System for Measuring Directional Responses of Sound Sources

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***Abstract*—This paper presents a MATLAB-based application for measuring the directional responses of sound sources and performing subsequent analysis. The system incorporates digital signal processing, microphone array calibration, and graphical representation of results. Additionally, an automatic calibration method utilizing an acoustic calibrator is introduced. The paper concludes with the presentation and discussion of measurement results obtained using the developed application.**

***Index Terms*—acoustic measurements, directional response, frequency response, sound source analysis, digital signal processing, automatic acoustic calibration, microphone array calibration, spherical plot, fractional octave analysis, anechoic chamber.**

I. Introduction

Acoustic measurements are very important to today's society. They are used in modern electronic and automotive research and development. At Brno University of Technology, an anechoic chamber is employed for such measurements. To ensure optimal performance, the anechoic chamber must be isolated from the control room. Conducting measurements of sound sources using commercially available software has become increasingly impractical. Many applications support only single-microphone measurements, while others require manual calibration, where an acoustic calibrator is placed on the microphone in the anechoic chamber, and the user must return to the control room to initiate the calibration process. This process becomes time-consuming when a large number of microphones is used, reducing the efficiency of the measurement process. The application described in this paper supports an array of up to 64 microphones, this makes manual calibration time consuming, so it is necessary to introduce an automated calibration procedure. The application described is a dedicated application designed to address these challenges. It has the same name as the master's thesis associated with it. The application is called System for Measuring Directional Responses of Sound Sources (SMDRSS) for brevity and pronunciation purposes, the abbreviation is intended to be read as the name Smithers. The current implementation of the application is tailored for measuring loudspeakers as sound sources and is compatible with audio interfaces supporting a large number of input channels, such as MADI, AVB, and DANTE. It is designed for measurement with a single microphone or with microphone arrays. Future development aims to extend its capabilities to include the measurement of musical instruments, incorporating automated onset detection.

Acoustic measurements are inherently prone to errors. For instance, the addition of a measurement microphone into the anechoic chamber requires a mounting solution, such as an arm which can introduce acoustic reflections into the microphone, thereby skewing the results. Additionally, the measurement system exhibits various non-linearities. The non-linearities can stem from analog-to-digital and digital-to-analog converters, air transmission, and the movement of speaker and microphone membranes. These non-linearities must be considered because many conventional signal processing techniques assume a linear system [1].

The application described in this paper enables the measurement of two-dimensional (2D) and three-dimensional (3D) directional responses, as well as the frequency response of the system at each measurement point. The primary measurement signal used is a one-second exponential sweep. However, the software is flexible and allows for the use of any measurement signal, supplied as a linear pulse code modulation (LPCM) wave file.

II. Defining Used Algorithms

A. Sound Pressure Level Calculation

The sound pressure level (SPL) is calculated using the root-mean-square (RMS) value of the signal captured by the microphone. The RMS value is computed in the discrete time domain using samples of the measured signal. The sound pressure level in dB(SPL) is then determined using the ratio of the RMS value obtained with an acoustic calibrator to the RMS value of the measured signal. The acoustic calibrator used in this study generates a 1000 Hz tone at 94 dB(SPL). The reference pressure for dB(SPL) is 20 μPa, which corresponds to the threshold of human hearing for a 1 kHz sinusoidal tone [1].

B. Automated Calibration Using Autocorrelation

The automated calibration process is a key feature that significantly reduces calibration time. Most software implementations require the user to place an acoustic calibrator on each microphone and manually initiate calibration each time in the software. In contrast, the proposed approach automatically detects the frequency and level of the calibration tone. The level detection is achieved by calculating the RMS value of each buffer. If the level exceeds -40 dB(FS), the autocorrelation algorithm is used to detect the frequency. If the level does not exceed -40 dB(FS) the signal is ignored. This reduces the risk of false frequency detection, which could trigger calibration. In addition the calibration tone using any calibration tool will always be the dominating one, especially in the anechoic chamber, this improves the accuracy of the used frequency detection algorithm. A test recording was measured to prove the viability of the process (see Fig. 1)

