PROCEEDINGS of the
23rd International Congress on Acoustics

9 to 13 September 2019 in Aachen, Germany

Nonlinearities in sound field control systems

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ABSTRACT

If a loudspeaker is driven at high levels, its output is likely to be affected by nonlinearities. The result is a distortion of the loudspeaker output which depends on the input signal. When looking at a sound cancellation or sound field control system the introduction of such nonlinearities lowers the performance. This paper investigates the extent of such effects and looks at strategies to counteract the problem. The paper specifically focuses on quantifying the effect of compression on the output of the system at the input frequency, finding a model to represent the coherent distortion and counteract accordingly. A comparison of different strategies to counteract the effect of nonlinearities is presented. The results show that a compensation based on an input voltage level adapted linear response of a loudspeaker can improve a sound zoning algorithm up to 20dB. Alternative compensation methods are discussed and show the benefit of a displacement level dependent compensation at low frequencies.

Keywords: Sound Field Control, Nonlinearities

1. INTRODUCTION

Sound field control systems rely on accurate estimations of the transfer function between source and the controlled zone and often assume linear time invariance of the source. However, real loudspeakers behave nonlinear at high output levels. This leads to compression of the fundamental frequency and a rise of distortion products. In a previous paper Ma (1) shows that nonlinear distortion degrades the acoustic contrast, a performance metric in a sound zoning system. Further he states that nonlinear effects on the fundamental component appear to be the main cause for this contrast loss.

This paper investigates first approaches to compensate this effect. In contrast to more sophisticated compensation models (2,3) this work aims on developing a fast and easy-to-implement solution to the problem. The proposed approach is based on a prediction of the change in linear transfer function at different voltage and displacement levels. Specifically, the compression and decompression of the linear transfer function due to changing level will be compensated. The complexity of the subject required limitations and simplifications at several points. Nevertheless, the outcome of the paper shows promising results of an improvement of up to 20dB on a broad range of frequencies. The effect of distortion products will be touched but not discussed in detail. The following sections focus on explaining the proposed compensation method and present experimental results on a real loudspeaker.

2. METHOD

2.1 Compensation Method

The magnitude and phase response of most loudspeakers is dependent on the input signal due to its nonlinear behavior at higher input levels. The first step of the proposed compensation method is to identify those dependencies. First, we measure the transfer function between voltage and near-field sound pressure of a given loudspeaker at different input gains to characterize the nonlinearity of a loudspeaker by its apparent linear transfer function at different levels. The level dependent changes can be used to calculate an approximation of the expected change in transfer function at any other level. This expected change can be described as a function of input level using a polynomial fitting function of fourth order. This expected change is used to compute a linear compensation filter

We introduce the estimation signals $x_i(t)$ with correspondent level $L_i = L(x_i(t))$. One level is chosen as reference level L_{ref}. The level can thereby either be a RMS (root-mean-squared) voltage or RMS voice coil displacement.

To compute the compensation filters one
1. measures transfer functions $H(f, L_i)$ at levels L_i (Fig.1, left),
2. computes the change in transfer function $\Delta H(f, L_i) = H(f, L_i)/H(f, L_{ref})$ compared to the reference level (Fig.1, right),
3. approximates ΔH using a polynomial regression (4th order), to get changes for a continuum of input levels: $\Delta\widetilde{H}(f, L)$. These are the linear compensation filters in frequency domain.

To apply the compensation filters one
1. measures the test signal level $L_{test} = L(u(t))$,
2. applies the compensation filter to the test signal $y(t) = IFFT\{\Delta\widetilde{H}(f, L_{test})FFT\{u(t)\}\}$, and
3. plays back $y(t)$ instead of $u(t)$.

In the following, the signals to estimate the compensation are called estimation signals. The signals the compensation is applied to will be referred to as test signals.

Figure 1 – Left: sketch of a loudspeaker magnitude response at four different levels. Right: differences in magnitude response at a single frequency for different levels together with polynomial regression.

