Simen Helbæk Kjølberg

Sound Levels at a Clarinetist's Ears During Solitary Practice

Master's thesis in Electronics Systems Design - Acoustics
Supervisor: Ulf Peter Svensson
Co-supervisor: Magne Skålevik
June 2023

Master's thesis

NTNU
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BREKKE STRAND

Abstract

This report presents sound pressure levels (SPL) measured at the ears of 4 clarinetists, individually playing structured practice sessions in a semi-anechoic chamber and in a practice room, along with the measured difference between sound power level (SWL) and the direct SPL at ears, for test signals in the dynamic *forte* measured in the semi-anechoic chamber. A quantification of uncertainty has been determined for the found parameters, including standard deviations and 95% confidence intervals. The sound level contributions of the direct and the reflected sound at the musicians' ears have additionally been determined for the practice room, using sound strength measurements with reference data of the head-related transfer function (HRTF) and the instrument's directivity index (DI) for the radiation angle towards the musician's ears. The findings have thus provided new insight of sound level contributions at the ears of clarinetists during solitary practice.

Daily, A-weighted exposure levels $L_{A,\mathrm{EX8h}}$ have been calculated from measurements of 25 minute practice sessions, based on average daily practice session durations of 2.1 hours, relating to published results by O'Brien *et al.* [1]. The average measured $L_{A,\mathrm{EX8h}}$ from practice sessions was found to be 82 dB re p_0 in the semi-anechoic chamber and 85 dB re p_0 in the practice room. As The Norwegian Labour Inspection Authority (Arbeidstilsynet) sets a legal limit of $L_{A,\mathrm{EX8h}} = 85$ dB re p_0 [2], the exposure from daily solitary practice in the practice room will alone maximize this limit. The average, C-weighted, maximum SPL $L_{CF\mathrm{Max}}$ between musicians was found to be 106 dB re p_0 in the practice room, higher than the measured 103 dB re p_0 in the semi-anechoic chamber. Although the level increase was expected due to room reflections, the dynamic range and uncertainty is observed to increase for a room of the given size ($V = 21.7\mathrm{m}^3$).

The measured power-pressure difference $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ at the dynamic *forte* was found to be 3.9 ± 0.3 dB (m = 32). The octave band level difference was however found to be lowest at the 1000 Hz band, but most stable in the 250 and 500 Hz bands. Results indicate that low values of $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ correspond with the musicians' decreased ability to hear the room response. This is verified through determination of the direct and reflected sound contribution at ears in the practice room, where measurements showed a slight increase of the direct sound compared to the reflected sound in the 1000 Hz octave band. For all other evaluated bands, the reflected sound level contribution was found to be higher than for the direct sound. The practice room used for measurements is however considered too small for an individual practice room, according to the volume limits given in ISO 23591:2021 [3]. As a result, the increase in SPL from the room response is relatively high.

The measurements and results presented in this report have certain limitations, related to sample size and methodology. Using a sound power measurement method for a free field environment over a reflective plane given in ISO 3744:2010 [4], the source has been defined as the entire musician playing the clarinet, as isolating the clarinet is difficult. Hence, the source and receiver overlap. Due to the non-stationary acoustic center of the clarinet, positioned above the floor, interference effects from floor reflection are also present in measurements. Simulations show up to a relative -2 dB SPL offset deviation in the 250 Hz band, for the measured sound pressure levels used for SWL measurements. Further research is needed to cast light on the Bb clarinet's acoustic centeroid, and the possibility of isolating the instrument. It would also be desirable with an in-depth study of the representative properties of an anechoic versus a semi-anechoic environment, aiming for the best practically viable representation of the direct sound of the clarinet. Finally, determining the $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ difference for other instruments would be of great interest.

Sammendrag

Denne rapporten presenterer målte lydtrykksnivå ved ørene til 4 klarinettister, fra individuelt framførte, strukturerte øvingssesjoner i semi-ekkofritt kammer og i et øvingsrom, sammen med den målte differansen mellom lydeffekt- og lydtrykknivå fra direktelyd målt ved ørene i semi-ekkofritt kammer, for testsignal i dynamikkgraden *forte*. En kvantisering av usikkerhet er oppnådd for de målte parametrene, inkludert standardavvik og 95% konfidensintervall. Lydtrykksbidragene fra direkte og reflektert lyd ved musikerenes ører har i tillegg blitt bestemt for øvingsrommet, ved bruk av Strength-målinger i kombinasjon med referansedata for "the head-related transfer function" (HRTF) og instrumentets direktivitetindeks (DI) for radieringsvinkelen mot musikerens ører. De presenterte resultatene har med dette bidratt til ny innsikt for lydtrykksbidrag ved ørene hos klarinettister under individuell øving.

Daglige, A-vektede eksponeringsnivå $L_{A,\mathrm{EX8h}}$ har blitt beregnet fra målinger av 25-minutters øvingssesjoner, basert på en gjennomsnittlig varighet på 2.1 timer for daglig individuell øving, knyttet til data publisert av O'Brien *et al.* [1]. Gjennomsnittlig $L_{A,\mathrm{EX8h}}$ fra øvingssesjoner ble målt til 82 dB re p_0 i semi-ekkofritt kammer og 85 dB re p_0 i øvingsrommet. I lys av Arbeidstilsynets juridiske grense på $L_{A,\mathrm{EX8h}} = 85$ dB re p_0 for A-vektet, 8-timers eksponeringsnivå [2], impliserer resultatene at alenebidraget fra daglig individuell øving vil maksimere denne grensen. Det gjennomsnittlige, C-vektede maksimallydtrykket $L_{CF\mathrm{Max}}$ mellom musikerene ble målt til 106 dB re p_0 i øvingsrommet, høyere enn de målte 103 dB re p_0 i semi-ekkofritt kammer. Selv om nivåøkningen i øvingsrommet var forventet på grunn av romrefleksjonsbidrag, ble det observert at usikkerheten og det dynamiske området øker for rom av den gitte størrelsen ($V = 21.7\mathrm{m}^3$).

Den målte differansen mellom lydeffekt- og direktelydtrykksnivå, $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ ved *forte*, ble funnet å være 3.9 ± 0.3 dB (m = 32). Oktavbåndsdifferansen ble derimot funnet å være lavest i 1000 Hz-båndet, men mest stabil i 250 og 500 Hz-båndene. Resultatene indikerer at lave verdier for $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ korresponderer med musikerenes nedsatte evne til å oppfatte romresponsen. Dette ble videre bekreftet gjennom bestemmelse av direktelyd- og refleksjonsbidragene ved ørene i øvingsrommet, hvor målinger viser en moderat økning av direktelydsbidraget sammenlignet med refleksjonsbidraget i 1000 Hz-båndet. For alle resterende oktavbånd som ble evaluert, ble refleksjonsbidraget vist å være høyere enn direktelydsbidraget. Øvingsrommet som ble brukt er forøvrig ansett som for lite i henhold til romvolumgrensene oppgitt i ISO 23591:2021 [3]. Som et resultat av størrelsen er økningen i de målte lydtrykksnivåene fra romresponsen relativt høy.

Resultater og målinger presentert i denne rapporten innehar bestemte begrensninger, relatert til utvalgsstørrelse og målemetodikk. Det har blitt benyttet en metode for måling av lydeffekt i fritt felt over en reflekterende overflate oppgitt i ISO 3744:2010 [4], som har ført til at kilden er definert som hele musikeren med klarinetten, ettersom isolasjon av klarinetten er utfordrende. Dermed eksisterer det en overlapp mellom kilde og mottaker. Grunnet et ikke-stasjonært akustisk senter av klarinetten som er posisjonert over gulvet, er også interferenseffekter fra gulvrefleksjoner tilstede i målingene. Simuleringer viser opptil et -2 dB relativt avvik i 250 Hz-båndet for målt lydtrykk benyttet i lydeffektberegninger, grunnet gulvrefleksjon. Videre forskning er nødvendig for å kaste lys over Bb-klarinettens akustiske senteroide, og muligheten for å isolere instrumentet i målinger. En detaljert studie av representativiteten for ekkofritt og semi-ekkofritt kammer ville også vært ønskelig, med et mål om å gi en best mulig representasjon av klarinettens direktelyd egnet for praktiske formål. Til sist ville det vært av stor interesse å identifisere differansen $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ for flere musikkinstrumenter.

Preface

This thesis concludes my Master's Degree in acoustics at the Department of Electronic Systems at NTNU. Working with this thesis has been a journey of long days, tiresome setbacks, ecstatic moments from solving analysis code bugs and deep dives into interesting research. Finishing this thesis feels like turning over the page from a lengthy chapter, ten years after my endeavor abroad into a Bachelor's Degree in Sound Technology, and five years after the beginning of my civil engineering studies at NTNU.

I want to thank my supervisors, professor Peter Svensson of NTNU and Magne Skålevik from Brekke & Strand, for great guidance throughout the work with the thesis and for providing me with just the right balance between academic freedom and helpful suggestions of direction. The practical guidance from Magne and the trusty academic input from Peter have been invaluable.

Much time has been spent planning, conducting and analyzing measurements. I want to thank Tim Cato Netland for helping facilitate all the necessary elements for measurements. Jesper Holsten, for inspiring me to dive into the object-oriented side of Python, to build my own set of analysis tools. Morten Andreas Edvardsen at Rambøll, for helping with brainstorming and pinpointing ideas at an early stage before commencing the thesis work. And my fullest gratitude go the participating musicians, who took the time out of their lives to contribute both to the future of clarinet acoustics, and to the resume of this future acoustics consultant.

Working with the thesis would be a much less bright experience if not for the support of the people surrounding me daily. I want to thank my classmates for making the years at NTNU a great and memorable time. A tip of the hat also goes to Brage Gaasø Samsonsen, as mutually agreed, for helping me maintain my humility at the tennis table in the NTNU break room.

Thanks go to the people at the Trondheim offices of Brekke & Strand, for providing me with a great daily environment for working with the thesis, both socially and facilitatory.

Big thanks go to my family for sticking up with me through my many years of studying, and especially my parents for sparking my interest in music and sound at an early age.

Finally I want to thank Merete, who has been a steady anchor during the years of my studies, even in chaotic times of switching jobs and moving apartments.

Simen Helbæk Kjølberg
6th of June 2023

Table of Contents

List of Figures

List of Tables

Abbreviations

Table 1: Abbreviations used in this report

Symbol	Quantity	Unit/Value
A	Equivalent Absorption Area	m^2
DF	Directivity Factor	
DI	Directivity Index	dB
f_s	Sampling Rate Frequency	Hz
G	Sound Strength	dB
G_{diff}	Sound Strength estimation from T and V in diffuse field	dB
L_A	A-weighted Sound Pressure Level	dB re 20μPa
L_{eqT}	Time-Equivalent Sound Pressure Level over a duration T	dB re 20μPa
L_{EX8h}	Noise Exposure Level Normalized to a Nominal 8h Working Day	dB re 20μPa
L_C	C-weighted Sound Pressure Level	dB re 20μPa
L_{WA}	A-weighted Sound Power Level	dB re 1pW
p	Sound Pressure	Pa
p_0	Reference Sound Pressure	20μPa
r_{ac}	Distance from ears to Acoustic Center	m
r_{crit}	Critical Distance (Hall Radius)	m
SD	Standard Deviation (of sound levels)	dB
SNR	Signal-to-Noise Ratio	dB
SPL, L_p	Sound Pressure Level	dB re 20μPa
SWL, L_W	Sound Power Level	dB re 1pW
T, T_{60}, T_{20}	Reverberation Time	s
T_{mid}	Mid-Frequency Reverberation Time	s
V	Room Volume	m^3
W	Sound Power	W
W_0	Reference Sound Power	1pW

AD	Analog-to-Digital (Conversion)	
CI	Confidence Interval	
CIPIC	Center for Imaging Processing and Integrated Computing	
HRTF	Head Related Transfer Function	
HRIR	Head Related Impulse Response	
IR	Impulse Response	
ISO	International Organization for Standardization	
PSA	Perforated Slot Absorber	
RSS	Reference Sound Source	
WAV	Waveform Audio File Format	
WHO	World Health Organization	

Chapter 1

Introduction

1.1 Motivation

Musicians in orchestras and woodwind bands constitute a unique group regarding daily exposure of sound, both due to levels and signal properties. Musical instruments can produce a great range of variety in properties like sound pressure level, directivity, frequency range and harmonic partials. Not only do the various instruments differ from each other, but most single instruments, like the Bb clarinet, will yield great variation in the mentioned parameters. Consequently, the clarinet will sound different based on factors like listener position, the power produced by the player, finger positioning, as well as physical instrument features [5]. From the perspective of a listener in the audience, the variation from the instrument can partially be masked by playing in a larger ensemble [6] or in a reverberant room. For the latter case, the diffuse field of a room will "smooth" out the level variations that occur from dynamic variation.

A musician however also spends much time practicing in solitude. In this situation, the sound levels reaching the ears of the musician themselves will include multiple varying contributions, which during solitary practice mainly is the direct sound and reflected sound from the room. These levels will depend on source related properties like power and directivity of the instrument, as well as room features such as strength (G) and reverberation.

Identifying the sound levels and the different level contributions at a musician's ears during solitary practice is of high interest, with two main motivations:

1. Considerations regarding the health conditions of musicians due to sound exposure.
2. Identifying signal properties of the musical instrument for room acoustics, instrument synthesis and other purposes related to music technology.

For health considerations, it is desirable to identify the levels which musicians are exposed to on a daily basis. Musicians have been found to show more noise induced hearing loss (NIHL) than what would typically be expected based on age and gender [7]. Reports also show a higher prevalence of tinnitus and hyperacusis, which were respectively diagnosed for 26.5% and 18.9% of groups of participating classical musicians (respectively 1567 and 503) in a study by Di Stadio et. al [8]. In addition to hearing damage, noise in excessive amounts or longer durations can cause other long term health problems. The World's Health Organization (WHO) reported noise as a leading cause of stress in Europe, subsequently leading to for example cardiovascular effects, reduced sleep quality among other major health issues [9].

Regarding the signal analysis of the radiated sound from the clarinet, the motivation for separating the direct and reflected signals at the musicians' ears can help yield a greater understanding of the Bb clarinet as a musical instrument. In previous research, sound levels at musicians' ears during solitary practice in non-anechoic conditions [1], and radiated sound power for various dynamic degrees have been measured [10]. Moreover, the international standard for acoustic rehearsal and recital rooms, ISO 23591:2021, provides a table of typical power levels (at the dynamic *forte*) radiated by various instruments, which is used as a reference in designing music rehearsal and practice rooms [3]. However, little data exists regarding the corresponding ear levels from the direct sound contributions. Information about the relevant contributions at the ears would be of great value for multiple purposes, including music room acoustics, auralization and simulations or modeling of musical instruments.

The Bb clarinet has been chosen as an isolated case study for investigating these concerns. The goal of this report is to quantify and identify the sound levels at the musician's ears, along with the sound power, both measured in an anechoic or semi-anechoic room. Measurements of sound levels should also be measured in a practice room, where a quantification should be made of the contributions from the direct and reflected sound. The conditions for the dynamic *forte* are of primary interest, relating to the information given in ISO 23591:2021. Specifically, a supplementing value for the expected A-weighted power-pressure difference $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ between the radiated power and the direct sound pressure level (SPL) contribution at the ears is desirable. A quantification and discussion of uncertainty should be included.

1.2 Background

This section will present previous research, along with other essential background information relevant to the topic. The key points discussed in this section is to be regarded as the foundation for how the methods, data selection, tests and measurements have been designed.

1.2.1 Previous Research

There has been conducted a number of studies related to the sound pressure levels at the ears of music performers. O'Brien *et al.* made measurements of professional orchestral musicians during solitary practice, using structured practice sessions of 23 minutes [1]. The sessions included tuning notes in specified dynamics, as well as technical warm-up routines and a piece of orchestral repertoire. The measured A-weighted, sound pressure levels were presented, with time-equivalent calculations for two duration contexts. These were the time-equivalent levels for the 23 minute practice sessions, $L_{A,\mathrm{eq23min}}$, along with calculated estimates of the average daily exposed levels based on 2.1 hour practice sessions per 8 hour work day, $L_{A,\mathrm{EX8h}}$. Measurements of maximum, C-weighted, instantaneous sound pressure levels, $L_{C,\mathrm{peak}}$, were also presented. As the measurements were made in practice rooms, the measured sound levels contained contributions from both direct and reflected sound, between which there were made no distinction. A more detailed description of relevant parameter definitions is given in Chapter 2.

The reported average level measured at the ears of musicians practicing the Bb clarinet was $L_{A,\mathrm{eq23min}} = 92$ dB re p_0. Based on an average rehearsal duration of 2.1 hours, this level corresponds to a daily exposure of $L_{A,\mathrm{EX8h}} = 86$ dB re p_0 for an 8 hour work day. It should be stressed that this estimate of $L_{A,\mathrm{EX8h}}$ does not include daily group rehearsal sessions, which will further increase the average daily exposure. For the loudest tuning notes (*ff: fortissimo*), the average measured level from a 15 second sustained note was $L_{A,\mathrm{eq15sec}} = 97$ dB re p_0. Moreover, the maximum instantaneous level was reported to be $L_{C,\mathrm{peak}} = 111$ dB re p_0. By comparison, The Norwegian Labour Inspection Authority (Arbeidstilsynet) sets a legal limit of $L_{A,\mathrm{EX8h}} = 85$ dB re p_0 for the daily A-weighted 8-hour exposure level, and a C-weighted maximum allowed peak level of $L_{C,\mathrm{peak}} = 130$ dB re p_0 [2]. The results found by O'Brien for the Bb clarinet, summarized in Table 1.1, is used as a reference to the findings in this report. Table 1.2 shows an indicative list of typical A-weighted sound pressure levels for selected activities and sound sources, as well as for certain dynamic notations, for reference.

Table 1.1: Sound pressure levels measured at clarinetists' ears, published by O'Brien *et al.* [1], with corresponding legal level limits set by The Norwegian Labour Inspection Authority (Arbeidstilsynet) [2]. Daily exposure level $L_{A,\mathrm{EX8h}}$ is calculated from assuming an average 2.1 hour duration of daily practice sessions, based on measured $L_{A,\mathrm{eq23min}}$.

	$L_{A,\mathrm{eq15sec}}$ (*ff*), [dB] re p_0	$L_{A,\mathrm{eq23min}}$ [dB] re p_0	$L_{A,\mathrm{EX8h}}$ [dB] re p_0	$L_{C,\mathrm{peak}}$ [dB] re p_0
O'Brien	97	92	86	111
Norwegian legal limit (Arbeidstilsynet)	-	-	85	130

Table 1.2: Examples of activities or sources with typical A-weighted sound pressure levels [11], with approximate dynamic strengths. Typical levels for dynamic notations published by Rindel are originally given as unweighted SPL values [12]. All levels are indicative.

$L_{A,\mathrm{eq}}$ [dB] re p_0	**Example of activity or source**	**Dynamic (Indicative)**
0	Softest sound audible to a person with normal hearing	
10	Normal breathing	-
30	Whispering	-
50	Rainfall	*ppp*
60	Normal conversation	*pp*
70	Freeway Traffic	*p*
80	Ringing Telephone	*mf*
90	Tractor	*f*
100	Factory machinery	*ff*
110	Leaf Blower	*fff*
120	Thunder	-
140	Airplane taking off	-

Previous research has also shown measurements of high stage levels for musical performance, meaning situations outside solitary practice. It was found by A. C. Gade that the average measured $L_{A\mathrm{eq,ears}}$ at the musicians' ears for a whole orchestra, ranged between 85 and 91 dB re p_0 for various classical pieces and in different music halls [13]. Moreover, R. Wenmaekers showed little reduction of noise exposure during performance from using simple physical measures like screening behind musicians or elevating the position of certain instrument groups [14]. Consequently, musicians will generally experience daily equivalent levels higher than those estimated from the solitary practice sessions alone.

1.2.2 Acoustics in Practice Rooms

Suitable room acoustics are essential to achieving good rooms for practicing music. For acoustic music instruments, meaning instruments not relying on electrical power amplifiers, the average SPL is dependent on factors like the type and number of instruments, dynamic expression of the instrument(s), room volume and reverberation [12]. Historically, music written for acoustic instruments has been performed in reverberant halls amplifying the music for an audience, while adjustments of levels were adressed through positioning of instrument groups and dynamic control of the players, often through the guidance of a conductor. Most music written for acoustic instruments therefore require some degree of room response as part of the musical expression.

Music rooms for individual practice, non-individual rehearsal and recital[1] are today projected according to the international standard ISO 23591:2021, which provides guidelines for room acoustic

[1]ISO 23591:2010 separates between rooms for rehearsal and recital, the latter not being relevant for this report. For the context of this study, the word "practice" is chosen to refer to technical instrumental work in solitude, whereas "rehearsal" is interpreted as a group of musician rehearsing music repertoire.

properties based on music category. The three categories currently used in ISO 23591 for practice and rehearsal rooms are given in Table 1.3 [3].

Table 1.3: Music categories for room types as given in ISO 23591:2021 [3]. Bb clarinet belongs to the "Loud" category, as indicated in bold fonts.

Music category	**Typical instruments and applications**
Quiet	Classical instruments, Vocal, Piano
Loud	**Wind instruments (Clarinet)**, Grand Piano
Amplified	Amplified instruments, Amplified Vocal

As a woodwind instrument, the Bb clarinet belongs to the "Loud" category in ISO 23591. For individual practice rooms of that specific category, criteria is given for properties like average room height, net volume and area, reverberation time and background noise level. Figure 1.1 shows criteria for T_{mid} reverberation time and room volume V, for the music categories as given in ISO 23591. Estimations of the sound strength parameter G, G_{diff}, are also indicated in the figure, relating to the pressure amplification properties of the room. The G_{diff} values are diffuse field estimations based on reverberation time and room volume, further elaborated in Section 2.8.

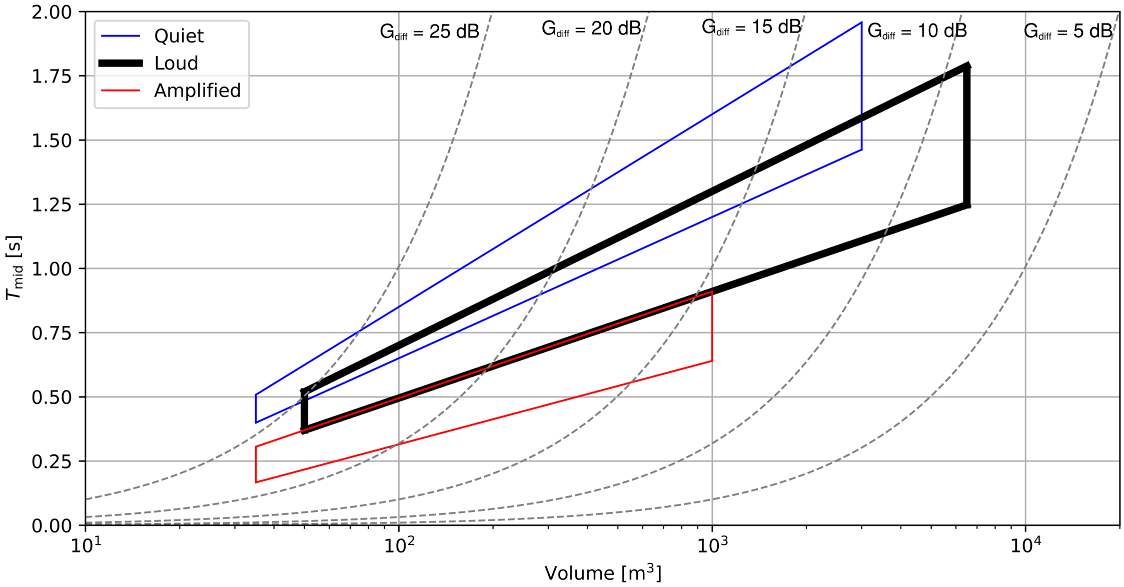

Figure 1.1: Limits and parameter guidelines of T_{mid} and room volume V, as provided in ISO 23591: "Loud" category used for Bb clarinet indicated with thicker black line. Sound strength G_{diff} estimate marked with dashed lines. Figure constructed from values given in ISO 23591:2021 [3].

Using the above figure for a suitable T_{mid} value range for a given room volume, a corresponding set of curves for the optimal reverberation values in octave bands is given by ISO 23591 for the selected music category. Figure 1.2 shows the reverberation time T in octave bands for the "Loud" category, as a percentage value $p\%$ of the T_{mid}, constructed from the standard.

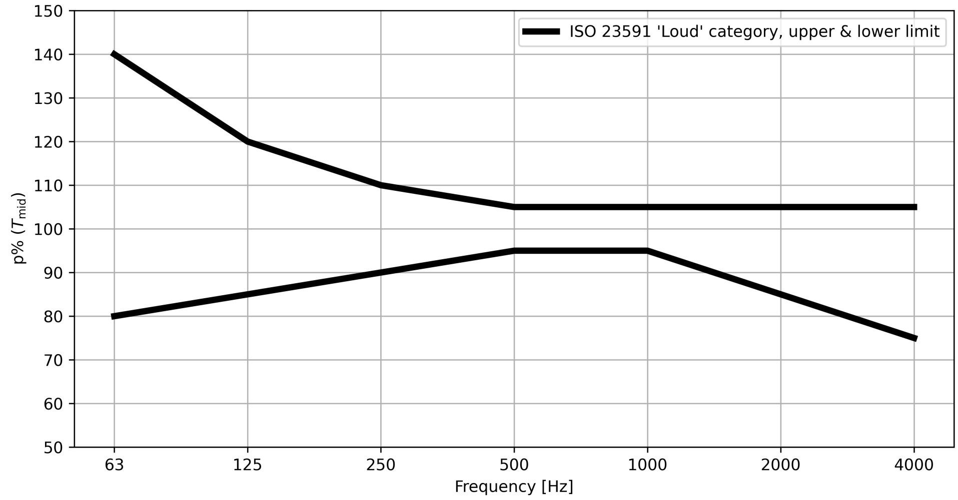

Figure 1.2: Upper and lower limits for reverberation time T in octave bands for the "Loud" category, given as a percentage value p of the T_{mid}. Figure constructed from values given in ISO 23591:2021 [3].

1.2.3 Acoustic Properties of the Clarinet

The Bb clarinet, as a woodwind instrument, has a characteristic build influencing its tonal qualities. It has a near-cylindrical shape, made by wooden parts attached by metal rings. The mouthpiece at the top of the instrument houses a wooden reed, attached with a ligature, excited to vibration by the mouth of the player. The mouthpiece is connected via the barrel to the upper and lower joints, where the tone holes and keys are used to produce various pitches. At the lower end of the instrument, a conical bell extends the width of the cylinder to a flare shape. Figure 1.3 shows an image of a Bb clarinet, marked with the relevant parts.

