threshold.

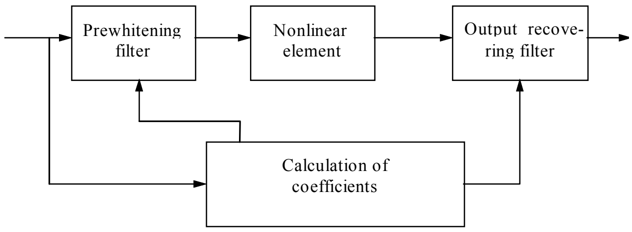

Fig.1 - Block-diagram of the pulse elimination system

In our algorithm the first possibility is implemented. A general block diagram of the pulse eliminating system is given in Fig. 1. In Fig. 2 the histogram of impulsive noise (a) and the corrupted signal (b) are shown.

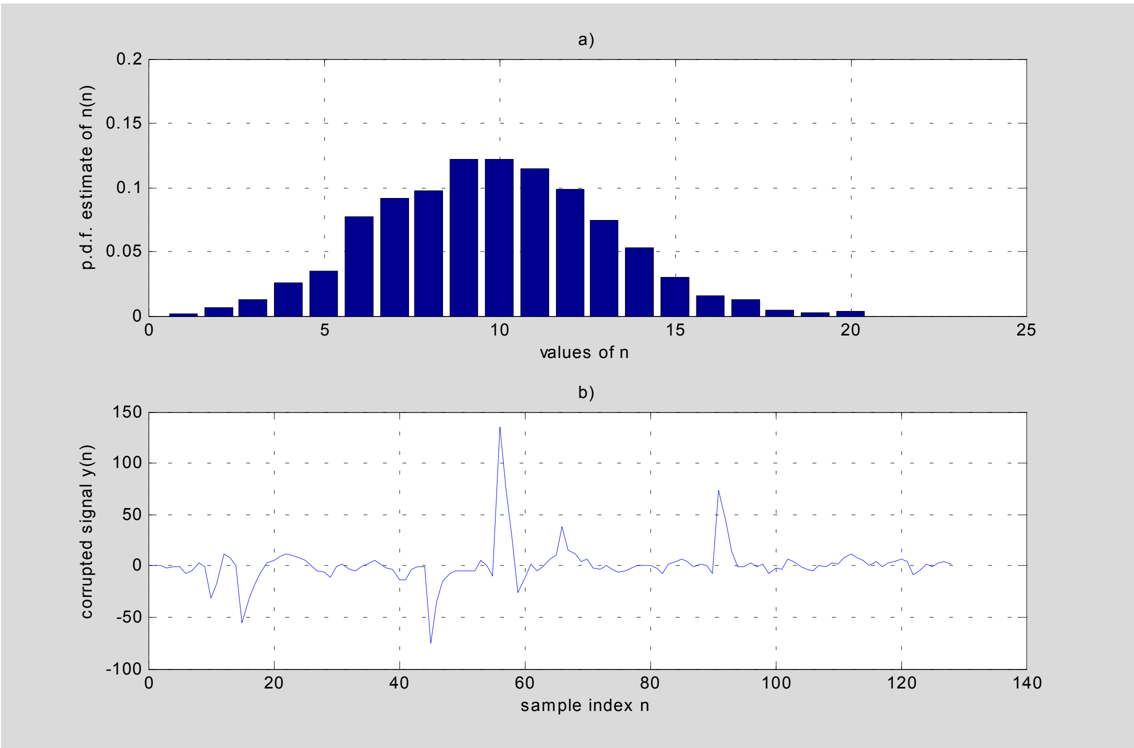

Fig. 2. Histogram of the disturbance (a) and corrupted signal (b)

