Internal use

Audio Engineering Society
Convention Express Paper 228

Presented at the 156th Convention
2024 June 15-17, Madrid, Spain

This Express Paper was selected on the basis of a submitted synopsis that has been peer-reviewed by at least two qualified anonymous reviewers. The complete manuscript was not peer reviewed. This Express Paper has been reproduced from the author's advance manuscript without editing, corrections, or consideration by the Review Board. The AES takes no responsibility for the contents. This paper is available in the AES E-Library (http://www.aes.org/e-lib) all rights reserved. Reproduction of this paper, or any portion thereof, is not permitted without direct permission from the Journal of the Audio Engineering Society.

Intermodulation distortion caused by loudspeaker system geometry variation, part 1

Juha Backman[1]

[1] *AAC Technologies Solutions Finland Oy, Lemminkäisenkatu 46, Turku, Finland*

Correspondence should be addressed to Author (juha.r.backman@gmail.com)

ABSTRACT

The paper discusses an intermodulation distortion phenomenon that can be observed especially in modern microspeaker designs. The distortion is heard when strong low-frequency signals are present at the same time as upper midrange and treble content. The distortion is shown through measurements and simulations to be due to the variation of the frequency response with the movement of the diaphragm. The upper midrange and treble response of typical microspeaker assemblies is determined by the Helmholtz resonance of the front cavity, and this varies strongly with diaphragm position since the relative change of the front cavity volume is large, but similar distortion is present in any device where the frequency response varies with the position of the diaphragm. The paper discusses two different acoustical solutions for reducing the distortion, suitable for microspeaker designs and other coupled-cavity loudspeakers, and a signal processing distortion reduction solution that is generally applicable to any system with position dependent frequency response.

1 Introduction

A few years ago, the manufacturers of small audio devices such as smartphones and tablets were intrigued by a form of distortion that got somewhat misleadingly named "piano distortion", as the phenomenon was apparently first observed in listening tests with piano music, although it is equally audible with any test material containing strong bass signals along with upper midrange sounds. Listening to the devices affected by this distortion reveals the harshness of the sound characteristic to intermodulation distortion, but the explanation of the physical cause and potential remedies of this have not been published until this work.

The typical devices where this kind of distortion is heard are loudspeakers with a flat diaphragm and a low front cavity, and an output port tuned to have a resonance at the upper limit of the used frequency range. The drivers when measured in the free field have usually rather benign distortion behaviour, with the harmonic distortion around the fundamental resonance being the only significant form of distortion. However, when the driver is measured in the actual acoustical environment used in the typical portable devices the harmonic distortion remains at a level determined mostly by the driver properties (taking the linear effects of the front cavity response on the distortion spectrum into account), but the intermodulation distortion, best characterized by multitone distortion, is at a substantially higher level.

The reason for the increased intermodulation distortion being observed only in recent years is the two-fold change in portable loudspeaker systems: on one hand, the driver technology has improved substantially, with the linear displacement of a current state-of-the art microspeaker being about

Internal use

twice the displacement of loudspeakers of earlier generations, from ten or more years ago, while the overall device design requirements have called for much narrower cavities in the front of the diaphragm.

The paper presents measurements from laboratory prototypes resembling typical portable devices. The measurement results and nonlinear simulations indicate that the observed intermodulation distortion is caused by the parametric variation of the front cavity upper midrange/treble Helmholtz resonance frequency with the low-frequency movement of the diaphragm, and driver related distortions or port flow noise are substantially below the midrange distortion caused by the cavity variation.

The algorithmic approach is more generally applicable to any form of intermodulation distortion where the variation of response with the movement of the diaphragm plays a central role, such as midrange response variation caused by inductance modulation or the narrow air spaces of typical compression drivers. The algorithmic distortion reduction can be used together with the acoustical methods.

2 Driver related intermodulation mechanisms

The mechanisms of intermodulation distortion in loudspeaker drivers are quite well known. Some of these mechanisms cause also harmonic distortion, but there are mechanisms such as inductance variation and the geometry variation induced distortion, the topic of this paper, that are observable only as intermodulation distortion. The main mechanisms are [1]:

- Variation of the magnetic flux as a function of displacement and of the voice coil current.
- Variation of the mechanical parameters (suspension stiffness, damping) with position
- Inductance modulation (displacement, current)
- Area modulation [2]

These mechanisms are observed in the output signal typically as amplitude modulation. Another mechanism, present in even an ideal loudspeaker with a large diaphragm velocity, is Doppler distortion, observable as frequency modulation [3]. The significance of Doppler distortion in microspeakers is discussed in Section 6.2.2.

