SIGNAL DENOISING USING SUBBAND THRESHOLDING

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ABSTRACT

Thresholding of subband signals is the modern method for signal denoising used in many technical branches. This article deals with denoising of stationary and non-stationary noise from useful signal. The signal measured on NMR tomography presents a good example of signal that needs removal of noise.

1 INTRODUCTION

Thresholding of subband signals is one of the more popular method for removing noise from useful signal. Fundamentals of subband signals thresholding is shown in figure 1 [1]. Using Analysis Filter Bank (AFB) the input signal is divided into series of subband signals. Partial subband signals are thresholded properly in order to optimally suppress noise without influence on the useful signal. The magnitudes of thresholds p_i are calculated within block Threshold Estimation (TE). In block of Synthesis Filter Bank (SFB) the subband signal are again synthesized into the resultant signal.

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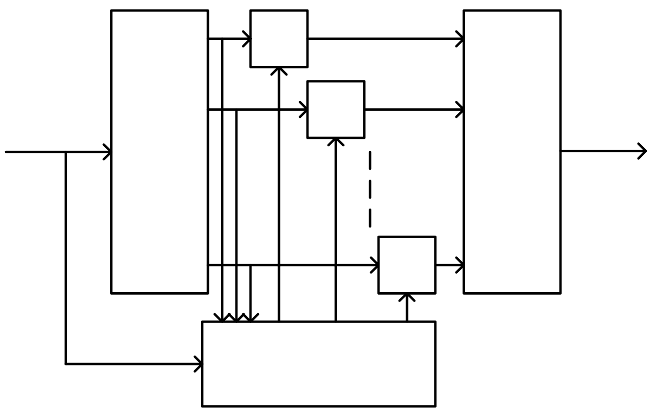

Fig. 1: *Block diagram of subband thresholding.*

Parameters selection of block parts in figure 1 depends especially on properties of input signal. Filter banks basis depends on distribution of power spectrum density. Type and order of filter banks is chosen by required decay in their stop band, efficient computing and the presence of transient effects. Filter bank without downsampling is commonly used. Using filter bank without downsampling reduces the number of numerical operation and required memory, but at the expense of a inferior result. The magnitudes of thresholds p_i are calculated by means of block TE on the basis of calculating the noise variance δ. For white noise derived equation $p = \delta\sqrt{2\ln(L)}$ was used, where L is length of the input signal. The value computed by $p = \delta\sqrt{2\ln(L)}$ is commonly to high and then we chose threshold magnitude by $p = \delta K$, where K is an empirically obtained constant. Choice of thresholding type is another important chapter. In this article, the choice was made only between soft and hard thresholding. Both of soft and hard thresholding has advantages and disadvantages.

The most demanding problem is computing of partial thresholds. If noise is stationary, then the computing of thresholds is relatively easy. On the other hand, if noise is non-stationary, computing of the thresholds is much harder and leads to time varying thresholds. Therefore we divide subband thresholding into two types, for stationary and non-stationary noise.

2 SUBBAND THRESHOLDING OF SIGNAL WITH STATIONARY NOISE

In the case where noise is stationary the noise does not change its properties, partial thresholds p_{i} are constants. For white noise, where all thresholds are identical, we have one global threshold. In other cases partial thresholds p_{i} are different and we must compute them separately. Threshold magnitude can be computed from input signals, or from partial subband signals which is more effective. The type and order of filter bank used makes nearly no difference, but the filters order has to be high enough so as to prevent signal aliasing between bands, which must be less then noise magnitude. Since transient effects do not happen, we can chose the order of the filter bank arbitrarily. Soft thresholding is better then hard thresholding, because step changes of the thresholded signals do not occur.