It has been elaborated by Klippel (4) that nonlinear behaviors of loudspeakers are dependent on different variables. These variables include current, velocity and displacement. An effective compensation strategy should be based on the level of a variable that is directly connected to the change in response. In this work, a simple approach of using the voltage (which is coupled to the current) and the displacement variable was investigated. In the following we focus on three variations of the compensation procedure:

1) *Sweep-voltage approach:* the transfer functions are measured with an exponential sweep excitation. The compensation is based on the RMS voltage level.

Measuring with sweep signals excites only one frequency at a time. This way mainly distortions at the fundamental frequency are recorded.

2) *Noise-voltage approach*: the transfer functions are measured with M-Noise[1], a noise with music like crest-factor. The compensation is based on the RMS voltage level.

In contrast to the *sweep-voltage approach,* this test signal is closer to music which allows interaction between frequencies and amplitudes when measuring the transfer function. The interpretation of the results of this approach is less obvious due to the random structure of amplitude and frequency content.

3) *Noise-displacement approach:* the transfer functions are measured with M-Noise. The compensation is based on the RMS displacement level.

Here, the calculation and application of the compensation is performed using voice coil displacement instead of the voltage level. This compensation assumes that changes in transfer function are related to displacement. In contrast to the two preceding approaches this approach applies more compensation for signals with high displacement (signals with strong low frequencies).

[1] https://m-noise.org

2.2 Performance Metrics

Sound cancellation applications are based on a primary wave cancelled by a secondary wave. A perturbation of amplitude or phase of the secondary wave, e.g. due to nonlinear distortion, directly translates to an achievable reduction of sound (ΔL_R. Fig. 2, right illustrates this ΔL_R as a function of amplitude and phase deviation (5). For no error in phase or amplitude, ΔL_R approaches negative infinity, which corresponds to total cancellation. Using the formula of ΔL_R the maximal achievable reduction of a sound cancellation application can be calculated given a certain error in transfer function. From Fig.4, already slight deviations in level and phase compromise a given sound cancellation application. Assuming a constant phase error of ±2.5° and a magnitude error of +1dB a given sound cancellation application could only achieve an attenuation of primary sound of around -17dB. A reduction of this errors to ±0.5°and +0.2dB improves the performance to -32dB.

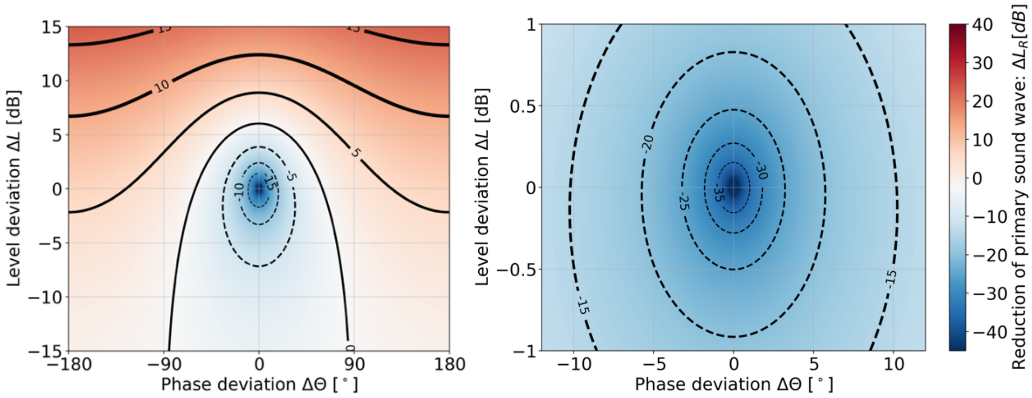

Figure 2 – Reduction in amplitude of a primary sound wave as a function of the difference of amplitude and phase between primary and secondary sound wave

3. Results

To test the discussed compensation strategies measurements were made using a 10-inch closed box subwoofer. This section first presents measurements of the variation of transfer function for changing level of this loudspeaker. Next, the compensation method is applied.

