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A fast and accurate adaptive-filter method to measure directivity of loudspeakers

Shun Nakamura[1], Gerald Enzner[1], Ali Fallah[1,2], and Steven van de Par[1,2]

[1]*Department of Medical Physics and Acoustics, Carl von Ossietzky Universität Oldenburg*
[2]*Cluster of Excellence "Hearing4All"*

Correspondence should be addressed to Shun Nakamura (shun.nakamura@uol.de)

ABSTRACT

In audio engineering, the speed and precision of loudspeaker directivity measurement is intrinsic to product development. Conventional methods for individual measurements are time consuming and may delay progress in loudspeaker design and evaluation. Our novel approach speeds up this process without sacrificing accuracy, using standard lab equipment of existing configurations. By utilizing a continuously rotating loudspeaker, our method quickly generates a full 360-degree directivity map, significantly reducing the conventional time required for comprehensive assessments. Our method uses the Normalized Least Mean-Square (NLMS) adaptive filter algorithm to identify rotating impulse responses and to reject turntable noise. In doing so, it delivers nearly infinite spatial resolution in the azimuth direction, thus offering detailed insights into loudspeaker performance across angles and frequencies. Extensive analysis with two fundamentally different loudspeaker devices confirms our method's effectiveness, allowing for rapid directivity evaluations in about a minute, which is a drastic improvement over the hours needed with traditional approaches. This advancement facilitates quicker product iterations and design, leading to enhanced acoustic quality in audio products from smart speakers to mobile devices.

1 Introduction

For measuring the quality of loudspeakers, relying solely on data from the forward axis is insufficient. Sound is radiated in all directions from the loudspeaker and the characteristics from the sides and back contribute to the entire sound performance. The Consumer Electronics Association's Standard Methods of Measuring Indoor Speakers (CEA-2034) defines the Spinorama test, which generates data predictive of how speakers might perform in a typical listening environment [1]. The Directivity Index (DI) of sound power, which quantifies the directionality of the loudspeaker, is crucial for estimating in-room frequency responses, closely correlating with consumer subjective preferences [2]. The Spinorama test requires frequency response measurements in 10° increments, both vertically and hori-

zontally, meaning a single loudspeaker must undergo 72 measurements to cover merely the horizontal and the median plane. Such extensive testing is laborious and time-consuming, particularly when modifications to the product necessitate repeated measurements.

Furthermore, the development of personal and portable audio devices such as smartphones, tablets, notebooks, studio monitors, multimedia equipment, and professional PA equipment necessitates precise and detailed 3D-field directivity data, which advanced 3D measurements may provide [3]. However, capturing a complete sphere at 1° angular resolution (for holography) demands an excessive 64,800 measurement points and at least 90 hours measurement time if only 5 seconds are considered per angle. Such exhaustive sampling is essential to accurately depict the directivity characteristics, especially for devices with complex acoustic outputs that significantly vary with direction. Despite this necessity, in practice, the total number of measurement points must be significantly reduced to keep the total measurement time acceptable.

Our study introduces a novel measurement method that significantly expedites the loudspeaker directivity assessment without sacrificing accuracy. By employing a continuously rotating loudspeaker, our method efficiently captures a continuous 360-degree directivity map in the azimuth direction, significantly reducing the measurement time by eliminating the traditional stop-and-go process and the need for direction stabilization in-between measurements. To accurately track the impulse responses at each angle from a rotating loudspeaker to a measurement microphone, we employ the NLMS adaptive filter algorithm, which effectively yields quasi infinite spatial resolution. This method not only streamlines the measurement process, but the result can also provide a more nuanced understanding of a loudspeaker's performance across the angles. The resulting data can be instrumental for rapid prototyping and iterative testing, thereby facilitating quicker advancements in loudspeaker design. We conducted comprehensive testing on two loudspeaker models, including the Dynaudio Core-7 studio monitor and a custom omni-directional loudspeakers (COD) developed in our lab [4], demonstrating that our method can deliver the respective directivity assessments in approximately one minute measurement time compared to the hours required by conventional methods.

The challenge of measurement time is similarly encountered in the acquisition of individual head-related transfer functions (HRTFs), which traditionally required extensive measurement time and patience, often amounting to several hours [5]. Recent advancements with continuously rotating apparatus have significantly reduced time consumption [6]. The NLMS algorithm was then used to extract the HRTF from continuously rotating measurements of probe noise played from single- or multiple fixed loudspeakers [7, 8].

