DEVELOPMENT OF ROOM IMPULSE RESPONSE PROCESSING SOFTWARE

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This paper describes the back and front-end development of an open source software for the calculation of EDT, T20, T30, C50, C80, Ts, Tt, EDTt and IACC acoustical parameters. These parameters are calculated from either a room impulse response or a recorded sine sweep. The software provides MMF and Schroeder's smoothing methods for obtaining the energy-time curve, and three background noise compensation for the later. The results are displayed in octave or third-octave bands. In order to validate the results, a comparison with two commercial softwares is made. For reverberation and clarity related parameters, good general results are obtained while IACC parameters shows greater discrepancies with commercial softwares. Tt, EDTt assessment can be made since it cannot be compared with commercial results.

Keywords: Room Acoustics, ISO 3382, Open source softwares, Python development

1. Introduction

A common goal in architectural acoustics is to characterize the acoustical quality of a close environment. This is necessary to assess the current acoustical state of an enclosure, as well as to propose solutions to potential problems based on predictions.
Although there are several methods to describe a room acoustically, the most widely used way is through the processing and analysis of its impulse responses (RIR). Currently, the most widespread method for obtaining RIR is from a Sine Sweep (SS). It is based on obtaining the

impulse response of the enclosure from the convolution between the recorded SS and its inverse filter.

The present work describes the development of a software in Python programming language for the processing of impulse responses and the calculation of the acoustic parameters commonly used derived from it. The development includes the design of signal processing algorithms, the graphical user interface (GUI) and the calculation of some of the acoustical parameters defined in the ISO 3382 standard [1]. These parameters are: T30, T20, EDT, C50, C80, T_s, T_t, EDT_t and IACC, calculated for octave and third octave bands.

This software is developed under the logic of open source software so that through the collaborative work of several users, optimization and maintenance can be achieved. This report presents the procedure used to design the program, as well as the results obtained and the validation of the code when comparing it with two recognized commercial programs: the Aurora plug-in for Audacity [2] and the EASERA [3].

This papers structures as follows: section 2 describes the state of art for IR processing, section 3 describes the theoretical framework in which this work base upon, section 4 describes the software and signal processing, section 5, 6 and 7 describes the results, its discussion and the conclusion of the work respectively.

2. State of Art

Nowadays, in order to characterize a room acoustically, its impulse response is processed using various software. The calculation process is usually different, so it is common to obtain large deviations depending on the used program.

The works of Jacobsen [4] and Rasmussen et al. [5] introduce the Inverted Time Analysis technique. Through it, the impulse response before filtering in thirds octave bands is temporarily reversed. In consequence, it avoids the typical ringing phenomenon which occurs in band-pass filters.

On the other hand, in the book "Diseño acústico de espacios arquitectónicos", Carrión [6] develops objective acoustic parameters commonly used in room acoustics, several of which are used in this work. In this book, the author defines one of the newest concepts in the area of acoustic measurements of rooms, the Texture of a room.

Bidondo and Pepino [7] continue to work with what was previously described. They define Early Reflections as each amplitude outlier present in an RIR, and the Mixing Time when its accumulated energy reaches 99%. They define descriptors that together determine the Texture of the room. These descriptors are Mixing Time, the Expected Texture, and the Distance Between Models.

Regarding impulse response processing programs, REW [8] is an open source software platform designed for the calculation of room acoustical parameters. It allows the stimuli generation and the measurement process within the software. On the other hand, Ángelo Farina developed AURORA module [2], a free tool that allows the rooms acoustic impulse responses to be measured and manipulated. It delivers several of the acoustic parameters used here from the study of the room impulse response. Another commonly used software for the IR processing is EASERA [3], developed by AFMG group. This software allows not only to obtain several of the acoustical parameters described in this work, but to obtain other parameters and descriptors of a signal.

3. Theoretical Framework

3.1 Room impulse response

The impulse response (IR) is defined as the temporal evolution of sound pressure observed in a room as a result of the emission of a Dirac impulse. The characteristics of any linear and time-invariant system are fully described by its impulse response *h(t)*. In practice, it is impossible to create and generate true Dirac delta functions, although very short transient sounds can provide good enough approximations for practical measurements. However, an alternative measurement technique is to use a maximum length sequence signal (MLS) (or other deterministic flat spectrum signals) and transform the measured response into an impulsive response [1].
Angelo Farina has developed an alternative method of impulse response measurement [9]. Under this method, a logarithmic sinusoidal sweep stimulus is used, in which the frequency varies exponentially as a function of time in the range of interest. This technique provides a considerable advantage over SNR compared to linear sinusoidal sweep, periodic pulse, or MLS techniques [10]. The logarithmic (or exponential) sine sweep can be described by applying Equation (1) [11].

$$x(t) = sin\left(\frac{2\pi f_i T}{ln\left(\frac{f_i}{f_f}\right)}\left(e^{\left[\frac{ln\left(\frac{f_i}{f_f}\right)*t}{T}\right]} - 1\right)\right) \tag{1}$$

where f_i and f_f is the initial and the final frequency, T is the sweep duration, and t are the time samples.
To obtain the IR of the room, it is necessary to convolve the measured signal with its inverse sweep filter. To generate the inverse filter, it is necessary to invert the time of the excitation signal and then apply (the inverse pulse) an amplitude envelope to reduce the level by 6 dB per octave [10].

3.2 Energy Time Curve (EDT)

An energy-time curve represents how the energy of sound decreases in a room as a function of time. From an analysis of an ETC, the acoustic behavior of the room can be predicted. ETC cannot be obtained directly from a real IR, but can be found by smoothing the room recorded IR. For this, two methods of smoothing are mainly used: Inverse Schroeder Integral or Moving Median Filter (MMF). The Inverse Schroeder Integral can be obtained by applying Equation (2)

$$ETC(t) = 10log\left(\frac{\int_t^{\infty} h^2(t)dt}{\int_0^{\infty} h^2(t)dt}\right) \tag{2}$$

where $h^2(t)$ represents the squared room impulse response.
In practice, it is not possible to integrate to infinity. This is the reason why, for this method, it is extremely important to find the correct limit for the integration. The limit must be defined at the exact point where the ETC drop crosses the noise floor. To find such a point, the Lundeby or Chu methods are often used.

