A Passive Method for the Determination of Acoustical Parameters in Occupied Rooms

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Introduction

The acoustical influence of the audience in concert halls and opera houses is still difficult to predict. Measurements of room acoustical parameters in occupied spaces are important for the design of new halls and renovations of existing halls. Conventional measurement methods of the occupied case are usually not very much appreciated by the audience because of the relatively high sound levels of the measurement signals. This rises the need for passive measurement methods.

A simple passive measurement method to determine the Reverberation Time is to analyze the decaying soundfield after a stop-chord during a concert [1]. This method only delivers the Reverberation Time and strongly depends on the played material. In [2] and [3] passive measurement methods using natural sound as test signal and based on cross-correlation are presented. This article carries on these ideas with a more elaborated algorithm which was tested in a virtual environment and in the well-known concert hall of the Stadtcasino Basel.

Method

In a definded performance setting the music signal x of a natural source on stage is used as test signal. A microphone is placed in the near-field of the source and provides the source signal x'. In the audience area—in the far-field of the source—a second microphone delivers the signal y. A block diagram of this setting is shown in Figure 1, where signals n_0 and n_1 are additive noise sources. h and g represent room impulse responses (RIR). Ideally, the measured source signal x' is equal to the source signal x. Then g corresponds to a Dirac delta function. Unfortunately, this is never true.

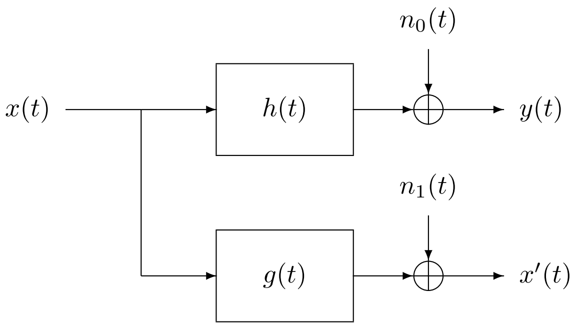

Figure 1: Block diagram of the setting

The approach to measure room acoustical parameters such as Early Decay Time, Reverberation Time and Clarity is based on estimating the RIR h and extracting the room acoustical parameters from the estimated impulse response. But the diffuse field component and early reflections in the RIR g badly influence the estimate of h and also the extracted parameters. Therefore several corrective processes had to be introduced to the algorithm.

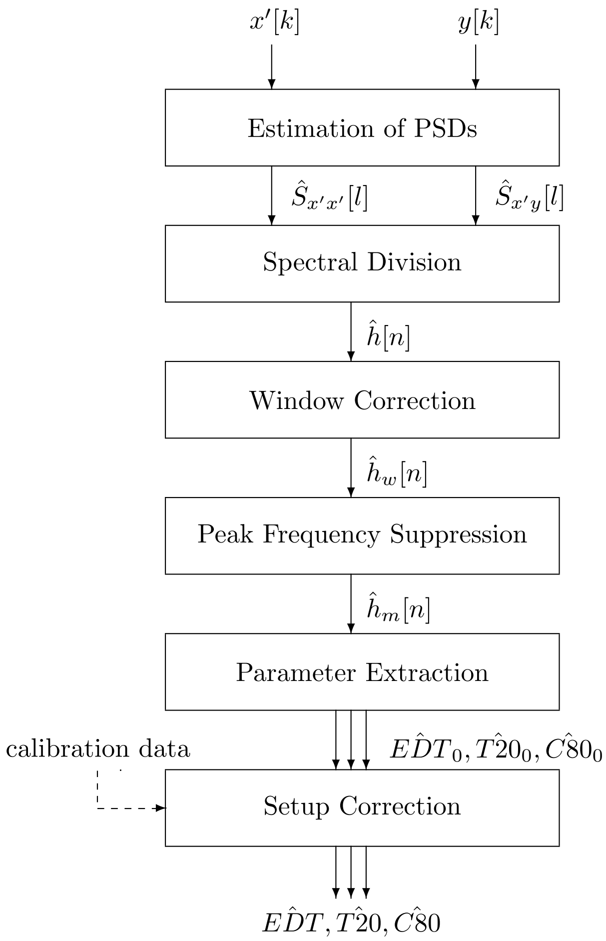

Figure 2: Algorithm for passive determination of room acoustical parameters with a close miked natural source

Algorithm

The developed algorithm is based on cross-correlation and uses additional information about the setup which can be found by a calibration measurement of the unoccupied space. Figure 2 shows a block diagram of the whole algorithm to compute the acoustical parameters from the two microphone signals x' and y.