Basically any mechanism that creates parameter (such as displacement) dependent frequency response variation causes intermodulation distortion. On the other hand, there are nonlinearity mechanisms such as hysteresis in magnet assemblies or piezoelectric transducers that is not clearly measurable as response variation, but cause intermodulation. Also amplifier crossover distortion (a problem of some early transistor amplifier designs) has no effect on frequency response, but is easiest to observe as intermodulation.

3 Intermodulation distortion measurement methods

The standardized measurements of intermodulation distortion are defined in IEC 60268-21:2018 [4]. Whenever applicable, the practices in this quite extensive standard should be used for documentation, but more flexible approach is necessary for experimental work.

Multitone measurements [5] are a common way of characterizing complex distortion, and they have been found to be very useful in this work.

Fixed frequency two-tone IMD measurements are useful for studying or visualizing isolated phenomena that are discovered using other methods. A more practical way of measuring the modulation is to use one fixed frequency and one sweeping frequency.

In addition to conventional measurement software, the measurement sweeps are easily generated, and the results are visualized, using audio editing software (in this example, Audacity freeware), Figure 1.

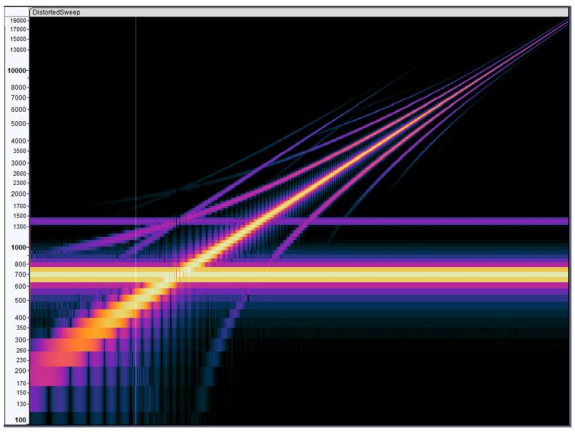


Figure 1. Result of an IMD sweep measurement visualized using an audio editor.

Internal use

4 Earlier work on boundary layer and parametric geometry effects

The effect of volume variation on the modulation distortion of a transducer (a compression driver) was first discussed by Harry F. Olson [6] [7].

The role of parametric variation of the values of acoustic subsystem parts with air or mechanical part movement versus nonlinearity of a subsystem in transducers has been best discussed in detail by Voishvillo [8], with focus on compression drivers.

One specific case related to microspeakers, the effect of parametric variation of flow resistance was studied in the earlier work by the present author [9] [10], but the work focused on low frequency harmonic distortion, which is affected by parametric effects much less than the intermodulation distortion, and as the focus of the work was on low frequency performance the model was simplified to omit the effect of cavity resonances.

5 Sample measurements

The existence of midrange intermodulation is clearly shown in a multitone measurement from a test loudspeaker with a 14*20 mm^2 outer dimensions flat diaphragm microspeaker with the back connected to a small sealed cavity and the front radiating through a cavity with a nominal internal height of 1.5mm. The output port was tuned to 5 kHz. All the measurements, unless otherwise stated, are made at 1V RMS input voltage.

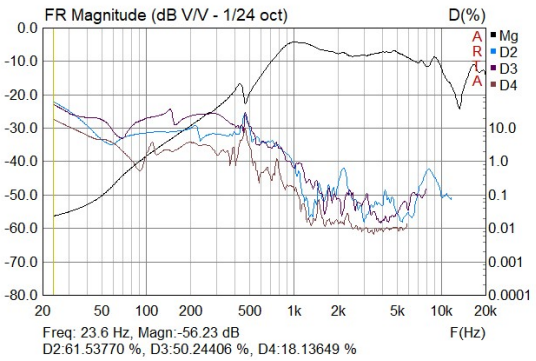


Figure 1. Frequency response and harmonic distortion components without a front cavity.