2 Conventional Loudspeaker Measurement Methods

Generally in acoustic measurement and more specifically to evaluate the performance characteristics of loudspeakers, the methods with Maximum Length Sequences (MLS) or Sweep Signals are widely employed [9] to capture frequency responses and directivities at discrete angles. The Fourier transform is applied in both cases to convert the time-domain measurement signals into the frequency domain, where the division of output and input spectrum delivers frequency responses to characterizes the system's acoustical properties. If required, the result is inversely transformed back to the time domain to deliver impulse responses (IR).

2.1 Maximum Length Sequences (MLS)

The MLS is a pseudo-random fullband binary sequence for efficient capture of wideband IR. The measurement with MLS can be beneficial for quick and reliable results. However, it is sensitive to the nonlinearity of loudspeakers (or generally the measured transfer functions) which may arise at higher output levels [10]. Adjusting for a good balance of high signal-to-distortion ratio (SDR) and high signal-to-noise ratio (SNR) may significantly complicate the measurement process.

Mathematically, the impulse response $h[n]$ using MLS can be computed via simple correlation, leveraging the ideal autocorrelation properties (i.e., constant spectrum) of MLS and thus avoiding spectral division:

$$h[n] = \mathrm{IFFT}\left(\mathrm{FFT}(y[n]) \cdot \mathrm{conj}(\mathrm{FFT}(x[n]))\right) \quad (1)$$

where $x[n]$ is the MLS signal, $y[n]$ is the recorded measurement signal from the speaker, and $\mathrm{conj}(\mathrm{FFT}(x[n]))$ denotes the complex conjugate of the Fourier transform of the MLS signal. The sequence perfectly correlates with itself at lag zero, such that $|\mathrm{FFT}(x[n])|^2 = 1$ and therefore spectral division $\mathrm{FFT}(y[n])/\mathrm{FFT}(x[n]) = \mathrm{FFT}(y[n]) \cdot \mathrm{conj}(\mathrm{FFT}(x[n]))$ appears as multiplication.

2.2 Sweep Signals

Sweep signals are valued for their ability to eliminate harmonic distortion from an acoustic measurement. Using sweep signals with exponentially increasing frequency further takes into account the requirement of more observation period needed for low frequencies and therefore enhances the accuracy of its deconvolution from the recorded measurement signal [11].

The exponential sine sweep mathematically is

$$x(t) = \sin\left(2\pi f_1 T\left(e^{\frac{t}{T}\ln\left(\frac{f_2}{f_1}\right)} - 1\right)\right) \tag{2}$$

where t is time, T is its total duration, and f_1 and f_2 are the start and end frequencies of the sweep, respectively. The desired impulse response $h(t)$ of the measured acoustic system is calculated by deconvolving the recorded measurement signal $y(t)$ with the inverse of the sweep signal $x(t)$ in the frequency domain:

$$h(t) = IFFT\left(\frac{FFT(y(t))}{FFT(x(t))}\right) \tag{3}$$

Because of the actual narrowband measurement taking place for each frequency at each time of the sweep duration, the harmonic distortion of the measured system will be displaced from the current frequency and also from the desired linear part of the resulting impulse response, allowing for precise segregation.

3 Continuous Measurement Setup

The setup of the proposed directivity continuous measurement method for loudspeakers is shown in Figures 1 and 2 from different perspectives of orientation. The speaker to be measured is placed on a turntable. In the case of a conventional loudspeaker, its front baffle is perpendicular to the reference axis pointing towards the measurement microphone in order to define 0° azimuth angle of the setup [1]. A turntable speed between 30 and 120 seconds for a single 360° revolution of the loudspeaker is considered. The turntable then rotates about the axis perpendicular to the reference axis, which refers to continuously varying azimuth θ of the setup. In order to ramp to constant speed of the turntable, the measurement should be started with the turntable in a slightly negative angular position. Using a sensor or other means, we can then define 0° and 360° as starting and ending points of a uniform measurement process. White noise is employed here as the excitation signal $x(n)$ for probing the changing angles fullband all the time during the measurement. One or more microphone recordings $y(n)$ can be acquired simultaneously depending on the desired number of elevations.