The first method, as described by Lundeby in his article [12], consists of an iterative algorithm that estimates the level of background noise and the EDT slope until the crossing point between the two is found. On the other hand, Chu's method [13] consists of subtracting the root mean square (RMS) value of the background noise from the energy before being integrated by Schroeder. The RMS value is calculated from the last 10% of the signal, which is assumed to be the range that contains the background noise.
The inverse Schroeder integral can be calculated without compensation for background noise. In that case, the limit of the integral corresponds to the length of the RIR (theoretically, the integral is calculated to infinity).
The Moving Median Filter (MMF) can be applied instead of the Schroeder method. In this case, the value of each sample is replaced by the value of the median of the adjacent samples. For this, it is necessary to establish a time window in which the calculation will be carried out. It has to be taken into account that, for correct results in lower frequencies, a 50 ms windows is expected to cover up to 20 Hz.
Due to the fact that the reverberation time is frequency dependent, IR filtering is necessary in different frequency bands. The IEC 61260 standard specifies the procedures necessary for the development of octave and third octave band filters [14]. In particular, for the third octave band filters, an additional step was implemented. As Rasmussen et al. [5], for short reverberation times, this type of filter can introduce high levels of ringing. To avoid this problem, the authors propose the Time Reversed Analysis method. It consists of inverting the impulse response time before and after filtering. Time-reversed analysis has been shown to more accurately approximate the results of reverberation time calculations.

3.3 Acoustical Parameters

3.3.1 Early Decay Time (EDT) and Reverberation Time (T20 and T30)

The reverberation time is defined as the time required for the sound pressure level to decrease by 60 dB from its initial level once the source is shut off. The slope of the IR can be determined from the slope of the line fitted by linear regression to a suitable portion of the curve. RT is normally obtained from the portion of the decay between –5 dB and –35 dB (T30) or between -5 dB and -25 dB (T20) below the maximum initial level. Decay times are calculated from these slopes by extrapolating to a 60 dB dip.
Early Decay Time is derived from the reverberation time decay curve; between 0 dB and 10 dB below the initial level. A 60 dB level fall-off is extrapolated from the first 10 dB level fall-off. The result determines the EDT.

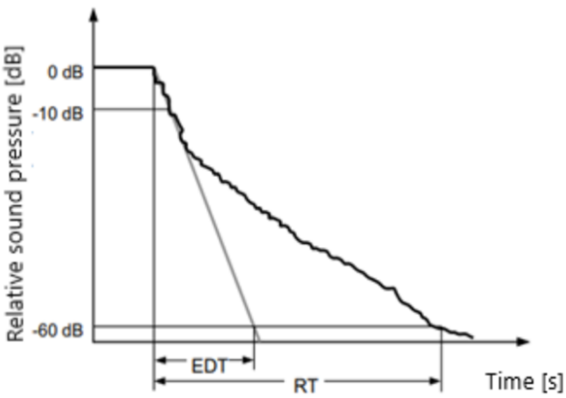


Figure 1: EDT and RT relation.

The EDT is more related to the subjective impression of liveness than RT. This means that, in all those points of a room with an EDT significantly lower than the RT, the room will result, from a subjective point of view, more subdued than would be inferred from the value of RT [1,15].

3.3.2 Clarity (C50 and C80)

The clarity parameters determine the relationship between the initial and final acoustic energy. This can be calculated for 50 ms or for 80 ms depending on whether the results are to be related to speech or music conditions respectively. These parameters give a general impression of speech intelligibility (C50) or the clarity of music (C80). Equation (3) shows the calculation expression determined in ISO 3382-1 [1].

$$C_t = 10log\left(\frac{\int_0^t h^2(t)dt}{\int_t^\infty h^2(t)dt}\right) \tag{3}$$

where *t* is 50 milliseconds or 80 milliseconds for C50 and C80 respectively

3.3.3 Center Time (Ts)

The center time (Ts) is defined by the ISO 3382-1 as the time corresponding to the center of gravity of the squared impulse response [1]. It represents the perceived balance of the room between clarity and reverberation. This parameter can be calculated by the Equation (4).

$$T_S = \left(\frac{\int_0^\infty t * h^2(t)dt}{\int_0^\infty h^2(t)dt}\right) \tag{4}$$

Where h^2 is the squared impulse response.

3.3.4 Inter Aural Cross Correlation (IACC)

Interaural cross correlation is measured binaurally with a dummy head [1]. Subjective studies on auditoriums have shown that the IACC correlates well with the subjective quality of “spatial impression” of a concert hall [16]. Although, as ISO 3382-1 points out, it is not a widely used parameter, binaural measurements are being used more and more over time. The IACC can be obtained from Equation (5).

$$IACF(\tau) = \left(\frac{\int_{t1}^{t2} hl(t)hr(t+\tau)dt}{\sqrt{\int_{t1}^{t2} hl^2(t)dt * \int_{t1}^{t2} hr^2(t)dt}}\right) \tag{6}$$

$$IACC = max|IACF(\tau)|$$

where *hl (t)* and *hr (t)* is the impulse response of the left and right channels respectively, and *t1* and *t2* are the integration time limits.

4. Software Development

This section describes the software at user level, then the signal processing procedure followed for obtaining the IR and the acoustical parameters. The fantasy name IR-ps is chosen for the software, and it stand for Impulse Response Processing Software.

For the development of the software, Python language is chosen. The software consists of two different windows. Figure 2 shows the main window in which there are 3 tabs. In the first tab “SS/IR File” there is first a section that allows the user to enter a recorded sine sweep or a room impulse response file. A file name display is included to indicate the loaded file. Then, the user has to select if the signal is mono or stereo. After that, If the sine sweep option is selected, the user is asked to insert information about the ideal sweep used which includes: start and end frequency (in Hz), duration (in seconds) and whether the sine sweep is linear or logarithmic. This information further use to create the inverse filter in the back-end part of the software. In the second tab, “representation”, the user can choose the way in which the IR will be processed: type of filtering (1/1 or 1/3 octave bands) and smoothing (MMF or Schroeder’s method with No Compensation, or Lundeby or Chu’s compensation for background noise). An status bar is included, which is linked to the calculate button in order to warn the user if something is missing or to inform if the process was done correctly.