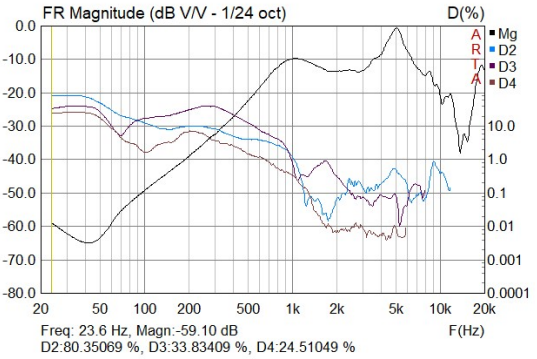


Figure 3. Frequency response and harmonic distortion components with a front cavity.

The harmonic distortion measurement shows some increase with the front cavity, although the third harmonic increase is partially an artefact due to the resonance boost of the front cavity, not an actual indicator of increased transducer nonlinearity. The harmonic distortion figures are not alarmingly in either case, and especially the distortion without the front cavity is actually quite low.

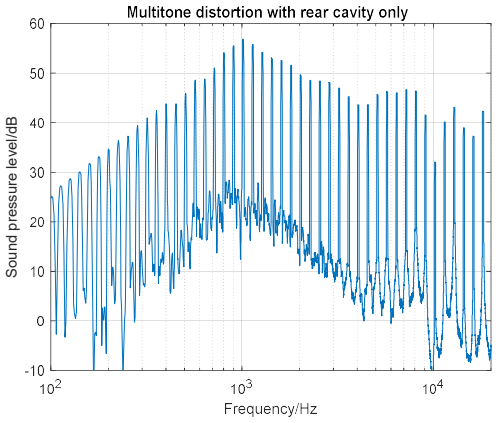


Figure 4. Multitone spectrum without a front cavity.

Internal use

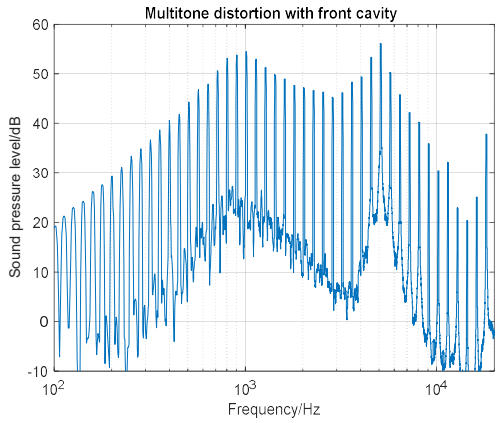


Figure 5. Multitone spectrum with a front cavity.

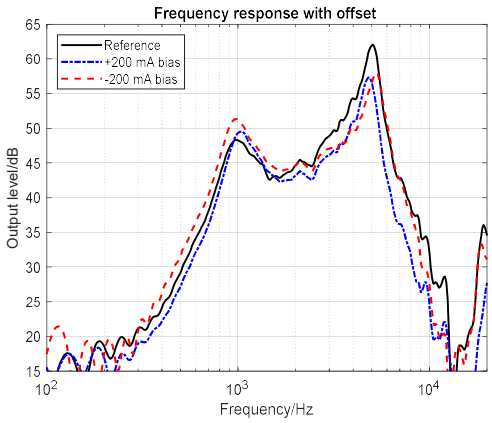


Figure 6. Frequency responses measured in the nominal position and with offsets generated using ±200 mA bias currents.

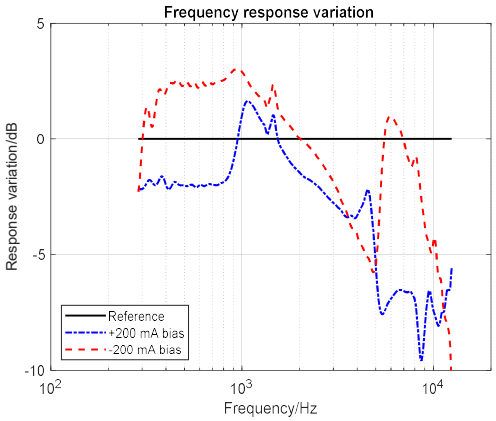


Figure 7. Deviation of the offset responses from the reference.

It is quite easy to convince ourselves that this increase of midrange intermodulation is a phenomenon related to the enclosure by comparing the multitone and harmonic distortion measurements of the same driver in the same rear cavity but with an open front. The low-frequency harmonic distortion behaves in a very similar way in both measurements, and the multitone distortion of the transducer itself shows no midrange irregularities. Harmonic distortion measurements show some differences between the open front and the resonator cavity, but the frequency dependence of the harmonic distortion is not similar to the intermodulation distortion.

