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Total Sound Power Estimation of Simultaneous Loudspeakers using Near-Field Pressure for Automatic In-Room Equalization

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ABSTRACT

This study presents an effective way to estimate the total sound power (TSP) of multiple loudspeakers. The TSP is associated with the average response of all loudspeakers in the whole room. This computation is useful for equalizing the loudspeakers and counteracting the adverse influences of the room in a global manner. The TSP is normally obtained with measurements on a large number of locations in the room. A simulation study presented in this paper suggests that in order to obtain an accurate TSP measurement, more than 36 microphone positions are needed. Previous studies have shown that the TSP of a single loudspeaker in the room can be estimated by using the near-field (NF) pressure and the velocity of the loudspeaker which can be obtained by various well-known methods. Typical audio reproduction with multiple subwoofers or full-range loudspeakers make use of "bass management". This concept simultaneously plays back the same low-frequency content e.g. below 80–200 Hz over the single or multiple subwoofers/loudspeakers. In this study we propose a method to estimate and equalize the TSP produced by multiple loudspeakers with a minimum number of microphones, e.g., for two loudspeakers, using only two NF microphones, one attached to each loudspeaker. The estimation is performed by obtaining simultaneous or sequential measurements of the multiple loudspeakers and computing the correct summation of the NF pressure responses of each of the loudspeakers. Encouraging results were obtained in a standard listening room with commercially available loudspeakers, predicting the TSP from 40–200 Hz with an accuracy of 2.7 dB SD.

1 Introduction

The response of a loudspeaker is severely altered when radiating sound in a room. The magnitude of the frequency response can show peaks and valleys up to 20 dB, especially in the frequency range where the wavelengths are comparable with the room dimensions (e.g., between 20 and 400 Hz). The interaction of sound waves with the boundaries of the room, build distinct zones with high sound-pressure levels, related to the room resonances, and zones with low sound-pressure levels related to anti-resonances where the sound is self-canceling. Previous studies have made use of multiple loudspeakers to control the sound field in the room. Santillán [1], and the author worked this problem around by constructing a traveling plane wave across the room, and neutralizing it at the other end of the room with extra loudspeakers in counter phase [2].

Multiple loudspeakers and signal processing were used to optimize the sound field in the room in [3]. Hill and

Hawksford [4] utilized a multiple-source array to exploit many degrees of freedom, e.g., an omnidirectional, and a multiple-driver box with three bipolar patterns to control the sound-field response over a listening area.

Several studies have introduced the room correction concept by measuring the combined loudspeaker-room frequency response [5, 6, 7]. A multiple-input multiple-output (MIMO) approach for room compensation over a 84×33×30 cm listening volume in the room using several support loudspeakers was proposed in [8]. A subjective evaluation of some of these methods has been presented by Olive et al. in [9]. The common characteristic of most of the cited approaches is that they make use of a number of microphone locations to measure the transfer function from the electrical loudspeaker terminals to one or more locations in the room, commonly called listening locations or "listening area". These methods can be prone to human error or customer inconvenience.

In order to equalize the response of the loudspeaker in the entire volume of the room, the total sound power (TSP) output radiated by the loudspeaker into the entire room is required, [10, 11]. The use of two microphone positions close to the driver's diaphragm to compute the TSP radiated by the loudspeaker into the room was proposed in [10]. The use of only one microphone positioned near the loudspeaker to estimate the TSP was proposed by the author in [12].

In audio reproduction it is common practice to use multiple loudspeakers intended to simultaneously reproduce the same low-frequency content, often called "bass management". To reduce the negative effect of the room, a solution to automatically assess the average response of the loudspeakers in the entire room volume without cumbersome measurements is needed.

In this paper we present a method to automatically estimate the TSP of multiple loudspeakers working simultaneously in a room with a minimum number of microphones (one per loudspeaker). In Section 2 the method to estimate the TSP is described, while in Section 2.4 loudspeaker room simulations are presented to test the TSP estimation. In Section 3 the results are presented using simulations and real loudspeaker-room measurements. In Section 4 the work is summarized, discussed and preliminary conclusions are presented.

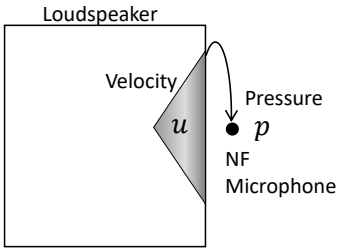


Fig. 1: Diagram of loudspeaker, velocity, and NF mic.

