Adaptive Subband Filtering Method for MEMS Accelerometer Noise Reduction

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ABSTRACT

Silicon microaccelerometers can be considered as an alternative to high-priced piezoelectric sensors. Unfortunately, relatively high noise floor of commercially available MEMS (Micro-Electro-Mechanical Systems) sensors limits the possibility of their usage in condition monitoring systems of rotating machines. The solution of this problem is the method of signal filtering described in the paper. It is based on adaptive subband filtering employing Adaptive Line Enhancer. For filter weights adaptation, two novel algorithms have been developed. They are based on the NLMS algorithm. Both of them significantly simplify its software and hardware implementation and accelerate the adaptation process.

The paper also presents the software (Matlab) and hardware (FPGA) implementation of the proposed noise filter. In addition, the results of the performed tests are reported. They confirm high efficiency of the solution.

Keywords: MEMS accelerometer, adaptive subband filtering, NLMS algorithm, vibration measurement, hardware filter

1 INTRODUCTION

Basing on the observation and analysis of a vibration spectrum it is possible to detect broken parts of a machine, determine a type of the failure and predict its future development. In the currently used measurement systems, the information on vibration magnitude is provided by expensive piezoelectric accelerometers. An alternative to these sensors can be silicon microaccelerometers produced using micromachined technologies. These devices are relatively cheap and may be a component of a complex and comprehensive vibration measurement system built as a single IC. Unfortunately, commercially available micromachined accelerometers have some limitations. One of the most significant is a relatively high level of the self-noise observed in the output signal [1]. The noise limits measurement resolution, which makes it impossible to use these sensors for precise diagnostic measurements.

A predominant noise component in the surface technology silicon microaccelerometers with low-noise acceleration detection circuits comes from Brownian motion of the proof mass. A study of the ADXL202 accelerometer produced by Analog Devices showed that amplitude-frequency characteristics of the observed noise signal from the sensor in standstill condition is constant in the frequency range up to about 2 kHz (Figure 1). This was confirmed by the analysis of autocorrelation function of the noise [1].

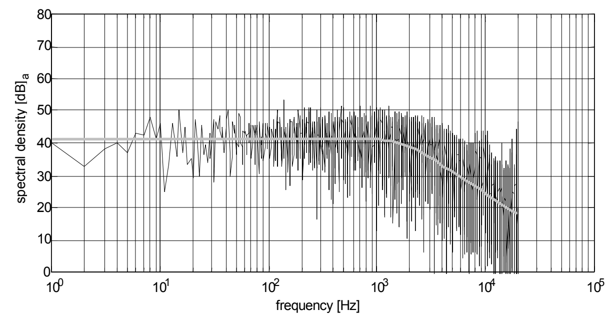


Figure 1: Power density spectrum of the accelerometer noise

Hence, in the given frequency range, the self-noise of the examined accelerometer can be treated as a white noise. The target application engages the use of MEMS accelerometers for turbogenerator vibration measurement. The most valuable sources of information for this type of machines are harmonics of vibration signal. The determination of their level and finding the correspondence with rotational frequency of the respective machine parts enables the identification of a failed part. The above observations indicate the possibility for the implementation of the Adaptive Line Enhancer (ALE) type circuit to reduce noise level.

2 ADAPTIVE LINE ENHANCERS

Adaptive Noise Canceller (ANC) circuits are based on the primary structure of an adaptive filter (Figure 2) [2][3]. The input of ANC adder is fed with the signal $x(n)$ to be filtered, which is a sum of the usable $s(n)$ and interfering $v(n)$ components, where $n \geq 0$ denotes subsequent time instants. It is assumed that the both components are uncorrelated with each other (they are independent).

The necessary condition on which the ANC circuit can work properly is that the $d(n)$ signal, which is applied at the input of the adaptive filter, is correlated with the usable component $s(n)$ and is not correlated with the interfering signal $v(n)$. However, in many real applications there is no possibility to separate the reference signal fulfilling this condition. Then the perfect solution is to apply the ALE filter. Here, the reference signal is the input signal $x(n)$ delayed in time (by d samples). The introduction of the

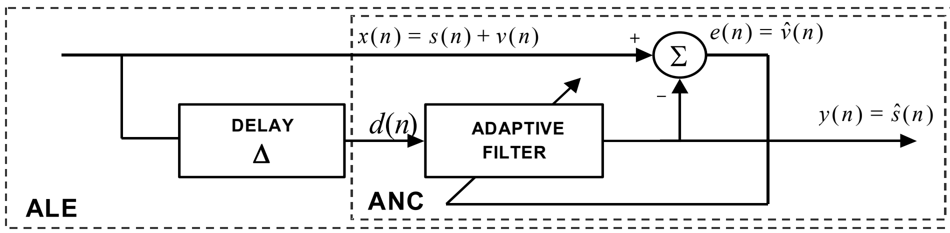


Figure 2: Structure of the Adaptive Noise Canceller (ANC) and Adaptive Line Enhancer (ALE).

delay is aimed to decorrelate the interfering component *v(n)* between the input signal *x(n)* and the reference signal *d(n)*.