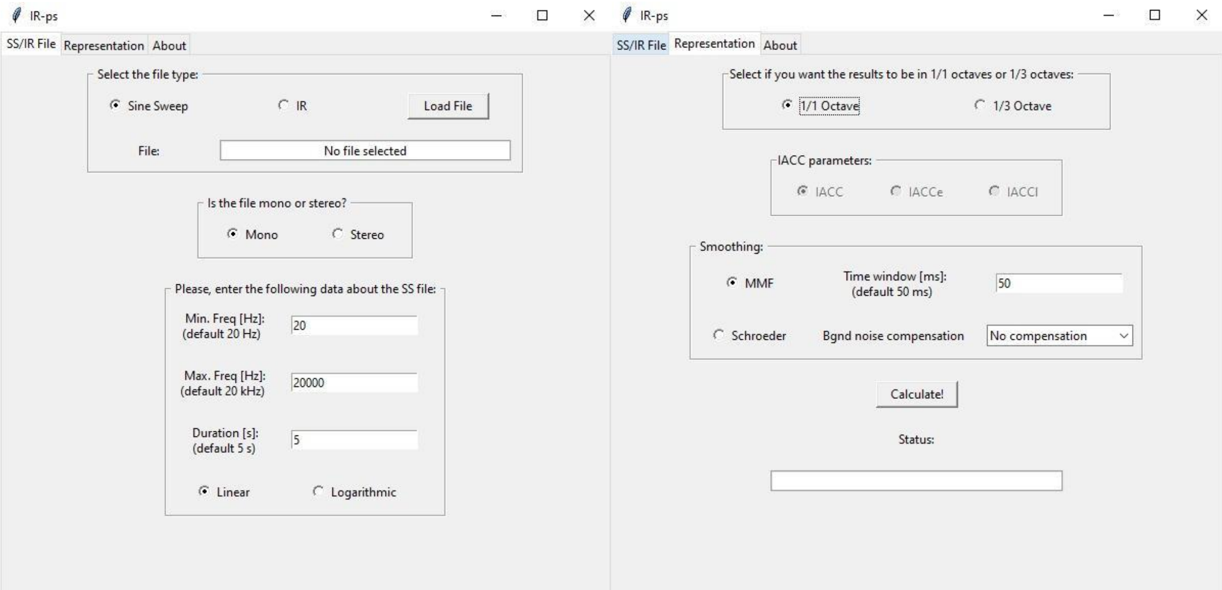
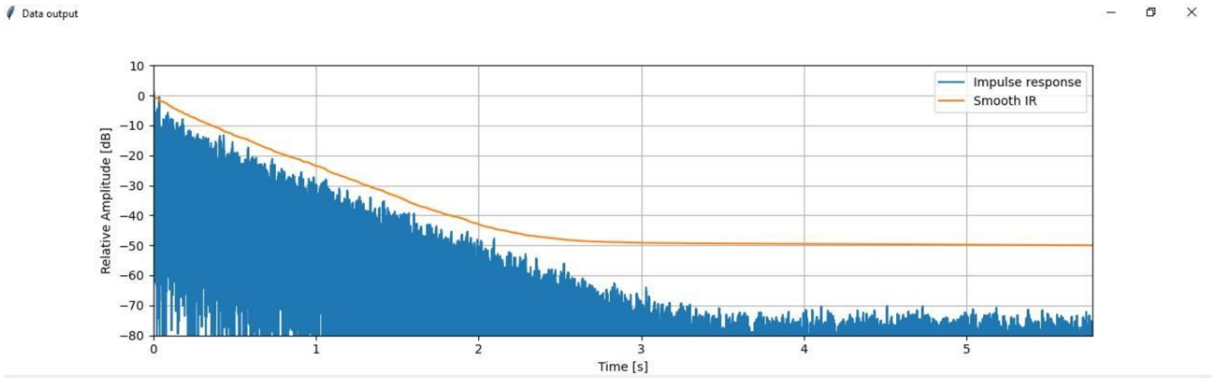


Figure 2: Main window representation. File type selection section (Left). Representation section (right).

If the file chosen is stereophonic, IACC options are available for selection. This option allows the user to choose between IACC, $IACC_e$ and $IACC_l$ option. A third tab, “About”, is included, in which the used can found information about the developers of the software and a brief explanation its use.

As seen in Figure 2, the first and second tab has several entry fields. In order to ease the user experience, and to give a correct example of the way of filling these fields, recommended default values are included.

The second window is reserved to display the results. Figure 3 shows its appearance. This window includes a display of the IR signal for the 1 kHz band, as well as the smoothed signal. Also a table which includes all the parameter displayed in the selected fractional octave band filter is added. At the bottom of the window, the export and return buttons are included. Export option generates an excel file with the table information, saved by a name chosen by the user. The return button goes back to the main window.



Parameter	31.5 [Hz]	63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1 [kHz]	2 [kHz]	4 [kHz]	8 [kHz]	16 [kHz]
EDT [s]	1.08	0.68	0.89	1.63	2.4	2.26	1.7	1.23	0.72	0.27
T20 [s]	1.8	1.23	1.32	2.09	2.89	2.8	2.1	1.51	0.75	0.47
T30 [s]	2.01	1.8	1.62	2.2	3.02	2.86	2.2	1.59	0.83	0.47
C50 [dB]	-1.44	2.11	5.43	2.89	-0.33	-0.23	2.18	4.36	7.79	14.94
C80 [dB]	3.01	8.69	9.99	6.47	2.37	2.8	4.9	7.56	13.57	22.99
EDTt [s]	11.37	12.57	13.57	10.14	8.32	8.35	10.49	13.29	22.63	32.63
Tt [s]	0.55	0.36	0.36	0.58	0.87	0.85	0.6	0.46	0.25	0.14
Ts [ms]	103.55	76.12	75.01	131.14	183.61	177.26	137.88	97.57	52.54	39.43

Export Return

Figure 3: Calculated acoustical parameter window.

4.1 Signal Processing

In order to clarify the used signal processing method, Figure 4 shows the signal flow of the developed program. The input signal can be IR or SS. In the second case, the program generates the inverse filter corresponding to the SS with which the IR will be obtained by convolution. To correctly generate the inverse filter, it is necessary to enter information about the ideal sweep used, as stated in the previous section. On the other hand, in either case, it is necessary to specify whether the loaded file is mono or stereo in order for the to process the arrays corresponding with the audio samples accordingly.

Both types of IRs are filtered by octave or third-octave band (according to the user's selection) through the same procedure, according to the IEC 61260 standard [14]. The filtering is made using a function script not developed by the authors of this paper but for José M. Renquena Plens, available for use in Github via General Public License [17]. If the IR is stereo, each channel is filtered separately as if it were individual audios. To perform the third octave filtering, each signal is first inverted in time to apply the Time Reversed Analysis presented in section 2, then the signal is filtered and, finally, the filter output is inverted again.

Once the IR is filtered in octave fractions, the smoothing process is carried out to obtain the energy-time curve. In this instance, the user must decide the smoothing method to implement, being able to choose between Schroeder's method and Moving Median Filter method (MMF). When selecting MMF, the user must enter the length of the window in milliseconds. 50 ms pre-loaded as default value in order to include 20 Hz as minimum frequency. On the other hand, when selecting the Schroeder method, the user can choose between the different types of background noise compensation: Lundeby or Chu's method, or no compensation at all. Finally, the different parameters can be obtained by implementing the formulas described in section 3.3. For the case of stereo signals, it is possible to calculate the IACC parameters and the mono parameters are calculated from the average between both channels.

