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Estimation of the sound power of multiple sources using SPL measurements and room acoustic simulations

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ABSTRACT

The sound power of sources is a key input to room acoustic simulations. However, existing measurement methods are generally unpractical as they require specific equipment, for instance a reverberation room in ISO 3743. A method is proposed to find the sound power spectrum of sources for room acoustic simulations, based on in situ sound pressure level (SPL) measurements at known locations. The measured data is fitted to an ODEON simulation of the measurement space via an optimization algorithm. The method requires an accurate numerical model of the environment. A notable advantage is the possibility of estimating multiple sources at once with the same set of SPL measurements. The proposed method is investigated in a dry room in which up to three sources are active simultaneously. It is found that at least three receivers per source are required to obtain robust estimations, and it is beneficial to place them in the vicinity of the sources. The accuracy of the estimation also depends on the relative level between the sources, as a too strong source will mask a weaker one. Finally, dry environments are more suited to measuring multiple sources, due to larger spatial variations in SPL.

1. INTRODUCTION

The characterization of sound sources is an essential aspect in applications such as industrial noise control or room acoustics. In particular, the sound power is a source property that directly affects the sound pressure level (SPL) in a given environment. In a room acoustic simulation context, it therefore constitutes key input data to a model. Multiple standardized methods have been developed to measure the sound power level of sources. A guideline between these methods is provided in ISO 3740 [1]. Direct methods are based on the measurement of the sound field surrounding the source and correcting for the environment. In ISO 3741, a spatially averaged SPL is measured in a reverberation room, from which one can infer the sound power of the source [2]. Several other methods consist of evaluating the sound field on a surface around the source; they make use of either sound pressure measurements (ISO 3744-3746 in free-field [3–5], ISO 3747 *in situ* [6]) or sound intensity measurements (ISO 9614 [7]). Alternatively, ISO 3743 is a comparison method in which the SPL is measured in a reverberation room in the presence of the source under test and with a known reference source [8]. These methods require specific facilities such as a reverberation chamber, or advanced measurement equipment such as an intensity probe or a scanning device.

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This study investigates how room acoustic simulations can be used to retrieve the sound power of one or multiple sources, based on SPL measurements at different locations. The SPL is relatively easy to measure, as most acousticians have access to one or several sound level meters. The method relies on a numerical model of the measurement environment, in which the simulated SPL is fitted to the measured data by varying the power of the sound sources. In principle, this method can work in any environment, as long as an accurate model of the space is available. In addition, it has potential for measuring more than one source at once, which can be advantageous in noisy environments. The study focuses on the estimation of up to three sources simultaneously. It explores the number and the location of the SPL measurement points to optimize the method. The challenges with regard to multiple source estimations are also explored.

2. METHODOLOGY

Let us consider an acoustic space in which N incoherent sources should be characterized in terms of sound power. We assume that an accurate numerical model of the space is available, and that it contains the sources at their proper locations, albeit with unknown power. The sources can have any geometry, as long as they are properly represented in the numerical model (e.g., line sources if the simulation allows it). We consider the problem in one given octave band. The SPL is measured at M known locations, and it is expressed in terms of squared sound pressures in a vector $\mathbf{y}_{meas}$ of size M. Given a vector $\mathbf{x}$ of size N containing the sound power levels of all active sources in dB, one can obtain a set of simulated squared sound pressures at the measurement positions, $\mathbf{y}_{sim} = y(\mathbf{x})$, where $\mathbf{y}_{sim}$ is a vector of size M. The problem therefore requires minimizing a cost function f, defined as

$$f(\mathbf{x}) = ||y(\mathbf{x}) - \mathbf{y}_{meas}||^2. \tag{1}$$

According to the principle of superposition, the squared pressure at a given position is a linear combination of the contributions of each source, under the assumption that the sources are incoherent. The function y can be rewritten as

