University of Edinburgh

Acoustics Project Report
Impulse Response Measurement and Analysis of
Reid Concert Hall

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Contents

1 Introduction and Background

An impulse response measurement can tell us room's acoustic parameters. By analyzing the acoustic parameters, reasonable rooms acoustic design and optimization can be applied in different venues. The aim in this report is to measure impulse response measurement and analyze acoustics impact of Reid Concert Hall.

1.1 Room Impulse Responses

The room impulse response is the transfer function between the sound source and the microphone. The sound emitted at the source point propagates in all possible directions and is reflected at the boundary of the room (which is shown in the figure 1). The impulse response at the receiver point consists of the direct sound reaching the receiver and the subsequent reflection. As a result, the room impulse response measurement is unique to the location of the source and receiver. To recover the original sound source, the received microphone signal can be convolved with the inverse of the room impulse response function.

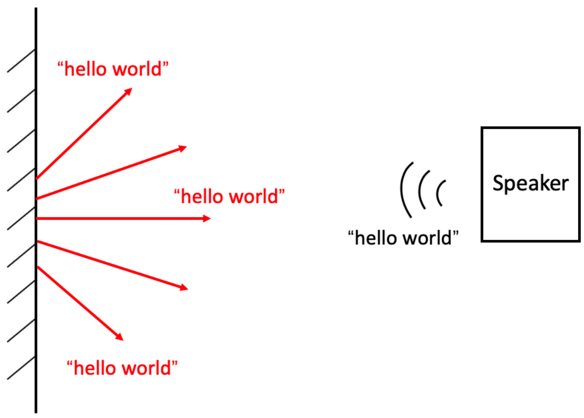

Figure 1: The propagation of sound sources

The rooms of the closed environment can regard as a linear [1], time-invariant[2] system (LTI system), and the entirety of an LTI system can be described by a single function called its impulse response. This function exists in the time domain of the system . For any input, the output of the LTI system is the convolution of the input signal and the room's impulse response. The relationship between input and output can be expressed as

$$y(t) = h(t) * x(t) \triangleq \int_{-\infty}^{\infty} h(\tau)x(t-\tau)d\tau \tag{1}$$

Where the impulse response is h(t), input is $x(t)$ and output is $y(t)$.

1.2 Room Acoustic Parameters

There are many meaningful room acoustic parameters are derived from room impulse responses [1].

Examples of room acoustic parameters include:

- Sound Intensity - which is related to sound levels.
- The reverberation time - which is related to the energy decay rate and the clarity.
- Definition and Centre Time - which are related to early to late energy ratios of the impulse response.

Reverberation time
Reverberation is a collection of sounds reflected from the surface in a closed space like a concert hall or a classroom. Reverberation time (RT)1.2 is the number of seconds it takes for the reverberation energy to decay to one

[1] A linear system is a system in which the outputs of a linear combination of its inputs are the same as a linear combination of the individual responses to these inputs.
[2] Time-invariant systems in which the output is not dependent on the input in the system when applied.

millionth (or 60dB) of its original value from the moment the sound signal stops (which was defined 100 years ago by W. C. Sabine. T). For example, if it takes 10 seconds for the sound in a room to decay from 100dB to 40dB, the reverberation time will be 10 seconds. This could also be written as RT_{60} time. The Reverberation time RT_{60} can be calculate by the Sabine equation [2]:

$$RT_{60} = \frac{24(In10)V}{C_{20}S_a} \tag{2}$$

Where T_{60} is the reverberation time (to drop 60 dB); V is the volume of the room; C_{20} is the speed of sound at $20degree$ (room temperature); S_a is the total absorption in Sabins.