2 Methods

2.1 Room-Loudspeaker Total Sound Power

The total sound power radiated into a room depends on the source, its properties and location, and on the properties of the room such as its size, shape and sound absorption [13]. The $TSP(f)$ radiated by a loudspeaker into the room in the frequency domain can be written as:

$$TSP(f) = \tfrac{1}{2} Real\left\{p(f)u^*(f)\right\}, \tag{1}$$

where $p(f)$ is the complex amplitude of the sound pressure at the source, and u^* is the complex conjugate of the velocity of the driver in the loudspeaker enclosure [10, 12]. When pu^* is real (pressure in phase with velocity) all the power is transmitted into the room, and when pu^* is imaginary (pressure and velocity in quadrature), no power is transmitted [12]. For spherical sound waves the complex particle velocity component u_r in the radial direction from the source can be calculated from equation:

$$u_r = \tfrac{p}{\rho c}\left(1 + \tfrac{1}{jkr}\right), \tag{2}$$

where p is the complex pressure, ρc is the characteristic impedance of air, r is the distance from the observation point, $k = \frac{2\pi}{\lambda}$ and j is the imaginary unit. We can note that far from the source, the sound pressure and particle velocity are in phase, just as in a plane wave, but when $kr << 1$, the particle velocity and the sound pressure are almost at 90°out of phase [14].

2.2 Total Sound Power Measurement

To obtain the actual TSP radiated by the loudspeaker into the room, the root-mean-squared (RMS) sound

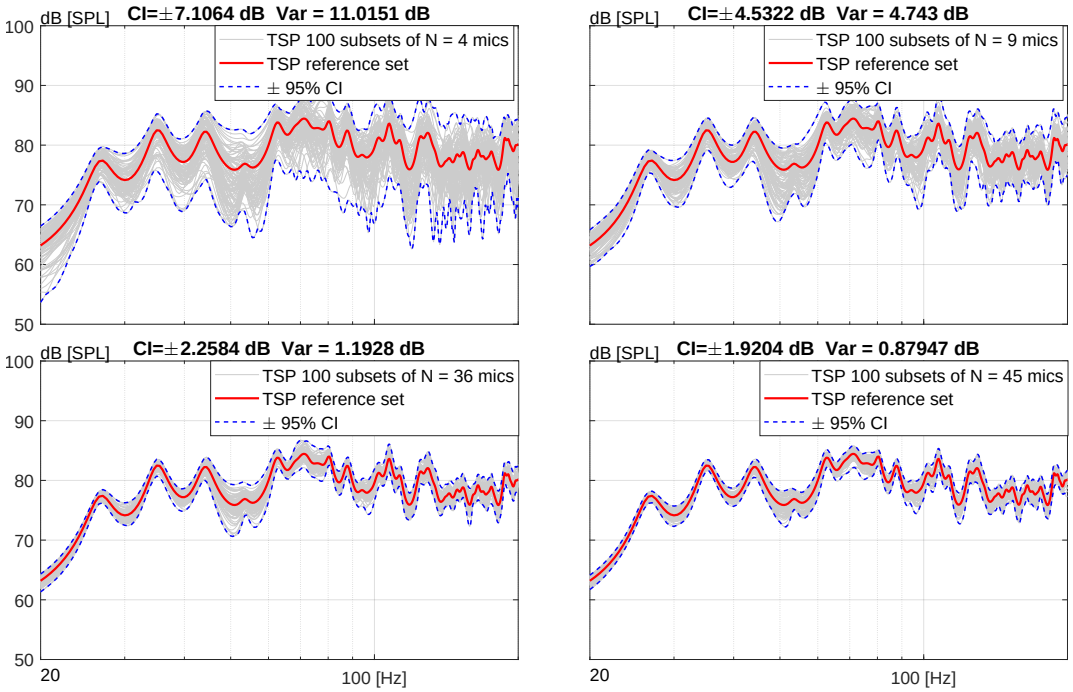


Fig. 2: FDTD Small Room simulation results, loudspeaker position S3, TSP computed on 100 subsets of N microphone positions. Upper left, using sets of N=4 mics. Upper right, using sets of N=9 mics. Bottom left, using sets of N=36 mics. Bottom right, using sets of N=45 mics.

pressure level at a number of microphones randomly distributed in the room can be computed as:

$$TSP_{actual}(f) = \sqrt{\frac{1}{N}\sum_{i=1}^{N}|p_i(f)|^2}, \qquad (3)$$

where p_i is the sound pressure in dB at N number of microphones randomly located in the room at discrete frequencies f.