$$y(\mathbf{x}) = \sum_{i=1}^{N} a_i(\mathbf{x}_i) y_i(x_{0,i}), \tag{2}$$

where $y_i(x_{0,i})$ is the vector of squared pressures obtained at the M receiver positions when only source i is active, given an arbitrary initial power level of $x_{0,i}$, and $a_i(\mathbf{x}_i)$ is a scaling factor related to the power of source i. The function y_i contains the source-receiver paths from source i to each receiver. It depends on the geometry of the room, the selected materials, the receiver positions, the position of source i and its directivity pattern. However, it is independent of the source's power $\mathbf{x}_i$. The terms $a_i(\mathbf{x}_i)$ can be related to the power and the initial power of source i as follows,

$$a_i(\mathbf{x}_i) = 10^{0.1(\mathbf{x}_i - x_{0,i})}, \tag{3}$$

and $y_i(x_{0,i})$ is expressed in terms of SPL L_i,

$$y_i(x_{0,i}) = p_0^2 10^{0.1 L_i}, \tag{4}$$

where p_0 is a reference pressure. Equation 2 then becomes:

$$y(\mathbf{x}) = p_0^2 \sum_{i=1}^{N} 10^{0.1(L_i + \mathbf{x}_i - x_{0,i})}, \tag{5}$$

so the objective function is

$$f(\mathbf{x}) = ||p_0^2 \sum_{i=1}^{N} 10^{0.1(L_i + \mathbf{x}_i - x_{0,i})} - \mathbf{y}_{meas}||^2. \tag{6}$$

In practice, the function is normalized by p_0^2, which can then be omitted in the formulation.

The proposed methodology consists of two steps. The first step calculates the source contributions for a given set of source powers, i.e. L_i for known $x_{0,i}$ values (typically 0 dB for every source). The second step is the actual search that aims at minimizing Equation 6, by varying the vector $\mathbf{x}$. The first step is independent of the unknown source power, and it requires the computation of each source-receiver path in the numerical model. This makes it quite expensive computationally, but it must be run only once. The second step corresponds to varying $\mathbf{x}$ in Equation 6. The evaluation of $f(\mathbf{x})$ is then very cheap, which makes it suited to optimization algorithms. Besides, analytical formulas for first and second derivatives of the cost function follow from Equation 6, which makes convex optimization methods attractive to handle the problem.

Sound power levels in dB are used as inputs of the optimization problem, as they allow for a wider search range of solutions. However, this makes the problem non-linear, so common search algorithms such as linear regression cannot be used in that case. Instead, a trust region (TR) solver from the SciPy library is used to solve the problem (see [9] for a description of the solver and [10] for insight on TR methods in solving non-linear constraint optimization problems).

As an additional step, the solution $\tilde{\mathbf{x}}$ is scaled so that the simulated and measured square pressures correspond to the same energy, i.e. $||y(\tilde{\mathbf{x}})|| = ||\mathbf{y}_{meas}||$. The scaling factor is typically very close to 0 dB, which occurs when $f(\tilde{\mathbf{x}}) \approx 0$.

Note that the span of all possible power values does not complete $\mathbb{R}^M$, meaning that for unrealistic values of $\mathbf{y}_{meas}$, there could be no configuration of powers $\mathbf{x}$ achieving $f(\mathbf{x}) = 0$. Nevertheless, this should not be an issue if the measurements are accurate and the room conditions are correctly modeled in the simulation. For the problem to have a physically acceptable solution, it is recommended to have more measurement points than unknown sources. This makes the problem overdetermined. Sound pressure levels are typically displayed in dB, but they are converted to squared pressures in the optimization (linear scale). Note that the sound powers are still expressed in dB, which makes the objective function non-linear. The problem is solved independently for each octave band.

3. EXPERIMENTAL TESTS

The method was tested with up to three simultaneously active sources in an acoustically treated room.

