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In-Room Low-Frequency Sound Power Optimization using Near Field Response

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ABSTRACT

The total sound power (TSP) produced by a loudspeaker can be severely affected when placed in typical living rooms. General approaches consider an equalization filter designed toward a desired target. The computation of the TSP requires measurements on a number of microphones spaced in the room. In this work, an automatic method to estimate the TSP without the use of numerous measurements is proposed. The proposed method includes a static microphone configured to measure the near-field sound pressure of the driver, and a controller to determine its velocity to automatically adjust the sound power levels to an acoustic environment. Results in typical living rooms show an average standard deviation error of 2.6 dB on the estimation of the TSP.

1 Introduction

The response of a loudspeaker is severely altered when radiating sound in a room. 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). This is due to the interaction of sound waves with the boundaries of the room, building distinct zones with high sound-pressure levels, related to the room resonances, and zones with low sound-pressure levels related to zones 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 di-polar patterns to control the sound field response over a listening area.

Several studies have introduced the room correction concept which uses a microphone to measure the combined loudspeaker-room frequency or impulse response at one or more locations in the room, [5, 6, 7]. A subjective evaluation of some of these methods has been presented by Olive et al. in [8]. When implementing these approaches in a commercial product, these methods can be prone to human error. Thus, research into automatic methods that avoid the inconvenience of tedious measurements is recommended. In order to equalize the response of the loudspeaker in the whole room, obtaining the TSP radiated to the room is required. This necessitates a number of microphone locations randomly distributed in the room.

In [9] the use of two microphone positions close to the driver's diaphragm are employed to compute the velocity and the acoustic power output radiated by the loudspeaker into the room. This method has been implemented in a product using a mechanical moving device attached to the speaker with one microphone allowing the two position measurements near the driver of the loudspeaker with one microphone.

This paper proposes a method for implementing a loudspeaker that automatically optimizes its TSP depending on its location in the room. Our proposed approach only requires one microphone in front of the driver's diaphragm, and the loudspeaker model.

2 Room-Loudspeaker Sound Power

The 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 [10]. The sound power $P(f)$ radiated by a small source into the room in the frequency domain can be written as:

$$P(f) = \tfrac{1}{2} Real\left\{Z_{rad}(f)u^2(f)\right\}, \tag{1}$$

where $Z_{rad}(f)$ is the acoustic radiation impedance, $u(f)$ is the volume velocity complex amplitude of the sound source [11], [10]. The acoustic radiation impedance is defined by

$$Z_{rad}(f) = \tfrac{p(f)}{u(f)}, \tag{2}$$

where $p(f)$ is the sound pressure complex amplitude at the source. If the right side of Equation. 2 is replaced in Equation. 1, then Equation 1 becomes:

$$P(f) = \tfrac{1}{2} Real\left\{p(f)u^*(f)\right\}, \tag{3}$$

which is half of the real part of the product between the near-field sound pressure complex amplitude p in front of the diaphragm, and the complex conjugate of the velocity $u^*$of the driver. 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.

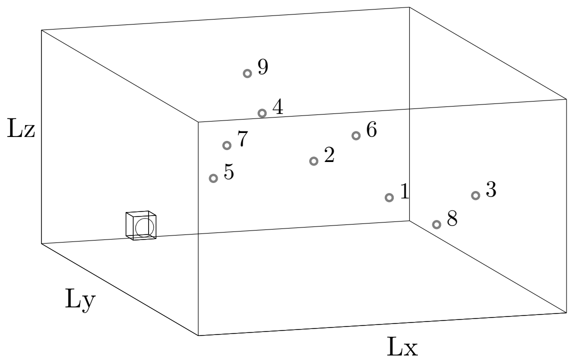

Fig. 1: Room simulated, loudspeaker and virtual microphone positions.

2.1 Total sound power (TSP)

To obtain the true TSP radiated by the loudspeaker into the room, the mean squared sound pressure level at a number of microphones randomly distributed in the room can be computed as:

$$P_{true}(f) = \sqrt{\frac{1}{n}\sum_{i=1}^{n} |p_i(f)|^2}, \tag{4}$$

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

Concerning the number of microphones needed to obtain a reliable measurement, Pedersen [12] has found that by using from 9 to 10 random microphone positions in the room, the RMS deviation from a reference estimate of the energy in the 3D sound field of 20 random microphone positions gets down to 1 dB. In this study it was decided to use 9 randomly chosen microphone locations for the true TSP computation.

2.2 FDTD Sound field room simulations

Initially room simulations were run using the finite differences in the time domain (FDTD) method. The FDTD procedure simulates sound wave propagation in the time domain. A graphical user interface (GUI) and script written in MATLAB has been utilized based on [13]. In this method the room 3D space is discretized in pressure and particle velocity points staggered in space, forming cells of size h. A sampling frequency $F_s = 8000$ Hz and a speed of sound $c = 343$ m/s was used in the simulation covering a total length $L = 8192$

samples or 1.02 seconds. The cell size h was set to 7.62 cm, giving a reliable result up to $\frac{c}{5\times h} = 900.3$ Hz.

