Application of digital filters for measurement of nonlinear distortions in loudspeakers using Wolf's method

Rafal Siczek

Wroclaw University of Technology, Wybrzeze Wyspianskiego 27, 50-370 Wroclaw, Poland
rafal.siczek@pwr.wroc.pl

The designing process of digital filters used for measurement of nonlinear distortions in loudspeakers by the Wolf's method is presented in the paper. The Wolf's method has been developed in 1953 and originally the analog filters was applied. This method requires particularly steep slopes of both band-elimination filters (at the transmitting part of the measurement system) and band-pass filters (at the receiving part). The digital filters allows for achievement of such steep slopes with a relatively small cost in comparison with analog ones. The very high attenuation in the barrage bands can be achieved. In the very narrow transition band any irregularities do not appear. The design procedures using MATLAB package have been described. Then, the digital filters allow for achievement of a high accuracy of measurement.

1 Introduction

The problem of digital filters designing with very steep slope in the rejection band is presented in this paper. These filter are used for measurement of nonlinear distortion in loudspeakers. The method has been developed by V.Wolf in 1953 [1]. Author in his original work used the analog filters. Next contribution for this method was given by S. Nikaido in 1973 [2], but in his work also analog filters were applied. The application of digital filters for measurement of nonlinear distortion in loudspeaker are published by Imaka and Ohga [3].

2 Method of measurement

The metod of measurement of nonlinear distortions using broad-band signal was proposed by V.Wolf in 1953. A block diagram of the measuring system is shown in Fig.1

To the input of measuring system the broadband signal (e.g. white noise) is given. This signal is filtered with the band-elimination filter (in our case 5/9 octave). This signal after amplification excites tested loudspeaker. The acoustical signal radiated by the loudspeaker is transduced into the electrical one by the measuring microphone. This signal is filtered with the band-pass filter with the same central frequency as the band-elimination filter at the input, but with slightly narrower bandwidth (in our case 1/3 octave). Because the loudspeaker is a nonlinear system, in analyzed frequency band a product of nonlinearity occurs. However, in this band occurs also the residual signal, caused by finite slope of both filters. This slope should be sufficiently steep, in order to keep residual signal level below the product of nonlinear distortion. If, instead of single pair of band-elimination and band-pass filter, the bank of filter pairs with different central frequencies is used, we can obtain the dependence of product of nonlinear distortion on frequency.
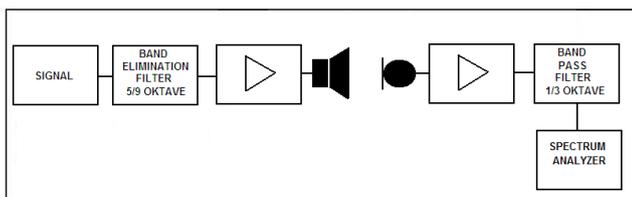

Fig.1 Principle of measurement

The advantage of this method of measurement of nonlinear distortions in comparison with classical methods i.e. harmonic or intermodulation distortion is better correlation with subjective evaluation of nonlinear distortion, because the spectral structure of broadband signal is similar to the structure of actual programs transmitted by loudspeakers. The disadvantage is complication connected with required very steep slope of used filters. If they are insufficient, the residual noise dominates over the product of nonlinearity.

3 Digital filters

3.1 Introduction

The ideal solution for measurement of nonlinear distortion using Wolf's method seems to be the application of digital filters. Nowadays, digital filters became more popular in comparison with the analog ones because of their properties. In relatively easy way the very steep slopes can be obtained. Also the signal attenuation in the attenuation band is very high. The transition band without ripples can be obtained. Application of digital filters in measuring system of nonlinear distortion should provide an increase of the accuracy of the measurement.

The Butterworth FIR filters have been chosen. They have flat frequency response in pass band, however the falling down of the frequency response at the end of this band is higher than other filters (e.g. Chebyshev filters). Also the steep of slope in the transition band is smaller than for Chebyshev filters.

The filters have been designed using Filter Design&Analysis Tool, which is included into MATLAB system.

3.2 Assumptions

The designed filters are Butterworth FIR filters. The attenuation band in band-elimination filter is equal to 5/9 octave, and pass band in band-pass filter is equal to 1/3 octave. At each end of the analysed band the reserve is equal to 1/9 octave. This reserve is necessary in order to provide maximum attenuation of residual signal during the measuring process. The slope of both filters is higher, when this reserve is narrower. The order of both filters has been calculated in such a manner that the energy of the residual signal was below 0.1% of the input signal in analyzed frequency band.

The calculated minimal values of orders of the filters are 4884 for band-elimination filter and 4242 for band-pass filter.

3.3 Calculation of the slopes of designed filters

As an example, the design of both filters: band-elimination and band-pass with common center frequency f_0=315Hz is presented.