3.1. Setup

Experimental tests were conducted in a rectangular laboratory room of dimensions 5.9 m×5.8 m×3.1 m. The walls and the ceilings are covered by porous absorbers, which results in a relatively low reverberation time (around 0.3 s). The Schroeder frequency of the room is 147 Hz. An ODEON model of the room was created for the simulation of SPL data. In order to improve the model's accuracy, ODEON's genetic material optimizer was used, where the absorption coefficients of the walls are adjusted to fit measured and simulated room acoustic parameters [11]. Early Decay Time (EDT), Reverberation Time (T_{30}), Centre Time (T_S), Clarity (C_{80}) and Definition (D_{50}) were used as parameters in the optimization, as suggested in [11]. The measured parameters were derived from impulse response measurements in the actual room. The geometrical model and the comparison between the adjusted model and the measurements are shown in Figure 1. A fair agreement is found between 250 Hz and 4000 Hz, where the Just Noticeable Difference (JND) error is between 1 and 3. The 63 Hz and 125 Hz bands are below the room's Schroeder frequency, and ODEON simulations are not fully reliable in this frequency range, as they cannot represent the room's modal behavior. This can explain the larger differences between the simulated and the measured room acoustic parameters, even after adjusting the boundary conditions. At 8 kHz, the larger discrepancy can be attributed to a lower signal-to-noise ratio in the measurements, as well as the lack of initial data for material absorption in this octave band (manufacturer data usually stops at 4 kHz).

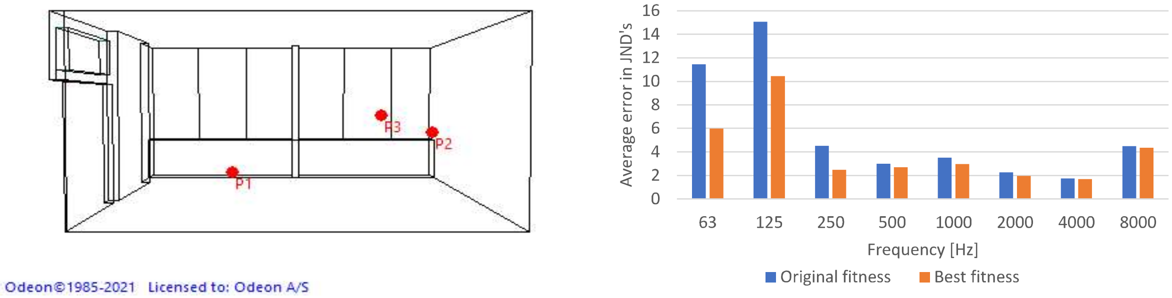

Figure 1: Left: ODEON model of the test room, including three source positions. Right: outcome of the material optimization in terms of JNDs (Just Noticeable Difference).

Up to three sources with different sound powers, are active in the test room. Their location is shown in Figure 1. These three sources are modeled as point sources in ODEON.

- Source 1 is an aerodynamic reference source from Brüel & Kjær type 4204 [12], placed on the floor.
- Source 2 is an omnidirectional loudspeaker source, Odeon Omni [13], which plays a stationary pink noise. Note that the sound power depends both on the loudspeaker's frequency response and the played signal.
- Source 3 is another Odeon Omni source, but it plays a noise signal with a modified spectrum.

The SPL is measured at different locations using 3 different sound level meters, directly in octave bands. An example is shown in Figure 2. For each source, we use 3 receiver positions relatively close to the source (approx. 1 m), 3 slightly further (approx. 2 m), and 2 much further (approx. 4 m). As only 3 sound level meters were available, the measurements are taken 3 by 3 consecutively, under the assumption that the sound field is stationary. The L_{eq} value is read on the sound level meters after 30 seconds of time-averaging for each position.

Figure 2: Picture of the experimental setup. Three sound level meters close to Source 2 (Odeon Omni).

3.2. Estimation of one source

The B&K source is estimated with different combinations of receivers, situated approximately between 1 m and 4 m from the source.