2.3 Number of Microphones for Actual TSP

Concerning the number of microphones needed to obtain a reliable "true" TSP measurement, a simulation study has been performed using the finite-difference time domain (FDTD) method, (see [15]). On this study five typical listening rooms have been simulated, included three of the standard rooms at the SRA Audio Lab. A compact loudspeaker has been set up on commonly used loudspeaker room locations S1, S2 and S3. The FDTD method is based in the time domain wave propagation inside the enclosure, which is discretized in a dense grid of pressure and velocity points. The impulse responses (IRs) on 7292 regularly spaced pressure points (virtual microphones) have been saved for the reference TSP computation grid. Points near

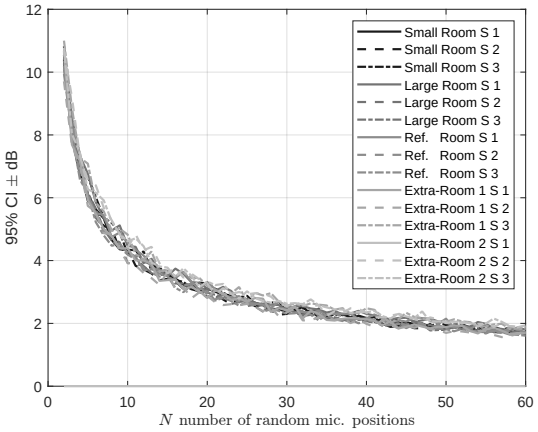


Fig. 3: 95% CI, 100 subsets of N mics vs. reference TSP. Five rooms with 3 loudspeaker positions.

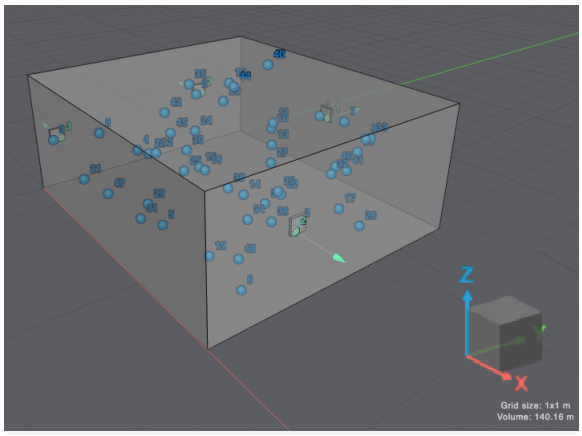
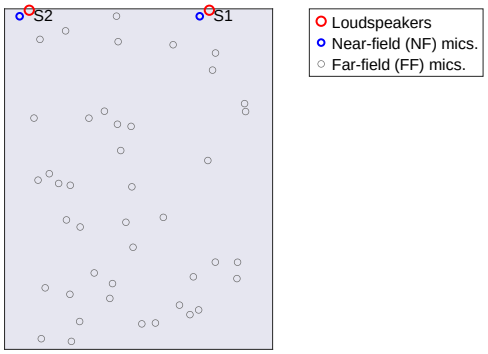
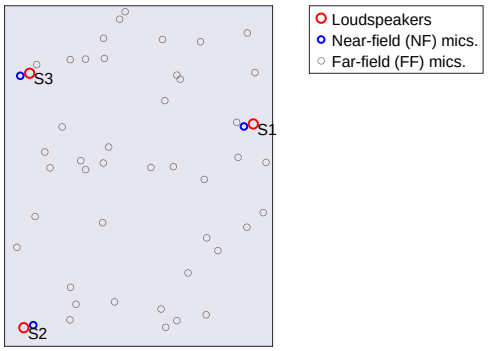


Fig. 4: Treble frame loudspeaker room simulation.



(a) Setup 1.



(b) Setup 2.

Fig. 5: Room simulations Setups: 5a) S1 and S2. 5b) S1, S2 and S3. Red circles, loudspeaker positions. Blue circles, NF mics. Gray circles, FF mics.

the sound source were discarded. From the reference grid, 100 separate subsets of N random microphone positions were chosen, including different heights. The SPL frequency responses from 20–200 Hz were used to compute the TSP using Equation (3).

In Figure 2 an example of the TSP, S3 loudspeaker position in "Small Room" computed over the 100 sets of N=4, N=9, N=36 and N=45 number of randomly chosen microphones are shown. The statistical analysis of the error has been performed per frequency bin. Since we have verified that these error distributions are not normal, bootstrapping was used by resampling 1000 times each frequency bin data with replacement. Then the 2.5 th and 97.5 th percentiles were computed as the CI bounds with the functions `bootci` and `prctile` in Matlab. This method is reliable and versatile because it doesn't assume a specific data distribution. In Figure 3 the error results of the five rooms are summarized. The average CI (over the frequency range between 20–200 Hz) between the computed TSP over the 100 sets and the reference TSP is compared against the N microphones required for the actual TSP computation. As observed in Figures 2 and 3, by increasing the number of microphones the TSP computation gets closer to the reference TSP. By using 36 microphones randomly distributed in the room the error start to decrease from 2.3 dB down to 1.8 dB by using 60 microphones.