6 An explanation of the midrange distortion

6.1 Frequency response variation

The best explanation for the observed midrange intermodulation around the resonance frequency of the front cavity appears to be the variation of the frequency response around the resonance when the diaphragm moves with low-frequency signals.

The effect can be measured by applying a static displacement to the diaphragm, for instance by adding a DC current to the measurement voltage signal. In this measurement the signal voltage was reduced to 100 mV to ensure that the displacement caused by the signal voltage is significantly smaller than the displacement from the bias current.

6.2 Other potential distortion mechanisms

6.2.1 Flow noise

Another possible hypothesis would be that the output port creates flow noise that would be then perceived as some kind of modulation distortion. This can be analysed by taking a more detailed look at the two-tone intermodulation spectrum. It is clearly visible that even if the low-frequency flow generates some broadband noise components, their level is well below the intermodulation distortion components and also, in this case, the slight noise peak is well above the port resonance frequency of 5 kHz, where the multitone distortion is the strongest.

Internal use

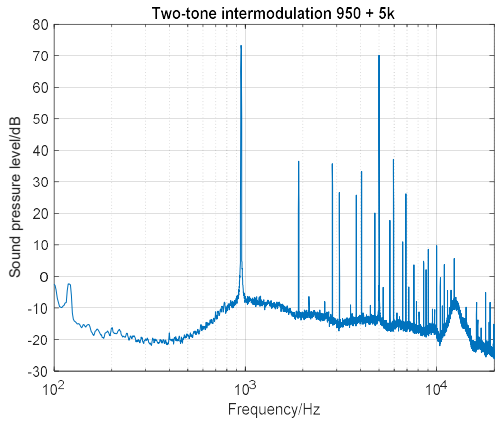


Figure 8. Two-tone intermodulation spectrum with the front cavity.

6.2.2 Doppler distortion

Doppler distortion in microspeakers is an interesting question. The displacements of microspeakers are small, but, on the other hand, the diaphragm velocity maximum occurs at a significantly higher frequency than in full-size loudspeakers. As an estimate of the order of magnitude, the peak velocity of a high quality microspeaker, vibrating at 1 kHz with a 1mm amplitude is 6.28 m/s = 0,018*c*, so the peak velocity cannot be regarded as vanishingly small as compared to the speed of sound. The peak velocity in this example is actually quite close to the perception threshold suggested by Fryer [11] for music, about 10 m/s. Measurements of Doppler distortion in microspeakers used in this test, however, do not show any conclusive results, which agrees with Klippel's analysis [12] showing that Doppler distortion in microspeakers is typically well below other distortions.

6.2.3 Air nonlinearity

The nonlinear compressibility of the air causes distortion through two known mechanisms: nonlinear compressibility of the air inside the enclosure [13], and nonlinear propagation of the sound [14], [15]. As the displacement of the diaphragm is a significant fraction of the enclosure volume, the internal air nonlinearity should be taken into account for general modelling, but it is affecting mostly low frequencies and does not explain the observed frequency dependence of the midrange distortion. Nonlinear propagation would require a longer port than typically found with microspeakers.

7 Acoustical distortion compensation

As the reason for the midrange intermodulation is found to be the variation of the loudspeaker cavity resonance, this immediately suggests two approaches for distortion reduction: suppressing the resonance, or reducing the variation of the resonance. (The obvious alternative of increasing the front cavity height is very seldom available in typical microspeaker applications.)

7.1 Resonance suppression

The suppression of the resonance can be achieved through attaching a damped resonator to the front cavity [16]. The resonator can be either tuned to suppress the resonance and reduce the intermodulation distortion, or the resonance can be tuned to a higher frequency to extend the front cavity response, but in this case it will be less effective in controlling the distortion.

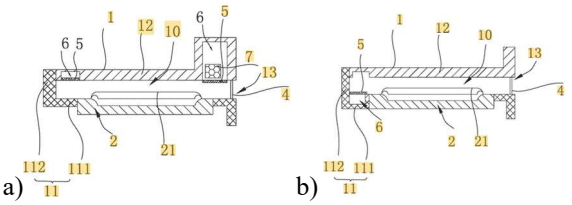


Figure 9. Examples of implementing a resonator in the output port (a) or behind the driver (b).

