Technical University of Denmark

RBA Toolbox report

Room and Building Acoustics Toolbox for MATLAB

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1 Introduction

Room acoustic measurements in general requires a location, measurement equipment, such as microphones and speakers, and a recording device. It is furthermore essential to follow a measurement procedure that comply with standards, specifically as *ISO-3382 Measurement of room acoustic parameters*. Room acoustic measurements can therefore become a tedious task when post-processing of data is also considered.

The Room and Building Acoustics Toolbox (RBA Toolbox for short) for MATLAB is build upon an idea of integrating measurements and post-processing of measurement data into one single platform. Room acoustic measurement software such as Dirac and Pulse is often used and is regarded as industry standard. The transfer of measurement data from this software to other platforms e.g. MATLAB for post-processing is however a tiresome process. The RBA Toolbox wishes to eliminate this process by doing both measurements and post-processing within MATLAB. This can save time for the room acoustic community who mostly use MATLAB for post-processing of data.

The RBA Toolbox is open distributed software. This is done to encourage students, and other acoustic engineers, to observe and dig into the processing principles of room acoustic measurements and post-processing. And hopefully to engage in making this toolbox even more powerful and versatile in the future.

The ideas behind the RBA Toolbox is described in this report along with an experimental validation of the software. It has been emphasized in the making of the RBA Toolbox that it must comply with ISO 3382. The experimental validation has been performed by comparing room impulse responses measured with Dirac software and the RBA Toolbox in 3 locations; a large reverberation chamber ($T_{60} \approx 5\,\mathrm{s}$ @1 kHz), a small room ($T_{60} \approx 1\,\mathrm{s}$ @1 kHz) and a scale model.

2 Acknowledgments

We would like to thank Antoni Torras-Rosell for providing us with several function for the toolbox, many useful discussions and helpful feedback. Bastian Epp, Jonas Brunskog and Cheol-Ho Jeong for answering so many of our questions.

3 Project Overview

The RBA toolbox has been developed specifically to fulfill the needs of students and academic staff working with room acoustics and processing data in MATLAB. It provides the basic framework for common measurement procedures and make the basic tools for data processing available.

The RBA toolbox aims to give

- a flexible, platform-independent, and easy-to-use measurement procedure that comply with ISO 3382
- a basic set of tools for common operations regarding post-processing of data
- a framework for ongoing development and sharing of code

In general room acoustic measurements can be divided into four seperate tasks or areas

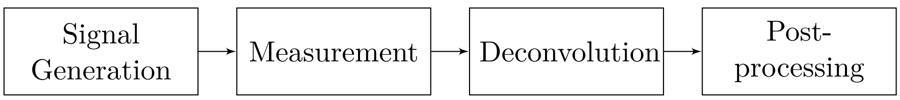

Figure 1: Steps in room acoustic measurements

The necesarry functions for signal generation and deconvolution was provided courtesy of Antoni Torras-Rosell [1]. So the main focus of the project has been to implement a method for performing the actual measurements using *MATLAB* and post-processing.

For measurements we ended up with a solution relying on an external toolboox *PsychToolbox*. This way it is possible to perform asynchrounous measurements on multiple platforms.
For post-processing it was necesarry to create functions for filtering the impulse reponse, calculating decay curves and determining reverberation times. Furthermore some calculations for room acoustic parameters such as Clarity, Definition and Early Decay Time has been implemented.

4 Measurements

MATLAB has built-in functions for audio playback and recording, using the `audiorecorder` and `audioplayer` objects. But when making room acoustic measurements it is necessary to do asynchronous, low-latency playback and recording. This is not possible using the built-in *MATLAB* functionality, and for this reason we have chosen to use the external toolbox *PsychToolbox*. The toolbox is designed for psychoacoustic measurements, giving a stimuli to a user while receiving an input from the user and has many implemented functions.
One of the things included in the toolbox is a low-latency, multi-platform audio driver based on the *portaudio*[1] library.
Using this we are able to play a signal and record the response of the room at the same time. Since the measurement procedure is the same, independent of the measurement signal, a single function `rbaMeasurement` is used for all measurements. It takes the desired measurement signal and outputs an ensemble average of the recorded signals, doing a number of transient measurements, defined by the user.

