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MEASUREMENT OF MICROPHONE HARMONIC DISTORTION USING PREDISTORTION TECHNIQUE

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ABSTRACT

Measurement of microphone nonlinear behavior has always been a challenging task due to a lack of linear acoustic sources. A recently developed technique that corrects a distorted output signal of an excitation device using predistortion of its input signal is used in this work. Thanks to this technique, a spectrally clean sinusoidal acoustic pressure is generated inside a closed box in which the microphone under test is placed together with a low-distortion reference microphone. The goal of this paper is twofold. First, show that the direct measurement of distortion is possible even for high-pressure levels; second, compare the harmonic distortion of several types of microphones such as studio electrostatic and electrodynamic microphones, MEMS microphones and laboratory microphones.

Keywords: *microphones, measurement, distortion*

1. INTRODUCTION

The quality of microphones that are used for audio industry such live production, or sound recording, may vary a lot depending on the type, i.e., the method they use to convert the air pressure variations of a sound wave to an electrical signal, and on the design and technology approach used for their fabrication [1]. Nonlinear distortion, one of the factors that influence their overall quality, is very often overlooked for two reasons. First, it is often considered very low, and thus negligible; second, it is difficult to measure. This paper shows how to measure the harmonic distortion of microphones directly and compares the distortion of several types of microphones used in the audio engineering industry.

The measurement of microphone non-linearities can be done either indirectly, by measuring the intermodulation distortion using two independent acoustic sources [2–4], or directly, by measuring the harmonic distortion using a single acoustic source. The direct method requires a reference microphone and a loudspeaker (or similar acoustic source) whose non-linear distortions are much smaller than that of the microphone under test. While low-distortion microphones (low-sensitivity condenser microphones or piezoresistive microphones) are available, low-distortion loudspeakers are much more challenging to make, especially at high sound pressure levels [5]. One solution to achieve high-pressure levels while keeping the source distortion low is to use a system of small cavities and standing wave tubes [6, 7]. Such a system can be usually used at low frequencies up to e.g, 200 Hz [8, 9], or is tuned to a specific frequency (e.g., 500 Hz) at which the pressure is maximized [7]. At this frequency, the influence of the non-linear behavior of the loudspeaker is reduced by a high acoustic impedance, limiting displacement-dependent non-linearities, and by an anti-resonant response of the tubes near the higher harmonics. This approach, limited for a given frequency, is also used in a commercially available product [7, 10].

This work uses a recently developed harmonic correction method [11] that corrects a harmonic distortion caused by an excitation device and cancels its harmonics to the background noise level. This adaptive method is used in this work to provide pure sinusoidal excitation, measured by low sensitivity, low distortion, small size reference microphone, and to allow direct measurement of the harmonic distortion of the microphones under test at different sound pressure levels and frequencies.