3.1 Mathematical System Model

The recorded signal $y(n)$ at discrete time n is described by means of a linear convolution model of the excitation signal $x(n)$ and the time-varying impulse response $h(m,\theta_n)$ at the momentary angle $\theta_n = \omega n T_s$, where $\omega = 2\pi/T_{360}$ is the angular frequency of the loudspeakers, T_{360} the duration of a 360° revolution, and $T_s = 1/f_s$ the sampling time interval. The symbol N represents the model length of the impulse response (IR) and η is any unavoidable observation noise:

$$y(n) = \sum_{m=0}^{N} x(n-m)h(m,\theta_n) + \eta(n) \tag{4}$$

3.2 Impulse Response Tracking by NLMS

The NLMS adaptive algorithm is used to track the impulse responses of the rotating loudspeaker. To express the computations, we denote excitation signal $x(n)$ and loudspeaker impulse responses $h(n,\theta_n)$ as vectors

$$\mathbf{x}(n) = [x(n), x(n-1), ..., x(n-N+1)]^T \tag{5}$$

$$\mathbf{h}(\theta_n) = [h(0,\theta_n), h(1,\theta_n), ..., h(N-1,\theta_n)]^T\ . \tag{6}$$

The rotating loudspeaker impulse responses are then estimated by the sample-wise update rule [12]

$$\widehat{\mathbf{h}}(\theta_{n+1}) = \widehat{\mathbf{h}}(\theta_n) + \mu\,\frac{e(n)\,\mathbf{x}(n)}{||\mathbf{x}(n)||^2} \tag{7}$$

$$e(n) = y(n) - \widehat{\mathbf{h}}^T(\theta_n)\mathbf{x}(n) \tag{8}$$

using the recorded signal $y(n)$, the excitation signal $\mathbf{x}(n)$, the previous estimate $\widehat{\mathbf{h}}(\theta_n)$ of the IR, and a suitable choice of the stepsize control factor $0 \le \mu \le 1$, typically $\mu = 0.5$ After few initial iterations, the rotating IR $\mathbf{h}(\theta_n)$ of the acoustic system is tracked and represented by the adaptive filter coefficients.

In doing so, the NLMS algorithm employs the error signal $e(n)$ between recording $y(n)$ and adaptive filter output $\widehat{y}(n) = \widehat{\mathbf{h}}^T(\theta_n)\mathbf{x}(n)$ mimicking the recording. Successful operation of NLMS is thus proved by error signal minimization. The error signal will then represent the unpredictable system noises, such as the

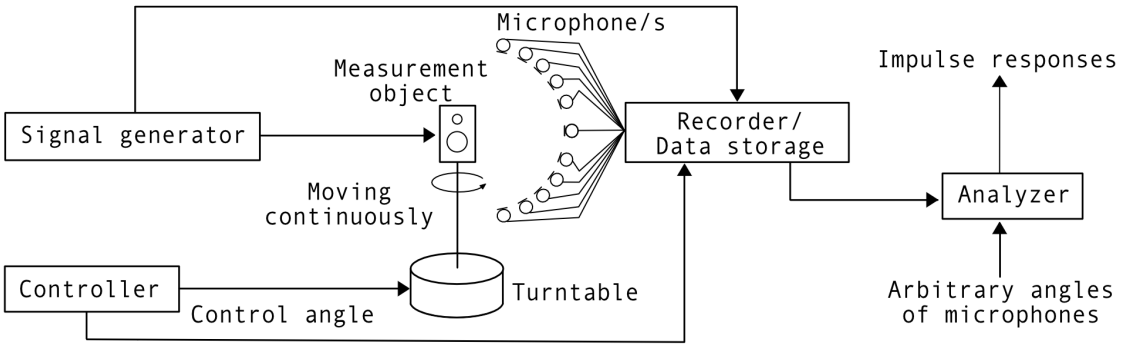


Fig. 1: Measurement setup (side view).

