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# Comparison of acoustic, electric and laser contactless displacement measurements applied to observe nonlinear behaviour of diaphragm movement in dynamic loudspeakers

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**Abstract** Dynamic loudspeakers introduce distortion into reproduced auditory signal, especially in case of operation with high power or low frequency. Assessment of distortion by measuring current of voice coil should allow to compensate or prevent the product of distortion. The paper contains a comparison of displacement capture by optical triangulation displacement sensor, acoustic signal acquired using microphone, voltage and current. In case of processing low frequency signals, acoustic signal captured by microphone is dominated by harmonics of the fundamental frequency, and the laser displacement sensor provides more accurate information about the diaphragm displacement. Harmonics recorded in current have the same character as the displacement signal for low frequency, but for frequencies above the resonance, acoustic signal is a better source of displacement information.

**Keywords:** dynamic loudspeaker; nonlinear distortion; laser displacement sensor.

## 1. Introduction

All dynamic loudspeakers introduce distortion into reproduced auditory signal. Depending on application, some distortion may be acceptable, contributing to characteristic coloration of sound [1]. That is why it is sometimes proposed to evaluate audio system using not synthetic, but music signals [2], and to apply auralization technique for evaluating nonlinear distortion in a subjective manner [3]. However, this kind of evaluation is easily influenced by other, non-technical factors [4,5]. Therefore, most works concentrate on objective measures and consider linear response of a loudspeaker as a desired behaviour, proposing various techniques to identify nonlinear effects and to help avoid or minimise them.

Klippel and Seidel [6] propose studying distortion in time domain using five measures to acquire both amplitude and phase information. It would allow to observe the distortion dependency on frequency, displacement, and other state variables. Displacement is assumed to be similar to the sound pressure in the near field. They also point possible sources of linear and nonlinear distortion, dividing the latter into located in the electro-mechanical part, triggered by improper functioning of the speaker, and not reproducible random distortion. Earlier work of Klippel [7] discusses modelling of nonlinear system to predict phase and amplitude response. In later work [8] a broader group of transducers is discussed and simulations are introduced to identify symptoms of particular distortions. A particular example is the compensator that considers nonlinear viscoelasticity of the suspension, discussed by Tian et al. [9]. Further application of modelling is presented by Iwai and Kajikawa [10]. They model nonlinear distortions to design a 3rd-order IIR filter, which compensates for distortions of a loudspeaker. Other attempts include works of Kajikawa [11], Schneider et al. [12], and Kaizer [13].

Optical sensors can be applied to measure vibration of a loudspeaker cone, as discussed by Hernandez et al. [14] or Klippel and Schlechter [15]. In the latter work, a scanning vibrometry is considered as an alternative to Doppler interferometry. Through applying a mechanical scanning system with rotational and linear actuators it is possible to analyse not only surround resonance, but also rocking and break-up modes. A similar method is presented in the work of Bellini et al. [16], but here the laser Doppler vibrometer is applied. Other laser-based approaches are presented in works of Moreno, Bog and Medina [17,18].

Loudspeakers can be characterised using Thiele/Small parameters [19]. Geiger [20] uses these parameters to design a closed loop system based on optical linear displacement sensor and voltage-controlled current source. The loop allows to control loudspeaker cone motion, and in turn, to reduce harmonic distortion. Pillonnet et al. propose to reduce distortion by directly controlling coil current [21].

In this approach no optical sensor is used, and the loudspeaker operation is controlled on the basis of large signal impedance model. Nonlinear inductance of voice coil is modelled and studied in the work of Dobrucki et al. [22], and the effect is considered a significant source of intermodulation distortion. Adaptive nonlinear control of loudspeakers is also considered in older works of Klippel [23,24].

Applying optical sensors is an effective way to observe and control loudspeaker operation. However, it is better suited for research purposes and may not be a practical approach in typical use cases, where additional equipment cannot be mounted on the speaker. Even though loudspeaker models can work, and improve distortion behaviour without direct, optical displacement or velocity information, using only electric parameters, optical sensors can be used for validation of such systems. Bai and Lee present sensorless, DSP-based observer to determine a speaker cone velocity, and compare it to measurements conducted using laser vibrometer [25].

Nonlinear behaviour of loudspeakers can be minimised by improvements within transducers only up to a certain point. Further improvements require better understanding and identification of nonlinear effects, and ability to design numerical models controlling adaptive linearisation systems. Models require information regarding loudspeaker state and operation. This kind of information can be obtained by measuring electric quantities, analysing acoustic signal, or using optical sensors. While all of these are possible in the laboratory, under real use conditions optical sensors, which can provide the most comprehensive data, are difficult to apply. Therefore, it is important to be able to substitute one kind of measurement with another, easier to carry out.

This work presents a set of measurements carried out using three techniques: acoustic, electric, and laser displacement sensor. Results in the form of time-domain signal waveforms, spectra, and harmonic distortion are compared for different signal levels and frequencies. Differences are discussed, so knowledge about area of applicability can be used to choose the most accurate method to observe nonlinear behaviour of loudspeakers depending on signal parameters.

## 2. Materials and Methods

The aim of the experiment was to capture, analyse and compare three types of loudspeaker response: acoustic signal, diaphragm movement, and voice coil current for variable power of the processing signal and variable frequency, especially low frequency. The experiment contains measurements of displacement captured by the laser displacement sensor, the acoustic signal captured by the condenser microphone, the voltage, and the current in the loudspeaker circuit. The current and movement of the diaphragm coincidence will be used to observe distortion in sound generated by loudspeakers without additional sensors.

