

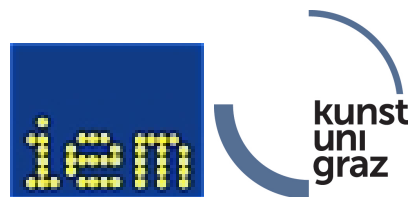
Sound Reinforcement, Laboratory, Assignment 2  
Lecture-Number: 17.0036

# Measurement of a variable curved Line-Source in a room and Analysis of its Decay over distance

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# 1. Introduction

This report investigates decay behavior by increasing the distance of a variable curved line source, measured during the laboratory „*Sound Reinforcement*“ in lecture hall i9, Inffeldgasse 13 on May 9th. As the device under test (DUT), the 3D-printed small-size variable curved line source invented by Lukas Gölles and fabricated by the Institute of Electronic Music and Acoustics (IEM) was chosen. The used line array (LA) configuration comprises twelve single loudspeaker enclosures, as shown in Fig. 1.1. The target design was a curvature that results in a decay of 0 dB per doubling of the distance (dod). The online tool, EnImSo line array desinger [Gö23], allows us to easily calculate the angles for an individual LA by specifying just the enclosure number and size, the suspension height, the listening area, and the decay target. After calculating an appropriate design, the continuous angles are adapted to the discrete possible angles of the DUT. The final design was precisely suspended, and each loudspeaker was excited individually with a sine-sweep and measured with pressure zone microphones (PZM) on floor level at 26 probe locations equally spaced by 0.5 m. The resulting impulse responses (IR) are post-processed, and the results are presented in this report.



Figure 1.1.: DUT: Line Array

## 2. Experimental Setup & Post-Processing

### 2.1. Experimental Setup

As the measuring environment, the acoustically treated lecture hall i9 ( $17.25\text{ m} \times 9.5\text{ m} \times 3.9\text{ m}$ ) with a floor space of  $164.7\text{ m}^2$  was chosen. The investigated LA consists of twelve 3D-printed cabinets [GZM23], each equipped with an SB acoustics SB65WBAC25-4<sup>1</sup>, 2.5-inch wide-band transducer. Each of the cabinets has an enclosure height of 0.082 meters, by which we were getting a total LA length of 0.984 m. We chose two different configurations to investigate the influence of a wall close to the back of the LA. In the first configuration, the LA is located in the room with more than 2 m distance to the closest wall. Secondly, the LA is placed as close as possible to the back wall (0.65 m). The mounting mechanism enables 11 different adjustable angles from  $0^\circ$  to  $10^\circ$ , in steps of  $1^\circ$ .

The web-based tool *EnImSo*<sup>2</sup> calculates the discrete splay angles for the specified configuration for a listening area of 10 m depth and an upper source height of 2 m with a target decay of 0 dB per dod. The calculated angles are displayed in table 2.1. These angles must adapt to match the LA’s possible mechanic angles to achieve the design’s overall curvature. For this purpose, getting a similar accumulated angle of all enclosures is very important. In this case, having a gain  $g = 0.27$ , the accumulated total array angle of the proposed design is  $34.8^\circ$ . The adapted angles are shown in the third column and accumulate to  $29.5^\circ + 4.8^\circ = 34.3^\circ$ . In the last column, the sample delays for each speaker are displayed, which are necessary to get a decay of  $-3\text{ dB}$  per dod. The array tilt angle describes the overall mounting angle of the array, often enforced by a bumper, which carries the whole array.

Enclosure number (From top to bottom)	Tilt angles <i>EnImSo</i> (0dB/dod)	Adapted cabinet angles ( $1^\circ$ -steps)	Sample delay (48 kHz) (-3dB/dod)
Array tilt angle	$11.3^\circ$	$6.5^\circ + 4.8^\circ = 11.3^\circ$	0
1	$0^\circ$	$0^\circ$	0
2	$0.2^\circ$	$0^\circ$	0
3	$0.3^\circ$	$0^\circ$	1
4	$0.4^\circ$	$1^\circ$	1
5	$0.5^\circ$	$0^\circ$	2
6	$0.6^\circ$	$1^\circ$	3
7	$0.8^\circ$	$1^\circ$	5
8	$1.2^\circ$	$1^\circ$	6
9	$1.7^\circ$	$2^\circ$	8
10	$2.8^\circ$	$3^\circ$	11
11	$4.9^\circ$	$4^\circ$	14
12	$10.1^\circ$	$10^\circ$	16
Accumulated	<b><math>34.8^\circ</math></b>	<b><math>34.3^\circ</math></b>	<b>67</b>

Table 2.1.: Calculated and adapted enclosure angles and in sample delays at 48 kHz

As no bumper exists for the prototyped array, a provisional suspension was built using a loudspeaker

<sup>1</sup><https://sbacoustics.com/product/2-5in-sb65wbac25-4/>

<sup>2</sup><https://enimso.iem.sh/post/line-array-designer-two-target/>

stand, a metal boom, and a chain. The final overall array tilt angle of  $6.5^\circ$  results from  $4.8^\circ$  angular trapezoidal shape of the enclosure and was precisely adjusted using a laser distance meter as shown in Fig. 2.1. To cover the listening area at  $0^\circ$  in the on-axis orientation of the LA, 26 evenly spaced



Figure 2.1.: Array tilt angle



Figure 2.2.: AKG PZM 6D being positioned

probe locations on the ground were chosen to avoid reflections from the floor. A spatial discretization of 0.5 m results in a one-dimensional coverage of 12.5 m, which exceeds the priority listening depth of 10 m to investigate the decay beyond the design boundary. Two AKG PZM 6D boundary microphones were used to get as close as possible to the floor, as shown in Fig. 2.2. The origin position (0 m) was directly below the LA to evaluate the spatial aliasing at higher frequencies ( $> 4$  kHz).