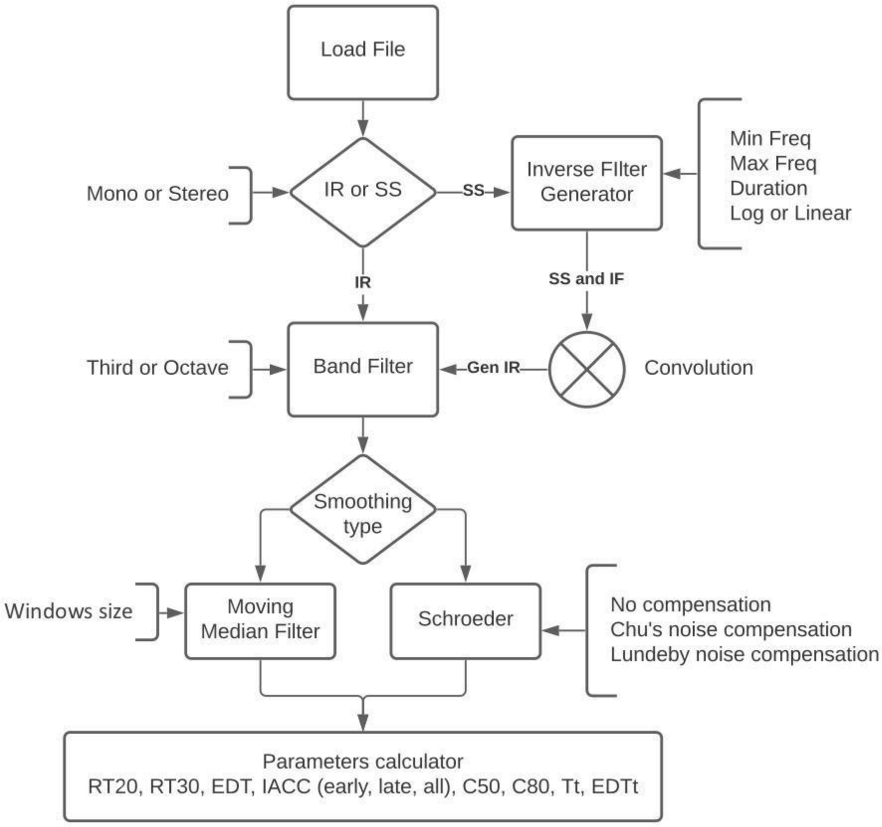


Figure 4: Software signal flow of IR-ps

5. Results

In this section, results of the calculated parameters are shown. In order to assess the consistency and certainty of the different parameters, calculation of the parameters by commercial software are included. As it was previously stated, the commercial software selected for the comparison are Audacity and EASERA. Two software are used in order to double check the results, and since Audacity results alone are inconsistent for certain parameters (as it will be discussed in the future). The evaluation of the results includes the calculations made with both MMF and Schoeder's smoothing methods. For the later No Compensation, Lundeby and Chu's compensation results are shown. All the results in this sections are represented in octave bands. EASERA results are display from 125 to 8000 Hz, since the software gives results in these bands. EDT, T20, T30, C50, C80, and Ts results are shown for a monoaural IR obtained from Terry's Typing Room assessment available in Open Air Lib [18]. IACC parameters are obtained from a binaural impulse response recorded in the Usina del Arte Symphony Hall, also available in Open Air Lib [19].
Considering reverberation related parameters, Figures 5 to 7 show the results for EDT, T20 and T30 parameters respectively.

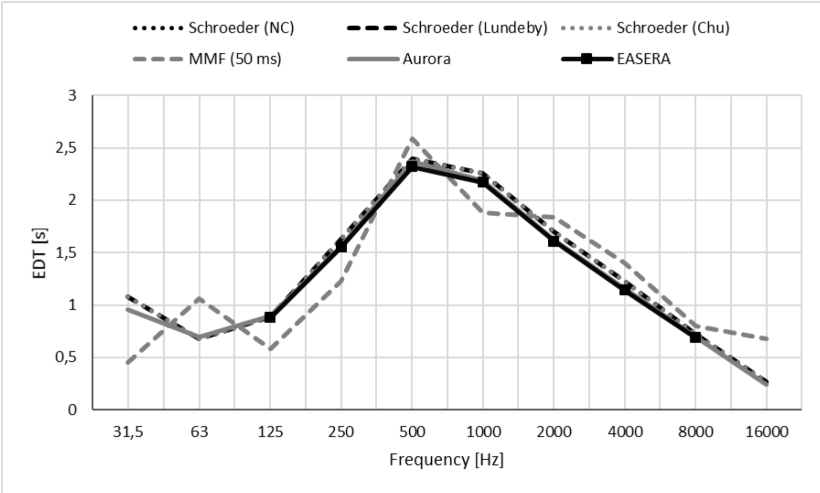


Figure 5: EDT results obtained via IR-ps compared with EASERA and Aurora.

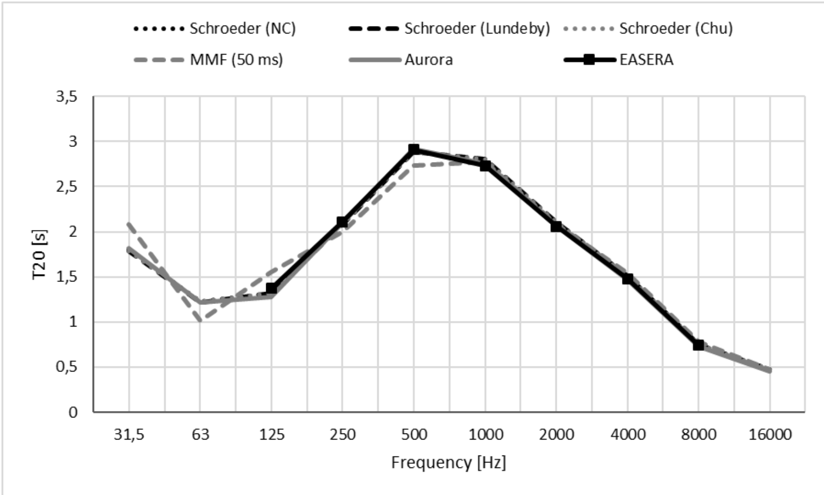


Figure 6: T20 results obtained via IR-ps compared with EASERA and Aurora.

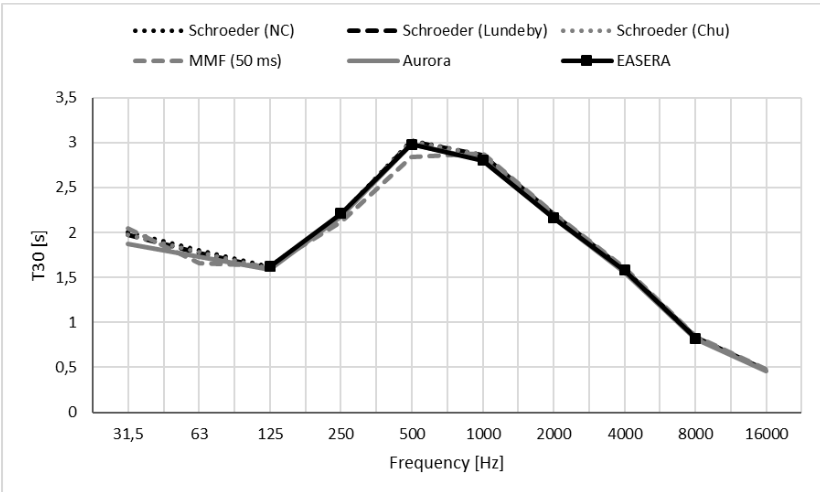


Figure 7: T30 results obtained via IR-ps compared with EASERA and Aurora

Figure 8 and 9 shows the calculations of C50 and C80 parameters respectively. Figure 10 shows the results for Ts parameter.