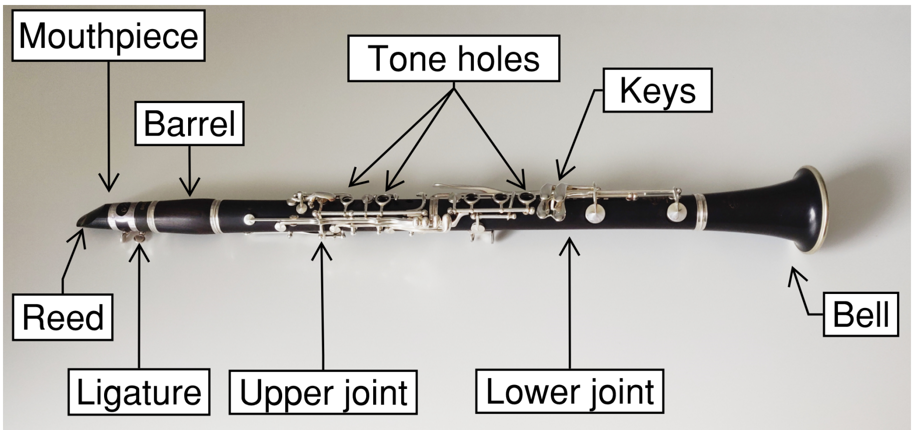

Figure 1.3: Photo of a Bb clarinet with indications of important parts.

The clarinet is often used as a classic example of a closed-open pipe, in educational texts for acoustics or physics (for example, see [15]). As with other wind instruments, the acoustic properties and sound characteristics of the clarinet results from the acoustic impedance Z measured close to the embouchure [5]. The reed, excited to vibration from the lips of the musician, acts as a closed end at the upper part of the instrument, operating with maximas of sound pressure (p). The bell at the bottom end acts as the open end of the instrument with maximas of volume velocity (U), and zero sound pressure. These properties lead to a set of resonances and anti-resonances where even harmonics of f_n (n = 2, 4, 6, ...) are attenuated. As a result, the sound of the Bb clarinet is characterized by a prominence of odd harmonic partials f_n (n = 1, 3, 5, ...) This characteristic is illustrated in Figure 1.4.

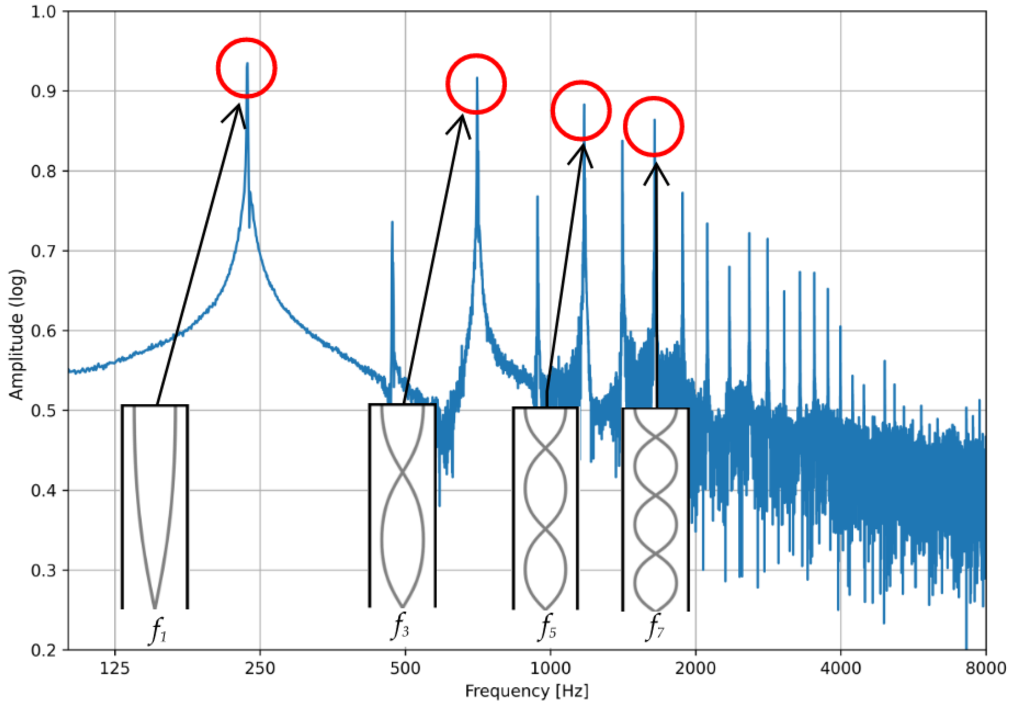

Figure 1.4: Harmonic partials of the Bb clarinet for a sustained note, with corresponding pressure wave resonance patterns. The fundamental frequency f_1 = 233 Hz is indicated, along with the odd harmonic partials f_3, f_5 and f_7. Wave pattern illustrations show the maximas of p at the top (reed) and zero pressure nodes at the bottom (bell), inspired by Figure 3 in *Clarinet Acoustics* by Dickens *et al.* [5].

The name of the Bb clarinet stems from the fact that the instrument is transposed by two semi-tones from a C note in what is commonly called a concert pitch, or a C *natura*. This means that when asked to play a C note, which is a common root note for western music scales, a clarinetist would produce a pitch corresponding to a Bb note on e.g. a grand piano, which is two semi-tones below a C note [16]. Figure 1.5 shows how a played C_4 on a Bb clarinet corresponds to a Bb_3 in concert pitch notation, both representing the same pitch with fundamental frequency f = 233 Hz.
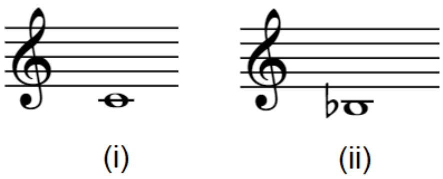
Figure 1.5: Comparison of notation for the same pitch with a shared fundamental frequency f = 233 Hz, displayed in notation for (i) a Bb clarinet, versus (ii) concert pitch notation (natura).

The Bb clarinet is among the wind instruments with largest range of musical pitch, with a span of more than 3.5 octaves [17]. Ranging from its lowest note, an E_3 (D_3 natura), it can produce notes up to a C_7 (Bb_6 natura). In terms of frequency range, this corresponds to a range between 147-1865 Hz for the fundamental frequencies, although the harmonic partials reach above this range.

The sound radiation of the Bb clarinet varies depending on pitch and dynamics (produced sound power). From measurements reported by Jürgen Meyer, the typical, frequency-unweighted sound power level at *forte* is shown to be $L_W = 93$ dB re 1pW, which is also the main reference for the typical power levels presented in ISO 23591:2021 [10, 3]. From a distance of 9 meters, Meyer however reported sound pressure levels played in different registers ranging from $L_A = 35$ dB re p_0 for certain octaves, to exceeding $L_C = 90$ dB re p_0 in the stronger registers of the instrument.

Regarding the directional patterns of the Bb clarinet, the instrument is often compared to the oboe, despite having different sound excitation mechanisms. For higher frequencies, the Bb clarinet is shown to radiate mainly in the direction of the bell, gradually becoming more omnidirectional as the frequency decreases [18]. Below 500 Hz, Meyer found the radiation pattern of the Bb clarinet to be approximately spherical [10]. The average directional properties were interestingly reported by Pätynen and Lokki to remain stable between various dynamics [18]. Figure 1.6 shows the directivity pattern (directivity index) in decibels for the Bb clarinet in selected 1/1 octave bands, from a side view of the musician.

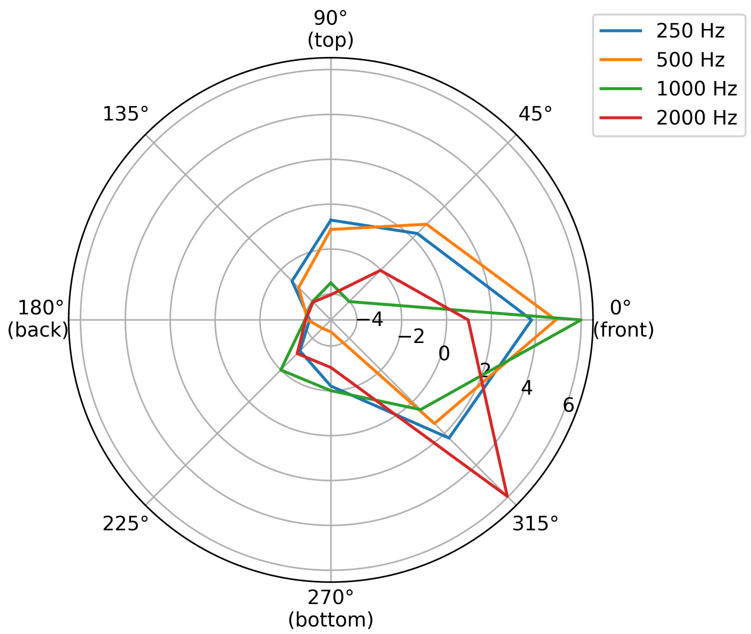

Figure 1.6: Directivity index (DI) in 1/1 octave bands of Bb clarinet from side view of musician, in decibels. Figure is reconstructed from values in Figure 11, Pätynen and Lokki [18], offset from normalized values to obtain logarithmic averages of 0 in each band.

Chapter 2

Theory

This chapter gives an overview of the relevant technical theory used for this report, related to measurements, parameters and estimation techniques. Methods and considerations for subjective analysis and interviews is described in Chapter 3.

2.1 Sound Pressure Levels

The sound pressure level (SPL), L_p, is a quantity for expressing the sound pressure p in relation to a reference sound pressure value. It is often expressed as

$$L_p = 10\log_{10}\left(\frac{p^2}{p_0^2}\right) \text{ [dB] re } p_0, [19] \tag{2.1}$$

where $p_0 = 20 \cdot 10^{-6}$ Pa is the reference pressure in air.

2.1.1 Diffuse Field SPL

For a room with a diffuse field, a theoretical SPL L_p at a given distance from the source can be expressed with:

$$L_p = L_W + 10\log_{10}\left(\frac{DF_S DF_R}{4\pi r^2} + \frac{4}{A}\right) \text{ [dB] re } p_0, \tag{2.2}$$

where the first term L_W is the sound power level (SWL) of the source, while the second and third term represent contributions from the direct sound and the diffuse field, respectively. The direct sound terms includes the directivity factor for the source, DF_S, and the receiver, DF_R, while r is the distance from the source in meters. In the diffuse field term, A is the equivalent absorption area of the room, given in m^2.

The SPL in Equation 2.2 is impacted by changes to the direct sound term or the diffuse term. If the distance r is doubled, the direct sound contribution drops by 6 dB, while a doubling of A leads to a 3 dB drop for the diffuse field term. As A is inversely related to the reverberation time, shown with Sabine's formula in Section 2.7, an increased reverberation time leads to an increased SPL from the diffuse-field term.

The distance $r = r_{crit.}$ is the critical distance where direct sound and diffuse-field term contributions in Equation 2.2 are equal.

2.1.2 Directivity

The directivity factor used for the source and receiver in Equation 2.2 is a ratio number, expressing the radiated/received energy compared to an omnidirectional source/receiver, for a certain angle of radiation or incidence. An omnidirectional source of receiver will have a $DF = 1$.

The directivity index, DI, is used to express the DF logartithmically, in decibels:

$$DI = 10 \log_{10} DF \text{ [dB]}. \tag{2.3}$$

2.1.3 Averaging and Summing Levels

The method of averaging and summing sound levels is dependent on the context of the measurements. For calculating the average exposure or radiated sound energy over time, or for determining the averaged measured level between simultaneous measurement positions, a logarithmic average level $\overline{L}$ for the sound energy is used (energetic average). Similarly, for summing simultaneous sound level contributions to a total level, the logarithmic sum level L_{tot} of the energy is used (energetic sum).

The calculation methods for logarithmic averaging and summing of n levels L_i are respectively given in equations 2.4 and 2.5.

$$\overline{L} = 10 \log_{10} \left(\frac{1}{n} \sum_{i=1}^{n} 10^{\frac{L_i}{10}} \right) \text{ [dB]} \tag{2.4}$$

$$L_{tot} = 10 \log_{10} \left(\sum_{i=1}^{n} 10^{\frac{L_i}{10}} \right) \text{ [dB]} \tag{2.5}$$

In a situation with repeated measurements of a source, an average level represents the mean of measured values from different moments in time. If the engineer attempts to determine a "true" level of a source by repeating measurements, an arithmetic average of the measured levels is used.

2.2 Time Equivalent and -Weighted Levels

Time equivalent levels are often used for one or several durations of stationary or non-stationary noise exposure, to express the equivalent average level exposure over a certain duration of time. The time-equivalent sound pressure level L_{eqT} for a duration T in seconds, is denoted as:

$$L_{\mathrm{eqT}} = 10 \log_{10} \left(\frac{1}{\mathrm{T}} \int_{t=\mathrm{T}}^{t} \frac{p^2(\xi)}{p_0^2} d(\xi) \right) \text{ [dB] re } p_0, [20] \tag{2.6}$$

where $p_0 = 20 \cdot 10^{-6}$ Pa is the reference pressure in air.

The above principle of time equivalent levels can be used for entire durations, or for time weighting measurements. For time weighted measurements, a series of time equivalent levels are calculated for subsequent time windows of a fixed duration. The commonly most used time-weightings are "Fast", F, and "Slow", S, corresponding to time windows of 0.125 seconds and 1 second. The fast and slow time weighted sound pressure levels L_F and L_S are expressed as:

$$L_F = 10\log_{10}\left(\frac{1}{0.125\text{s}}\int_{t=0}^{0.125\text{s}}\frac{p^2(t)}{p_0^2}d(t)\right)\text{ [dB] re } p_0,\ [20]$$
$$L_S = 10\log_{10}\left(\frac{1}{1\text{s}}\int_{t=0}^{1\text{s}}\frac{p^2(t)}{p_0^2}d(t)\right)\text{ [dB] re } p_0.\ [20] \tag{2.7}$$

2.3 Daily Exposure Levels

ISO 1999:2013 provides guidelines for calculating noise exposure levels, related to the estimation of noise-induced hearing loss. The noise exposure level normalized to a nominal 8 hour working day, L_{EX8h}, is given by:

$$L_{\text{EX8h}} = L_{\text{eqT}} + 10\log_{10}\left(\frac{\text{T}}{\text{T}_0}\right)\text{ [dB] re } p_0, [21] \tag{2.8}$$

where T is the time of noise exposure in hours, and $\text{T}_0 = 8\text{h}$ is the reference time for a working day.

2.4 Frequency Weighting

Frequency weighting is used to mirror human perception of loudness for SPL. The most commonly used weighting curves are A- and C-weighting. The weighting curves act as an offset to the sound levels depending on the frequency, illustrated in Figure 2.1 [20].

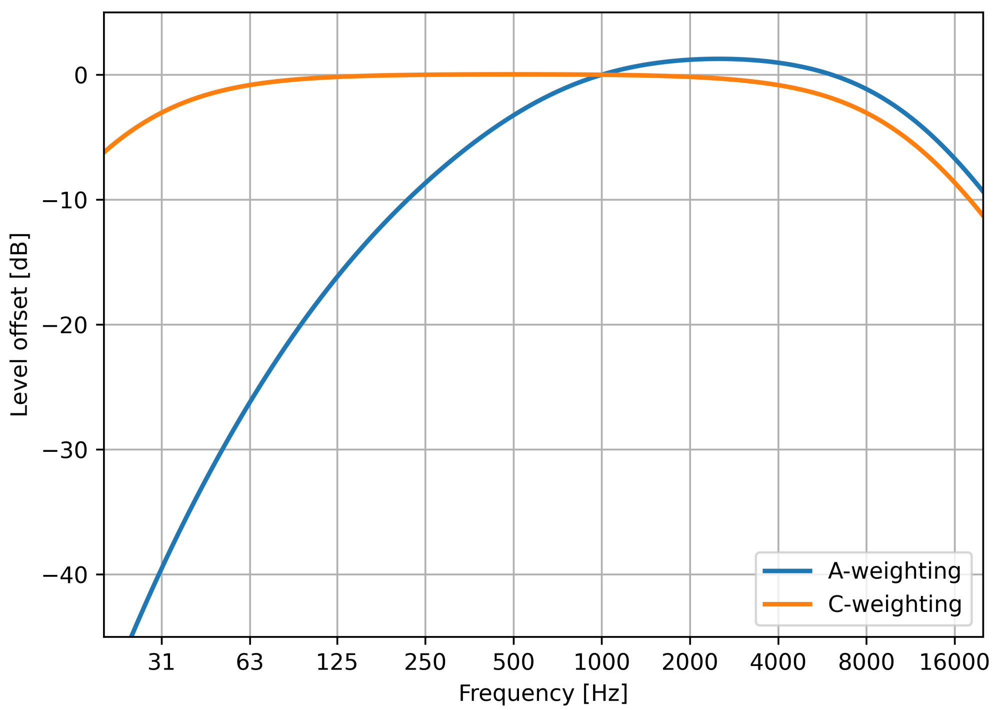

Figure 2.1: Frequency weighting curves used for offsetting sound levels [20]. Blue curve shows the A-weighting, orange curve shows C-weighting.

For lower SPL, humans are less sensitive to lower frequencies and more sensitive in frequencies around the resonance area of the ear canal. This is reflected through the A-weighting curve. When SPL is higher, the differences in sensitivity decrease and the perceived loudness flattens across the frequency spectrum, which is reflected through the C-weighting curve.

The weighting curves can be applied to both SPL and sound power levels. As an example, the SPL for frequency weighting A and time weighting F is written as L_{AF}.

A common notation for A- or C-weighted level values is to add a suffix to the decibel unit, like "dB(A)" or "dB(C)". For this report, this will not be used, and the frequency-weightings of levels will instead be indicated by subscripts, as $L_{A(...)}$, $L_{WA(...)}$, $L_{C(...)}$ or $L_{WC(...)}$.

2.5 Maximum Sound Pressure Levels

Maximum sound pressure levels are used to express the upper extremity of the SPL over the duration of one or several measurements. The maximum SPL is commonly expressed as the maximum time-weighted sound level, for either "Fast" or "Slow" weighting. The maximum level is then the highest measured time-weighted level from a series of time-weighted SPLs. As an example, the maximum SPL for frequency weighting C and time weighting F is written as $L_{CF\max}$. [20].

The maximum SPL can also be expressed through the maximum instantaneous SPL within a stated time interval. This is often used with a C frequency weighting, commonly referred to as the maximum C-weighted peak SPL, $L_{C,\text{peak}}$. [1]

2.6 Sound Power

The sound power level (SWL) L_W is used to express the radiated sound power of a source. From a known sound power value W, the sound power level can be expressed as

$$L_W = 10\log_{10}\frac{W}{W_0}\ \text{[dB] re 1pW}, [19] \tag{2.9}$$

where W is given in Watts and $W_0 = 1$ pW is the reference power value.

2.6.1 Methods of Determining Sound Power

Several methods of identifying L_W from measurements exist. Using a free field environment over a reflective field, the sound power level can be estimated from measurements of L_p. This method is described in NS-EN ISO 3744:2010 "Acoustics: Determination of sound power levels and sound energy levels of noise sources using sound pressure — Engineering methods for an essentially free field over a reflecting plane" [4].

Following the method described in ISO 3744, the sound source is placed in a fixed position on the horizontal plane. A hypothetical measurent surface is defined either as a parallelepiped, cylinder or a hemisphere centered around the source. For the hemispherical measurement surface, a radius r equal to or greater than twice the characteristic source dimension, d_{O}, should be used. If the source is positioned centered on a point origo on the reflective plane, d_{O} is defined as the length from origo to the furthest corner of the minumum sized parallelepiped just enclosing the source.

Having defined a hemispherical measurement surface, a set of measurement positions are distributed along the surface to measure L_p. The number of positions should ideally be 10, based on

[1]The maximum, C-weighted peak or instantaneous SPL, $L_{C,\text{peak}}$, is commonly used for legal limits related to noise at workplaces and events, in Norway and in some other countries.

key positions relative to the source center point given in ISO 3744. These positions can be found in Table E.1, Appendix E.

The sound power level L_W is then found by using

$$L_W = \overline{L_p} + 10 \log_{10} \frac{S}{S_0} \text{ [dB] re 1pW}, \tag{2.10}$$

where $\overline{L_p}$ is the logarithmic average sound pressure level from all measurement positions, while S is the area in m^2 of the measurement surface and S_0 is the reference surface area of 1 m^2.

Following the procedure described in ISO3744, $\overline{L_p}$ should be calculated using correction terms for background noise and for the influence of the measurement environment. Using the uncorrected averaged level for the source under test $\overline{L'_{p(\text{ST})}}$, the corrected, time-averaged sound pressure level is found by

$$\overline{L_p} = \overline{L'_{p(\text{ST})}} - K_1 - K_2 \text{ [dB] re } p_0, \tag{2.11}$$

where K_1 and K_2 are the background noise and the environmental correction terms given in ISO 3744, respectively. K_1 is applied if the ratio ΔL_p between the signal and background noise is 15 dB or less. As the value of K_2 increases with reverberation time, the environmental correction term is considered negligible for an anechoic or a semi-anechoic chamber.

The A-weighted sound power level, L_{WA}, can be identified by either applying the frequency weighting curve given in Figure 2.1 or by using A-weighted sound pressure levels to find the averaged $\overline{L'_{pA(\text{ST})}}$. In the latter case, applying the A-weighted levels directly in equations 2.11 and 2.10 will yield L_{WA}.

2.6.2 Standardized Uncertainty

ISO 3744:2010, Chapter 9, provides guidelines for determining uncertainty for sound power levels. The sound power level uncertainty $u(L_W)$, in decibels, is estimated from the total standard deviation σ_{tot} in decibels:

$$u(L_W) \approx \sigma_{tot} \text{ [dB]. [4]} \tag{2.12}$$

The total standard deviation σ_{tot} is further expressed from standard deviation components related to technical reproducibility, σ_{R0}, and uncertainty from instability of operating and mounting conditions of the source, σ_{omc}, both expressed in decibels. The reproducibility component σ_{R0} can be found through round robin tests, through mathematical modeling using known uncertainty information about all measurement components or from general reference data provided in ISO 3744. For the source related uncertainty σ_{omc}, information is obtained through repeated measurements. The expression for determining σ_{tot} is given as:

$$\sigma_{tot} = \sqrt{\sigma_{R0}^2 + \sigma_{\text{omc}}^2} \text{ [dB]}, \tag{2.13}$$

where in the context of a person playing a musical instrument, σ_{omc} is interpreted to represent the variations inherent to the musical instrument and from the player(s).

By definition, sound power measurements made in accordance with NS-EN ISO 3744:2010 correspond to an engineering grade of accuracy (grade 2).

2.7 Reverberation

The reverberation time T, or T_{60}, is the time in seconds taken for the level of the SPL in a room caused by reflections to drop by 60 dB. The reverberation time can either be estimated from mathematical formulas or measured directly. A expression for estimating T_{60} is the classical formula by Sabine:

$$T_{60} = 0.161\text{s/m}\frac{V}{A}\ [\text{s}], [22] \tag{2.14}$$

where V is the room volume in m^3 and A is the equivalent absorption area in m^2.

In addition to T_{60}, T_{20} and T_{30} are often used for measurements. These are both estimates of the T value, based on extrapolating the measured time taken for a 20 dB or 30 dB drop, respectively.

The parameter T_{mid} is often used to represent the reverberation time in the midrange frequencies, 500 Hz and 1 kHz. For given reverberation times in 1/1 octave bands, the T_{mid} is expressed as:

$$T_{mid} = \frac{T_{500\text{Hz}} + T_{1000\text{Hz}}}{2} [\text{s}]. \tag{2.15}$$

Knowing the reverberation time T and the room volume V of a room, the Schröeder frequency f_c can be determined. The frequency f_c indicates a (non-abrupt) transition frequency between the range of individual room mode domination, where the wavelengths are comparable to the dimensions of the room, to a range where the modes overlap in such a high density that individual modes are no longer distinguishable. The Schröeder frequency is given by:

$$f_c = 2000\sqrt{\frac{T}{V}}\ [\text{Hz}]\ [23]. \tag{2.16}$$

2.8 Sound Strength

The sound strength, G, is a measure of the sound amplification properties of a room. It is expressed from the ratio between the total SPL in the diffuse field, L_p, and the direct SPL from the isolated sound source at a 10 meter distance, $L_{p,dir,10m}$:

$$G = L_p - L_{p,dir,10m}\ [\text{dB}].[22] \tag{2.17}$$

An estimation of the sound strength $G_{\text{diff}} \approx G$ in the diffuse field can be acquired, by combining the above equation with equation 2.2 and the expression for A in equation 2.14:

$$G_{\text{diff}} = 10\log_{10}\frac{T}{V} + 45\ [\text{dB}], \tag{2.18}$$

assuming an omnidirectional source and receiver ($DF_S = DF_R = 1$). For the G_{diff} estimate, it is also assumed that the diffuse field term is dominating ($r >> r_{crit.}$), so that the direct sound contribution can be neglected.

2.9 Head Related Transfer Functions

The Head Related Transfer Function (HRTF) is used to represent the acoustic effect related to the presence of a listener's head and ears. The HRTF is defined as the ratio between the spectral[2] sound pressure $p_{ears}(f)$ at the ears of the listener, and the spectral sound pressure $p_{absent}(f)$ measured in a position between the ears, in the absence of the listener [24]. Mathematically, the HRTF for a frequency f is expressed as:

$$HRTF(f) = \frac{p_{ears}(f)}{p_{absent}(f)} \ . \tag{2.19}$$

The HRTF can further be expressed in decibels as an offset of the absolute sound pressure value, relating to the squared pressures. In this context, the logarithmic expression for the HRTF is:

$$HRTF(f) = 10\log_{10}\left(\frac{p^2_{ears}(f)}{p^2_{absent}(f)}\right) \text{ [dB]}. \tag{2.20}$$

2.10 Determining SPL Contributions in a Non-anechoic Room

In a non-anechoic room, the total SPL measured at a persons' ears, $L_{p,ears,room}$, consists of SPL components from the direct sound, $L_{p,ears,room,dir}$, and the reflected sound, $L_{p,ears,room,refl}$:

$$L_{p,ears,room} = 10\log_{10}\left(10^{\frac{L_{p,ears,room,dir}}{10}} + 10^{\frac{L_{p,ears,room,refl}}{10}}\right) . \tag{2.21}$$

For a musician playing an instrument with a generated SWL $L_{W,instrument,room}$, applying the specific data for the source, receiver and room into equation 2.2 gives an analytic expression for $L_{p,ears,room}$:

$$L_{p,ears,room} = L_{W,instrument,room} + 10\log_{10}\left(\frac{DF_{instrument}DF_{ears}}{4\pi r^2_{ac}} + \frac{4}{A_{room}}\right) \text{ [dB] re } p_0, \tag{2.22}$$

where the instrument and ears are the source and receiver(s) respectively, and r_{ac} is the distance between the musicians' ears and the acoustic center (centroid) of the instrument, in meters. The frequency-dependent distance r_{ac} can be identified in an anechoic or semi-anechoic room by manipulating equation 2.2 with the data from the anechoic measurements:

$$r_{ac} = \sqrt{10^{\frac{L_{W,instrument,anechoic} - L_{p,ears,anechoic} + \Delta L_{p,HRTF} + \Delta L_{p,DI} - 11}{10}}} \text{ [m]}. \tag{2.23}$$

For easening the separation of the direct and reflected terms in equation 2.22, we introduce two constants D and R:

$$\begin{aligned} D &= \frac{DF_{instrument} \cdot DF_{ears}}{4\pi r^2_{ac}} \ , \\ R &= \frac{4}{A_{room}} \ . \end{aligned} \tag{2.24}$$

The direct and reflected sound contributions at the ears in a non-anechoic room, $L_{p,ears,room,dir}$ and $L_{p,ears,room,refl}$, will each contain the single D and R term, respectively. Using equation 2.22 to express the non-anechoic, instrument SWL as

[2] The Fourier transform of the sound pressure.

$$L_{W,instrument,room} = L_{p,ears,room} - 10\log_{10}(D+R) ,$$

the direct and reflected SPL contributions at the ears can be expressed from the total measured SPL at the ears:

$$\begin{aligned} L_{p,ears,room,dir} &= L_{p,ears,room} + 10\log_{10}\left(\frac{D}{D+R}\right) , \\ L_{p,ears,room,refl} &= L_{p,ears,room} + 10\log_{10}\left(\frac{R}{D+R}\right) . \end{aligned} \tag{2.25}$$

For constants D and R, it is assumed that r_{ac} is found from anechoic or semi-anechoic measurements and that $DF_{instrument}$ and DF_{ears} are known, while R cannot be known directly. If G_{room} is known from separate measurements of the non-anechoic room made with an omnidirectional loudspeaker, R can be estimated by inserting the radiated SWL of the loudspeaker $L_{W,lsp}$ into the SPL terms of equation 2.17. Assuming an omnidirectional measurement microphone and an average measurement distance r_G, G_{room} can be expressed as:

$$G_{room} = 10\log_{10}\left(\frac{1}{4\pi r_G^2} + R\right) - 31\text{dB} . \tag{2.26}$$

By assuming that $r_G >> r_{crit}$ such that the direct contribution of the G_{room} measurements can be ignored, the measured $G_{measured}$ can be approximated as the reflected contribution of the room strength, $G_{room,refl}$:

$$G_{room} \approx G_{measured} = G_{room,refl} = 10\log_{10}(R) - 31\text{dB} . \tag{2.27}$$

The empirical estimate of R can then be expressed as:

$$R = 10^{\frac{G_{measured}-31}{10}} . \tag{2.28}$$

Chapter 3

Method

3.1 Test Procedure

The main objectives of the test can be categorized into a subset of goals given below, concerning the Bb clarinet and the sound levels at the ears of the musician. The parameters of interest for the tests are further explained in Section 3.1.1, while the general outline of the test is introduced in Section 3.1.2.