3.1 Loudspeaker frequency response at different input levels

The measured transfer functions for measurement signals with different RMS voltage levels are shown in Fig. 3.

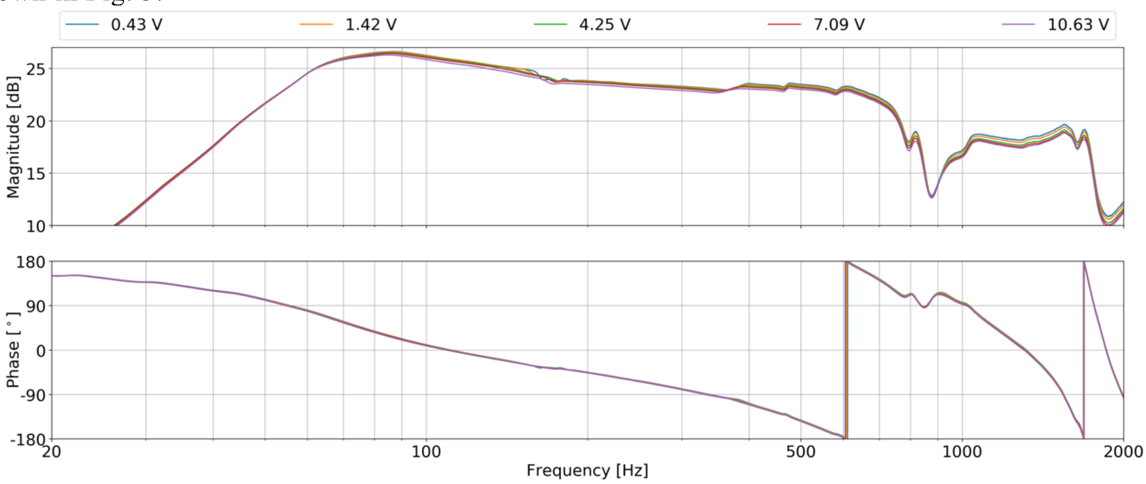

Figure 3 – 10-inch subwoofer frequency response measured in near-filed at 5 different levels with 15 second exponential sweeps.

The considered frequency range has been limited to 20Hz to 2000Hz, which is the effective range of this speaker. The amplitude response shows a rather flat curve from 70 to 700Hz. The resonance of the loudspeaker lies between 60-70 Hz which explains the drop in transfer function below that frequency (6). At higher frequencies (above 400Hz) the response shows irregularities due to the construction of the loudspeaker box. Another irregularity can be observed around 160 Hz. To identify a change in response we chose 0.43 V as a reference measurement. The change in response is the difference to the reference response. Fig. 4 shows the differences in transfer function to the reference level at 0.43V.