From the 22kHz sampled microphone signals x' and y the cross power spectral density $\hat{S}_{x'y}$ and the power spectral density $\hat{S}_{x'x'}$ are estimated by Welch's method [4]. A 2^{18}-point Hanning window w and a 50% window overlap are used. Basically all transformations between time and frequency domain are performed by FFT or IFFT respectively.

Then the **Spectral Division** is applied. To prevent division by zero, a division saturation is introduced as shown in (1), where a is a small constant.

$$\hat{H}[l] = \frac{\hat{S}_{x'y}[l]\,\hat{S}_{x'x'}[l]}{\hat{S}^2_{x'x'}[l] + a} \tag{1}$$

Back in the time domain, the influence of the time window function w is corrected. The **Window Correction** is simply a multiplication by an envelope function e, which is given by (2) where * denotes the linear discrete convolution.

$$e[n] = \frac{1}{\frac{w[n]*w[n]}{\|w\|^2}} \tag{2}$$

Interfering sound can lead to very narrow-banded notches in the spectrum of the RIR g. The implicit inversion of g in the spectral division converts these notches into narrow-banded peaks in the spectrum of $\hat{h}_w$. These peaks are clearly audible and sound like discrete room resonances. For an appropriate estimation of the room acoustical parameters these discrete peak frequencies are detected in the spectrum of $\hat{h}_w$ as described in [5] and filtered out with narrow-banded notch filters in the **Peak Frequency Suppression**.

In the **Parameter Extraction** section the modified impulse response $\hat{h}_m$ is truncated and octave band filtered. The noise-corrected Clarity $\hat{C80}_0$ is calculated and linear regression lines of noise-corrected Schroeder curves provide the Early Decay Time $\hat{EDT}_0$ and the Reverberation Time $\hat{T20}_0$.

The diffuse sound part in g leads to temporal smearing of the estimated impulse response $\hat{h}_m$. This temporal smearing involves a systematic overestimation of the decay parameters and a systematic underestimation of the clarity parameter. These systematic errors were investigated by simulations in Matlab. The simulations showed that these errors can be predicted from the setup parameters

- distance source to the source microphone r_s,
- directivity of the source Q_{source},
- directivity of the microphone Q_{mic} and
- volume of the room V.

For instance, relationship (3) could be found for the absolute error of $\hat{T20}_0$.

$$\Delta\hat{T20}_0 \propto \frac{E_{g,\mathrm{diff}}}{E_{g,\mathrm{dir}}} \propto \frac{r_s^2}{Q_{\mathrm{source}} Q_{\mathrm{mic}} r_H^2} \tag{3}$$

$E_{g,\mathrm{dir}}$ and $E_{g,\mathrm{diff}}$ denote the energy of the direct sound part of the RIR g or the diffuse sound part respectively.

Relation (3) also contains the critical distance r_H and the directivity factors of the source Q_{source} and the source microphone Q_{mic}.

From (3) follows for the relative error of $\hat{T20}_0$

$$\delta\hat{T20}_0 = \frac{\Delta\hat{T20}_0}{T} \propto \frac{r_s^2}{Q_{\mathrm{source}} Q_{\mathrm{mic}} V} \,. \tag{4}$$

Relation (4) shows the possibility to compensate this temporal smearing effect by introducing a **Setup Correction** on the calculated room acoustical parameters. If some of the setup parameters are unknown, which is usually true, the setup correction coefficients can be found by a calibration measurement of the unoccupied space.

Maximal Microphone Distance

The potential of the proposed method is highly depending on the distance of the source microphone to the source r_s. The distance $r_{s,\max}$ was defined to be the microphone distance which still allows the determination of EDT and $T20$ with relative errors of $\pm 10\%$ and $C80$ with an absolute error of ± 1dB.

The algorithm was tested within a virtual environment consisting of synthetical RIRs which were convolved with dry music signals. For $r_{s,\max}$ the following empiric estimating formula could be found:

$$r_{s,\max} \cong 0.3\sqrt{Q_{\mathrm{mic}} Q_{\mathrm{source}}}\, r_H \qquad [\mathrm{m}] \tag{5}$$

Measurement Results

Measurements with the new method were carried out in the unoccuppied concert hall of the Stadtcasino Basel. For verification purpose also conventional measurements with an omnidirectinal loudspeaker were performed. All measurements were made with omnidirectional microphones and had a duration of one minute.

Figure 3 shows the room acoustical parameters $C80$, EDT and $T20$ measured by the new method with a male singer as acoustical source. Four measurement results with different distances r_s of the source microphone to the singer's mouth are shown. In the octave bands 250Hz to 4kHz the measured parameters show high consistency for different distances r_s. The differences of the $C80$ between the conventional reference measurement and the new method originate from the directivity of the singer.

