Impact of changes in the parameters of digital filters for loudspeaker nonlinear distortion using the broadband noise method

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Abstract

This article presents the continuation of tests on the measurements of the product of nonlinear distortion in dynamic loudspeakers with the use of digital filters. As demonstrated in the recent measurement sand the analysis of the results obtained on the tests on loudspeaker nonlinear distortions, the parameters of digital filters used in the broadband noise method affect significantly the obtained results. This research presents results of the tests with using Chebyshev's bandstop and bandpass filters of types I and II. These filters significantly improve the quality of the measurements of nonlinear distortions in a dynamic loudspeaker by the broadband noise method.

Introduction

A system modeled using a mathematical description can be used in studies on parameters of electronic devices without using actual devices. A mathematical model is used very often as a substitute of a real device to be tested, which lowers significantly costs of measurements. Similarly to other electronic models, also a loudspeaker can be described using equations describing its performance and parameters. The mathematical model of a dynamic loudspeaker is based on its equivalent circuit.

As demonstrated in the recent measurements [1][2][3][4][5] and the analysis of the results obtained on the tests on loudspeaker nonlinear distortions, the parameters of digital filters used in a broadband noise method affect the results obtained significantly. Until now, there have been tests carried out mainly on the product of nonlinear distortions in a dynamic loudspeaker with the use of Butterworth's filters and measurements performed with the use of Chebyshev's filters constitute a minor part of these tests.

The research of nonlinear distortions will be divided in two parts: simulation and measuring. The simulation part will consist in the making of measurements and initial simulations in order to investigate the appropriateness of the measurement method used. To this end certain mathematical models of loudspeakers will be used. The measurement part will consist in making the measurements of real loudspeaker.

Measurement method

The broadband signal is given to the input of measuring system. This signal is filtered with the band-elimination filter. In our case the eliminated bandwidth is equal to 5/9 octave. This signal after amplification excites loudspeaker model or real loudspeaker under test. The signal from the output of the nonlinear block is given to the bandpass 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). In analyzed frequency band only the product of nonlinearity should occur [6][7].

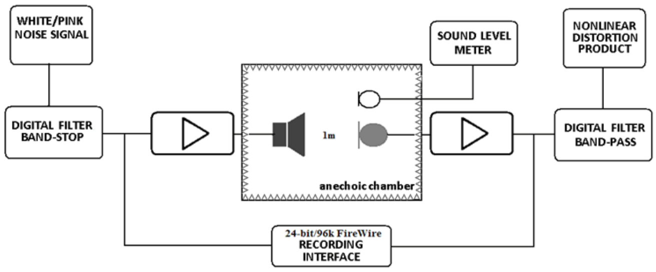

Figure 1: Method of measurement with a real loudspeaker

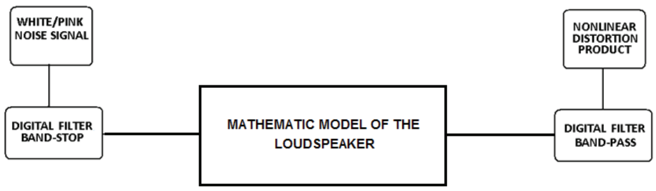

Figure 2: Method of measurement with a loudspeaker model

Digital filters

Digital filters designed for the purpose of this work need to comply with several basic guidelines resulting from the nature of nonlinear distortions and the used measurement method, that is:

- the digital filter on the transmitting side of the system needs a slightly broader band than the bandpass filter on the receiving side, which results from the need to compensate for the influence of residual noise
- the digital filters should be designed in 1/3 octave bands, within the frequency gap of 20 Hz – 5 kHz, for the purpose of measurement and modelling
- the system's sampling frequency is 44.1 kHz – this is linked with the frequency of sampling the sound card used for measurement and the same frequency setting in the entire measurement system
- the filters should be implemented in a program, which allows for full compatibility with SIMULINK/MATLAB software and generating filter codes in C/C++ language as well as Code Composer Studio - this makes implementing the designed digital filters on signal processors significantly easier.

Filters with unlimited impulse response were used for the purpose of the work, after prior research. They are a lot faster and less computationally complex than filters with limited impulse response.

The bandstop filter at the transmitting side should have broader bandwidth than the bandpass filter at the receiving side because of finite slope of both filters which causes the appearance the residual noise in the measuring band.