A rectangular room with dimensions $Lx = 4.88$ m, $Ly = 6.40$ m and $Lz = 2.74$ m, similar to one of the reference listening rooms at Samsung Audio Lab. was simulated. The frequency-dependent absorption coefficients utilized in all walls are typical for drywalls on studs, (see Table 1). Nine randomly chosen virtual microphone positions were simulated to collect impulse responses. The frequency response (FR) was obtained by computing the Fast Fourier Transform (FFT) on the impulse responses, padded to $N = 16000$ samples to procure a $\Delta f = 0.5$ Hz.

Table 1: FDTD wall absorption coefficients.

Frequency (Hz)	20	125	250	500	1000
Abs. Coeff.	0.12	0.12	0.12	0.08	0.06

The simulated loudspeaker was constructed by defining a volume of $0.38 \times 0.38 \times 0.38\,\text{m}^3$ with dummy pressure points inside the box covered by rigid walls with the components of velocity set to zero. The driver is represented by a rectangular surface area of $0.23 \times 0.23\,\text{m}^2$ defined by the component velocity points moving in the y direction, (see Figure 2). These points are prescribed with the velocity in the time domain of a linear small-signal model of a 12-inch driver in a closed box.

The loudspeaker model employed in this study simulates the linear output of a driver for a given input wave by applying a linear small-signal model, seen in Figure 3. The representation utilizes the small signal

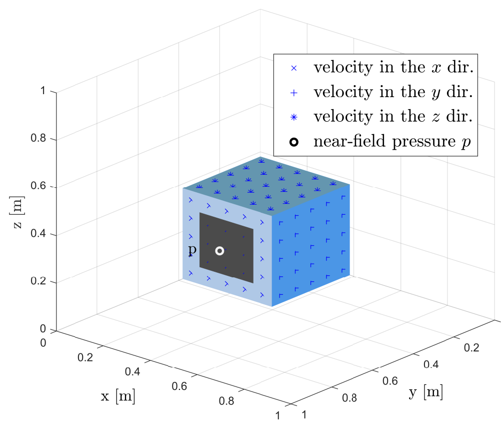

Fig. 2: Loudspeaker box for the FDTD model.

Table 2: FDTD Loudspeaker center coordinates.

Loudspeaker coordinates (m)	x	y	z
S1	1.26	0.19	0.19
S2	2.40	0.19	0.19

parameters obtained by a transfer function measurement from voltage to velocity using the laser Klippel analyzer system. The loudspeaker was measured in its enclosure.

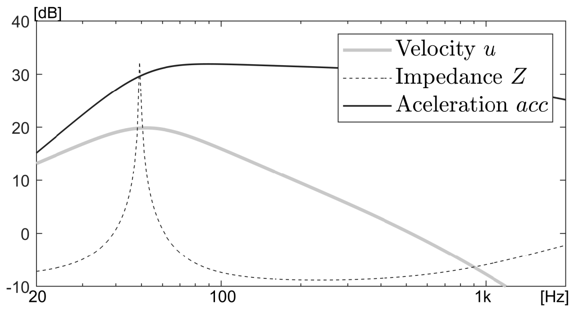

Fig. 3: Model of the 12-inch driver in a closed box.

Simulation Results

Two typical subwoofer positions in the room were simulated, the center coordinates can be found in Table 2. In the left graph of Figure 4 the gray curves show the frequency responses at the nine microphone positions of loudspeaker location S1, and the black curve shows the "true" TSP computed with Equation 4.

To estimate the TSP using Equation 3, the complex response in the frequency domain of the next available pressure point in front of the driver by $h/2$ =3.81 cm in the FDTD simulation was used as p. For u, the complex velocity response of the 12-inch driver in a closed box model, (shown in Figure 3) was used. As we can observe in the right graph of Figure 4 the relationship between the near-field pressure p and the velocity of the driver sustains about 90 degrees along 20 to 200 Hz. One should be aware that the model of the loudspeaker computes the velocity of the driver at the driver itself, and the near-field pressure p is recorded at $h/2 = 3.81$ cm from the driver, therefore a small delay exists in the near-field pressure p response.

The estimated TSP of loudspeaker position S1 using Equation 3 is shown in left graph of Figure 5. It is clear