### 2.1. Test stand

The experimental setup was presented in Figure 1, consisting of a laser displacement sensor, condenser microphone, data acquisition card, and audio interface. The equipment is listed in Table 1, and its parameters are given in Table 2. The current measurement was obtained by measuring the voltage across a 0.2 [Ω] resistance. The voltage was measured directly on the loudspeaker connector, so the true value of the voltage was measured. The additional series resistance leads to a worse damping factor, but its value is minor and can be considered equivalent to the wire resistance in practical applications. The measurement based on a triangulation laser displacement sensor provides a voltage signal of displacement. The principle of measurement was presented in Figure 2. This type of transducer measures displacement, as opposed to an optical vibrometer, which measures acoustic velocity. The measurements were carried out in the electronics laboratory, which was not adopted acoustically, but the test stand was outfitted with acoustic absorption materials, with the aim of preventing the first acoustic reflection. The measurement point was located in the center of the dust cap, marked with a red dot in Figure 4. The picture of the loudspeaker under test is presented in Figure 3.

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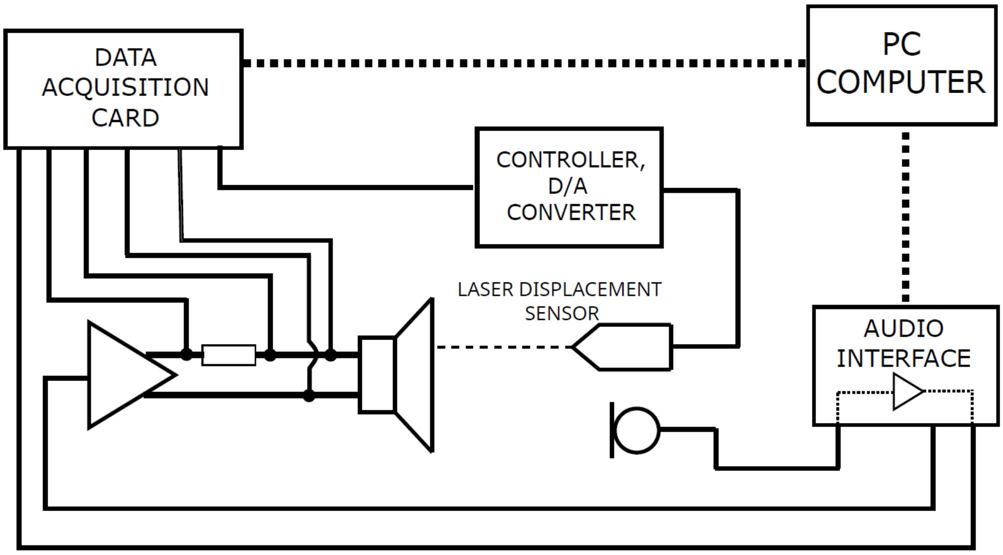