1. Measuring the SPL at the musicians' ears for practice sessions and the dynamic *forte*, in semi-anechoic chamber and in a practice room,

2. Identifying the A-weighted power-pressure difference, $L_{WA,\text{instrument}} - L_{A,\text{ears}}$, in 1/1 octave bands at the dynamic *forte*, for the direct sound in a semi-anechoic chamber, with a quantified uncertainty,

3. Determining the SPL contributions at the musicians' ears of the direct and the reflected sound, $L_{A,\text{ears,room,dir}}$ and $L_{A,\text{ears,room,refl}}$, in 1/1 octave bands in a practice room.

3.1.1 Parameters of Interest

For the SPL measurements at the musicians' ears during practice sessions, the A-weighted, time-equivalent SPL $L_{A,\text{ears,eqT}}$ was chosen as the main parameter of interest, where T represents the duration in seconds of the entire session or of a specific section. The maximum SPL with frequency weighting C, time-weighting F, $L_{CF,\text{Max}}$ was chosen as the maximum SPL unit for measurements. Relating to the measurements made by O'Brien [1], time equivalent levels were also used for calculation of sound level exposure for an 8 hour work day based on daily practice sessions of 2.1 hours.

The A-weighted SWL and SPL were chosen as main parameters of interest for the direct sound power-pressure difference measured at the musicians' ears, $L_{WA,\text{instrument}} - L_{A,\text{ears}}$, relating to human perception of frequencies as discussed in Section 2.4. For practical music room situations, the obtained $L_{WA,\text{instrument}} - L_{A,\text{ears}}$ will give an expected SPL at a musician's ears from the direct sound, $L_{A,\text{ears}}$, given a typically radiated SWL of the instrument.

The A-weighted levels were also chosen for determining the SPL components at the musicians' ears in a practice room, $L_{A,\text{ears,room,dir}}$ and $L_{A,\text{ears,room,refl}}$. Identifying the contribution from the room requires separate room acoustic measurements, in which G is measured.

3.1.2 Test Outline

The test consisted of two parts; one in a semi-anechoic chamber and one in a regular practice room. For both locations, measurements were made for test signals in the dynamic *forte* and for a structured practice session. A total of 4 clarinetists participated in the study, 2 of which play professionally, whereas the remaining 2 were hobbyists playing in amateur wind bands.

Measurements were made of the SPL at the musicians' ears in both locations for the *forte* test signals, relating to the reference data given in ISO 23591:2021 [3]. In addition, SWL was determined in the semi-anechoic chamber, both for test tones in *forte* and during practice sessions.

For structuring the practice sessions, a procedure similar of that of O'Brien was used [1]. A detailed description of the test material, constituting of test signal scale runs and practice sessions, is given in Section 3.2, while details for the measurement setup, implementation and analysis are given in the following sections. Table 3.1 gives an overview of the measured parameters for the various measurement signals, in each location.

Table 3.1: Overview of measurements and parameters measured in semi-anechoic room and in practice room. "X" marks that a parameter is measured in the given location.

Measurement	**Parameter(s)**	**Semi-anechoic Chamber**	**Practice Room**
Test signal (Scale runs in *forte*)	L_{WA}, $L_{WA} - L_{A,\text{ears}}$	X	-
	$L_{\text{Aeq25s,ears}}$	X	X
Practice session (Warm-up, Music repertoire)	L_{WA}, $L_{WA} - L_{A,\text{ears}}$	X	-
	$L_{A,\text{eqT,ears}}$	X	X
	$L_{CF,\text{Max}}$	X	X
Room Acoustic Parameters	G, T_{20}	-	X

3.2 Music Test Material

In each measurement location, the participants would play individually on separate occassions, a full set of the following musical parts given below. More detailed explanations of the sections are given in the following sections.

- Warm-up (5 minutes, free playing)
- Music piece practice (10 minutes, "Gavotte")
- Music piece practice (10 minutes, "Menuett")

Practice session

- Scale runs in *forte*

Test signal

3.2.1 Practice Session

The warm-up sections and the practice of music repertoire were chosen as test material to represent a normal practice session for a musician. The selected musical pieces, both arrangements for the clarinet from compositions of J. S. Bach, were chosen both due to levels of required technicality as well as tonal and dynamic ranges. By having a variety of dynamics and spreading the notes across more than one octave, the chosen pieces can represent a balanced representation of typical practice material. For both "Gavotte" and "Menuett", the dynamic ranges span from *piano* to *forte*, with tonality spanning over two octaves. In addition, neither of the pieces are particularly fast-paced, such that it would be possible for both professionals and hobbyists to perform a proper practice session for each piece during two 10-minute spans.

3.2.2 Test Signal (Scale runs in *forte*)

For the test signal, or scale runs, the participants were presented with a visually aiding metronome at 70 bpm and asked to play a two-octave major scale in a consistent *forte* as illustrated with musical notation in Figure 3.1. The scale run was played in sustained quarter-notes from the lowest C note of the clarinet, corresponding to a Bb_3 natura with a fundamental frequency of 233 Hz. When reaching the C note two octaves higher (Bb_5 natura, fundamental frequency of 466 Hz), the clarinetists were instructed to take a breathing pause of free duration before repeating the scale in reverse direction, back down to the low C.
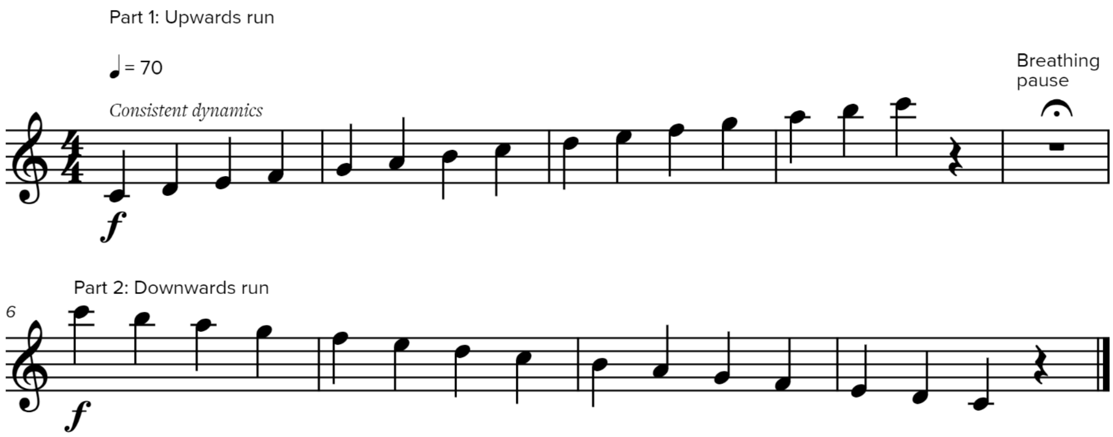
Figure 3.1: Scale-runs in *forte*, notation for sound power measurements. The lowest note, C, played on a Bb clarinet corresponds to a Bb_3 natura with fundamental frequency of 233 Hz.

For analysis purposes, the pause between the upwards and downwards runs was removed from the measurements. The combination of the up- and downward scale runs was finally defined as one single measurement, with a total duration of 25 seconds. The participants each did several scale run measurements for different settings in the two measurements locations, as described in the following sections.

3.3 Sound Pressure Level Measurements

3.3.1 Microphone Positions

A total of 4 microphones were used to measure the averaged sound pressure level at or close to the musicians, as listed in Table 3.2. The same setup of microphones was used for both the semi-anechoic chamber and the practice room. Two lavalier microphones were mounted outside the ear canals in a fixed position, one on each ear, as shown in Figure 3.2.

Table 3.2: Microphones used for sound pressure level measurements, with respective positions. All microphones are omnidirectional.

Microphone	**Position**
Lavalier, Ear-Mounted	Left Ear
Lavalier, Ear-Mounted	Right Ear
Free-Field, Condenser	Clip-On, Instrument Mounted, 12 cm from instrument body
Free-Field, Condenser	1.50 meter horizontal distance from musician, 1.20 meters above floor (eye-height)

Figure 3.2: "On-ear" mounted lavalier microphone.

An omnidirectional microphone was placed 1.50 meters in front of the musician, at 1.20 meters above the floor, approximating eye level height for a sitting position. The horizontal distance was measured from the capsule of the microphone to the hand position on the clarinet. The choice of a 1.50 meter distance was made for direct comparison with the measurements of O'Brien [1]. Moreover, an omnidirectional "clip-on" microphone was mounted directly on the lower part of the instrument, measuring the sound pressure levels at an approximately 12 cm distance from the body of the clarinet. The microphone was positioned to receive sound radiating both from the bell and the tone holes. Figure 3.3 shows the clip-on microphone construction and the 1.5 meter reference microphone.

Figure 3.3: Left: Clip-on microphone, mounted on instrument in approximately 12 cm distance from clarinet body. Right: Reference microphone placed at 1.5 meter horizontal distance from musician.

3.3.2 Signal Chains

The microphones were mounted and connected using signal paths as illustrated with simplified block diagrams in Figure 3.4. The ear-mounted lavalier microphones and the instrument-mounted "clip-on" microphone (i) made use of a direct connection between an audio interface with integrated 48 Volt supply, while the 1.50 meter reference microphone (ii) used an external signal conditioner. A stationary audio interface connected to a computer with a recording software was used in the semi-anechoic chamber, while a portable recording interface was used for the practice room. Both interfaces featured built-in analog-to-digital converters (ADC), using a sample rate $f_s = 48$ kHz. A detailed list of equipment is given in Appendix F, while a detailed overview of used microphone signal chains can be found in Appendix G.

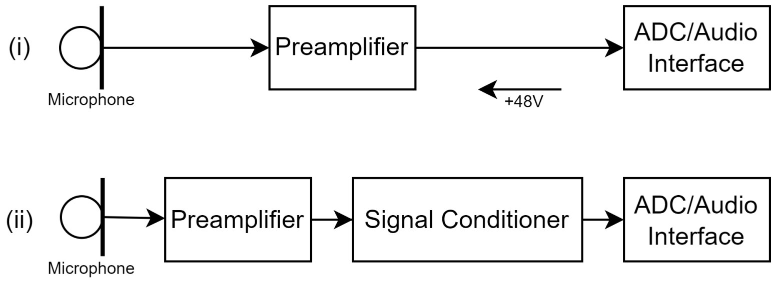

Figure 3.4: Block diagrams of microphone signal chains for SPL measurements. (i): Signal chain for ear mounted lavalier microphones and instrument-mounted clip-on microphone. (ii): Signal chain for reference microphone at 1.50 horizontal distance.

3.3.3 SPL Analysis

The SPL was measured for all 4 microphones for each section of the music test material presented in Section 3.2, in both locations. Before each measurement session, the microphones were calibrated according to the method which will be discussed in Section 3.6.

For the *forte* scale run test signals, the time-equivalent SPL $L_{A,\mathrm{eq25s}}$ was determined for each complete run, using the calculation methods for time and -frequency weighting described in Sections 2.2 and 2.4. The measurements were repeated n times per musician, performed as single number ratings and in 1/1 octave bands. In the semi-anechoic chamber, the $L_{A,\mathrm{eq25s}}$ measurements were made simoultaneously with measurements of the SWL L_{WA}, further explained in Section 3.4.3, resulting in $n = 8$ measurement repetitions for each musician due to source rotations in fixed degree increments. For the practice room, the number of repeated measurements of $L_{A,\mathrm{eq25s}}$ per musician was $n = 3$, having the musicians seated in a fixed position.

For the practice session sections, separate measurements were made for each of the three musical sections presented in 3.2.1 - a 5 minute free warm-up section, along with smaller practice sessions of the two pieces "Gavotte" and "Menuett", with 10 minutes spent for each. The time averaged SPLs $L_{A,\mathrm{eq5min}}$ and $L_{A,\mathrm{eq10min}}$ were respectively measured for the warm-up and for the music pieces, along with the maximum SPL, $L_{CF,\mathrm{Max}}$, according to the calculation method described in Section 2.5. In addition, 8 hour exposure levels $L_{A,\mathrm{EX8h}}$ were calculated from the time-equivalent SPL at the loudest ear, using Equation 2.8. The daily exposure level calculations assumed an average 2.1 hour duration of daily practice sessions, with the same time-equivalent intensity as the measured 25 minute practice session, such that $L_{A,\mathrm{eq2.1h}} = L_{A,\mathrm{eq25min}}$, relating to the results published by O'Brien [1].

The analysis of measurements were done with a combination of spreadsheets and analysis code written in Python. The latter are available in Appendix H, presented as separate scripts.

3.4 Semi-Anechoic Chamber Measurements

The first part of the test took place in a semi-anechoic chamber, for studying the direct signal from the clarinet. This section will explain the details of the measurement setup, room layout and preparation, as well as a short discussion on the choice of using a semi-anechoic chamber.

3.4.1 Measurement Setup

The chamber used for the measurements had a volume of 106 m^3, with ceiling height of 3.6 meters. 24.5 m^2 of reflective chipboards were placed on the floor in an anechoic chamber, thus making it semi-anechoic. The boards were installed as tightly as possible, reducing any gaps and slits inbetween.

A chair was mounted in a fixed position on the floor, where the musicians would be seated for the entirity of the measurements. 4 microphones for SPL measurements were installed, as described in Section 3.3. In addition, an array consisting of 11 microphones (named M1-M11) were used to measure the generated SWL from the instrument. Figure 3.5 illustrates the positions of microphones M1-M11 from a top and side view, while the exact microphone positions can be found in Appendix E.

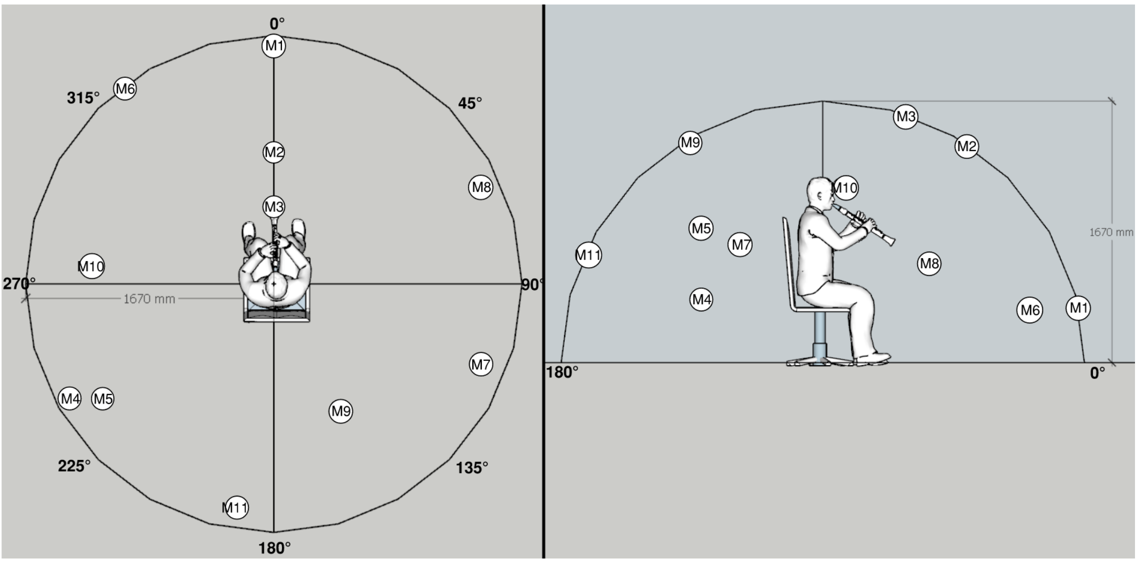

Figure 3.5: Visualization of microphone positions M1-M11 in hemispherical measurement surface, in semi-anechoic environment from top view (left) and side view (right). Top view shows angular indications around the player, used for source rotations.

Six of the microphones in the array were mounted using regular microphone stands on the floor, while the remaining 5 were mounted on horizontal metal rods cut to customized lengths. The rods themselves were mounted to vertical track poles and stabilized using ropes. Absorptive foam was placed on the track poles where possible, to reduce the amount of reflections and edge diffraction effects. All 11 microphones were positioned on a hemispherical surface, in a radius of 1.67 meters from the floor center point beneath the chair. Figure 3.6 shows a photo from the semi-anechoic chamber with the described setup, while further details of the SWL measurements are discussed in Section 3.4.3. A detailed list of the equipment used can be found in Appendix F.

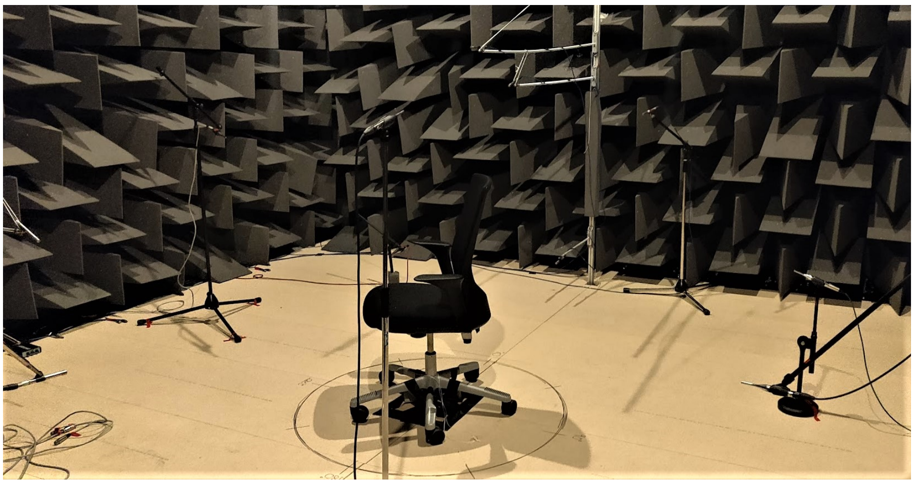

Figure 3.6: Photo from the semi-anechoic chamber with the described setup for sound power level measurements.

3.4.2 Measurement Procedure

Measurements were made of SPL and SWL for the test signal and practice session presented in Section 3.2, for each musician. SPL parameters were measured and calculated as described in Section 3.3.3, while the technical method used for SWL measurements is presented in Section 3.4.3 below. The test signal scale runs described in Section 3.2.2 were measured $n = 8$ times for each musician, with increasing rotation angles of the player. A set of guide marks were drawn on the floor, indicating angles with 45 degree increments around the player, from 0 to 315 degrees, as illustrated in the top view in Figure 3.5. The musicians were asked to perform the complete scale run while facing in the 0 degree direction, then rotate their position to face the 45 degree mark and repeat the measurement, and so on until they have completed a full circle. The microphone array remained stationary between the rotations. Performing this procedure for all 4 musicians, the total number of test signal measurements in semi-anechoic chamber resulted in $n = 32$.

For the practice sessions, measurements were made of SPL and SWL while the musicians conducted the musical sections as described in Section 3.2.1. Each musical section was performed and measured once for each musician, facing the 0 degree direction for the entirity of the session.

3.4.3 Sound Power Measurements

The sound power level was measured using the method provided in NS-EN ISO 3744:2010 [4], with the calculation procedure summarized in Section 2.6.1.

A hemispherical measurement surface with a radius of 1.67 meters was used for the array microphones M1-M11, using the center point on the floor beneath the chair as a reference point for the source. The microphone positions are largely based[1] on positions 1-10 and 18 from the key positions given in ISO 3744:2010, which can be found in Appendix E. The microphones were placed in such a way to avoid symmetrical axis positions, where possible.

In defining the characteristic source dimension, d_O, an approximation was necessary due to the nature of the sound source. The methods in ISO 3744 are based on centering the measurement surface around a point on the floor. However, as the musician is playing in a sitting position, the acoustic center of the instrument will exist somewhere above the floor, at a horizontal distance in front of the musician. The average hand position of the musicians was measured to be 83.5 cm vertically above the floor, and 30 cm horizontally in front of the center axis at the chair base. As the acoustic center position of the Bb clarinet is non-stationary, it was chosen to use the vertical distance as the d_O, such that $d_O = 83.5$ cm, resulting in a surface radius of $r = 2d_O = 1.67$m. In addition, the dimensions of the surrounding room restricted the maximum radius of the measurement surface, making it impractical to choose a larger d_O. Aiming to reduce the error originating from the acoustic center position, it was chosen to perform repeated measurements of the test signal scale runs, with rotations of the player. For the pracice session sections, measurements were however only made with the players facing the 0 degree direction.

For each test signal rotation and for each practice session section, the A-weighted SPL L_A was measured at each microphone in the heimspherical array. At the end of the sessions, the background noise was measured using the same microphone array to check for correction needs. The A-weighted SWL, L_{WA}, was then calculated for each test signal rotation and for each practice session section, using the calculation method given in Section 2.6.1.

The Python code written to determine the SWL, SPL and power-pressure difference from measurements can be found in Appendix H.11.

[1] The positions of the microphone array could not be absolutely based on the provided key positions given in ISO 3744. This was mainly due to lack of sufficient space, as well as the available number of suitable microphone stands. As 5 of the microphones were mounted using horizontal rods, restricting the positioning of those microphones, it was instead chosen to use an appoximation of the positions in ISO 3744.

3.4.4 Sound Power Level Uncertainty

For the determination of the sound power level uncertainty $u(L_W)$, an approach based on the model described in Section 2.6.2 is used. It is assumed that by performing measurements of clarinetists in a number of rotation angles, where the players are all instructed to produce the same signal (scale-run in *forte*, 70bpm over 25s), once for every musician in every rotation angle, the standard devation $\sigma_{L_W,\mathrm{measured}}$ in decibels derived directly from the measured results will contain both standard devation components σ_{R0} and σ_{omc}. By this method, it is assumed that:

$$u(L_W) \approx \sigma_{tot} \approx \sigma_{L_W,\mathrm{measured}}\ [\mathrm{dB}]\ . \tag{3.1}$$

3.4.5 SWL Signal Chains

The microphones used for SWL measurements were either free-field capsules, mounted on preamplifiers and connected through signal conditioner amplifiers, or +48V driven free-field microphones. Figure 3.7 illustrates the simplified signal chains for the relevant microphone groups. Both types of chains are assumed to be sufficiently linear and frequency transparent for the purpose, such that the difference is assumed negligible. A detailed overview of the microphone signal chains used can be found in Appendix G, while model specifics is given in the equipment list in Appendix F.

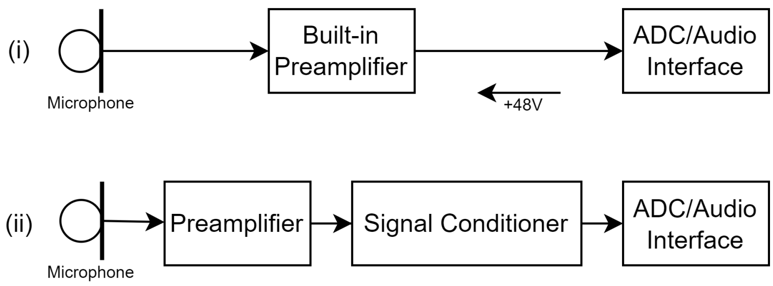

Figure 3.7: Block diagrams of microphone signal chains for SWL measurements. Signal chains for microphones (i): M6-M10, and (ii): M1-M5, M11.

The signals were recorded into a stationary audio interface connected to a computer, to WAV-format. The recorded files were sampled, quantized and stored with a bit depth of 24 bit and a sample rate f_s = 48 kHz. All microphones were calibrated before the start of each measurement session, using the procedure described in Section 3.6.

3.4.6 Semi-anechoic vs. Anechoic

A semi-anechoic chamber is by definition an anechoic chamber with a reflective floor surface. This can be referred to as a "free-field environment over a reflective surface", which emulates an outdoor environment. By removing the reflections of the walls and ceiling, the contribution of the direct sound from the chosen sound source is partly isolated for the measurements, with the contribution of the floor reflections.

The reasoning for choosing a semi-anechoic environment for the test, instead of anechoic, was threefold:

1. Using a free field environment over a reflective plane allows for the use of NS-EN ISO 3744:2010 for measuring sound power.
2. Musicians will always sit or stand above some horizontal surface (floor/ground), hence the complete sound source was defined to include floor reflections.

3. Contribution symmetry: The main parameter of interest was the A-weighted power-pressure difference, $L_{WA,\text{instrument}} - L_{A,\text{ears}}$. As the ISO 3744 method described in Section 2.6.1 calculates L_{WA} from averaged measurements of L_A, contributions from floor reflections will be present in both terms. Hence the error source originating from floor reflections can be assumed to be reduced for the power-pressure difference.

3.5 Practice Room Measurements

The second location used for measurements was a practice room, representative of a daily practice environment for the involved musicians. This section describes the procedure and setup used for the field measurements in the room.

3.5.1 Room Layout

The room in which the measurements took place had a floor area of 8.60 m^2 with a height of 2.52 meters, resulting in a volume of 21.7 m^3. The room had a geometry based on a classic "shoebox" shape, with the exception of a curvature in one corner. Figure 3.8 shows an illustration of the rehearsal room, along with dimensions and relevant acoustic objects. The position of the musician and a reference microphone is also shown, which will be discussed further in Section 3.5.2.