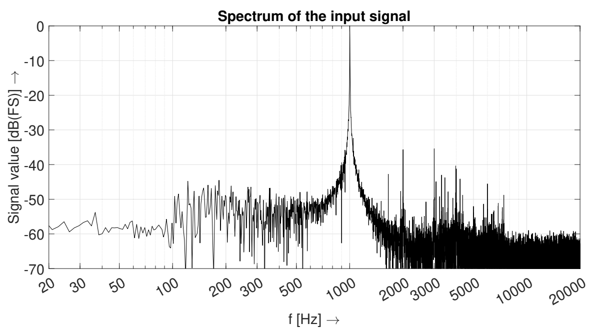

Fig. 1. Spectrum of the recorded test signal

The frequency detection is performed using the autocorrelation algorithm. First, the autocorrelation formula (1) is applied to the signal [2]:

$$R_{\mathrm{XX}}[n] = \sum_{\mathrm{m}=0}^{N-1} x[m] \cdot x[(m+n)_{\mathrm{mod}(N)}], \tag{1}$$

where $x[m]$ is the input signal, n is the lag, and N is the signal length. The number of samples in the autocorrelation function is quite high, only several peaks are required to detect the frequency of the signal. To reduce computational requirements, the number of samples in the autocorrelation function is trimmed to include only the dominant peaks. The number of samples required to capture a specific number of peaks is given by:

$$N_{\mathrm{S}} = N_{\mathrm{P}} \cdot \frac{f_{\mathrm{S}}}{f_{\mathrm{E}}}, \tag{2}$$

where N_{P} is the number of peaks, f_{S} is the sampling frequency, and f_{E} is the expected frequency of the calibration tone.

After trimming, the signal is normalized (see Fig. 2) and rectified by setting all samples with a value below zero to zero (see Fig. 3). The final step involves identifying local maxima and their positions, calculating the mean distance between peaks, and determining the frequency using:

$$f = \frac{f_{\mathrm{S}}}{\Delta N_{\mathrm{avg}}}, \tag{3}$$

where f_{S} is the sampling rate and ΔN_{avg} is the average distance between peaks in samples.

The application reports ΔN_{avg} as 48 samples. Given a sampling rate f_{S} of 48 kHz (used for recording the test signal, as depicted in Fig. 1), the detected frequency f is 1000 Hz, as calculated using (3).

A circular buffer records the signal until calibration is complete. If the detected frequency f calculated using (3) matches the calibration frequency for a sufficient number of buffers (approximately 0.5 seconds), the RMS value of the recorded signal is used as the calibration constant. The 0.5-second recording duration ensures sufficient precision in the RMS calculation. This approach allows the user to remain in the anechoic chamber, calibrating each microphone sequentially or multiple microphones simultaneously using multiple calibrators.

Potential improvements include the use of a bandpass filter tuned to the calibration tone frequency before autocorrelation, which could enhance reliability in noisy environments. To clarify the filtered signal would not be used for the RMS calculation, it would be used for the frequency detection and the unprocessed signal would be used to calculate the RMS for the calibration constant.

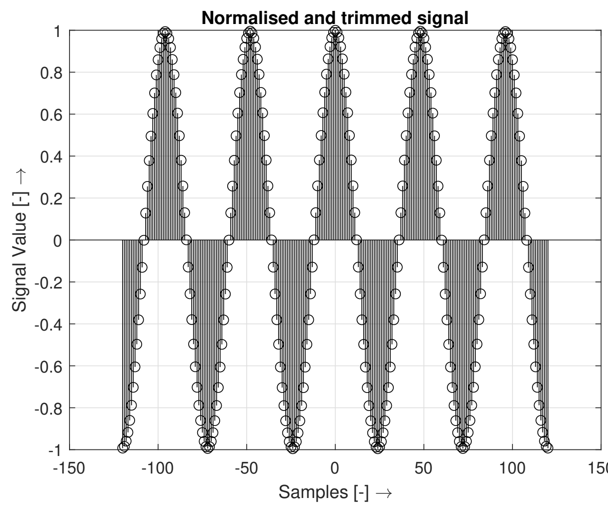

Fig. 2. Trimmed and normalised autocorrelation function of the recorded signal

C. Transfer Function Calculation

The transfer function is calculated using [2]:

$$H(\omega) = \frac{Y(\omega)}{X(\omega)}, \tag{4}$$

where $X(\omega)$ represents the spectrum of the input signal (flowing into the system under analysis) and $Y(\omega)$ represents the spectrum of the output signal (flowing out of the analysed system). While this equation is precise for a linear system, using it for a non-linear systems introduces inaccuracies. The inaccuracies introduced in non-linear systems are minimal for this application, though they should still be considered.