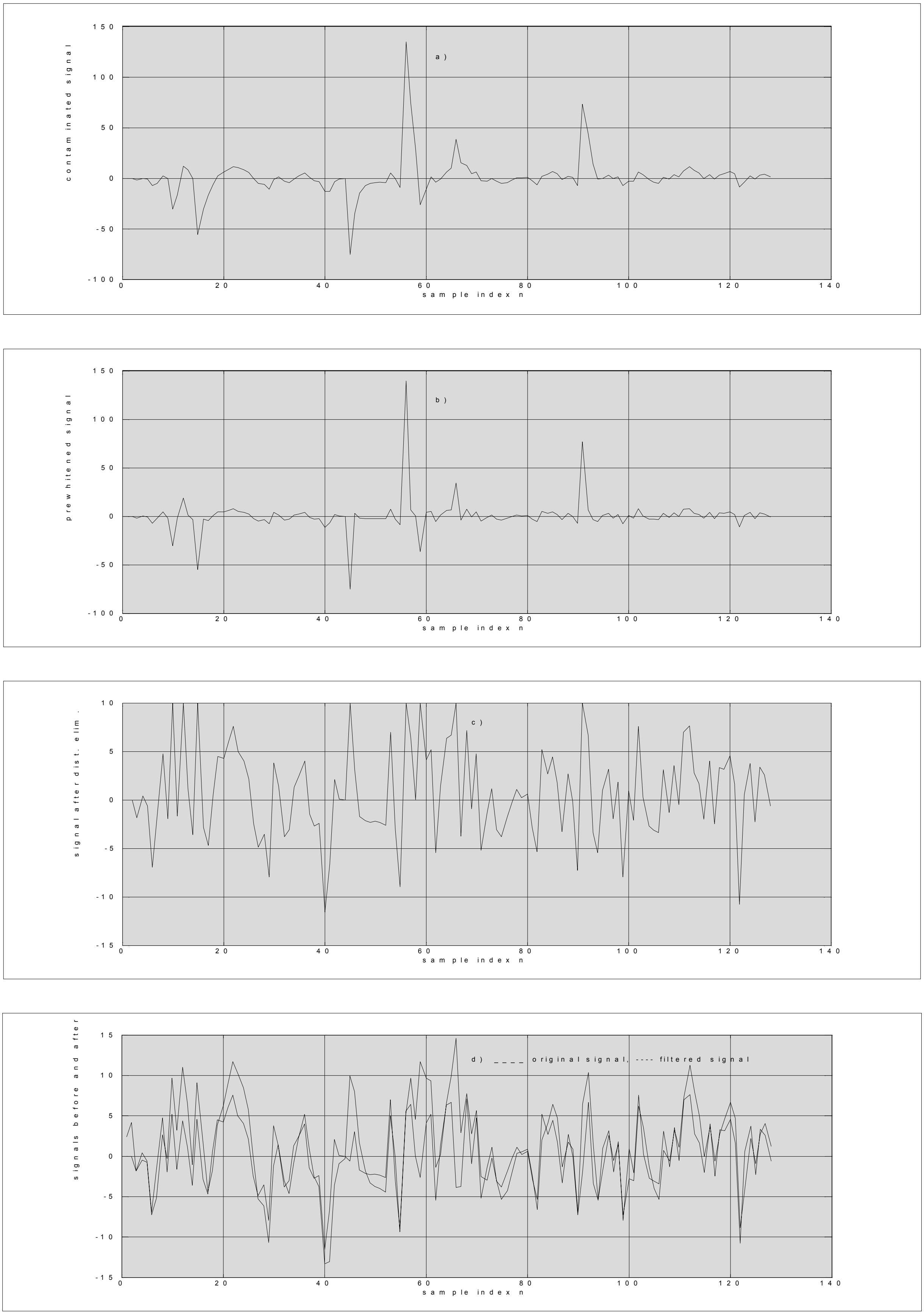

Fig. 3. Exact filtering - contaminated signal (a), prewhitened signal (b), signal after elimination of distortion (c) and recovered (filtered) signal compared with original pattern (d)