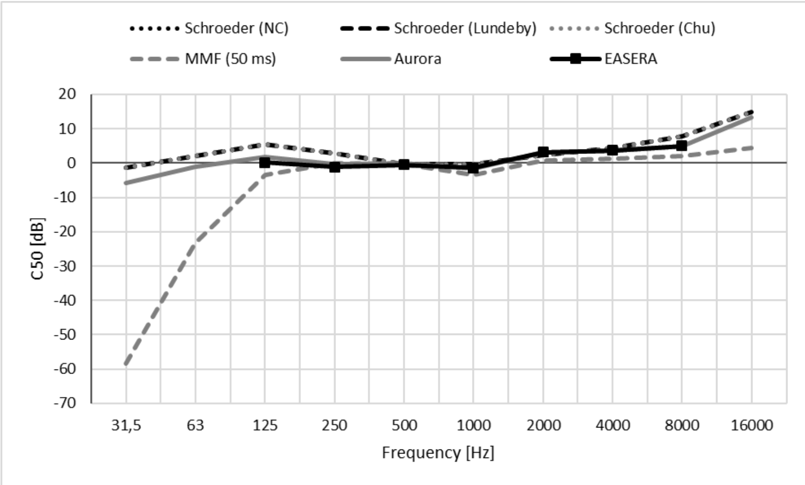


Figure 8: C50 results obtained via IR-ps compared with EASERA and Aurora

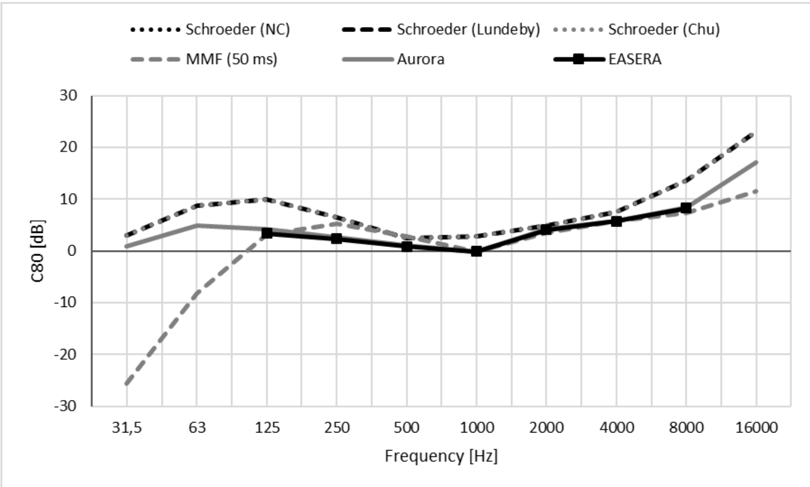


Figure 9: C80 results obtained via IR-ps compared with EASERA and Aurora

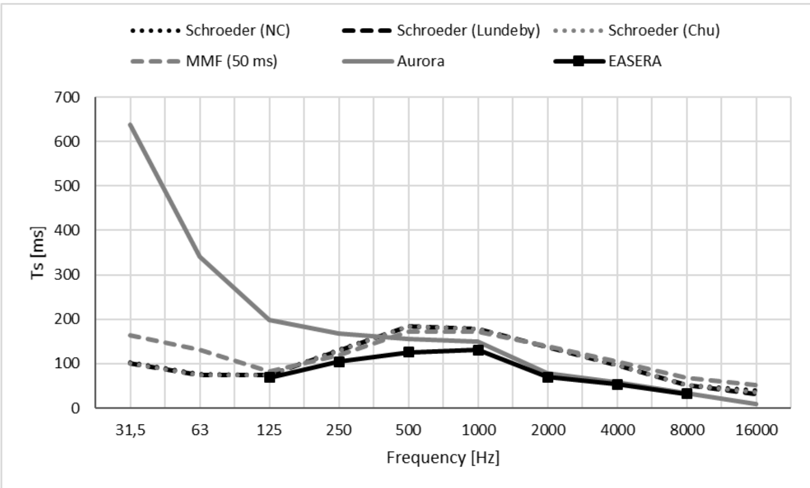


Figure 10: Ts results obtained via IR-ps compared with EASERA and Aurora

As stated previously, IACC parameters are calculated for its assessment from a binaural recording. When processing this recording via Audacity, the results were discarded since the values were greater than 1 (above 20 for some bands). Hence, the results obtained with IR-ps software for IACC are compared only with those calculated with EASERA. Figure 11 shows these results.

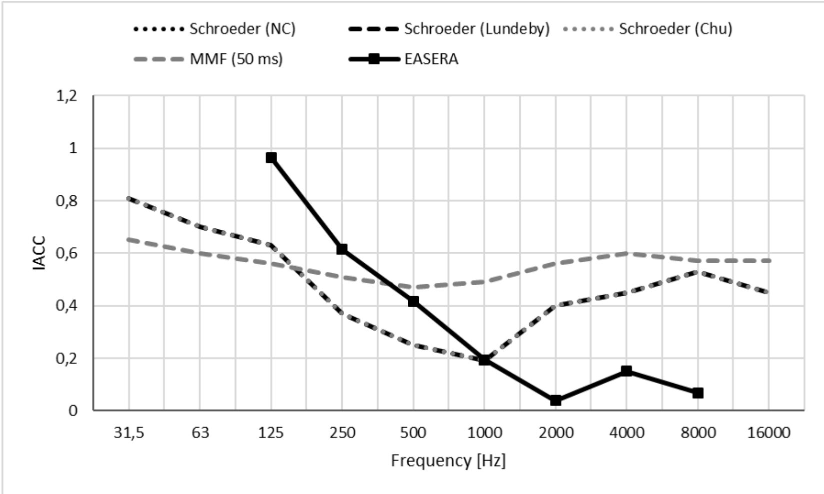


Figure 11: Ts results obtained via IR-ps compared with EASERA and Aurora

The full values for the calculated parameters via IR-ps, Aurora and EASERA are shown in Table A1 of the appendix. This table also includes $IACC_e$, $IACC_l$, T_t and EDT_t. These last two parameters are not shown in this section since neither Aurora nor EASERA has the option for its calculation. Hence, the comparison possibility between IR-ps results and commercial software is not available for these parameters.
It is useful for the assessment of the overall quality of the calculations to present the differences between the results obtained by the developed software with those obtained with a commercial one. For this, T30 and C80 parameters are selected. Then, Aurora's results for these parameters are chosen as reference, and the relative difference between Aurora's and IR-ps' (for all smoothing options) results is calculated. Figure 12 and 13 shows the results of this comparison for T30 and C80 respectively.