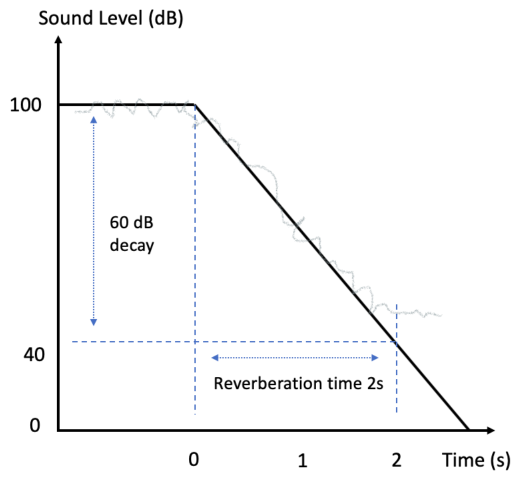

Figure 2: The definition of reverberation time

Measuring T_{60} time accurately is often difficult because it may not produce enough consistent and stable sound levels, especially in larger rooms or Spaces [3]. To solve this problem, it is more common to measure T_{20} and T_{30} times, and then multiply them by 3 and 2 respectively to get the total T_{60} time. The common equations for the calculation of T_{20} and T_{30} both are as following:

$$T = 10lg\frac{\int_t^{\infty} h(t)^2 dt}{\int_0^{\infty} h(t)^2 dt} \quad [dB] \tag{3}$$

Where T_{20} is defined between -5dB and -25dB and T_{30} is defined between -5dB and -35dB.

Early decay time
Early decay time (EDT) has turned out to be better correlated with the reverberance perceived during running speech and music [4]. EDT can be obtained from reverberation time decay curve; A short EDT is a good indication of speech clarity, as the combination of early reflections that reach the listener within 50 milliseconds with direct sound improves speech clarity [3].

Clarity
Clarity describes the degree to which every detail of the performance can be felt. The clarity factor for speech (often referenced as C_{50} is the ratio of sound power present within the first 50 ms of the impulse response, when early reflections occur, to the sound power present thereafter [3].
The equation for the calculation of C_{50} is as following [1]:

$$C_{50} = 10lg\frac{\int_0^{50ms} h(t)^2 dt}{\int_{80ms}^{\infty} h(t)^2 dt} \quad [ms] \tag{4}$$

2 The Studied Space

Reid Concert Hall, a 218-seat small music venue, built in 1859. It is located in the city of Edinburgh, Scotland, in the United Kingdom . The hall is used for lectures and rehearsals by the orchestra and chorus. Many concerts and recitals are often held here.

2.1 Interior Appearance

Figure 3: Interior appearance of the concert hall

Main concert hall space with outstanding classical decorative scheme with coffered semi-elliptical tunnel vault ceiling, grand dentilled cornice, cornice bands to mid height and plain pilasters between windows. Raked floor to west and piano lift to basement [5].

2.2 Motivation

In this project, we aim to explore architectural and environment acoustics of the interior of Reid Concert Hall by measuring the room impulse response of Reid Hall and analysing impacts of the room acoustic parameters.

The Reid concert hall was designed as a 'dry acoustic' with little reverberation and the 1978 organ was designed to complement the building to create a fine acoustic clarity [5]. According to these characteristics (little reverberation and fine acoustic clarity), we decided to focus on the reverberation and clarity both factors and want to verify the performance of these characteristics in Reid concert hall.

3 Technology and Apparatus of Measurement

For the measurement processing of IR, the input signal are generated using a computer, the response to the scan is recorded with a microphone and deconvolved to obtain the system impulse response.

3.1 Sine Sweep Technology

In order to measure the impulse responses, several methods can be used such as MLS (Maximum Length Sequence), IRS (Inverse Repeated Sequence), Time-Stretched Pulses and Sine Sweep [6]. The aimed space for this project is Reid concert hall. In quiet environments the sine sweep method seems to be the most appropriate [7], so we chose sine sweep method to scan the input signal.

Sine sweep is a sinusoidal function that changes frequency over time (the equation is shown as following). We play the sinusoidal sweep in the room, record the result and then deconvolve the result by reversing the sinusoidal sweep in time to get the reverberant pulse.

$$x(t) = sin(\frac{2\pi f_1 T}{In(\frac{f_1}{f_2})}\left[exp\frac{In(\frac{f_2}{f_1})t}{T} - 1\right]) \tag{5}$$

Where f_1 is the starting frequency, f_2 the endind frequency, and T the duration of the chirp.