At the beginning of a new measurement, two channels are set up. One for playback of the measurement signal and another for recording.

```
playHandle = PsychPortAudio('Open', [], [], latency, fs, nrChannels);
recHandle = PsychPortAudio('Open', [], 2, latency, fs, nrChannels);
```

It is possible to use a single channel for simultaneous playback and recording with `PsychPortAudio`, but the *PsychToolbox* documentation warns that this may cause problems on some hardware configurations.
For convenience, the system default input and output devices are used. This means that if an external sound card is needed e.g. for measurements with high sampling rates, the sound card must be selected in the System Preferences of the Operating System.

Playback with *PsychToolBox* works by filling a playback buffer with the desired signal and initiate playback using the `'Start'` command. Prior to recording a recording-buffer must be allocated. It is important that this buffer is large enough that it does not become full during measurements. The recordingbuffer is periodically emptied and the content is appended to a vector `recordedAudio`.

The procedure of a measurement is shown in Figure 2. The number of repetitions depends on the desired number of averages. After measurements are done, the recorded signals are aligned and the mean is calculated.

[1] For information visit `http://www.portaudio.com`

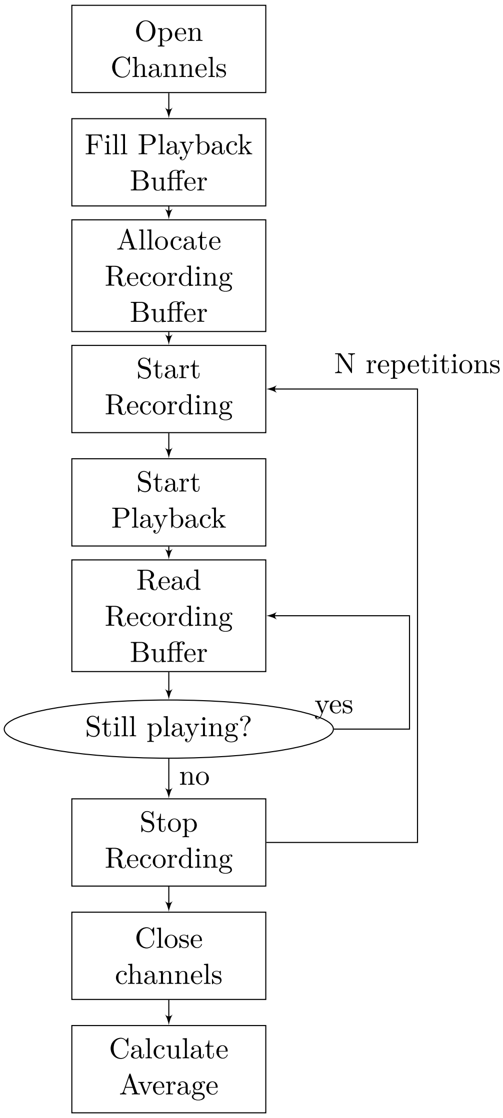

Figure 2: Diagram overview of rbaMeasurement.

The signals are aligned by calculating the cross-correlation between the measurement signal and the recorded signal using the `rbaCrossCorr`-function. `rbaCrossCorr` replaces the built-in function `crosscorr`. It uses the following relation.

$$\mathcal{F}\{f \star g\} = \mathcal{F}\{f\}\mathcal{F}\{g\}$$

Which allows for faster computation than `crosscorr`. Similarly the functions `rbaLinearConv` and `rbaCircularConv` are included to perform faster convolution operations.