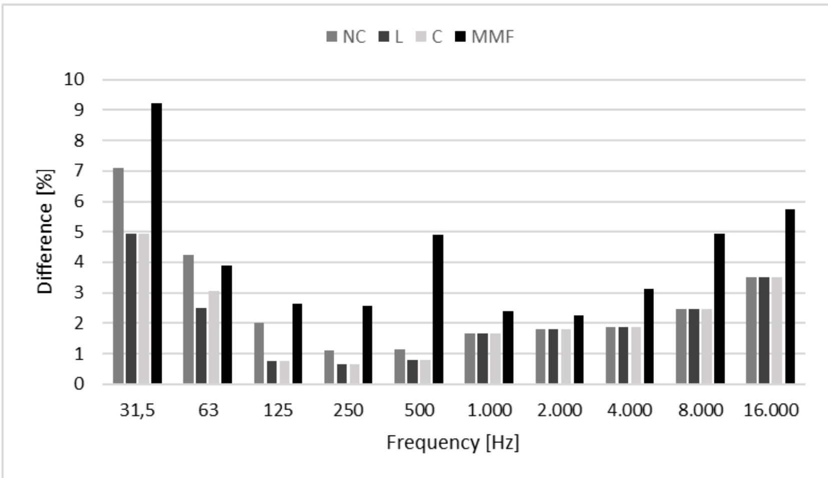


Figure 12: relative difference between T30 values obtained with IR-ps against Audacity's reults expressed as percentage. NC = no compensation, L = Lundeby, C = Chu, MMF = moving median filter.

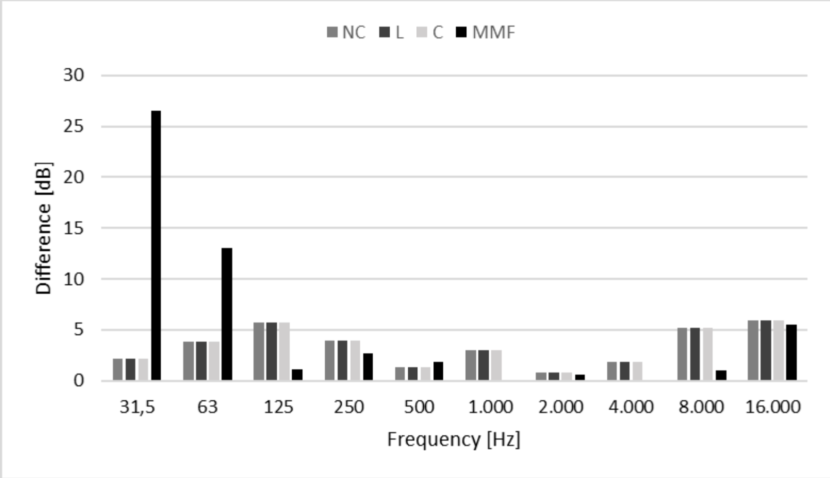


Figure 23: relative difference between C80 values obtained with IR-ps against Audacity's reults in dB. NC = no compensation, L = Lundeby, C = Chu, MMF = moving median filter.

In order to assess the consistency of the results obtained from a sine sweep recording, it is decided to compare the parameters obtained via processing a recorded sine sweep, against those obtained from an IR calculated from the same sine sweep in a commercial software. For this, Aurora module for Audacity is used, since it enables the user to reproduce a sine sweep, and obtain an IR via convolution of the recorded sine sweep with an inverse filter generated by the module. The recorded sine sweep file and the IR file that Aurora generates are processed with the corresponding options in IR-ps and the obtained parameters are compared with parameters obtained by Aurora module of the IR generated in Audacity. Figure 14 shows these process in order to ease the interpretation of the results. The recorded sine sweep corresponds with a home-made recording of a bedroom made in one position only for the source and microphone. In order to protect the source from potential damage, since a commercial monitor (Thonet and Vander 140 w monitor) is used, the sine sweep (logarithmic) is limited from 125 to 16.000 Hz with a length of 20 seconds. It is important to remark that the aim of the recording is not assessing the acoustical parameters per se, but to obtain a recorded sine sweep and a IR.

Figure 15 shows the results of the parameters obtained via processing the recorded sine sweep with IR-ps, along with those obtained from processing the IR (generated via convolution of the recorded SS with its inverse filter in Aurora) in IR-ps as well. The parameters calculated in Aurora are added for comparison. In Figure 15 "parameter IR-ps" corresponds with the results from the recorded sine sweep in IR-ps, "parameter IR Aurora" corresponds with the IR obtained via convolution processed in IR-ps, and "parameter Aurora" are the results from the calculation with Aurora. IR-ps calculations are made with Schoeder's smoothing with no compensation.

Full parameter comparison in made in table A2 of the appendix. Stereo comparisons are not included due to the impossibility of replicate the measurements in stereo format.

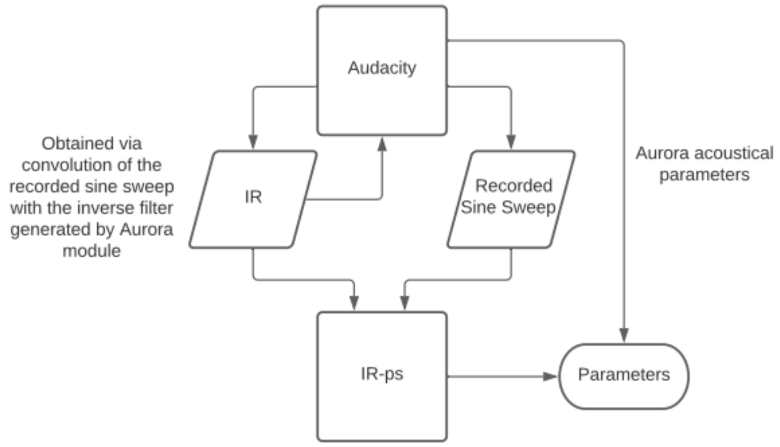


Figure 14: Process carried out for the assessment of the sine sweep results.