The equations for calculation of -3dB limiting frequencies for the band-elimination filter with bandwidth of attenuation equal to 5/9 octave are:

$$\frac{f_2}{f_1} = \sqrt[9]{2^5} \qquad (1)$$

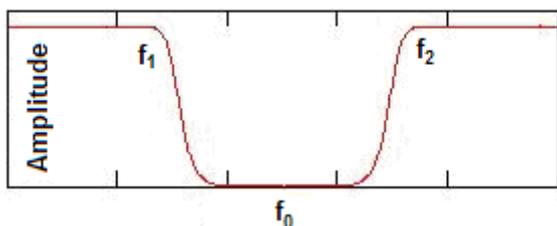$$f_1 \cdot f_2 = f_0^{\,2} \qquad (2)$$

This yields f_1=260Hz, f_2=382Hz

Fig.2 Band-elimination filter

Similarly, the equations for calculation of -3dB limiting frequencies for band-pass filter with bandwidth equal to 1/3 octave:

$$\frac{f_2^{'}}{f_1^{'}} = \sqrt[3]{2} \qquad (3)$$

$$f_1^{'} \cdot f_2^{'} = f_0^{\,2} \qquad (4)$$

This yields $f_1^{'}$=280Hz, $f_2^{'}$=354Hz
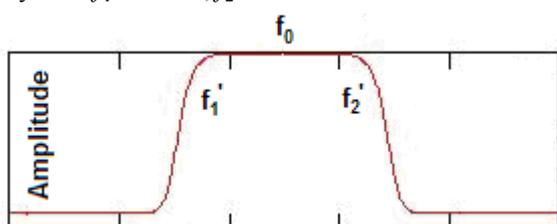
Fig.3 Band-pass filter

3.4 Design of the band-elimination filter

The frequency response of the filter with center frequency f_0=315 Hz is presented in Fig. 4.
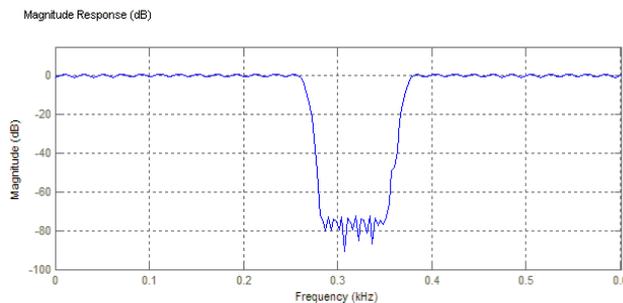

Fig.4 Frequency response of the designer band-elimination filters

3.5 Design of the band-pass filter

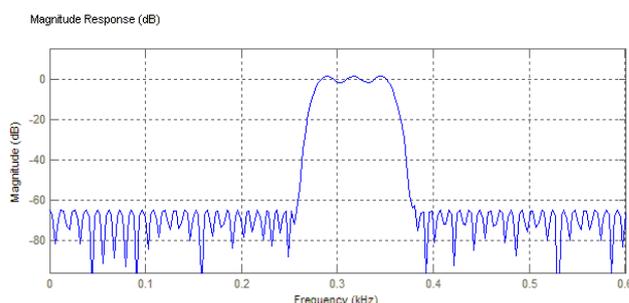The similar calculation as presented in 3.4 was done for the band pass-filter

Fig.5 Frequency response of the designer band-elimination filters

3.6 Verification of correct operations of the designed filters

The verification of the correct operations of the designed filters has been made using the white noise as the input signal. The measuring setup is presented at Fig. 6.
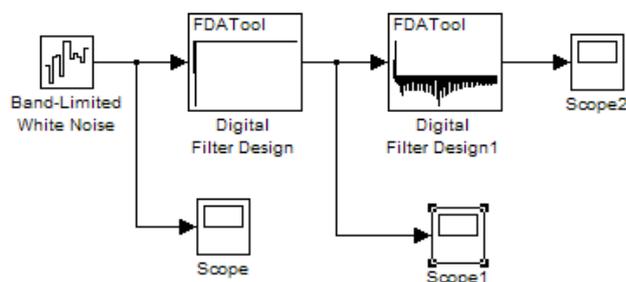