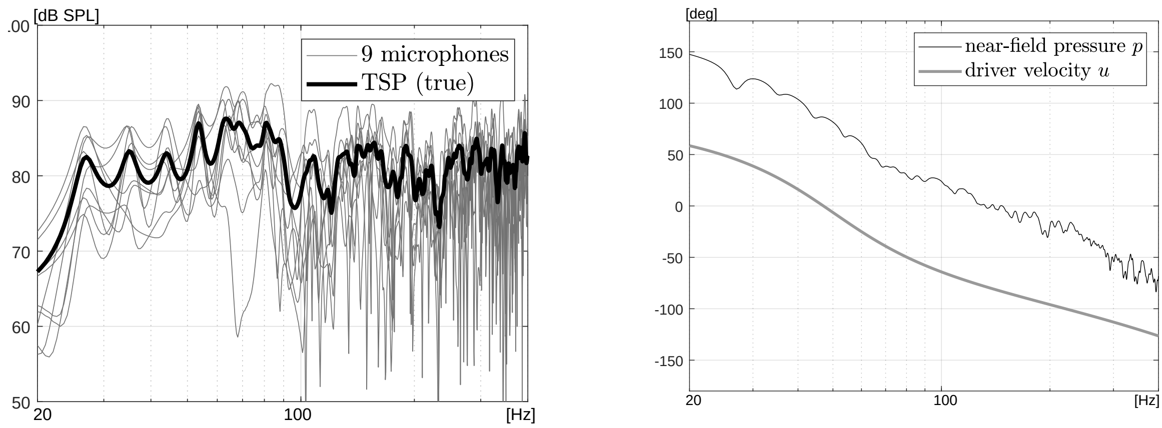

Fig. 4: Simulation, loudspeaker position S1. Left graph: frequency response. Right graph: phase.

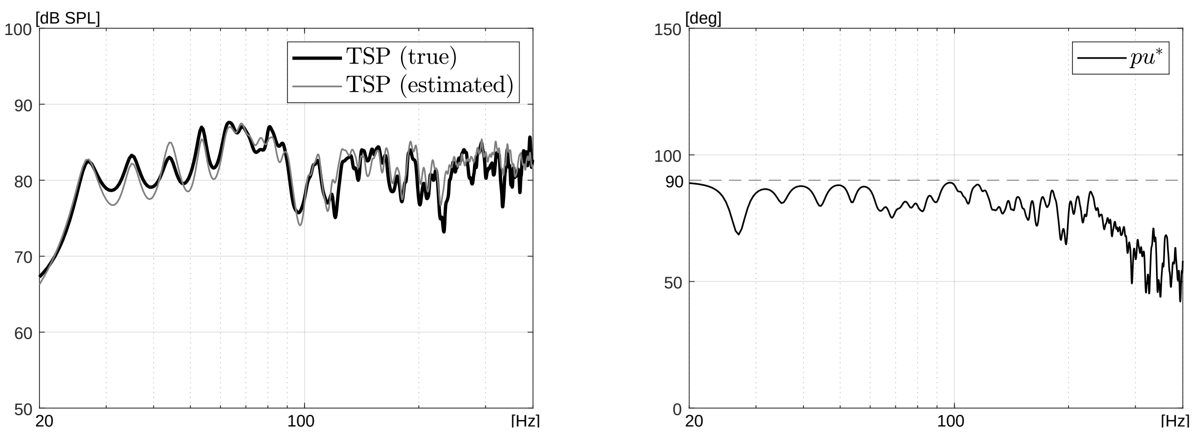

Fig. 5: Simulation, estimated TSP loudspeaker position S1. Left graph: frequency response. Right graph: phase.

that the estimation of the TSP is very close to the "true" TSP. In the right graph of the same figure, the phase of product pu^* gets very close to 90 degrees around some frequencies, at these particular regions, the TSP response presents a reduction in the TSP.

In the left graph of Figure 6 the result of the TSP estimation of loudspeaker position S2 simulation is shown. As we can observe in the light gray curve, the estimation follows the "true" TSP everywhere but from 30 to 50 Hz, and around 117 Hz where a loss in power is observed. Looking at the right plot of Figure 6, the phase of the product pu^* touches the 90 degrees around the same range from 30 to 50 Hz, and also at 117 Hz. In general, the TSP estimation in both positions, S1 and S2 is very close to the "true" TSP, but at some frequencies presented small discrepancy. It is also observed that in both cases when the phase of the product pu^* approaches the 90 degree line, the TSP is weaker than the "true" TSP, which is not really true. The simulations have shown that when utilizing only one pressure point to predict the TSP, the accuracy of the phase differences between pressure and velocity are crucial.

3 TSP Estimation

As observed in Section 2.2, simulations shown that the TSP can be estimated with the use of only one pressure point in front of the driver of the loudspeaker and with the velocity of the system. This confirms that a compact sound source is a velocity source, therefore the velocity of the system does not depend on the position in the room. To estimate the TSP, we propose the use of one microphone attached to the loudspeaker to acquire the