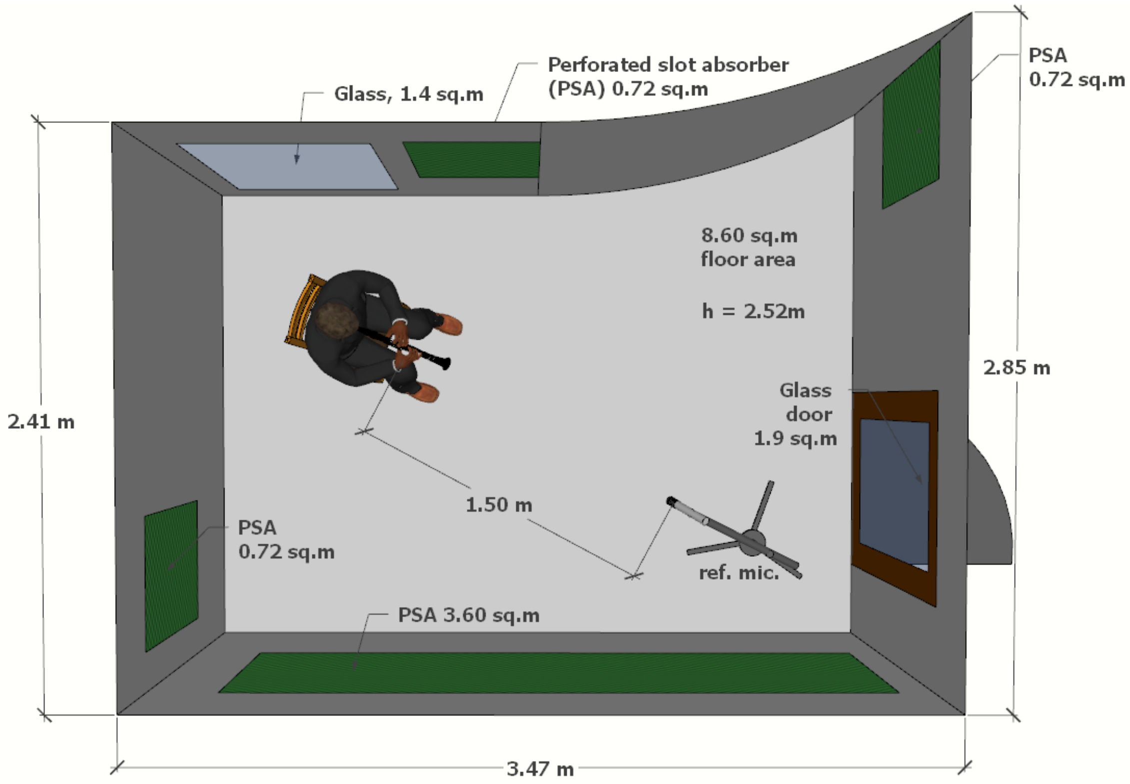

Figure 3.8: Illustration of rehearsal room used for field measurements. Existing perforated slot absorbers (PSA) previously installed in the room are marked in green.

The room walls were made of gypsum, with a glass window and door positioned as shown in the figure above. No absorbers were mounted in the ceiling, leaving a reflective surface resembling painted gypsum, while the flooring consisted of vinyl.

On a daily basis, this room was usually being used by wind band professionals. Thus some acoustic treatment had been installed at an earlier time by the owners of the building. Hanged on the walls, there was pre-installed a total of 5.8 m^2 perforated slot absorbers (PSA) made of gypsum, with a perforation grade of approximately 15%.

3.5.2 SPL Measurements

Measurements were made of SPL for the test signal and practice session presented in Section 3.2, for each musician, using the setup and procedure described in 3.3. The SPL parameters were measured and calculated as described in Section 3.3.3. For the test signal scale runs in *forte*, the musicians were asked to repeat the scale runs 3 times, resulting in $n = 3$ test signal measurements per musician. The practice session sections were performed and measured once per player. All measurements were made with the musicians facing the same direction, seated in the same position throughout the entire measurement session.

The analysis of measurements were done with a combination of spreadsheets and analysis code written in Python. The latter are available in Appendix H, presented as separate scripts.

3.5.3 Room Acoustic Measurements

To identify the SPL contribution from the room, there were made room acoustic measuments of the practice room on a separate occasion, to determine the reverberation, T_{20}, and strength, G, in 1/1 octave bands. The measurements were made using impulse response (IR) measurements with a pre-calibrated setup, including an omnidirectional speaker, a free-field omnidirectional microphone and the room acoustic analysis software ODEON [25]. The chair was removed prior to the measurements and no persons were present in the room during the procedure. Figure 3.9 illustrates the signal chain of the measurement setup used for room acoustic measurements. Information about model specifics of the used equipment can be found in Appendix F.

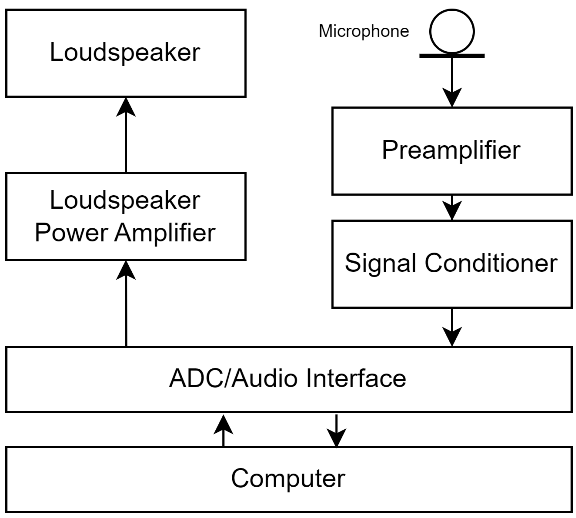

Figure 3.9: Block diagram of signal chain for measurements of reverberation, T_{20}, and strength, G. Microphone and loudspeaker are both omnidirectional.

A total set of 6 measurements was done in the practice room, using 3 receiver positions for each of 2 source positions. The positions of the omnidirectional loudspeaker were chosen such that any symmetrical distances to the room surfaces was avoided, both in between the axes and between the two source positions. A sinusoidal sweep of 8.0 seconds ranging from 16Hz to 16kHz was played from the software through an audio interface, sending the signal further through a power amplifier connected to the loudspeaker. The measurement microphone was similarly positioned to avoid distance symmetry between axes, other receiver positions and relative to the source. Aiming to keep the receiver positions in diffuse field, a distance between the source and receiver of $r_{SR} > 1.0$

meters was used. Using a guessed value of $T = 0.5$ seconds, the critical radius was estimated to $r_{crit} = 0.37$ meters, using equations 2.2 and 2.14 with an omnidirectional source and receiver. It was therefore assumed that $r_{SR} > 1.0$m would fulfill $r_{SR} >> r_{crit}$. Figure 3.10 shows a picture from the practice room during the room acoustic measurements, using the omnidirectional loudspeaker and microphone.

Figure 3.10: Photo of room acoustic measurements in practice room, using pre-calibrated IR setup with omnidirectional microphone and loudspeaker.

The full measurement setup was calibrated in a reverberation chamber at an earlier date, to ensure measurement of absolute pressure values, necessary for obtaining G. The calibration procedure is based on creating a calibration file from measurements in diffuse field, which is later matched up against reference measurements in situ at a specified distance. A full description of the calibration procedure can be found in [26], referred to by Odeon as the "2-step calibration procedure".

3.5.4 Determination of Sound Level Contributions at Ears

The SPL contributions in the practice room at the musicians' ears of the direct and the reflected sound, $L_{A,\text{ears,dir}}$ and $L_{A,\text{ears,refl}}$, were calculated in 1/1 octave bands using the method described in Section 2.10. To calculate the average distance from the musicians' ears to the acoustic center of the instrument r_{ac}, the average power-pressure difference $L_{WA,\text{instrument}} - L_{A,\text{ears}}$ from test signal measurements in the semi-anechoic room was used. The directivity of the instrument was obtained using published reference data, from measurements by Pätynen and Lokki [18], reconstructed as DI curves for octave bands in the range 250-2000 Hz in Figure 1.6. The incidence angle of 135 degrees elevation was used, to approximate the propagation path from the clarinet to the midpoint between the musicians' ears.

For the HRTF, significant variations will exist between players due to differences in head shape and size, ears, shoulders and other anatomic features. It was therefore decided to use average data from a published database, with an incidence angle of the source corresponding to the instrument position. HRTF values from 45 people (90 ears) published by the Center for Imaging Processing and Integrated Computing (CIPIC) [27] were logarithmically averaged for an incidence elevation angle of -45 degrees. Figure 3.11 shows the averaged HRTF curve in 1/1 octave bands applied for the calculations, approximating the offset in decibels relative to the SPL without any head present.

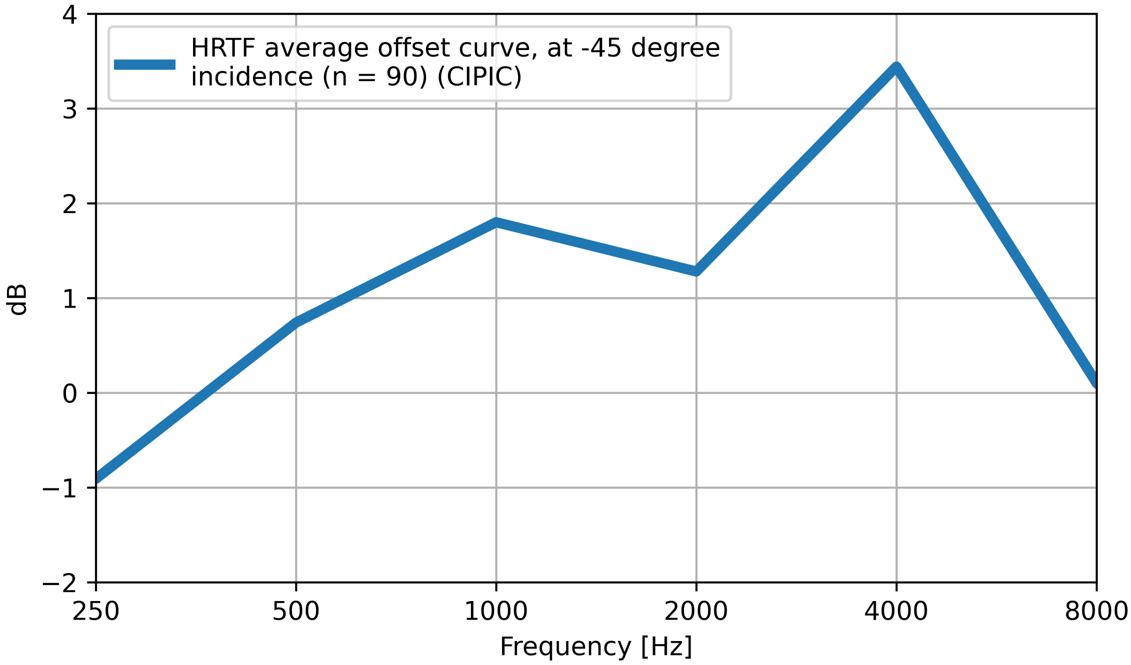

Figure 3.11: HRTF curve at incidence elevation angle of -45 degrees, logarithmically averaged from 45 people (90 ears), in 1/1 octave bands. Reference database published by CIPIC [27].

The directivity values in 1/1 octave bands obtained from the instrument and the HRTF were respectively used for the source and receiver directivities DF_S and DF_R, calculating the constant D in equation 2.24. Measurements of G from the practice room were applied to determine the empirical estimate of the constant R, given in equation 2.28.

For the SPL measured at the musician's ears in the practice room, $L_{A,\text{ears,room}}$, a logarithmic average of $L_{A,\text{ears,eq25s,room}}$ test signal measurements was used ($n = 12$). The SPL contributions $L_{A,\text{ears,room,dir}}$ and $L_{A,\text{ears,room,refl}}$ were finally calculated using equations 2.25.

3.6 Calibration

For obtaining the absolute pressure values, a calibrator was used prior to each measurement session. The calibration was done for both SPL microphones and microphones in the array for SWL measurements. A known sound pressure value of 94 dB re 20μPa was played from the calibrating device and recorded thorugh each single microphone, allowing for the measurements to be scaled accordingly later in the analysis process. The Pytshon script used for calibration can be found in Appendix H.3. Microphone adapters corresponding to the various microphone sizes were used, introducing a loss of approximately -2 dB. The variation in adapter size affected these offsets for the calibration pressure values, which were accounted for in the scaling process. Table 3.3 shows the calibration values used, including known or assumed offsets, for the various microphone capsule sizes.

Table 3.3: Calibration values used for various microphone capsule sizes. The known reference values are listed on the calibrator itself for the specific capsule sizes.

Microphone adapter size	**Microphones**	**dB value (re 20μPa)**	**Reference**
1/2”	M1-M5, M11, Clip-on, 1.5m ref	92.1	Known
1/4”	M6-M10	91.9	Known
1/8”	Left, Right	91.9	Assumed

Chapter 4

Results

This chapter reports results found from implementing the methods given in Chapter 3.

4.1 Sound Pressure Levels Measured at Ears

This section presents the SPL measurements from practice sessions in the semi-anechoic chamber and in the practice room.

4.1.1 Time Averaged Levels

The equivalent, A-weighted sound pressure level $L_{A,\mathrm{eqT}}$ was recorded in both measurement locations for the given sections of the structured practice sessions. Table 4.1 shows the measured SPL at ears and at 1.5 meter distance for warm-up section and musical pieces "Menuett" and "Gavotte", averaged between musicians (n=4), with 95% confidence intervals assuming a normal distribution. Time interval T is indicated for each section. The measured SPL per musician during each section can be found in Appendix A.

Table 4.1: Measured SPL $L_{A,\mathrm{eqT}}$ at ears and at 1.5 meter distance, in semi-anechoic chamber and in practice room. Values averaged between musicians (n = 4), with 95 % normal distribution confidence intervals.

Practice Session Section	**Semi-anechoic chamber**				**Practice room**			
	Left Ear $L_{A,\mathrm{eqT}}$ [dB] re p_0	Right Ear $L_{A,\mathrm{eqT}}$ [dB] re p_0	Ears Avg. $L_{A,\mathrm{eqT}}$ [dB] re p_0	1.5m Ref. $L_{A,\mathrm{eqT}}$ [dB] re p_0	Left Ear $L_{A,\mathrm{eqT}}$ [dB] re p_0	Right Ear $L_{A,\mathrm{eqT}}$ [dB] re p_0	Ears Avg. $L_{A,\mathrm{eqT}}$ [dB] re p_0	1.5m Ref. $L_{A,\mathrm{eqT}}$ [dB] re p_0
Warm-up (T = 5min)	89 ± 10.3	89 ± 9.3	89 ± 9.6	81 ± 8.7	91 ± 9.2	92 ± 9.6	92 ± 9.4	87 ± 9.8
"Menuett" (T = 10min)	88 ± 6.3	89 ± 5.4	88 ± 5.4	81 ± 4.7	90 ± 9.2	91 ± 8.7	91 ± 8.8	87 ± 8.4
"Gavotte" (T = 10min)	86 ± 8.9	87 ± 7.4	87 ± 8.3	79 ± 6.1	89 ± 9.4	89 ± 8.7	90 ± 9.4	84 ± 7.9

4.1.2 Time Equivalent- and Daily Exposure Levels

The daily exposure levels $L_{A,\mathrm{EX8h}}$ were calculated based on a 2.1 hour practice session per 8 hour work day, relating to the results published by O'Brien. The 2.1 hour equivalent $L_{A,\mathrm{eq2.1h}}$ was calculated by determining the equivalent level $L_{A,\mathrm{eq25min}}$ at the loudest ear for the entire practice session, and assuming a constant average level throughout 2.1 hours, i.e. $L_{A,\mathrm{eq2.1h}} = L_{A,\mathrm{eq25min}}$.

Table 4.2 shows equivalent levels from averaged exposure from measurements compared to the data for Bb clarinet published by O'Brien in [1].

Table 4.2: Averaged sound exposure from practice sessions, in A-weighted equivalent levels over duration of sessions. Direct comparison with data from Table I in published work of O'Brien [1], including 8-hour equivalent level estimated from an average daily 2.1 hour practice session, and inter-aural difference between ears in decibels. Levels marked (*) are 23-minute equivalent levels from O'Brien's measurements.

Measurement Location/Origin	Left Ear $L_{A,\mathrm{eq25min}}$ [dB] re p_0	Right Ear $L_{A,\mathrm{eq25min}}$ [dB] re p_0	1.5 m ref. $L_{A,\mathrm{eq25min}}$ [dB] re p_0	Estimated exposure after 2.1 h $L_{A,\mathrm{EX8h}}$ [dB] re p_0	Inter-aural difference [dB]
Semi-anechoic chamber	87	88	80	82	1
Practice Room	90	91	86	85	1
O'Brien (see [1])	92*	92*	86*	86	0

4.1.3 Maximum Levels

The maximum C-weighted, "Fast" time weighted levels $L_{CF\mathrm{Max}}$ from the entire practice sessions were obtained in both locations. For each musician, the maximum ear level throughout all sections of the practice session was registered. Table 4.3 shows the average maximum level $L_{CF\mathrm{Max}}$ between the players, with 95% confidence intervals assuming a normal distribution and standard devations in decibels (n = 4). The average of inter-aural differences between maximum levels are also provided in decibels, with confidence intervals and standard deviations. Values of $L_{CF\mathrm{Max}}$ for each musician in every practice session section can be found in Appendix B.

Table 4.3: Average of maximum levels $L_{CF\mathrm{Max}}$ between players from practice sessions, with average of inter-aural differences for maximum levels, both with 95% normal distribution confidence intervals and standard deviations in decibels (n = 4).

Measurement Location	$L_{CF\mathrm{Max}}$ [dB] re p_0	$L_{CF\mathrm{Max}}$ SD [dB]	Inter-aural diff. [dB]	Inter-aural diff SD [dB]
Semi-anechoic chamber	103 ± 5.3	3.3	1.9 ± 2.2	1.4
Practice room	106 ± 7.6	4.8	2.4 ± 1.8	1.1

The maximum C-weighted peak level $L_{C\mathrm{peak}}$ at the ears during practice sessions was measured to 115 dB re p_0 in the practice room, and 111 dB re p_0 in the semi-anechoic chamber. For comparison, the maximum $L_{C\mathrm{peak}}$ found in a practice room for Bb clarinet by O'Brien was 111 dB re p_0 [1].

4.2 Power-Pressure Difference

The measurement setup was installed in the semi-anechoic laboratory as described in Section 3.4.1. The following sections present the results from measurements of sound power and pressure for the various parts of the test procedure.

4.2.1 Test Signal in *forte*

Measurements were conducted for the test signals in *forte*, of SPL and SWL in the semi-anechoic chamber and of SPL in the practice room. Regarding dynamics, the musicians were asked to perform the runs with as much dynamic consistency as possible. Prior to the measurement, the musicians were informed on a general basis that the subjective experience of a semi-anechoic room would differ from their daily rehearsal room environments. They were however asked to trust the "feel" of the instrument when choosing the dynamic for their *forte*. No further instructions were given regarding the dynamics.

Table 4.4 shows the arithmetic averages and standard deviations (SD) of 25 second scale-run measurements of time-equivalent, A-weighted sound pressure levels in both locations, in addition to the sound power level and power-pressure difference, in dB, from the semi-anechoic chamber. The 95% confidence intervals (CI) are also stated with the averaged values, assuming a normal distribution. For the power-pressure difference, a logarithmic average is used between the left and right ear levels for $L_{A,\text{eq25s,ears}}$. The background noise was measured to $L_{A,\text{background noise}} = 17$ dB re p_0, meaning that no corrections were necessary for determining the SWL. Data have been added for SWL from Meyer [10], and for SPL at the musicians' ears from O'Brien [1] for comparison. An overview of the measured sound power, ear pressure levels and the power-pressure differences measured in the semi-anechoic chamber for each musician during the test signals is available in Appendix C.

Table 4.4: Arithmetically averaged dB values with 95% confidence intervals and standard deviations from measurements of test signal in *forte*; A-weighted sound pressure levels at ears, $L_{A,\text{eq25s,ears}}$, in semi-anechoic chamber and practice room, as well as sound power L_{WA}, and $L_{WA} - L_{A,\text{eq25s,ears}}$ difference for scale runs in *forte*, in semi-anechoic chamber. Semi-anechoic values are averaged from measurements of each musician in 45 degree increments in source rotation (m=32), while practice room values are averaged from 3 measurements of each musician in a single position (m=12). Confidence intervals assume a normal distribution. (*) Reference data added from Meyer (SWL) with unweighted level [10], and (**) from O'Brien (SPL at ears) for 15 second test tones in *fortissimo (ff)*.

Measurement Location/ Origin	**SWL** L_{WA}		**Left Ear** $L_{A,\text{eq25s}}$		**Right Ear** $L_{A,\text{eq25s}}$		**Power-pressure diff.** $L_{WA} - L_{A,\text{eq25s,ears}}$	
	Avg. ± CI [dB] re 1pW	SD [dB]	Avg. ± CI [dB] re p_0	SD [dB]	Avg. ± CI [dB] re p_0	SD [dB]	Avg. ± CI [dB]	SD [dB]
Semi-anechoic (f) (m=32)	95 ± 0.6	1.7	91 ± 0.9	2.6	92 ± 0.8	2.1	3.9 ± 0.3	0.8
Practice room (f) (m=12)	-	-	94 ± 2.0	3.2	93 ± 2.3	3.8	-	-
Meyer (f)	93*	-	-	-	-	-	-	-
O'Brien (ff)	-	-	97**	-	96**	-	-	-

The measured SPL, SWL and power-pressure differences were also determined in 1/1 octave bands. As the lowest note played during the test signals corresponds to a fundamental tone $f = 233$ Hz, the bands below 250 Hz are ignored. Figures 4.1 and 4.2 show the determined SWL, L_{WA}, and SPL, $L_{A\text{eq25s,ears}}$, calculated in 1/1 octave bands from arithmetic averages of all test signal measurements (m=32). The ear level $L_{A\text{eq25s,ears}}$ is calculated from averages between the left and right ears of the players, as the levels were found to be similar for the test signal measurements (median interaural difference 0.9 dB). Figure 4.3 shows the corresponding arithmetic average of power-pressure differences, $L_{WA} - L_{A\text{eq25s,ears}}$, in 1/1 octave bands. All figures show the distribution of the underlying data illustrated as box plots, where the middle and outer quartiles are represented with boxes and whiskers, respectively. The median within each octave band is marked with an orange line, while outliers are illustrated as circles.

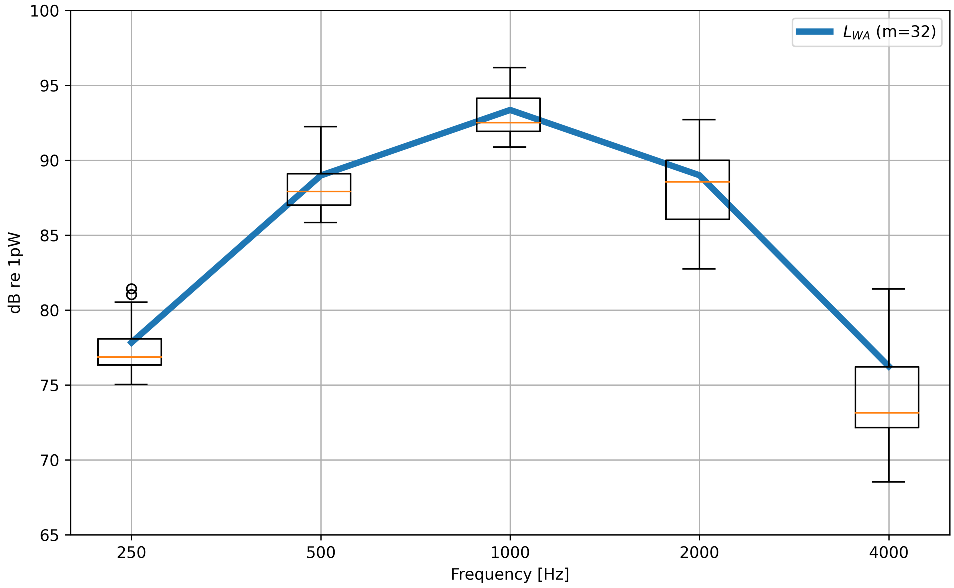

Figure 4.1: L_{WA} A-weighted octave band levels, for scale runs in *forte*, in semi-anechoic laboratory. Values are averaged from measurements in 45 degree increments in source rotation (m=32). Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (orange line) and outliers (circles).

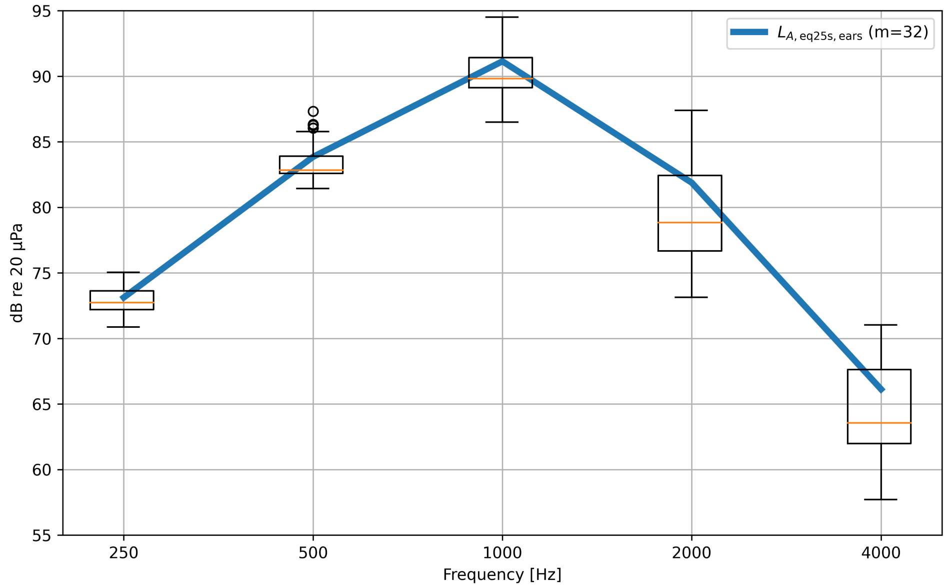

Figure 4.2: $L_{A\mathrm{eq25s,ears}}$ A-weighted octave band levels at ears, for scale runs in *forte*, in semi-anechoic laboratory. Values are averaged from measurements in 45 degree increments in source rotation (m=32). Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (orange line) and outliers (circles).