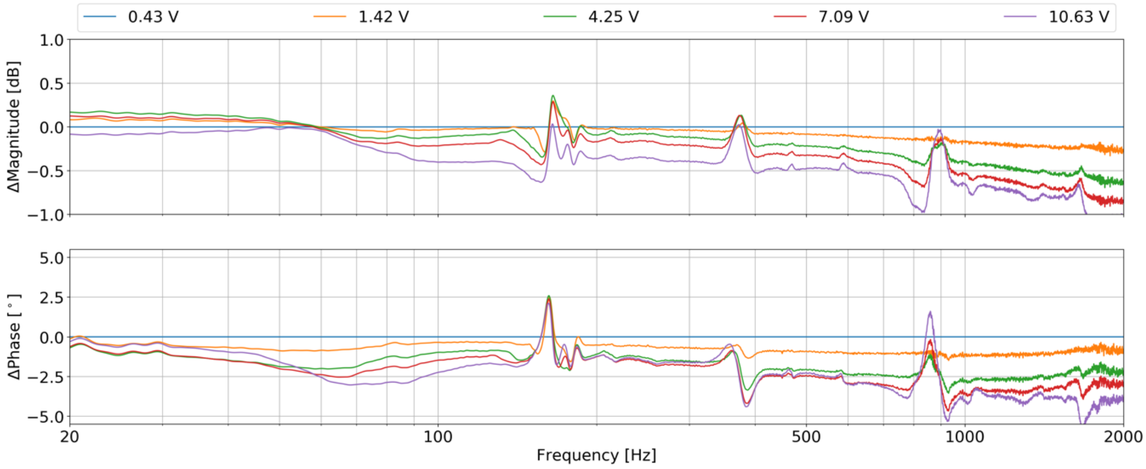

Figure 4 – Differences of magnitude and phase response at different levels relative to responses at 0.43V.

Above 60 Hz the transfer function decreases in amplitude and has a negative phase shift for increasing level. At high frequencies, this effect might be mainly a product of voltage dependent nonlinearities, as the voice coil displacement is small. At lower frequencies, the displacement increases which allows displacement and velocity dependent nonlinearities to have an effect. Below 60Hz the transfer function amplitude shows a rise up to 4.25V and drops at higher levels. This behavior is related to a downwards shift in the impedance resonance frequency. Above 5V dampening effects due to high displacements start to dominate (e.g. voice coil leaving the magnet or reluctant force of the suspension (4)). The shape of the various peaks in Fig. 4 indicate a shift in frequency of (anti-)resonances.

3.2 Sweep-Voltage Compensation

Fig.5 shows the measured transfer function with applied compensation of the *sweep-voltage approach*. The differences in transfer function for varying level are smaller than the changes observed in Fig. 4. Around the area of certain resonances in the speaker (e.g. around 160Hz) the performance of the compensation is less effective.

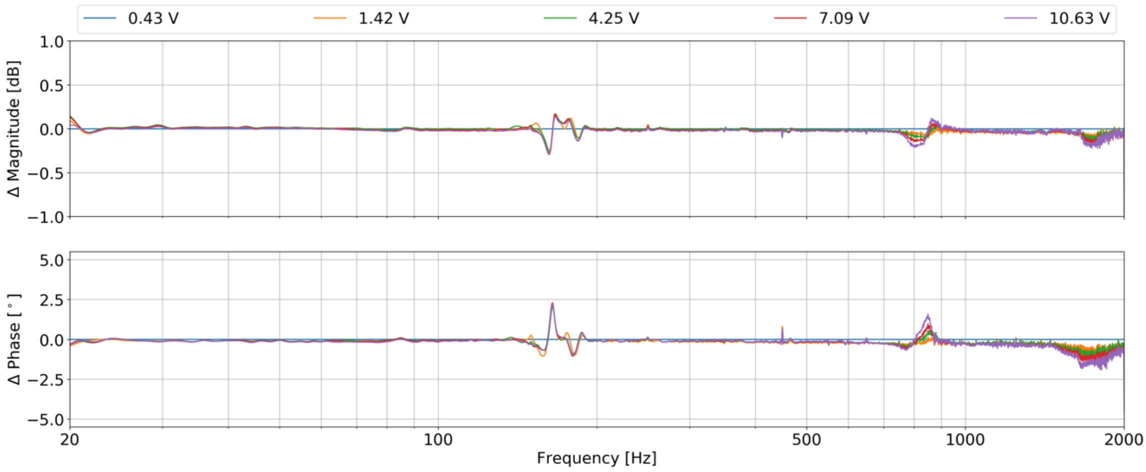

Figure 5 – Differences in loudspeaker response (Fig.1) with applied compensation filters at different levels relative to measurement at 0.43V

In order to relate the influence of such a compensation to a sound cancellation problem, the achievable reduction with and without compensation is shown in Fig.6 (as explained in Sec.2.2).