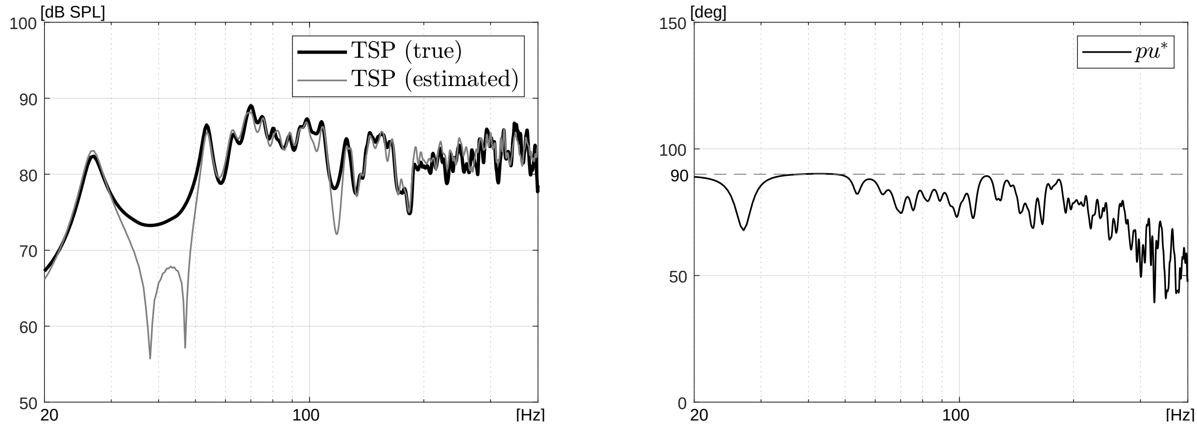

Fig. 6: Simulation, estimated TSP loudspeaker position S2. Left graph: frequency response. Right graph: phase.

near-field pressure. Simultaneously, the velocity of the system can be computed by acquiring the current of the loudspeaker, or computed offline by modeling the motion of the loudspeaker (including its enclosure).

3.1 Prototype

A prototype has been built using a 12-inch subwoofer in a 0.34×0.34×0.37 m closed enclosure. A 1/2-inch GRAS condenser microphone was attached with a wooden fixture in front of the centre of the diaphragm, at approx. 5 cm from the dust cap.

Fig. 7: Subwoofer prototype.

3.2 Multitone Test Signals

In order to perform measurements in the field and obtain the complex frequency response at the near-field microphone and at the randomly distributed microphones in the room, the multitone method was chosen. A nonlinear device under test (DUT) excited by a stimulus can be assumed to have an additive behavior where the non-linear components of the response Y_{NL} are added to the linear components Y_L [14]. The sum of Y_{NL} and Y_L, combined with measurement noise nz will produce the measured response Y. In order to extract the best linear approximation (BLA) response Y_L, we need to feed a repetitive signal u that is a sequence of $M \times (P+1)$ multitone signals $y_m^p(t)$ $(m = 1, 2, \cdots, M;$ $p = 0, 2, \cdots, P)$. The signal is comprised of a sum of harmonic sines with randomized phases and whole periods. During the measurement, the signal u is sent to the DUT and appropriate outputs (current i, excursion x, pressure p, etc.) are recorded. The post-processing to extract the BLA of the DUT is done by subsequent averaging of the FRs over P and M. By post-processing

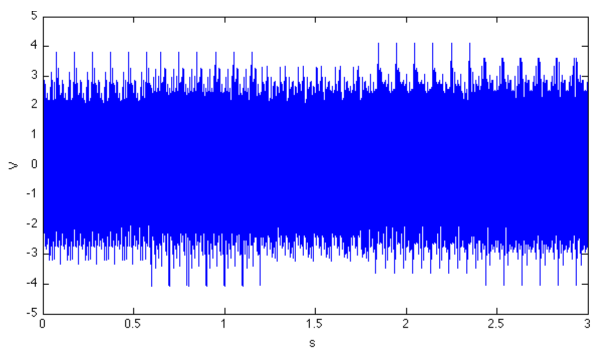

Fig. 8: Example of sequential multitone with $M = 5$ realizations of $P + 1 = 6$ periods each.

the ratio of $G = Y_L/U$ in the frequency domain, the algorithm will effectively linearize the FR of the DUT. The algorithm only performs averaging of the last P periods, instead of all $P+1$ periods because the first period $p = 0$ contains some transient information from the $(m-1)$'th realization, therefore these segments are discarded to have P periods with steady-state conditions of the loudspeaker. An example of a sequential multitone signal is shown in Figure 8. We note that a major advantage of using multitones vs sine sweeps is that the crest factor and spectral distribution of the stimulus signal can be approximated to typical audio content (music, speech, movie track, etc.) by tuning the amplitudes and phases of the individual tones.

3.3 Velocity Phase Correction

Typically at low frequencies and small distance to the source, the near-field pressure is close to 90 degrees ahead of the velocity, meaning that $kr \ll 1$ where k is the wave number and r is the distance to the source. The divergence from perfect quadrature between pressure and particle velocity reflects the fact that some power is radiated into the room.

In Figure 9, (light gray curve) the phase of product pu^* of a measurement in a real living room of the subwoofer prototype is presented. As it can be observed, the phase of product pu^* (original) is not constant across frequencies, and presents a downward slope. One would think that this is due to the propagation delay between the microphone and the driver's diaphragm. Another suggestion is that in Equation 3, the pressure and velocity should be measured at the acoustic centre of the sound source, which for a compact source, and at low frequencies is in front of the diaphragm (approx. 10 cm). The situation presented in [9], where the velocity is computed using the difference between two microphone positions in front of the loudspeaker reinforces the idea of acquiring both pressure and velocity in front of the drivers diaphragm, where the acoustic centre is.