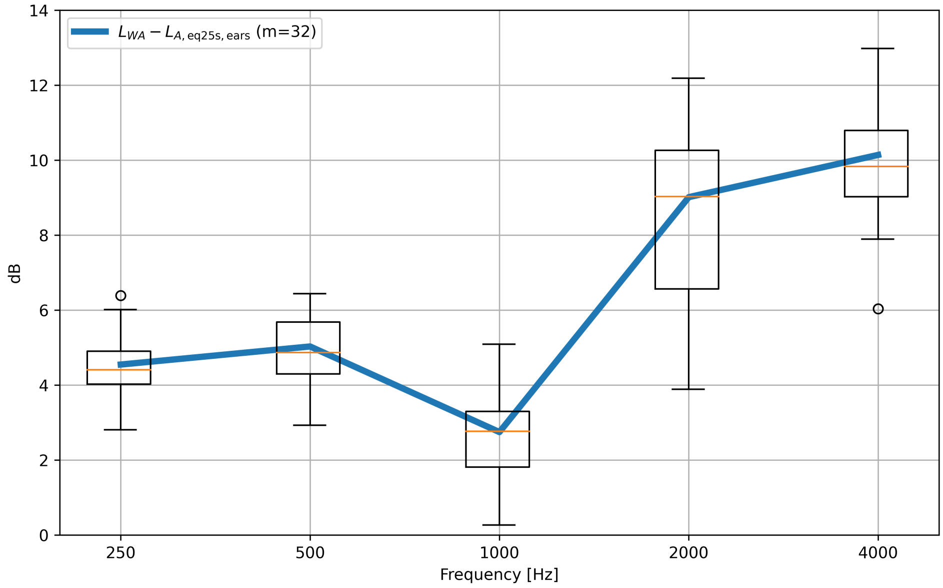

Figure 4.3: Arithmetic average of difference between power and ear pressure levels ($L_{WA} - L_{A\mathrm{eq25s,ears}}$) in 1/1 octave bands, for scale runs in *forte*, in semi-anechoic laboratory (m=32). Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (orange line) and outliers (circles).

Figure 4.3 shows that the determined power-pressure difference is close to 5 dB in the 250 and 500 Hz bands, before dropping by 2 dB in the 1000 Hz band and then increasing significantly for higher frequencies. From the sample variance given by the box plots, the power-pressure difference however remains the most stable for the 250 and 500 Hz bands. Table 4.5 shows the same power-pressure difference as arithmetic average values, with 95% confidence intervals assuming a normal distribution, in decibels (m = 32).

Table 4.5: Arithmetic averages of octave band values and single number ratings of A-weighted power-pressure difference for direct sound at ears $L_{WA} - L_{A\mathrm{eq25s,ears}}$, sound power level L_{WA} and direct sound pressure level at ears $L_{A\mathrm{eq25s,ears}}$, with 95% confidence intervals (m = 32). Measurements of test signal in *forte* in semi-anechoic chamber.

Parameter	**Octave band center frequency**					**Average single number rating**
	250	500	1000	2000	4000	
$L_{WA} - L_{A\mathrm{eq25s,ears}}$ [dB]	4.5 ± 0.3	5.0 ± 0.3	2.7 ± 0.4	9.0 ± 0.8	10.1 ± 0.5	3.9 ± 0.3
L_{WA} [dB] re 1pW	78 ± 0.7	89 ± 0.7	93 ± 0.5	89 ± 1.0	76 ± 1.4	95 ± 0.6
$L_{A\mathrm{eq25s,ears}}$ [dB] re p_0	73 ± 0.4	84 ± 0.6	91 ± 0.8	82 ± 1.6	66 ± 1.3	92 ± 0.8

4.2.2 Practice Session, Single Rotation Angle Measurements

The SPL measurements from practice sessions in the semi-anechoic chamber were analyzed with time weighting F ("Fast"), in order to determine the variation of the average between both ear levels, produced by the clarinet while producing sound. The practice session measurements in the semi-anechoic chamber were measured in a single rotation angle, with the musician facing forward in the 0 degree direction, without any repetitions in other source rotation angles. To avoid influence by rests, silent parts were gated out. This was done by determining the radiated SWL L_{WAF} for each 0.125s time window, and removing all windows where $L_{WAF} < 65$ dB re 1pW before calculating an average value. Table 4.6 shows the A-weighted, "Fast" time weighted SPL

$L_{AF,\text{avg,gated}}$ at both ears, with the corresponding SWL $L_{WAF,\text{avg,gated}}$ value for each musician, logarithmically averaged from the entire practice sessions (25 min) and with standard deviations (SD) in decibels. The SD shows the variation of the levels throughout the entirity of the practice sessions. Values for each musician from each section of the practice session can be found in Appendix D, while the ungated, time-equivalent sound pressure levels can be found in Appendix A.

Table 4.6: Logarithmic average levels from entire practice session in semi-anechoic chamber, without rests; A-weighted, "Fast" time weighted SPL $L_{AF,\text{gated}}$ and SWL $L_{AF,\text{avg,gated}}$, averaged from values where $L_{WAF} \geq 65$ dB re 1pW. Bottom row shows the arithmetic average values between musicians, and standard deviations from 0.125s time windows from all musicians.

Person No.	**Left Ear**		**Right Ear**		**Sound Power Level**		**Power-pressure Diff.**	
	$L_{AF,\text{avg,gated}}$ [dB] re p_0	SD [dB]	$L_{AF,\text{avg,gated}}$ [dB] re p_0	SD [dB]	$L_{WAF,\text{avg,gated}}$ [dB] re 1pW	SD [dB]	$L_{WAF,\text{avg,gated}}$ $-L_{AF,\text{ears,avg,gated}}$ [dB]	SD [dB]
1	79	5.8	81	5.8	86	6.4	4.9	1.4
2	78	5.2	81	5.4	85	5.6	5.1	1.3
3	89	6.8	90	7.0	94	7.0	4.9	2.0
4	85	5.7	86	5.7	91	5.9	5.8	2.1
Total	83	7.2	84	6.9	89	7.3	5.2	1.8

The gating of values below $L_{WAF} = 65$ dB re 1pW was necessary to obtain representative results from when the clarinet is played. Figure 4.4 shows an example histogram of the calculated L_{WAF} values for Person 1, ungated, for the total 25 minute duration of the entire practice session in the semi-anechoic chamber. The figure shows a minority of occurances where L_{WAF} is in the immediate area below 65 dB re 1pW, but a 16% occurance peak in the level range around 40 dB re 1pW. The occurances in this lower level range correspond to rests and breathing pauses of the musician, which is why the levels below the occurance dip at 65 dB re 1 pW have been gated out, for the purpose of measuring the variation from dynamics of the instrument.

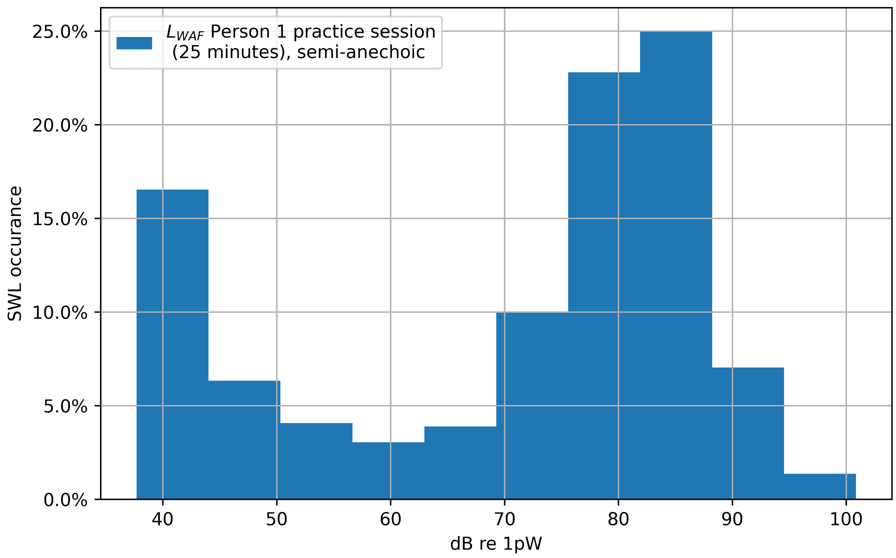

Figure 4.4: Example histogram for L_{WAF} values from entire 25 minute practice session in semi-anechoic chamber, from Person 1. Occurances approximately below 65 dB re 1 pW originate from rests and breathing pauses.

Filtering low levels due to rests and pauses was in addition found to have a stabilizing effect in determining the power-pressure difference. Figure 4.5 shows an example from one of the musicians during the "Gavotte" practice section, of how the $L_{WAF} - L_{AF,\text{ears}}$ difference value over time is affected by becoming less stable for lower values of L_{WAF}. An equivalent power level $L_{WA,\text{background}}$ have additionally been calculated, using equation 2.10 with the measured background noise $L_{A,\text{background}}$ in the semi-anechoic chamber, indicated in the figure.

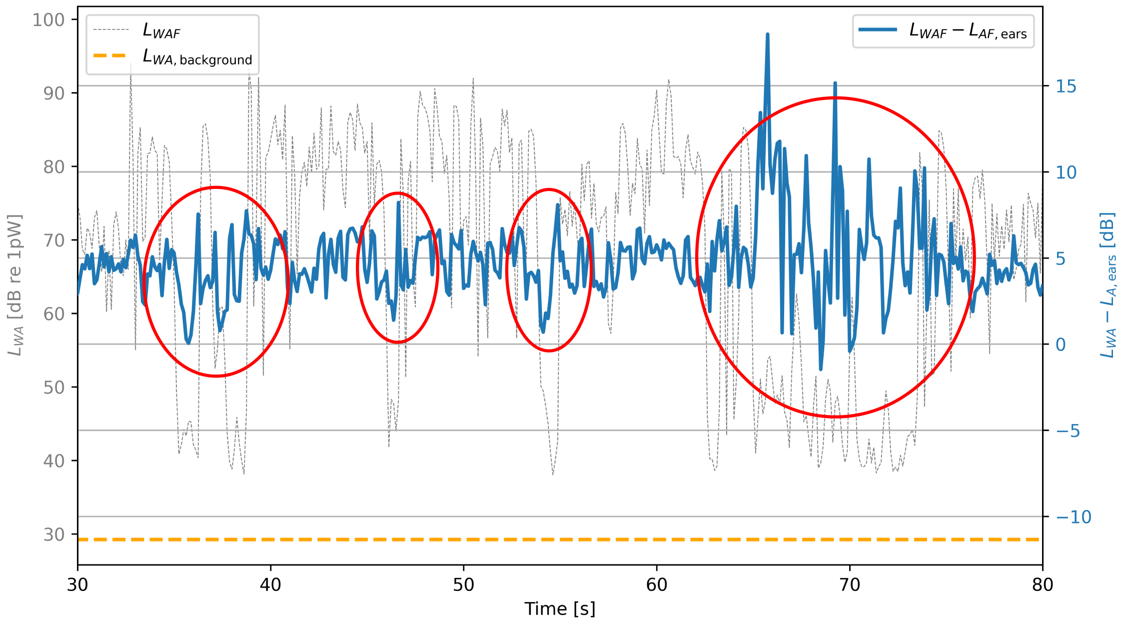

Figure 4.5: Example from Person 1, during practice of music piece "Gavotte". Blue curve and right y-axis shows $L_{WAF} - L_{AF,\text{ears}}$ over time, with the corresponding L_{WAF} value indicated with grey dashed lines and the left y-axis. Background noise equivalent SWL $L_{WA,\text{background}}$ shown with yellow dashed curve. Occurances where $L_{WAF} - L_{AF,\text{ears}}$ is less stable during rests and breathing pauses indicated with red circles.

4.2.3 Comparison of Measurements in Single Angle vs. Full Rotation

A comparison was done between the test signal measurements in *forte* for all source rotations (m = 32) and for the subset of measurements at the front angle at 0 degrees (m$_{front}$ = 4). This was done to help determine the credibility of power-pressure measurements performed for the practice session sections, in the semi-anechoic chamber. A quantification was made of the average measured power-pressure difference for both situations, with a 95% confidence interval assuming a normal distribution, and the deviation between the average values, presented in Table 4.7. The Python script written to compare the situations can be found in Appendix H.12.

Table 4.7: Comparison of test signal measurements in all source rotation angles versus front angle only (0 degrees), with 95% confidence intervals and percentage deviation of average values.

Parameter	**Octave frequency bands [Hz]**				
	250	**500**	**1000**	**2000**	**4000**
$L_{WA} - L_{\text{Aeq25s,ears}}$, all rotations (m = 32) [dB]	4.5 ± 0.3	5.0 ± 0.3	2.7 ± 0.4	9.0 ± 0.8	10.1 ± 0.5
$L_{WA} - L_{\text{Aeq25s,ears}}$, front angle (m$_{front}$ = 4) [dB]	5.0 ± 1.2	5.1 ± 1.0	4.1 ± 2.3	9.1 ± 4.1	10.2 ± 1.2
Deviation of average value [dB]	+0.5	+0.1	+1.4	+0.1	+0.1

As shown in table above, the highest deviation between the average measured power-pressure difference is found in the 1000 Hz band, with +1.4 dB. The remaining bands have lower deviations, with deviation values of +0.1 dB for octave bands 500, 2000 and 4000 Hz. The highest uncertainty is however found at the 2000 Hz band for the front angle measurements, with a 95% confidence interval of ± 4.1 dB.

4.3 Sound Level Contributions at Ears in Practice Room

4.3.1 Room Acoustic Measurements

The reverberation time T_{20} and sound strength G were measured in the practice room, using the method described in Section 3.5.3. Figure 4.6 shows the measured T_{20} in octave bands, in comparison with the corresponding limits for the "Loud" music category from Figure 1.2, given in ISO 23591:2021. The variation of measurements (m = 6) is indicated with box plots.

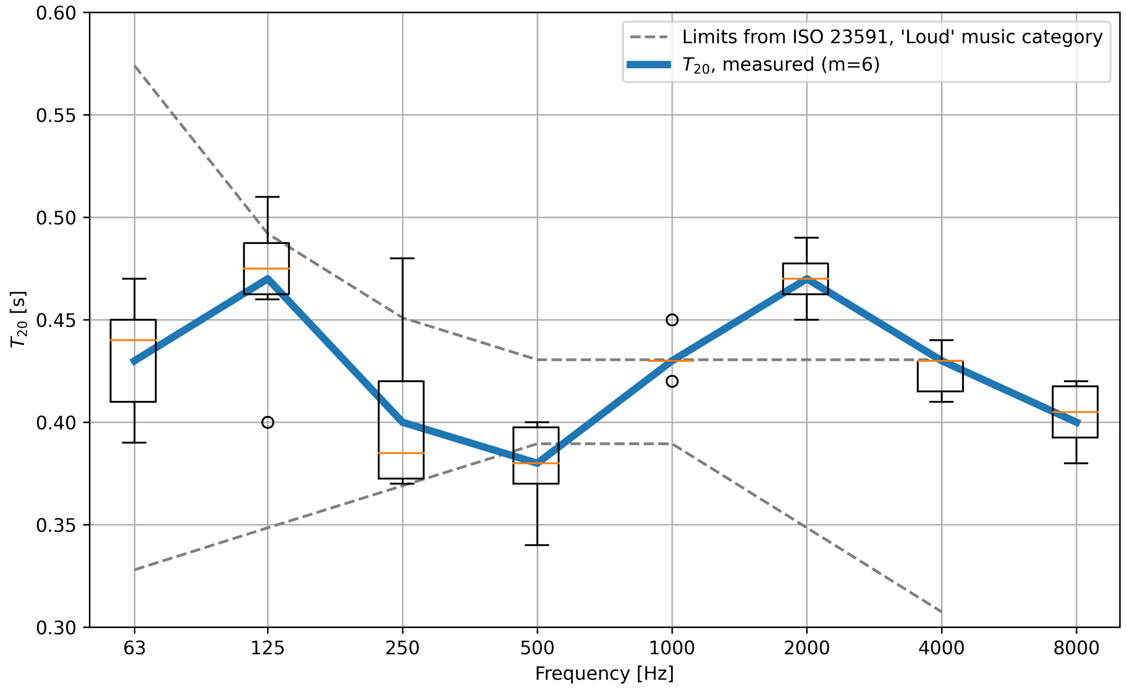

Figure 4.6: Reverberation time T_{20} measured in practice room. Blue line shows average value from measurements (m = 6), while dashed gray lines show upper and lower limits for "Loud" music category in ISO 23591, derived from the measured T_{mid} value. Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (orange line) and outliers (circles).

The T_{mid} value from the above figure is 0.41 seconds, from which the upper and lower reverberation limits have been derived. As shown in the figure, the measured T_{20} is too high in the 2000 Hz band and moderately low in the 500 Hz band, relative to the limits provided in ISO 23591. The room volume $V = 21.7$ m^3 is however lower than the minimum volume of 50 m^3 which is given for individual practice rooms in the "Loud" music category [3].

The Schröeder frequency f_c for the practice room was calculated to 275 Hz, using Equation 2.16 with the value of T_{mid}. The reverberant field is therefore expected to be dominated by room modes below this frequency. Given an increase of variation among the measured reverberation times in the bands below 500 Hz, this is likely caused by constructive and destructive interference effects from modes in lower frequencies.

Figure 4.7 shows the logarithmic average of the measured sound strength, G, from the practice room. Using the measured T_{mid} value, the critical distance $r_{crit.}$ was found to be 0.41 meters for mid frequencies, fulfilling $r_{SR} >> r_{crit}$ for all measurements as described in Section 3.5.3. The plotted G is shown to range between 27 dB and 31 dB for bands up tp 2000 Hz, after which the strength is significantly reduced for the upper bands. The variation from meaurements are here also indicated in the figure through box plots (m = 6), which is shown to be the most stable in the 500 - 2000 Hz bands.

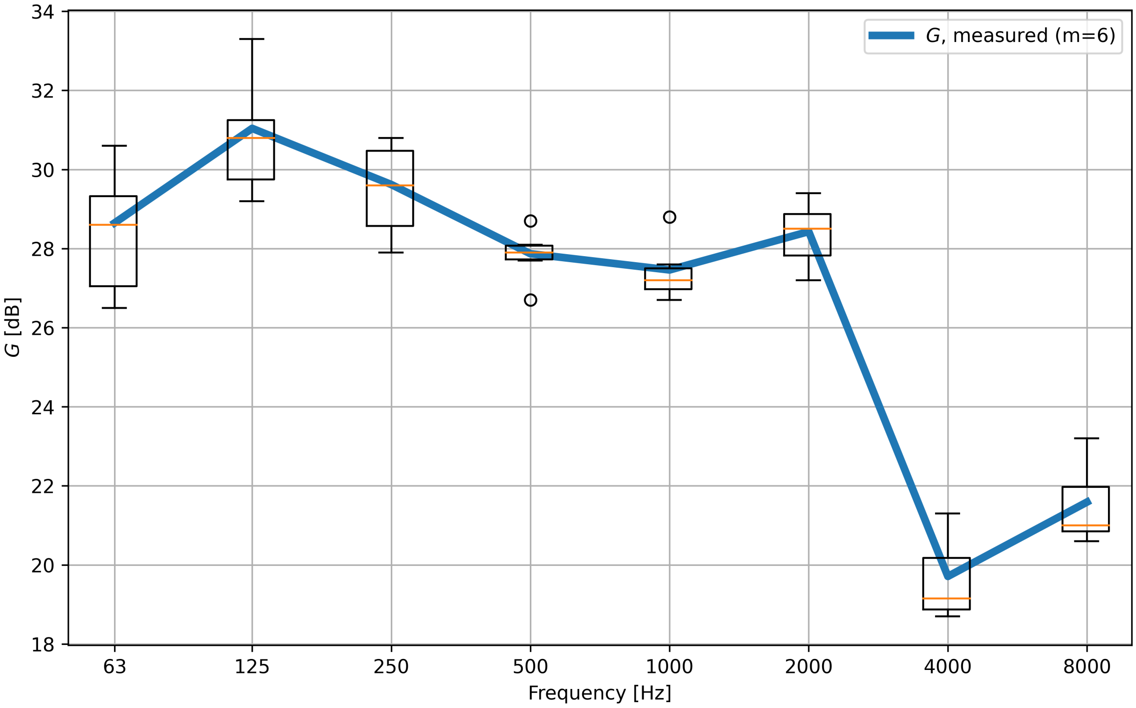

Figure 4.7: Sound strength G, measured in practice room. Blue line shows logarithmic average of measurements (m = 6). Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (orange line) and outliers (circles).

4.3.2 Determination of Levels at Ears

The G results were combined with existing measurements of SPL at the musicians ears for the test signals in the practice room, using the reference data for HRTF and instrument directivity, and the method described in Section 3.5.4. The average distance between the ears and acoustic center of the clarinet, r_{ac}, was calculated in 1/1 octave bands, using the average power-pressure difference for the test signals shown in Figure 4.3 along with the directivity reference data. The SPL contributions from the room and the direct sound, $L_{A,\text{ears,room,dir}}$ and $L_{A,\text{ears,room,refl}}$, were finally determined, using r_{ac}, the G measurements and the reference directivity data for DF_S and DF_R. Figure 4.8 shows the measured $L_{A,\text{ears,room}}$ with the determined $L_{A,\text{ears,room,dir}}$ and $L_{A,\text{ears,room,refl}}$ in the practice room, in 1/1 octave bands, for test signal scale runs in *forte*. The relevant parameters found or measured are provided in Table 4.8.

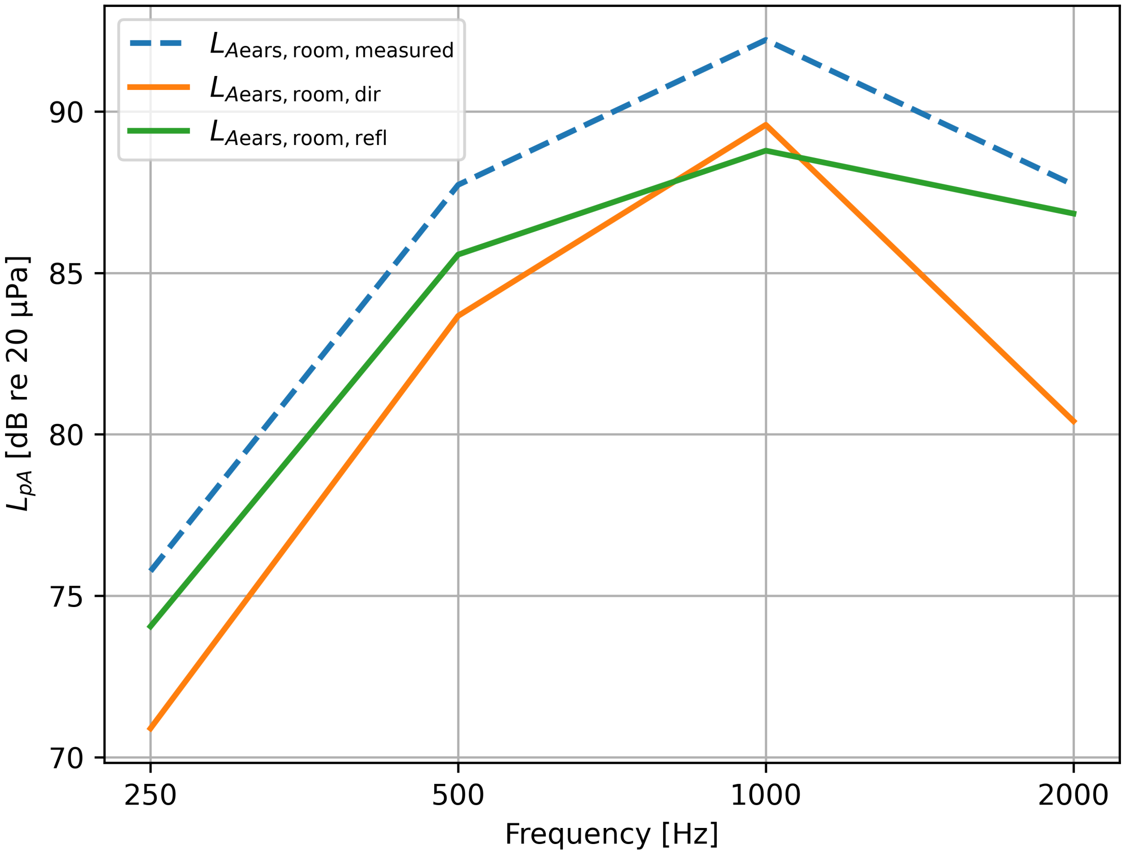

Figure 4.8: Sound pressure level contributions at ears for 1/1 octave bands, in decibels. Blue dashed line shows measured, total SPL at ears, while green and orange lines indicate the determined contributions of the reflected and the direct sound, respectively.

Table 4.8: Measured sound pressure levels at ears, $L_{A,\text{ears,room}}$, and found distance from ears to acoustic center of clarinet, r_{ac}, used for calculated estimate of ear level contributions. Bottom row shows critical distance r_{crit} for an omnidirectional source and receiver based on T_{20} measurements.

Parameter	**Octave band frequency center [Hz]**			
	250	500	1000	2000
$L_{A,\text{ears,room}}$ [dB re 20 μPa]	76	88	92	88
r_{ac} [m]	0.31	0.38	0.30	0.58
r_{crit} [m]	0.42	0.43	0.40	0.38

Chapter 5

Discussion

This chapter will discuss the results presented in Chapter 4, with influencing factors and the implications that follow.

5.1 Ear Levels During Practice Sessions

5.1.1 Equivalent Levels

Table 4.2 shows that the 8-hour exposure level $L_{A,\mathrm{EX8h}}$ from practice sessions averages to 82 dB in the semi-anechoic chamber, and 85 dB in the practice room. The published data from O'Brien, measured in a practice room, found a corresponding value of $L_{A,\mathrm{EX8h}} = 86$ dB for the Bb clarinet by comparison [1]. As The Norwegian Labour Inspection Authority (Arbeidstilsynet) sets a legal limit of 85 dB for the A-weighted 8-hour exposure level [2], an average practice session of 2.1 hours in the practice room alone will maximize the daily limit, based on the measurements presented in this report. Any additional noise exposure during the work day is likely to push the total $L_{A,\mathrm{EX8h}}$ above the legal limit, which for the case of O'Brien's data is the case for the practice session levels alone.

The findings in this report support data from earlier research, investigating the surrounding problems concerning noise exposure of professional musicians. Di Stadio *et al.* found a prevalence of tinnitus (26.5% of 1567) and hyperacusis (18.9% of 503) in a study among classical musicians [8], which can be linked to exposure of high sound levels. These prevalence numbers are higher than those of the general population, which were estimated to be 9.6% for tinnitus in the US [28], and 8-9% for hyperacusis on a global scale [29]. As the results in this report show, the levels from the individual practice sessions alone border on the legally permitted daily exposure. However, Jansen *et al.* found in a different study that although many participating musicians complained about tinnitus and hyperacusis, the location of the tinnitus could not be linked directly to the respective instruments of the musicians. It was further reported that the tinnitus usually was perceived in high frequency areas, which often is associated with NIHL [7]. This implies that for professional musicians in general, there are no immediate indications supporting the direct sound as the sole contributor of hearing loss among musicians. For the practice session measurements made of the Bb clarinet in this study, the 8-hour exposure level $L_{A,\mathrm{EX8h}}$ shows an increase of 3 dB from a semi-anechoic chamber to the practice room, implying an energetic doubling from the reflected and diffuse sound contribution. Hence, the sound levels at the ears are observed to increase due to room reflections, and the previous studies indicate further significant exposure from daily activities outside of the individual practice.

Regarding the level difference between the semi-anechoic chamber and the practice room, other factors than the additional reflection contributions will have an impact on the measured levels in the two locations. The human performance during the two 25 minute practice sessions will differ in terms of structure, intensity and stochastic variations. It was also shown in a study by Tor Halmrast that musicians, when placed in an especially absorptive room, will unconsciously compensate by 1-2 dB through their playing, compared with a more reflective practice environment [30]. The performed intensity and the measured levels will thus be influenced by the musicians' perception of the room.