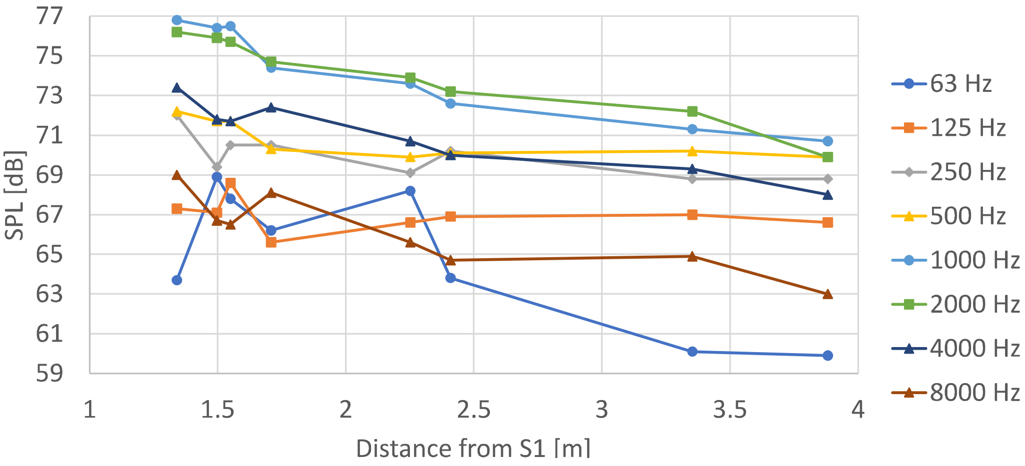

Figure 3: Measured SPL data in octave bands when Source 1 is active, plotted as a function of distance to Source 1.

The measured SPL data is displayed in Figure 3, as a function of distance from Source 1. In all octave bands, spatial variations are visible. The 500 Hz band is the one with the least variations across bands, with only 2.3 dB difference between the minimum and maximum values. At high frequencies, a range of about 6 dB is recorded. Generally, the dependence of SPL with distance is complex, as it also depends on the reflections in the room, and hence its boundary conditions. Below Schroeder's frequency (147 Hz), we expect a relatively strong modal behavior, which can explain the large SPL variations for the 63 Hz band. From 125 Hz to 500 Hz, the measured SPL plateaus after about 2 m. In these bands, the SPL becomes fairly uniform when sufficiently far away from the source. This could indicate that the sound field in the room is fairly diffuse in these frequency bands. At 1 kHz and above, the measured SPL decays with distance, which is explained by the high sound absorption of the walls and ceilings in these octave bands. The sound field is then comparable to a free field case.

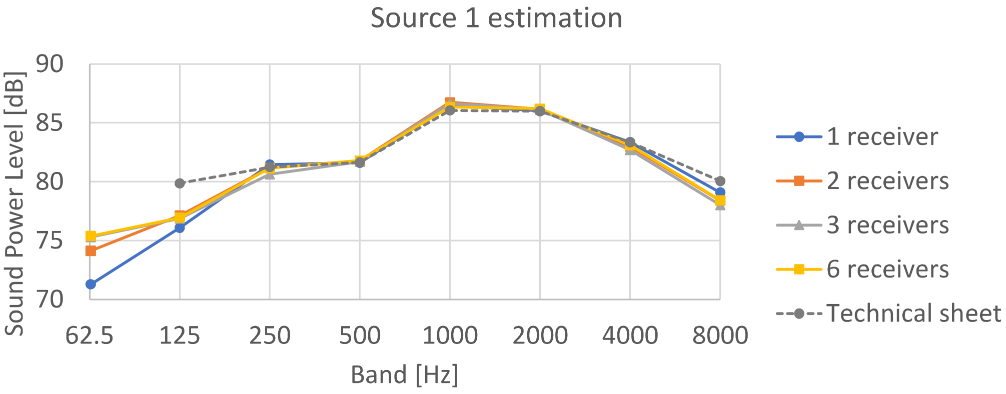

Figure 4: Estimation of Source 1 with different numbers of receivers. Comparison with reference data from B&K's technical sheet [12].

Figure 4 shows examples of the estimation of the source's power level for different numbers of receivers. As reference, sound power data from the source's technical sheet is also plotted [12]. For one receiver, the estimation is accurate between 250 Hz and 4 kHz. Larger errors can be seen in octave bands with larger differences between the model and reality, as already evidenced in Figure 1 (below Schroeder's frequency and at 8 kHz). Figure 3 also shows that the measured SPL is overall lower at 63 Hz, 125 Hz and 8 kHz, which makes the estimation more sensitive to background noise for these octave bands. Using more receivers leads to a more accurate solution, especially in the 125 Hz band, although the estimated power level is still 3 dB lower than the technical sheet value. Very little difference is found between the estimation with 3 receivers and with 6 receivers, which indicates that the algorithm already yields its best estimation for 3 receivers.