III. Defining Measurements

A. Single Point Analysis

This measurement type is used to calculate and analyse the frequency response of the measured system using (4). The

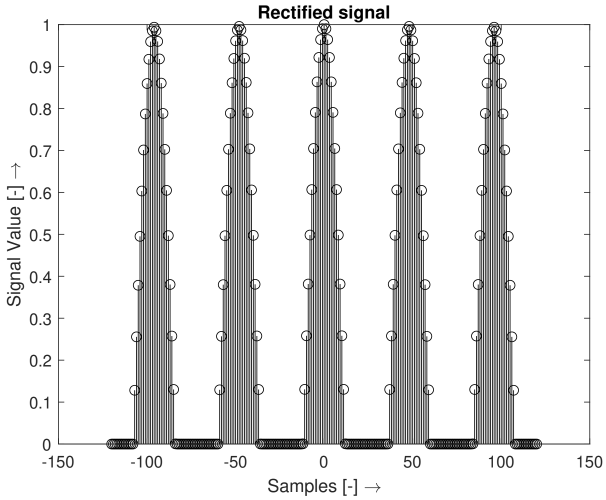

Fig. 3. Rectified autocorrelation function of the recorded signal

frequency response coefficients are subsequently utilized for fractional octave analysis. Fractional octave analysis subdivides the transfer function into octave bands, which can be used for further analysis. The most common fractional octave analysis is 1/3-octave analysis. The RMS value of the sound pressure level within each fractional octave band is computed using:

$$X_{\mathrm{RMS}} = \sqrt{\frac{4}{N}\sum_{\mathrm{k}=0}^{N-1} X_{\mathrm{k}}^2}, \tag{5}$$

where X_{k} is the spectral coefficient and N is the signal length. The number four in the mean calculation needs to be added, because the value of the spectral coefficient X_{k} is one half of the signal amplitude, this means that the RMS value needs to be multiplied by two, and this two is moved into the square function to make it readable. For brevity, the fractional octave analysis is not demonstrated in this paper. However, (5) is employed to calculate the sound pressure levels for specific third-octave bands, which are used to generate the 2D and 3D directional responses of the measured system.

B. Directional Response

This measurement determines the sound pressure level as a function of the angle relative to the reference axis, using multiple measurement points. It can display 2D directional characteristics in either the horizontal or vertical plane using a polar plot, or 3D directional characteristics using a spherical plot. The directional characteristic is computed for a specific third-octave frequency band, following the guidelines outlined in [3]. In this measurement, 0 degrees of elevation and 0 degrees of azimuth correspond to the direction of the acoustic axis [4]. The directional response is modelled using the solution to the wave equation for a circular piston membrane [1], given by:

$$\eta(\varphi) = \left|\frac{2\mathrm{J}_1(kR\sin(\varphi))}{kR\sin(\varphi)}\right|, \tag{6}$$

where φ is the azimuth angle, J_1 is the first-order Bessel function, k is the wavenumber, and R is the radius of the circular piston membrane. The wave number is calculated using:

$$k = \frac{2\pi f}{c_0}, \tag{7}$$

where f is the frequency of vibration, and c_0 is the speed of sound in air.

IV. Results

This section presents the measurement results obtained using the described measurement system. The equipment used includes:

- Measurement microphone
- Brüel & Kjær 9640 microphone turntable
- Brüel & Kjær NEXUS Conditioning Amplifier
- Computer running the SMDRSS application
- RME Digiface Dante and RME Micstacy audio interfaces
- EVENT Electronics ALP5 active speaker (see Fig. 4)

The measurement was conducted in six passes using a single microphone, rotated in 5-degree azimuth increments, resulting in 73 measurements per pass. The 73rd measurement was included to connect the first and last points of the polar plot. Alternatively, the first measurement can be duplicated as the 73rd point to enclose the plot. The choice between these approaches is left to user preference. These passes were performed at elevations of 0°, 5°, 10°, 20°, 30°, and 40°, with the microphone positioned 1 meter from the speaker's acoustic center [4]. This setup resulted in a total of 438 measurement points, ensuring sufficient spatial coverage for accurate directional response analysis.