5.1.2 Maximum Levels

The maximum ear levels presented in Table 4.3 show that average $L_{CF\mathrm{Max}}$ between the musicians was found to be 3 dB higher in the practice room compared to the semi-anechoic chamber, for practice sessions. The variation between the players is although higher in the practice room, with a 1.5 dB increase of the standard devation. A slight increase of 0.5 dB is also present for the average inter-aural difference in the practice room. It is possible that the room is not only contributing to a general level increase, but also increasing the range of exposed SPL from introducing more parameters. Changes in the musician's position and direction, or in the reflective and absorptive effects due to the size of their body, are likely to impact the increase of $L_{CF\mathrm{Max}}$ from the room.

The maximum C-weighted peak level $L_{C,\mathrm{peak}}$ measured during the practice sessions was also found to be higher in the practice room compared to the semi-anechoic chamber. Although this is expected due to a level increase from room reflections, the difference could also originate from human variation between performances. None of the registered $L_{C,\mathrm{peak}}$ values did however exceed the legal limit of $L_{C,\mathrm{peak}} = 130$ dB re p_0 set by the The Norwegian Labour Inspection Authority (Arbeidstilsynet) [2].

5.2 Sound Power Measurements

The octave band power-pressure differences presented in Figure 4.3 show a relatively stable value in the 250 and 500 Hz octave bands. This is possibly related to the findings by Meyer, who reported the directivity of the Bb clarinet to be close to omnidirectional below 500 Hz [10]. In addition, the fundamental frequencies of the tones played in the scale cover a range of 233-932 Hz, meaning that the fundamental frequencies can be assumed to be the primary contributors of these octave bands. This can be verified through Figure 1.4, which illustrates the same low C at the beginning of the scale used for test signal measurements. The next prominent partial harmonic is f_3, which is approximately 700 Hz for the lower C tone, hence not related to the 250 Hz band. As the fundamental tone increases for the remaining notes of the scale, this partial will not contribute to the 500 Hz band either.

The variation of $L_{WA} - L_{A,\mathrm{ears}}$ increases further for higher frequencies, implying a larger variation from both the instrument radiation and the surrounding environment above 500 Hz. The sources of the variation in measurement can be categorized into the following groups:

- L_{WA} measurements
- Non-stationary acoustic centra r_{ac} between tones
- Differences between instruments (n=4)
- Variation from HRTF, absorptive and reflective effects from body in between players
- Variation in player performance and in between players

5.2.1 Sound Power Uncertainty and Error

For the L_{WA} measurements, the uncertainty stems from both the measurement technique and from variation of the source, as described in Chapter 9 "Measurement Uncertainty" in ISO 3744:2010

[4]. Moreover, the choice of acoustic centra around which the microphone positions M1-M11 are centered will have an impact on the measured L_{WA}. The source was defined as the seated musician playing the clarinet, as isolation of the instrument is highly challenging. This means that there is a significant deviation between the positions of the center point of the chosen hemispherical measurement surface, and that of the true centeroid of the clarinet, positioned above the floor in an unknown horizontal position away from the hemisphere center. Therefore there are three essential error sources that must be specifically considered for the sound power measurements:

1. The distance between the acoustic centeroid and the array will vary for each microphone position, in both the horizontal and the vertical plane,

2. A source position above a reflecting floor will introduce interference effects,

3. The musician itself will act as an acoustic barrier, absorber or reflector.

The importance and the consequences of the the mentioned points will be discussed in the sections below.

5.2.2 Acoustic Center Position

The error originating from the non-stationary acoustic centeroid can partially be argued to be resolved for the horizontal plane, from performing 8 rotations of the source at even angle incrementations. The average distance between the true acoustic centeroid and the array points will approach the center of the hemisphere, thus reducing the error.

Moreover, microphones are not positioned symmetrically in relation to every source rotation position. This means that the measurement of a non-omnidirectional source will vary inbetween source rotations. On the contrary, by applying an array setup which is not perfectly symmetrical, errors from source directivity will be ”smoothed” out from the averaging of even source rotation increments.

5.2.3 Floor Reflection Effects

Regarding the floor reflections, the interference effect will be largest when the source has an unobstructed reflection path to the receiver. As the musician and chair are covering parts of the reflection area, the degree of interference will be much depending on the absorption introduced by the seated musician, across the frequency spectrum. For lower frequencies, the wavelengths are large compared to the musician and chair, potentially resulting in little absorption, whereas higher frequencies might be absorbed to a higher degree. Thus, interference effects might be reduced for high frequencies, for some reflection paths.

To explore the interference effect of the floor reflection, a Python simulation was performed for the measurement situation in the smei-anechoic chamber, where an omnidirecitonal source was positioned at 30 cm in front of the center, 83.5 cm above a reflecting surface, corresponding to the chosen characteristic source dimension d_O and roughly to the hand position of the musicians. The receiver positions were modelled for the L_{WA} measurement array, for each of the 11 microphones and in all 8 rotations of the musician (m = 88). The pressure impulses of the direct and the reflected signals were both scaled with a factor of $\frac{1}{r}$, corresponding to their respective propagation distances. The 1/1 octave band IR values of all 88 positions were finally averaged. Figure 5.1 shows the averaged simulated impulse response in octave bands, at the chosen receiver point, with the resulting sound pressure level offset from the interference. The said Python script can be found in Appendix H.6.

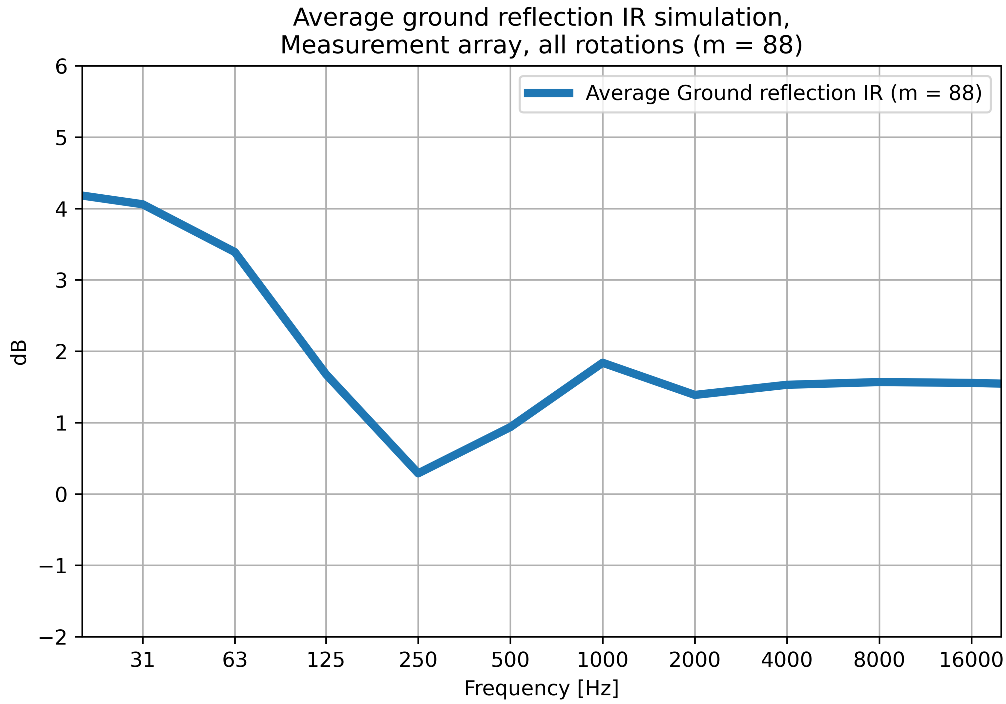

Figure 5.1: Simulation of average sound pressure level offset IR in decibels from floor reflection interference, due to difference between direct and reflected paths, in octave bands. Calculated for all array microphone positions, in each rotation of the musician (m = 88).

As shown in the above figure, the simulations show a general level increase below +2 dB in all bands of interest, but with a 1-2 dB reduction towards the 250 Hz band, compared to the remaining frequency range of interest. A higher increase can be observed for the lower frequencies, due to longer wavelengths. For the 1000 Hz band a slight peak can be observed, before the offset stabilizes at approximately +1.5 dB for the octave bands in higher frequencies. From this, the largest relative offset deviation will be present at the 250 Hz band and partially the 500 Hz band, with a maximum, relative deviation value below 2 dB.

The same simulation was run individually for the single shortest and longest distances between the source and array microphone positions. For the longest distance, the negative deviation at the 250 Hz band were closer to 6 dB compared to the surrounding bands. For the shortest distance however, the 125 Hz octave band value showed a -2 dB offset, while the values from (and including) 250 Hz stabilized to an approximate offset between +0.5-1 dB. Thus, the curve dip at 250 Hz is not present for the microphones with the shortest distance to the musician. The relative values between the remaining bands seem to be less deviating according to the simulations, although with a general level offset below at a maximum value +2 dB.

In reality, the high frequency level offsets might be somewhat lower due to air absorption, surface imperfections and absorption from the musician. It must also be acknowledged that the simulation only holds for a musician in a seated position. If the musician is standing, the difference between the direct and reflected paths will be larger, leading to a negative interference peak at a lower frequency.

The effect of floor reflections for the clarinet was also discussed by J. Meyer. He found that from the perspective of a listener, the intensity in harmonic components above 1500 Hz will increase resulting from the reflections [10]. This effect, and the interference effects discussed above, are however much less prominent from the position of the musicians' ears due to the relative difference of the direct and reflected propagation path distances. This is shown through an additional Python simulation, presented in Figure 5.2, where the listener position was set to 1.25 meters above the center point of the floor. As shown in the figure, the curve dip is lower in frequency than for the average array position, but still with maximum relative deviation between offsets below 2 dB. The Python script for simulating the IR offset at the ear receiver position can be found in Appendix H.7.

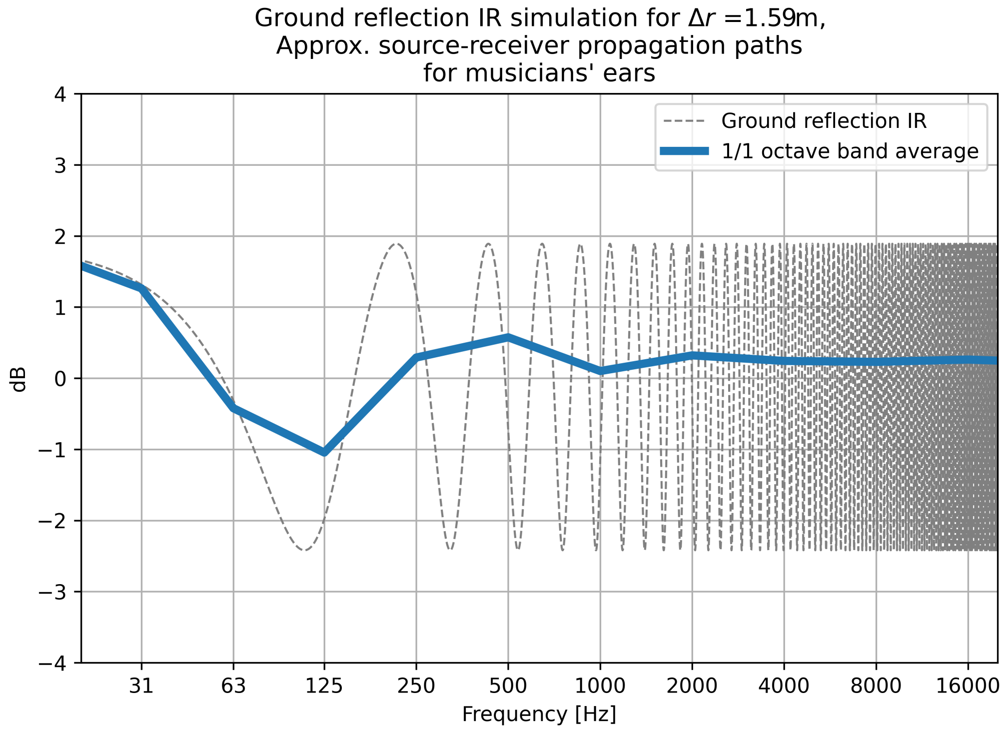

Figure 5.2: Simulation of pressure offset IR in decibels from floor reflection interference, for an approximate distance difference $\Delta r = 1.59$m between direct and reflected paths, from instrument center source to receiver position at musicians' ears.

5.2.4 Implications on Room Perception and Instrument Directivity

The results presented in Section 4.2.2 show that the power-pressure parameter is non-static with variations over time, over the course of a musical piece or an entire practice session. These variations lead to two new insights concerning both the musical instrument properties and the level perception of the musician.

Firstly, the most variation of $L_{WA} - L_{A,\text{ears}}$ occurs in the moments where the instrument is silent, between the musician's playing. In Figure 4.5, a trend can be seen where the power-pressure difference drops with a small decay after the musician stops playing, before it quickly rises to a higher value and starts alternating in an unpredictable manner. In these occurances, the musicians hear more of the room response, due to a lack of direct sound. The response from any reflective surfaces and resonances, in combination with non-stationary noise from breathing, cloth rustling and background noise, results in the high values and unpredictable behavior of $L_{WA} - L_{A,\text{ears}}$. Therefore, the power-pressure difference can be argued to reflect what the musician is hearing: a low and stable power-pressure difference while the musician is playing means that they hear a lot of the direct signal and not much from the room, while the moments of higher or less stable $L_{WA} - L_{A,\text{ears}}$ are the time windows where the musician "registers" the response of the room.

Secondly, although the power-pressure difference is more stable during playing of the instrument, there is always some degree of variation. This is also the case even for longer, stationary tones, indicating that the directivity of the instrument is non-stationary. It was found by M. Skålevik through array measurement analysis of violin and oboe that the directivity of harmonic partials changed over time, over the course of a single tone with a fixed pitch [31]. Although the oboe was found to have less variation than the violin in its fundamental frequency f_1, the higher order harmonics showed a significant variation in directivity. An example of this can be seen in Figure 25 in [31]. As the properties of the clarinet was found to be strongly comparable to those of the oboe by Pätynen and Lokki [18], the results presented in this report support the assumption that the directivity of the Bb clarinet behaves similarly. The test signal measurements performed with source rotations do however suggest that the average directivity over time remains stable, for the 250 Hz and the 500 Hz bands.

5.2.5 Single Angle Measurements

The comparison between test signal measurements performed in all source rotations and in the front angle only, presented in Section 4.2.3, are essential for evaluating the credibility of practice session measurements performed from a single rotation angle. Table 4.7 shows a deviation of +1.4 dB for the measured average power-pressure difference in the 1000 Hz band, while the 2000 Hz band holds the largest uncertainty (±4.1 dB) for the value measured at the front angle. Consequently, the presented single angle measurements from practice sessions can not be expected to be fully representative for the mentioned octave bands. For the remaining bands, the respective deviations and uncertainties are however found to be significantly lower, such that results for those bands are more likely to be more accurate for the sample selection. It must however be acknowledged that the sample size of the single angle measurement is quite low with a small statistical significance (m = 4), such that those results must be interpreted as mere indications instead of being representative for the whole population.

5.3 Practice Room Measurements

5.3.1 Practice Room Acoustics

The room volume of the practice room used for measurements, 21.7 m^3, is lower than the minimum volume limit of 50 m^3 given by ISO 23591 for individual practice rooms of the "Loud" music category. Figure 5.3 shows the volume and mid-frequency reverberation time T_{mid} for the practice room, in relation to the limits given by ISO 23591. According to the standard, the room can therefore be argued to be unsuitable for the given use. By extrapolating the "Loud" limit curves in Figure 5.3 down to 21.7 m^3, the measured T_{mid} value is shown to be too high even for the extrapolated limits. For the "Quiet" music category however, with identical octave band reverberation limits relative to T_{mid} as the "Loud" category, the measured T_{mid} value will almost satisfy the requirements for an extrapolated set of limit values. The corresponding lower room volume limit given in ISO 23591 for the "Quiet" category is 35 m^3, meaning that the practice room is considered too small according to the standard - even if one were to argue that the Bb clarinet is closer to a "Quiet" instrument.

Despite the small room volume, the participating musicians informed that the specific room was being used for practice on a daily basis. The two professional musicians further informed that they often were assigned even smaller rooms for their daily practice routines. As shown in Equation 2.18, a reduction in room volume V leads to an increase of the estimated G_{diff} value. This is further strengthened by inspecting the T and V contributions of the equation. While T typically ranges between 0-1 seconds for small and medium sized rooms, the range of V is much higher, meaning that the room volume will be the strongest contributor to the estimated sound strength G_{diff}. As a result, one can expect louder SPL in smaller rooms.

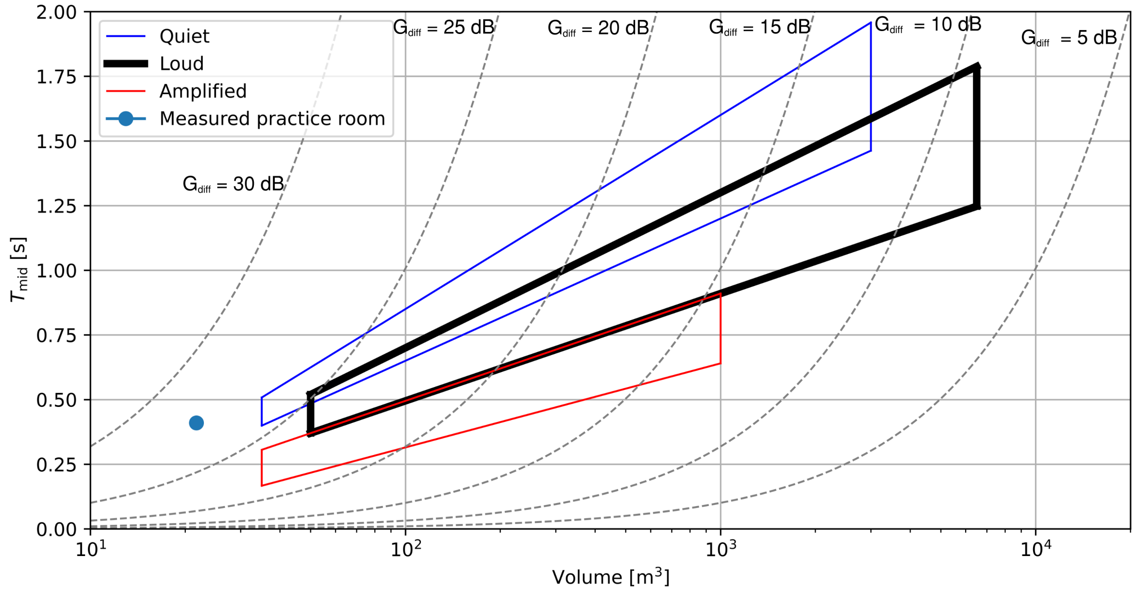

Figure 5.3: Limits and parameter guidelines of T_{mid} and room volume V, as provided in ISO 23591, with measured T_{mid} and V of practice room indicated with blue dot. "Loud" category used for Bb clarinet indicated with thicker black line. Sound strength G_{diff} estimates marked with dashed lines. Figure constructed from values given in ISO 23591:2021 [3].

5.3.2 Ear Level Contributions

The calculated sound level contributions in the practice room presented in Figure 4.8, show that the reflected sound level was found to be higher than the direct signal for most evaluated octave bands, except for 1000 Hz. From the discussion in Section 5.2.4, a low power-pressure difference $L_{WA} - L_{A\mathrm{eq25s,ears}}$ corresponds with the musicians' decreased ability to hear the room response, due to a more prominent direct sound level contribution. The determined octave band values of $L_{A,\mathrm{ears,room,dir}}$ and $L_{A,\mathrm{ears,room,refl}}$ presented in Figure 4.8 partially verify that this is the case for the 1000 Hz band. Although $L_{A,\mathrm{ears,room,dir}}$ was not found to be significantly higher than $L_{A,\mathrm{ears,room,refl}}$, this is the only evaluated band where the reflected sound is not the most prominent.

Further connecting the sound level contributions to the determined power-pressure difference, is the dominance of the reflected sound level at the 2000 Hz band. The value of $L_{WA} - L_{A\mathrm{eq25s,ears}}$ is found to be higher for the frequency bands above 1000 Hz, corresponding to a less significant contribution of the direct signal. Similarly, the power-pressure difference which was found to be moderate in the 250 and 500 Hz bands correspond to a less significant dominance of the reflected sound level.

Moreover, considering the diffuse field conditions is relevant for evaluating the direct and reflected level contributions. The assumption of $r_G >> r_{crit}$ was fulfilled for all evaluated bands, seeing as the r_{crit} for an omnidirectional source and receiver was found to range between 0.38 m and 0.43 between 250-2000 Hz. Still, the estimated distance between the ears and the acoustic center of the clarinet shown in Table 4.8, r_{ac}, was found to have comparable values, ranging between 0.30 m and 0.58 m for the same bands. This is an additional implication that the ears of the clarinetists are in a position relative to the clarinet where neither of the direct or reflected levels are fully dominating in the given practice room, from a theoretical standpoint.

5.4 Limitations

The limitations of measurements and the scope must be considered for evaluating the results brought to light in this report. This section will discuss limitations related to sample size, test material and methodology, as well as challenges related to source properties.

5.4.1 Sample Size and Instrument Model Distinction

The results presented in this report suffer from a limited sample size. The repeated measurements using multiple musical pieces and several source rotation angles do help increasing the data amount and the repeatability of the test, but the amount of musicians (n = 4) is still low. Considering that the musicians each used one single (privately owned) instrument means that the sample size of different clarinets used is the same as the number of musicians. Hence, any deviations originating from the charactestics of different clarinet makes, brands or models are unknown.

5.4.2 Isolated Instrument Measurements

SWL measurements could have been made in a fully anechoic chamber, centered around the clarinet, for a potentially better estimate of the isolated instrument. The chosen method of using a semi-anechoic chamber for sound power measurements was done partly due to practical purposes, being able to use a well-defined and standardized method. It was also chosen due to the consideration that a clarinetist will usually play standing or sitting above a reflective surface. As a consequence, the measurements do not represent the isolated clarinet, but a more complex source including the entire musician playing with floor reflections. The acoustic center offset from the floor, the lack of a defined characteristic source dimension d_{O} and the floor reflections themselves contribute to further imprecision, discussed in Section 5.2.3.

Even if an anechoic chamber was used, with or without an alternative method for measuring the radiated SWL, isolating the clarinet from the player during measurements would be very challenging. The musician, or the acoustic influence from the presence of the musician, will likely have to be included as part of the source definition.

5.4.3 Power-Pressure Difference in Practice Room

Measurements were not conducted of the radiated SWL in the practice room. This was due to space restrictions, in addition to the acoustic properties of the room which is neither largely reverberant nor (semi-)anechoic. Hence, the power-pressure difference could not be measured directly during the sessions in the practice room. It would be interesting to see how the measured difference between SWL and ear SPL in the practice room compares to the values from the semi-anechoic chamber.

5.4.4 Test Material

A limited selection of musical material was used for measurements, covering dynamics between *piano* and $forte$. The results are therefore restricted by the dynamic notation in the music, as well as the distribution of notes across the tonal register. Supplementing the results with more musical test material with a larger span in dynamics and tonal register, could grant an increased insight into the sound level properties over time of the Bb clarinet.

5.4.5 Uncertainty of SWL Measurements

For expressing the uncertainty and standard deviation of the measured sound power levels, the assumption has been made that the sample standard devation $\sigma_{L_W,\text{measured}}$ from measurements will be an approximate representation of the standardized uncertainty, $\sigma_{L_W,\text{measured}} \approx \sigma_{tot} \approx u(L_W)$. A limitation of this approach is however that there is missing information on how the underlying uncertainty is distributed between the repeatability and the source variation components, respectively represented by σ_{R0} and σ_{omc}.

It is possible to obtain this information through repetitions of measurements for σ_{omc}, and a set of round robin tests or mathematical modeling for σ_{R0}. Using a reference sound source (RSS) with a stationary and constantly emitted noise level, it is likely that one could approximate σ_{R0} through repeated measurements of the RSS.

5.5 Measurement Error Sources

Several sources of error must be acknowledged for the performed measurements. Some error sources related to the sound power measurements in semi-anechoic chamber have been discussed in Section 5.2. Other relevant errors for the contents in this report will be discussed here, for which the origins range from stochastic to deterministic, as well as from human variation.

5.5.1 Human Variation

The musicians involved are at all times influenced by internal and external factors, further influencing their playing style and dynamics. Fatigue, concentration and mood will likely have an impact on the produced power and the dynamic consistency of their playing. In addition, the visual and auditive surroundings will also influence how the musicians play their instrument. It was shown by Halmrast that musicians have a general tendency to unconsciously compensate their playing when placed in a very absorptive environment, as a result of the reduction of room response, which they usually rely on [30]. Halmrast observed that installing significant amounts of absorptive materials in a test room (from ”moderatly absorbing” to ”well absorbing”) would reduce the overall SPL from a constant reference source by 3-5 dB, but the effective reduction resulting from the players' compensation was found to be 1-2 dB less. As the two locations used for the tests in this report have fundamentally different reverberation characteristics, the potential influence of unconscious compensation from the players must be acknowledged.

5.5.2 Microphone Positioning

For measurements both in semi-anechoic chamber and in the practice room, influences from microphone positioning and mounting will be present. Vibrations from the floor, from the musicians themselves or disturbances in the building, could both decrease the precision of positioning as well as introducing low-frequency noise. For positioning the microphones, a laser distance measurement device was used with a precision of 0.01 meters, impacting the precision of the original microphone positions. Deviations from distance measurements, especially from the source, will also impact the measured sound pressure, with an approximate uncertainty of $\Delta p \propto (\frac{1}{r_{measured}} - \frac{1}{r_{true}})$. This is also true for imprecise positioning of the source, of which the true acoustic center is unknown and non-stationary for the Bb clarinet.

Regarding the array used for SWL measurements in the semi-anechoic chamber, symmetrical positions of microphones can result in unwanted interference effects. Microphones M1-M3 and M4-M5 were respectively mounted along the same axes with horizontal rods from track poles, due to space limitations. As a result, microphones M1-M3 share the same x-coordinate, while M4-M5 share their y-coordinate. As a worst consequence, the microphones might share effects from positive or negative interference patterns at certain frequencies due to similar reflection path distances. These effects are however estimated to be quite low, since the only reflective surface is the floor, which relates to the z-coordinate.

5.5.3 Effects from Musician

The presence of the musician itself will introduce acoustic effects altering the measured results. For higher frequencies, the musician will act as an absorber and a barrier, depending on their body size, clothing and hair. Other frequencies might reflect off the body of the musician, leading to level deviations from interference.

In addition to having a direct impact on measured levels, the barrier effect of the musicians further complicates the basis of modeling and calculation of ear levels. A reference data set has been used for the directivity of the clarinet, using an average instrument DI published by Pätynen and Lokki [18]. Such directivity patterns are usually obtained from measurements around the musician, meaning that the mentioned effects from the body of the musician will be present for the applied directivity data. Hence, there will be an error in the directivity for the angle of incidence, since the levels of interest in this report are localized at the ears of the musician, not behind the musician. This leads to an unknown error regarding the DI values applied for determining the direct and reflected sound level contribution at the ears, $L_{A,\mathrm{ears,room,dir}}$ and $L_{A,\mathrm{ears,room,refl}}$, in the practice room.