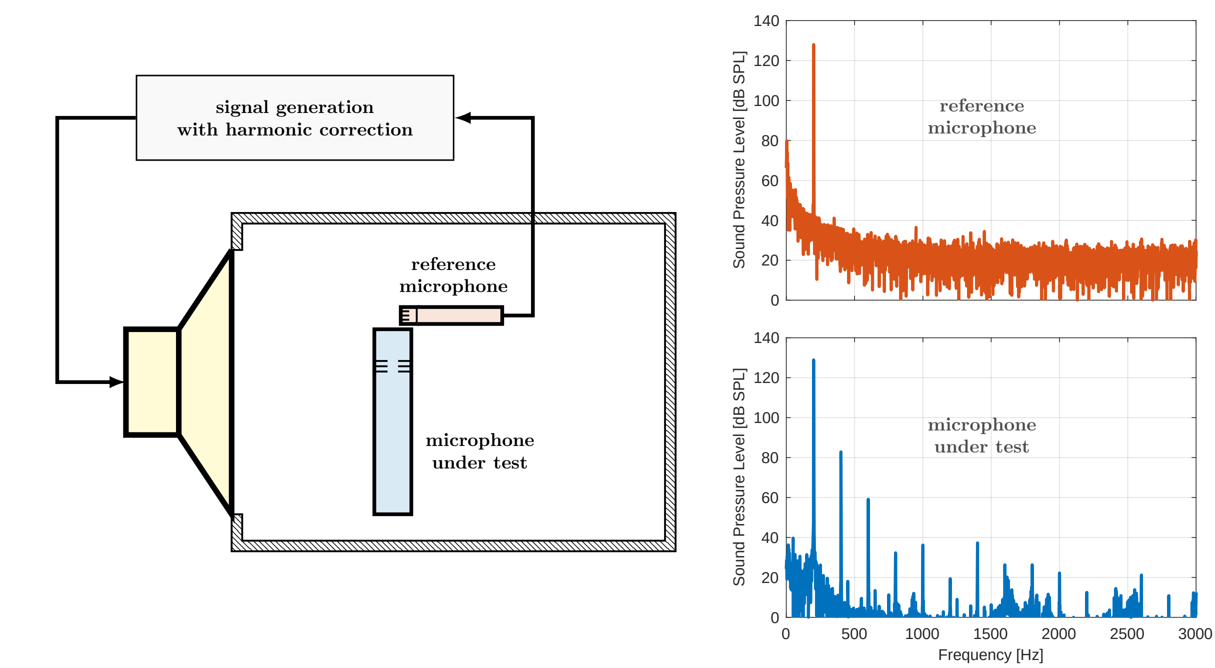


Figure 1. Schematic representation of the method (on the left) with spectra of the reference microphone (on the right above, in red) and the microphone under test (on the right below, in blue). Note, that the spectra of the reference microphone consists of a pure sine wave with higher harmonics suppressed to the noise level.

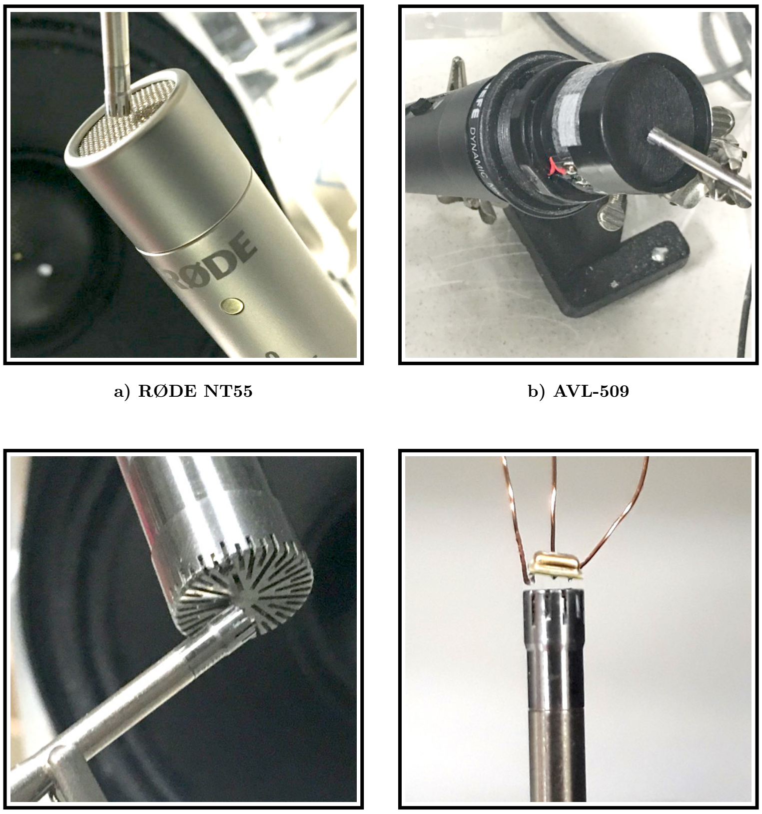
The method has been tested successfully on MEMS microphones [12].

The harmonic distortion is measured on four different microphones. These microphones and the measurement setup are presented in section 2. Next, the harmonic correction method is presented in section 3. The graphs illustrating the dependence of the sound pressure levels of the higher harmonics on the sound pressure level measured with the reference microphone are presented for different frequencies in section 4.

2. MEASUREMENT SETUP

The measurement setup is schematically depicted in Fig. 1. The microphone under test and the reference 1/8" Pressure Microphone GRAS 40DP are placed in a sealed box of dimensions [cm] (18.7 x 18.6 x 19). An 8" loudspeaker, together with an off-the-shelf amplifier, provides the pressure excitation inside the box. A preamplifier GRAS Type 26AC and a conditioning amplifier Brüel&Kjær Type 2609 are used to ensure proper handling of the reference microphone signal. The signal generation and acquisition are provided using an RME Fireface 400 sound card.