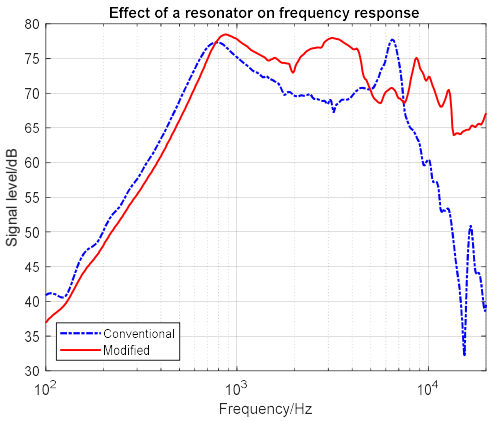


Figure 10. Frequency response of test loudspeakers with or without the resonator. The loudspeaker in this test is designed for a higher front cavity resonance than the loudspeaker used in other tests.

Internal use

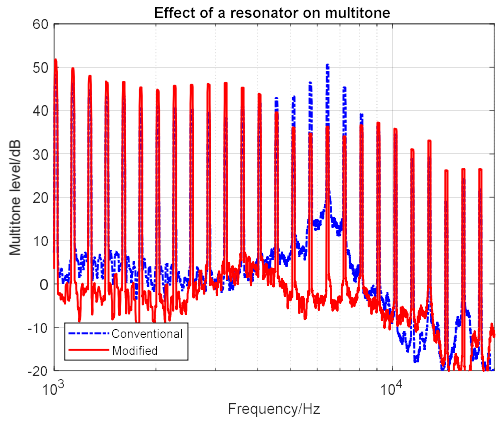


Figure 11. The effect of the resonator on the multitone spectrum.

7.2 Resonance variation reduction

The method is based on making also the acoustical inductance of the front port to vary with the displacement of the diaphragm [17]. This is achieved by creating a small constriction on the cavity near the exit port, but inside the edge of the driver. This constriction creates a small, variable inductance which keeps the resonance frequency more constant than in conventional solultion, while having only a minimal effect on the boost provided by the front cavity resonance.

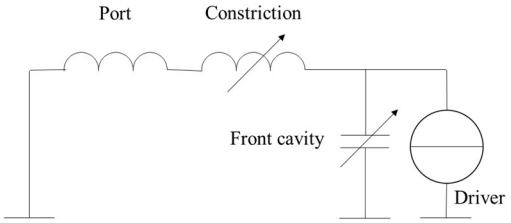


Figure 12. Simplified equivalent circuit of the front cavity with a variable constriction.

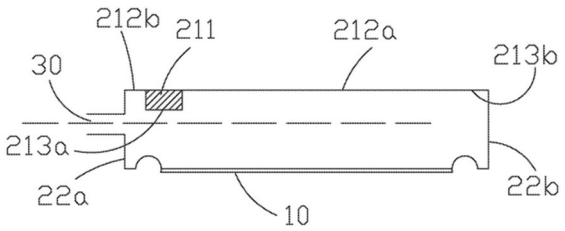


Figure 13. One way of implementing a constriction that causes an acoustic inductance varying with the diaphragm movement.

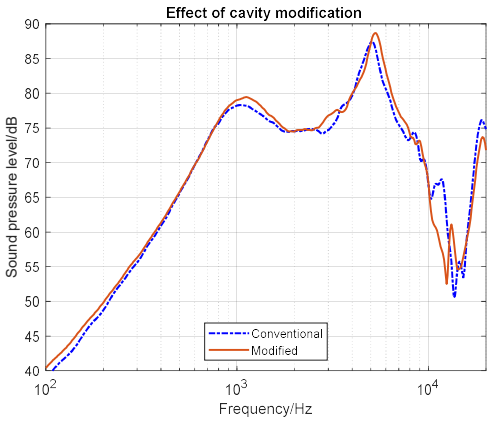


Figure 14. Effect of front cavity structure modification on the frequency response.