5.5.4 Frequency Responses of Measurement Elements

Deterministic (non-stochastic) errors due to impulse/frequency responses originating from various elements will also partly influence the results. The HRTF, discussed in Sections 2.9 and 3.5.4, is complex and different for every person. As the incidence angle of the instrument will vary due to movement of the musician, the HRTF will also be impacted further. The uncertainty of the HRTF variations, and its deviations from the averaged HRTF estimate curve used for the ear level calculations, will introduce errors to the measurements which cannot be fully quantified. In addition, the microphones that were used will have a non-flat frequency response. Although all microphones were calibrated prior to measurements, the response originating from the microphone capsule characteristic and from the mounted capsule adapters (discussed in Section 3.6) will introduce an error for each octave band. Errors from microphone capsules can be estimated using frequency responses from published datasheets relating to the used models, however it is not possible to know the true error for each individual microphone (including mounted capsule adapters) without designated tests.

5.6 Further Work

Several aspects encountered in the work of this thesis should be further investigated. An in-depth study should be conducted on the acoustic centeroid position of the Bb clarinet. The findings in this report, along with the dynamic directivity discussed by Skålevik, implies a non-stationary acoustic center position [31], varying with the input of the musician. Increasing the knowledge on the centeroid position would provide a better framework for performing systematic measurements of sound power and and related parameters, in addition to give a higher general understanding of the Bb clarinet acoustics.

There is also a need for evaluating alternative approaches to the definition of the clarinet source, possibly with alternative methods for sound power measurements. To find the best representation of the Bb clarinet as a sound source, an evaluation should be made of whether isolating the instrument from the player is possible, and if deemed desirable for practical purposes. A comparison is also needed of the representative properties of an anechoic versus a semi-anechoic environment for best representing the direct sound of the instrument.

If an accurate method is found feasible for conducting sound power measurements in small practice rooms, it would be desirable with comparable measurements of the power-pressure difference made directly in non-anechoic environments. The results will however serve as comparison values for indication, where the reflected sound from a practice room is present at the musicians' ears.

Moreover, the reference data used for the clarinet's DI suffer from an unknown degree of inaccuracy for the context of this report, due to measurement positions away from the player. Systematic research is needed on the directivity properties of the clarinet relative to the ears of the player. A more accurate investigation on the close-range directivity including typical effects from HRTF, in combination with research on the acoustic centeroid as discussed above, will grant an increased insight to the radiation properties of the Bb clarinet relative to the musician.

Finally, the results and measurements presented in this report have limitations which could be adressed in further research, by increasing the sample size and range of input data. Measurements of more musicians would be of great interest to verify the findings of this report, using a greater data population to achieve a greater statistical significance. It would also be desirable with measurements conducted in a range of room types and sizes, for specified clarinet model variations, and for alternative musical pieces, focusing on specific dynamics and tonal ranges. Seeing as the Bb clarinet here is a chosen case study, measuring the power-pressure difference at *forte* for additional musical instruments would also be of high value, expanding the reference data provided in ISO 23591.

Chapter 6

Conclusion

Sound pressure levels at the ears of 4 clarinetists have been measured individually in semi-anechoic chamber and in a practice room. 2 of the musicians are professional players with daily practice sessions in solitary and in groups, while the remaining 2 are active amateurs with experience from orchestra and wind bands. The musicians have been asked to perform a 25 minute structured practice session in both locations, for which the A-weighted equivalent sound pressure level $L_{A,\mathrm{eq25min}}$, daily exposure level $L_{A,\mathrm{EX8h}}$ and C-weighted maximum level $L_{CF\mathrm{Max}}$ have been measured. Also, sound power measurements have been performed in semi-anechoic chamber, for test signals in the dynamic *forte* with 8 source rotations. Measurements were done with an 11-microphone array using a method for a free-field environment over a reflective plane, as described in ISO 3744:2010 [4], approximated with a semi-anechoic chamber. The average difference between the A-weighted sound power level and the direct sound pressure at the ears, $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$, has followingly been determined. The test signals were additionally measured in a single position in the practice room. Room acoustic measurements in the practice room of sound strength, G, and reverberation time, T_{20}, were combined with the registered sound pressure levels at the ears to determine the contributions of the direct and reflected sound, $L_{A,\mathrm{ears,room,dir}}$ and $L_{A,\mathrm{ears,room,refl}}$. From the presented results, new insight has been gained regarding the sound level contributions at the ears of clarinetists during solitary practice.

The ear sound pressure levels for practice sessions and test tones have previously been measured by O'Brien *et al.* in non-anechoic rooms for various instruments in an orchestra, including the Bb clarinet [1]. While also measureing equivalent and maximum sound pressure levels, he found from a questionnaire that the average daily duration of practice sessions lasted 2.1 hours, from which a daily exposure level $L_{A,\mathrm{EX8h}}$ was calculated. The framework of the practice session measurements and analysis in this report has therefore been similarly structured for comparison purposes. In addition, a set of reference sound power levels are given for the dynamic *forte* in ISO 23591:2021, used for projecting rooms for music rehearsal, recital and practice [3]. Relating to this information, the measurements of the sound power-pressure difference are therefore based on this dynamic.

The ear sound pressure levels presented in Table 4.2 shows that the 8-hour exposure level $L_{A,\mathrm{EX8h}}$ from practice sessions averages to 82 dB re p_0 in the semi-anechoic chamber, and 85 dB re p_0 in the practice room. The Norwegian Labour Inspection Authority (Arbeidstilsynet) sets a legal limit of 85 dB for the A-weighted 8-hour exposure level [2], indicating that the daily noise exposure from solitary practice alone will maximize this limit. The results published by O'Brien *et al.* were similar, with a daily exposure level of $L_{A,\mathrm{EX8h}} = 86$ dB re p_0. Moreover, the average, maximum level $L_{CF\mathrm{Max}}$ was found to be 106 dB re p_0 in the practice room and 103 dB re p_0 in semi-anechoic chamber. The level increase from the surrounding room is expected, but the uncertainty and range of levels between the musicians increase as a result from the additional room parameters. The maximum, C-weighted instantaneous peak level $L_{C\mathrm{peak}}$ from all musicians was however found to be 115 dB re p_0, from the practice room measurements. This is lower than the maximum legal peak level limit set by The Norwegian Labour Inspection Authority of $L_{C\mathrm{peak}} = 130$ dB re p_0.

The $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ difference for the dynamic *forte* was found to be 3.9 ± 0.3 dB in single number rating. By investigating the octave band levels, the power-pressure difference was found to be most stable in the 250 and 500 Hz bands, with respective values of 4.5 ± 0.3 dB and 5.0 ± 0.3 dB. The difference value was found to decrease in the 1000 Hz band and then increase significantly for higher frequencies, but with a greater uncertainty in all bands above 500 Hz. Uncertainty values related to sound power levels have been obtained through retrieving the standard deviations from measurements, and calculating 95% confidence intervals assuming a normal distribution. A more detailed determination of uncertainty components is possible for further tests by applying a standardized uncertainty approach, distinguishing between technical repeatability represented by σ_{R0} and the variation of the source represented by σ_{omc}.

From test signal measurements in the practice room, the level contribution from room reflections was found to be higher in most octave bands compared to the direct sound. The exception is the 1000 Hz band, where the direct signal was slightly higher. This phenomenon relates to the low power-pressure difference that was found in the same band, indicating that the clarinetists' awareness of the room response is at its minimum in this frequency range.

The performed measurements are only indicative for the specific practice room used for this report, but it must be noted that the room is considered too small according to volume limits given in ISO 23591:2021 for individual practice rooms. As a result, the sound strength G and the resulting increase in sound pressure level from the room is relatively high.

One significant challenge for the tests is to define the source in a reasonable manner. Completely isolating the clarinet from the musician is probably impossible, and the musician will likely always be situated over a plane with some reflective properties. Hence, the extended source was defined as the clarinet and the musician, also enabling the use of ISO 3744. As a result, the line between source and receiver becomes somewhat undefined due to an overlap of the two. The acoustic centeroid of the source also becomes relevant, seeing as this is non-fixed and will exist somewhere above the floor surface.

From this, the effects originating from floor reflections in the process of determining the sound power level should be further investigated. A simulation script shows that the resulting average impulse response from the interference between direct and reflected signal paths for the measurement array yields a maximum relative offset deviation of -2 dB, for the sound pressure level in the 250 Hz octave band. Further simulations showed however that the interference effects from floor reflections are moderately less prominent from the position of the musician's ears, but with a maximum relative offset deviation of -2 dB for the 125 Hz band.

In general, the measurements conducted in this report suffer from a limited sample size. Further measurements should be performed for several musicians, in multiple room types and for a varied set of clarinet models and musical pieces. The uncertainty concerning the acoustic centeroid of the clarinet implies that there is a need for research on the non-stationary acoustic center position, and how it varies with the input of the musician. Finally, determining the $L_{WA,\mathrm{instrument}} - L_{A,\mathrm{ears}}$ difference for other instruments would be of great interest, relating to the information given in ISO 23591:2021 for the dynamic *forte*.

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Appendix A

Sound Pressure Level Measurements from Practice Sessions

The equivalent, A-weighted sound pressure level $L_{A\text{eq}}$ was recorded in both a semi-anechoic chamber and in a practice room, for the given sections of the structured practice sessions. Tables A.1 - A.3 show the measured SPL at ears and at 1.5 meter distance for warm-up section and musical pieces "Menuett" and "Gavotte", per person.

Table A.1: Measured SPL $L_{A,\text{eq5min}}$ at ears and at 1.5 meter distance, in semi-anechoic chamber and in practice room, during warm-up.

Person	**Semi-Anechoic**				**Practice Room**			
No.	Left Ear $L_{A,\text{eq5min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq5min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq5min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq5min}}$ [dB] re p_0	Left Ear $L_{A,\text{eq5min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq5min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq5min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq5min}}$ [dB] re p_0
1	83	84	83	76	84	82	83	78
2	78	80	79	73	84	86	85	81
3	92	93	92	85	94	95	95	90
4	90	90	90	82	94	93	93	90

Table A.2: Measured SPL $L_{A,\text{eq10min}}$ at ears and at 1.5 meter distance, in semi-anechoic chamber and in practice room, during rehearsal of music piece "Menuett".

Person	**Semi-Anechoic**				**Practice Room**			
No.	Left Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq10min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq10min}}$ [dB] re p_0	Left Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq10min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq10min}}$ [dB] re p_0
1	83	85	84	77	83	83	83	79
2	83	85	84	78	83	86	85	81
3	91	92	91	83	93	95	94	89
4	88	89	88	82	93	92	93	89

Table A.3: Measured SPL $L_{A,\text{eq10min}}$ at ears and at 1.5 meter distance, in semi-anechoic chamber and in practice room, during rehearsal of music piece "Gavotte".

Person No.	**Semi-Anechoic** Left Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq10min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq10min}}$ [dB] re p_0	**Practice Room** Left Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Right Ear $L_{A,\text{eq10min}}$ [dB] re p_0	Ears Avg. $L_{A,\text{eq10min}}$ [dB] re p_0	1.5m Ref. $L_{A,\text{eq10min}}$ [dB] re p_0
1	79	81	80	74	80	81	81	77
2	79	82	81	76	83	84	83	79
3	90	91	91	82	92	93	93	87
4	87	87	87	81	91	90	91	86

Appendix B

Maximum Levels from Practice Sessions

The maximum C-weighted, "Fast" time weighted levels $L_{CF,\mathrm{Max}}$ were obtained in semi-anechoic chamber and in a practice room, for each section of the structured practice session. Tables B.1 - B.3 show the maximum levels $L_{CF,\mathrm{Max}}$ for each of the musicians, for warm-up section and musical pieces "Menuett" and "Gavotte". The inter-aural differences between maximum levels are also provided, in decibels.

Table B.1: $L_{CF,\mathrm{Max}}$ for left and right ears of all musicians, with inter-aural differences between maximum levels, during warm-up.

Person No.	**Semi-Anechoic Chamber**			**Practice Room**		
	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]
1	96	97	1.7	100	96	3.9
2	92	96	3.9	98	101	3.1
3	105	106	0.9	108	107	1.2
4	103	103	0.1	109	106	3.6

Table B.2: $L_{CF,\mathrm{Max}}$ for left and right ears of all musicians, with inter-aural differences between maximum levels, during rehearsal of music piece "Menuett".

Person No.	**Semi-Anechoic Chamber**			**Practice Room**		
	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]
1	98	97	1.2	99	96	3.0
2	94	99	4.3	101	99	2.1
3	104	106	1.2	105	108	2.8
4	100	101	0.4	105	107	1.6

Table B.3: $L_{CF,\mathrm{Max}}$ for left and right ears of all musicians, with inter-aural differences between maximum levels, during rehearsal of music piece "Gavotte".

Person No.	**Semi-Anechoic Chamber**			**Practice Room**		
	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]	Left Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Right Ear $L_{CF,\mathrm{Max}}$ [dB] re p_0	Inter-aural diff. [dB]
1	97	100	2.7	95	95	0.2
2	92	95	2.9	95	98	2.8
3	103	104	0.7	106	108	2.0
4	96	97	0.7	103	103	0.7

Appendix C

Test Signal Measurements in Semi-Anechoic Chamber

The A-weighted sound power L_{WA}, sound pressure levels $L_{A,\mathrm{eq25s}}$ at ears and power pressure difference $L_{WA} - L_{A,\mathrm{eq25s,ears}}$ for the direct sound at the musicians' ears were meaured for test signals in *forte*, in a semi-anechoic chamber. For each musician, the measurements were performed for m = 8 source rotations. Table C.1 shows the measured, averaged values for each musician, with standard deviations from source rotations (m = 8), and 95% confidence intervals, assuming a normal distribution. For the power-pressure difference, a logarithmic average is used between the left and right ear levels for $L_{A,\mathrm{eq25s,ears}}$.

Table C.1: Averaged dB values of A-weighted sound power L_{WA}, sound pressure levels $L_{A,\mathrm{eq25s}}$ at ears and $L_{WA} - L_{A,\mathrm{eq25s,ears}}$ difference for scale runs in *forte*, in semi-anechoic chamber. Values are averaged from measurements in 45 degree increments in source rotation, with standard deviations in dB given between angles (m=8), and 95% confidence intervals assuming a normal distribution.

Person no.	SWL L_{WA}		Left Ear $L_{A,\mathrm{eq25s}}$		Right Ear $L_{A,\mathrm{eq25s}}$		Power-pressure diff. $L_{WA} - L_{A,\mathrm{eq25s,ears}}$	
	Avg. [dB] re 1pW	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB]	SD [dB]
1	95 ± 0.5	0.6	91 ± 0.4	0.5	92 ± 0.3	0.4	4.0 ± 0.5	0.6
2	94 ± 0.3	0.4	88 ± 0.4	0.5	91 ± 0.7	0.8	4.1 ± 0.5	0.6
3	98 ± 0.3	0.4	95 ± 0.3	0.3	96 ± 0.2	0.2	2.9 ± 0.3	0.4
4	95 ± 0.6	0.7	90 ± 0.3	0.4	91 ± 0.3	0.3	4.7 ± 0.6	0.7

Appendix D

Power-Pressure Difference for Practice Sessions, Single Position Measurements

The A-weighted sound power level, ear sound pressure levels and power-pressure differences for the direct signal at ears were measured in a single source rotation in the semi-anechoic chamber. Low levels originating from rests and breathing pauses have been gated away, such that any time windows where $L_{WAF} < 65$ dB re 1pW have not been evaluated. Tables D.1 - D.3 provide the measured values of the A-weighted sound power L_{WAF}, sound pressure levels L_{AF} at ears and $L_{WAF} - L_{AF,\mathrm{ears}}$ difference for the warm-up section and the musical pieces "Menuett" and "Gavotte", for each musician. The value of $L_{AF,\mathrm{ears}}$ used for the power-pressure difference is an average of the left and right ear levels.

Table D.1: Averaged dB values of A-weighted, F time-weighted sound power L_{WAF}, sound pressure levels L_{AF} at ears and $L_{WAF} - L_{AF,\mathrm{ears}}$ difference for 5 minutes of warm-up, in semi-anechoic chamber. Only time windows where $L_{WAF} \geq 65$ dB re 1pW have been evaluated. Standard deviations presented in decibels.

Person No.	L_{WAF}		**Left Ear** L_{AF}		**Right Ear** L_{AF}		$L_{WAF} - L_{AF,\mathrm{ears}}$	
	Avg. [dB] re 1pW	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB]	SD [dB]
1	86	5.4	80	5.0	82	5.1	5.0	1.4
2	81	4.5	74	4.6	77	4.8	5.2	1.4
3	95	6.5	90	6.3	91	6.9	5.1	1.7
4	92	5.2	87	5.6	87	5.4	6.0	1.8

Table D.2: Averaged dB values of A-weighted, F time-weighted sound power L_{WAF}, sound pressure levels L_{AF} at ears and $L_{WAF} - L_{AF,\text{ears}}$ difference for 10 minutes rehearsing music piece "Menuett", in semi-anechoic chamber. Only time windows where $L_{WAF} \geq 65$ dB re 1pW have been evaluated. Standard deviations presented in decibels.

Person No.	L_{WAF}		**Left Ear** L_{AF}		**Right Ear** L_{AF}		$L_{WAF} - L_{AF,\text{ears}}$	
	Avg. [dB] re 1pW	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB]	SD [dB]
1	87	6.5	80	5.9	82	5.8	5.0	1.4
2	87	5.8	80	5.3	83	5.3	5.1	1.4
3	94	7.0	88	6.6	90	6.7	5.1	2.2
4	91	6.2	85	5.8	86	5.8	6.1	2.3

Table D.3: Averaged dB values of A-weighted, F time-weighted sound power L_{WAF}, sound pressure levels L_{AF} at ears and $L_{WAF} - L_{AF,\text{ears}}$ difference for 10 minutes of rehearsing music piece "Gavotte", in semi-anechoic chamber. Only time windows where $L_{WAF} \geq 65$ dB re 1pW have been evaluated. Standard deviations presented in decibels.

Person No.	L_{WAF}		**Left Ear** L_{AF}		**Right Ear** L_{AF}		$L_{WAF} - L_{AF,\text{ears}}$	
	Avg. [dB] re 1pW	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB] re p_0	SD [dB]	Avg. [dB]	SD [dB]
1	84	6.5	77	5.7	79	5.9	4.8	1.2
2	84	5.0	76	4.6	80	4.7	5.1	1.2
3	93	7.2	88	7.3	89	7.3	4.6	1.9
4	90	5.9	84	5.6	85	5.7	5.4	1.9

Appendix E

Sound Power Measurement Positions

The microphone positions M1-M11 used in the sound power measurement array is given below in below in Table E.1. In the right column are the corresponding, or closest, microphone positions provided in ISO 3744:2010 [4]. Positions M1-M5 are positioned using horizontal rods of fixed lengths mounted to stationary poles, while M6-M11 are individually positioned on the floor using single microphone stands.

Table E.1: Microphone positions used for sound power measurement array in semi-anechoic chamber, with corresponding or close key positions provided in Table B.1 of ISO 3744:2010 [4], in reference to center point (origo). Values from ISO 3744 calculated from a measurement surface radius of $r = 1.67$m. All coordinate values given in millimeters.

Microphone Name	**Microphone Position (x,y,z) [mm]**	**ISO 3744:2010 Corresponding Mic. Position Number**	**ISO 3744:2010 Corresponding Mic. Position (x,y,z) [mm]**
M1	(0, 1626, 334)	1	(-267, 1603, 367)
M2	(0, 909, 1389)	7	(434, 1086, 1186)
M3	(0, 512, 1580)	10	(-167, 167, 1653)
M4	(-1403, -786, 413)	3	(-1303, -919, 518)
M5	(-1165, -786, 871)	18	(-718, -1019, 1119)
M6	(-1000, 1300, 334)	2	(-1303, 1002, 334)
M7	(1386, -534, 752)	5	(1386, -534, 752)
M8	(1386, 668, 635)	6	(1386, 668, 635)
M9	(434, -835, 1386)	9	(434, -835, 1386)
M10	(-1236, 117, 1119)	8	(-1236, 117, 1119)
M11	(-267, 1500, 689)	4	(-267, -1503, 685)

Appendix F

Equipment Lists

In the three bullet list sections below are given the equipment used for measurements. Equipment used for sound pressure measurement is the same for both semi-anechoic chamber and for the practice room.

F.1 Semi-anechoic chamber

- Sound power measurements:
  - 24.5 m^2 chipboards
  - Microphone array:
    - 11x free-field microphones
      - 6x Norsonic Nor1201 Preamplifier (M1-M5,M11)
      - 2x Norsonic Nor1225 Microphone
      - 3x Bruel & Kjaer 4190 Microphone
      - 1x Bruel & Kjaer 4165 Microphone
      - 5x nTi M4261 Integrated Microphone & Preamplifier (M6-M10)
    - 6x Microphone stands
    - 5x Threaded metal rods with microphone attachments
    - 6x Bruel & Kjaer 1708 Signal Conditioner
    - XLR cables
    - BNC cables
    - Makita LD050P laser distance measurement device
- Sound pressure measurements:
  - 2x Sennheiser MKE-1EW lavalier microphones (Left/Right) with ear mounts, integrated pre-amplifier
  - 1x Sony ECM-50PS Clip-on lavalier microphone, integrated pre-amplifier
  - 1x Norsonic Nor1201 Preamplifier (1.5 ref microphone)
  - 1x Norsonic Nor1225 Microphone (1.5 ref microphone)
  - 1x Microphone stand (1.5 ref microphone)
  - 1x Bruel & Kjaer 1708 Signal Conditioner (1.5 ref microphone)
  - 1x BNC-cable (1.5 ref microphone)
  - XLR-cables
- Lynx Aurora Sound Interface

- Chair
- Angular markings on floor
- Note stand
- Bruel & Kjaer 94dB Calibrator & Capsule Size Adapters
- Computer with Adobe Audition audio recording software

F.2 Practice Room

- Sound pressure measurements: As in F.1
- Chair
- Note stand
- Bruel & Kjaer 94dB Calibrator & Capsule Size Adapters
- Nor140 Sound Analyzer (Background Noise Measurement)
- Sound Devices MixPre-6 4channel portable audio recorder

F.3 Room Acoustic Measurements

- Norsonic Nor276 omnidirectional speaker
- Norsonic Nor280 power amplifier
- Speaker stand
- Speak-on cable
- 1x Norsonic Nor1201 Preamplifier
- 1x Norsonic Nor1225 Microphone
- 1x Microphone stand
- 2x XLR cables
- 1x Bruel & Kjaer 1708 Signal Conditioner
- 1x BNC-cable
- Bruel & Kjaer 94dB Calibrator & Capsule Size Adapters
- USB and power cables
- Roland Studio-Capture sound interface
- Computer with ODEON Combined 17

Appendix G

Microphone Signal Chains

Table G.1 provides information about the microphones and signal chains used for measurements. Specifics regarding models can be found in the equipment list, given in Appendix F.

Table G.1: Information and signal chains for measurement microphones.

Microphone(s)	**Type**	**Preamplifier**	**Amplification**	**Classification**
M1-M5, M11	Free-field	Mounted	External Signal Conditioner Amp.	Class 1
M6-M10	Free-field	Integrated	+48V from sound interface	Class 2
Left, Right Ear	Lavalier	Integrated	+48V from sound interface	-
Clip-on	Lavalier	Integrated	+48V from sound interface	-
1.5 ref	Free-field	Mounted	External Signal Conditioner Amp.	Class 1

Appendix H

Python Code

This chapter provides relevant Python code written for analyzation, calculations and simulations related to the presented results. The mathematical methods are based on the theory presented in Chapter 2. Certain levels and parameters have been calculated externally using spreadsheets, also based on the methods described in Chapter 2. For sound power levels and sound pressure levels, the measured values have been stored in external csv-files, before being re-imported and analyzed for the various purposes.