It seems that the velocity and phase would also include a frequency dependent delay, due to the physical characteristics of the source, and the point where the pressure is acquired. If this is not properly handled, it would lead to an incorrect sound power estimation. In our investigation we are using the velocity of the model of the loudspeaker, and just one acquisition point for the pressure in front of the diaphragm (approx. at 5 cm). To accurately estimate the TSP of the loudspeaker in the room, it was decided to de-trended the relative phase of velocity and pressure, provided that the angle associated between pressure and velocity is about 90 degrees. The trend was evaluated by fitting a 6th order polynomial using the least squares method.

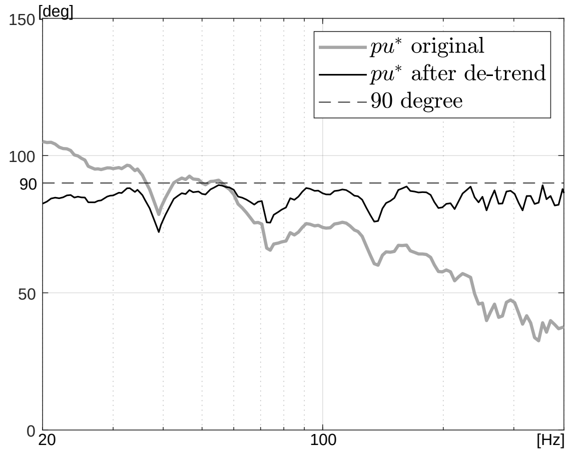

Fig. 9: Phase of product $pu*$, before and after de-trend.

In Figure 9, (black curve) the result after de-trending the phase of the velocity is shown. This procedure allows the use of the microphone not only in front of the diaphragm, but also attached to the loudspeaker box, closer to the diaphragm. More details on this procedure can be found in [15]. After this preparation step, the TSP is computed using Equation 3 and finally, to obtain the quantity in decibels, the following equation is used:

$$P_{dB}(f) = 10 \cdot \log_{10}\left[\frac{P(f)}{P_0}\right], \tag{5}$$

where P_0 is the reference sound power (10^{-12} W) in acoustic Watts.

In Figure 10, the TSP of the loudspeaker in a real living room is presented. For verification purposes, the "true TSP" has been computed from measurements of nine microphones randomly spaced in the room. The estimated TSP has been normalized in level to be as close as possible as the "true TSP" in the range from 30 to 100 Hz. As it can be observed the estimated sound power and the sound power measured with nine microphones are very similar, with a slight deviation around 25 Hz and 55 Hz and from 100 to 200 Hz.

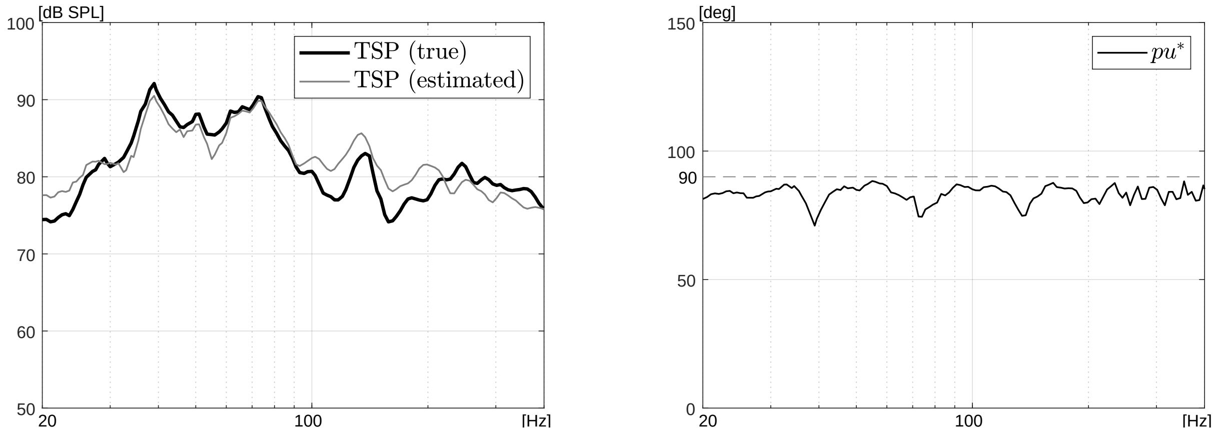

Fig. 10: Real Room 4, estimated TSP, loudspeaker position S2. Left graph: frequency response. Right graph: phase.

4 Equalization Strategy

In this section, the TSP equalization process including the room gain estimation and equalization target will be detailed.

4.1 Room gain

To obtain a natural, sound room equalization, the room gain can be approximated with the help of the velocity, the estimated TSP and the near-field sound pressure. Similar procedures have been presented in [5]. The near-field sound pressure frequency response (NFP) in dB SPL is normalized with the estimated TSP around 200 Hz. Then the NFP is subtracted from the estimated TSP. Finally a shelving function is fitted to obtain the room gain. In Figure 11, the target curve shows the room gain particular to Room 4, loudspeaker position S2.