When performing a measurement, there are two ways to take the reverberation in the room into account. One is to build up a sound field in the room by playing a measurement signal prior to measuring. This is called the steady state method. By doing this the measurements can be performed in succession with little to no pause between excitation sequences (as long a it is periodic), and an MLS sequence can be played over a long period at almost non-perceivable levels, but still end up with a high SNR.

The other way is to play the measurement signal and wait for the response to decay. This is called a transient measurement. The transient method requires that a delay between measurements are inserted, which must correspond to the reverberation time in the room. It takes up more time, but produces very stable results. The RBA Toolbox only supports the transient method, due to stability problems in the implementation of the steady state method.

5 Post-processing

5.1 Filters

An important aspect of room acoustic post-processing is to analyze the impulse response in octave or third-octave frequency bands. This means that it is necessary to filter the impulse response prior to calculating reverberation time or other parameters.
The filters are constructed using the MATLAB Signal-processing Toolbox and are 3rd order Butterworth-filters. The cut-off frequencies are the limits of the frequency band so for octave band filters the cut-off frequencies are

$$f_{lower} = f_c \cdot 2^{-1/2}$$
$$f_{upper} = f_c \cdot 2^{1/2}$$

Where f_c is the center frequency of the given frequency band.
In an article in *Journal of Sound and Vibration* Jacobsen [2] describes the influence of a bandpass filter on the decay curve. The delay in the filter skewers the decay curve, leading to miscalculations of the reverberation times. This happens unless the impulse response of the filter is significantly shorter than the impulse response of the system. Generally the decay is affected unless the following relation is fulfilled

$$BT_{60} > 16$$

Where B is the bandwidth of bandpass filter and T_{60} is the reverberation time of the system. This means that the problem is biggest at low frequencies in rooms with short reverberation times. And especially when using 1/3-octave bandpass filters.
In a letter to the editor [3], a solution is proposed. The effect of the filters can be reduced by time-reversing the signal prior to filtering. For this reason the function `rbaIR2OctaveBands`, which is used to filter impulse-response, has the option to time-reverse the signal before filtering.

5.2 Cropping impulse responses

Normally cropping of an impulse response, i.e. getting rid of onset and tail of noise, is a users job, and many times the user do okay. But it is desirable to get consistent crops, when dealing with many impulse responses, and therefore some effort has been put into an automatic procedure for cropping.

Locating the onset

The onset of the impulse response can be located numerically. The method proposed in the ISO-3382-1, uses a max-search in the normalized squared impulse response, which locates the peak. The onset is then defined as the first sample in the impulse response that reaches this maximum $-20\,\mathrm{dB}$[4].

The functionality is implemented in `rbaCropIR`, and figures 3 and 4 show an example of an impulse response before and after cropping with `rbaCropIR`.

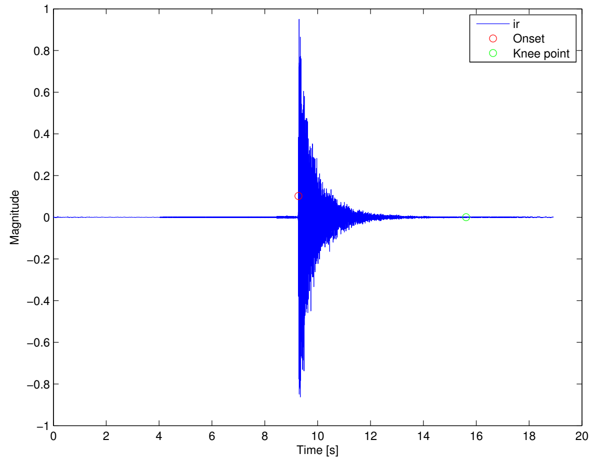

Figure 3: Impulse response of the reverberation chamber at DTU, measured with RBA, with onset and knee point marked.