The adaptive algorithm for the calculation of respective filter coefficients tends to eliminate the uncorrelated component. As a result, the amplitude-frequency characteristic of the filter adopts the shape of the characteristic of the comb filter, which passes harmonic components present in a signal being processed. The output signal *y(n)* is the estimate of the usable component *s(n)* of the vibration signal.

The analysis of the ALE filters properties shows that their noise reduction efficiency depends not only on the filter parameters but also on the processed signal features. For the ALE filter using the LMS algorithm it was proved [4] that the theoretical SNR improvement is described by the relationship:

$$SNG = \frac{SNR_{out}}{SNR_{in}} = \frac{L}{2} \frac{\sum_{m=0}^{M-1} A_m^2 [1 + (4/L)(\sigma_v^2 / A_m^2)]^{-2}}{\sum_{m=0}^{M-1} A_m^2 \cdot \sum_{m=0}^{M-1} [1 + (4/L)(\sigma_v^2 / A_m^2)]^{-2}} \quad (1)$$

It is thus proportional to the number L of filter coefficients. Additionally, it depends on the amplitude A_m (of power) of the respective harmonics in the processed signal, their number M and the interfering signal power.

As mentioned above, in the target application, the MEMS accelerometers measure turbogenerator vibrations. In the measured signal, for the frequency range of 10 Hz ÷ 6 kHz, one can observe over 100 harmonics. As a consequence, a significant SNR improvement requires the use of filters with large number of coefficients. However, some limitations appear here. The first of them is the necessity for the application of the digital ICs with large computational power to implement the filter. Moreover, in [4], it has been proved that there exists some finite value $L = L_{lim}$, for which the ability of the ALE circuit to reduce noise is the highest. Thus the maximum SNR improvement for the signal containing M harmonics is limited to the value:

$$SNG = \frac{L_{\lim}}{4M} \quad (2)$$

This effect is a result of the adaptive filter transmittance noise, which is caused by an error in the filter coefficients adaptation.

The ALE filter order affects also the frequency resolution, which determines the filter ability to differentiate between the respective harmonics in the sampled signal. In [1], it has been showed that for the signal containing M equidistant harmonics there is a necessity to apply the filter with the number of coefficients satisfying the inequality [2][4]:

$$L > 2M \quad (3)$$

The above considerations indicate that it is possible to obtain high efficiency of the ALE-NLMS filter in terms of the SNR improvement for signals that are sampled with low frequency and whose useable component contains small number of harmonics. This conclusion was a basis for the development of the adaptive subband filtering method for the vibration signal of large rotating machines.

3 ADAPTIVE SUBBAND FILTERING OF NOISE

The principle of operation of the circuit for adaptive signal processing in frequency subbands is illustrated in the below figure.

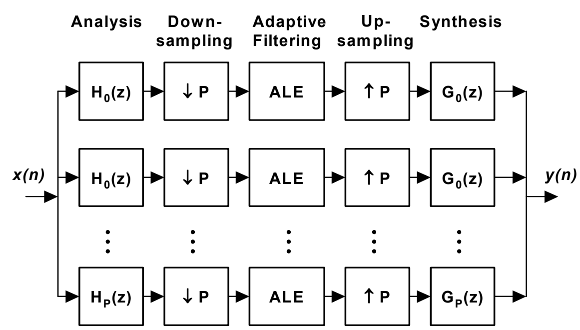


Figure 3: Structure of the system for adaptive signal processing in frequency subbands.

In the applied method, the signal is divided into 16 frequency subbands with the use of the suitably designed [1] bank of bandpass analysis filters. After the filtration, each subband signal occupies 16 times narrower band as compared to the input signal. As a result, all these signals are 16 times up-sampled. The sampling frequency for each signal is reduced by the factor of 16 with the use of reducers (down-sampling). The signals processed in this way are filtered with ALE filters. Upon appropriate processing, the frequency of each subband signal is increased with the use of expanders (up-sampling). The resultant signals are merged together in the bank of bandpass synthesis filters forming output signal of the filter.