Fig. 2.3 sketches the experimental setup in a side view. The blue points on the ground indicate the microphone positions for the measurement. The dashed lines are the aiming directions of each speaker. It is visible that the LA was supposed to deliver an even SPL for the distances ranging from 1 m to 10 m as all aimings (apart from the lowest cabinet) are directed towards this area.

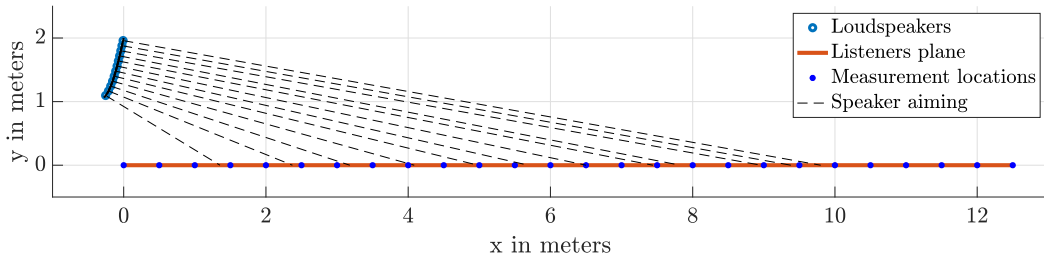


Figure 2.3.: Sketch of the geometry of measurement and aiming angles

## 2.2. Measurement Setup

In order to obtain the IR from each cabinet to each probe location, we used the multi-exponential-sine-sweep (MES) measurement [MBL07]. The MES is gainful because of its low crest factor and the possibility of suppressing any harmonic distortion. Furthermore, measuring multiple source-receiver combinations is time-efficient. Having 12 sources and 26 receivers in two configuration positions results in  $12 \times 26 \times 2 = 624$  single sweep measurements. Using the MES method (measuring all 12 speakers simultaneously) and two microphones simultaneously, we reduced the number of measurements to  $1 \times 13 \times 2 = 26$ .

The MES  $\mathbf{x}[n]$  was created using the *Python* script 'sweep\_generation.py', having a sweep-length of 10 s and a time difference between the sweeps of 2 s. To avoid crackles at the end of a sweep, the starting frequency of 20 Hz was adapted such that we start and stop at 20 kHz with a zero phase value.

A second *Python* script 'measurement.py' executed the measurement by playing back the sweep and recording the room responses from the microphones.



Figure 2.4.: *RME Fireface UFX+*



Figure 2.5.: *sonible d24*

The used audio interface *RME Fireface UFX+* (Fig. 2.4) converted the audio stream to MADI. An optical fiber cable sends the 12-channel audio to the *sonible d24* (Fig. 2.5) amplifier. The 12 speakers are connected via NL8 multipin cables to the amplifier. The two AKG PZM 6D boundary microphones recorded the MES, including the room reflections and speaker characteristics. These signals are sent back to the audio interface via XLR and converted to digital. Then, the measurement script multiplies the Fourier transformation of the inverse of the original sweeps  $\mathbf{x}_{\text{inv}}[n]$  with the Fourier transformation of the recorded sweeps  $\mathbf{y}[n]$ ,

$$\mathbf{H}(\theta) = \mathcal{F}\{\mathbf{y}[n]\}\mathcal{F}\{\mathbf{x}_{\text{inv}}[n]\}, \quad (2.1)$$

and the IRs  $\mathbf{h}[n]$  can then easily be obtained by applying the inverse Fourier transformation to the resulting transfer functions  $\mathbf{H}(\theta)$ , such that

$$\mathbf{h}[n] = \mathcal{F}^{-1}\{\mathbf{H}(\theta)\}. \quad (2.2)$$

## 2.3. Post-Processing

After obtaining the raw IRs from measurements, further signal processing steps are necessary to evaluate the data. At first, a microphone level mismatch between the two microphones must be compensated (See Fig. 2.6). The IRs are dynamically truncated to 250 samples to remove room reflections. The 12 individual IRs are summed in the complex domain after applying the Fourier transformation. Furthermore, the individual speaker can be equalized with two peak filters to get a flat frequency response of the direct sound. By adding extra delay, as mentioned in the section 2.1, the decay can be electronically modified to  $-3$  dB per dod. To correct the spectral behavior of the LA, a whitening filter with  $-3$  dB/octave from 4 kHz towards lower frequencies is used (See Fig. 2.8). The root mean square (RMS) is calculated to show the overall impression of the whole array over the distance. The spectrum is also split into 1/3-octave bands using a filter bank to observe different frequency ranges (See Fig. 2.8). An A-weighting filter is applied to match the human hearing perception (See Fig. 2.8) for the RMS plot. A waterfall plot displays the spectral evolution over the distance.