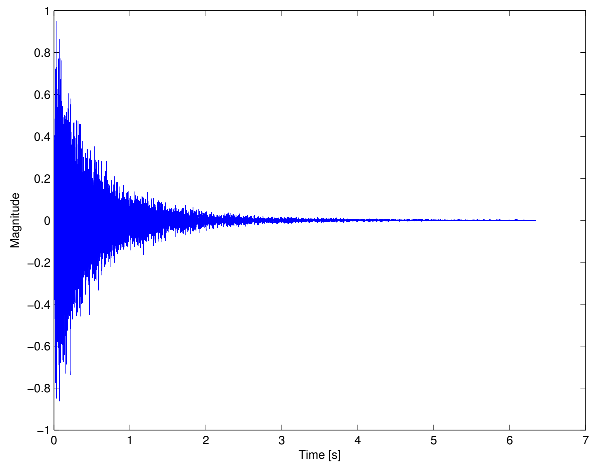

Figure 4: The same impulse response after being cropped.

This method is however, not precise enough when acoustic parameters are computed from early energy ratio such as Clarity, C_{80}, Definition, D_{50}, and Centre Time, T_S. According to Defrance and Polack [5], an onset change of 2 ms can result in a 50% difference of the computed C_{80} value. Of several methods discussed in the paper only a few are suited for our purposes, but more investigations are needed in order to determine the best choice for future implementation. In order to achieve consistency in the computation of Clarity, Definition and Centre Time, a more practical solution, also described by Defrance and Polack [5], is chosen: The onset of the impulse response is determined from the index of the maximum value minus 5 ms (this is also the approach used in the `Midas` package [5]. This is implemented into the functions and will not be obvious to the user, but only ensure more consistent results.

5.3 Lundeby

In line with the importance of finding the peak of the impulse, the intersection between the decay and the noise floor is also important. By precise location of this point, called the knee point, a larger dynamic range is obtainable for calculations of decay curves.

Lundeby has provided an iterative method for calculation of this knee point [6]. Several other methods have also been investigated, but they were found to be either too computationally heavy or too simple to provide reliable results. The lundeby method are seamlessly implemented into `rbaSchroeder`, but can be overwritten by an input, defining a knee point to be used instead. In figure 3, the knee point from the lundeby algorithm is shown.

However, there are big differences in the decay curves, between finding the knee point for a broadband signal, and find one for each frequency band. An example is shown in figures 5, 6, and 7.

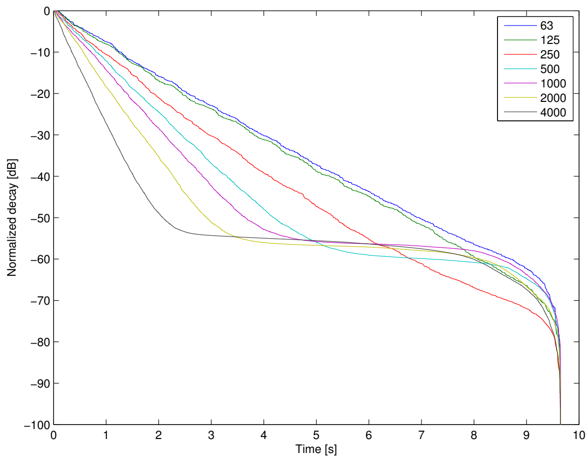

Figure 5: Decay curves from impulse response, only cropped at the onset. No knee points are found or set.

Figure 5 shows the decay curves of an impulse response, which has only been cropped at the onset. The signal is then filtered in octave bands, and the decay curves are found by a simple schroeder integration. As seen in the figure, there is about 50 dB of dynamic range.