Fig.6 Measuring setup for verification of the correct operations of filters

The input signal (white noise) id presented in Fig.7

It is RMS value is approximately equal to 30 units. This signal after filtering by band-elimination filter is presented in Fig.8. Because the eliminated frequency band is relatively narrow the RMS value did not significantly change. The signal filtered by band-pass filter is presented in Fig.9. The RMS value is approximately equal to 0.03 unit, it means that it is 0.1% of input signal.
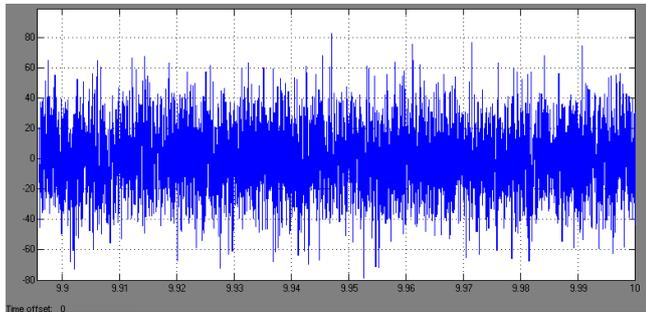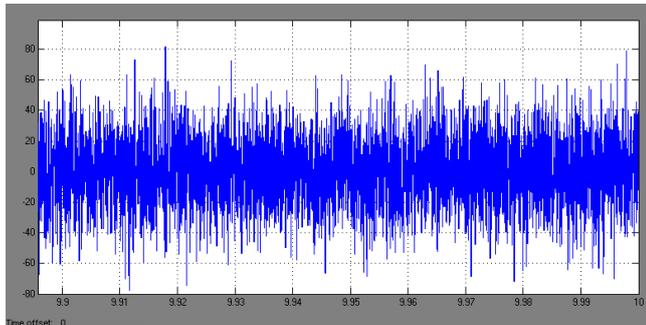
Fig.7 Input measuring signal – white noise

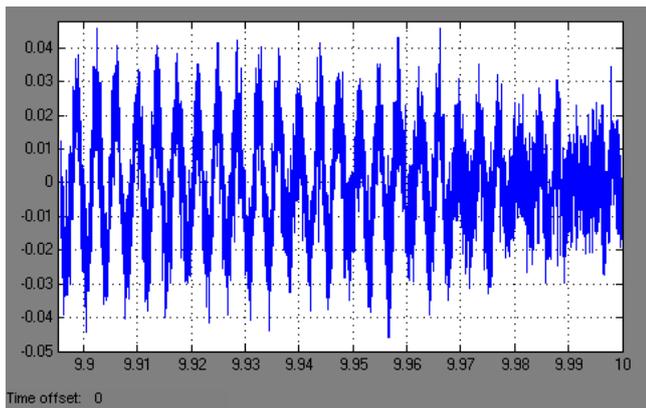Fig.8 Input measuring signal after filtering by the band-elimination filter

Fig.9 The residual signal at the output of band-pass filter

4 Simulation of the measurement of nonlinear distortions using Simulink

The simulation of measured process has been done using SIMULINK software. SIMULINK is a tool with own graphic interface, which uses MATLAB for computing process. It is applied for versatile analyses of dynamical systems.

The measuring block diagram is presented in Figs. 10 and 11. The input signal is filtered by the designed band-stop filter. The filtered noise is given to the nonlinear block, which simulates loudspeaker nonlinearity. In the next step, the signal is filtered by the designed band-pass filter. At each step of signal processing, the result is observed using Scope block.

As the input signal the white noise presented in Fig. 7 has been applied.
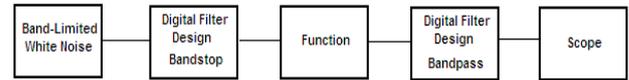

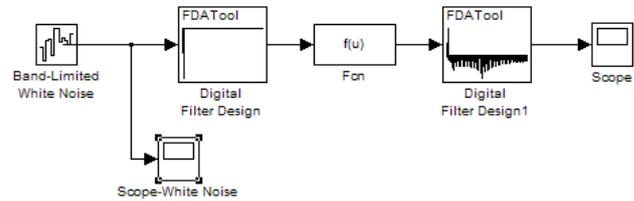Fig.10 General block diagram of the measuring system

Fig.11 Block diagram of the measuring system –Simulink

The results of simulations are presented below. All nonlinearities was considered as static. They are nonlinearities of third order what is typical for loudspeakers. First three examples are symmetrical nonlinearities weak, medium and strong respectively. The fourth example is a medium nonlinearity with asymmetry.

4.1 Example No. 1

Inserted nonlinearity is given by the following equation:

$$f(u) = u + \frac{u^3}{30000} \qquad (5)$$

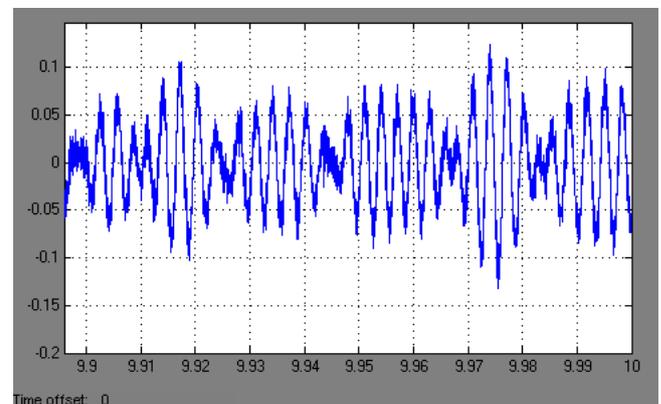The output signal is presented in Fig.12

Fig.12 The signal at the output of band-pass filter (example No. 1)

Although the nonlinearity is weak, the level of the signal is significantly higher than the level of residual signal. The difference is small, then this weak nonlinearity is lower limit of nonlinearity possible to measurement.