### 2.3.1. Microphone Offset Correction

Fig. 2.6 shows the RMS SPL ( $L_{p,\text{RMS}} = 20\log_{10}\left(\frac{\sqrt{\frac{1}{N}\sum_{n=0}^{N-1}h(n)^2}}{p_0}\right)$ ) of the individual IRs at the measured locations before post-processing for the two examined configurations. The second microphone shows

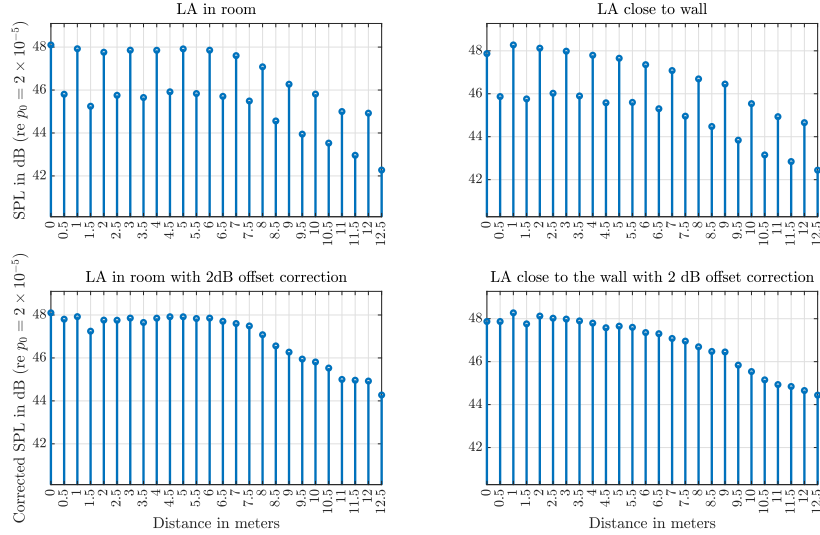


Figure 2.6.: Offset correction of 2 dB for the second microphone

a level difference of  $-2$  dB compared to the first microphone. A simple offset correction with,

$$h_{\text{mic},2}(n) = 10^{\frac{2 \text{ dB}}{20}} h_{\text{mic},1}(n) \quad (2.3)$$

is performed to compensate for the mismatch. The corrected RMS SPL is displayed in the lower two plots.

### 2.3.2. Truncation

The IRs are dynamically truncated to 200 samples after the prominent peak to avoid first reflections from the room's boundary surfaces. Therefore, a peak detection algorithm determined the location of the most prominent peak of each IR, and a rectangular window of 250 samples was applied. The detected peak is located at sample position 51 of the window to include the transient process. Fig. 2.7 shows an exemplary recorded IR with two first reflections visible between samples 13600 and 13700, the truncation window of 250 samples, and the resulting IR without the reflections.

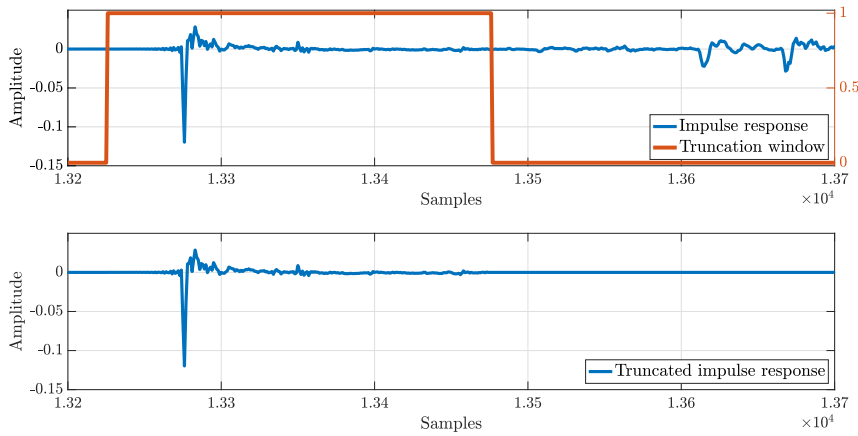


Figure 2.7.: Top: Original IR (blue) with rectangular window (amber); Bottom: truncated IR

### 2.3.3. Filter

In total, four different filters are applied to the IRs.

- The IRs are filtered with two peak filters with the following parameters to equalize the loudspeaker towards a flat direct sound frequency response [GZM23].

$$f_{\text{EQ},1} = 2.8 \text{ kHz}, \quad g_{\text{EQ},1} = -6 \text{ dB}, \quad Q_{\text{EQ},1} = 0.8$$

$$f_{\text{EQ},2} = 22 \text{ kHz}, \quad g_{\text{EQ},2} = -10.5 \text{ dB}, \quad Q_{\text{EQ},2} = 1.69$$

- As far as the LA shows a pink noise behavior towards low frequencies, a spectral correction filter (SCF) up to 4 kHz with a magnitude  $|H_{\text{SCF}}(f)|$  was used.

$$|H_{\text{SCF}}(f)| = \min\left(\sqrt{\frac{f}{4000}}, 1\right) \quad (2.4)$$

- A  $\cos^2$  third-octave filter bank was used for smoothing and separating the frequency bands.
- The A-weighting filter transfer function<sup>3</sup> in the Laplace-domain is given by

$$G_A(s) = \frac{k_A s^4}{(s + 129.4)^2 (s + 676.7)(s + 4636)(s + 76617)^2}, \quad \text{with} \quad k_A = 7.39705 \times 10^9. \quad (2.5)$$

