Measurement, modelling and compensation of nonlinearities in hearing aid receivers

Timm Schaer, Stephan D. Ewert, Jörn Anemüller, Birger Kollmeier
Universitaet Oldenburg, Medizinische Physik, 26111 Oldenburg, Germany, Email: timm.schaer@uni-oldenburg.de

Introduction

Hearing aid receivers operate over a large dynamic range and reach output levels of up to 131 dB. Electromagnetic receivers are commonly used in hearing aids since they offer a very high efficiency. Receivers are typically characterized by their linear transfer function. These linear distortions are easily measured with standard methods and if compensation is applied it is often limited to the magnitude transfer function. The current study addresses a more complete characterization of the hearing aid receiver by measurement of the nonlinear transfer function. Nonlinear distortions of the receiver are output level dependent and might negatively influence measures like speech intelligibility of an aided hearing impaired person, particularly at high levels. Moreover, nonlinear distortions are problematic in the context of future "high-fidelity" hearing aids and compensation methods are of interest. Here, a fast and efficient sine-sweep method proposed by Farina [2] was employed to estimate the linear and nonlinear transfer function. The measurement was repeated at output levels ranging from 31 dB SPL to 131 dB SPL. From the data, nonlinear input-output functions can be constructed or alternatively, the diagonal elements of a Volterra-series based description of the nonlinear transfer function can be derived. Three nonlinear receiver models with different level complexity are suggested and tested.

Method

Nonlinear systems can not be described by the impulse response or the transfer function only. The linear theory of systems must be enhanced by nonlinear parts. The Volterra series is one possibility to describe nonlinear systems [1]:

$$y(t) = \sum_{k=0}^{K} H_k \left\{ x(t) \right\}. \tag{1}$$

The output signal $y(t)$ of the overall system is a sum of K single systems with the system function $H_k\{\cdot\}$ defined as follows:

$$H_k \left\{ x(t) \right\} = \int_0^{T_k} \cdots \int_0^{T_k} h_k(\tau_1, \ldots, \tau_k) \prod_{j=1}^{k} x(t - \tau_j) d\tau_j, \tag{2}$$

where $h_k(\tau_1, \ldots, \tau_k)$ is referred to as the Volterra kernel of order k. For practical use, the continuous form is transformed into the discrete form

$$y(n) = \sum_{m}^{M} h_1(m) x(n-m) + \sum_{m_1}^{M} \sum_{m_2}^{M} h_2(m_1, m_2) x(n-m_1) x(n-m_2) + \ldots, \tag{3}$$

where $h_1(m)$ and $h_2(m_1, m_2)$ are the first- and second-order Volterra kernel, respectively. The kernels have the dimension $\mathbb{R}^{M \times \ldots \times M}$. The first-order kernel is identical to the impulse response for linear systems.

An approximation of the system was used by Farina [2]. The kernels are reduced to the main diagonal $\tilde{h}_k(m) = \text{diag}\left\{h_k(m_1, m_2, \ldots, m_k)\right\}$ and Eqn. 3 is simplified to

$$\tilde{y}(n) = \sum_{m=1}^{M} \tilde{h}_1(m) x(n-m) + \sum_{m=1}^{M} \tilde{h}_2(m) x^2(n-m) + \ldots + \sum_{m=1}^{M} \tilde{h}_K(m) x^K(n-m). \tag{4}$$

This system is like an enhanced Hammerstein model of higher order [4]. In the classical Hammerstein model, the signal $x(n)$ is raised to power two and filtered with an impulse response. In the present case, the signal $x(n)$ is raised to the power of K and filtered with the respective impulse response h_K. The K different parts are summed up and form the output signal $y(n)$.

