Calibration of Room Impulse Response Measurement Chain with Free-field Simulated Models

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Abstract

Auralization techniques, used in many applications within the acoustics context, are powerful tools that allow for recreating sound stimulus via convolution of an anechoic recording (of the desired sound signal) with the binaural room impulse response (BRIR) of a specific environment. Nevertheless, the frequency response (FRF) of the instrumentation employed in the BRIRs (or RIRs) evaluations may generate a negative impact (coloration) in the auralization results. In this context, this study proposes a method to extract the free-field response of the measurement chain. The technique consists of finding the direct sound, the first reflection inside the RIR and designing a hybrid window (with smooth decay) to avoid spectrum leakages. To test effectiveness, five distinct sound sources were used to obtain the BRIRs of an auditorium/lecture room at the Federal University of Santa Maria, Brazil. The frequency coloration caused by the instrumentation can be corrected with an appropriate inverse filtering process (extracted from the instrumentation free-field response) to properly convolve the anechoic recordings. Thus, the BRIRs would contain less impact from the instruments FRFs. Ultimately, the following step is the comparison concerning corrected and uncorrected outcomes.

Introduction

In the study of room acoustics, there are several methods of characterization to predict the acoustic field inside different environments. In the experimental scope, obtaining the Room Impulse Response (RIR) or Binaural Room Impulse Response (BRIR) [1] allows for the calculation of different parameters such as TR, EDT, D50, C80, etc [2, 3]. This approach is based on the basic principle that a source-receiver system in an environment can be considered a Linear and Time-Invariant System (LIT) [4]. Thus, it can be fully characterized by its Impulse Response (IR).

To measure the RIR (or BRIR) for a specific source-receiver position, it is necessary to have a source in place of the transmitter and a microphone (or torso simulator) in the receiver position. Figure 1 illustrates a typical measurement chain used in room acoustics. A signal is generated by the measurement software and converted from the digital to the analog domain by the digital-to-analog converter (DAC). In the following, the signal passes through an amplifier (AMP) to power the sound source transducer. The environment is excited and the microphone (mic) acquires the signals, sending them to an analog-to-digital converter (ADC). Accordingly, the recorded data is processed in the software to perform the necessary operations to estimate the RIR. This enables convolving the RIR with a desired anechoic recording, simulating as if this recording were reproduced inside that given room.

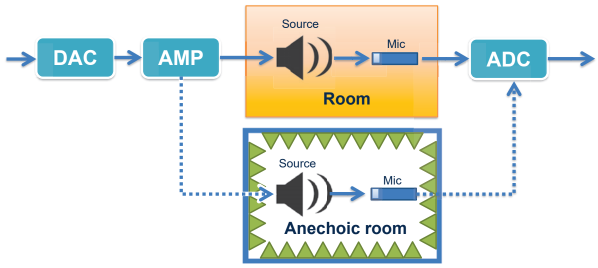

Figure 1: Typical measuring chain for RIR extraction.

Observe Figure 1. Each element of the measurement chain has its own IR, introducing undesirable colorations into the final RIR. In other words, the obtained RIR is the result of the convolution of each IR of the measurement chain – since the elements are assumed to be linear and time-invariant.

Fortunately, it is possible to use signal processing tools to partially correct the measurement chain response. Thus, virtual results would become more faithful in comparison to the real situation. That is, the sound source and listener would be virtually well placed inside the room (creating a verisimilar feeling of the physical/real situation). To accomplish this idea, it is necessary to know the response of the measurement chain (MC). Therefore, the design of a suitable inverse filter is needed (to pursue the elimination of the MC influence over the RIR).

Strictly speaking, it is usually necessary to use an anechoic chamber to obtain the chain response (note the dashed path in Figure 1). However, this makes calibration expensive and often impracticable. Subsequently, this research investigates the efficacy of Simulated Free-Field methods (SFF) to obtain the anechoic response of measurements in room acoustics. Thus, it is divided into two main parts: the first is a simulation of the SFF methods (convolving ideal RIRs with anechoic responses of typical loudspeakers). And, the second is the measurement of a MC in two different rooms with distinct sound sources and different source-receptor distances. In conclusion, it was possible to compare the effectiveness of the SFF methods on correcting the response of simulated and experimental RIRs.