All the proposed filter magnitudes are shown in Fig. 2.8

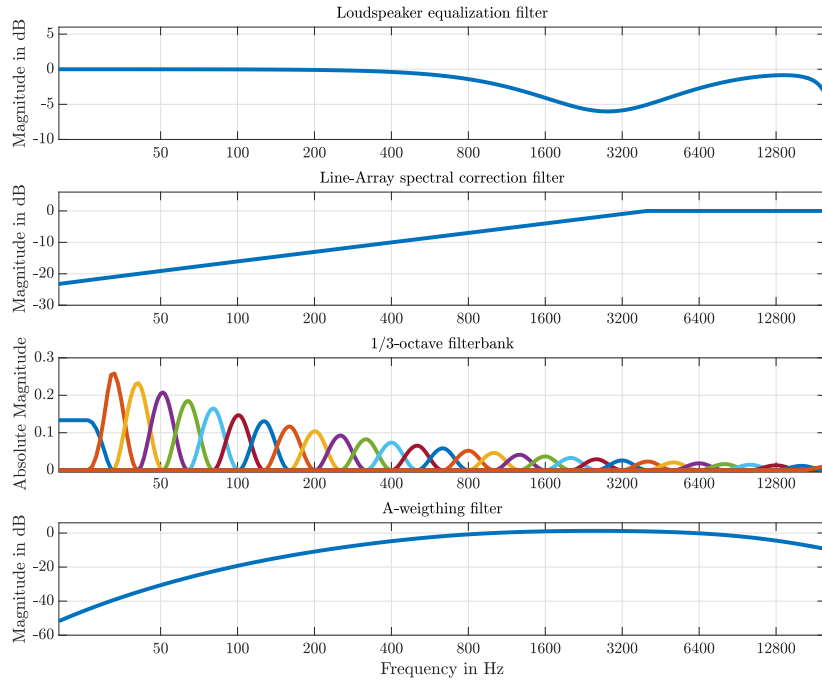


Figure 2.8.: Frequency responses of the applied filters. From top to bottom: -Loudspeaker equalization, - Spectral LA correction filter, - 1/3-octave filterbank, -A-weighting filter

<sup>3</sup><https://en.wikipedia.org/wiki/A-weighting>



## 3. Results

This chapter includes all the results from the measurements. At first, the Sound Pressure Level (SPL) in the distance of the LA is evaluated and related to the behavior of familiar point sources. The frequency dependency over distance is investigated by viewing separate 1/3-octave bands and waterfall plots, providing an overview of the evolution. Moreover, the behavior of the wall close behind the LA and the electrical delay of the individual speaker to achieve a different decay are evaluated.

### 3.1. Decay Behavior

As described in section 2.3, the post-processing is done, and the complex transfer functions of the 12 speakers are summed up per microphone location. Fig. 3.1 shows the RMS SPL values over the distance for the two configurations. The x-axis is implemented as semi-logarithmic (linear from 0 m to 1 m, and then logarithmic) to simplify the observation of doubling the distance. With the proposed

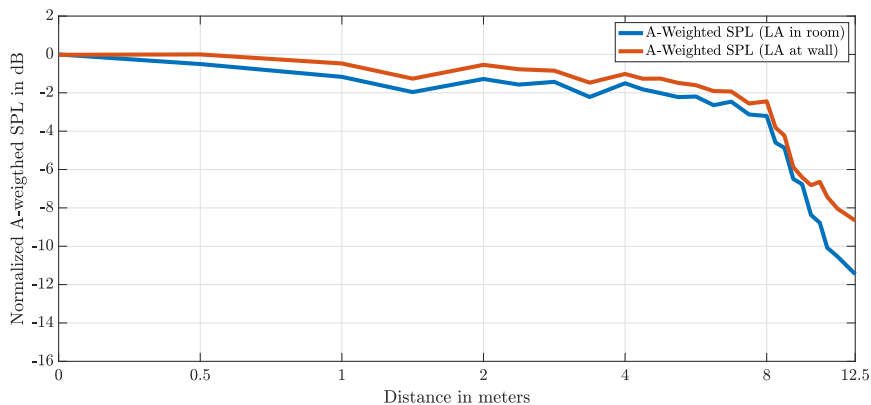


Figure 3.1.: A-weighted RMS SPL for the two configurations

design, the target of 0 dB per dod is nearly achieved for the configuration in the room and close to the wall. Up to 8 m the level varies by only  $\pm 1$  dB. Between 2 m and 8 m the variation is even less than  $\pm 1$  dB. As expected, the level drops drastically after 8 m by approximately 12 dB per dod for the configuration with the LA in the room. The increasing values towards the origin are because of the unidirectional directivity of low frequencies, which especially affect the near field, which is also responsible for the overall higher values of the LA close to the wall.

By comparing the LA Fig. 3.1, with Fig. 3.2, the effect of the LA, concerning level decay in distance is remarkable. This figure shows the decay of only the uppermost and lowest speaker. The typical  $1/r$  decay of  $-6$  dB per dod is visible from a certain distance. The lowest speaker starts to decay with  $1/r$  from 2 m and the uppermost speaker from about 4 m.

### 3.2. Spectral Decay

Besides the SPL, the spectral behavior per dod is also important for a balanced sound experience. Therefore, seven meaningful 1/3 octave bands from 125 Hz to 8 kHz are picked to evaluate the LA.