2.4 Loudspeaker-Frame Room Simulations

The acoustic room simulation software Treble SDK [16] was utilized to test the estimation of TSP. A geometric model and sound absorption characteristics of the actual standard "Large Listening Room" of dimensions; L= 7.32 m, W= 6.4 m, H= 2.74 m was imported to the software. A simplified geometric model from a commercially available loudspeaker "Music Frame" was imported with only the left woofer on the enclosure (as seen in Figure 4). The actual Music Frame loudspeaker uses two 87 mm woofers sharing the same enclosure, for faster computation time we decided to

Table 1: Simulations: loudspeaker center coordinates

	Loudspeaker Pos.	Coordinates (x,y,z) m
Setup 1	S1	0.03, 4.89, 1.596
Setup 1	S2	3.75, 6.37, 1.596
Setup 2	S1	2.54, 5.94, 0.850
Setup 2	S2	6.94, 4.81, 0.850
Setup 2	S3	6.92, 0.46, 0.820

include only one woofer for the simulation study. This loudspeaker is normally hung on the walls with a fixture and two stoppers having the loudspeaker driver as close as 3 cm facing the wall. This loudspeaker has an embedded array of four near-field (NF) microphones equally spaced in the bottom part, along the width of the frame for voice control commands. Two setups within the same room dimensions were simulated as detailed in Table 1 and Figure 5. For the actual TSP computation 45 far-field (FF) randomly located (including height) microphones were set in the room. For the NF microphones the left-most receiver was set on each loudspeaker geometry to estimate the TSP. A 3D sub-model geometry of the frame loudspeaker was imported. The Treble engine was set to use a boundary velocity layer source together with the wave-based solver. This enables highly accurate sound source characteristics in the low frequencies including near-field effects [16]. In the software the loudspeaker source had a flat frequency spectrum from 20 to 500 Hz. Then the source was defined by a 2nd-order high-pass and a 10th-order low-pass both Butterworth filters processed in a forward/backward manner to zero out the phase of the filter. The output results of the simulations were 2 s length IRs sampled at 32 kHz fs sampling rate. The speed of sound c was set to 343 m/s.

2.5 TSP Estimation of Single Loudspeaker

As described in Section 2.1, to estimate the TSP of a single loudspeaker in a room the NF pressure and the velocity of the driver in the enclosure are needed to use in Equation 1. The volume velocity of the simulated loudspeaker is calculated by using a Dirac delta function $\delta(t)$ filtered in cascade by a 2nd-order high-pass filter with fc at 20 Hz, and a 10th-order low-pass filter with fc at 500 Hz, both Butterworth filters processed in a forward/backward manner to zero out the phase of the filter. The Fourier transform of the band-pass filtered Dirac is then divided by the angular frequency $\omega = 2\pi f$ multiplied by the inverse of the imaginary unit j as in

$$U(f) = \frac{\delta(f)}{j \cdot 2\pi \cdot f}, \tag{4}$$

where $\delta(f)$ is the frequency domain band-pass filtered Dirac delta function. Then $U(f)$, and the transfer function at the NF virtual receiver as $p(f)$, in the frequency domain, are evaluated in Equation 1 to compute the TSP. In Figure 6 the estimated TSP of loudspeaker S2 of Setup 1 (see Table 1) with only one NF microphone

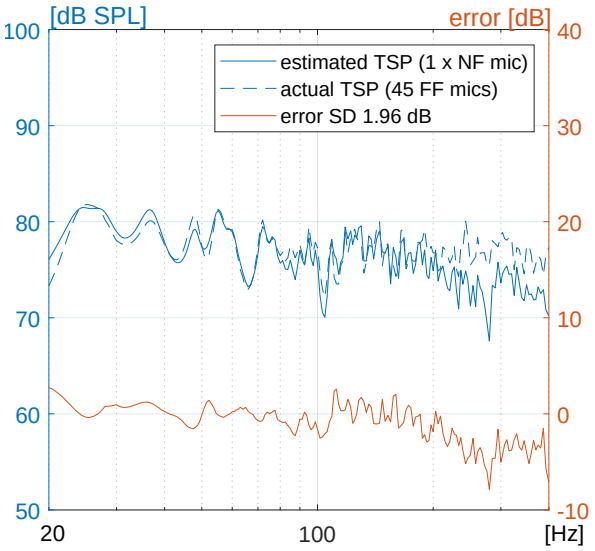


Fig. 6: Simulations: Loudspeaker position S2 on Setup 1. Estimated TSP with one NF mic. (continuous curve). Actual TSP computed with 45 FF mics. (dotted curve). Error (lower continuous curve).

is shown. In the same figure the actual TSP computed with 45 FF microphones is presented. As seen in the error curve between estimated and actual TSP of Figure 6, the accuracy below 400 Hz is about 1.96 dB SD and about 1.15 dB SD below 200 Hz.

2.6 TSP Estimation of Multiple Loudspeakers

When two identical loudspeakers S1 and S2, with driver velocities $u_1 = u_2$, are utilized to reproduce simultaneously the same low-frequency content a phenomenon is observed. The resulting sound field will be affected by their location in the room, the phase and amplitude response, and the absorption characteristics of the room.