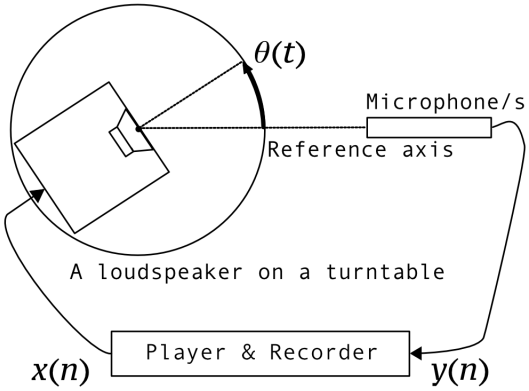


Fig. 2: Bird's-eye view of the measurement setup with a fixed microphone (actually multiple microphones) and a rotating loudspeaker.



Fig. 3: Measurement setup in anechoic chamber with the COD speaker developed by the Acoustic Group at Carl von Ossietzky Universität.

turntable motor, ambient noise, or nonlinear distortions separated from the impulse response representation.

For reproducibility of the proposed method, the procedures of continuous rotation and NLMS algorithm are illustrated in Figure 4 regarding the typical appearances of recording and error signal. We consider the two aforementioned loudspeaker models for which we can expect fundamentally different characteristics of the related signals. For the COD speaker, its omnidirectionality is here recognized easily from the constant envelope of the recorded signal as a function of time, whereas the Core7 clearly depicts attenuation when recorded from the rear in the middle part of the measurement. In both cases, the NLMS error is minimized in proportion to the recorded signal, which may indicate a nonlinear part of the loudspeaker signal.

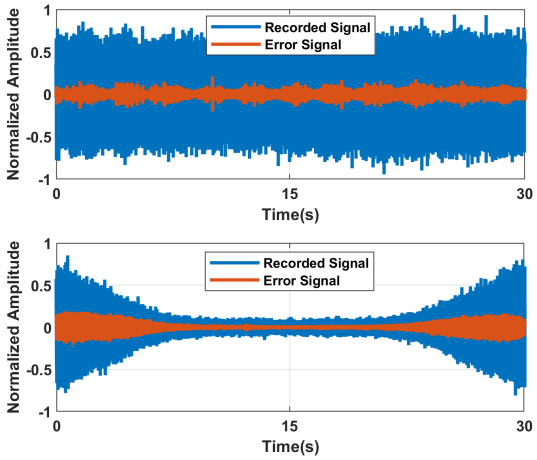


Fig. 4: Recorded signals and error signals for the COD (top) and Dynaudio Core7 speakers (bottom).

4 Results

As shown by the description of the NLMS algorithm in Section 3.2, the native result of the proposed measurement method are the acoustic system impulse responses from loudspeakers to measurement microphones. Therefore, in what follows, we initially present results to demonstrate the quality of the obtained impulse responses, then convert the impulse responses to frequency responses, and finally assemble the entire set of frequency responses obtained from all directions into directivity plots of both loudspeaker models under investigation. For reference, all these loudspeaker representations are in parallel obtained also with the sweep method under otherwise similar measurement conditions and rendered here for comparison.

4.1 Impulse Responses

Figure 5 presents impulse responses of the new (adaptive filter) and conventional (sweep) measurement methods applied to the COD speaker at discrete angles of 0° and 90°. Apart from strong similarity of new and conventional methods, the two angles demonstrate the expected omni-directionality of the speaker. The Core-7 speaker in Figure 6 as well shows very consistent impulse responses of new and conventional measurements at 0° and 90°. As expected for Core-7, we see a considerable directional dependence of the impulse response. Our choice of the adaptive-filter length was $N = 2000$ and the results are here truncated to 250 samples merely for presentation.

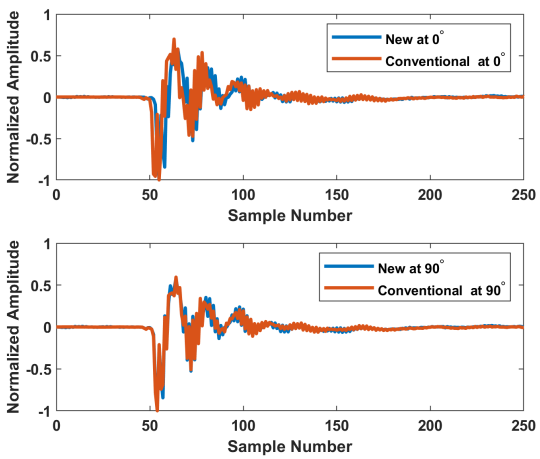


Fig. 5: Impulse responses from COD speaker at angles of 0° (top) and 90° (bottom).