4.2 Example No. 2

Inserted nonlinearity is given by the following equation:

$$f(u) = u + \frac{u^3}{3000} \qquad (6)$$
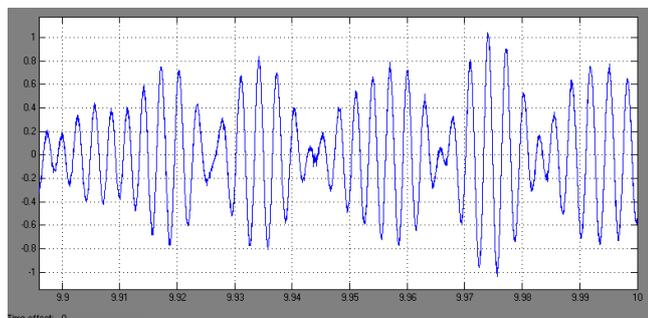
The output signal is presented in Fig.13

Fig.13 The signal at the output of band-pass filter (example No. 2)

This nonlinearity is easy for measurement.

4.3   Example No. 3

Inserted nonlinearity is given by the following equation:

$$f(u) = u + \frac{u^3}{300} \tag{7}$$
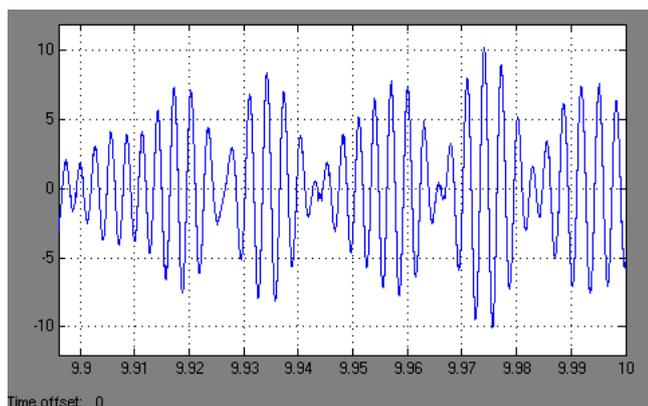
The output signal is presented in Fig.14

Fig.14 The signal at the output of band-pass filter (example No. 3)

This nonlinearity is very strong and signal level in the analyzed band is almost the same as the exciting signal.

4.4   Example No. 4

Inserted nonlinearity is given by the following equation:

$$f(u) = u + \frac{u^2}{100} + \frac{u^3}{3000} \tag{8}$$

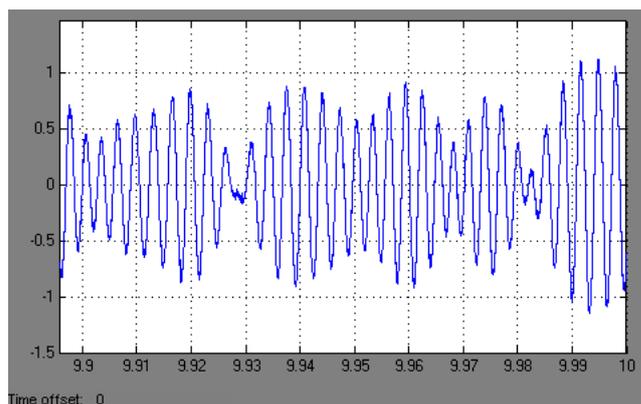The output signal is presented in Fig.15

Fig.15 The signal at the output of band-pass filter (example No. 4)

Comparing the Figures 13 and 15 it can be observed that asymmetry causes an increase of the nonlinearity product.

5   Conclusion

The method of measurement of nonlinear distortions in loudspeakers using digital filters has been presented. Using MATLAB software packet developed by MathWorks it is possible to design necessary filters as well as to simulate the measuring process.

References

[1] V. Wolf, "Dynamic Method of Nonlinear Distortion Investigation" (in Russian), *Radiotechnika.* vol. 8, no. 2 pp. 27 – 37, 1953.

[2] S. Nikado, "On Distortion Measurement of Loudspeakers Using Programmed Sound" *Reports of the 6th International Congress on Acoustics,* vol. III D, Electroacoustics and Architectural Acoustics, Tokyo, pp. 21-28 1968.

[3] K. Imaoka and J. Ohga, "A New Digital Measurement for Distortion of Acoustical Devices", 121st AES Convention, October 2004, preprint 6216.