Fig. 4. The EVENT Electronics ALP5 active speaker used as the sound source

A. Single Point Frequency Responses

Figures 7, and 8 show the frequency response of the measured system at different azimuths at 0-degree and 40-degree elevations. Higher frequencies exhibit greater directionality,

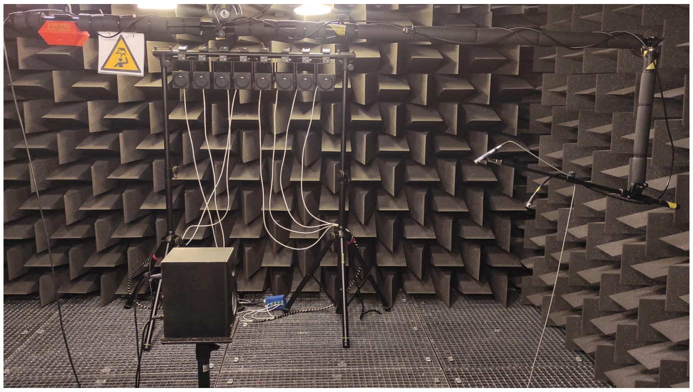

Fig. 5. The measurement setup used to gather data for this paper

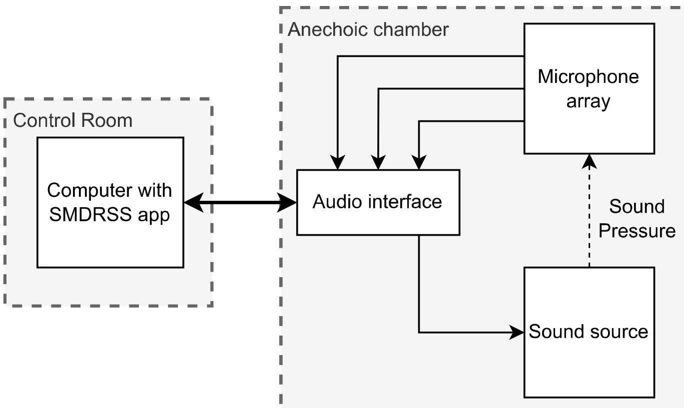

Fig. 6. Schematic of the signal flow and connections between devices during measurement

leading to a decrease in level as the angle from the acoustic axis increases.

B. Directional Responses in 2D

Figures 9, 10, and 11 present the measured directional responses, compared to a mathematical model, at an elevation of 0 degrees. The theoretical curves are based on the circular piston membrane model, as given by (6), where the radius corresponds to the respective speaker cone radii (in meters) for both the woofer and the tweeter. The plots also include the normalized RMS values of the measured sound pressure.

Figure 9 shows a measurement at a low frequency, where the speaker's crossover causes most of the sound to originate from the woofer, which has a cone diameter of 5 inches. The model does not account for the bass reflex port if the speaker enclosure, which introduces distortion in the shape of the response curve. Figure 10 represents a transitional frequency, where both the woofer and tweeter contribute to the output, resulting in a directional pattern that lies between their respective modelled responses. At higher frequencies, as shown in Figure 11, only the tweeter, with a cone diameter of 1.2 inches, is active.

Figures 12 and 13 show the directional response at 0-degree elevation, while Figure 14 depicts the response at a 40-degree elevation. The mentioned figures demonstrate the same directional effects observed in the frequency response measurements.

C. Directional Responses in 3D

This measurement utilizes all 438 recorded points to generate the 3D directional response. The 3D directional response is represented as a spherical plot, requiring azimuth, elevation, and sound pressure level as inputs. Figures 15 and 16 show the response at 15.85 kHz. In the mentioned figures, the coordinate center—which coincides with the acoustic center of the speaker—is marked with an 'X' and the acoustic axis is indicated by a dashed line in the top view.

V. Conclusion

This paper introduced the challenges associated with acoustic measurements, particularly those related to calibration in an anechoic chamber. The issues encountered during calibration were identified, and corresponding solutions were presented. Fundamental problems inherent to acoustic measurements were also discussed, alongside the introduction of a dedicated application designed to address these challenges.

Key algorithms for acoustic measurements, including sound pressure level calculation and transfer function computation, were defined. Manual calibration of a microphone array typically takes 10 to 15 seconds per microphone. In contrast, the automated method proposed in this paper reduces the calibration time to approximately 2 seconds per microphone. Moreover, the operator no longer needs to exit the chamber after calibrating each microphone; instead, the entire array can be calibrated before leaving. These improvements significantly reduce both the time and effort required for the calibration process.

The measurement types employed in this study—single-point analysis, 2D directional response, and 3D directional response—were outlined, demonstrating the versatility of the proposed system. The results of these measurements were presented and analysed, highlighting the system's effectiveness in capturing detailed acoustic characteristics.

The proposed system streamlines the calibration process, enabling researchers to focus on measurement procedures rather than manual calibration tasks. By supporting a wide range of equipment and adaptable measurement scenarios, this application has the potential to accelerate advancements in acoustic research.

Acknowledgement

The author would like to express gratitude to doc. Ing. Jiří Schimmel, Ph.D., for his supervision and guidance throughout this study.