**Figure 1**. The test stand diagram.

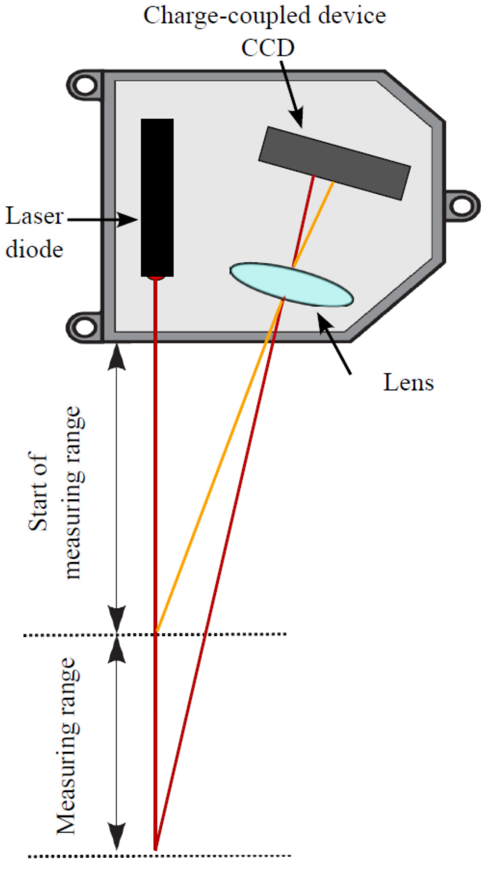


**Figure 2.** The displacement laser sensor diagram. Source: elaboration of the author, based on [26].

**Figure 3**. The test stand with the loudspeaker.

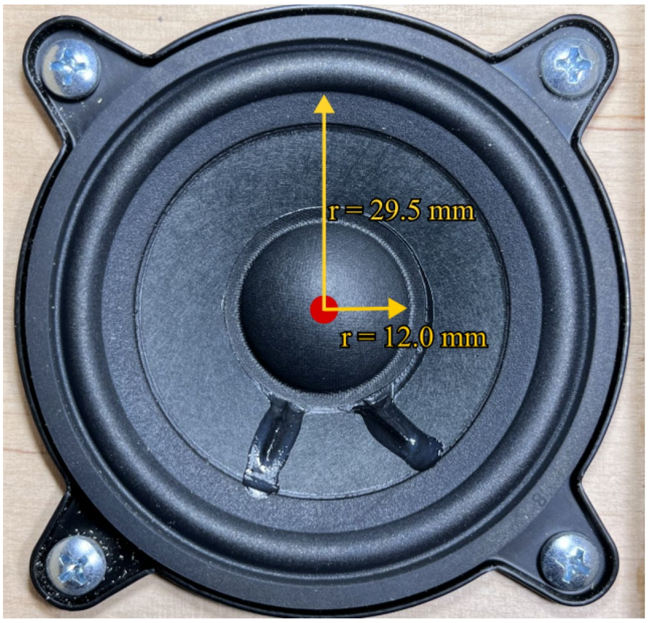


**Figure 4.** The dimensions of the diaphragm with the position mark of the measurement point.

**Table 1.** Apparatus used.

| Device | Manufacturer | Model |
|---|---|---|
| Data acquisition card | National Instruments | 9215 |
| Laser displacement sensor | Micro-Epsilon | ILD2300-50 |
| Controller, D/A converter | Micro-Epsilon | CSP2008 |
| Audio power amplifier | Park Audio | CF700-4 |
| Microphone | Audix | TM1PLUS |
| Audio interface | RME | FIREFACE UC |
| Loudspeaker | Faital-Pro | 3FE25 |

**Table 2.** Operating parameters.

| Device | Parameters |
|---|---|
| Data acquisition card | sampling frequency: 110497 Hz per channel,<br>input range: ±10 V. |
| Laser displacement sensor | sampling frequency: 49000 Hz,<br>input range: 24.225 ÷ 39.225 mm (0.75 mm/V). |
| Controller, D/A converter | input sampling frequency: 49000 Hz,<br>output sampling frequency: 100000 Hz,<br>output range: −10 ÷ +10 V (0.75 mm/V). |
| Audio power amplifier | AB class power amplifier,<br>power rating: 2 x 220 W @ 8 Ω,<br>frequency range(0, - 0.2 dB): 20 ÷ 20000 Hz,<br>total harmonic distortion ( 1kHz ): -90.5 dB,<br>total harmonic distortion ( 20Hz…20kHz ): -74.0 dB,<br>sensitivity: 0.775 V,<br>SNR: 100 dB,<br>slew rate: 40 V/μs. |
| Microphone | sensitivity: 6.4 mV/Pa @ 1kHz,<br>frequency response: 20 ÷ 25000 Hz ± 2 dB. |
| Audio interface | sampling frequency: 44100 Hz,<br>microphone input gain: −18 dB. |
| Loudspeaker | nominal diameter: 80 mm,<br>frequency range: 100 ÷ 20000 Hz,<br>maximum power handling: 40 W,<br>maximum excursion: 1.83 mm,<br>maximum excursion before permanent damage: 7.9 mm [27],<br>inner dimensions of cabinet: 157 × 88 × 80 mm,<br>volume of cabinet: 1.11 dm3,<br>type of cabinet: closed,<br>resonance frequency: 180 Hz. |

## 3. Methods

The aim of the experiment was to capture, analyse and compare three types of loudspeaker response: acoustic signal, diaphragm movement, and voice coil current for variable power of the processed signal, and variable frequency, especially low frequency. It should allow us to predict displacement distortion by analysing the current and voltage in the loudspeaker circuit. The low frequency band is assumed to be one octave below, up to one octave above the resonance frequency, for the scope of this study (90 ÷ 360 Hz).

A specific test signal was required due to specific test conditions. The displacement value of the diaphragm was close to the maximum excursion before damage to the suspension or voice coil. For higher frequencies, the current carried by the voice coil causes a growth of temperature, which could lead to overheating. Therefore, the excitation signal was a sine wave of approximately 300 ms duration containing full sine periods to avoid discontinuity spikes at the end and beginning. The excitation signal contained a break to cool the voice coil, to prevent damage, after each 300 ms signal block. The maximum voltage level at 0 dB was 68.94 V peak-to-peak (reference value). Measurements were carried out for levels: 0, −3, −6, −9, −12, −18, −24, −30, and −36 dB. Frequencies ranged from 53 to 4220 Hz at 1/6 octave resolution. The time delay in the displacement signal is caused by digital processing in the laser sensor equipment, whereas the acoustic signal delay is due to the time it takes for the acoustic wave to propagate between the loudspeaker and the microphone. An example of captured signals is presented in Figure 5. The time shifts of the signals have been compensated for easier comparison of waveform shape. The time delay was calculated based on cross-correlation.