Simulated Free-field Simulation and Validation

The tested SFF methods used were the Time Delay Spectrometry (TDS) developed by Heyser [5] and RIR windowing techniques, explained by Muller and Massarani [6]. As later explained, during the simulations the windowing method showed improved results (see Figure 6). For this reason, it was adopted as the standard process.

To work with the SFF models, the processing shown in the flowchart of Figure 2 was followed. At first, simulated room impulse responses ($\mathrm{RIR}_{\mathrm{simulated}}$) were created to serve as known basic cases.

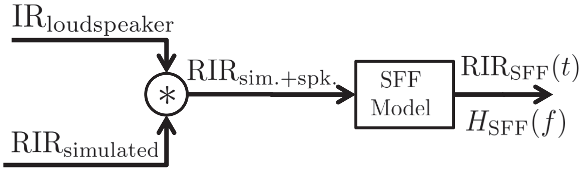

Figure 2: Overview of the SFF simulation. The sign "$*$" denotes the convolution operation and the SFF block denotes the SFF method applied to the full $\mathrm{RIR}_{\mathrm{sim.+spk.}}$.

Originally, the RIRs were obtained by three different simulation methods:

1. In the first method[1], the source-receiver system is placed at h_{SR} meter above an infinite reflector plane, see Figure 3. This model results in a RIR containing two pulses, direct sound and first reflection (representing a plausible simulation of outdoor measurements). In this way, it is possible to evaluate the required height of the system for a good estimation of the frequency response via SFF.

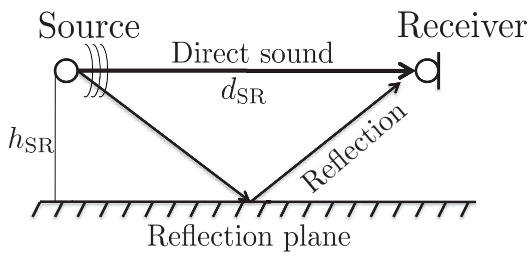

Figure 3: Simulated RIR, Method 1.

2. The second method considers a rectangular room using Image-Source Mirror (ISM) [1, 2, 7], as shown in Figure 4. This model results in an RIR containing the first reflections and reverberant tail for a rectangular room. Therefore, it is representative for the application of SFF models, given the common number of rooms available with these proportions. Moreover, the use of the source-receiver system under symmetry conditions allows better temporal separation between direct sound and first reflections.

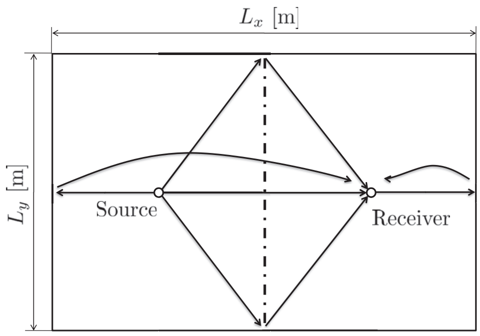

Figure 4: Simulated RIR, Method 2 (ISM).

3. The third approach evaluates a 2D model using Finite Elements Method (FEM) [8]. This method was used to obtain RIRs based on acoustic wave theory. Thereupon, the timestamp of reflections is not in form of delta functions $\delta(t)$, like in Methods 1 and 2. The wave's direction of arrival (DOA) does not exist. Thus, the outcome of this method, as well as measurements, needs more sophisticated ways of determining the reflections.

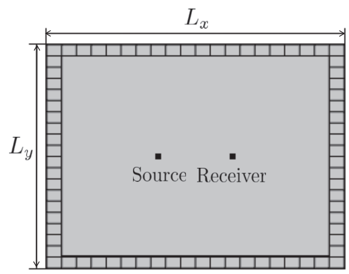

Figure 5: Simulated RIR, Method 3 (FEM).