Measurements with a grand piano as acoustical source were as well performed and showed promising results. But to make conclusive statements more experiments with a grand piano are required. The main problem of the grand piano as acoustical source is to get a good representative source signal x'. The geometrical size of the source involves a minimal microphone distance, which implies the use of unidirectional microphones or even multiple microphones for a good estimate of the source signal.

Conclusion and Outlook

A passive measurement method for the determination of room acoustical parameters in occupied rooms was presented. The new measurement method had been tested in a concert hall with a solo singer as acoustical source. The parameters EDT, $T20$ and $C80$ in the octave bands 250Hz to 4kHz could be determined with good accuracy.

Future work will consider more experiments with other natural source types (e.g. grand piano). Furthermore, practical measurements during concerts in occupied concert halls and opera houses should be carried out.

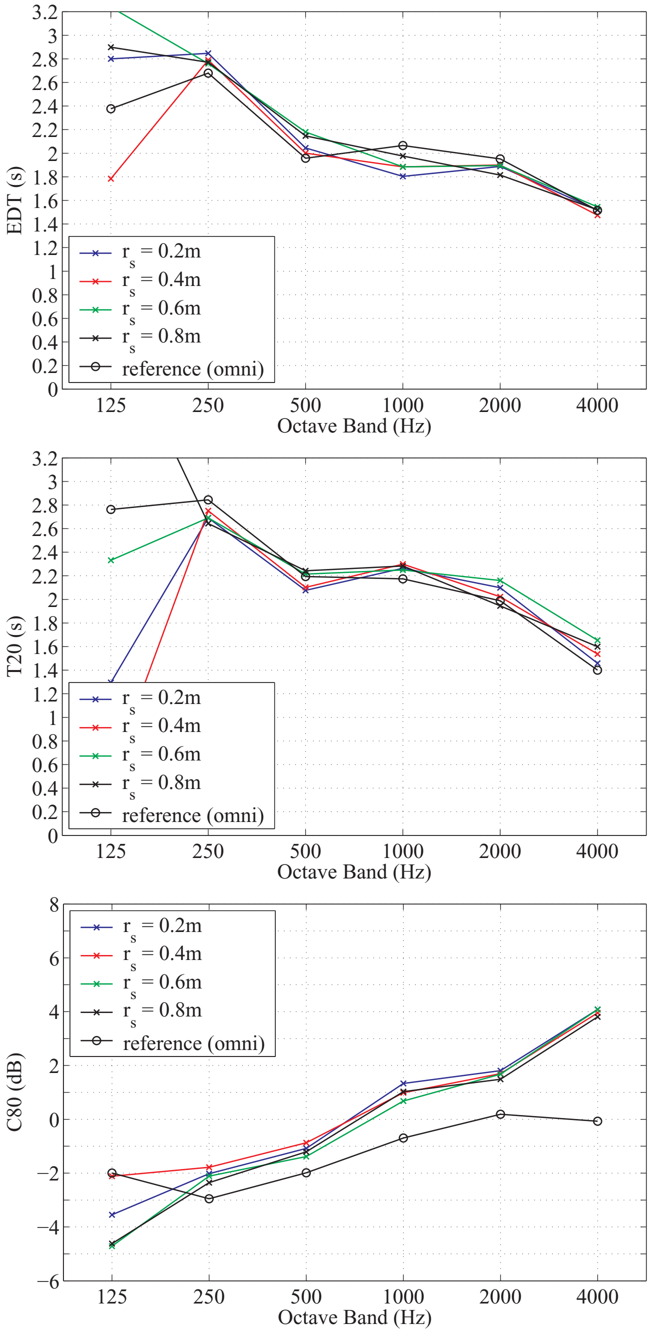

Figure 3: Measured room acoustical parameters with a solo singer as natural source and different source microphone distances r_s. The parameters of a conventional measurement with an omnidirectional loudspeaker are shown as reference.

References

[1] Concert Halls and Opera Houses, L. Beranek, Springler, New York, 2004

[2] Messung akustischer Parameter mit Musiksignalen in besetzten Konzertsälen, M.R. Schroeder et al., DAGA '76, 1976

[3] Verwendung von Stimme, Gesang und musikinstrumenten zur Bestimmung raumakustischer Parameter, R. Machner, DAGA '06, 2006

[4] The use of fast Fourier transforms for the estimation of power spectra: A method based on time averaging over short modified periodograms, P.D. Welch, IEEE Transactions on Audio and Electroacoustics, 1967

[5] Verwendung von Musiksignalen zur Bestimmung raumakustischer Parameter, R. Pieren, Master's thesis, ETH Zurich, Switzerland, 2008