As an additional test, different combinations of up to 5 receivers were computed. Figure 5 shows the standard deviation for each number of receivers. As expected, using more receivers leads to less variability in the data, hence more precise results. If we exclude the 63 Hz, band, the standard deviation is below 0.5 for all bands when at least 3 receivers are included in the optimization problem. This also indicates that the estimation becomes sufficiently repeatable for 3 receivers. Using more receivers still slightly improves the precision.

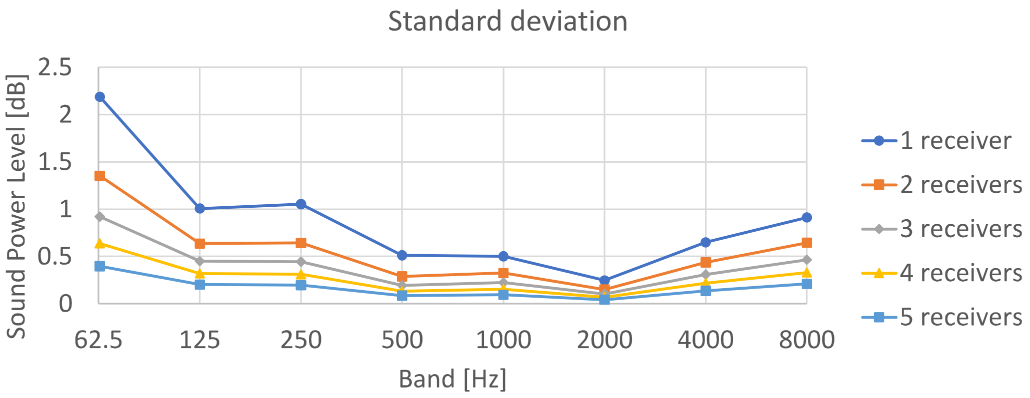

Figure 5: Estimation of Source 1 with different numbers of receivers. Standard deviation for different number of receivers.

According to these results, three receivers are sufficient to obtain an accurate and robust estimation of one given source power. However, some differences can still be seen between the technical data and the estimation (even with up to 8 receivers in the model). These differences can be due to mismatch between the numerical model and reality (calculation method, boundary conditions, source properties, background noise). The estimation is the best the optimization algorithm can yield, given the input data. Therefore, in the rest of the study, we use the estimated sound power with a large number of receivers (8) for each of the 3 sources, as shown in Figure 6. It is seen that Source 1 (B&K) has the highest power. Source 2 and Source 3 have comparable levels, but they have slightly different powers, as they play back different signals.

3.3. Estimation of two sources

Now that the three sources were estimated separately in order to obtain reference data, we study a scenario where two sources are simultaneously active, namely Source 1 and Source 2. According to the findings of Section 2, we study different cases with at least 6 receivers, i.e. 3 receivers per source:

- 6 receivers close to Source 1
- 6 receivers close to Source 2
- 3 receivers close to Source 1, 3 receivers close to source 2
- 6 receivers close to Source 1, 6 receivers close to source 2

The last case has more receivers than the other ones, as it can show whether using more measurement data can improve the estimation. We use the estimation data from Figure 6 as reference for each source power level.