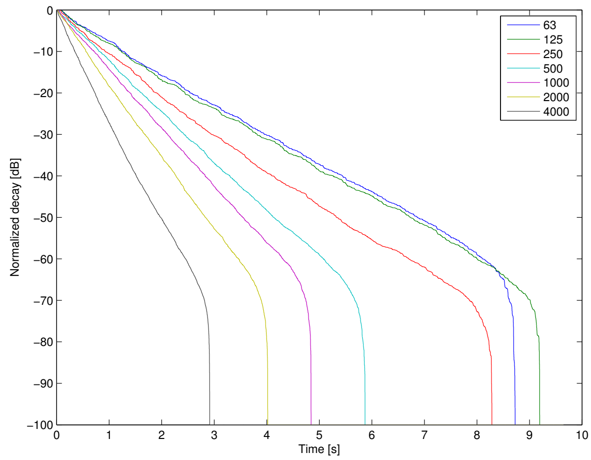

Figure 6: Decay curves from impulse response, cropped at the onset and in frequency bands.

Figure 6 shows the decay curves of the same impulse response, but where the lundeby method has been used to find a knee point in each octave band before performing the schroeder integration (the default behavior of `rbaSchroeder`). It is evident, that the dynamic range is increased from the previous example. Now there is enough dynamic range to calculate a full T_{60}, i.e. 65 dB, in almost all bands.

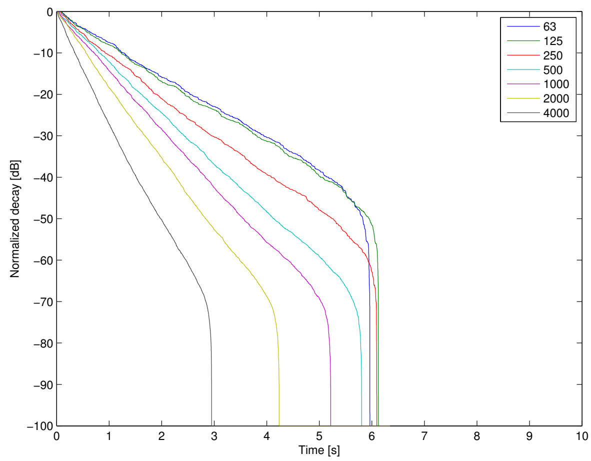

Figure 7: Decay curves from impulse response, cropped at the onset and on the broadband signal.

The last example of decay curves are shown in figure 7, where the same impulse response has been cropped at the knee point of the broadband signal, instead of finding knee points for each frequency band. This results in a far smaller dynamic range for low frequencies, especially below 500 Hz, where a loss of 10 dB of dynamic range is seen, compared to the last example.

It is clear that care must be taken when performing crops to the right parts of the signal at the right time of processing. The best results are seen, when cropping the broadband signal at the onset, and afterwards cropping at the knee points for each frequency band. The default behavior of `rbaCropIR`, `rbaIR2OctaveBands`, and `rbaSchroeder` encourages this method (see demoDecayCurve.m), but RBA packages are capable of doing it exactly the way the user wants.

6 Verification

In order to verify the functionality of the toolbox, a number of measurements have been conducted with the Dirac system as the reference. The two systems are compared by doing measurements with the (exact) same setup except for the computer hardware and software. Multiple measurements (each with several pre-averages) were recorded and stored for later post-processing. Two sets of impulse responses, obtained with Dirac and the RBA toolbox, respectively, are compared by computing the rms-deviations between impulse responses of the same set,

$$RMS^{\mathrm{Dirac}} = \sqrt{\left\langle \left(h_i^{\mathrm{Dirac}} - h_j^{\mathrm{Dirac}} \right)^2 \right\rangle} \quad \text{for} \begin{cases} i = 1, 2, \ldots, n-1 \\ j = i+1, \ldots, n \end{cases} \tag{1}$$

and similar for the RBA toolbox measurements. n is the measurement number (containing several pre-averages).

6.1 Transient sweep method

The measurements were carried out in the large reverberation chamber at DTU with a $V = 215$ m^3 with T_{60} peaking at around 8 s at low frequencies. Measurements with a logarithmic sine sweep were generated and recorded by Dirac and the RBA Toolbox. The reproducibility was emphasized by doing 5 measurement with each 5 averages of 5.46 s sweeps with the same source and receiver positions that comply with ISO 3382.