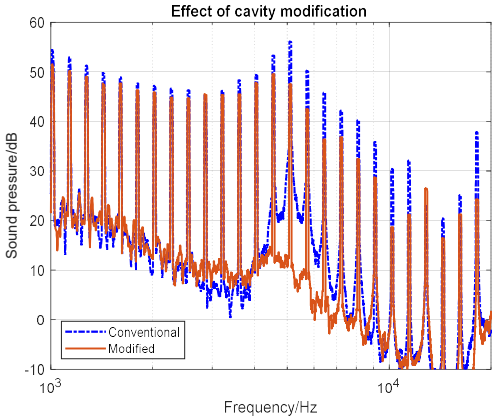


Figure 15. Effect of front cavity structure modification on the multitone distortion.

8 Distortion compensation algorithm

The acoustical solutions are effective for limiting the effects of single front cavity resonance variation, but they impose constraints on the mechanical design and are ineffective against multiple moving resonances or e.g. response variation due to position dependent voice coil inductance. These limitations prompted the development of an algorithmic approach for controlling the general response variation related to displacement.

The algorithmic approach is based on using one or more variable biquad filters controlled by a computational model of the diaphragm displacement, along with a delay compensation to ensure that the filter control and the low frequency diaphragm movement are precisely aligned. The filters equalize

Internal use

the difference between the frequency response at the nominal position and the displaced position. The biquad coefficients are determined from a simulation model of the loudspeaker unit or from measurement of an actual loudspeaker.

The typical loudspeaker nonlinearity controller, intended to reduce the driver nonlinearity, has a driver model and a nonlinear, time variant controller. The nonlinearity model can be updated using current sensing.

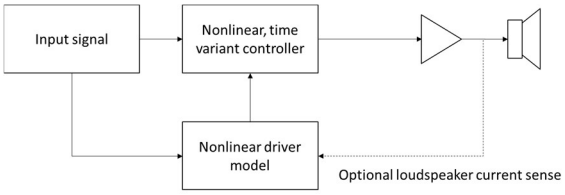


Figure 16. Generic structure of typical loudspeaker nonlinearity compensators.

When reducing the high-frequency intermodulation, the system actually behaves in a linear manner with respect to the high-frequency signals (superposition principle applies to the high-frequency content), so the equalizer can be linear, but time variant and controlled by the low-frequency system displacement model.

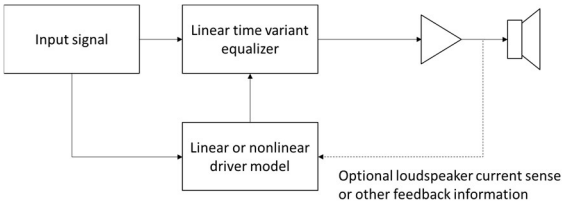


Figure 17. Generic structure of a parameter dependent frequency response compensator.

The equalizer is essentially a pole-zero type compensator, which is implementable as a combination of one or more adjustable biquad blocks, Figure 16. The filter coefficients behave quite smoothly as the function of displacement, so the control model can be implemented e.g. as an interpolated lookup table or a low-order polynomial.

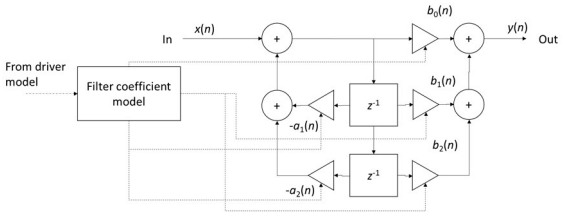


Figure 18. The structure of a variable biquad block.

When the filter coefficient lookup tables are formed using typical curve fitting algorithms, the DC gain of the filters may fluctuate slightly. The distortion performance of the algorithm at low frequencies can be improved by normalizing the coefficients for unity DC gain. Also, in practical implementations it may be advisable to reduce the compensation at low frequencies, either by implementing a bypass filter on the compensation signal path or by implementing an adjustable filter section that reduces the low-frequency compensation.

Although test signals, such as a modulated multitone, are very useful in confirming the functionality of the algorithm, the most realistic test is to use actual music or speech signals. One example of simulation test results is given below in Figure 17. Simulink is used for both the nonlinear transducer and front cavity model and for implementing the compensation filter. The distortion is analysed here as the difference of the output signal computed using a linear loudspeaker model without compensation and the output signal computed using a nonlinear loudspeaker model, with or without compensation.