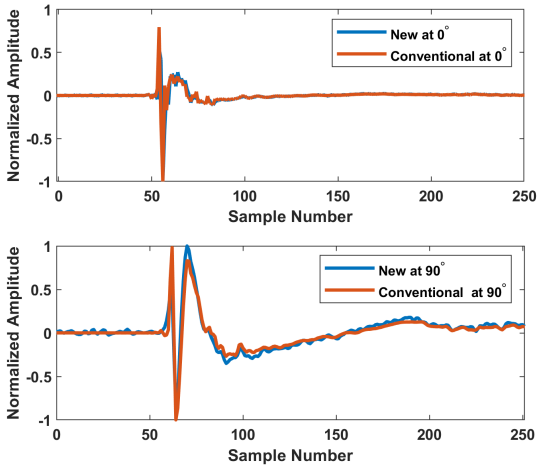


Fig. 6: Impulse responses from Core7 speaker at angles of 0° (top) and 90° (bottom).

4.2 Frequency Responses

Figures 7 and 8 provide the comparison of the corresponding frequency responses with remarkable consistency for both speakers and both directions, albeit the considerably different measurement methods. The spectral similarity includes the representation of low frequencies as well as the fine structure in the mid range except for a small glitch with Core7 at around 220 Hz. Further small differences indeed can be noticed at very high frequencies, but we may not attribute either method as right or wrong. Apart from the good agreement of new and conventional measurements, the frequency responses nicely indicate the converse focus of the loudspeaker designs for omni-directionality in Figure 7 and for flat spectrum on 0° axis in Figure 8.

4.3 Directivity Plots

Figure 9 finally renders a comparison of directivity maps with octave-band frequencies and A-weighting as per ANSI S1.42-2001 [13]. The similarity of new and conventional measurements is somewhat amazing in this representation for both loudspeaker models. Thereby, it should be noted that the conventional measurement of 360 discrete angles takes in the order of 2 hours time, while the new continuous recording is completed in just about a minute (i.e., raw measurement time, not including experimental setup and NLMS computations). Very similar outcomes of new and conventional measurements is the main result of this paper, but the measurement speed our striking advantage.

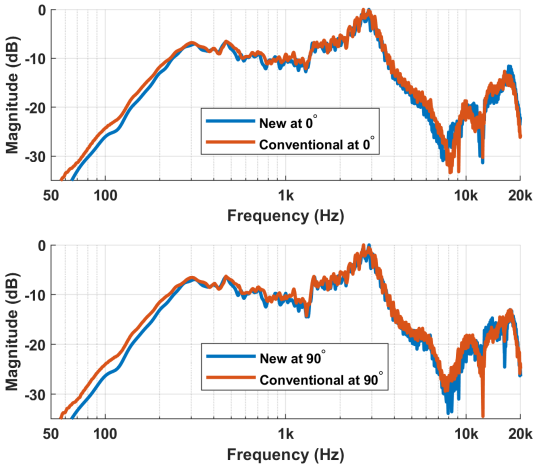


Fig. 7: Frequency responses from OMS speaker at angles of 0° (top) and 90° (bottom).

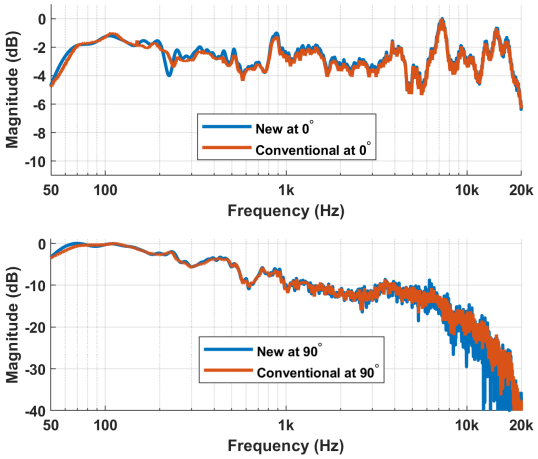


Fig. 8: Frequency responses from Core7 speaker at angles of 0° (top) and 90° (bottom).