4.2 Equalization

The room gain is added to the acceleration from the loudspeaker model to construct the equalization target. The method for designing the filter to optimize the TSP response was based on the technique proposed by Ramos and López [16] using a minimum-phase filter constructed to correct the response from 20 to 300 Hz. The equalization filter was constructed using infinite impulse response (IIR) filters with 14 biquads or parametric equalization (PEQ) filters. The result of the equalization process is presented in Figure 11.

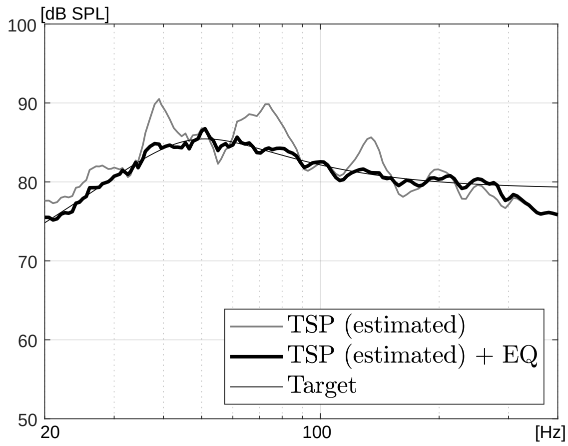

Fig. 11: Equalization result. Room 4, S2

5 Experimental Test

The subwoofer prototype has been measured in 11 living rooms including two listening rooms at the Samsung US Audio Lab, which included a total of 60 loudspeaker positions. The loudspeaker was located in about four to five typical positions in the room, including at least one in the corner of the room. The near-field pressure was measured on each position using multitone as a test signal. In order to verify the results, nine microphones distributed randomly in the room were set to measure and calculate the "true" TSP for each subwoofer position. For each loudspeaker position, the TSP was estimated utilizing the proposed method.

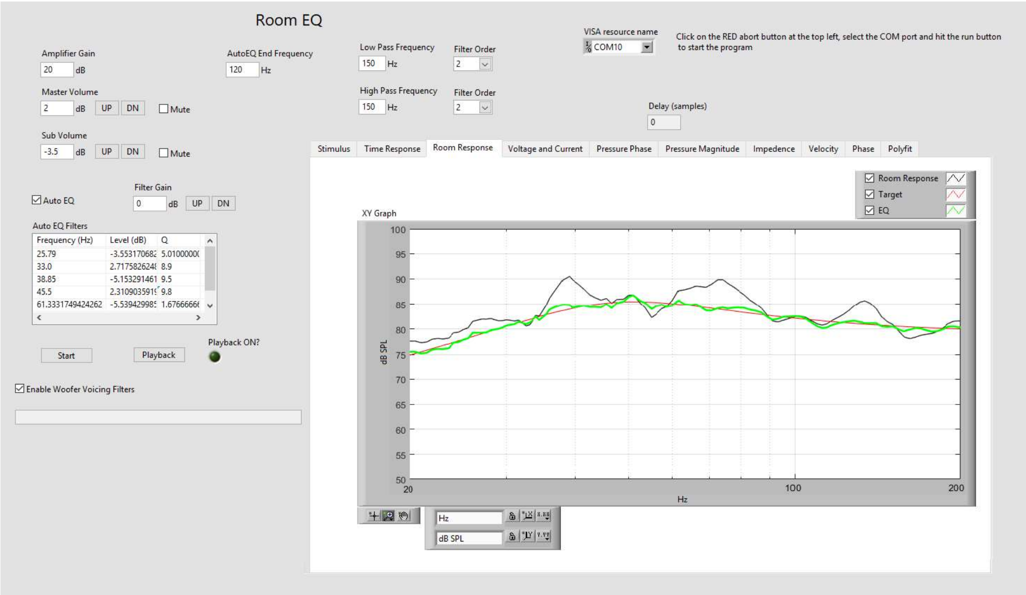

Fig. 12: Front end of the GUI hardware measurement tool.

5.1 DSP Hardware and Software Tool

For the real-time implementation, playing and recording audio signals, the Analog Devices ADSP-21469 hardware development kit was utilized. The user initiates the test employing a GUI built in LabVIEW which runs on a PC (seen in Figure 12). After the test is run, the algorithm generates the parametric equalization (PEQ) filters to optimize the TSP. The hardware sends back the TSP, target and the generated filter frequency responses to the GUI, for easy viewing of the graphs. The user can control via the GUI whether to pass the output signal through PEQs or bypass it for a convenient A/B testing.

6 Results

The errors between the estimated and the "true" TSP with the nine microphones distributed in the room were calculated. The overall TSP estimation error curves from the total of 60 cases over 11 living rooms is shown in Figure 13. The mean error and confidence interval are shown in the same figure. The estimation of the TSP presented an average standard deviation error of 2.61 dB from 22 to 100 Hz, and 2.67 dB from 22 to 200 Hz. The estimation error seem to worsen as the frequency increases from 100 to 400 Hz. In Figure 14, a sample of four results loudspeaker positions from the 60 cases over the 11 living rooms are shown.