Considering the simulations, this paper will only present the process used in Method 1. Further results and details about the RIR simulation and SFF models can be consulted in Brum's undergraduate thesis [9].

The applied windowing consisted in separating the direct and reflected sound. The time difference between them is called Initial Time-Delay Gap (ITDG), usually depicted in a reflectogram plot [2, 3]. The basic process seeks and identifies local maxima, enabling the separation. This approach has worked well with the aforementioned Methods 1 and 2, which can output reflectograms. Nevertheless, for Method 3 and measurements, an energy-based approach called *Local Energy Ratio* [10] had to be utilized due to the proximity of local maxima in the IRs. This overcomes the limitations of the basic peak detection process, making the approximate localization of the direct sound possible. Finally, a hybrid time-window (rectangular combined with Blackman-Harris) is constructed and applied. Observe the upper part of Figure 6.

The following step needs an instrumentation IR for testing. Thus, a free-field measured response of a Yamaha HS50M loudspeaker was used to calculate its impulse response ($\mathrm{IR}_{\mathrm{loudspeaker}}$). To include the influence[2] of the room into

[1]Similar to the ray tracing method.

[2]This procedure is reciprocal, i.e. it also includes the influence of the sound source response in the simulated RIRs.

the $\mathrm{IR}_{\mathrm{loudspeaker}}$, it was convolved with the $\mathrm{RIR}_{\mathrm{simulated}}$. This yields (observe Figure 2)

$$\mathrm{RIR}_{\mathrm{sim.+spk.}} = \mathrm{RIR}_{\mathrm{simulated}} * \mathrm{IR}_{\mathrm{loudspeaker}} \,, \qquad (1)$$

where $*$ denotes the convolution operator [4].

Accordingly, the proposed SFF models are applied to $\mathrm{RIR}_{\mathrm{sim.+spk.}}$, resulting in $\mathrm{RIR}_{\mathrm{SFF}}(t)$ in time domain or $H_{\mathrm{SFF}}(f)$ in frequency domain. The comparison between $\mathrm{IR}_{\mathrm{loudspeaker}}$ and $\mathrm{RIR}_{\mathrm{SFF}}$ (i.e. between the original IR and the resulting IR after the removal of the room response) was carried out, showing the similarities. This represents the effectiveness of the SFF applied.

Figure 6 shows the total $\mathrm{RIR}_{\mathrm{sim.+spk.}}$ simulated (for Method 1 and $h_{\mathrm{SR}} = 1$ m, blue line); hybrid time-window (dashed line); windowed $\mathrm{RIR}_{\mathrm{SFF}}$ (orange line) and TDS response (red line). The sound received was simulated above an infinite reflector plane, thus, its impulse response is the superposition of two Dirac functions. That is

$$\mathrm{RIR}_{\mathrm{simulated}} = \delta(t - \tau_1) + \sqrt{1 - \alpha} \cdot \delta(t - \tau_2) \,, \qquad (2)$$

where τ_1 is the direct sound arrival time, τ_2 is the first reflection arrival time and α is the absorption coefficient of the plane.

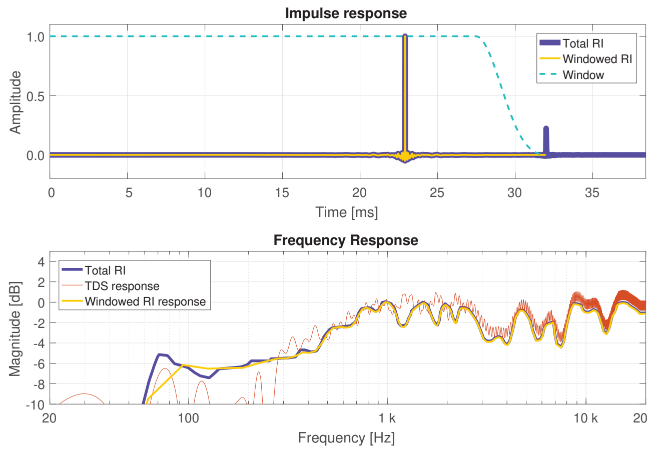

Figure 6: Simulated results for Method 1.