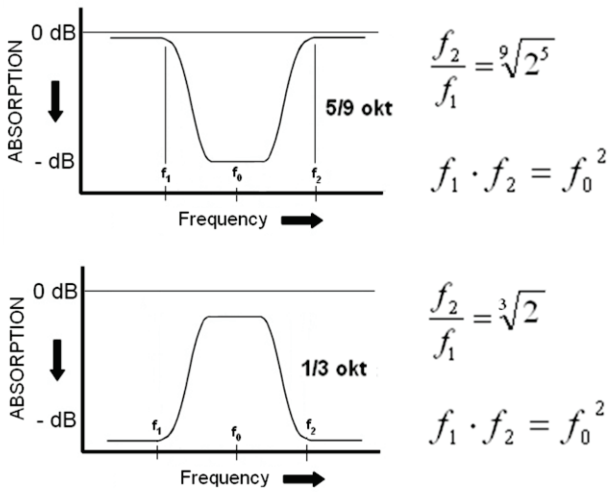

Figure 3: Digital filters bandwidth

The figure below shows an example of a digital filter designed in FDATool

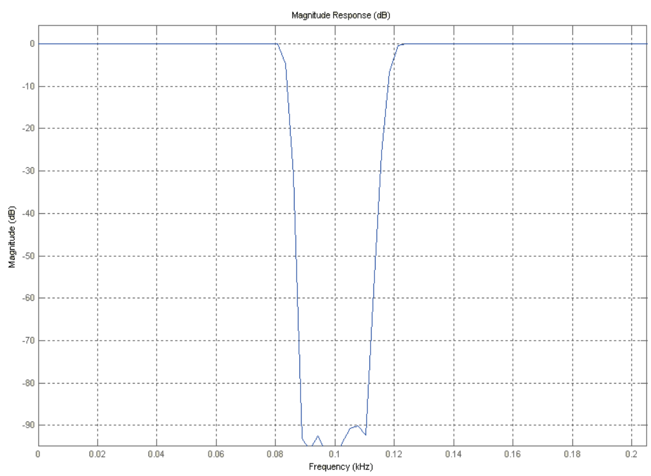

Figure 4: Chebyshev digital filter – 100Hz

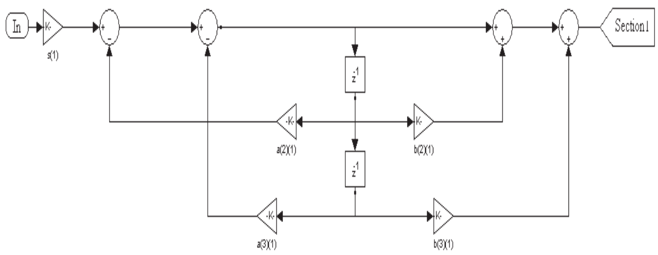

Figure 5: Chebyshev filter – type II – first section

Dynamic loudspeaker models

A dynamic loudspeaker is a strongly nonlinear device. A way how it works and its construction generate in many places significant nonlinear distortions in the processed band. We can distinguish a nonlinearity caused by the suspension stiffness, nonlinearity connected with the force factor *Bl*, and nonlinearity of the voice-coil inductance L_E. On the basis of equivalent circuit and mathematical equations three loudspeaker models have been developed in the Matlab/Simulink software.

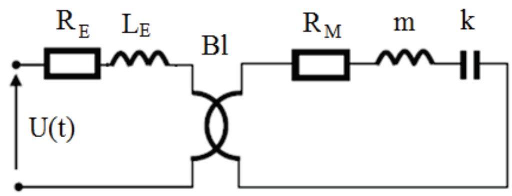

Figure 6: Loudspeaker - equivalent circuit

Mathematic model of the loudspeaker - the voice-coil inductance

The loudspeaker is characterized with following system of differential equations (1)(2)(3), the dependence of the voice-coil inductance is modeled by the linear function (4). The voice coil inductance depends nonlinearly not only on displacement but also on electric current. This mathematical model of the dynamic loudspeaker is implemented in the Matlab/Simulink software.

$$L_{E0}(1+ax)(1+3bi^2)\frac{di}{dt} = E(t) - R_E i \tag{1}$$

$$\frac{dx}{dt} = v \tag{2}$$

$$\frac{dv}{dt} = -\frac{R_M}{m_{ms}}v - \frac{K_{ms}}{m_{ms}}x + \frac{Bl}{m_{ms}}i + \frac{L_{E0}a}{2m_{ms}}i \tag{3}$$