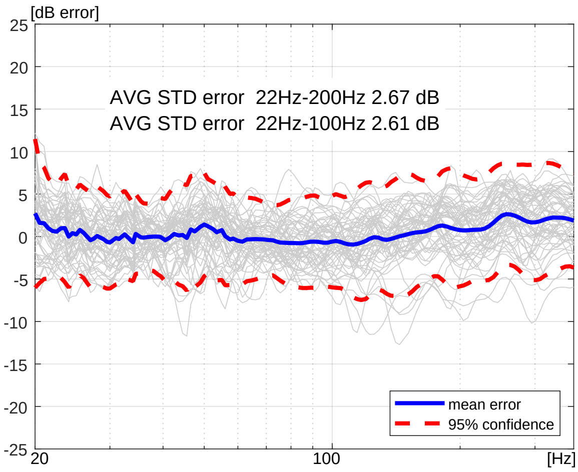

Fig. 13: Overall TSP estimation error from the total of 60 cases over 11 living rooms.

7 Discussion

As shown in Section 6, the estimation of the TSP in the room can be achieved with only one microphone in the near-field of the loudspeaker driver. The phase differences between pressure and velocity are crucial to predict accurately the TSP. It is especially important when the phase difference gets close to 90 degrees, because a small error can result in a big discrepancy

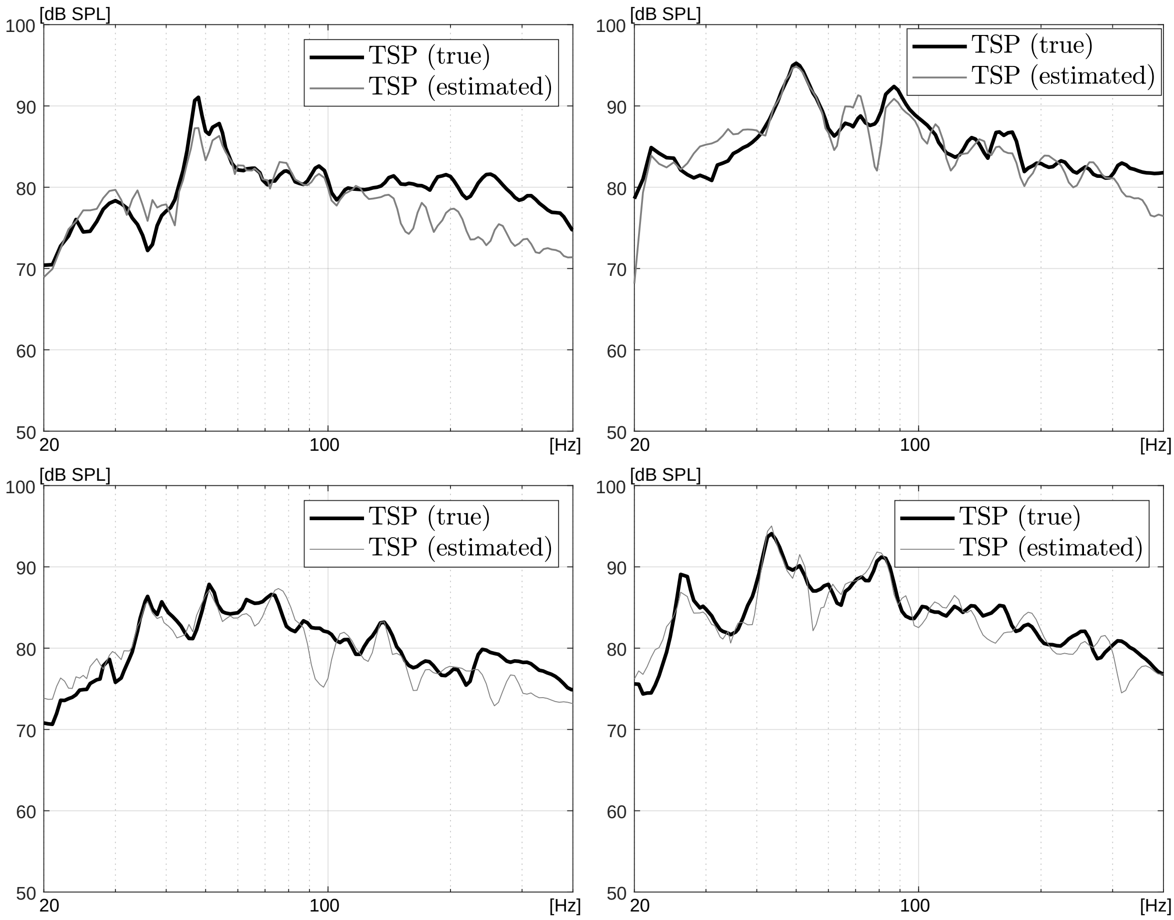

Fig. 14: Results TSP estimation. Top row from left to right, Room 2 - S4. Room 3 - S2. Bottom row left to right, Room 4 - S6, Room 5 - S1.

in the TSP estimation. Concerning the acquisition of the velocity of the system, the transfer function from voltage to velocity can be used (loudspeaker model). Another alternative is by measuring the current i, and computing the velocity from the electrical equation: $u = (V - R_e i)/Bl$, in the frequency domain, where R_e is the electrical resistance, V is the voltage input and Bl is the force factor.