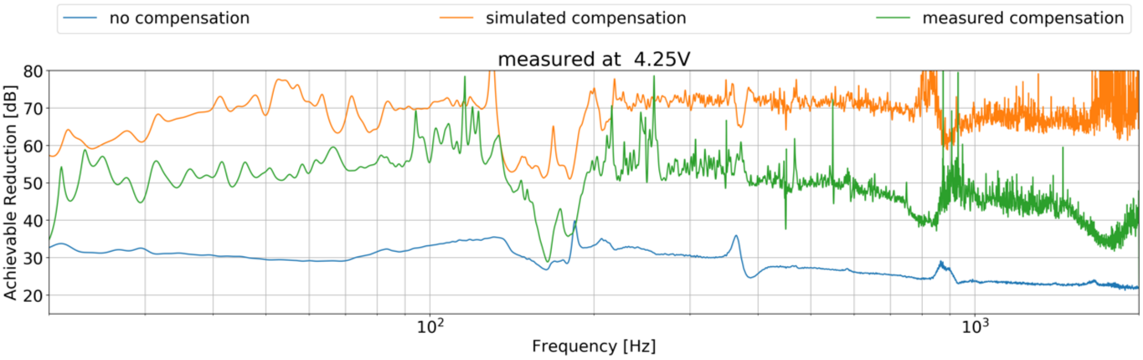

Figure 6 – Achievable reduction calculated from an error in transfer function. Simulated and measured for a sweep measurement at 4.25V compensated for a reference at 0.43V

The Figure shows that, for a sweep test signal, the compensation using the *sweep-voltage approach* improves a sound cancellation problem by 10-30 dB on a broad range of frequencies. It can be seen that the compensation tends to perform worse in the areas with resonances. Note that the test signal matches the estimation signal which leads to a good performance of the compensation. Fig.7. shows the performance of the sweep-volt compensation filter applied to other test signals.

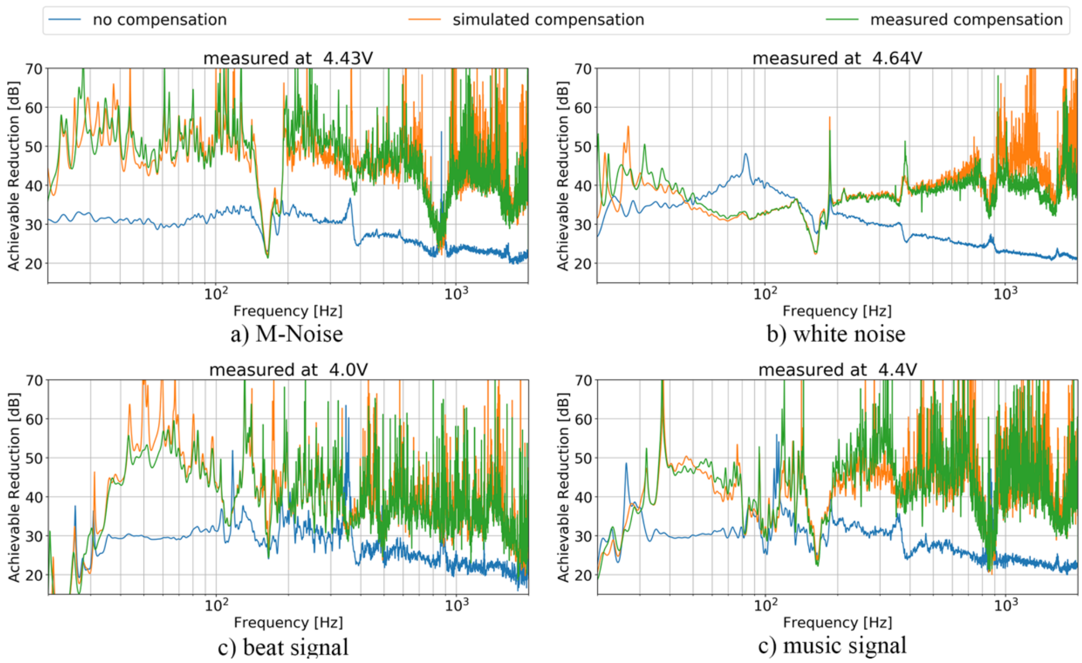

Figure 7 – Achievable reduction calculated from the error in transfer function for different signals. Simulated and measured for signals around 4 - 5 V compensated for a reference at 0.43V