The input signal is a 20-second and exponential sine sweep that scans audible frequencies from 20Hz to 20000Hz at a constant amplitude. With sine sweeps, it is common to add a segment of silence after each signal, for avoiding the time aliasing problem [7]. So a sufficient duration 10s is added at the end of the Sine Sweep signal in this project.

PARAMETERS	DEFAULT
INITIAL FREQUENCY	20HZ
FINAL FREQUENCY	20000HZ
SSWEEP DURATION	20S
DURATION OF THE SILENCES INSERTED AFTER EACH SWEEP	10S

Table 1: Sine Sweep Parameter Setting in the Project

3.2 Measurement Theory of the Impulse Response

The input signal was emitted spatially separated by speaker and recorded by a microphone. According to the equation 1 (which is mentioned in the chapter 1), by convolving the input signal x(t) with the impulse response h(t), we can get the output signal y(t). In order to convert sine sweep responses y(t) to impulse responses h(t), we assume there is a proper inverse filter f(t), which can transform input signal x(t) into a delayed Dirac's delta function (t) by convolving. The generated method of the Inverse Filter f(t) is illustrated detailly in the paper [7].

$$x(t) \otimes f(t) = \delta(t - K) \tag{6}$$

Deconvolution of the impulse response is then achieved by linearly convolving the output y (t) under test with this inverse filter f (t):

$$y(t) \otimes f(t) = h(t) \tag{7}$$

3.3 Software and Equipment of Measurement

Software for measuring impulse responses: Audacity 2.0

- a free, open source digital audio editor and recording application.
- using for calculating inverse filter and recording of sweep response.

Sound source: Genelec 1029A Speaker

- Frequency Response: 70 to 18000 Hz
- Signal-To-Noise Ratio: 90 dB
- Output Level (SPL): 100 dB
- Audio Amplifier: integrated
- The sound source used for impulse response measurement should be as close to omnidirectional, and this speaker is non-omnidirectional. We replace omnidirectional sound source by adjusting the angle of the speaker

Receiver: AKG - C414 XLS

- Omnidirectional polar pattern microphones
- Audio Frequency Bandwidth: 20 to 20000 Hz
- Sensitivity: 23 mV/Pa
- Equivalent Noise Level: 6 dB-A
- Signal to Noise: 88 dB-A

Optimus Sound Level Meter: 162C Optimus Red sound level meter

Field Recorder: Zoom - F8 - 6871

The following is the schematic diagram of the measurement system in this project.