,where:

$$L_E(x,i) = L_{E0}(1+ax)(1+3bi^2) \tag{4}$$

Mathematic model of the loudspeaker - the suspension stiffness and Bl

In this model we can distinguish a nonlinearity caused by the suspension stiffness and nonlinearity connected with the force factor *Bl.*

$$R_E i_z + Bl\frac{dx}{dt} = E(t) \tag{5}$$

$$m_{ms}\frac{d^2x}{dt^2} + R_M\frac{dx}{dt} + K_{ms}x = Bli_z \tag{6}$$

are suspension stiffness and force factor, respectively. Parameters μ_1 and μ_2 are called coefficients of nonlinearity and x_{01} and x_{02} are asymmetries of nonlinearities of *Bl* and

K respectively.

$$K_{ms} = K_0[1 + \mu_2 (x - x_{02})^2] \qquad (7)$$

$$Bl = Bl_0[1 - \mu_1 (x - x_{01})^2] \qquad (8)$$

are suspension stiffness and force factor, respectively. Parameters μ_1 and μ_2 are called coefficients of nonlinearity and x_{01} and x_{02} are asymmetries of nonlinearities of Bl and K respectively. The mathematical model of the dynamic loudspeaker described by equation system (1) is designed in the Simulink software as a flow-graph. The solution of the equation system is displacement of the moving coil versus time. In order to obtain the acoustic pressure the following formula has been used:

$$p_{ak} = \rho_0 \frac{S}{2\pi d} \frac{d^2 x}{dt^2} \qquad (9)$$

where: S – effective surface of the diaphragm, d - distance between loudspeaker and observation point (d = 1 m), ρ_0 - density of the air. The acoustic pressure level is computed in each 1/3-octave bandwidth. It is a measure of nonlinear distortion.

Results of simulations and measurements with using a Butterworth filters

The following graphs show examples of measurements and simulations of loudspeaker nonlinear distortions with using Butterworh filters in presented method [2][3]. We can see that with the increase of the nonlinear coefficients and input noise power the level of distortions increases.

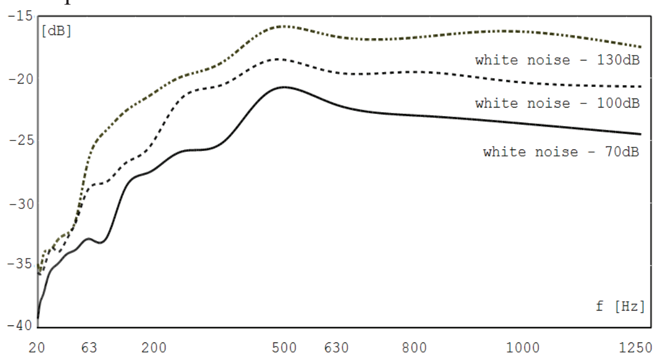

Figure 7: Nonlinear distortion product versus white noise power - measurements

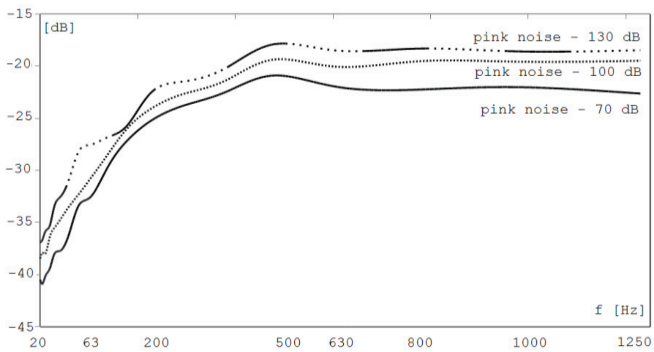

Figure 8: Nonlinear distortion product versus pink noise power - measurements

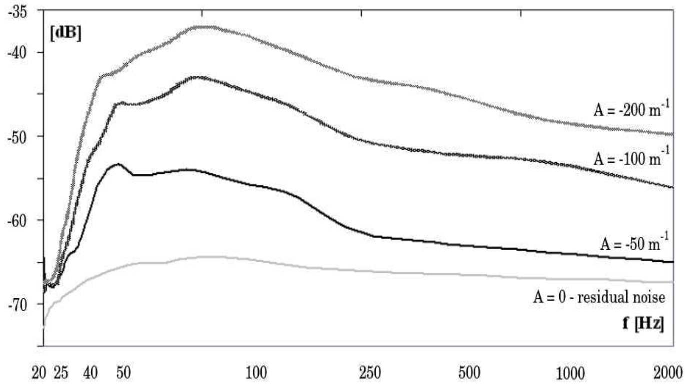

Figure 9: Nonlinear distortion product versus nonlinear *a* parameter - measurements