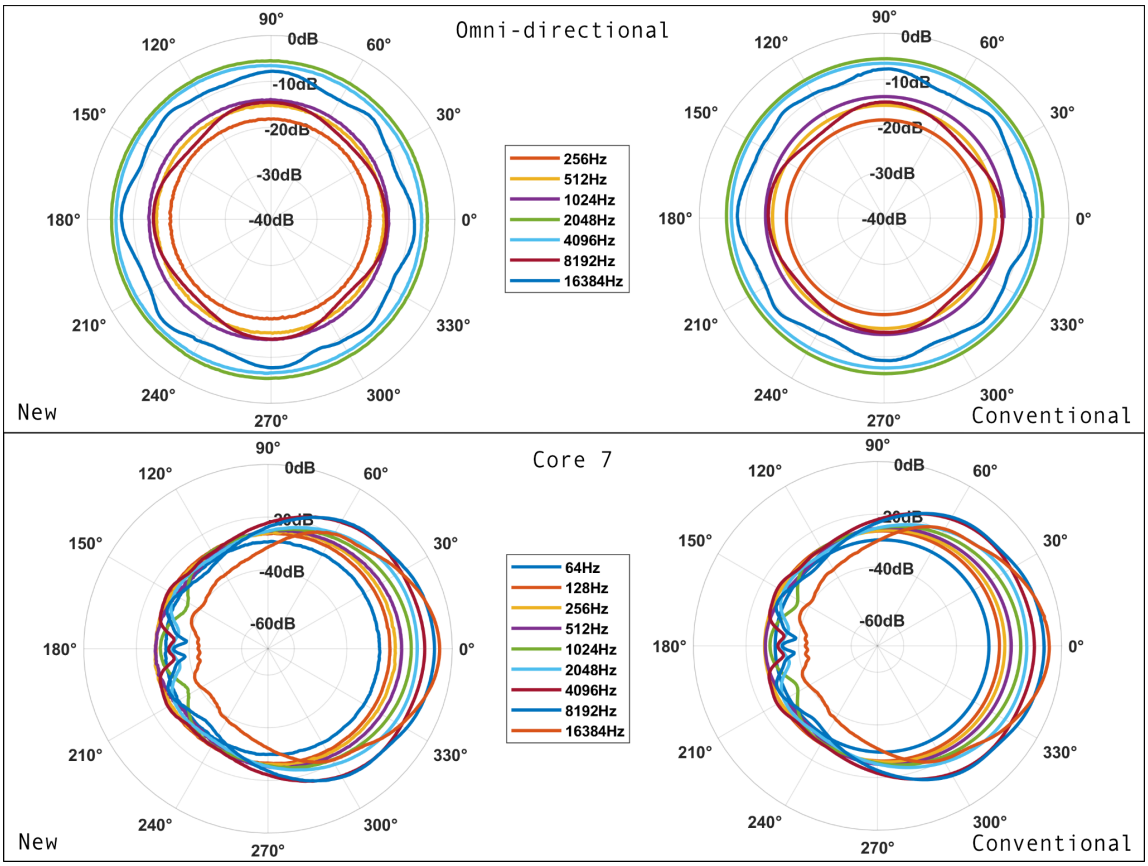


Fig. 9: Directivity plot with octave band representation of the Core7 and the COD speaker. Top left: Proposed fast continuous method. Top Right: conventional "stop-and-and go" method for 360 angles. Bottom left: proposed fast continuous method. Bottom right: conventional "stop-and-and go" method for 360 angles.

5 Conclusion

A new method for efficiently measuring loudspeaker characteristics with very high spatial resolution was presented in this paper. The proposed method relies on a continuously rotating apparatus for saving measurement time and on an adaptive computational algorithm for continuous impulse response representation of the measured device in the azimuth direction. With reference to similar techniques earlier presented for obtaining continuous-azimuth head-related transfer functions, we merely use in the order of one minute recording time for one 360 degree revolution of the loudspeaker device, which constitutes the primary advantage of the proposed method over traditional time-consuming measurement of discrete angles. Under these conditions, the paper delivers evidence of a by and large equivalence of the rapid measurement with the traditional sweep method in terms of impulse responses, frequency responses, and directivity maps. We believe that the amenities of this new measurement technology will be useful in various fields of acoustic product development, validation, or individualization.

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