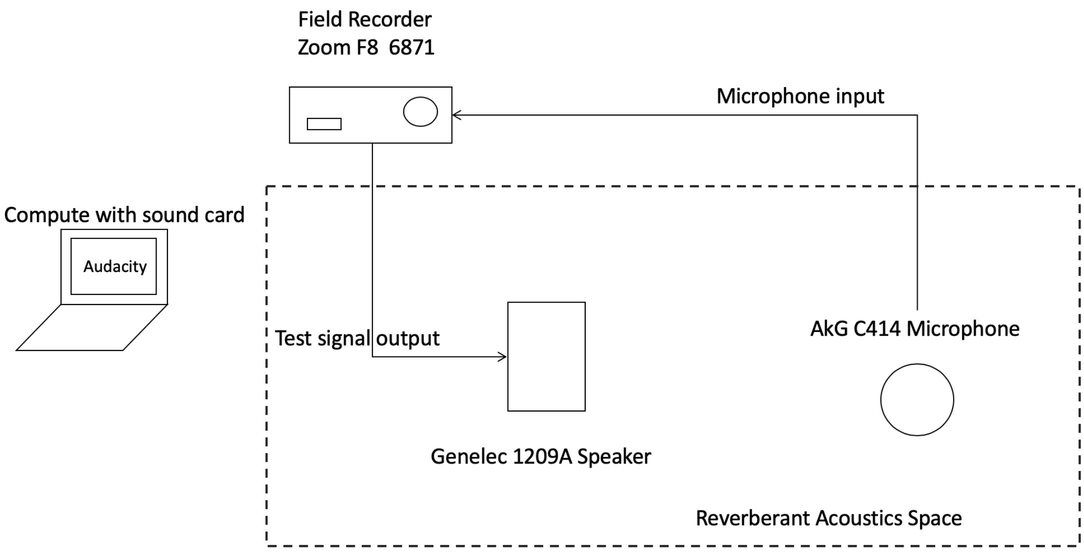

Figure 4: Schematic diagram of the measurement system

3.4 Implementation and Experimental Setup

We set totally 16 experiments of measurement of impulse response by changing the position of receiver and sound source with different angles, the detail experiment information is in the following table 2. Aftering experiments, we got the 16 sweep responses y(t). In order to obtain a more reasonable omnidirectional source signal (limitation of non-omnidirectional sound source), we averaged the sweep response with different angle of source sweep responses. After deconvolving between the recorded sweep response and the original sweep by the Aurora Sweep Convolver Interface in Audacity 2.0, we finally outputted the 6 impulse responses which are illustrated in the table 3.

Experiment Index	Angle of the Sound Source	Position Figure
1	Facing to stage	Figure 5
2	Facing to left of hall	Figure 5
3	Facing to seats	Figure 5
4	Facing to right of hall	Figure 5
5	Facing to stage	Figure 6
6	Facing to seats	Figure 6
7	Facing to stage	Figure 7
8	Facing to seats	Figure 7
9	Facing to stage	Figure 8
10	Facing to left of hall	Figure 8
11	Facing to seats	Figure 8
12	Facing to right of hall	Figure 8
13	Facing to stage	Figure 9
14	Facing to left of hall	Figure 9
15	Facing to seats	Figure 9
16	Facing to right of hall	Figure 9

Table 2: Experiment group information for measurement of impulse responses

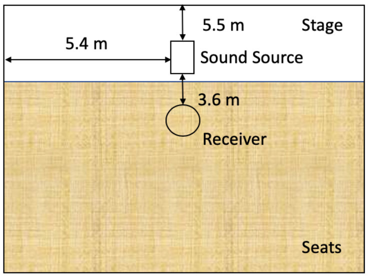

Figure 5: Sound source in the stage center and receiver in the front of seats area

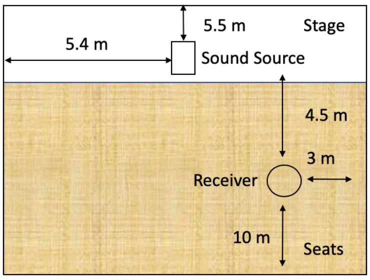

Figure 6: Sound source in the stage center and receiver in the right of seats area

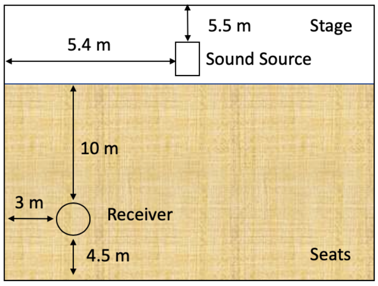

Figure 7: Sound source in the stage center and receiver in the left of seats area

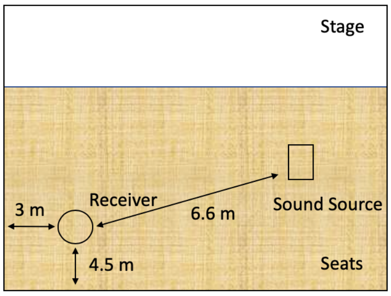

Figure 8: Sound source in the right of seats area and receiver in left bottom of seats area

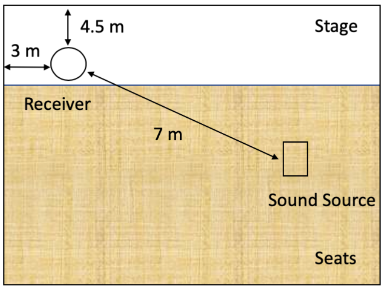

Figure 9: Sound source in the right of seats area and receiver in left top of stage area