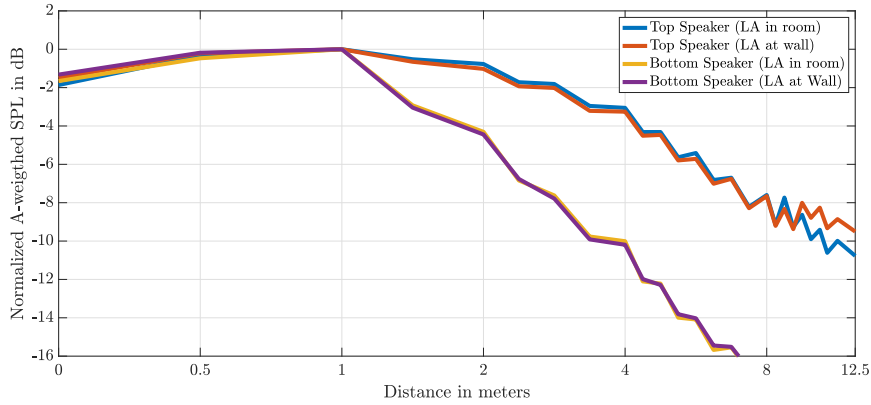


Figure 3.2.: A-weighted SPL RMS, Top/Bottom loudspeaker

### 3.2.1. 1/3 Octave Bands

Fig. 3.3 shows the decay of different frequency bands. The two lowest bands (125 Hz and 250 Hz) show almost no impact of the LA. From 1 m to 2 m, these frequencies decay with  $-6$  dB per dod, which implies the behavior of a point source. At 500 Hz, the array begins to act as a line source, which depends on the ratio between the wavelength and the LA length. A line source starts to act if the wavelength reaches the dimension of the array length. For frequency bands above 1000 Hz the LA the

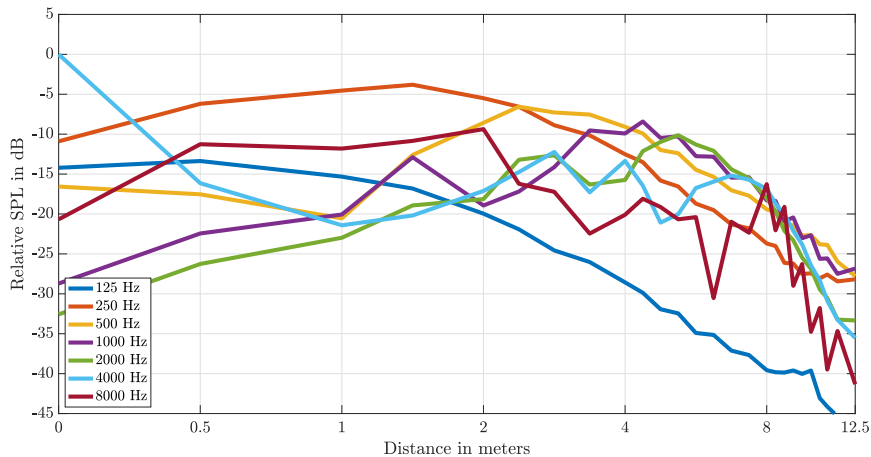


Figure 3.3.: 1/3-octave decay, LA in Room

decay seems to increase up to 6 m to 8 m. As mentioned before, in the area of interest is between 1.4 m to 8 m the frequencies [2000 Hz, 4000 Hz] only varies by  $\pm 5$  dB. The 1000 Hz frequency band varies about  $\pm 10$  dB, which indicates a spectral unbalance at these frequencies over the decay process. In the front area, less direct sound arrives due to the directivity pattern of the single cabinet.

For better visibility of the decay behavior of the LA, the different frequency band decays are shifted such that they have the same level at a representative middle distance of 5.5 m. Fig. 3.4 underlines the nearly flat decay of the upper-frequency bands in the area of interest, while the level loss of the lower bands is significantly higher. This figure shows that the decay beyond 8 m increases for higher frequency bands and is less for the lower ones. It can be explained by the dissipation of the air, which is stronger for higher frequencies. Also, the wave propagation behavior of low frequencies could influence the sound pressure closer to the wall. The peaks of the 4000 Hz band at 7 m and the 8000 Hz

band at 8 m could be even predicted by viewing at the nearby aiming points in Fig. 2.3 at these locations. At the origin 0 m (below the LA), the extremely different levels for the 4000 Hz and 8000 Hz

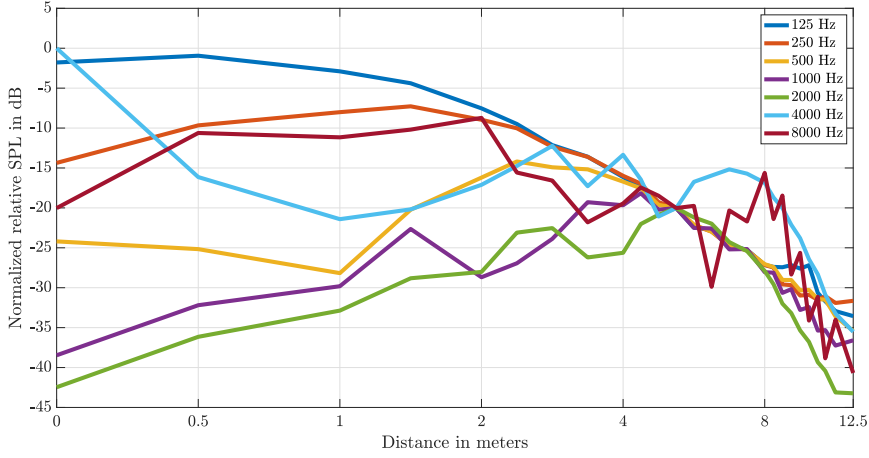


Figure 3.4.: 1/3-octave decay, normalized to 5.5 m, LA in Room

indicate the spatial aliasing of the LA which correspond to the distance between neighboring speakers in the array. This interference between two centers of neighboring membranes theoretically starts at  $d_{\text{speakers}} > \lambda_{\text{aliasing}}$  where  $d_{\text{speakers}}$  is the distance between the two centers. In our case,

$$d_{\text{speakers}} = 0.082 \text{ m} = \lambda_{\text{aliasing}}, \quad \text{corresponds to,} \quad f_{\text{aliasing}} = \frac{c}{\lambda_{\text{aliasing}}} = \frac{340 \frac{\text{m}}{\text{s}}}{0.082 \text{ m}} = 4146.34 \text{ Hz} \quad (3.1)$$

such that we can assume that the spatial aliasing starts, as observed, at around 4 kHz. The interference pattern below at above the line source builds up as peaks whenever  $d_{\text{speakers}}$  is a multiple of the whole wavelength and dips when  $d_{\text{speakers}}$  causes a phase difference of  $180^\circ$ .