Next, a sinusoidal signal is used to measure the harmonic distortion of each of the four tested microphones at two frequencies (200 Hz, 2 kHz) and different sound pressure levels going from 90 to 130 dB SPL with a 2 dB step. A harmonic correction [11], whose details are explained in section 3, is applied to ensure a pure harmonic acoustic pressure inside the box. Consequently, the pressure signal measured with the reference microphone contains no higher harmonics (suppressed to the noise level) as illustrated in Fig. 1. Note also, that the reference 1/8" Pressure Microphone GRAS 40DP is a low sensitivity microphone with negligible distortion (below noise-floor level) at sound pressure levels below 130 dB SPL. For the sake of comparison, the voltage signal at the output of the microphone under test is recalculated to the equivalent sound pressure level using its sensitivity. (see Fig. 1).



a) RØDE NT55 b) AVL-509

c) GRAS 46AF d) TDK ICS-40180

Figure 2. Pictures of each microphone under test inside the box. The reference microphone is placed as close as possible (not touching) to the microphone under test.

Four different microphones are measured: (a) a 1/2" RØDE NT55 condenser 'pencil' studio microphone, (b) an AVL-509 dynamic studio microphone, (c) a 1/2" GRAS 46AF laboratory condenser microphone, and (d) a MEMS microphone TDK ICS-40180. The picture of these microphones placed next to the reference microphone during measurement are shown in Fig. 2.

3. CORRECTION OF SOURCE DISTORTION

An important part of the direct measurement of the harmonic distortion of microphones is the linearity of the source and the linearity of the reference microphone. While the linearity of the reference microphone can be ensured by choosing a microphone with low sensitivity and low distortion, the linearity of the sound source (a loudspeaker) is much more difficult to achieve.

A recently developed method [11] can correct the harmonic distortion caused by such a sound source and cancel its harmonics into the background noise. The source is excited by a periodic signal, the harmonic distortion obtained using the reference microphone is analyzed in the frequency domain, and the source signal is modified so that the harmonic distortion measured using the reference microphone decreases. This section describes the main steps of the harmonic correction method. For a more detailed explanation, please see [11] or an online document with video demonstrations at `https://ant-novak.com/pages/predist_meas/`. [13].

Let us consider a loudspeaker (a nonlinear system) that excites an enclosed box, as depicted in Fig. 1. The pressure at the reference microphone is noted $p(t)$. The loudspeaker is driven by an electrical signal $u(t)$ generated by a data acquisition device (a sound card) and an amplifier. For a linear loudspeaker, if the input voltage signal $u(t)$ is sinusoidal, the output pressure $p(t)$ is also a sinusoidal signal of the same frequency. However, if the loudspeaker is nonlinear, the phenomenon of harmonic distortion may occur: additional components corresponding to integer multiples of the fundamental input frequency are then generated in the pressure signal $p(t)$.

The basic idea of harmonic correction is to inject harmonic components into the voltage signal $u(t)$ signal so that undesirable harmonic components of the pressure signal $p(t)$ disappear. If the loudspeaker behaves quasilinearly at low amplitudes and becomes nonlinear at higher levels, which is the case for most loudspeakers, the algorithm of harmonic correction can be described by the next three steps.

- First, we estimate the loudspeaker frequency-dependent pressure sensitivity; in other words, the linear FRF (Frequency Response Function) between the pressure and the voltage. This measurement is done using a low-amplitude (low enough to avoid distortion) and large-band signal, such as a multi-tone signal. This FRF is used in the following steps to estimate the amplitude and phase of the harmonics to be injected into the input signal.
- In the second step, a sinusoidal signal $u(t)$ is generated at the input of the loudspeaker, and the pressure signal $p(t)$ is acquired. A sequence of signal $p(t)$ (e.g., 4800 samples) is used to calculate the Fourier series coefficients using FFT (Fast Fourier Transform) algorithm. The amplitude and phase of higher harmonics of the signal $p(t)$ are then used to predict which harmonics (amplitude and phase) should be injected into the signal $u(t)$ for the next iteration. This is done using the linear FRF estimated in the first step of the procedure. The higher harmonics to be injected are calculated as the ratio of the harmonics of signal $p(t)$ and the FRF at the particular frequency of each harmonic considered. This ratio is then subtracted (added with the opposite phase) from the signal $u(t)$.
- Finally, the harmonic correction algorithm is computed recursively over successive non-overlapping sequences (of the same number of samples) of $p(t)$. Each time, the ratio between the remaining unwanted harmonics of the signal $p(t)$ and the FRF is subtracted from the signal $u(t)$ until undesirable harmonics are reduced to the level of the background noise.