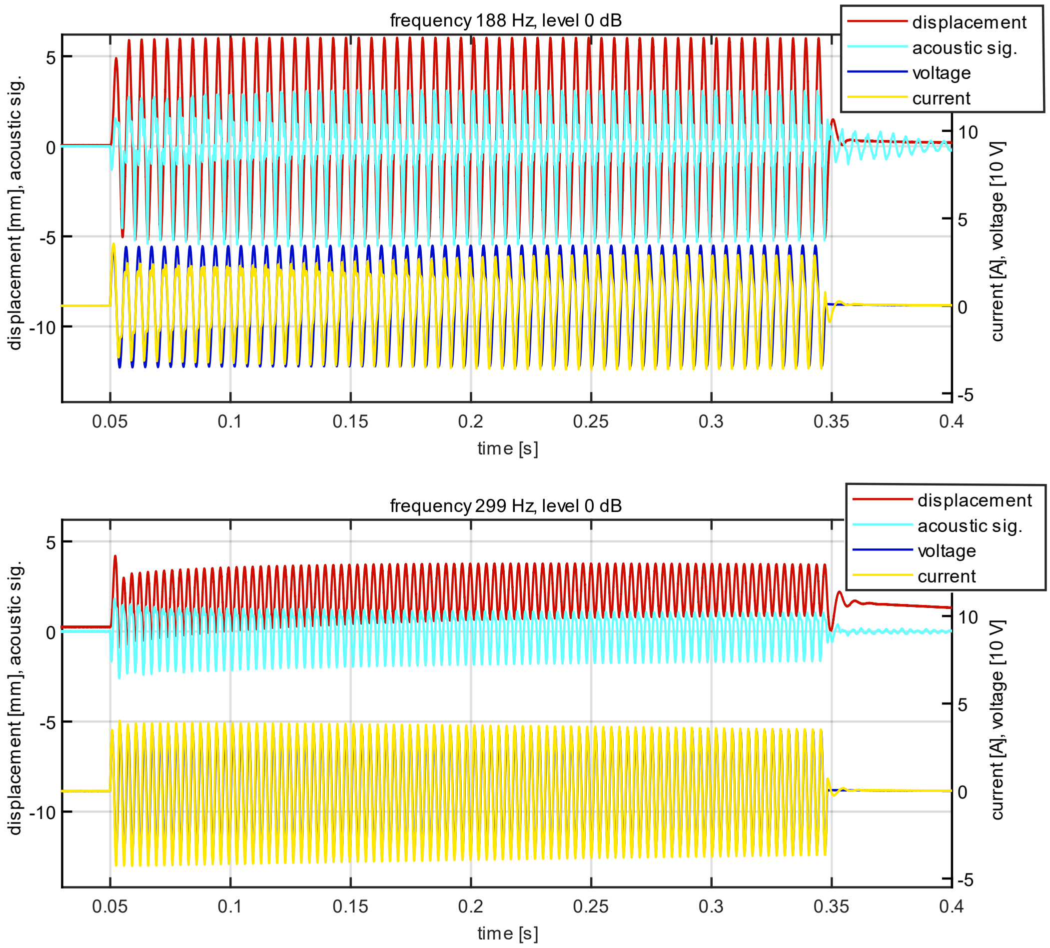


**Figure 5.** The waveforms recorded for the maximum power of excitation signal – full 300 ms block. The waveforms placed above represent the displacement and acoustic signals, while the waveforms placed below represent the voltage and current.

In this study, the focus was on measuring the displacement of the diaphragm because, in the low-frequency range (below resonance frequency) and within the linear operating range, the excursion is proportional to the force acting on the diaphragm:

$$|\mathbf{x}| \propto |\mathbf{F}_c| C_{ms} , \tag{1}$$

where: $\mathbf{x}$ [mm] – excursion of the voice coil, $C_{ms}$ [mm/N] – mechanical compliance of the loudspeaker suspension.

Moreover, $\mathbf{F}_c$ – force acts on voice coil as proportional to current:

$$|\mathbf{F}_c| = Bli_c , \tag{2}$$

where: $Bl$ [N/A] – force factor, $i_c$ [A] – electric current in the voice coil.

The power spectral density (PSD) plots presented were computed using the Blackman window with a length of 32768 samples. The centre of the window was located at the midpoint of the duration of the excitation signal.

Total harmonic distortion (THD) was calculated using the following expression:

$$\mathrm{THD} = 20 \log_{10} \frac{\sqrt{\sum_{k=2}^{n} U_k^2}}{U_1} , \tag{3}$$

where: $U_1$ – effective value of the fundamental component, $U_k$ – effective value of the k$^{th}$ harmonic component, $k$ – index of harmonic component, in range from 1 to 7.

## 4. Results

As we can see in Figures 5 and 6, the waveforms captured using three different methods have different shapes. Particularly for a frequency of 67 Hz and level of 0 dB, the fundamental frequency is unnoticeable in the acoustic signal. In Figure 7 the fundamental frequency is weaker than harmonics that are the result of the distortion. The loudspeaker efficiency at 67 Hz is small, due to the dimensions of the diaphragm being significantly smaller than the wavelength.