The impulse responses h_K can be measured by exciting the nonlinear system with an exponential sweep signal

$$x(t) = \sin\left(\frac{\omega_1 T}{\ln\left(\frac{\omega_2}{\omega_1}\right)} \left(e^{\left(\frac{t}{T} \ln\left(\frac{\omega_1}{\omega_2}\right)\right)} - 1 \right) \right), \tag{5}$$

where ω_1 is the lower cutoff frequency and ω_2 is the upper cutoff frequency. The signal has the duration T. In the following description the variable frequency with time is denoted with ω_v. The output signal of a nonlinear system excited with a sinusoidal input signal contains harmonic distortions at integer multiples of the fundamental frequency ω_v. The output of the system is:

$$\hat{y}(t) = \hat{h}_1(t) * \sin(\omega_v t) + \hat{h}_2(t) * \sin(2\omega_v t) + \ldots. \tag{6}$$

The exponential sine-sweep method of Farina [2] allows to directly measure the impulse responses $\hat{h}(t)$ for the different harmonics. With algebraic conversions, the main diagonals of the Volterra kernels, $\tilde{h}(t)$ in Eqn. 4 can be derived [2].

Measurement

The measurements were carried out with the hearing aid receiver *(EJ60005-00)* form Knowles, built in the hearing aid *(Acuris P)* from Siemens Audiologische Technik. The output of the receiver was connected to the ear simulator (Brüel & Kjær, Type 4157) via the tubing. The signal in the ear simulator was measured by the microphone (Brüel & Kjær, Type 2669) and amplified by a G.R.A.S. (Type 12AA). The analog signals from the amplifier was digitalized and processed by the RME Fireface 800 and send to the computer. The playback and recording of the signals was controlled by MATLAB with SOUNDMEXPRO. A sampling frequency of 192 kHz was used for the measurement to cover a large frequency range required for high-order harmonics at high frequencies.

The sine-sweeps had a duration of 8 s and were averaged 9 times. The measurements were repeated for levels between 31 dB SPL and 131 dB SPL in steps of 1 dB.

Total Harmonic Distortions (THD)

The nonlinearity of a system can be described by the total harmonic distortion (THD), which is usually measured at a frequency of 1 kHz. From the measured impulse responses $\hat{h}_k$ of Eqn. 6, the THD can be extracted for each frequency and level. The result is plotted in Fig. 1. The distortions of the receiver are high for low levels and low frequencies. For the mid range of frequencies (200 Hz to 4 kHz) the distortions increase for levels from 110 dB. The high frequency range shows also a high degree of distortions for low levels.

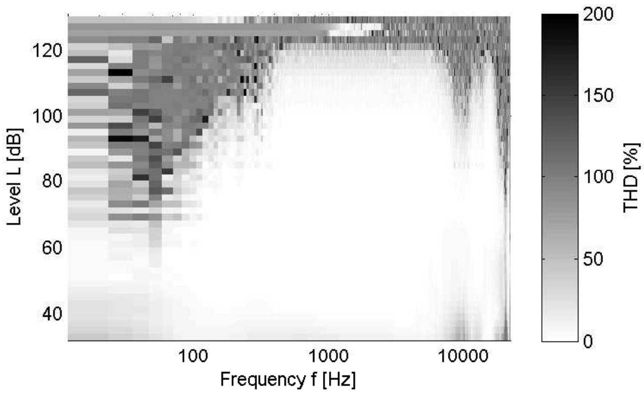
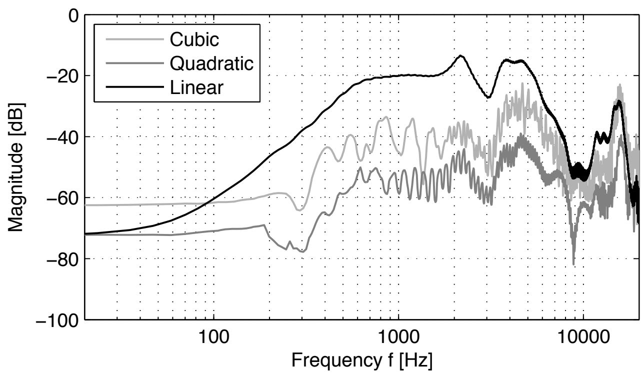