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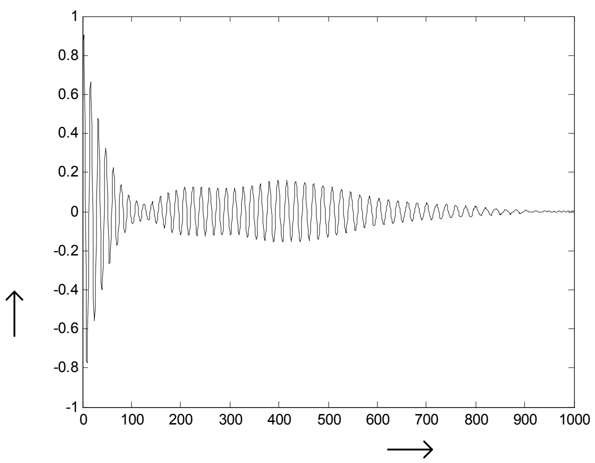

Fig. 2: *Free induction decay signal.*

An example of signal noised by stationary non-white noise is the Free Induction Decay (FID) signal measured in Nuclear Magnetic Resonance (NMR) tomography in the Institute of Scientific Instruments in Brno [2], [3]. The FID signal, shown in figure 2, is a complex signal with decreasing magnitude. Noise is stationary and then the S/N ratio decreases as well. Bearing in mind the approximately exponential distribution of power spectrum density, we use the analysing filter bank and the synthesizing filter bank of octave spectrum division – wavelet base. Threshold magnitudes are computed as K multiple of noise variance δ computed at the close of subband signals, where useful signals is zero. K constant chosen is relatively small, so as to not distort the FID signals.

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3 SUBBAND THRESHOLDING OF SIGNAL WITH NON-STATIONARY NOISE

An example of signal noised by non-stationary noise is the signal of Instantaneous Frequency (IF) computed by derivation of instantaneous phase of FID signal, shown in figure 3. The filter bank of octave spectrum division is used, again with respect to the spectrum density used. The noise in IF signal is non-stationary, it gradually increases. Due to this noise characteristic the values of thresholds p_i must be modified as related to the instantaneous noise variance in individual subbands. Computing thresholds p_{i} is a very complex process and generally requires operator intervention to define the difference between useful and disturbing signal. Use of combination of thresholds and deletion of the whole sub-band signal parts has proved beneficial.. Due to thresholds p_{i} time-dependence, step changes in subband signals occur. Any change, especially step change, is accompanied by a transient effect. The length and magnitude of transient effect is directly related to the filter bank used. Therefore we use filter banks with low order (Haar, Daubechies order 2 ÷ 5). In this way we eliminate transient effects, but at the expense of signal aliasing between individual bands, because the digital filter decay in the stop band is low. We have to compromise between attenuation of the useful leaking signal together with noise or retention of both.

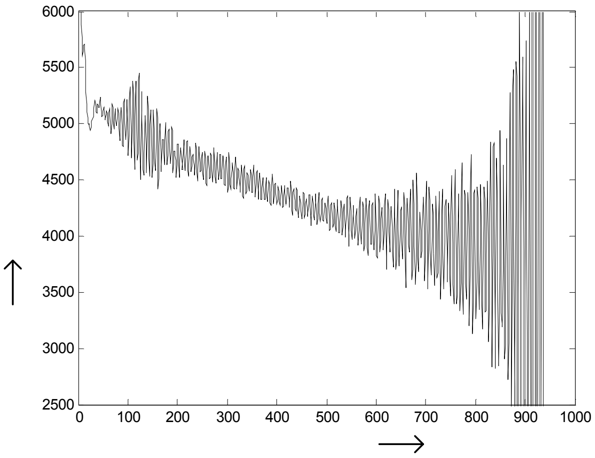

Fig. 3: *Instantaneous frequency of FID signal without filtering.*

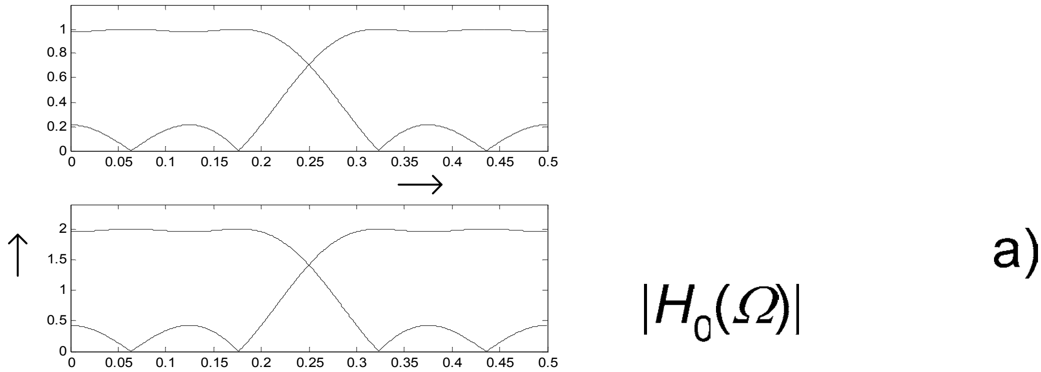

Fig. 4: *Characteristics of filter bank designed by employing Remez algorithm for N = 6*
a) Analysis b) Synthesis.