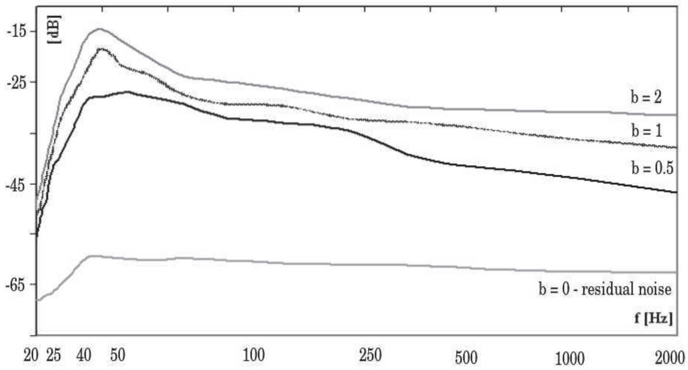

Figure 10: Nonlinear distortion product versus nonlinear *b* parameter - measurements

Butterworth ve Chebyshev

The following graphs show examples of measurements and simulations of loudspeaker nonlinear distortions with using Butterworh and Chebyshev filters in presented method. The following graphs [Figure 11 and 12] shows the difference in measurements. Nonlinear distortion product achieves lower values with using Chebyshev filters (type I and II). It follows that the use of this filters will result in more accurate measurements of nonlinear distortion.

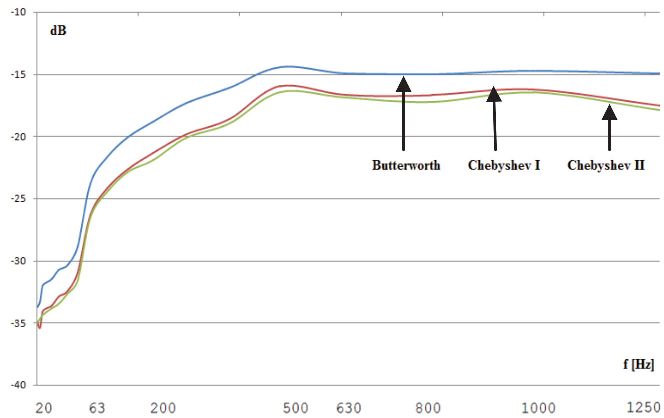

Figure 11: Chebyshev vs Butterworth filters – simulation process

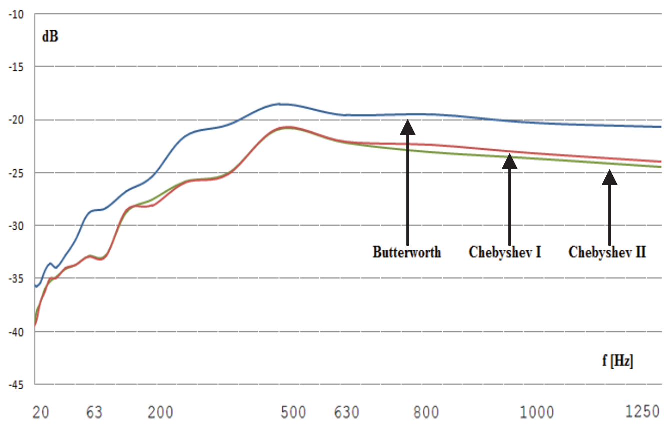

Figure 12: Chebyshev vs Butterworth filters – measurement process

Conclusion

The broad-band noise method of measurements of nonlinear distortions in loudspeakers using digital filters has been presented. The results of simulation of nonlinear distortion caused by force factor and suspension stiffness and results of nonlinear distortion caused by nonlinear voice-coil inductance are presented. We can see the difference in the measurements using the Butterworth and Chebyshev filters.

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