One way to obtain the joint TSP of the two loudspeakers is by measuring both sources simultaneously with a large number of microphones, e.g., 36-45 mics. Another way is to measure them separately with the same large number of microphones, then adding at the corresponding microphone IRs in the time domain or in the frequency (complex) domain the two sources offline. These two approaches are cumbersome and prone to errors.

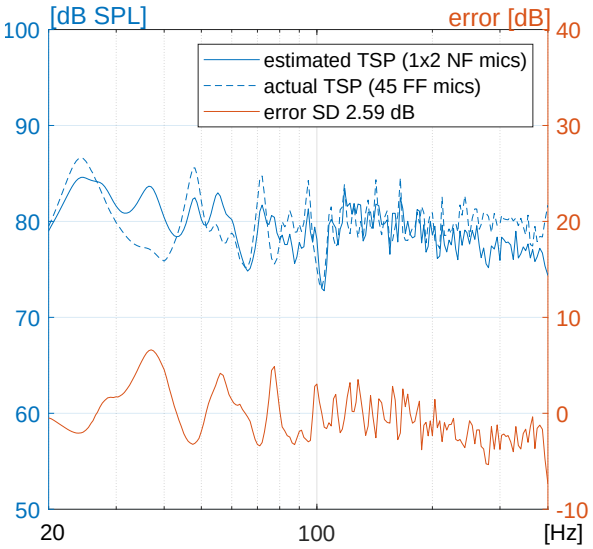


Fig. 7: Simulations: TSP of Setup 1 with two simultaneous loudspeakers. Continuous curve, estimation with two NF mics. (one at each loudspeaker) Dotted curve, actual joint TSP computed with 45 FF mics.

As explained in Section 2.5 one would think intuitively that the TSP of two simultaneous loudspeakers could be estimated with the addition of the two NF microphone transfer functions of the individual transfer functions $H_{SR1\to NF1}$ and $H_{SR2\to NF2}$, (one at each loudspeaker) measured separately in the room. Then adding the two transfer functions in the time domain or in the complex domain as

$$p(f) = H_{SR1\to NF1}(f) + H_{SR2\to NF2}(f), \qquad (5)$$

then use $p(f)$ in Equation 1. This example is illustrated in Figure 7 where the TSP was estimated as explained above using the same velocity $U(f)$ for both loudspeakers. As observed this intuitive method does not work, the estimation of the TSP of both simultaneous loudspeakers was wrong because it seems that the interaction of both sources in the room is not captured by the addition of both NF responses.

2.6.1 Simultaneous Measurements

If both loudspeakers are measured simultaneously as seen in Figure 8, the contribution of both sources in both NF mics would include the interaction of both

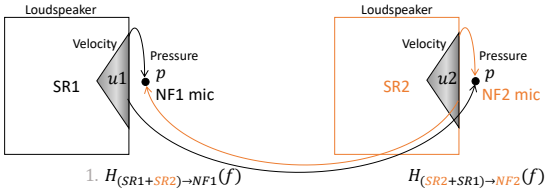


Fig. 8: Simultaneous measurement of S1 + S2 for TSP estimation.

loudspeakers in the room. Then the two NF transfer functions can be added as

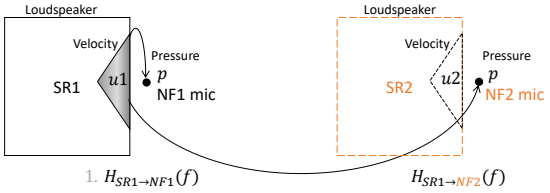
$$p(f) = H_{(SR1+SR2)\to NF1}(f) + H_{(SR2+SR1)\to NF2}(f). \qquad (6)$$

2.6.2 Sequential Measurements

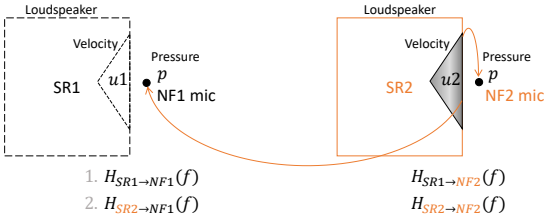
In case the measurement of each loudspeaker is done sequentially, then the transfer function of each loudspeaker to each NF microphone is needed to conform the correct addition as in

$$p(f) = 10^{\left(\frac{nS/2}{2}\right)} \cdot \left[H_{SR1\to NF1}(f) + H_{SR2\to NF1}(f)\right] + \left[H_{SR1\to NF2}(f) + H_{SR2\to NF2}(f)\right], \qquad (7)$$

where the term $10^{\left(\frac{nS/2}{20}\right)}$ is used to correctly scale the TSP and nS is the number of loudspeakers to be added.



(a) S1 measurement.



(b) S2 measurement.