IR experiments	Average	Position Figure
1	1,2,3,4	figure 5
2	1,3	figure 5
3	5,6	figure 6
4	7,8	figure 7
5	9,10,11,12	figure 8
6	13,14,15,16	figure 9

Table 3: The 8 impulse responses with summing sweep response with different angle of source sweep

3.5 Data Processing Software

In this project, we mainly use both two tools (Custom-Built Matlab Script and Aurora Acoustical Parameters Modules) together for data processing and analysis. Both of them can be used to calculate the T_{30}, T_{20} and C_{50} and so on. The drawback of Aurora module is that we can not control over the octave-band analysis i.e. on the filters used, or on the error of the linear fits but it is easier to operate than matlab script. In this case we can calculate the analyzed values by Aurora Modules and use matlab script to evaluate and delete the unreliable values.

4 Results and Analysis

4.1 The Results after Data Processing

The ISO standard does not recommend analysis of the original impulse response, but instead recommends octave bands [8]. The audible frequency range can be separated into unequal segments called octaves. The octave band provides a filtering method that divides the audible spectrum into smaller segments of octaves, allowing you to identify different noise levels at each frequency. The band center frequencies we used are (in Hz): [31.6, 63.1, 125.9, 251.2, 501.2, 1000, 1995.3, 3981.1, 7943.3].

The model we explored is a reverberant tail of sounds in a room. This kind of model does not include the direct sound or early reflections and noise floor [9] and according to Sabine's theory, the decay curve should be exponential and is should be a almost straight line on a logarithmic graph. Thus, we highly simplified the model and only focused on reverberant tail. Example about Decay curves obtained is shown in the figure 4.1.

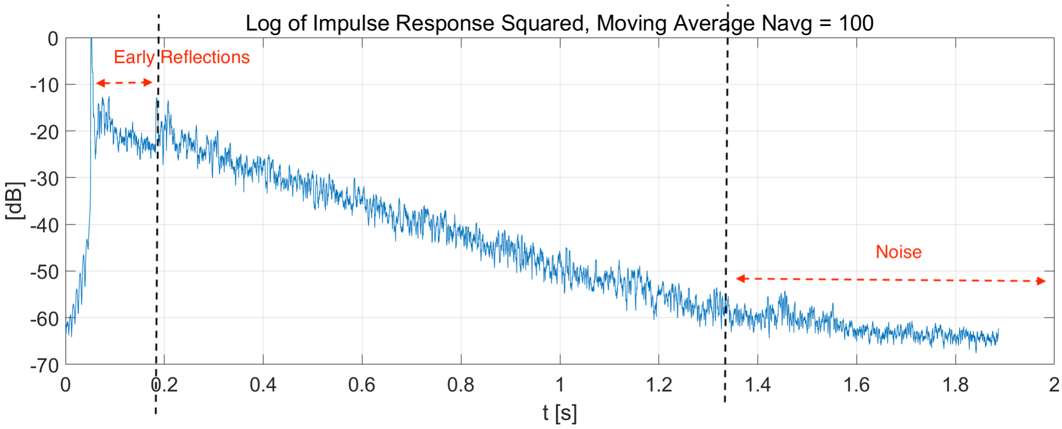

Figure 10: Decay curves obtained after cutting the parts of early reflection and noise for IR group 1

Figure 4.1 illustrates the octave-band-filtered decay curves obtained from different sweep frequency for IR experiment 1. It is apparent that the results for T_{30} are unreliable in the the first three octave band figures. We delete these these values and the remaining values are used to calculate the T_{30}, T_{20}, C_{50} and EDT separately. The same operations were applied to the remaining IR experiments to calculate the corresponding acoustics parameters in the table 4.