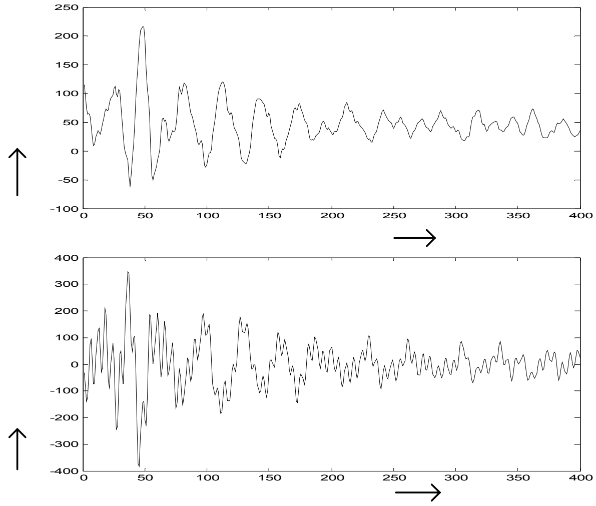

Fig. 5: *Fifth subband signal obtained by a) Remez filter bank N* = 6
b) Daubechies filter bank N = 6.

The solution to the low decay in the stop band of Haar, Daubechies types filter banks is the utilization of other filter banks with greater decay in their stop band. As optimal seems to be employing filter banks designed by Remez algorithm [4], [5], [6]. The advantage is on the one hand in evidently the highest possible decay for the tolerance field and the filter order. On the other hand, it is the direct selection of the filter bank tolerance field. The characteristic of

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the filter bank designed by employing the Remez algorithm is shown in figure 4. Subband signals obtained by Remez filter bank and by Daubechies filter bank are shown in figure 5. We see that the subbad signal obtained by Remez filter bank includes less noise.

4 CONCLUSION

Figure 6 shows the course of the instantaneous frequency signal processed by the wavelet filtering. It is apparent that the noise was suppressed significantly. A signal with useful length of 900 samples was acquired.

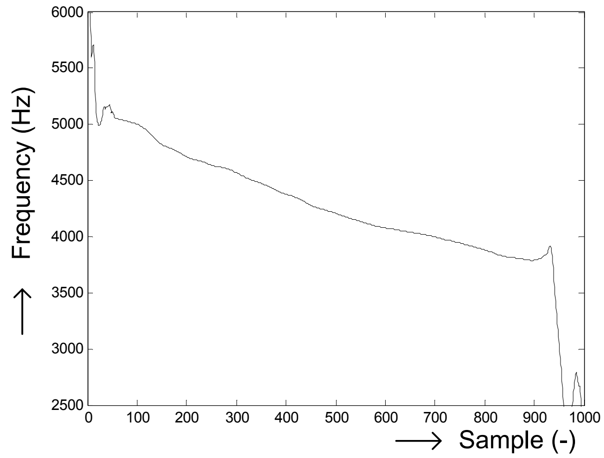

Fig. 6: *Instantaneous frequency of FID signal processed by wavelet filtering.*

REFERENCES

[1] Vaidynathan P. P.: Multirate System and Filter Bank, Englewood Cliffs, Prentice-Hall, New Jersey 1993.

[2] Bartusek, K., Gescheidtova, E.: Instantaneous Frequency of Spin Echo Method for Gradient Magnetic Field Measurement in MR Systems, Journal of Electrical Engineering, vol. 53, pp.49-52, 2002.

[3] Vlaardingerbroken, J.: Magnetic Resonance Imaging, Springer-Verlag, 2000.

[4] Fliege, N. J.: Multirate Digital Signal Processing, John Wiley & Sons, Chichester 1996.

[5] Vích, R., Smékal, Z.: Digital Filters, Academia, Praha 2000. (In Czech)

[6] Mitra, S. K.: Digital Signal Processing, McGraw-Hill, 1998.

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