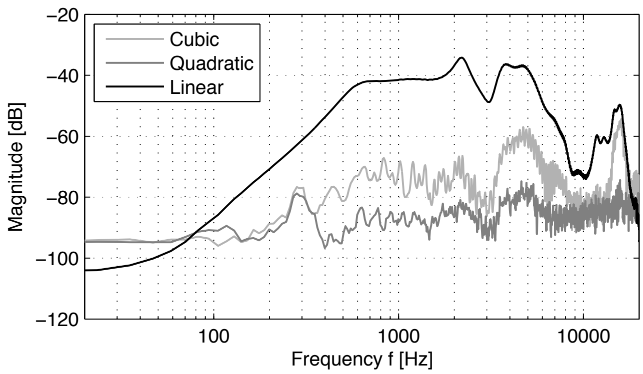
Figure 1: Total harmonic distortion of the hearing aid receiver in % plotted over the frequency f and the level L. The greyrange is from 0 % to 200 %.

Characteristics of the nonlinear transfer function of the receiver

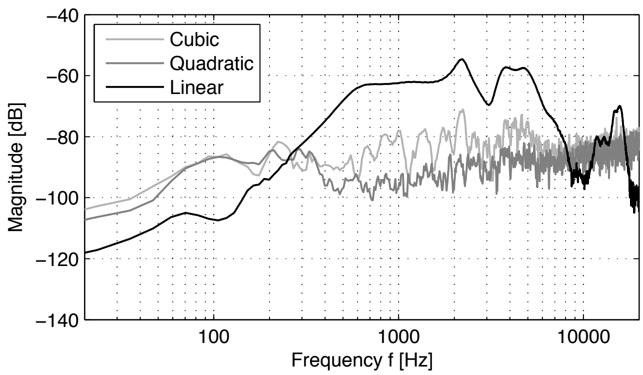
The transfer functions of the receiver were measured for the level range from 31 dB SPL to 131 dB SPL. Figures 2a)-c) show the transfer function for three levels 120 dB, 100 dB and 80 dB.



(a) Nonlinear Transfer functions of the receiver for 120 dB SPL.



(b) Nonlinear transfer function of the receiver for 100 dB SPL.



(c) Nonlinear transfer function of the receiver for 80 dB SPL.

Figure 2: Nonlinear transfer functions of the receiver for 120 dB SPL, 100 dB SPL and 80 dB SPL.

It is obvious that odd-order nonlinear transfer functions are higher than those of even order. For the two measurements at high levels, the quadratic and cubic transfer functions are similar to the linear transfer function. For the low level this structure is not visible and most likely below the measurement noise floor. Overall, the receiver has only very low nonlinear distortions in this level region.

Model

Three different approaches to model the nonlinear hearing aid receiver are suggested. The goal of this models is to describe the receiver with only a few parameters.

Model 1 - Nonlinear filtering with Look-Up-Table

This approach was suggested by Farina [2] with the nonlinear transfer function derived for only one level. The model used here (see Fig. 3) takes the measured main diagonals of the kernels up to order 3 as a Look-Up-Table for the nonlinear filtering. The signal is processed in an overlap-add scheme with a block length of $N = 8192$ (43 ms) and $\frac{N}{2}$ overlap. The level in the current block is estimated by its root-mean-square value (RMS). The transfer functions are selected depending on the current level.

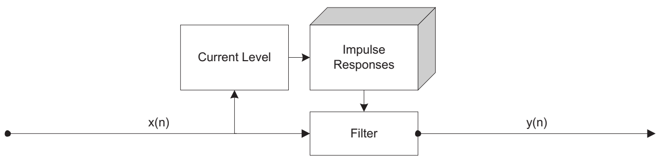


Figure 3: Model 1 - Block diagram of the model with nonlinear filtering with Look-Up-Table.

Model 2 - Hammerstein model with linear filtering and compression

In this approach shown in Fig. 4a), the system is described by a linear filter with subsequent instantaneous compression. It is known as the Hammerstein model [4]. The linear impulse response for the filter is the measured linear impulse response. The compression characteristic is taken from the input-output characteristic of the levels L at 1 kHz (see Fig. 4b)). For low levels, the hearing aid receiver shows a small expansion and for high levels a saturating compression.