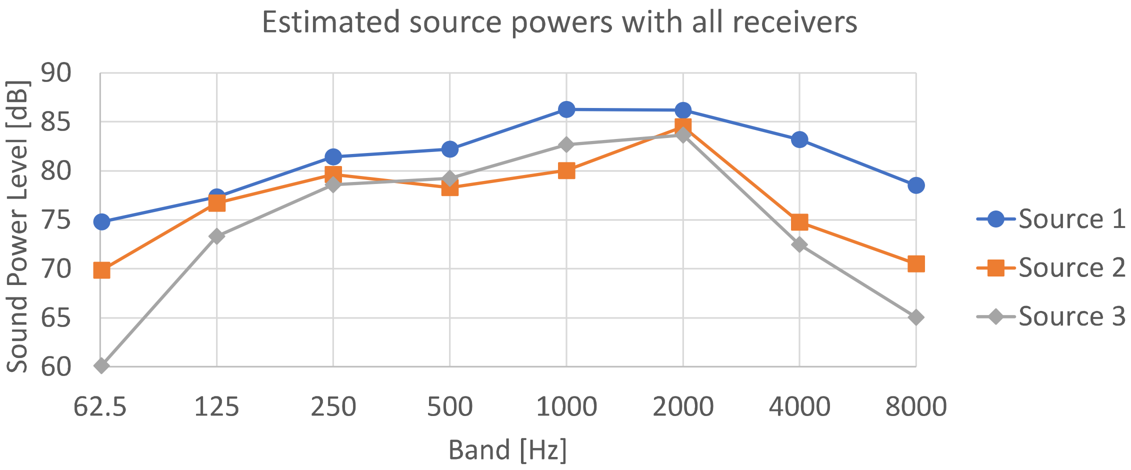

Figure 6: Estimation of Source 1, Source 2 and Source 3 with all receivers. This data is used as reference in the rest of the paper.

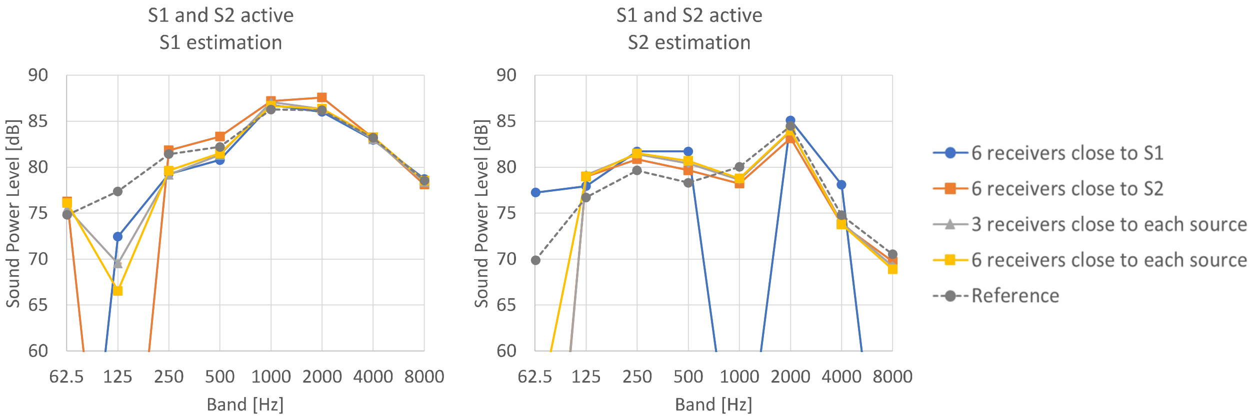

Figure 7: Estimation of Source 1 and Source 2 with different receiver combinations.

The estimated powers for the four cases are presented in Figure 7. With 6 receivers close to Source 1, Source 1 is well estimated from 1 kHz and above. However, at lower frequencies, its estimated power is slightly lower than the reference. Note that a way too low power is obtained at 63 Hz, but this can be attributed to the mismatch between the model and reality at low frequencies. The estimation of Source 2 is not as accurate, with the largest errors seen at 63 Hz, 1 kHz and 8 kHz. An explanation is the difference in power between the two sources. For instance, at 1 kHz, Source 1 is expected to have a power of about 86 dB and source 2's power level should be 80 dB. Therefore, the sound field at the receivers close to Source 1 is mostly due to the contribution of Source 1, so the algorithm tends to discard Source 2. Positioning receivers closer to Source 2 leads to a better estimation of Source 2, between 125 Hz and 8 kHz. The estimation of Source 1 is not too affected, because overall it has a higher power, so its contribution is still significant at the receiver positions close to Source 2. Still, Source 1 is completely discarded at 125 Hz, where Source 2 is slightly overestimated, but as discussed earlier, the estimation is not reliable in this frequency band. With receivers distributed both close to Source 1 and Source 2, both sources are fairly well estimated. Below 1 kHz, the algorithm tends to overestimate Source 2 and underestimate Source 1. This could indicate that the algorithm tends to favor comparable powers between the sources. This is particularly pronounced at 125 Hz, where again the numerical model is not entirely accurate. With 12 receivers, the estimation of both sources is quite comparable with the previous case, the main differences being visible below the Schroeder frequency (63 Hz, 125 Hz).