Fig. 9: Sequential measurement of multiple loudspeakers for TSP estimation.

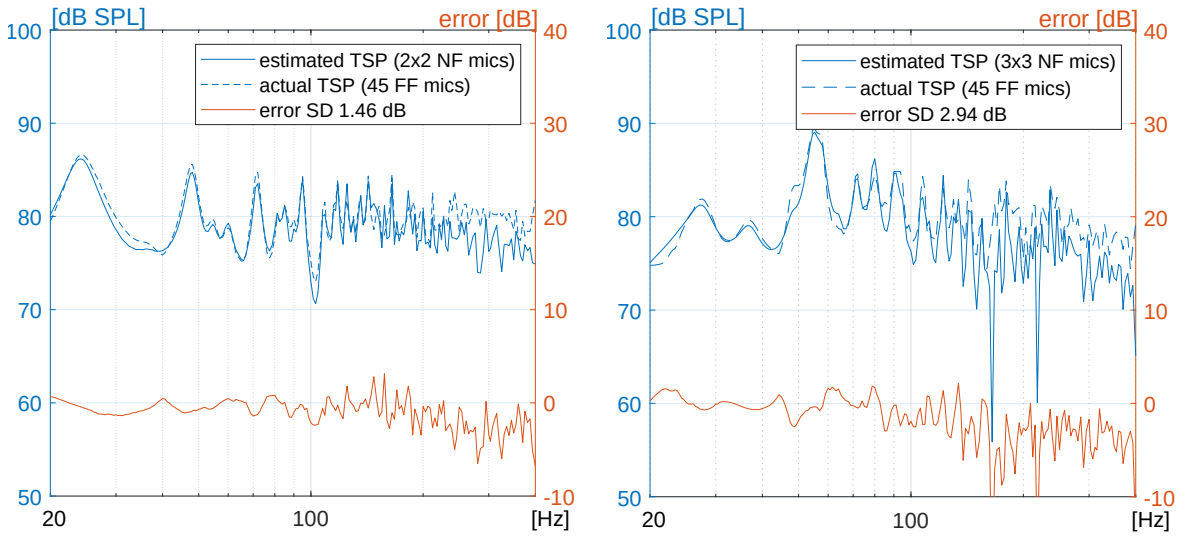


Fig. 10: Simulation results, TSP estimation. Left, Setup 1 with 2 simultaneous loudspeakers. Right, Setup 2 with three simultaneous loudspeakers. Continuous curves, estimation with NF mics. Dotted curves, actual TSP computed with 45 mics.

Then $p(f)$ can be used in Equation 1 using the same velocity $U(f)$ for both loudspeakers.

2.7 Equalization

Once the TSP was estimated an equalization filter was designed using the approach described in [17]. Sixteen parametric equalization filters (PEQ) in cascade were designed to match the TSP to a target response in the range of 40 to 200 Hz. The filter was designed in a manner that will follow the equalization target but will not boost the response in case of a notch in the TSP frequency response.

2.8 Real In-room Loudspeaker

Three commercially available loudspeakers "Music Frame" were set in the "Large Listening Room" of the same dimensions as the simulated room in Section 2.4. The listening room conforms with the ITU-R Rec. BS.1116 standard. The loudspeaker coordinate locations are shown in Table 2. The "Music Frame" works as a stereo loudspeaker with two woofers, two mid-range drivers and two tweeters. The audio input was routed to both woofers and only one mid-range and tweeter on each loudspeaker with their corresponding crossover and voicing filters included. The velocity of the loudspeaker was obtained by measuring the displacement transfer function from the electrical terminals to the woofers via a laser device. A miniature microphone (MEMS) was attached to the loudspeaker in the same position as the internal microphone of the loudspeaker to acquire the NF IR. The relative phase of velocity and pressure was de-trended, assuming that the angle associated between pressure and velocity is about 90 degrees. The trend was evaluated by fitting a 9th-order polynomial using the least squares method (see [12] for more details). For verification of the estimated TSP, 36 Beyerdynamic MM-1 omnidirectional condenser measurement microphones were randomly set in the room. The FF and NF microphones were connected to an RME A/D converter working on a PC laptop computer via AVB network. The audio digital output from the PC was sent to the loudspeakers via optical cable. The IR's were acquired by sequential measurements using the logarithmic sweep method described in [18].