The rms deviations are shown in Figure 8. The measurements obtained with the RBA toolbox are within the confidence intervals of the Dirac rms deviations, except for two outliers. Tables of acoustic parameters are found in Tables 1-5 and plots of T_{60} values are compared to Diracs derived values in Figure 9. T_{30}, EDT, D_{50} and T_S values are all very similar to Diracs values. Larger errors are seen when comparing C_{80} values. These deviation most likely arise from a different computation of the onset of the impulse response [5]. The Schröeder frequency of the room is given by

$$f_{\mathrm{sch}} = 2000\sqrt{T_{60}/V} \approx 300 \text{ Hz} \tag{2}$$

The rms-deviations are shown in Figure 10. All RBA measurements fall within the confidence interval of the Dirac measurements, except for one outlier.

[Hz]	63	125	250	500	1000	2000	4000	8000
Mean	7.51	7.43	6.11	5.01	4.33	3.51	2.38	1.43
Std. Dev	0.41	0.35	0.07	0.08	0.07	0.08	0.06	0.04
Dirac	7.42	7.36	5.97	4.98	4.31	3.36	2.35	1.41

Table 1: T_{30} [s] values for octave bands from measurements with RBA and Dirac.

[Hz]	63	125	250	500	1000	2000	4000	8000
Mean	7.26	6.64	5.37	5.09	4.36	3.51	2.11	1.25
Std. Dev	0.90	0.84	0.10	0.12	0.11	0.09	0.04	0.04
Dirac	7.87	7.33	5.54	5.24	4.49	3.60	2.18	1.22

Table 2: EDT [s] values for octave bands from measurements with RBA and Dirac.

[Hz]	63	125	250	500	1000	2000	4000	8000
Mean	-6.52	-5.72	-5.48	-4.57	-3.83	-3.43	-1.39	2.14
Std. Dev	0.24	0.72	0.16	0.12	0.15	0.06	0.12	0.11
Dirac	-5.61	-6.34	-5.07	-3.69	-3.56	-3.13	-1.25	2.06

Table 3: C_{80} [dB] values in for octave bands from measurements with RBA and Dirac.

[Hz]	63	125	250	500	1000	2000	4000	8000
Mean	0.15	0.13	0.18	0.22	0.18	0.23	0.31	0.44
Std. Dev	0.01	0.02	0.01	0.01	0.01	0.00	0.01	0.00
Dirac	0.22	0.11	0.20	0.25	0.22	0.25	0.34	0.43

Table 4: D_{50} values for octave bands from measurements with RBA and Dirac.

[Hz]	63	125	250	500	1000	2000	4000	8000
Mean	0.47	0.44	0.36	0.34	0.30	0.24	0.15	0.09
Dirac	0.47	0.47	0.37	0.33	0.28	0.24	0.15	0.09

Table 5: T_S [s] values for octave bands from measurements with RBA and Dirac.