The ripples in FRF obtained via TDS (comb filtering effect) occur because the low-pass filter does not completely reject the reflection effect. That is, it was not capable of removing the beats on the output of the modulators. To improve these results, the sweep rate of the excitation signal must be adjusted in conjunction with the cutoff frequency of the low-pass filter. Some aspects of the TDS basically depend on the critical sense of the experimenter, and are difficult to automate. In this context, the robustness of the time-windowing techniques is evident.

Experimental Measurements

The experimental tests were performed following the measurement chain depicted Figure 7. Both measurement sites are located inside the Federal University of Santa Maria (UFSM), RS, Brazil. The first environment was the Recording Studio Sérgio Assis Brasil (SAB). The room has dimensions of 12 m x 8 m x 8 m, with acoustic absorbers on the walls and ceiling. The floor is made of smooth concrete, providing specular reflections and very low acoustic diffusion. Therefore, the microphone[3] (receiver) was positioned laying on the floor. This was the closest condition possible to an anechoic situation. The measurements were carried out with source-receiver distances of 1 m, 2 m, 4 m and 8 m, to verify the 6 dB decay per each distance doubled. Moreover, a symmetry axis was adopted, see Figure 8.

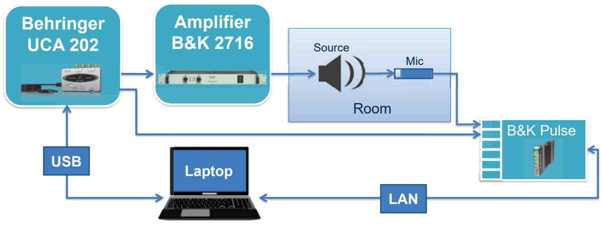

Figure 7: Measurement setup flowchart.

The second measuring environment was Room 355, an auditorium of the UFSM Technology Center (11.75 m x 6.48 m x 3.32 m). Two sets of measurements took place in this room:

1. on axis, similar as the SAB configuration, but the mic was placed 1.5 m above the ground (see Figure 8), and
2. *in situ*, where the sound source remained still in the position of the lecturer and 5 receivers were configured in the chairs (a typical room application), as observed in Figure 9.

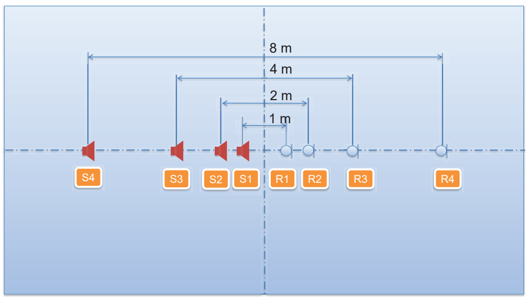

Figure 8: Measurement site with on axis source-receiver conditions (distances of 1 m, 2 m, 4 m and 8 m).

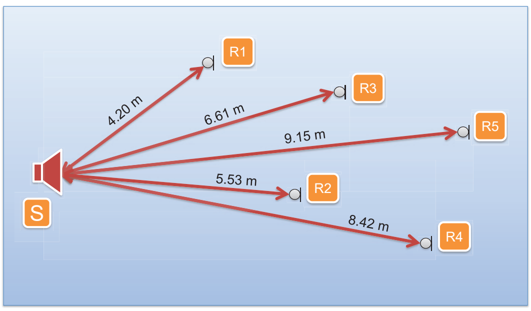

Figure 9: Measurement *in situ* in Room 355 (the sound source remained still in the position of the speaker and 5 receivers were configured in the chairs).