The presented method for the signal processing in the frequency subbands enables the use of the ALE filter for the signals containing many harmonics, including a complex signal typical of rotating machine vibrations. The increase in the filter operation efficiency arises from the reduction in the width of the frequency band of the processed signal and from the decrease in its sampling frequency. The band limitation reduces the number of harmonics present in each subband. The decrease in sampling frequency results in higher frequency resolution of the filter.

4 ALGORITHMS OF FILTER WEIGHT ADAPTATION

After the preliminary analyses, it was decided to apply the NLMS (Normalized Least Mean Squares) algorithm [3]. This choice is the trade-off between algorithm and hardware implementation simplicity, on the one hand and weight adaptation precision and speed of convergence to the optimal solution, on the other hand. In the considered algorithm the update of the filter coefficients vector ***w*** is performed in accordance with the equation:

$$\mathbf{w}(\mathrm{n}+1) = \mathbf{w}(n) + 2\mu_0 \gamma \mathbf{x}(n) e(n) \tag{4}$$

where: $\mathbf{w}(n) = [w_0, w_1, ..., w_{L-1}]$ – vector of coefficients (weights) for the filter of order L, $\mathbf{x}(n) = [x(n), x(n-1), ..., x(n-L-1)]$ – vector of the subsequent input signal samples, applied to the respective filter weights, μ_0 – adaptation coefficient, γ – normalizing coefficient. Normalizing coefficient value is determined from the relationship [3]:

$$\gamma = \frac{1}{\chi + \|\mathbf{x}(n)\|^2} \tag{5}$$

where: $\|\boldsymbol{x}(n)\|^2$ – squared norm of the vector of L samples of a signal fed into the filter weights, χ – small constant (it is often assumed that $\chi = 0.01$).

In practice, the value of the squared signal norm is calculated basing on the relationship [3]:

$$\|\mathbf{x}(n)\|^2 = \mathbf{x}(n)\mathbf{x}^T(n) \tag{6}$$

Unfortunately, this method is time-consuming and requires the use of many resources when applied in hardware. Considerable hardware resources are also required to perform the division operation that appears in the equation (5).

In order to simplify the selected method for updating the filter coefficients, two modifications of the NLMS algorithm were proposed. The first of them - RP-NLMS takes into account the fact that in theoretical considerations it is often assumed that:

$$\|\mathbf{x}(n)\|^2 \cong L\sigma_X^2 \tag{7}$$

The signal variation σ_X^2 at time instant n can be replaced by its estimate determined recursively [1]:

$$\tilde{\sigma}_X^2(n) = \alpha\tilde{\sigma}_X^2(n-1) + (1-\alpha)x^2(n) \tag{8}$$

Which for the signal with zero mean value is equivalent to:

$$\tilde{P}(n) = \alpha\tilde{P}(n-1) + (1-\alpha)x^2(n) \tag{9}$$

It was suggested that the filtration coefficient α can be determined from the equation which relates its value to the filter order and the scaling factor g accelerating filter response rate for temporary signal amplitude changes [1]:

$$\alpha = \exp(-g/L), \qquad g = 0, 1, 2, \ldots \tag{10}$$

The use of the presented method considerably simplifies the procedure for finding the value of $\|\boldsymbol{x}(n)\|^2$, since it requires only 4 multiplication operations and one addition (instead of L multiplications and L-1 additions required previously).

In the search for further simplification of the algorithm, it was noticed that the decrease in the precision of γ factor value determination does not cause significant deterioration in adaptive filter properties. As a result, it is possible to introduce the modification in the algorithm which consists in quantizing the values of this factor. In order to do it, the subsequent values of γ were determined for certain values of $\|\boldsymbol{x}(n)\|^2$ (in the range from 2^{-7} to 2^{12}) basing on the equation (5) and they were stored in the table. The χ constant could be neglected, thanks to assuming that $\gamma \leq \gamma_{max}$. The choice of the table element used in a given algorithm iteration is made basing on the current value $\tilde{P}(n)$. Here, it is worth to mention that for a given value of μ_0 it is possible to store the whole expression $2\mu_0 / \|\boldsymbol{x}(n)\|^2$ in the table. In the case, when the values of μ_0 and γ are powers of 2, the multiplication by these factors can be replaced in hardware implementation by a bitwise shift operation. This algorithm has been called T-NLMS.