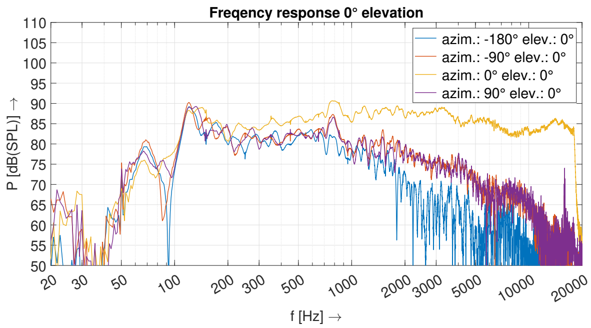

Fig. 7. Frequency response at different azimuths and an elevation of 0 degrees

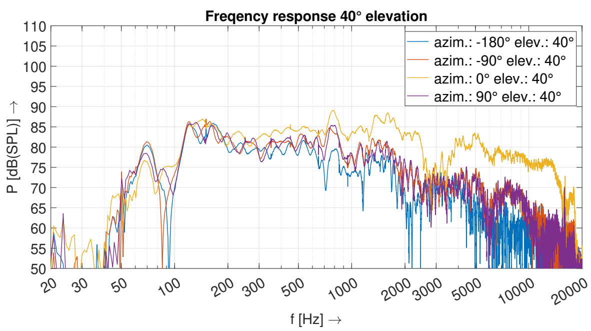

Fig. 8. Frequency response at different azimuths and an elevation of 40 degrees

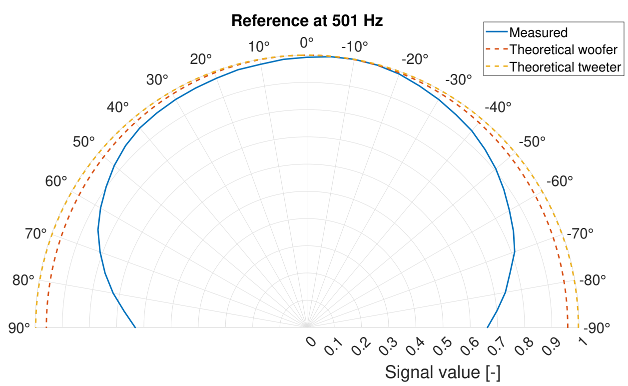

Fig. 9. Comparison of the theoretical and measured directional responses at 500 Hz

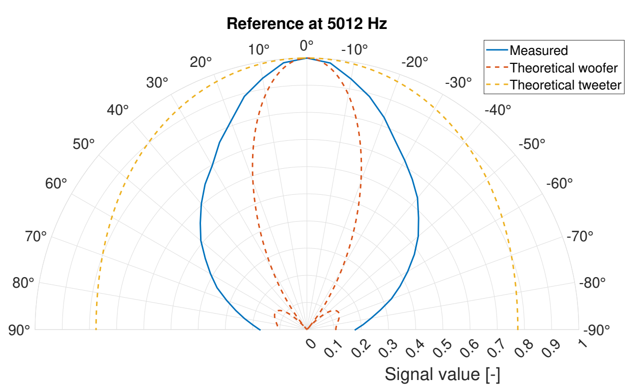

Fig. 10. Comparison of the theoretical and measured directional responses at 5000 Hz

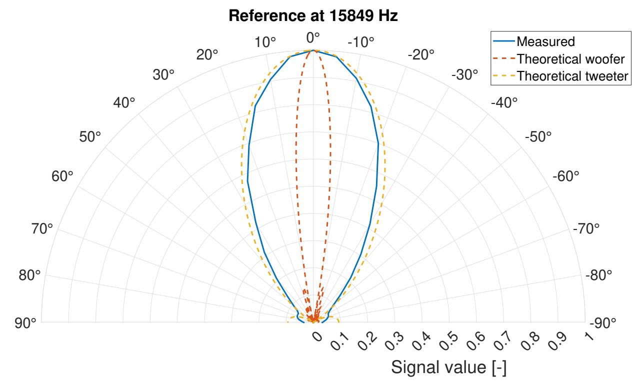

Fig. 11. Comparison of the theoretical and measured directional responses at 15000 Hz