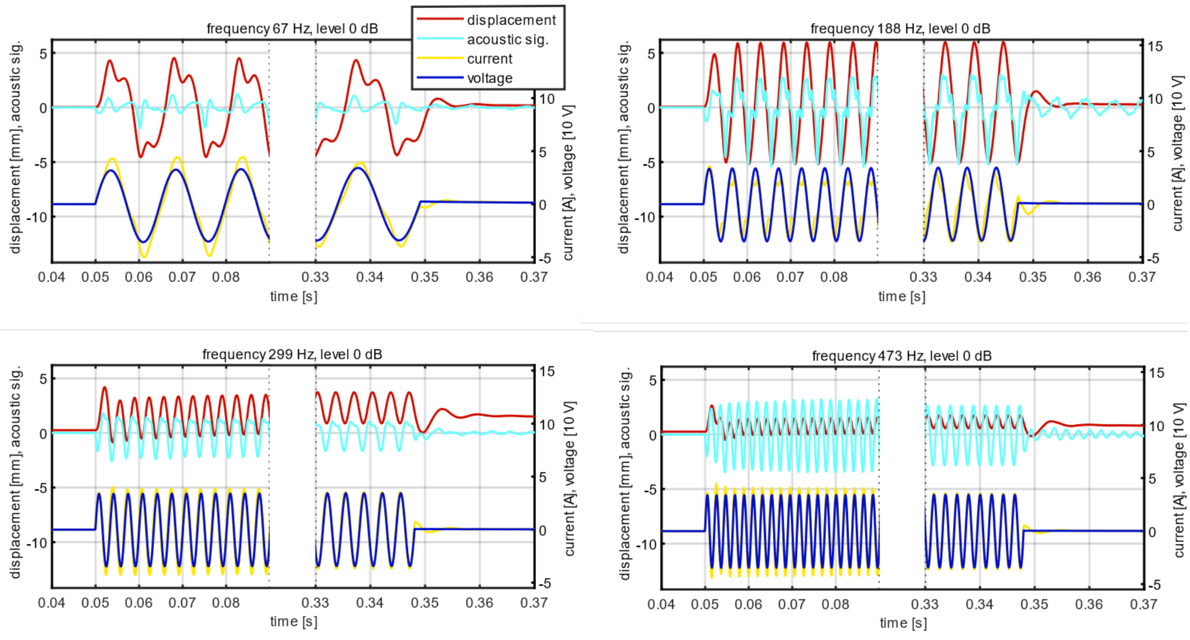


**Figure 6.** The waveforms recorded for the maximum power of excitation signal. The waveforms placed above refer to the displacement and acoustic signal, placed below current and voltage.

In Figure 5 a slow-changing displacement is visible; this phenomenon is described in [28, 29]. This phenomenon is shown in Figure 12 for a frequency of 750 Hz. The left part of Figure 12 presents waveforms for a series of signal excitation levels, and the right part presents displacement of the middle point of oscillation, determined by moving mean. As the level of the excitation signal increases, the slow-changing displacement caused by the resistance force increases as well.

Figures 7, 8, 9, 10 show PSD for frequencies 67, 188, 299, and 473 Hz. For all types of signal, the relationship of the levels of second and third harmonics is similar, and it can be observed in current as well. For a frequency of 473 Hz the amplitude of oscillation is small, and levels of harmonics are close to the level of noise floor.

Figure 11 shows the variability of THD value for different levels of excitation and frequencies. Acoustic signal and current have a similar character, but THD of current reach maximum for resonance frequency response, because the fundamental frequency is smallest in the resonance state.