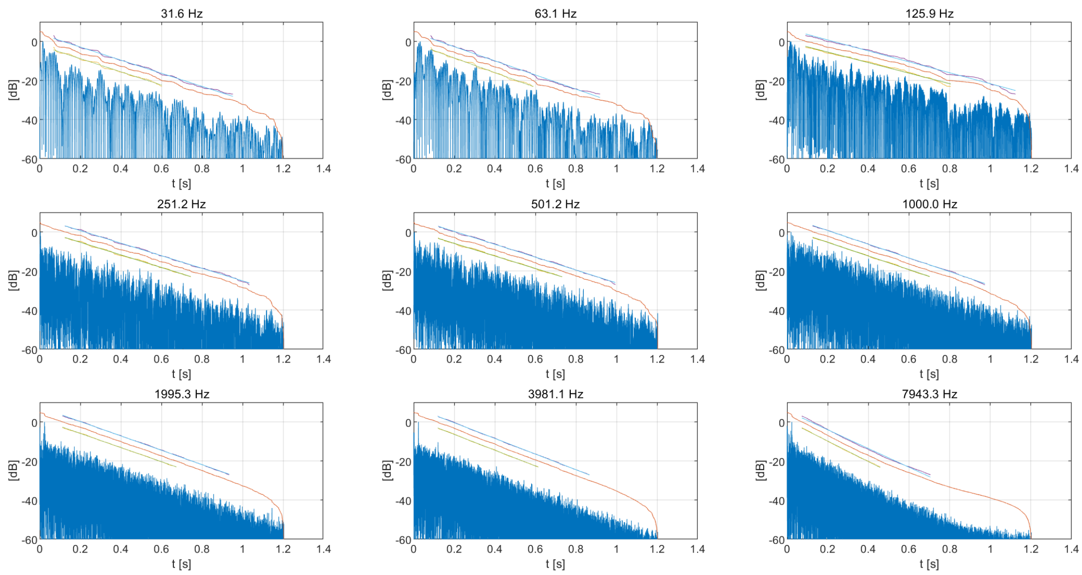

Figure 11: The octave-band-filtered decay curves for IR experiment 1; blue and green lines are fit T_{30} and T_{20} respectively; orange and purple lines are schroeder integral T_{30} and T_{20} respectively; red line is schroeder integral.

Parameters	31.6	63.1	125.9	251.2	501.2	1000	1995.3	3981.1	7943.3
C50[dB]	8.53	0.35	0.91	-0.44	-1.84	-2.20	-0.292	-0.85	1.72
EDT(s)	0.50	0.96	2.08	1.88	1.54	1.70	1.72	1.52	0.95
T_{20}	1.83	1.46	2.16	1.76	1.83	1.68	1.63	1.47	0.97
T_{30}	-	-	-	1.70	1.72	1.64	1.57	1.45	0.98

Table 4: The 8 impulse responses with summing sweep response with different angle of source sweep; the symbol '-' represents unreliable value.

4.2 Evaluation of Reverberation Time

Figures 4.2 4.2 4.2 4.2 show the results of parameters for three different receiving positions. Finally, the octave band results for each location are averaged (after deleting the unreliable values according to results of the matlab scripts analysis) to obtain more global indices of room reverberation. The average value obtained is shown in table 5.

Figure 12: T_{20} of three different receiving positions for each octave band.

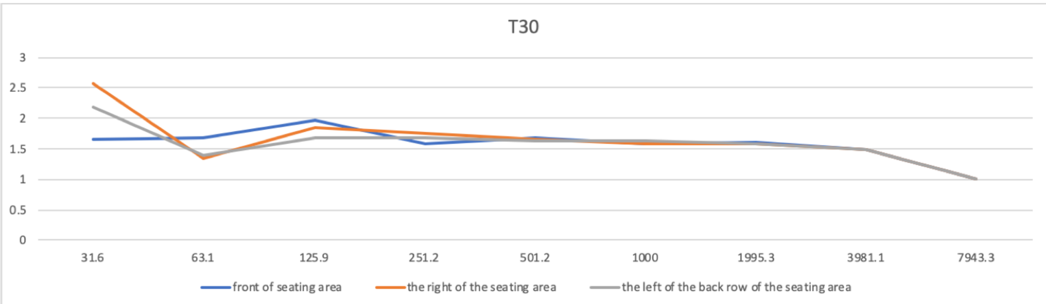

Figure 13: T_{30} of three different receiving positions for each octave band.