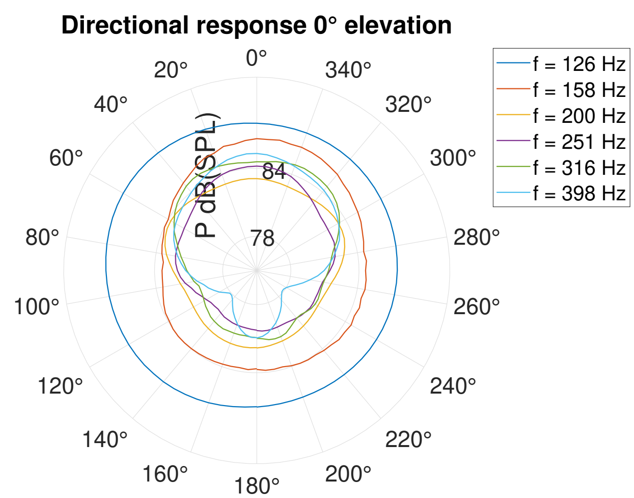

Fig. 12. Directional response of the measured system at low frequencies for a 0-degree elevation

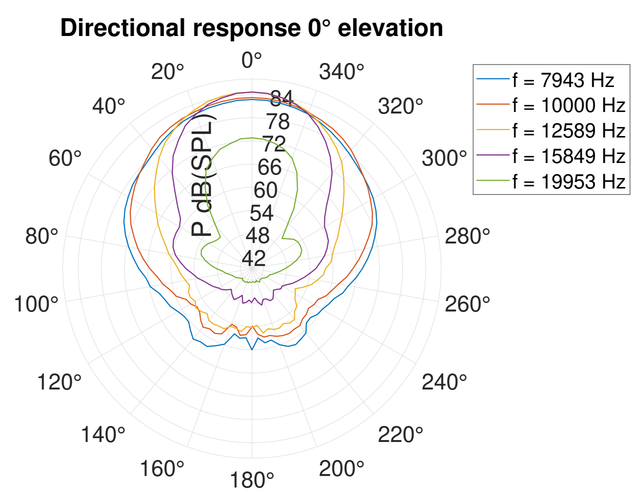

Fig. 13. Directional response of the measured system at high frequencies for a 0-degree elevation

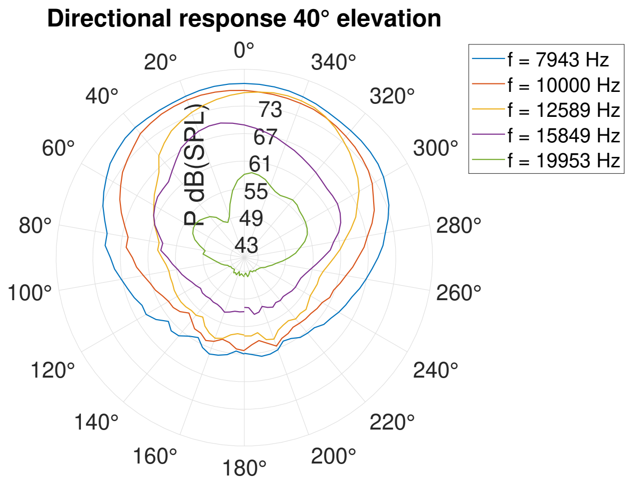

Fig. 14. Directional response of the measured system at high frequencies for a 40-degree elevation

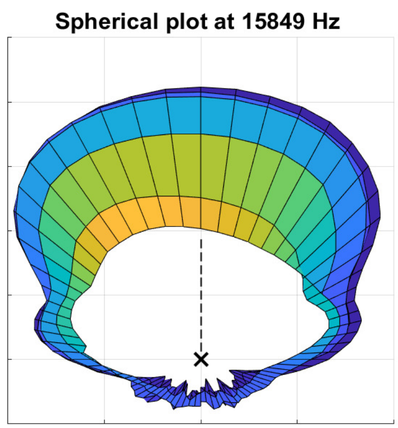

Fig. 15. Top view of the directional response of the measured system at 15849 Hz

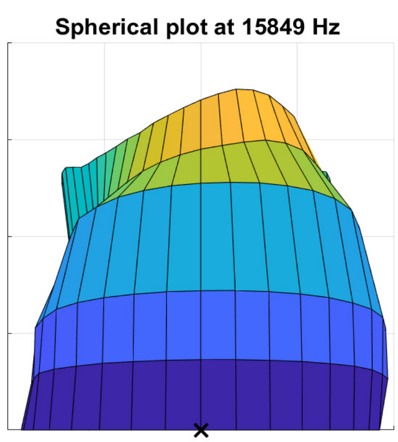

Fig. 16. Front view of the directional response of the measured system at 15849 Hz

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