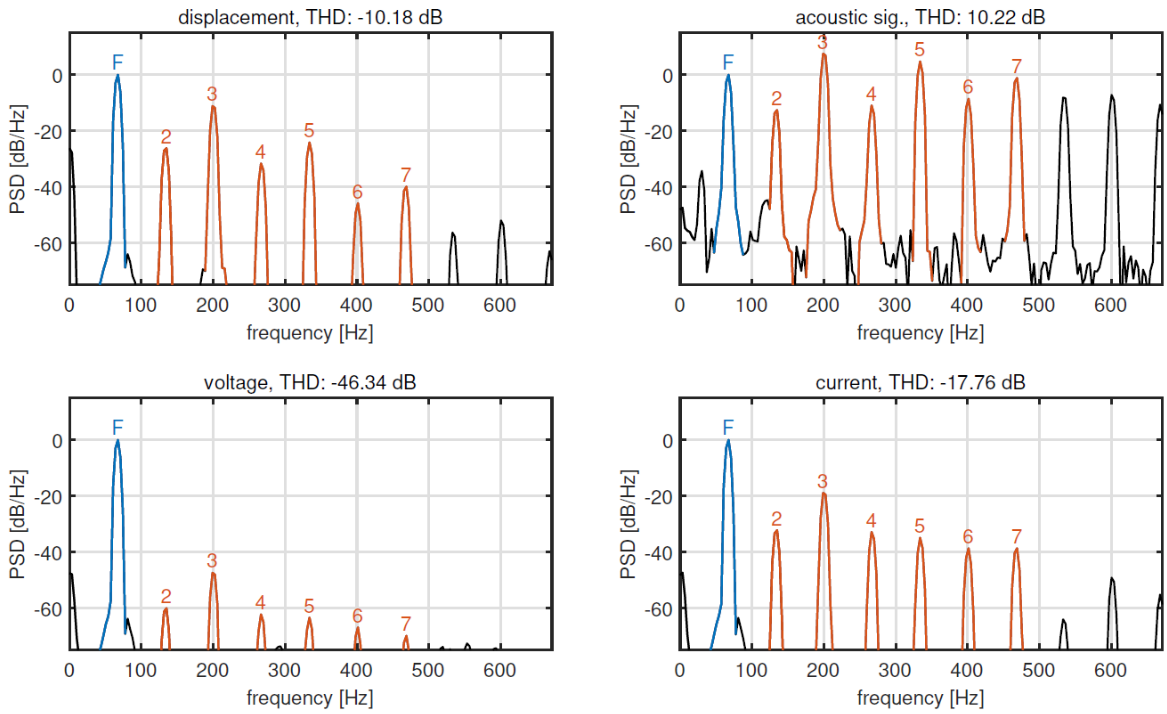


**Figure 7.** PSD of recorded signals: displacement, acoustic, voltage, current. Frequency 67 Hz, level 0 dB.

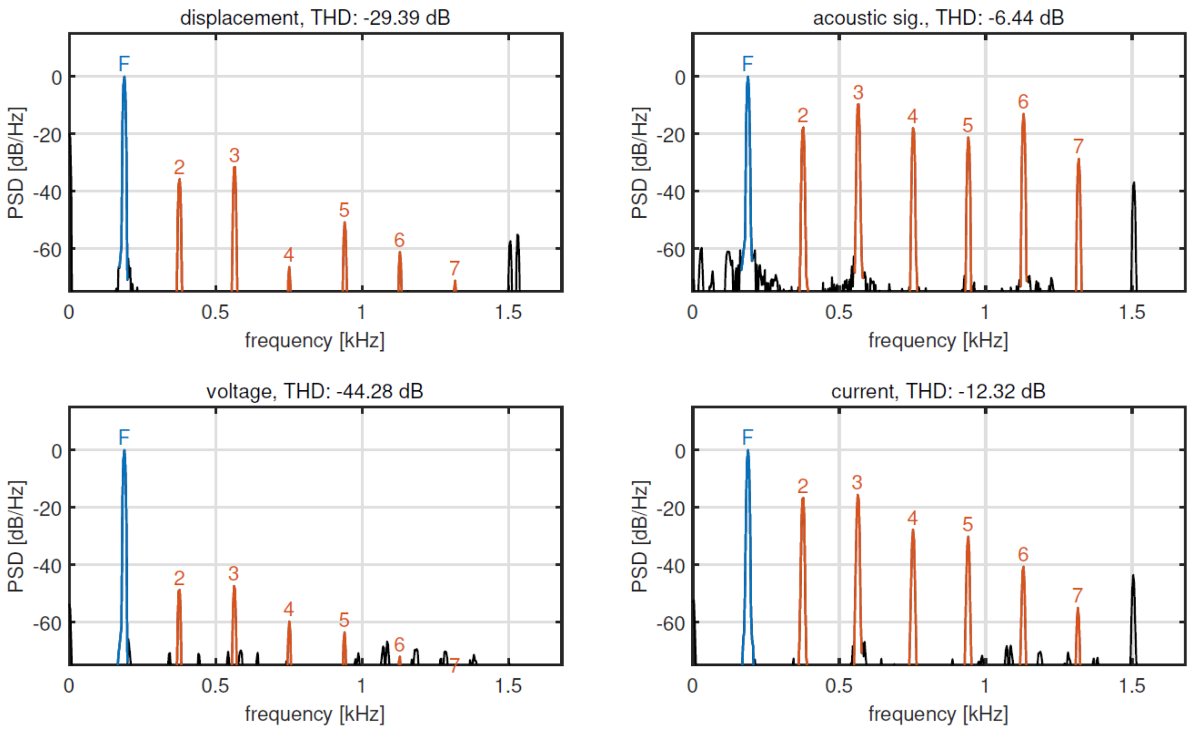


**Figure 8.** PSD of recorded signals: displacement, acoustic, voltage, current. Frequency 188 Hz, level 0 dB.

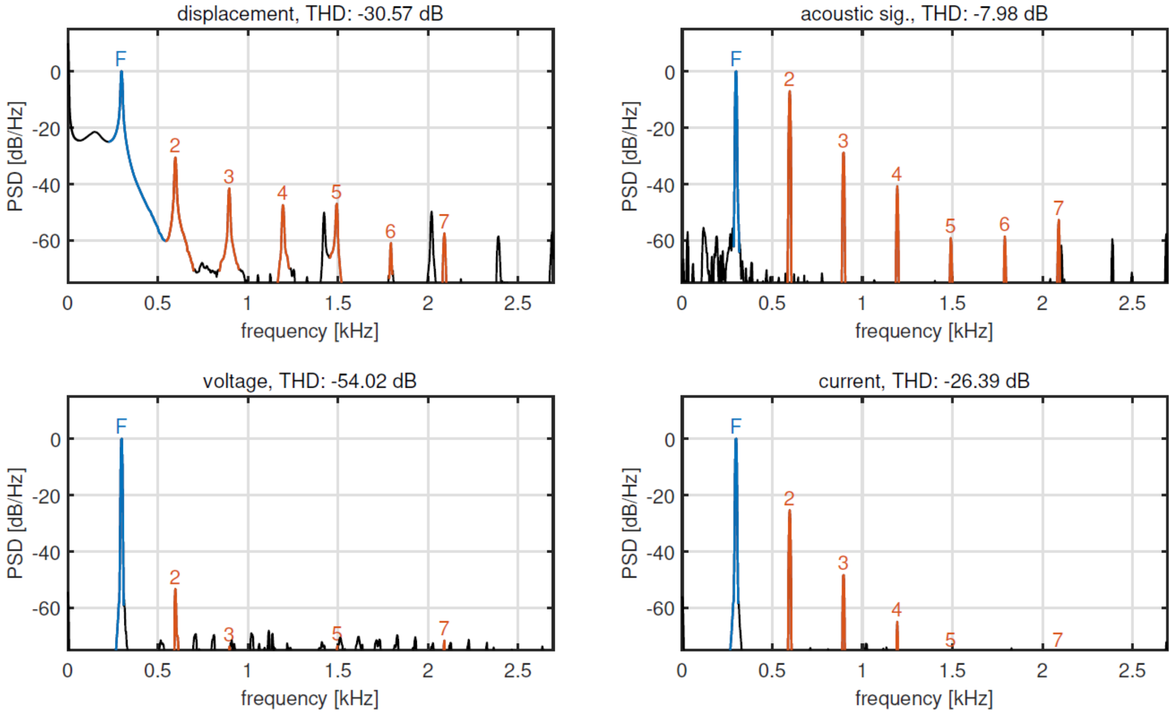


**Figure 9.** PSD of recorded signals: displacement, acoustic, voltage, current. Frequency 299 Hz, level 0 dB.

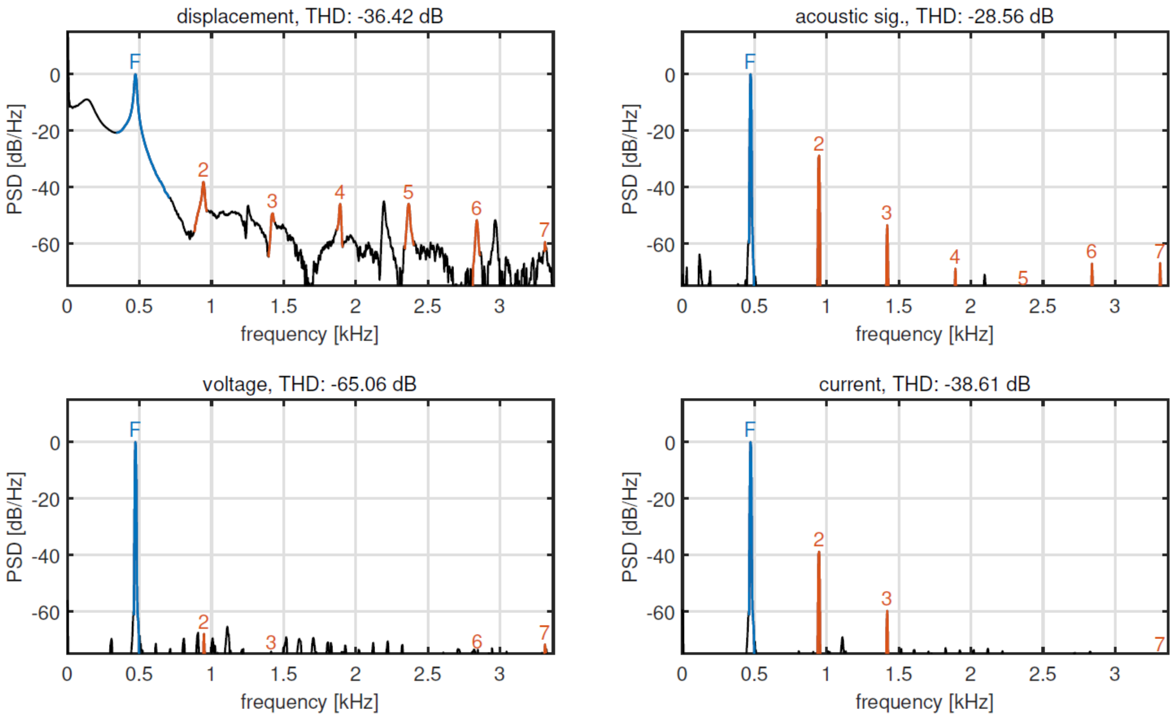


**Figure 10.** PSD of recorded signals: displacement, acoustic, voltage, current. Frequency 473 Hz, level 0 dB.

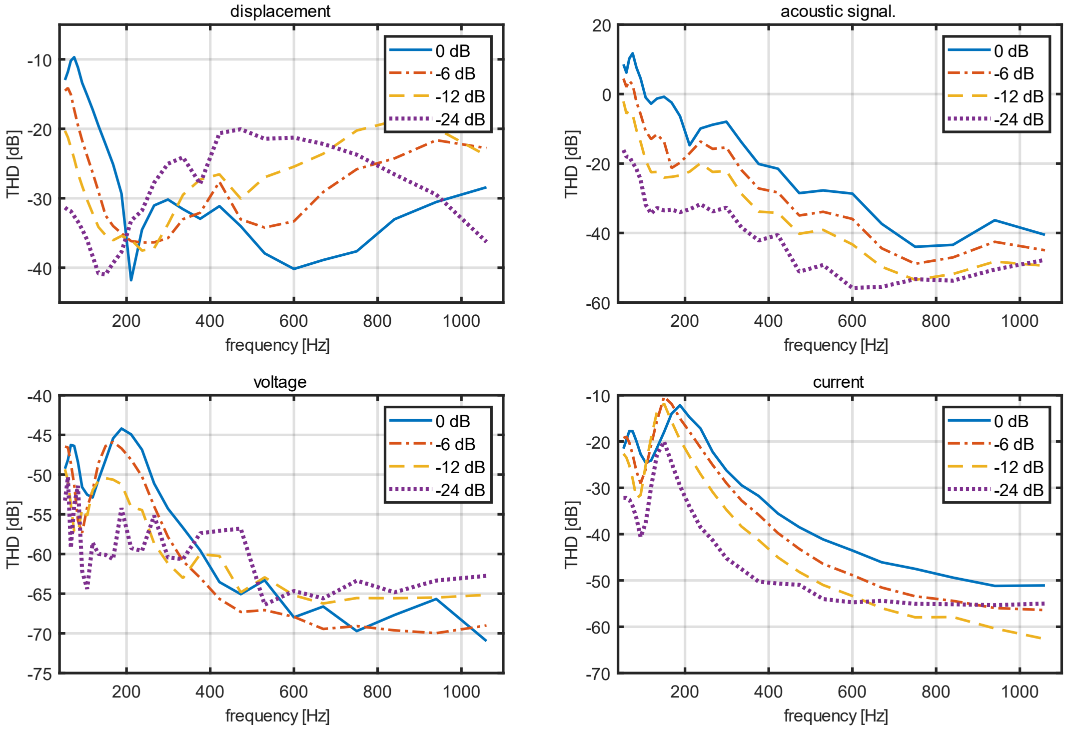


**Figure 11.** THD determined for 7 harmonics. THD is given in dB, frequency ranges from 53 to 1060 Hz, levels of excitation: 0, −6, −12, −24 dB.

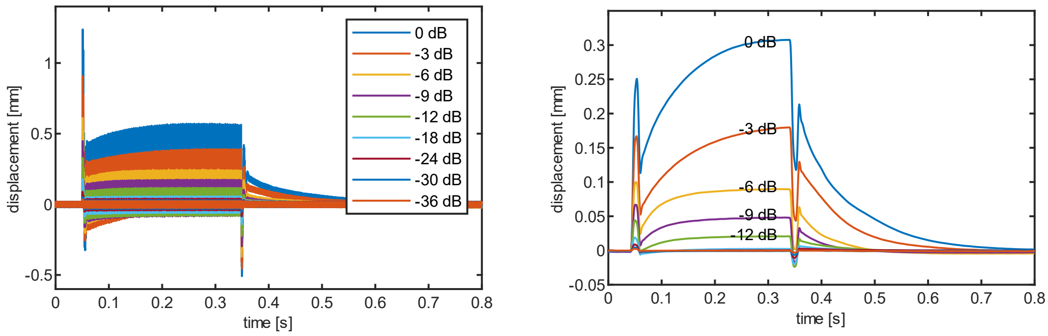


**Figure 12.** The displacement waveform for the frequency of 750 Hz, left: without low-frequency filtering, right: obtained by use of moving mean.

## 5. Conclusions

The laser displacement sensor works well in the range of low frequencies and slowly changing displacements, where the acoustic velocity is small but the displacement is relatively high. As can be seen in Figure 5, the slow build-up of displacement caused by reluctance force cannot be observed in acoustic signal or current. This displacement of the vibration midpoint is important for the linear range of the diaphragm movement, as it introduces asymmetry to the force action. It is possible to observe the level of distortion by measuring harmonic content in the voice coil current, but the distortion analysis may become less accurate for frequencies close to the resonance frequency.