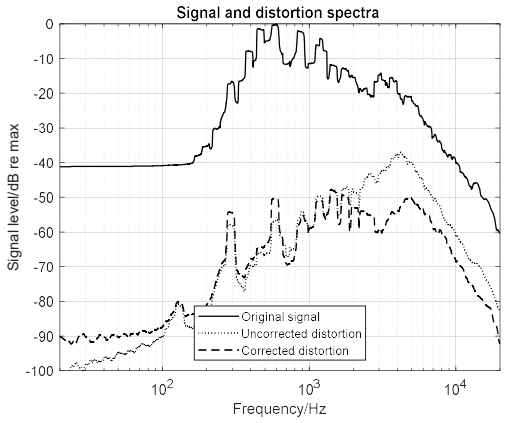


Figure 19. Distortion reduction of a simulated music signal using the algorithm.

Internal use

9 Conclusions

This paper has shown that a coupled cavity loudspeaker enclosure can easily generate a significant amount of intermodulation distortion. The distortion, however, is controllable through a suitable acoustical design or signal processing.

References

[1] Wolfgang Klippel, "Tutorial: Loudspeaker Nonlinearities – Causes, Parameters, Symptoms", *J. Audio Eng. Soc*., vol. 54, no. 10, October 2006, pp. 907 – 939.

[2] Wolfgang Klippel, Joachim Schlechter, "Dynamic Measurement of Transducer Effective Radiation Area", *J. Audio Eng. Soc*., vol. 59, no. 1/2, January 2011, pp. 44 – 52.

[3] George L. Beers, H. Belar, "Frequency-Modulation Distortion in Loudspeakers", *Proceedings of the IRE*, vol. 31, no. 4, April 1943, reprint *J. Audio Eng. Soc*., vol. 29, no. 5, May 1981, pp. 320 – 326.

[4] IEC 60268-21:2018, "Sound system equipment – Part 21: Acoustical (output-based) measurements."

[5] Eugene Czerwinski, Sergei Alexandrov, Alexander Voishvillo, Alexander Terekhov, "Graphing, Interpretation, and Comparison of the Results of Loudspeaker Nonlinearity Measurement", J. Audio Eng. Soc., vol. 52, no. 4, April 2004, pp. 332 – 357.

[6] Harry F. Olson, "Horn Loud Speakers, Part II. Efficiency and Distortion", *RCA Review*, vol. 2, no. 2, October 1937, pp. 265 – 277.

[7] Harry F. Olson, *Acoustical Engineering*, Van Nostrand Company, 1957, pp. 225 – 226.

[8] Alexander Voishvillo, "Nonlinear Versus Parametric Effects In Compression Drivers", *115th Convention of the Audio Eng. Soc*., New York, October 10 – 13 2003, Preprint 5912.

[9] Juha Backman, "Distortion from Boundary Layers", *103rd Convention of the Audio Eng. Soc*., September 26 – 29 1997, New York, Preprint 4617

[10] Juha Backman, "A Non-Linear Model of a Small Transducer", *Audio Eng. Soc. UK 13th Conference: Microphones & Loudspeakers*, March 1998, pp. 65 – 69.

[11] Peter A. Fryer, "Simulation and Investigation of Doppler Distortion", *56th Convention of the Audio Eng. Soc*., Paris, March 1 – 4 1977, Preprint 1197.

[12] Wolfgang Klippel, "Modeling the Large Signal Behavior of Micro-Speakers", *133rd Convention of the Audio Eng. Soc*., October 26 – 29 2012, Preprint 8749.

[13] Harry F. Olson, "Analysis of the Effects of Nonlinear Elements Upon the Performance of a Back-Enclosed, Direct Radiator Loudspeaker Mechanism", *J. Audio Eng. Soc.*, vol. 10, no. 2, April 1962, pp. 156 – 162.

[14] Eugene Czerwinski, Alexander Voishvillo, Sergei Alexandrov, Alexander Terekhov, "Air-Related Harmonic and Intermodulation Distortion in Large Sound Systems", J. Audio Eng. Soc., vol. 47, no. 6, June 1999, pp. 427 – 446.

[15] Guy Lemarquand, Michel Bruneau, "Nonlinear Intermodulation of Two Coherent Acoustic Progressive Waves Emitted by a Wide-Bandwidth Loudspeaker", *J. Audio Eng. Soc.*, vol. 56, no. 1/2, January 2008, pp. 36 – 44.

[16] Benedict Slotte, "Speaker Box", US Patent US11265629B2, granted 2022-03-01.

[17] Juha Backman, "Loudspeaker", US Patent US20240048893A1, granted 2024-02-08.