According to these examples, the two sources can be estimated simultaneously, and the result is most reliable if the SPL is measured in the vicinity of both sources. The problem is nevertheless more challenging than the estimation of a single source, and one can expect a few dBs of error.

3.4. Estimation of three sources

In this section, the three sources are active simultaneously. As done in Section 3, the SPL is measured at 3 receiver locations per source, i.e. 9 receivers in total. In addition, the estimation is also made for 6 receivers per source (18 receivers in total), so as to investigate the influence of more measurement data.

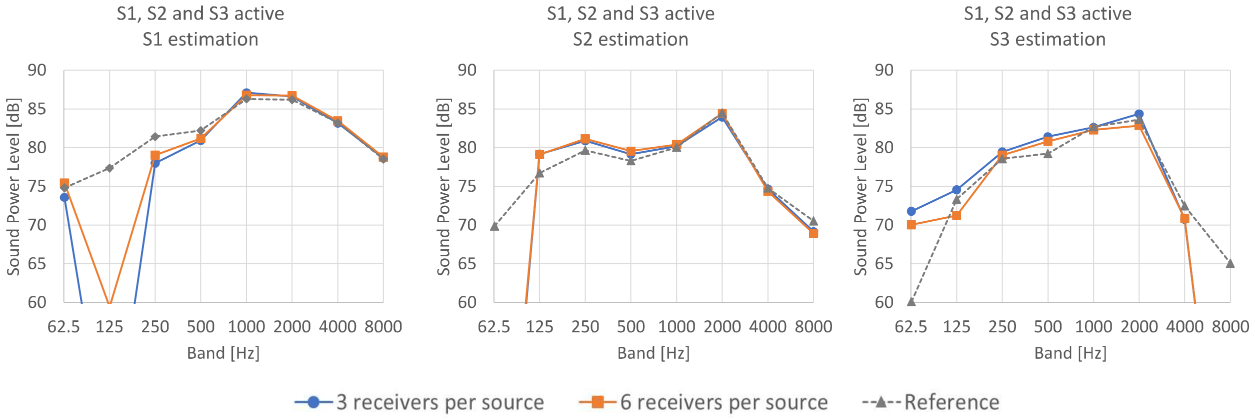

Figure 8: Estimation of Source 1, Source 2 and Source 3 with different receiver combinations.

Figure 8 shows the estimation of the power of three sources, compared with the reference power measured independently from each source (from Figure 6). Source 1 is fairly well estimated from 500 Hz and up. At lower frequencies, it tends to be underestimated. Source 2 is also quite accurate from 250 Hz and up, with more uncertainties at low frequencies. Source 3 is slightly less accurate than the two others, probably because it has a relatively lower power than the two others. In particular, it is completely discarded at 8 kHz, where the power of Source 3 is significantly lower than the two other sources, according to Figure 6. Therefore, the sound field can be already well explained with only the contributions of Sources 1 and 2. In that sense, Source 3 is masked from the solution in this octave band. Only minor improvements can be seen when using 6 receivers per source, e.g. for Source 1 at 125 Hz, but it is still greatly underestimated.

Once again, the results show that three receivers close to each active sources should be sufficient to properly estimate 3 sources simultaneously. The estimation is particularly challenging at low frequencies, due to the mismatch between the model and reality. If the power is very different between the sources, the algorithm also tends to favor stronger sources while discarding weaker sources.

4. DISCUSSION

The results of the previous section show that the proposed methodology makes it possible to estimate noise sources simultaneously, provided that sufficient relevant SPL data is measured.