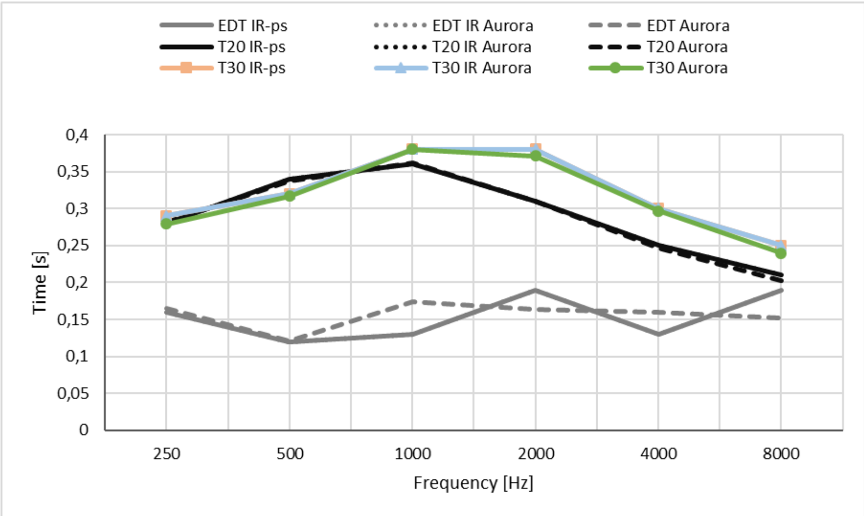


Figure 15: ETD, T20 and T30 calculated for the assessment of sine sweep processing in IR-ps

6. Discussion

From the analysis of the results of the calculated parameters from an IR with IR-ps in comparison with those obtained via commercial software, it can be seen that all parameters except IACC, follows the same tendency. Even more, the appreciable differences between commercial and non-commercial results is considerably low for the majority of the parameters. For reverberation related results, the two smoothing methods, and all three compensations for Schroeder's method, give similar results, being MMF the smoothing method with greater differences in EDT parameter, although the tendency is similar. Clarity related parameters results are sparser but maintain the general tendency, except for MMF smoothing method in lower frequencies. Figures 12 and 13 allow to quantify the difference for T30 and C80 parameters. For T30, the difference overall difference is less than 10% for all band, being less than 3% for central frequencies. In this case, MMF smoothing method presents larger differences for all bands. C80 parameter shows general differences between 2 and 6 dB, except for MMF in the lower bands. These discrepancies could be explained by taking into

account that MMF smoothing method is more primitive than Schroeder's method. The time window could be another factor even though 50 ms was used for all calculations.
Central Time parameter presents differences for the results in EASERA and Audacity software. The explanation for this differences is out of reach since the calculation method of these software is unknown. The results here presented are similar to EASERA's, and again for lower frequencies MMF smoothing method shows greater differences.
Inter Aural Cross Correlation is the parameter which results are more different than those calculated by commercial software. In this case, only EASERA's parameters are for comparison as Aurora's result values are not reasonable (larger than 1). For this parameter, neither the values nor the general tendency matches the reference. Even more, differences between both smoothing methods are observed. Yet the calculation algorithm is reviewed, the source of difference is not found. At last, the fact that Aurora's values are odd and different of those calculated with EASERA, may explain the difficulty of obtaining this parameter.
Last, EDT_t and T_t results cannot be compared with a commercial software since neither EASERA nor Aurora has the option for its calculation. This comparison and the validation of these parameters is impossible without this possibility.
Analyzing the results of the calculations made from a sine sweep, it can be seen from Figure 15 that both the IR obtained from Aurora and the IR obtained in IR-ps give the same results, and the values of these parameters compared with those calculated with Aurora follows the same general tendency. This results helps to validate the process of calculation of the IR carried out by IR-ps, from a recorded sine sweep via convolution of this signal with its inverse filter.

7. Conclusion

In this paper, the development process of an open source code for the calculation of EDT, T20, T30, C50, C80, Ts, Tt, EDTt and IACC acoustical parameters from a recorded impulse response or a sine sweep, and the assessment of the results via comparison with commercial software is described. From the obtained results, it can be said that the main objective is achieved successfully, even though the are some point to discuss.
In general basis the results are consistent and follows the general tendency of those calculated by commercial options, being this fact more marked for reverberation and clarity related parameters. MMF smoothing, especially in low frequencies, shows greater differences with the expected values. For stereo calculations, IACC parameters also shows discrepancies with the results from commercial options. Last, no comparison, hence no validation, can be done for EDT_t or T_t parameters, since its calculations are not available in commercial software.
In general, as IR-ps is a first version, several improvements can be made in the code in order to optimize calculation times, for example for MMF smoothing. Also, even though it is far beyond the scope of this works, a full quality assessment from the user's perspective can be made in order to improve and protect the software from potential misuse from the user. Also, other functionalities as real time play and record of the sine sweep, or custom export options can be added in future versions in order to make the software more complete. As IR-ps is an open source software, based in an ever growing language as it is Python, further improvements can be made for any person with the knowledge or interest.

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Appendix

Table 1: Full values of the acoustical parameters calculated with IR-ps, Aurora and Audacity presented in this paper

Parameter	Source	31.5 [Hz]	63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1.000 [Hz]	2.000 [Hz]	4.000 [Hz]	8.000 [Hz]	16.000 [Hz]
EDT [s]	S_NC	1,08	0,68	0,89	1,63	2,4	2,26	1,7	1,23	0,72	0,27
	S_L	1,08	0,68	0,89	1,63	2,4	2,26	1,7	1,23	0,72	0,27
	S_C	1,08	0,68	0,89	1,63	2,4	2,26	1,7	1,23	0,72	0,27
	MMF	0,45	1,06	0,58	1,23	2,59	1,88	1,84	1,4	0,8	0,68
	Aurora	0,961	0,699	0,895	1,561	2,369	2,187	1,616	1,158	0,695	0,242
	EASERA	-	-	0,88	1,55	2,32	2,17	1,61	1,14	0,69	-
T20 [s]	S_NC	1,8	1,23	1,32	2,09	2,89	2,8	2,1	1,51	0,75	0,47
	S_L	1,79	1,22	1,32	2,08	2,89	2,8	2,1	1,51	0,75	0,47
	S_C	1,79	1,23	1,32	2,08	2,89	2,8	2,1	1,51	0,75	0,47
	MMF	2,08	1,01	1,55	2	2,73	2,77	2,08	1,54	0,78	0,47
	Aurora	1,816	1,217	1,273	2,105	2,917	2,758	2,062	1,473	0,731	0,457
	EASERA	-	-	1,37	2,11	2,91	2,73	2,06	1,48	0,74	-
T30 [s]	S_NC	2,01	1,8	1,62	2,2	3,02	2,86	2,2	1,59	0,83	0,47
	S_L	1,97	1,77	1,6	2,19	3,01	2,86	2,2	1,59	0,83	0,47
	S_C	1,97	1,78	1,6	2,19	3,01	2,86	2,2	1,59	0,83	0,47
	MMF	2,05	1,66	1,63	2,12	2,84	2,88	2,21	1,61	0,85	0,48
	Aurora	1,877	1,727	1,588	2,176	2,986	2,813	2,161	1,561	0,81	0,454
	EASERA	-	-	1,62	2,21	2,98	2,8	2,16	1,58	0,82	-
C50 [dB]	S_NC	-1,44	2,11	5,43	2,89	-0,33	-0,23	2,18	4,36	7,79	14,94
	S_L	-1,44	2,11	5,43	2,9	-0,33	-0,23	2,18	4,36	7,79	14,95
	S_C	-1,44	2,11	5,43	2,9	-0,33	-0,23	2,18	4,36	7,79	14,94
	MMF	-58,39	-23,39	-3,5	-0,01	-0,39	-3,57	0,76	1,31	2,1	4,52
	Aurora	-5,894	-1,017	1,879	-0,251	0,012	-1,541	3,216	3,627	4,972	13,295
	EASERA	-	-	0,2	-1,1	-0,5	-1,4	3,1	3,7	5	-