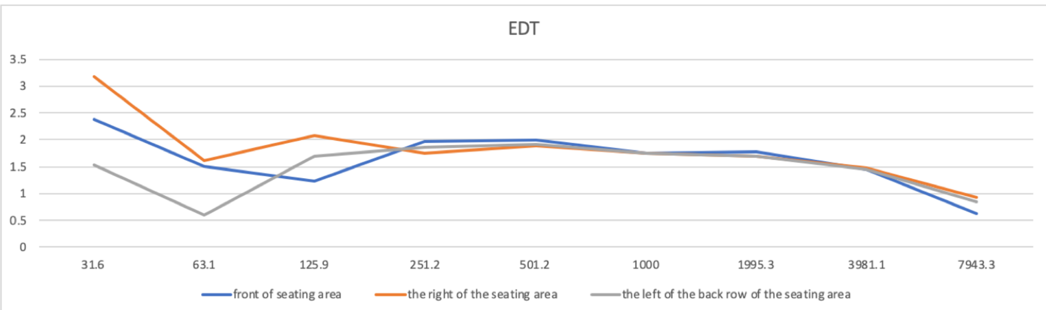

Figure 14: EDT of three different receiving positions for each octave band.

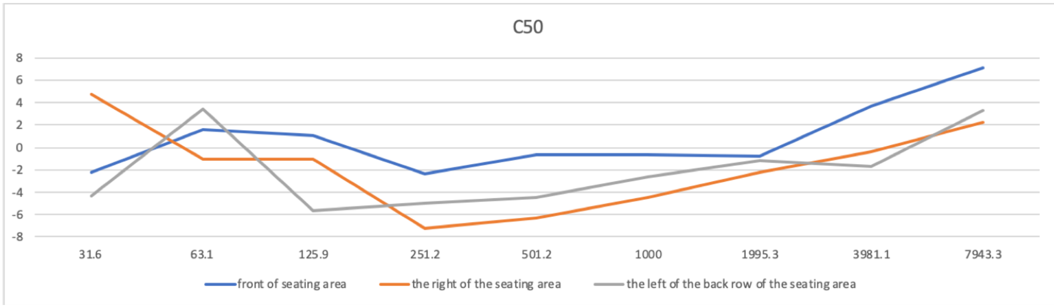

Figure 15: C_{50} of three different receiving positions for each octave band.

Receiver Position	T_{20}	T_{30}	EDT	C_{50}
front of seating area	1.68	1.58	1.64	0.75
the right of the seating area	1.69	1.65	1.82	-1.76
the left of the back row of the seating area	1.70	1.59	1.48	-2.02

Table 5: Octave bands average together to obtain more global indices to the room's reverberation

Using table 5 as a starting point, several apparent characteristics are as following:

- The sound source from the speaker in the stage center, the reverberation times T_{20} received at three different positions are almost the same (1.68, 1.69 and 1.79). The reverberation time of the concert hall does not change significantly with the changes of room location.

- There are significant differences in EDT at different receiving positions (1.65, 1.82, 1.48).

- Differences between EDT and T_{20} are obviously different at different positions, indicating that there is a clear difference between early reverberation attenuation and late attenuation at different positions.

- In the right of the seating area and the left of the back row of the seating area, C_{50} at these two locations are negative. A higher C_{50} value can provide clearer speech, that is, when consulting someone or conducting any other conversation that focuses on speech intelligibility. In other words, the intelligibility of the audience will be significantly different in front of the stage and in the seating area.

- The average T_{20} and EDT values of this concert hall (T_{20}=1.69, EDT=1.65 seconds respectively) indicate that their reverberation time is not quiet reasonable according the suggested parameter values (see figure 4.2). Besides, the figure 4.2 shows that T_{20} varies greatly across the entire frequency, especially at certain frequencies (31.6, 63.1, 236.9Hz).