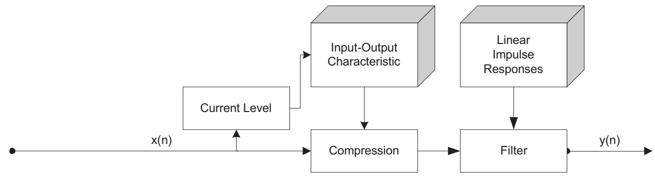
Model 3 - Nonlinear parallel filtering

Figure 5a) shows the third approach where the signal $x(n)$ is filtered in parallel with three impulse responses up to order 3 for three different levels of 80 dB, 100 dB and 130 dB. The three filtered signals are weighted depending on the current level with the weighting function shown in Fig. 5b) and added up to the output signal $y(n)$.

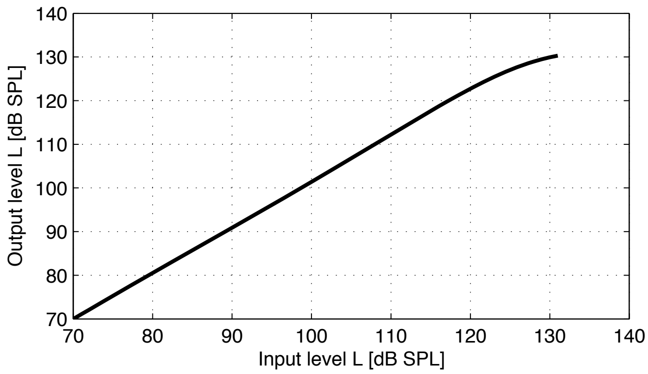
Model analysis

The three suggested models in the complexity of the required measurements of the receiver and the computational load required for the model simulations. The models were tested with the sine-sweep method by Farina and the simulation results were compared to the measurements.

The simulation results for model 1 are shown in the upper panel of Fig. 6a). In this case, the receiver of the hearing aid has to be measured for all possible levels and the computing load is quite high due to the block-based filtering. The states of the filters used for a preceding block do not match necessarily the impulse



(a) Block diagram of the Hammerstein model with linear filtering and compression.



(b) Input-output characteristic of the receiver for $f = 1$ kHz.

Figure 4: Model 2 - Hammerstein model with linear filtering and compression.

response in the current block if the level has changed. This introduces artefacts to the simulated output. Block-based processing causes a time delay in the output signal.

Model 2 Fig. 6b) uses only one linear impulse response but requires the measurement of the input-output characteristic of the receiver for all possible levels. The computing load is small due to linear filtering and compression afterwards. This model has no time delay.

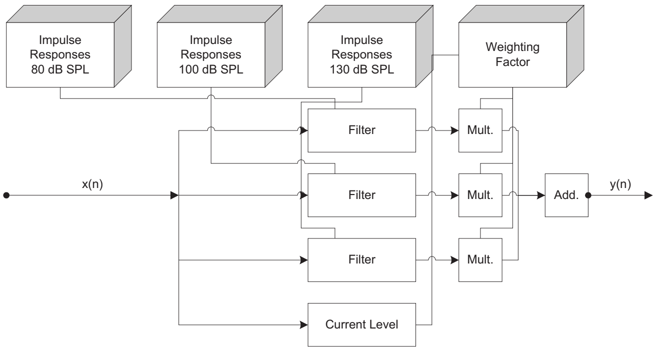
Model 3 Fig. 6c) uses only three nonlinear impulse responses for three different levels. To select the appropriate level regions, the measurement of the input-output characteristic is required. The signal is filtered three times and summed up with a weighting function, which causes moderate computational load. Phase shifts between the different impulse responses could cause cancellations in the output signal. The model has no time delay.