[3] Brüel & Kjær Type 4189 1/2-inch free-field microphone.

From the three measurement conditions (SAB on axis; Room 355 on axis and *in situ*), the RIRs were properly windowed and the frequency responses were estimated. Thereupon, finite impulse response (FIR) inverse filters were created aiming to correct the influence of the measurement chain.

Figure 10 shows the correction inverse filters for the dodecahedral sound source that was used during the measurements. The OmniPower Sound Source Type 4292 by Brüel & Kjær is usually utilized in room acoustics applications. Thus, considering the size of its loudspeakers and the frequency range indicated by the manufacturer, an *operational range* for correction was defined from 50 Hz to 10 kHz. The idea is to not touch the amplitudes which the sound source does not properly radiate, possibly avoiding the increase in unwanted noise.

The measurement in SAB Studio was chosen as the reference, since it is considered a *silent room*[4]. However, the comparison among the filters shows that is possible to extract a correction filter using simulated free-field methods (SFF) also for other conditions. It is important to notice that better results via SFF are achieved when the direct sound and the first reflections are more apart (increased ITDG). That being the case, the source-receiver distance can be optimized for each case.

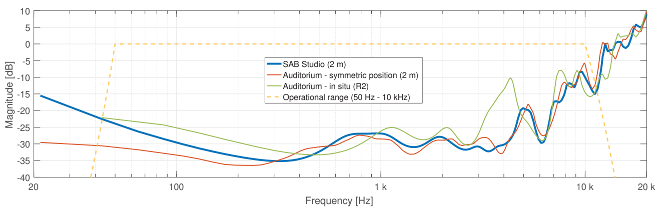

Figure 10: Frequency response of correction filters obtained from the three measurement conditions using dodecahedral sound source (B&K Type 4292).

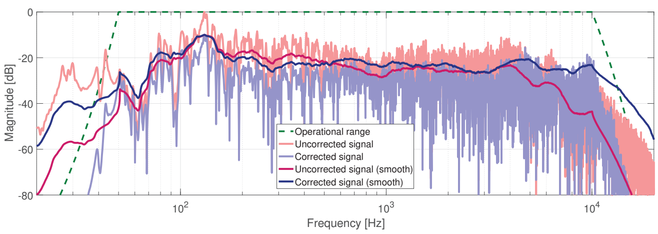

Figure 11: Comparison between corrected and uncorrected spectra (applying the inverse filtering).

In Figure 11 it is possible to observe one example of the inverse filtering application. Room 355 *in situ* RI measurement (R2) was convolved with a linear sweep (flat spectrum), and, in the following, convolved with the inverse filter (estimated in SAB Studio). That is,

$$\mathrm{RIR}_{\mathrm{corr.}} = (\mathrm{RIR}_{\mathrm{R2,\ uncorr.}} * \mathrm{IR}_{\mathrm{l.sweep}}) * \mathrm{IR}_{\mathrm{inv.filter}}\,. \quad (3)$$

The plot contains: operational range (dashed line), uncorrected signal in red and corrected signal in blue. The narrowband is depicted in light colors and smoothed versions in dark colors.

[4]Here the term *silent room* is employed because the anechoicity of the room was not previously project. Thus, it would not be appropriate to call it an *anechoic room*.

Final Considerations

Results obtained in the simulations and experiments have shown that the SFF methods can be useful to correct the influence of the measurement chain in RIRs. However, the presented approach may require a sufficiently large environment with an optimized source-receiver position (increased ITDG). Therefore, it is possible to apply the technique in many rooms available in universities and schools (given the common geometry).

An important point to be reviewed is the useful frequency range of the sound source. The experimenter must be aware of its response to avoid corrections in frequencies in which the source does not radiate sufficient sound power to excite the room. In this case, it is interesting to combine a bandpass filter in the correction, defining an operational range. Accordingly, corrections will not be applied to bands outside the range of the measurement system. Without this restriction, unwanted noise may be increased, yielding a worsening in the scenario.

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