3 Results

3.1 Simulations

In Figure 10 the simulation results of TSP estimation for Setup 1, with S1+S2, and Setup 2, with S1+S3+S4

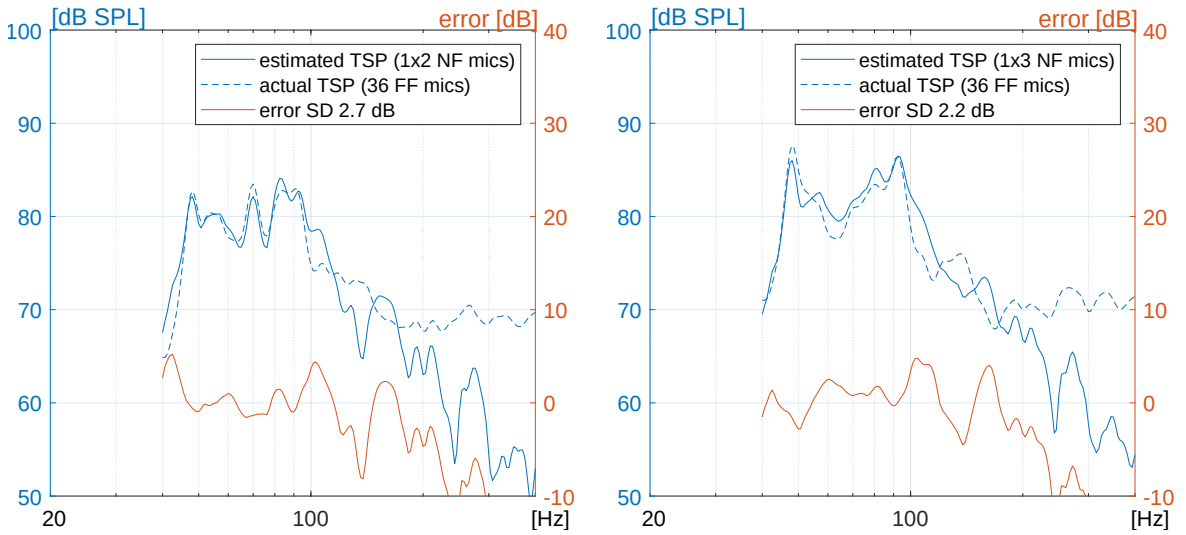


Fig. 11: Real In-room loudspeaker TSP estimation. Left, with S1+S4 loudspeakers. Right, with with S1+S3+S4 loudspeakers, both in Setup 2. Continuous curves, estimation with NF mics. Dotted curves, actual TSP computed with 36 FF mics.

loudspeakers are shown. The actual TSP computed with 45 FF microphones and the error curves are also shown in the same figure. The SD was computed from 20 to 400 Hz (also presented on each panel of Fig. 10).

3.2 In-room Measurements

In left panel of Figure 11 the TSP, result of two simultaneous loudspeakers in the real room, estimated with their NF microphones is shown. The actual TSP computed with 36 FF microphones and the error curves are also shown in the same figure. In right panel of Figure 11 the estimated TSP result of three simultaneous loudspeakers in the real room is shown. The SD was computed from 40 to 200 Hz (also shown in both panels). The designed equalization filters and estimated TSP+EQ for two and three loudspeakers are shown in the upper row of Figure 12. The error and target curves are shown in all panels of Figure 12. The SD was computed from 40 to 200 Hz (also shown in figure legends). The verification of the estimated EQs applied to the actual TSP measured with 36 FF microphones in the room are shown in lower row of Figure 12, for two, and three loudspeakers, respectively.

Table 2: In-room loudspeaker center coordinates

	Loudspeaker Pos.	Coordinates (x,y,z) m
Setup 2	S1	0.75, 6.30, 1.36
Setup 2	S3	7.22, 1.60, 1.68
Setup 2	S4	0.10, 2.00, 1.36

4 Discussion and Conclusion

Although the practical tests were performed in only one real room, and with a few loudspeaker positions, the result of this work is encouraging. More experiments should be performed in different room shapes and with loudspeaker placements away from the walls. The Music Frame loudspeaker is a complex system that is intended to be placed mostly on-wall. A simple case with regular single driver loudspeakers should be tested in order to verify the performance of the method, and possible improvements. Future research should also include the subjective evaluation of, e.g., the TSP equalization against an equalization result of measurements of a small listening area.

The concept of sound power radiated by a compact loudspeaker in a room was described in Section 2.1. The TSP can be estimated with a NF microphone measurement and the loudspeaker's velocity. A study to