Conclusion

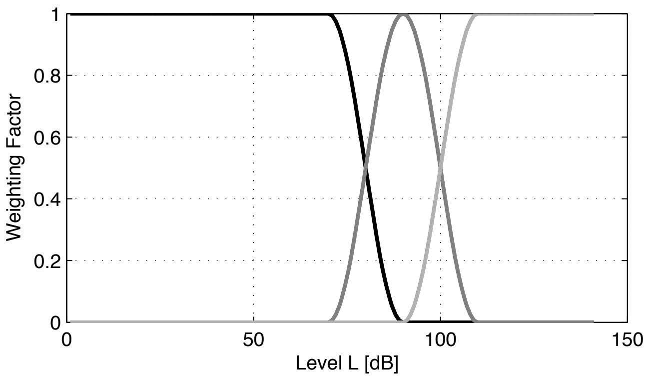
In this work it has been shown that the hearing aid receiver under test is a nonlinear system. The nonlinearity occurs mainly for levels higher than 100 dB. The sine-sweep method by Farina to measure the nonlinearity of a system is very powerful and easy to implement. The measurement of a receiver for a large level range can, however, take up to 3 h. The three suggested models require a different number of measurements and have different computational loads.

Acknowledgements

This work was supported by the Bundesministerium für Bildung und Forschung (BMBF) project "Modellbasierte Hörsysteme".



(a) Block diagram of the model with nonlinear parallel filtering.



(b) Weighting factor for three different level domains.

Figure 5: Model 3 - Nonlinear parallel filtering.

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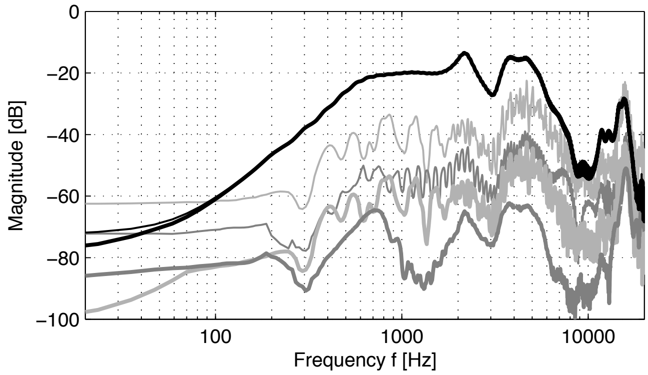
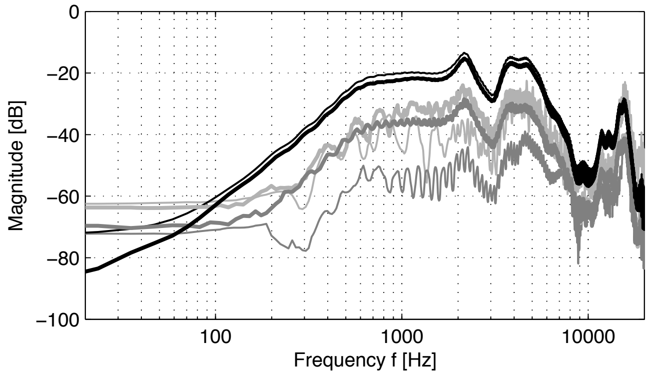
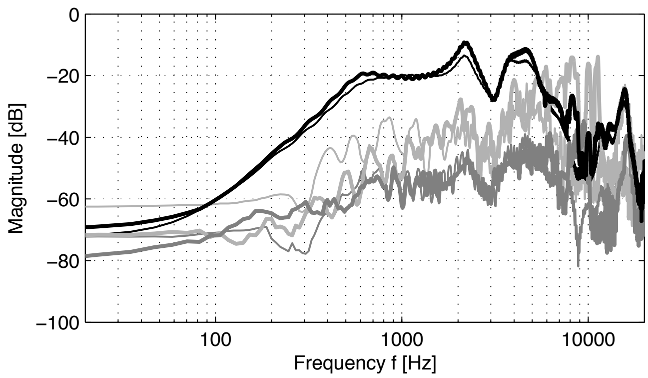


(a) Transfer function of the receiver for 120 dB SPL (thin lines) in comparison with the simulation of the model 1 - Nonlinear filtering with Look-Up-Table.



(b) Transfer function of the receiver for 120 dB SPL (thin lines) in comparison with the simulation of the model 2 - Hammerstein model with linear filtering and compression.



(c) Transfer function of the receiver for 120 dB SPL (thin lines) in comparison with the simulation of the model 3 - Nonlinear parallel filtering.

Figure 6: The comparison of the measurement with the three different models is show for the level 120 dB SPL.