4. RESULTS

The four selected microphones are measured using the measurement setup described in section 2. The harmonic distortion of each microphone is measured for two different frequencies (200 Hz and 2 kHz) and for input sound pressure levels between 90 and 130 dB SPL. The equivalent sound pressure level of the second harmonic is depicted in Fig. 3 and the third harmonic in Fig. 4. The levels of these harmonics vary a lot from microphone to microphone.

The strongest distortion exhibits the MEMS microphone (violet triangle marks in Figs. 3 and 4). Its 2nd harmonic (Fig. 3) reaches 92 dB SPL for the input level of 124 dB SPL and saturates for input sound pressure levels exceeding 125 dB SPL. Both 200 Hz and 2 kHz excitations lead to very similar results. Third harmonic (Fig. 4) is hidden under the noise-floor of 40 dB SPL up to the sound pressure level of the excitation signal 115 dB SPL and than increases to reach 83 dB SPL and 92 dB SPL for 200 Hz and 2 kHz respectively, for 124 dB SPL excitation.

The 1/2" GRAS 46AF laboratory condenser microphone shows the lowest distortion out of the four tested microphones (yellow plus marks in Figs. 3 and 4). For both 200 Hz and 2 kHz measurements the 2nd harmonic does not exceed 80 dB SPL, and the 3rd harmonic does not exceed 55 dB SPL for 130 dB SPL excitation.

The 1/2" RØDE NT55 condenser 'pencil' studio microphone shows also a quite low distortion (blue x-marks in Figs. 3 and 4). Its 2nd harmonic distortion is approximately 9 dB higher compared to the 1/2" laboratory microphone but 15 dB below the MEMS microphone. The 3rd harmonic does not exceed 60 dB SPL for 200 Hz excitation and is hidden in noise-floor for 2 kHz.

Finally, the AVL-509 dynamic studio microphone (red circle marks in Figs. 3 and 4), exhibit low distortion for 200 Hz, but much higher distortion for 2 kHz. The 2nd harmonic at 200 Hz is as low as the distortion of the 1/2" laboratory microphone, while for 2 kHz it is as high as the distortion of the MEMS microphone with saturation near 120 dB SPL. The third harmonic is very low for 200 Hz (close to the distortion of 1/2" RØDE studio microphone), but very high for 2 kHz even for lower sound pressure levels between 110 and 120 dB SPL where the 3rd harmonic

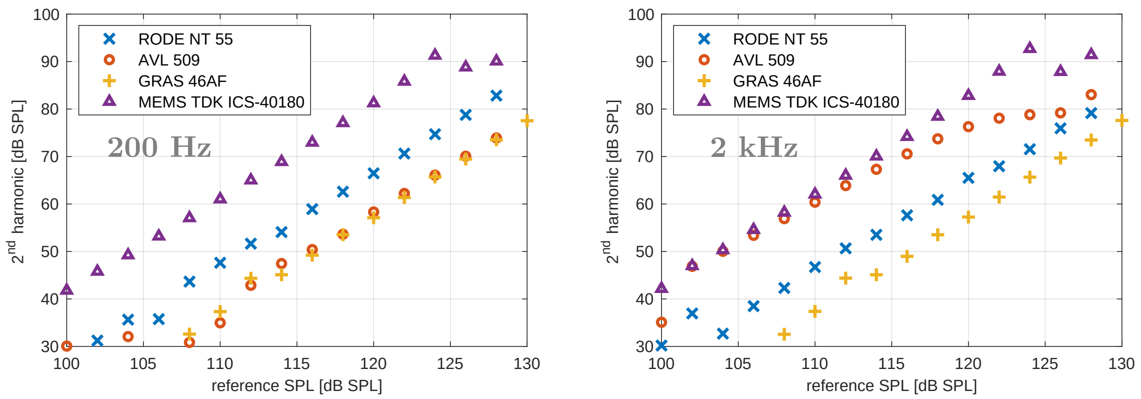


Figure 3. Sound pressure level of the second harmonic component as a function of the sound pressure level measured by the reference microphone (left: 200 Hz excitation, right: 2 kHz excitation).

is between 50 and 70 dB SPL.