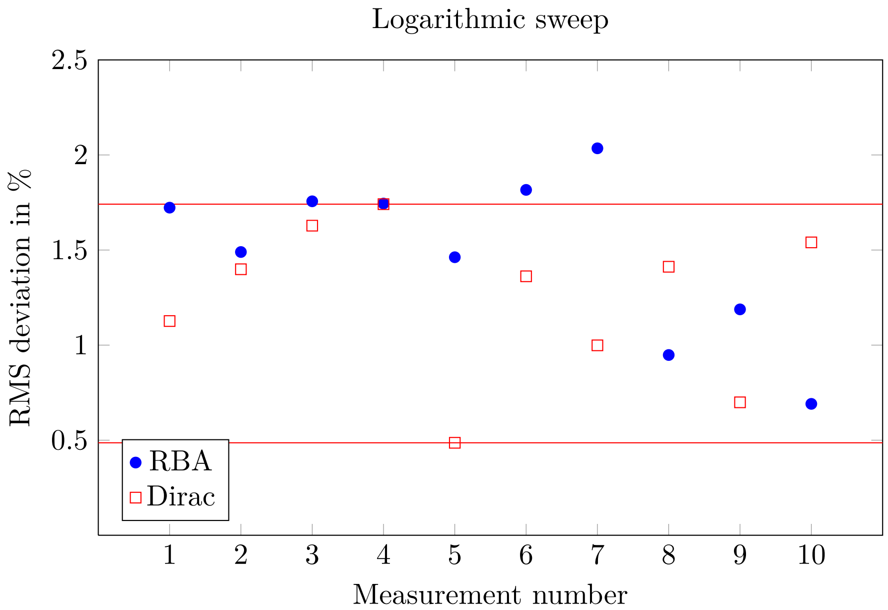

Figure 8: RMS deviation in %. The horizontal lines indicates the [5%, 95%] confidence intervals of the Dirac measurements.

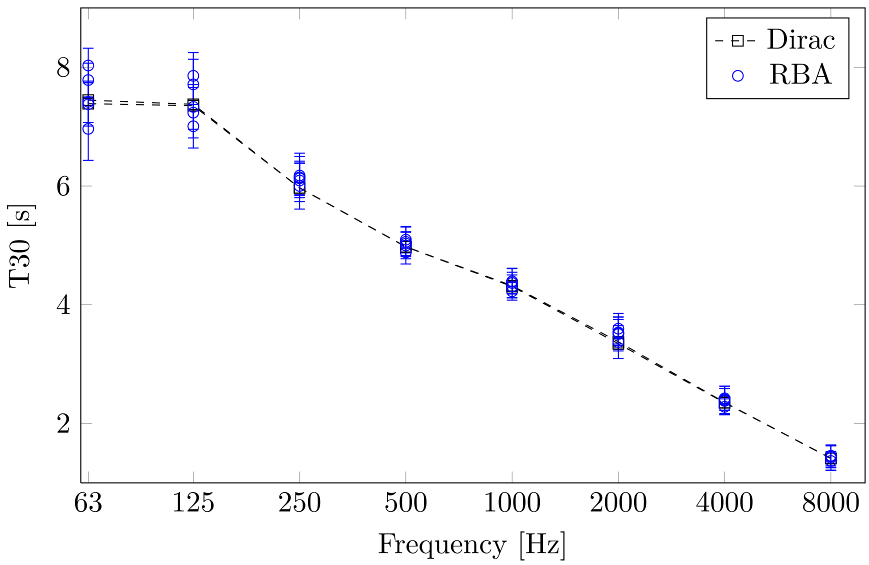

Figure 9: T_{30} comparison between RBA and Dirac. Both systems used logarithmic sweeps of 5.46 s and 5 pre-averages. A total of 5 RBA measurements and 2 Dirac measurements are shown

6.2 MLS measurements

The measurements were conducted in a small room $V = 43\ \mathrm{m}^3$ with a $T_{60} \approx 1$ s @ 1 kHz. 6 measurements with MLS sequences of 10.92 s were used to record the room impulse response. The impulse response were averaged over two measurements.

The rms-deviations are shown in Figure 10. All RBA measurements fall within the confidence interval of the Dirac measurements, except for one outlier.