Parameter	Source	31.5 [Hz]	63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1.000 [Hz]	2.000 [Hz]	4.000 [Hz]	8.000 [Hz]	16.000 [Hz]
C80 [dB]	S_NC	3,01	8,69	9,99	6,47	2,37	2,8	4,9	7,56	13,57	22,99
	S_L	3,01	8,69	9,99	6,47	2,37	2,8	4,9	7,56	13,57	23,01
	S_C	3,01	8,69	9,99	6,47	2,37	2,8	4,9	7,56	13,57	23
	MMF	-25,68	-8,16	3,16	5,21	2,88	-0,1	3,56	5,77	7,41	11,58
	Aurora	0,809	4,834	4,275	2,578	1,002	-0,16	4,141	5,685	8,408	17,087
	EASERA	-	-	3,4	2,3	0,8	-0,1	4,1	5,7	8,3	-
Ts [ms]	S_NC	103,55	76,12	75,01	131,14	183,61	177,26	137,88	97,57	52,54	39,43
	S_L	101,25	74,24	73,4	130,05	183,33	177,15	137,87	97,55	52,35	30,77
	S_C	101,56	74,93	73,39	129,99	183,29	177,14	137,87	97,55	52,35	35,76
	MMF	164,3	131,56	82,85	119,2	172,98	171,99	139	105,27	69,04	51,95
	Aurora	637,531	340,436	197,539	168,892	155,089	149,586	77,624	57,633	33,638	9,397
	EASERA	-	-	68,31	104,09	125,39	131,05	70,01	53,36	31,94	-
Tt [s]	S_NC	0,55	0,36	0,36	0,58	0,87	0,85	0,6	0,46	0,25	0,14
	S_L	0,55	0,36	0,36	0,58	0,87	0,85	0,6	0,46	0,25	0,14
	S_C	0,55	0,36	0,36	0,58	0,87	0,85	0,6	0,46	0,25	0,14
	MMF	0,55	0,36	0,36	0,58	0,87	0,85	0,6	0,46	0,25	0,14
EDTt [s]	S_NC	11,37	12,57	13,57	10,14	8,32	8,35	10,49	13,29	22,63	32,63
	S_L	10,86	12,89	13,73	9,84	8,17	8,28	10,49	13,3	22,93	45,38
	S_C	10,9	12,33	13,65	9,89	8,18	8,29	10,49	13,3	22,91	34,88
	MMF	12,93	13,27	14,01	10,21	8,34	8,06	9,67	11,83	18,19	27,49
IACC	S_NC	0,81	0,7	0,63	0,37	0,25	0,19	0,4	0,45	0,53	0,45
	S_L	0,81	0,7	0,63	0,37	0,25	0,19	0,4	0,45	0,53	0,45
	S_C	0,81	0,7	0,63	0,37	0,25	0,19	0,4	0,45	0,53	0,45
	MMF	0,65	0,6	0,56	0,51	0,47	0,49	0,56	0,6	0,57	0,57
	EASERA	-	-	0,964	0,614	0,417	0,193	0,038	0,151	0,069	-

Parameter	Source	31.5 [Hz]	63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1.000 [Hz]	2.000 [Hz]	4.000 [Hz]	8.000 [Hz]	16.000 [Hz]
IACCe	S_NC	0,94	0,81	0,72	0,55	0,39	0,32	0,62	0,66	0,71	0,48
	S_L	0,94	0,81	0,72	0,55	0,39	0,32	0,62	0,66	0,71	0,48
	S_C	0,94	0,81	0,72	0,55	0,39	0,32	0,62	0,66	0,71	0,48
	MMF	0,64	0,76	0,64	0,68	0,63	0,75	0,85	0,85	0,77	0,68
	EASERA	-	-	0,975	0,614	0,509	0,266	0,086	0,222	0,088	-
IACCl	S_NC	0,21	0,06	0,02	0,04	0,04	0,07	0,18	0,19	0,2	0,02
	S_L	0,21	0,06	0,02	0,04	0,04	0,07	0,18	0,19	0,2	0,02
	S_C	0,21	0,06	0,02	0,04	0,04	0,07	0,18	0,19	0,2	0,02
	MMF	0,65	0,28	0,03	0,06	0,06	0,15	0,33	0,37	0,38	0,13
	EASERA	-	-	0,927	0,657	0,228	0,118	0,078	0,076	0,107	-

Table 2: Full parameter comparison of the results of processing the recorded SS and the IR generated in Aurora with IR-ps, against Aurora results

Parameter	Source	250 [Hz]	500 [Hz]	1.000 [Hz]	2.000 [Hz]	4.000 [Hz]	8.000 [Hz]
EDT [s]	IR-ps	0,16	0,12	0,13	0,19	0,13	0,19
	IR Aurora	0,16	0,12	0,13	0,19	0,13	0,19
	Aurora	0,165	0,121	0,174	0,164	0,16	0,152
T20 [s]	IR-ps	0,28	0,34	0,36	0,31	0,25	0,21
	IR Aurora	0,28	0,34	0,36	0,31	0,25	0,21
	Aurora	0,282	0,337	0,362	0,31	0,246	0,202
T30 [s]	IR-ps	0,29	0,32	0,38	0,38	0,3	0,25
	IR Aurora	0,29	0,32	0,38	0,38	0,3	0,25
	Aurora	0,279	0,317	0,38	0,371	0,297	0,24
C50 [dB]	IR-ps	27,18	28,55	22,67	26,3	28,94	31,26
	IR Aurora	27,18	28,55	22,67	26,3	28,94	31,26
	Aurora	13,456	17,147	12,69	16,814	17,783	20,535

Parameter	Source	250 [Hz]	500 [Hz]	1.000 [Hz]	2.000 [Hz]	4.000 [Hz]	8.000 [Hz]
C80 [dB]	IR-ps	39,31	36,84	33,58	35,63	41,09	44,16
	IR Aurora	39,32	36,84	33,58	35,63	41,08	44,14
	Aurora	19,359	20,712	18,975	21,901	24,588	27,599
Ts [ms]	IR-ps	19,49	15,63	21,32	27,85	20,16	16,66
	IR Aurora	19,49	15,63	21,32	27,85	20,16	16,66
	Aurora	84,644	44,28	27,621	14,837	10,486	6,483
Tt [s]	IR-ps	0,08	0,08	0,1	0,11	0,08	0,07
	IR Aurora	0,08	0,08	0,1	0,11	0,08	0,07
EDTt [s]	IR-ps	66,46	158,07	135,29	152,38	176,95	209,45
	IR Aurora	8,93	9,98	8,51	9,97	11,35	12,08