The laser displacement sensor is applicable for measuring distortion at frequencies lower than the resonance frequency, where the processing efficiency of the loudspeaker is notably lower compared to the band above the resonance frequency. For higher frequencies, the method based on the microphone is more appropriate. The authors of the papers [14, 15] use laser displacement sensors to determine the parameters used in the models. As presented in this paper, the measurement of signal containing components above the resonant frequency is not very accurate due to the insufficient resolution of the triangulation sensors. This is confirmed by the waveforms recorded in Hernandez et al. work [14]. The main limitation of the displacement measurement method is the real resolution of the sensor. The displacement amplitudes for higher frequencies are smaller; as a result, the value of the displacement signal-to-noise ratio is lower for higher frequencies, leading to an overstatement of the value of harmonic components. Above the resonance frequency, the level of distortion increases with decreasing signal level and increasing frequency, opposite to distortions in acoustic signal. For frequencies higher than resonance, the THD characteristics for acoustic and current signals exhibit similarities. However, at the resonance frequency, the current component achieves a minimum value, while displacement maintains a high amplitude. The products of displacement distortions contribute significantly to the current, resulting in the current THD characteristic reaching a maximum value. These phenomena can be observed in Figure 11.

Regarding the issue of area of applicability mentioned in the introduction, the method based on the microphone is the best method for frequency above resonance. The method based on measurement displacement provides good results for frequencies below resonance, where the sensitivity of transducers is poor. Similarly to a microphone or displacement sensor, the method based on the current of the voice coil does not ensure accurate information about the level and pattern of distortion, but it does not require additional technical solution. The amplifier can be easily equipped with current measurement facilities, it does not require additional units in the loudspeaker cabinet. Therefore, the authors propose to use voice coil current, as source information about the state of the loudspeaker.

Based on studies by Pillonnet et al. [21], it was demonstrated that changing the method of supplying voltage to current can reform the processing process. In this paper, we show that the acoustic and displacement products of the distortion are correlated with current distortions. This leads to conclusion that the signal of current contains information about distortion. Such information can be used to filter the signal in a way that, while maintaining the amplifier's configuration as a voltage source, reduces distortions occurring in the current.

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### Additional information

The author(s) declare: no competing financial interests and that all material taken from other sources (including their own published works) is clearly cited and that appropriate permits are obtained.

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