Parameter	**Symbol**	**Chamber music**	**Symphony**
Hall size	V/N	2500 m^3/300 seats	25 000 m^3/2000 seats
Reverberation time	T	1.5 s	2.0–2.4 s
Early decay time	EDT	1.4 s	2.2 s
Strength	G	10 dB	3 dB
Clarity	C	3 dB	−1 dB
Lateral energy fraction	LEF	0.15–0.20	0.20–0.25
Interaural cross correlation	$1-\mathrm{IACC}$	0.6	0.7
Early support	$\mathrm{ST}_{\mathrm{early}}$	−10 dB	−14 dB

Figure 16: Suggested position-averaged values of objective room acoustic parameters in unoccupied concert halls for classical music [4]

5 Conclusions

The EDT and T_{20} values recommended in the figure 4.2 are hundreds of milliseconds lower than the results, it is worthy to consider the acoustic treatment of the interior of the concert hall. Reverberation time can be influenced by the size and shape of the room, the absorption of the sound-absorbing ceiling, and the amount of sound scattering and absorption of furniture and equipment in the room and so forth. We can reduce reverberation time by considering changing the interior materials of the concert hall, such as replacing the materials of the floor and ceiling with more sound-absorbing materials.

In Reid concert hall, the different locations have a great effect on the clarity. The audience in the front of the stage can hear the sound more clearly, while the effect in the back of the seat is worse. From a subjective perspective, EDT is more important because EDT is related to the perceived reverberation, and T is related to the physical properties of the auditorium, i.e. EDT can better correlate with the reverberations felt during speech and music playback [2]. This clarity is reflected in the form of an early decay time (EDT), which measures the initial 10dB decay rate in the decay curve (EDT = $6(t_{-10})$). The shorter the EDT means the higher the clarity [4]. Although in this project, the speakers used in this project were non-omnidirectional, which may have increased the EDT bias, placing the performer at the center rather than on the stage may be a good option to improve the perceived clarity around the room.

Besides, the Reid concert hall was designed with little reverberation which was mentioned in chapter 2; however reverberation times of experiment are significantly higher than the recommended values.

References

[1] Constant Hak, Remy Wenmaekers, and L. Luxemburg. "Measuring Room Impulse Responses: Impact of the Decay Range on Derived Room Acoustic Parameters". In: *Acta Acustica united with Acustica* 98 (Nov. 2012). DOI: 10.3813/AAA.918574.

[2] Claus Christensen, George Koutsouris, and Jens Rindel. "The ISO 3382 parameters: Can we simulate them? Can we measure them?" In: vol. 20. June 2013.

[3] Brad Rakerd et al. "Assessing the Acoustic Characteristics of Rooms: A Tutorial With Examples". In: *Perspectives of the ASHA Special Interest Groups* 3.19 (2018), pp. 8–24. DOI: 10.1044/persp3.SIG19.8.

[4] Anders Gade. "Acoustics in Halls for Speech and Music". In: *Springer Handbook of Acoustics, ISBN 978-0-387-30446-5. Springer-Verlag New York, 2007, p. 301* -1 (Jan. 2007), p. 301. DOI: 10.1007/978-0-387-30425-0_9.

[5] *Description of Reid Concert Hall.* URL: http://portal.historicenvironment.scot/designation/LB27995.

[6] Angelo Farina. "Simultaneous measurement of impulse response and distortion with a swept-sine technique". In: *Audio Engineering Society Convention 108.* Audio Engineering Society. 2000.

[7] Guy-Bart Stan, Jean-Jacques Embrechts, and Dominique Archambeau. "Comparison of different impulse response measurement techniques". In: *Journal of the Audio Engineering Society* 50.4 (2002), pp. 249–262.

[8] BSEN ISO and BRITISH STANDARD. *Acoustics—Measurements of Room Acoustics Parameters.* 2012.

[9] Rama Ratnam et al. "Blind estimation of reverberation time". In: *The Journal of the Acoustical Society of America* 114.5 (2003), pp. 2877–2892. DOI: 10.1121/1.1616578. eprint: https://asa.scitation.org/doi/pdf/10.1121/1.1616578. URL: https://asa.scitation.org/doi/abs/10.1121/1.1616578.