For M-Noise (a) the compensation reaches an improvement of 10-20dB on a broad range of frequencies. Applied to white noise (b) the compensation shows improvement at higher frequencies (above 200Hz) but not between 50-190Hz, where the displacement is high. As two different signals can have the same voltage level but different displacement levels, the choice of level is critical. The compensation might be ineffective, if the voltage and displacement levels of the estimation and test signal are different. We assume that at low frequencies the compression due to nonlinearities is mostly dependent on displacement and at higher frequencies there is more dependency on voltage. This might be the reason for the poor performance from 50-190Hz, since displacement is high but the voltage level is used to calculate the compensation. The presence of high displacement below 50Hz suggests an inefficient compensation below 50Hz. but Fig.7 shows an improvement in achievable reduction. This is no contradiction since an over- or underestimation of a change in transfer function can still lead to a compensation filter that improves the error in transfer function compared to the uncompensated state. The compensation filter improves cancellation for both beat[2] (c) and music[3] (d) signals on a broad range of frequencies of 5-20dB. Below 30-40 Hz the beat and music signals do not contain enough energy to make meaningful statements.

In general, the results indicate that the proposed compensation shows an improvement on a broad range of frequencies. The extent of the improvement however is dependent on the frequency content and time variance of the applied signal.

3.3 N*oise-voltage* and *noise-displacement approaches*

As has been stated in Sec. 2.1, compensations based on different models were investigated. Apart from *sweep-voltage approach,* a *noise-voltage approach* and a *noise-displacement approach* were tested. The comparison of those three approaches is shown in Fig. 8. White noise was chosen as a test signal due to the difference in spectra compared to the estimation signals used to calculate the compensation. White noise has a flat spectrum in contrast to exponential sweeps and M-Noise, which have decaying spectra with increasing frequency.