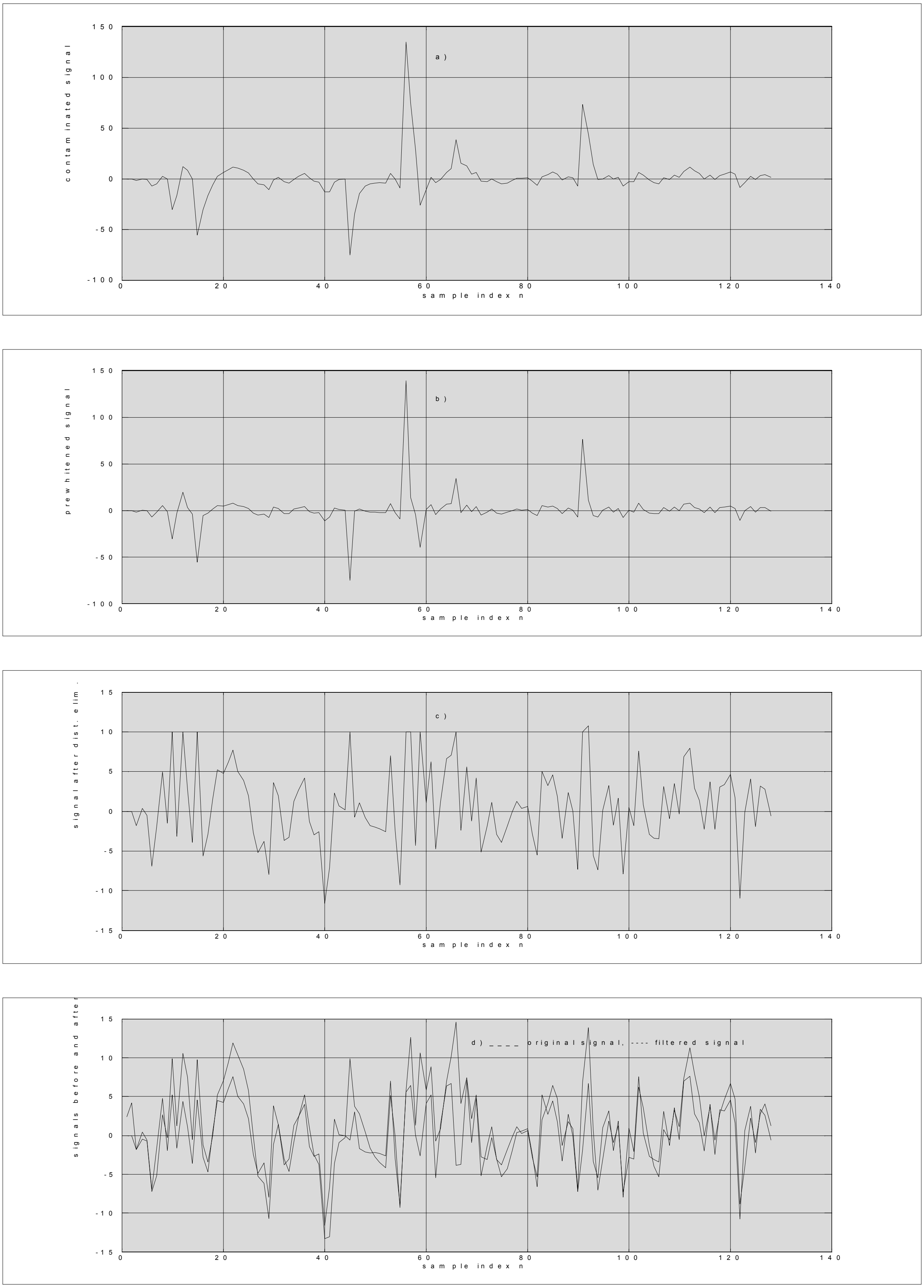

Fig. 4. Approximate filtering - contaminated signal (a), prewhitened signal (b), signal after elimination of distortion (c) and recovered (filtered) signal compared with original pattern (d)

3. SIMULATION RESULTS

In order to illustrate the above described idea of elimination of disturbances two situations were analyzed: exact filtering (prewhitening and recovering operations correspond exactly to the spectrum of input) and approximate filtering (both operations were replaced by first order difference formulae). In both cases the input signal was shaped by a second order IIR filter as determined by Equ. 2, with b_1=-0,5 and b_2=0,06. The results are shown in Figs. 3 and 4. The rms value of difference between the original and the filtered signal, resulting from Figs. 3d and 4d, equals 3,59. It means that the order of filtering does not influence substantially the quality of filtering.

It must be noted that the threshold value in "limiter" procedure was optimized in the simulation algorithm and equalled appr. 12 units of the input signal (it has been shown by the author that the procedure is practically "robust" with respect to the threshold within a certain interval). Also the first order filter in the approximate filtering case was adapted and assumed as b_1=-0,45 (for b_2=0).

4. CONCLUSIONS

The application of some well known concepts of nonlinear filtering was discussed in the contribution. It has been shown that the idea of prewhitening and subsequent nonlinear filtering of signals contaminated by impulsive noise can be implemented using some very simple arithmetical and logical operations. Nevertheless the parameters of filtering procedure should be possibly adapted to the properties (spectrum) of the filtered signal. It seems to be convenient to implement such algorithms using typical signal processors.

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