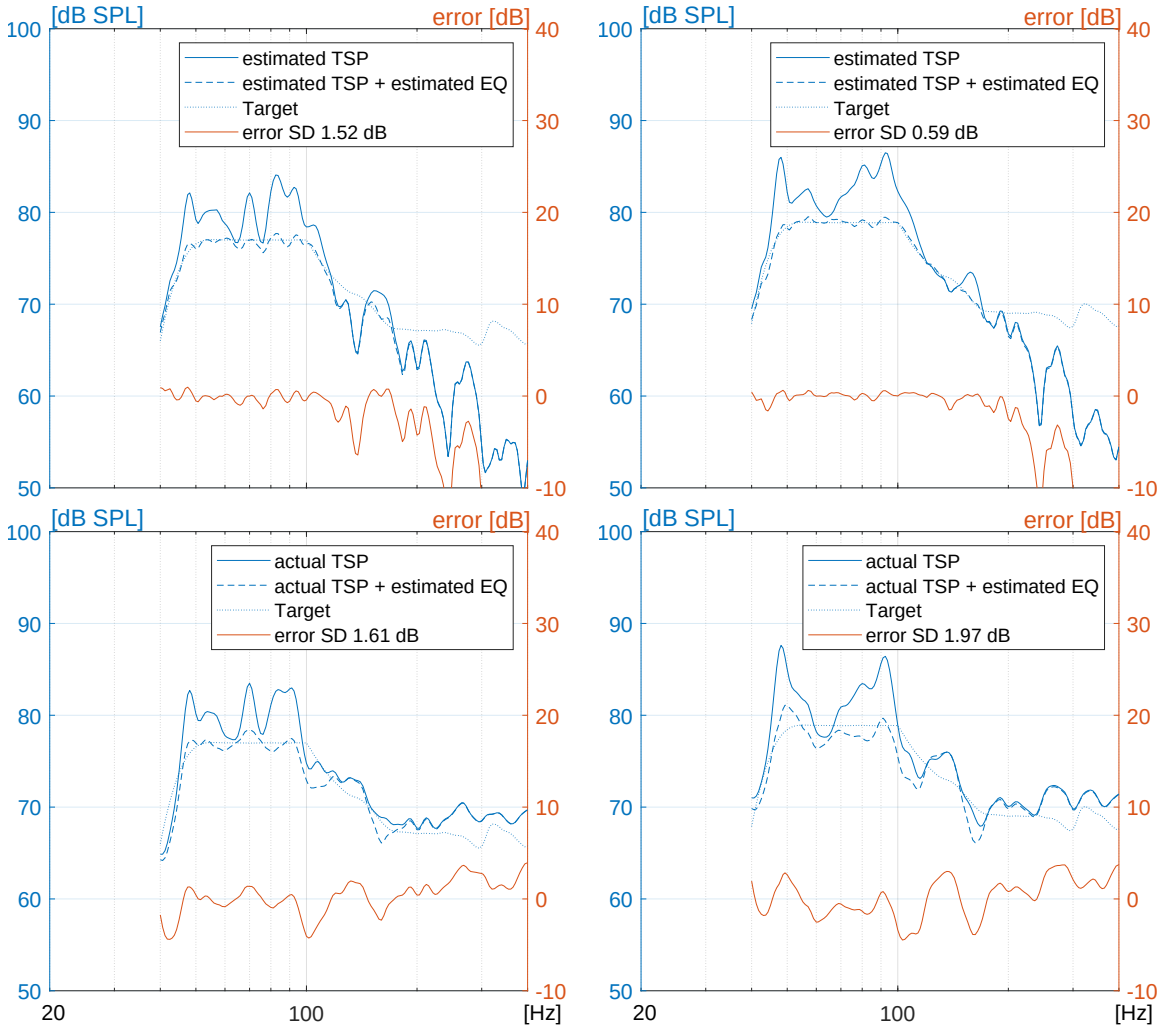


Fig. 12: Upper row, equalization of estimated TSP, Real In-room loudspeakers. Lower row, verification of estimated EQ. Left column, loudspeakers S1+S4. Right column, loudspeakers S1+S3+S4. Continuous curves, estimated/actual TSP. Dashed curves, estimated/actual TSP + estimated EQ. Dotted curves, EQ target.

find the number of microphones necessary to obtain a reliable TSP measure in a room was presented in Section 2.3, the study showed that to have an accuracy less than 2.3–2.5 dB 95% CI, more than 36 microphones randomly distributed in the room should be used.

The correct estimation of TSP radiated with simultaneous loudspeakers with a minimum number of NF microphones is explained in Section 2.6 by adding the measured complex frequency responses at the NF microphone positions simultaneously. When the measurement of both loudspeakers is sequentially done, then add the four transfer functions at both NF microphones from both loudspeakers, corresponding to the transfer function from loudspeaker active to its NF microphone and the same loudspeaker to the NF microphone at the other loudspeaker, and vice-versa.

When assuming identical loudspeakers, the same velocity can be utilized for the TSP estimation. This has

been verified with simulations and in one real room measurement.

Results of simulations shown a SD from 1.5–2.6 dB in the TSP estimation with a minimum number of NF microphones. In-room measurements with commercially available loudspeakers the TSP estimation presented a SD from 2.2–2.7 dB from 40-200 Hz, and 1.8-1.9 dB from 40-120 Hz. Finally two examples of room equalization filters with the estimated TSP were presented. A SD of 1.6–2.2 dB from the equalization target measured in the room with 36 FF microphones was obtained for two and three simultaneous loudspeakers respectively.

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