5. DISCUSSION

The measurement technique described and used in this paper requires only an ordinary loudspeaker, a sealed enclosure to maximize the sound pressure, a microphone with low sensitivity and low distortion, and a sound card. The main disadvantage that usually accompanies the measurement of nonlinear microphone distortion, i.e., the nonlinear behavior of the source (loudspeaker), is overcome by a signal processing technique that cancels out the harmonic distortion caused by the loudspeaker.

However, there is another disadvantage related to the distance between the reference microphone and the microphone under test. For the measurement of low frequencies (e.g., 200 Hz), the sound pressure distribution inside the sealed box is almost uniform, and the pressure measured by the reference microphone is the same as the sound pressure captured by the tested microphone. At higher frequencies (e.g., 2 kHz), the assumption of equal pressure inside the box is false due to the modal behavior inside the box. Not only can the sound pressure level be different at the two microphone positions, but the harmonic correction algorithm may also not work correctly. In some situations, when one or more harmonics fall to the local minimum of the pressure distribution at the reference microphone position, these harmonics may not be fully corrected.

The MEMS microphone can easily be placed very close to the diaphragm of the reference microphone, minimizing the distance between the two membranes. However, for other microphones tested in this article, for which the position of the membrane is not well determined or not easily accessible, the high-frequency results may be corrupted by this modal phenomenon.

Future work will focus on this issue. A smaller enclosure would allow the shifting of the problem to higher frequencies. However, it may be difficult to fit the microphones under test at a smaller volume because of their size. Another possible solution to overcome this problem is a free-field solution using powerful sounding speakers.

6. SUMMARY

To summarize, this work shows that a direct measurement of the harmonic distortion of the microphone is possible using an ordinary electrodynamic loudspeaker, sealed box, and a low sensitivity low distortion reference microphone. The results show that the MEMS microphone has a much higher distortion than other microphones. The lowest distortion is provided by the laboratory measurement condenser microphone. The distortion of the studio dynamic microphone exhibit an important frequency dependency. For the low frequency measurement (200 Hz), the dynamic microphone exhibits a low distortion, while for high frequencies (2 kHz) its distortion is much higher.

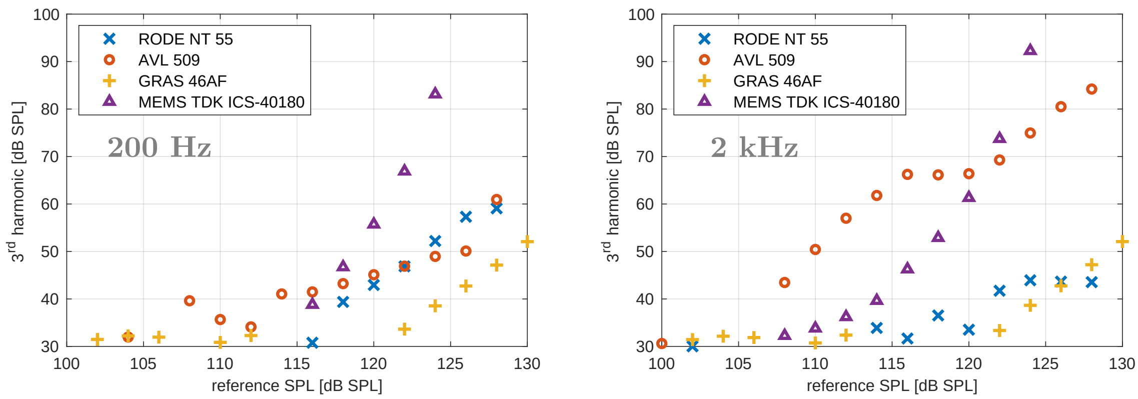


Figure 4. Sound pressure level of the third harmonic component as a function of the sound pressure level measured by the reference microphone (left: 200 Hz excitation, right: 2 kHz excitation).

7. ACKNOWLEDGMENTS

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