5 APPLICATION OF SUBBAND ADAPTIVE FILTER FOR NOISE REDUCTION IN TURBOGENERATOR VIBRATION SIGNAL

In the first step, the study of proposed method effectiveness was conducted with the use of the modeled signal, which was generated in Matlab (Figure 4a). The performed tests made it possible to verify the correctness of filter operation as well as to determine its properties and behavior in the stationary environment. The stationarity of the test signal allowed finding the maximum amplification error and the time of MSE (Mean Squared Error) minimization. The filtration efficiency was evaluated basing on the value of the SNR improvement factor:

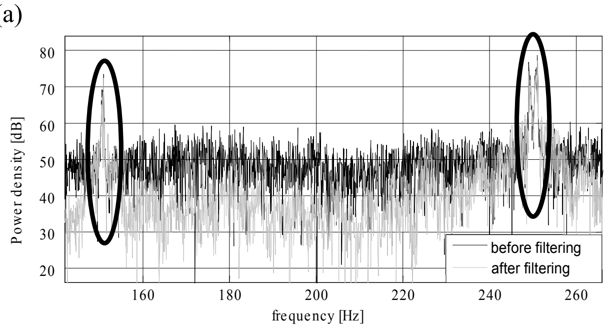
$$SNG = \frac{SNR_{out}}{SNR_{in}} = 10\log\frac{P_{vin}}{P_{vout}}[dB] \tag{11}$$

where: $P_{V\ in}$ – power of noise at the filter input, $P_{V\,out}$ – power of noise at the filter output.

For the exemplar filter configuration (number of frequency subbands: 16, filter order: 128; algorithm: T-NLMS; delay: Δ = 67; adaptation coefficient: μ_0 = 0.125; scaling factor: g = 16; analysis and synthesis filters' attenuation in stopband: 70 dB), the mean ability of noise stochastic component reduction equal to 11 dB was achieved [1]. During tests with the modeled vibration signal, due to its known parameters and stationarity, it was possible to determine the maximum filter amplification error (0.1dB for the exemplar configuration) and time of adaptation to the optimal solution (2080 iterations).

The next step was to verify the method usefulness for the reduction of the level of noise present in a real vibration signal of turbogenerator stator core. For the described filter, the mean SNR improvement of the order of 10 dB was achieved. The effect of filter operation on vibration spectrum is shown in the below figures. The first one additionally presents the circuit usefulness for filtration of complex signals containing periodic components which are not harmonics of the fundamental frequency. The second figure shows the capability of the filter (with large number of coefficients) to detect harmonic components masked by a noise.

(a)



(b)

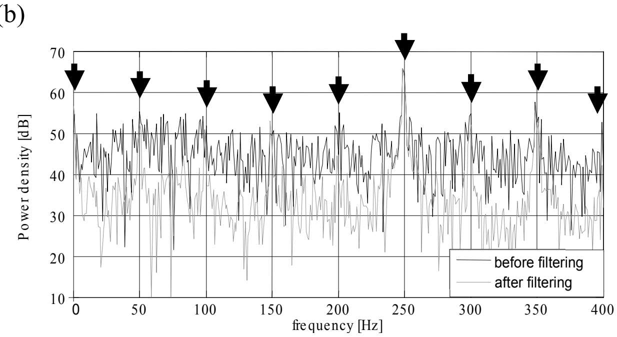


Figure 4: Effect of signal processing using 128-order filter for denoising of components other than 50 Hz harmonics (a) and using 1024-order filter, which enables the detection of low SNR components (b).

The final step of the studies was to implement the denoising filter part, which is responsible for the adaptive noise level reduction in one of the subbands, in FPGA device [1]. The basis for the hardware implementation of this filter was also the T-NLMS algorithm. The filter parameters were the same as for the software implementation. The result of the circuit operation is presented in the Figure 5. It confirms the possibility for hardware implementation of a fully functional adaptive subband filter that reduces the level of noise in periodic vibration signal which is recorded with the use of MEMS accelerometers.

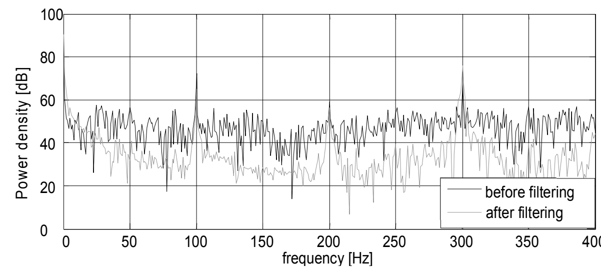


Figure 5: Exemplary result of the hardware-implemented adaptive noise filter operation

6 SUMMARY

The paper describes adaptive method for the reduction of the level of stochastic interference present in the turbogenerator vibration signal that is recorded with the use of micromachined accelerometers. The method applies the Subband Adaptive Line Enhancer circuit. The conducted research indicates the high effectiveness, parametrizability and the possibility for a relatively simple hardware implementation of the method. The proposed method can be employed in rotating machine diagnostic systems that apply harmonic analysis of the vibration signal. The implementation of the method enables the use of MEMS sensors in these systems.

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