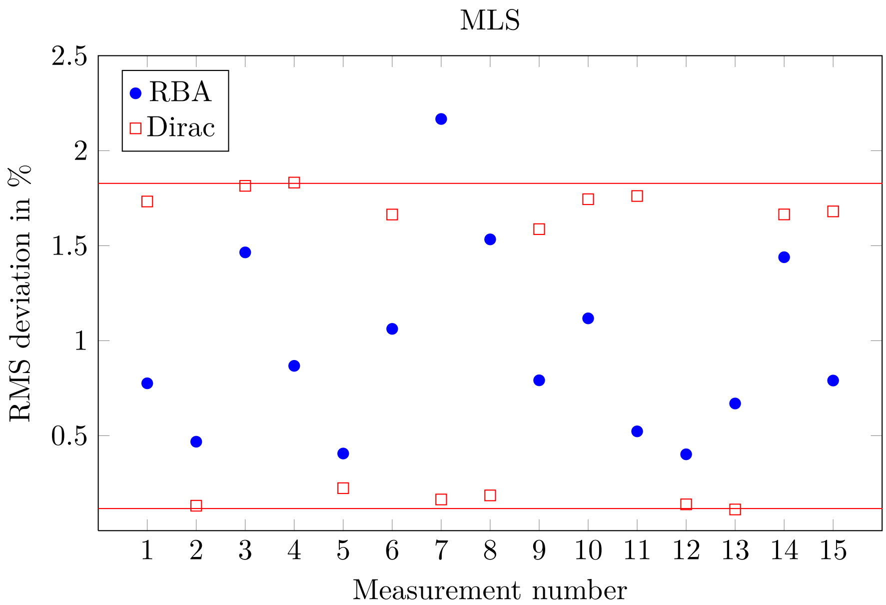

Figure 10: RMS deviation in %. The horizontal lines indicates the [5%, 95%] confidence intervals of the Dirac measurements.

6.3 Scale model

Scale model measurements can be of great interest for e.g. the design of a concert hall or absorption measurements where a full-scale room is not available. The idea behind this procedure is, that the basic parameters: length, frequency and time are scaled according to the model size. The connection between the model parameters and full-scale parameters (capital letters) in a

$1/\sigma$ scale model is

$$\text{Length} \quad L = l\sigma \tag{3}$$

$$\text{Time} \quad T = t\sigma\frac{c'}{c} \tag{4}$$

$$\text{Frequency} \quad F = \frac{f}{\sigma}\frac{c'}{c} \tag{5}$$

$$\text{Sampling rate} \quad F_s = \frac{f_s}{\sigma} \tag{6}$$

where c' is the speed of sound in the scale model [7].

Sound is absorbed during propagation due to molecular relaxation, viscosity and heat conduction [8]. Sound absorption in air depends on frequency, temperature, humidity, atmospheric pressure and hence distance of propagation. It follows that the conversion of an impulse response to full scale requires taking the absorption discrepancy into account. If m_{mod} denotes the air attenuation coefficient in the scale model and m_{full} denotes the full scale, then the discrepancy can be determined by applying a time dependent gain factor $H_n(t)$ to the octave band filtered impulse response around f_n

$$H_n(t) = 10^{b_n t/20} \tag{7}$$

$$b_n = m_{\text{mod}}(10f_n)c' - Km_{full}(f_n)c \tag{8}$$

where c' and c is the speed of sound in the measurement and reference, respectively [8].

The implementation of this conversion of impulse responses from scale models has been included in the RBA toolbox. It has however, not yet been thoroughly tested and it is therefore not possible to verify its functionality.

7 Conclusion

The RBA toolbox has been developed over 13 weeks as part of a special course. The process has involved many different aspects: discussions on the scope of the toolbox and the primary user base, researching, choosing and implementing algorithms and testing, validation of measurement procedure and a lot of bug fixing.

It has been shown that the RBA toolbox can provide users with a reliable collection of functions for data processing along with a validated procedure for measurements of the room impulse response.

It has been necessary to leave out several functions and ideas in order to reach the deadline. Some of which we hope will be implement at a later stage is

- Steady-state measurements will decrease measurement time and make long MLS recordings in noisy surrounding possible. The preliminary work has already been done but testing is still needed before this functionality can be implemented.

- Adding a noise compensation option in `rbaSchroeder` which could be useful to get more dynamic range from decay curves. The preliminary work has been done.

- Making the toolbox independent of *PsychToolBox* would decrease file size greatly and make installation easier for the user.

It is our hope that students and academic staff will adopt the RBA toolbox and contribute with bug fixing and additional functionality.

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