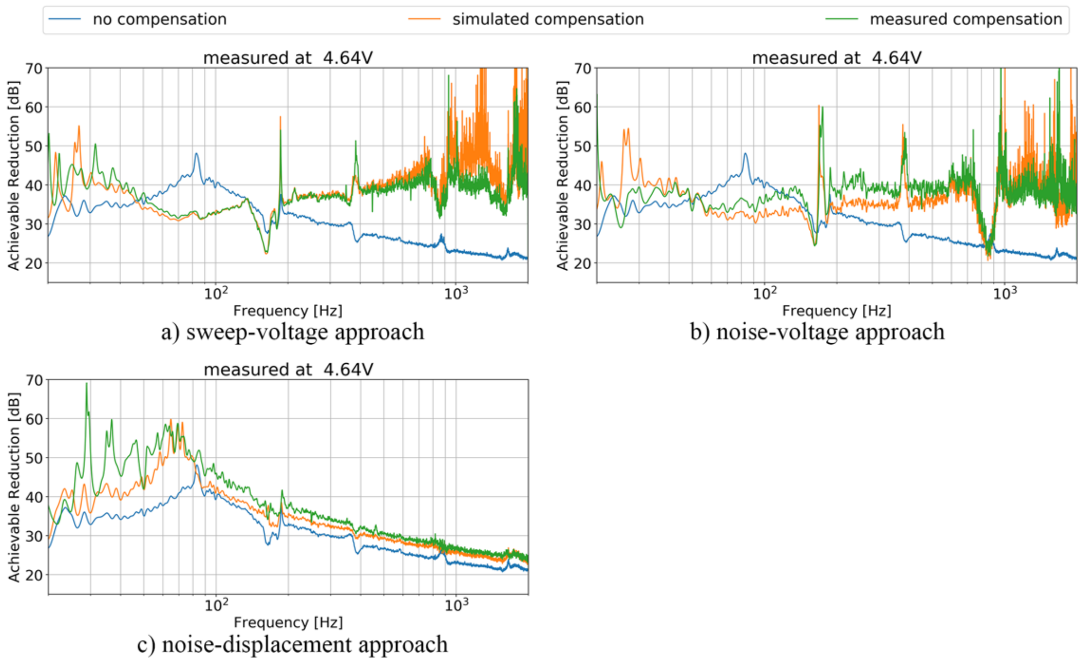

Figure 8 – Achievable reduction calculated from error in transfer function for different compensation approaches. Simulated and measured for a signal at 4.64V compensated for a reference at 0.43V.

Fig. 8 shows that for a white noise test signal and the *sweep-voltage approach* (a) or *noise-voltage approach* (b) there is no improvement visible at frequencies from 50-200Hz. For higher frequencies, 200-2000Hz, a consistent improvement of 10-20dB is present. The compensation based on displacement levels (c) shows an improvement of 5-10dB at low levels (below 80 Hz), while higher frequencies show less improvement (1-4dB). This behavior indicates that at lower frequencies a change in response is mainly dependent on displacement while at higher frequencies it is rather connected to voltage levels.

[2] A simple generated beat signal using kick drum, snare drum and hihat samples in a 4/4 signature standard beat accompanied by a simple increasing bass line.
[3] The music test signal was chosen such that it contained strong low frequency content, dynamic changes and a broad frequency spectrum

4. Discussion

The presented approach compensates a nonlinear loudspeaker system using level dependent changes in the linear transfer function. Results show that such a compensation can improve a sound cancellation system by several dB on a broad range of frequencies. Experiments with other loudspeakers show similar results. The compensation was performed for 10 different voltage and displacement levels spanning a range of around 24dB. While the results show the compensation around one RMS voltage level (4-5 V), the measurements at other levels show similar results.

The approach is rather simple and therefore limited. Especially if estimation and test signals have large spectral differences, the improvement from the compensation approach is small. This is assumed to be due to the dependency of nonlinear effects on multiple variables (e.g. current, displacement, etc.). A more sophisticated approach should be based on a compensation based on multiple variables.

Distortion products have not been considered in the presented compensation. It should be noted that such artifacts compromise a sound field control strategy as has been discussed by Ma (7). Even though might not be the primary issue, they become more relevant the more the compression of the fundamental is compensated.

The frequency resolution to calculate the compensation was 1Hz. The usage of a high resolution uncovers details and allows to compensate narrowband changes like resonance shifts. The effect of compression of the fundamental due to nonlinearities is expected to be similar for similar frequencies. Only looking at the compression of the fundamental, a coarse resolution might not only perform better but also limit the computational cost.

Two of the discussed compensation methods are dependent on the RMS voltage of the signal. In section 2.1 it is argued that instead of current level the voltage level is used to calculate and apply the compensation due to their interconnection. The relation between current and voltage is dependent on the impedance. But the impedance changes with level (e.g. DC resistance or resonance frequency shift). The interconnection between voltage and current is therefore level dependent. Using the RMS current instead of RMS voltage might improve the compensation strategy.

Addressing such limitations could further improve the presented compensation approach of nonlinearities in sound cancellation systems.

5. Conclusion

We showed that nonlinear effects in loudspeakers can be partly compensated by applying a level dependent linear compensation filter. A compensation approach based on a sweep test signal measured at different voltage levels was able to counterbalance the effect of (de-)compression on the excited frequency. The compensation was applied to different signals. For test signals with similar frequency content as the estimation signal, results show an improvement in sound cancellation of 10-20dB on average from 20-2000Hz. If test and estimation signal have different spectra the compensation approach shows only small improvements.

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