### 3.2.2. Spectral Evolution over Distance

The spectrum at the different probe locations is displayed using a three-dimensional waterfall plot to investigate the spectral evolution along the observed distance. In Fig 3.5, it is especially striking that from what frequency range, the LA acts as a line source and as a point source. From the 180 Hz peak at the origin an edge extends till 1000 Hz at 5 m, which indicates the boundary for point source behaviour. From this boundary, the LA acts as a point source for the level decay with  $1/r$ . This underlines the observation in section 3.2 of an uneven decay process between 500 Hz and 1000 Hz. Above 1000 Hz, the decay remains, some minor ripples neglected, stable from 1.5 m to 8 m, which indicates the line source behavior. The spatial aliasing in the origin is also clearly visible from 4000 Hz. Due to the designed curvature, from 8 m, a significant decay can be observed in the waterfall plot, especially in the higher frequency range.

## 3.3. Comparison of Mounting the LA in the Room and Close to Wall

In order to investigate the influence of a sound hard wall behind the speakers, the LA is placed 0.65 m in front of the room's back wall. Fig. 3.6 shows the waterfall diagram of this second configuration. Compared to Fig. 3.5, the boost at the lower frequencies within the first meter is unique. The observation of a higher overall level from section 3.1 can be confirmed, especially for greater distances. Even the 0 dB per dod, seem to exceed the 8 m boundary to 9.5 m for higher frequencies. An increase in level for the low frequencies is observable at the farthest measurement locations, which can be explained

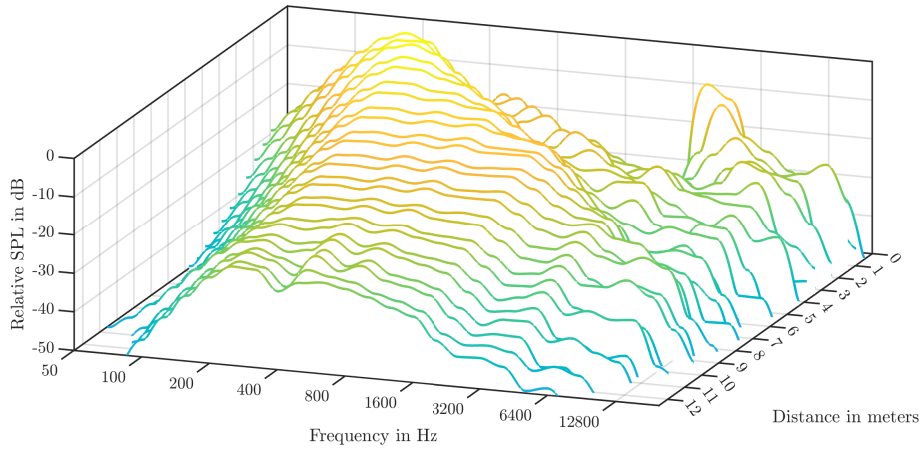


Figure 3.5.: Spectral decay in a waterfall plot, LA in the Room

by the better coupling of the room's eigenfrequencies (room modes) caused by the placement of the array in the pressure zone close to the wall. This gets underlined by the dip at 400 Hz at 12.5 m, which also indicates room resonance. This dip is also visible in Fig. 3.5 but is less pronounced in the freely positioned configuration.

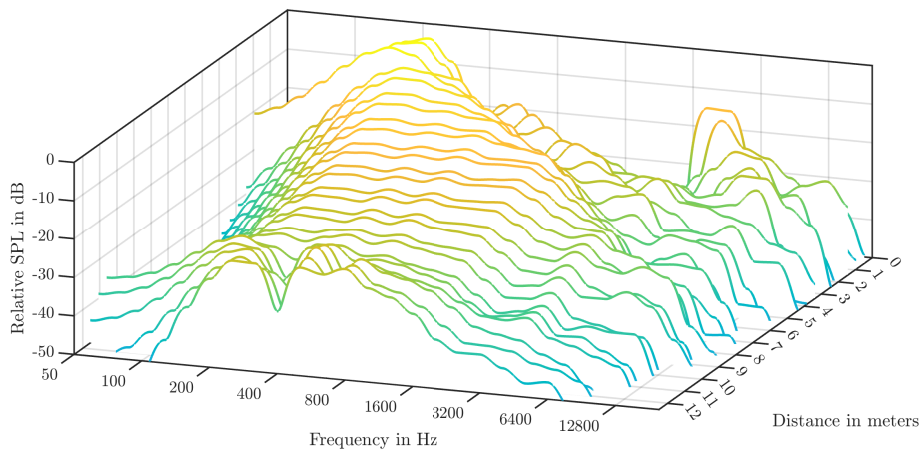


Figure 3.6.: Spectral decay in a waterfall plot, LA close to the wall

### 3.4. Electronically Adjustment of Decay Behavior

Driving each speaker with a separate amplifier and processing channel allows to electronically change the decay per dod after mounting and adjusting the splay angles by adding extra delay times, ascending towards the lower hanging speaker, the virtual curvature increases, and the decay per dod changes. As shown in Tab. 2.1, the *EnImSo Line-Array Designer* calculates extra delay times for specified decay behavior. For a decay of  $-3$  dB per dod, the resulting delay values are displayed in the fourth column of Tab. 2.1. By adding these sample delays to the IRs of the twelve speakers before summing up each probe location, the results correspond to an electronic delay during the measurement. Fig. 3.7 shows the SPL RMS over the distance, including the delay for the two configurations. The dotted black graph indicates the theoretical  $-3$  dB per dod, starting from 1.5 m, where the LA starts acting. In