One challenge for multiple source estimation is characterizing sources with very different powers. In such a case, some regions may only be affected by the strongest source in terms of SPL, which makes it difficult to estimate the weaker source. As an illustration, we revisit the case with 6 receivers close to Source 1 from Figure 7. At 1 kHz, Source 2 was attributed a very low power level. Figure 9 taken from the ODEON software shows the estimation of Source 2. The black vertical bar indicates the range of power variation which allows the cost function to remain within 10% of its optimal value. At 1 kHz, the bar is extremely large, which shows that the power Source 2 is completely insignificant in fitting the measured and simulated SPL data. This illustrates that the setup is not suited to estimate the weaker source. Measuring the sound field close to the weaker source can mitigate the issue in some cases, as the weaker source contributes more prominently to the sound field in that region. Nevertheless, it can also be preferable to estimate the sources separately (i.e. switching off the stronger source) if a better accuracy is required.

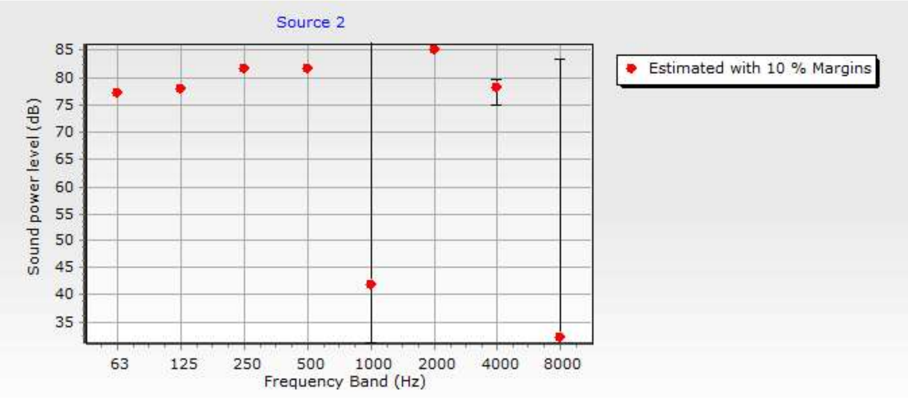

Figure 9: Estimation of Source 2 when Source 1 and Source 2 are active, and 6 receivers are placed close to Source 1. ODEON's output graph.

An advantage of the tested setup is that the environment is acoustically dry. Due to sound's spatial decay, considerable variations of SPL can be measured across the room, as evidenced in Figure 3, which facilitates the solving of the problem. In a too reverberant environment, the method is not expected to perform as well, as the SPL will be more uniform across the room, which means that the same information is measured at any location. This is also why it is recommended to measure SPL close to the sources, so as to favor the influence of direct sound and reduce the amount of mixing between different sources in the measurements.

Another key aspect is the accuracy of the numerical model. This includes the geometrical model, the choice of boundary conditions, the sources' properties (e.g., directivity) but also the calculation algorithm. In the present study, ODEON was used to calculate the sound propagation from sources to receivers. As a hybrid geometrical acoustic algorithm, it is not well-suited to frequency bands with low modal overlap, which explains the errors at low frequencies in the presented results.

5. CONCLUSION

A method was presented to estimate the sound power of one or multiple noise sources in a given environment. The method requires the *in situ* measurement of SPL at different locations, and it relies on fitting this measurement data to the output of a numerical acoustic simulation of the same space (ODEON, in the scope of this paper).

The presented experimental results show the validity of the method and lead to the recommendation of at least three SPL measurements in the vicinity of each active source, in order to obtain a robust and accurate sound power estimation. The estimation becomes challenging at low frequencies, due to model mismatch, as well as in regions with lower SNR. In addition, it is more difficult to estimate sources with different powers, as the weakest sources tend to be masked by the stronger ones.

The optimization problem could be facilitated by adding prior knowledge to the search. This is case dependent, but it could include the directivity of the sources, smoothing across frequency bands, or assigning a predefined spectrum if the source's nature is already known.

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