The presented technology does not need a mechanical or moving device to estimate the TSP. It only requires one measurement; therefore, no user interaction is needed. More over the initial measurement could be performed using the music program as excitation signal, to obtain the near-field pressure complex response. Fully automatic room equalization can be performed to improve the quality of the reproduced sound.

Preliminary studies have shown that our proposed method is robust to other near-field microphone positions, attached to the loudspeaker box, and not restricted to be in front of the driver. Recent investigations performed by the authors have shown that the proposed technology can also be utilized in vented boxes.

8 Summary

The room-loudspeaker sound power radiation concept has been analyzed. Simulations have been carried out in order to find the possibility to estimate the TSP in the room by using only one microphone attached in the near-field of the loudspeaker driver. The method to estimate the TSP has been described and evaluated by measurements in 11 typical US living rooms. The method uses the near-field sound pressure and the velocity from the linear model of the loudspeaker in its enclosure. Results have shown an average standard deviation error in the estimation of the TSP of 2.6 dB from 22 to 200 Hz.

The proposed equalization method automatically optimizes the TSP depending on its location in the room, maintaining the room gain but reducing the prominent room resonances. If the proposed method is implemented in a product, the user would not have to perform tedious measurements in the room. Informal listening tests reported satisfactory results improving the reproduced sound quality, especially in Clarity. The elimination of the boomy sound typical of loudspeakers in living rooms was evident.

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References

[1] Santillán, A. O., "Spatially extended sound equalization in rectangular rooms," *The Journal of the Acoustical Society of America*, 110(4), pp. 1989–1997, 2001, doi:10.1121/1.1401740.

[2] Celestinos, A. and Nielsen, S. B., "Controlled Acoustic Bass System (CABS) A Method to Achieve Uniform Sound Field Distribution at Low Frequencies in Rectangular Rooms," *J. Audio Eng. Soc*, 56(11), pp. 915–931, 2008.

[3] Welti, T. and Devantier, A., "Low-Frequency Optimization Using Multiple Subwoofers," *J. Audio Eng. Soc*, 54(5), pp. 347–364, 2006.

[4] Hill, A. J. and Hawksford, M. O., "Chameleon Subwoofer Arrays—Generalized Theory of Vectored Sources in a Closed Acoustic Space," in *Audio Engineering Society Convention 128*, 2010.

[5] Abildgaard Pedersen, J. and El-Azm, F., "Natural Timbre in Room Correction Systems (Part II)," in *Audio Engineering Society Conference: 32nd International Conference: DSP For Loudspeakers*, 2007.

[6] Abildgaard Pedersen, J. and Thomsen, K., "Fully Automatic Loudspeaker-Room Adaptation - The RoomPerfect System," in *Audio Engineering Society Conference: 32nd International Conference: DSP For Loudspeakers*, 2007.

[7] Bharitkar, S. and Kyriakakis, C., "Objective Function for Automatic Multi-Position Equalization and Bass Management Filter Selection," in *Audio Engineering Society Convention 119*, 2005.

[8] Olive, S., Jackson, J., Devantier, A., and Hunt, D., "The Subjective and Objective Evaluation of Room Correction Products," in *Audio Engineering Society Convention 127*, 2009.

[9] Pedersen, J. A., "Adjusting a Loudspeaker to Its Acoustic Environment - The ABC System," in *Audio Engineering Society Convention 115*, 2003.

[10] Kleiner, M. and Tichy, J., *Acoustics of small rooms*, CRC Press, Taylor & Francis Group, 2014, ISBN 9780415779302.

[11] Salava, T., "Acoustic Load and Transfer Functions in Rooms at Low Frequencies," *J. Audio Eng. Soc*, 36(10), pp. 763–775, 1988.

[12] Pedersen, J. A., "Sampling the Energy in a 3-D Sound Field," in *Audio Engineering Society Convention 123*, 2007.

[13] Celestinos, A. and Nielsen, S. B., "Low-Frequency Loudspeaker–Room Simulation Using Finite Differences in the Time Domain—Part 1: Analysis," *J. Audio Eng. Soc*, 56(10), pp. 772–786, 2008.

[14] Pintelon, R. and Schoukens, J., *System identification: a frequency domain approach*, John Wiley & Sons, 2012, ISBN 9780470640371.

[15] Celestinos, A., "Auto-equalization, in-room low-frequency sound power optimization," U.S. Patent US20180262175A1, Sep. 13, 2018.

[16] Ramos, G. and López, J. J., "Filter Design Method for Loudspeaker Equalization Based on IIR Parametric Filters," *J. Audio Eng. Soc*, 54(12), pp. 1162–1178, 2006.