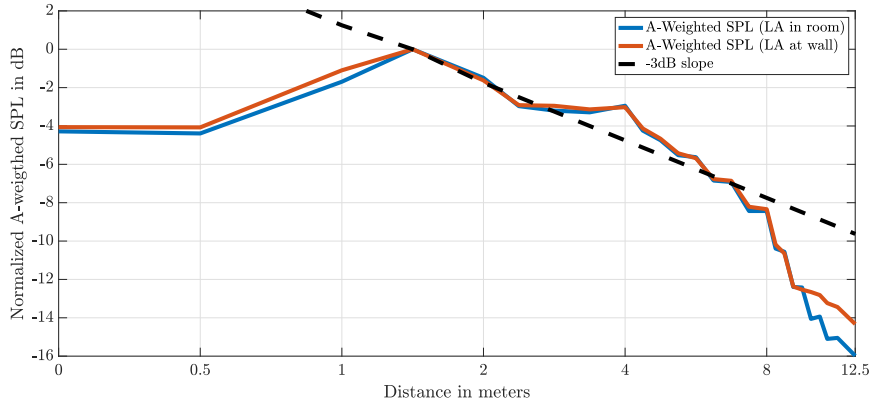


Figure 3.7.: A-weighted RMS SPL for the two configurations, electronically delayed to  $-3\text{ dB/dod}$

the distance of interest from 1.5 m to 7 m, it is visible that the delays lead to a decay of  $-3\text{ dB per dod}$ , except small outliers around 4 m of maximal 2 dB. As expected, exceeding the 8 m boundary, the SPL drops with the same decay of  $-12\text{ dB per dod}$ , as already observed in Fig. 3.1. Compared to the results without delay, it stands out that the SPL in the first half meter is 2 dB less with the delay applied. Without normalization (see Appendix A.1), it becomes clear that after 0.5 m, the SPL increases and exceeds the original configuration by 2 dB. Due to the delay, the theoretical curvature is higher, which causes a higher SPL between 0.75 m and 2.5 m and a lower SPL between 4 m and 8 m. In other words, the additional electronic curvature causes the  $-3\text{ dB per dod}$ . On the view of the

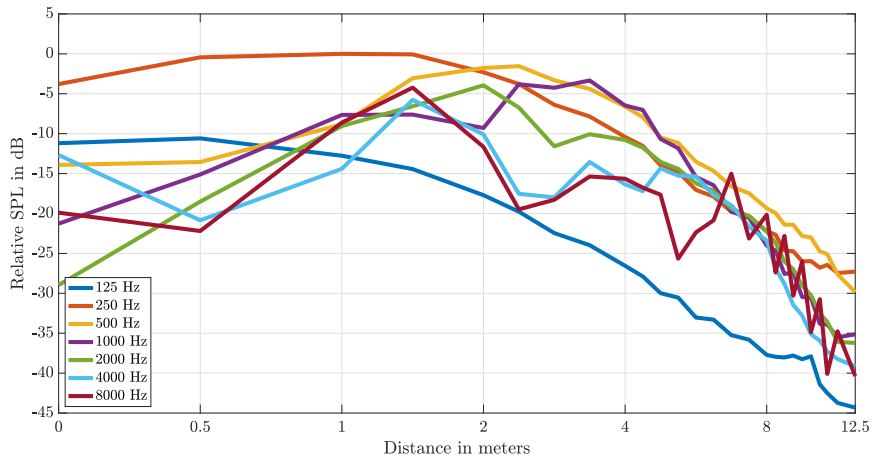


Figure 3.8.: 1/3-octave decay, LA in Room, electronically delayed to  $-3\text{ dB/dod}$

frequency behavior over the distance in Fig. 3.8, a steeper decay can also be observed for all frequency bands except for the 1000 Hz band. For 1000 Hz the decay firstly starts from 4 m distance. Compared to Fig. 3.3 and Fig. 3.1, the first point of impact of the LA seems to be closer to the array at 1.4 m, which is also explained by the more curved array.

## 4. Discussion & Conclusion

This chapter summarizes all the findings and further questions. A subjective listening comparison of the LA with a horn-driven point source was performed and is discussed in section 4.1.

### 4.1. Discussion

The results of the processed measurements in the laboratory show the advantage of a small-sized LA concerning coverage. The spectral unbalances over the observed area were visible in the measurements. A subjective listening evaluation was done by lifting the top speaker of the array to 3.7 m, such that the listening height is at 1.7 m. The spectral unbalance was mainly perceptible for frequencies between 1000 Hz and 8000 Hz at a distance between 4 m and 8 m. Especially the boost around 4000 Hz at 7 m distance (as observable in Fig. 3.4) was definitely audible.

As mentioned, the LA was compared to a point source speaker mounted in a ceiling corner. The direct sound frequency response was not equalized during listening, so the point source sounds slightly more balanced. Also, the highly elevated location of the point source and the reverberation in the room let the point source decay adequately compared to the array. Only from a distance of 6 m, the decay of the LA at high-frequencies was noticeably less compared to the point source. The spatial aliasing mentioned in section 3.2 below the LA is negligible as no listener is located there in a typical sound reinforcement application. The ripples at higher frequencies in the more distant measurement locations were not audible. The interference between the discrete membranes also causes these. Compared to the point source, having a 12-inch low-driver for a balanced sound experience, a low-frequency extension in addition to the LA is necessary.

### 4.2. Conclusion

The investigated miniature LA was measured in the laboratory, and the results were evaluated. The splay angles, calculated by the curvature design software *EnImSo Line-Array Designer*, led to the desired decay behavior for the overall SPL. Concerning single-frequency bands, the decay deviates from the design. It could be demonstrated that even with electronic delay, the decay could be precisely adjusted after mounting if the processing and amplification units are available. Due to the length of the examined LA, the lower frequency bound for line-source behavior was slightly higher ( $> 1000$  Hz) than expected ( $> 500$  Hz). This can be attributable to the room acoustics or the not exact discrete splay angles.

The effect of a wall in the direct vicinity behind the array only affected the low-frequency behavior of the LA. Compared to a point source, the advantage of the examined LA is the coverage over distance, especially in the higher frequency range. Due to the small speaker size and one chassis design, the spatial aliasing frequency is relatively high, but the low-frequency range is weaker than other systems.

# Bibliography

- [GZM23] L. Gölles, F. Zotter, and L. Merkel, “Miniature line array for immersive sound reinforcement,” in *Audio Engineering Society Conference: AES 2023 International Conference on Spatial and Immersive Audio*, Aug 2023. [Online]. Available: <http://www.aes.org/e-lib/browse.cfm?elib=22156>
- [Gö23] L. Gölles. (2023, Apr.) Line array designer. Institute of Electronic Music and Acoustics Graz. [Online]. Available: <https://enimso.iem.sh/post/line-array-designer/>
- [MBL07] P. Majdak, P. Balazs, and B. Laback, “Multiple exponential sweep method for fast measurement of head-related transfer functions,” *Journal of the Audio Engineering Society*, vol. 55, pp. 623–637, 07 2007.

# A. Appendix

## A.1. Not Normalized, A-weighted SPL RMS

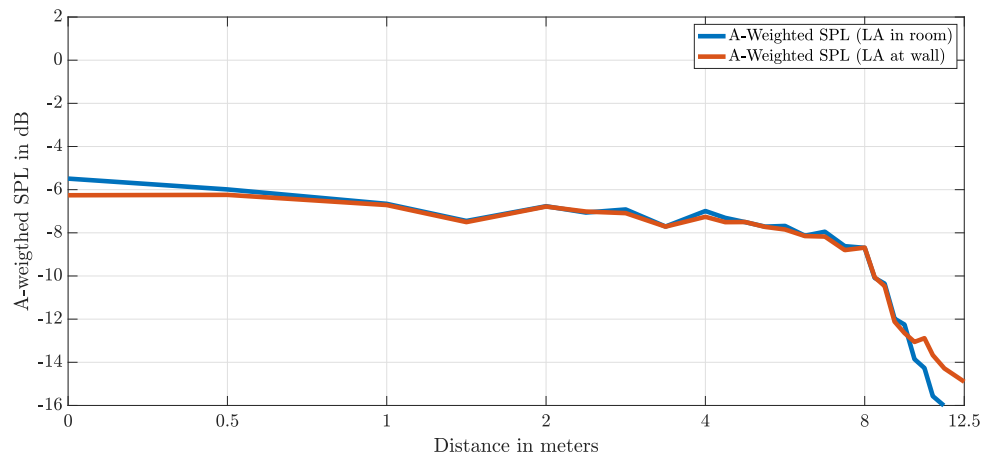


Figure A.1.: Not normalized SPL RMS

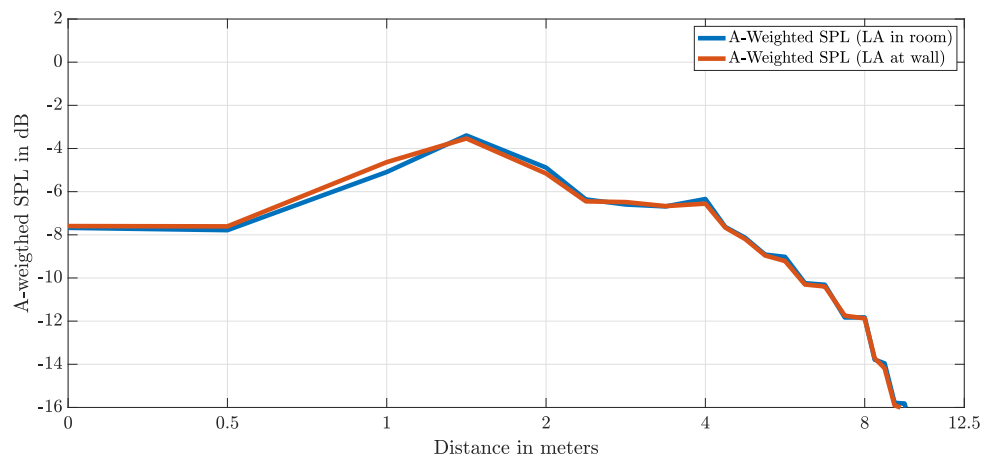


Figure A.2.: Not normalized SPL RMS, electronically delayed to  $-3$  dB/dod