H.1 A_weighting.py

The functions in this Python-script are used for calculating A-weighted sound levels in 1/3 and 1/1 octave bands.

```python
from functions_constants import *
from import_scale import *
import matplotlib.pyplot as plt
import numpy as np
from matplotlib import pyplot as plt
import matplotlib.ticker as ticker
import numpy as np
from numpy import fft as fft
import math as m

#Some necessary functions:
def calculate_rms(sig):
    return m.sqrt(np.mean(sig**2))

def calculate_spl(p_rms, p0):
    if p_rms == 0:
        answer = -1000
    else:
        answer = 20*m.log10(p_rms/p0)
    return answer

def bandsPowerSpec(powerSpec, f_m, f_lim, freq):
    temp = 0
    bandPowerSpec = [0]*len(f_m)
    n = 0
    for i in range(len(powerSpec)):
        if freq[i] < f_lim[n]:
            pass
        elif (freq[i]>=f_lim[n]) and (freq[i]<f_lim[n+1]) and (i<(len(freq)-1)):
            temp += powerSpec[i]
        elif (i<(len(freq)-1)):
            bandPowerSpec[n] = temp
            n += 1
            temp = powerSpec[i]
        else:
            temp += powerSpec[i]
            bandPowerSpec[n] = temp
    return bandPowerSpec

def get_A_band_levels(sig):
    numZerosRec = 0
```

```
    fftRec = fft.fft(sig, len(sig) + numZerosRec)
    fftRecLength = fftRec.size
    fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec = [x ** 2 for x in abs(fftRec)]
    # Creating 1/3 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
           250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
           16000, 20000, 25000]
    # A-weighting factors
    a_weight = [-10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -85.4, -77.8, -70.4,
                -63.4, -56.7, -50.5, -44.7, -39.4, -34.6, -30.2, -26.2, -22.5, -19.1, -16.1, -13.4, -10.9, -8.6, -6.6,
                -4.8, -3.2, -1.9, -0.8, 0, 0.6, 1.0, 1.2, 1.3, 1.2, 1, 0.5, -0.1, -1.1, -2.5, -4.3, -6.6, -9.3,
                -10 ** 9]
    # Border frequencies
    f_lim = [0.9, 1.1, 1.4, 1.8, 2.2, 2.8, 3.5, 4.4, 5.6, 7, 8.9, 11.2, 14.1, 17.8, 22.4, 28.2, 35.5, 44.7, 56.2, 70.8,
             89.1, 112, 141, 178, 224, 282, 355, 447, 562, 708, 891, 1122, 1413, 1778, 2239, 2818, 3548, 4467, 5623,
             7079, 8913, 11220, 14130, 17780, 22390, 28184]
    # Creating the bins
    oneThirdPowerSpec = bandsPowerSpec(powerSpecRec, f_m, f_lim, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in oneThirdPowerSpec]
    # Calculating A-weighted band levels
    a_weighted_levels = []
    for i in range(len(a_weight)):
        a_weighted_levels.append(band_levels[i] + a_weight[i])
    return a_weighted_levels

def get_A_band_levels_1_1_oct(sig):
    numZerosRec = 0
    fftRec = fft.fft(sig, len(sig) + numZerosRec)
    fftRecLength = fftRec.size
    fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec = [x ** 2 for x in abs(fftRec)]
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
           250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
           16000, 20000, 25000]
    # A-weighting factors
    a_weight_oct = [-10 ** 9, -10 ** 9, -10 ** 9, -77.8, -56.7, -39.4, -26.2, -16.1, -8.6, -3.2, 0, 1.2, 1.0, -1.1, -6.6, -10 ** 9]
    # Octave band freq lists:
    f_m_oct = f_m[::3]
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680, 11360, 22000, 44000]

    # Creating the bins
    octPowerSpec = bandsPowerSpec(powerSpecRec, f_m_oct, f_lim_oct, freqRec)

    # Plotting the power spectrum in octave bands:
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerSpec]

    # Calculating A-weighted band levels
    a_weighted_levels = []
    for i in range(len(a_weight_oct)):
        a_weighted_levels.append(band_levels[i] + a_weight_oct[i])
    return a_weighted_levels
```

H.2 C_weighting.py

The functions in this Python-script are used for calculating C-weighted sound levels in 1/3 and 1/1 octave bands, related to determining $L_{CF,\text{Max}}$.

```
from functions_constants import *
from import_scale import *
import matplotlib.pyplot as plt
import numpy as np

from matplotlib import pyplot as plt
import matplotlib.ticker as ticker
import numpy as np
from numpy import fft as fft
import math as m

#Some necessary functions:
def calculate_rms(sig):
    return m.sqrt(np.mean(sig**2))

def calculate_spl(p_rms, p0):
    if p_rms == 0:
        answer = -1000
    else:
        answer = 20*m.log10(p_rms/p0)
    return answer

def bandsPowerSpec(powerSpec, f_m, f_lim, freq):
    temp = 0
    bandPowerSpec = [0]*len(f_m)
    n = 0
    for i in range(len(powerSpec)):
        if freq[i] < f_lim[n]:
            pass

        elif (freq[i]>=f_lim[n]) and (freq[i]<f_lim[n+1]) and (i<(len(freq)-1)):
            temp += powerSpec[i]
        elif (i<(len(freq)-1)):
            bandPowerSpec[n] = temp
            n += 1
            temp = powerSpec[i]
        else:
            temp += powerSpec[i]
            bandPowerSpec[n] = temp
    return bandPowerSpec

def get_C_band_levels(sig):
    numZerosRec = 0
    fftRec = fft.fft(sig, len(sig) + numZerosRec)
    fftRecLength = fftRec.size
    fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec = [x ** 2 for x in abs(fftRec)]

    # Creating 1/3 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
           250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
           16000, 20000, 25000]
    # C-weighting factors
    c_weight = [-10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -10 ** 9, -21.3, -17.7, -14.3,
                -11.2, -8.5, -6.2, -4.4, -3.0, -2.0, -1.3, -0.8, -0.5, -0.3, -0.2, -0.1, 0.0, 0.0, 0.0,
                0.0, 0.0, 0.0, 0.0, 0, 0.0, -0.1, -0.2, -0.3, -0.5, -0.8, -1.3, -2.0, -3.0, -4.4, -6.2, -8.5, -11.2,
                -10 ** 9]
    #Border frequencies
    f_lim = [0.9, 1.1, 1.4, 1.8, 2.2, 2.8, 3.5, 4.4, 5.6, 7, 8.9, 11.2, 14.1, 17.8, 22.4, 28.2, 35.5, 44.7, 56.2, 70.8,
             89.1, 112, 141, 178, 224, 282, 355, 447, 562, 708, 891, 1122, 1413, 1778, 2239, 2818, 3548, 4467, 5623,
             7079, 8913, 11220, 14130, 17780, 22390, 28184]
    # Creating the bins
    oneThirdPowerSpec = bandsPowerSpec(powerSpecRec, f_m, f_lim, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in oneThirdPowerSpec]

    # Calculating C-weighted band levels
    c_weighted_levels = []
    for i in range(len(c_weight)):
        c_weighted_levels.append(band_levels[i] + c_weight[i])
    return c_weighted_levels
```

H.3 Calibration.py

The functions given in this Python-script is used for automatic calibration of measurements. The scripts "functions_constants.py" and "import_scale.py" call these functions for analysis of any measurement, depending on the measurement name and file.

```
from matplotlib import pyplot as plt
import numpy as np
import scipy.signal as signal
from functions_constants import *
from scipy.io import wavfile

def find_high_peaks(sig):
    idx = list(signal.find_peaks(sig)[0])
    return sig[idx] #Returns values of high peaks

def find_low_peaks(sig):
    idx = list(signal.find_peaks(-sig)[0])
    return sig[idx] #Returns values of low peaks

def find_peak_differences(sig):
    highs = find_high_peaks(sig)
    lows = find_low_peaks(sig)
    max_idx = min(len(lows), len(highs))
    return highs[0:max_idx] - lows[0:max_idx]

def get_scale_factor(sig, dB):
    ref_rms = Lp2amp(dB)
    diffs = find_peak_differences(sig)
    median_rms = 1/np.sqrt(2) * (np.median(diffs)/2)
    scale_factor = ref_rms/median_rms
    return scale_factor
```

H.4 Clarinet_Directivity.py

This Python-script is used to determine the estimated DI of the clarinet's radiation angle, towards the musician's ears, as presented in Figure 1.6. Reference data from Pätynen and Lokki [18].

```
from functions_constants import *
import numpy as np

bands = [250, 500, 1000, 2000]
angle_list = [0, 45, 90, 135, 180, 225, 270, 315]
#0: top
#90: front
#180: bottom
#270: back

#data is from Patynen/Lokki: "Instrument directivity" - acta acustica 2010
_250hz_normdb = [-4.5, -3.5, 0, -1.5, -6, -7, -8, -6.5]
_500hz_normdb = [-6, -4, 0, -3.5, -9.5, -9.5, -9, -8]
_1000hz_normdb = [-9.5, -10, 0, -5.5, -8, -8, -10, -10]
_2000hz_normdb = [-10, -8, -5, 0, -9, -9, -10, -10]

power_ref = 1

_250hz_norm_power = [power_ref * 10**(x/10) for x in _250hz_normdb]
_500hz_norm_power = [power_ref * 10**(x/10) for x in _500hz_normdb]
_1000hz_norm_power = [power_ref * 10**(x/10) for x in _1000hz_normdb]
_2000hz_norm_power = [power_ref * 10**(x/10) for x in _2000hz_normdb]

norm_powers = [_250hz_norm_power, _500hz_norm_power, _1000hz_norm_power,
    _2000hz_norm_power]
scaled_band_levels = []

for i, powers in enumerate(norm_powers):
    print("Band: ", 250*(2**i), "Hz")
    meanpow = np.mean(powers)
    print("Mean power before scale:" ,meanpow)
    power_scale_factor = power_ref/meanpow
    print("Power scale factor:", power_scale_factor)
    scaled_powers = [p * power_scale_factor for p in powers]
    meanpow = np.mean(scaled_powers)
    print("Scaled powers:", scaled_powers)
    print("Mean power after scale:", meanpow)
    scaled_band_levels.append([10*np.log10(x) for x in scaled_powers])
    print("Scaled levels: ", scaled_band_levels[i])
```

```
    print("Scaled dB mean: ", round(avg_db(scaled_band_levels[i]),1))
    indent()

band_DI_angleofincidence = []
band_DF_angleofincidence = []

for i, band in enumerate(scaled_band_levels):
    print("Band: ", 250 * (2 ** i), "Hz")
    print([round(x,1) for x in band])
    DI_angleofincidence = round(band[-1],1)
    band_DI_angleofincidence.append(DI_angleofincidence)
    print("DI at angle of ear incidence:", DI_angleofincidence)
    DF_angleofincidence = round(10**(DI_angleofincidence/10),2)
    band_DF_angleofincidence.append(DF_angleofincidence)

indent()
print("Bands [Hz]:", bands)
print("DI:", band_DI_angleofincidence)
print("DF:", band_DF_angleofincidence)
```

H.5 functions_constants.py

This Python-script provides general lists, functions and constants used by most other scripts.

```
import math as m
import numpy as np
from scipy.io import wavfile
import pandas as pd

df = pd.read_csv("timestamps.csv") #Contains manually written information about start and stop times for various
# sections

fs = 48000
filelist = ["M1",
            "M2",
            "M3",
            "M4",
            "M5",
            "M6",
            "M7",
            "M8",
            "M9",
            "M10",
            "M11",
            "Left",
            "Right",
            "Clips",
            "1.5m ref"]

quarterinch_and_lavalier_list = ["Left", "Right", "Clips", "1.5m ref"]

anglelist = [str(x*45) for x in range(0,8)]
takelist = ["Take1", "Take2", "Take3"]

p0 = 20*10**(-6)
W0 = 1*10**(-12)

#1/3 oct band center frequencies
f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100,
       125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000,
       5000, 6300, 8000, 10000, 12500, 16000, 20000, 25000]

#1/1 oct band center freq
f_m_oct = f_m[::3]
f_m_oct.append(32000)

def get_start_stop_times(person, location, type, df):
    start = int(df.loc[(df["Person"] == person) & (df["Location"] == location) & (df["Type"] == type), "Start"])
    stop = int(df.loc[(df["Person"] == person) & (df["Location"] == location) & (df["Type"] == type), "Stop"])
    return start, stop

def Lp2amp(Lp):
    global p0
    return p0 * 10 ** (Lp / 20)

def amp2Lp(amp):
```

```python
    global p0
    return 20 * np.log10(amp / p0)

def Lw2W(Lw):
    global W0
    return W0 * 10 ** (Lw / 10)

def W2Lw(W):
    global W0
    return 10 * np.log10(W / W0)

def calculate_rms(sig):
    return m.sqrt(np.mean(sig**2))

def importwav(person, location, type, file):
    filename = "data/" + location + "/" + person + "/" + type + "/" + file + ".wav"
    #Importing files and sampling frequencies
    fs, sig = wavfile.read(filename)
    return sig

def importwav_soundpower(person, location, type, file, subtype): #subtype is used
    differently in this case
    filename = "data/" + location + "/" + person + "/" + type + "/" + subtype + "/" +
        file + ".wav"
    #Importing files and sampling frequencies
    fs, sig = wavfile.read(filename)
    return sig

def indent():
    print("\n")

def avg_db(db_list):
    temp = np.average([10**(db/10) for db in db_list])
    if temp == 0:
        return -1.00000000e+09
    else:
        return 10*np.log10(temp)
```

H.6 ground_reflection_array.py

This Python script simulates the interference effect from ground reflections for the measurement array used in SWL measurements, in the semi-anechoic chamber. The average IR is calculated from octave band IR curves from 11 microphones, in 8 rotations of the musician (m = 88).

```python
import numpy as np
import matplotlib.pyplot as plt
from HRTF_average import get_octave_levels
from HRTF_average import get_octave_average_squared_levels
from HRTF_average import bandsAvgPowerRatioSpec
from A_weighting import *
import math as m

fs = 48000
c = 343

def get_band_levels_1_1_oct_from_FIR(powerFIR, fs, freqRec, p0):
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50,
        63, 80, 100, 125, 160, 200,
           250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000,
               6300, 8000, 10000, 12500,
           16000, 20000, 25000]
    # Octave band freq lists:
    f_m_oct = f_m[::3]
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680,
         11360, 22000, 44000]
    # Creating the bins
    octPowerSpec = bandsAvgPowerRatioSpec(powerFIR, f_m_oct, f_lim_oct, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerSpec]
    return band_levels

def rotate(pos, angle): #Clockwise rotation around the z-axis
    px, py, pz = pos
    qx = m.cos(angle) * (px) - m.sin(angle) * (py)
    qy = m.sin(angle) * (px) + m.cos(angle) * (py)
    qz = pz
    return [qx, qy, qz]

```

```python

#Array positions (x,y,z):
array_positions = [
    [0, 1.626, 0.334],
    [0, 0.909, 1.389],
    [0, 0.512, 1.580],
    [-1.403, -0.786, 0.413],
    [-1.165, -0.786, 0.871],
    [-1.0, 1.3, 0.334],
    [1.386, -0.534, 0.752],
    [1.386, 0.668, 0.635],
    [0.434, -0.835, 1.386],
    [-1.236, 0.117, 1.119],
    [-0.267, 1.50, 0.689]
]

all_array_positions = []

for k in range(0,8): #Rotating the array in 8 positions, yielding (m = 88) microphones positions
    rotated_array = [rotate(mic, k*(m.pi/4)) for mic in array_positions]
    for mic in rotated_array:
        all_array_positions.append(mic)

#Source position:
source_pos = [0, 0.30, 0.835] #Rough estimate, assuming 45 deg incidence angle towards ears. Ears are assumed located directly above center point.

octave_bands_ratio_all = []
for microphone in array_positions:
    dir_dist = np.sqrt((microphone[0]-source_pos[0])**2 + (microphone[1]-source_pos[1])**2 + (microphone[2]-source_pos[2])**2)
    refl_dist = np.sqrt((microphone[0]-source_pos[0])**2 + (microphone[1]-source_pos[1])**2 + (microphone[2]+source_pos[2])**2)

    # Create timevec and insert impulses:
    time_vec = [0]*fs
    dir_time = dir_dist/c
    dir_n = round(dir_time*fs)
    refl_time = refl_dist/c
    refl_n = round(refl_time*fs)
    time_vec[dir_n] = 1/dir_dist
    time_vec_ref = [x for x in time_vec]
    time_vec[refl_n] = 1/refl_dist #Without losses. Could potentially include air loss, but additional losses will also be present for higher frequencies due to body parts blocking the sound<

    # Full FFT
    #direct sig
    numZerosRec = 0
    fftRec_dir = fft.fft(time_vec_ref, len(time_vec_ref) + numZerosRec)
    fftRecLength = fftRec_dir.size
    fftRec_dir = m.sqrt(2) / fftRecLength * fftRec_dir[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec_dir = [x ** 2 for x in abs(fftRec_dir)]

    #Reflected sig
    numZerosRec = 0
    fftRec_refl = fft.fft(time_vec, len(time_vec) + numZerosRec)
    fftRecLength = fftRec_refl.size
    fftRec_refl = m.sqrt(2) / fftRecLength * fftRec_refl[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec_refl = [x ** 2 for x in abs(fftRec_refl)]
    powerSpecRec_ratio = [refl/dir for refl, dir in zip(powerSpecRec_refl, powerSpecRec_dir)]
    pressureSpecRec_ratio = [refl/dir for refl, dir in zip(fftRec_refl, fftRec_dir)]

    #OCTAVE BANDS
    octave_levels_ratio = get_band_levels_1_1_oct_from_FIR(powerSpecRec_ratio, fs, freqRec, 1) #If the reference value is 1: then the level is calulated from a ratio value, not a pressure value
    octave_bands_ratio_all.append(octave_levels_ratio)

octave_bands_T = np.transpose(octave_bands_ratio_all)
avg_octaveband_values = [avg_db(band) for band in octave_bands_T]

#Plotting
fig, ax = plt.subplots()
linewidth = 4
#ax.plot(freqRec, [10*np.log10(f) for f in powerSpecRec_ratio], linestyle="--", linewidth=1, color="gray")
ax.plot(f_m_oct,avg_octaveband_values, linewidth=linewidth)
ax.legend(["Average Ground reflection IR (m = 88)", "1/1 octave band average"])
ax.set_xlabel("Frequency [Hz]")
ax.set_ylabel("dB")
```

```
plt.title("Average ground reflection IR simulation,\nMeasurement array, all rotations (m = 88)")
ax.set_xscale("log", base=2)
ax.set_xticks(f_m_oct)
ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
ax.set_ylim([-2,6])
ax.set_xlim([20, 20000])
fig.set_figwidth(8)
fig.set_figheight(5)
ax.grid(True)
plt.savefig("Plots/Ground_reflection_avg_allarraypositions.png", dpi=600)
plt.show()
```

H.7 ground_reflection_ears.py

This Python script simulates the interference effect from ground reflections in semi-anechoic chamber, for a receiver position at the musician's ears. The IR is calculated in octave bands from a full resolution interference curve.

```
import numpy as np
import matplotlib.pyplot as plt
from HRTF_average import get_octave_levels
from HRTF_average import get_octave_average_squared_levels
from HRTF_average import bandsAvgPowerRatioSpec
from A_weighting import *

fs = 48000
c = 343

listener_y_pos = 1.25
listener_x_pos = 0

source_y_pos = 0.835
source_x_pos = 0.3

dir_dist = np.sqrt((listener_y_pos-source_y_pos)**2 + (listener_x_pos-source_x_pos)**2)
refl_dist = np.sqrt((listener_y_pos+source_y_pos)**2 + (listener_x_pos-source_x_pos)**2)
print(dir_dist)
print(refl_dist)

# Create timevec and insert impulses:
time_vec = [0]*fs
dir_time = dir_dist/c
dir_n = round(dir_time*fs)
refl_time = refl_dist/c
refl_n = round(refl_time*fs)
time_vec[dir_n] = 1/dir_dist
time_vec_ref = [x for x in time_vec]
time_vec[refl_n] = 1/refl_dist #Without losses. Could potentially include air loss, but additional losses will also be present for higher frequencies due to body parts blocking the sound

# Full FFT
#direct sig
numZerosRec = 0
fftRec_dir = fft.fft(time_vec_ref, len(time_vec_ref) + numZerosRec)
fftRecLength = fftRec_dir.size
fftRec_dir = m.sqrt(2) / fftRecLength * fftRec_dir[0:fftRecLength // 2]
freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
powerSpecRec_dir = [x ** 2 for x in abs(fftRec_dir)]

#Reflected sig
numZerosRec = 0
fftRec_refl = fft.fft(time_vec, len(time_vec) + numZerosRec)
fftRecLength = fftRec_refl.size
fftRec_refl = m.sqrt(2) / fftRecLength * fftRec_refl[0:fftRecLength // 2]
freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
powerSpecRec_refl = [x ** 2 for x in abs(fftRec_refl)]

powerSpecRec_ratio = [refl/dir for refl, dir in zip(powerSpecRec_refl, powerSpecRec_dir)]
pressureSpecRec_ratio = [refl/dir for refl, dir in zip(fftRec_refl, fftRec_dir)]

def get_band_levels_1_1_oct_from_FIR(powerFIR, fs, freqRec, p0):
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
           250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
           16000, 20000, 25000]
```

```
    # Border frequencies
    f_lim = [0.9, 1.1, 1.4, 1.8, 2.2, 2.8, 3.5, 4.4, 5.6, 7, 8.9, 11.2, 14.1, 17.8, 22.4, 28.2, 35.5, 44.7, 56.2, 70.8,
             89.1, 112, 141, 178, 224, 282, 355, 447, 562, 708, 891, 1122, 1413, 1778, 2239, 2818, 3548, 4467, 5623,
             7079, 8913, 11220, 14130, 17780, 22390, 28184]

    # Octave band freq lists:
    f_m_oct = f_m[::3]
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680, 11360, 22000, 44000]

    # Creating the bins
    octPowerSpec = bandsAvgPowerRatioSpec(powerFIR, f_m_oct, f_lim_oct, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerSpec]
    return band_levels


#OCTAVE BANDS
octave_levels_ratio = get_band_levels_1_1_oct_from_FIR(powerSpecRec_ratio, fs, freqRec, 1) #If the reference value is 1: then the level is calulated from a ratio value, not a pressure value

#Plotting
fig, ax = plt.subplots()
linewidth = 4
ax.plot(freqRec, [10*np.log10(f) for f in powerSpecRec_ratio], linestyle="--", linewidth=1, color="gray")
ax.plot(f_m_oct, octave_levels_ratio, linewidth=linewidth)
ax.legend(["Ground reflection IR", "1/1 octave band average"])
ax.set_xlabel("Frequency [Hz]")
ax.set_ylabel("dB")
plt.title("Ground reflection IR from $\Delta r$ ="+str(round(refl_dist - dir_dist,2))+"m\nApprox. source-receiver distance at musicians' ears")
ax.set_xscale("log", base=2)
ax.set_xticks(f_m_oct)
ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
ax.set_ylim([-6,6])
ax.set_xlim([20, 20000])
fig.set_figwidth(10)
fig.set_figheight(7)
ax.grid(True)
plt.savefig("Plots/Ground_reflection_listener_ears.png", dpi=600)
plt.show()
```

H.8 HRTF_average.py

This Python-script imports and calculates the average HRTF offset at an incidence elevation angle of -45 degreed, from reference dataset provided by CIPIC [27].

```
import sofar as sf
import pyfar as pf
import numpy as np
import matplotlib as mpl
import matplotlib.pyplot as plt
from A_weighting import *
from functions_constants import *
import os

directory = "HRTF Database/CIPIC"
sofa_filenames = []

for filename in os.listdir(directory): #Retrieving CIPIC data from folder
    f = os.path.join(directory, filename)
    sofa_filenames.append(f)

def get_octave_levels(sig, fs):
    numZerosRec = 0
    fftRec = fft.fft(sig, len(sig) + numZerosRec)
    fftRecLength = fftRec.size
    fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec = [x ** 2 for x in abs(fftRec)]
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
```

```
            250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
            16000, 20000, 25000]
    # Octave band freq lists:
    f_m_oct = f_m[::3]
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680, 11360, 22000, 44000]
    # Creating the bins
    octPowerSpec = bandsPowerSpec(powerSpecRec, f_m_oct, f_lim_oct, freqRec)
    # Plotting the power spectrum in third octave bands:
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerSpec]
    return band_levels

def bandsAvgPowerRatioSpec(powerSpec, f_m, f_lim, freq):
    temp = []
    bandPowerSpec = [0]*len(f_m)
    n = 0
    for i in range(len(powerSpec)):
        if freq[i] < f_lim[n]:
            pass
        elif (freq[i]>=f_lim[n]) and (freq[i]<f_lim[n+1]) and (i<(len(freq)-1)):
            temp.append(powerSpec[i])
        elif (i<(len(freq)-1)):
            bandPowerSpec[n] = np.mean(temp)
            n += 1
            temp = []
            temp.append(powerSpec[i])
        else:
            temp.append(powerSpec[i])
            bandPowerSpec[n] = np.mean(temp)
    return bandPowerSpec


def get_octave_average_squared_levels(sig, fs):
    numZerosRec = 0
    fftRec = fft.fft(sig, len(sig) + numZerosRec)
    fftRecLength = fftRec.size
    fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
    freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
    powerSpecRec = [x ** 2 for x in abs(fftRec)]
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
            250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
            16000, 20000, 25000]
    # Octave band freq lists:
    f_m_oct = f_m[::3]
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680, 11360, 22000, 44000]
    # Creating the bins
    octPowerRatioSpec = bandsAvgPowerRatioSpec(powerSpecRec, f_m_oct, f_lim_oct, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerRatioSpec]
    return band_levels

def bandsPowerSpec_fs(powerSpec, f_m, f_lim, freq, fs):
    temp = 0
    bandPowerSpec = [0]*len(f_m)
    n = 0
    for i in range(len(powerSpec)):
        if freq[i] < f_lim[n]:
            pass
        elif (freq[i]>=f_lim[n]) and (freq[i]<f_lim[n+1]) and (i<(len(freq)-1)):
            temp += powerSpec[i]
        elif (i<(len(freq)-1)):
            bandPowerSpec[n] = temp
            n += 1
            temp = powerSpec[i]
        else:
            temp += powerSpec[i]
            bandPowerSpec[n] = temp
    return bandPowerSpec

def get_band_levels_1_1_oct_from_FIR(powerFIR, fs, freqRec, p0):
    # Creating 1/1 octave band list
    # Center frequencies
    f_m = [1, 1.25, 1.6, 2, 2.5, 3.16, 4, 5, 6.3, 8, 10, 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200,
            250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500,
            16000, 20000, 25000]
    # Octave band freq lists:
    f_m_oct = f_m[::3]
```

```
    f_m_oct.append(32000)
    f_lim_oct = [0.71, 1.42, 2.84, 5.68, 11, 22, 44, 88, 177, 355, 710, 1420, 2840, 5680,
        11360, 22000, 44000]
    # Creating the bins
    octPowerSpec = bandsAvgPowerRatioSpec(powerFIR, f_m_oct, f_lim_oct, freqRec)
    band_levels = [calculate_spl(m.sqrt(x), p0) for x in octPowerSpec]
    return band_levels

if __name__ == "__main__":
    all_the_oct_band_levels = []

    for i, file in enumerate(sofa_filenames):
        data_ir, source_coordinates, receiver_coordinates = pf.io.read_sofa(file)
        coor = source_coordinates[600].get_sph()[0] #Retrieving source coordinates of
            correct incidence angle
        coor = [np.rad2deg(coor[0]), np.rad2deg(coor[1]), coor[2]]
        coor = [round(x, 1) for x in coor]
        fs = round(data_ir[600].sampling_rate)
        numZerosPadding = 200*4

        # Left ear:
        fftRec = fft.fft(data_ir[600].time[0], len(data_ir[600].time[0]) +
            numZerosPadding)
        fftRecLength = fftRec.size
        fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
        freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
        fftRec = fftRecLength * fftRec/m.sqrt(2)
        powerSpecRatio_l = [x ** 2 for x in abs(fftRec)]
        oct_band_levels_l = get_band_levels_1_1_oct_from_FIR(powerSpecRatio_l, fs,
            freqRec,1) #Calculating octave band level offset

        #Right ear
        fftRec = fft.fft(data_ir[600].time[1], len(data_ir[600].time[1]) +
            numZerosPadding)
        fftRecLength = fftRec.size
        fftRec = m.sqrt(2) / fftRecLength * fftRec[0:fftRecLength // 2]
        freqRec = fft.fftshift(fft.fftfreq(fftRecLength, 1 / fs))[fftRecLength // 2:]
        fftRec = fftRecLength * fftRec / m.sqrt(2)
        powerSpecRatio_r = [x ** 2 for x in abs(fftRec)]
        oct_band_levels_r = get_band_levels_1_1_oct_from_FIR(powerSpecRatio_r, fs,
            freqRec, 1) #Calculating octave band level offset

        all_the_oct_band_levels.append(oct_band_levels_l)
        all_the_oct_band_levels.append(oct_band_levels_r)

    avg_octave_band_levels = [0]*len(f_m_oct)

    for i, band in enumerate(f_m_oct):
        level = avg_db([band_levels[i] for band_levels in all_the_oct_band_levels])
        avg_octave_band_levels[i] = level

    print(f_m_oct[7:-2])
    print(avg_octave_band_levels[7:-2])
    avg_octave_band_levels[0] = avg_octave_band_levels[1]

    #Plotting
    fig, ax = plt.subplots()
    linewidth = 4
    ax.plot(f_m_oct,avg_octave_band_levels, linewidth=linewidth)
    ax.set_xlabel("Frequency [Hz]")
    ax.set_ylabel("dB")
    ax.set_xscale("log", base=2)
    ax.set_xticks(f_m_oct)
    ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
    ax.set_ylim([-2,4])
    ax.set_xlim([250, 8000])
    ax.grid(True)
    ax.legend(["HRTF average offset curve, at -45 degree\nincidence (n = 90) (CIPIC)"])
    fig.set_figwidth(7)
    fig.set_figheight(4)
    fig.savefig("Plots/HRTF_offset_incidenceangle.png", dpi=600)
    plt.show()
```

H.9 import_scale.py

This Python-script defines the "Measurement" class used to import, scale and analyze measurements from various locations, while also calculating simple sound pressure level parameters. The class is dependent on a structured filing system in a "Location/Person/MeasurementType"-hierarchy, with consistent naming of the microphones.

```python
from A_weighting import *
from C_weighting import *
try:
    import cPickle as pickle
except:
    import pickle
from datetime import datetime

fs = 48000

class Measurement():
    def __init__(self, location, person, type, file, subtype=None, printstatus=False, overwrite=False):
        if subtype == None:
            subtype_string = ""
        else:
            subtype_string = subtype
        self.id = str(location+person+type+subtype_string+file)
        self.printstatus = printstatus
        try:
            if overwrite:
                if self.printstatus:
                    print("Manual overwrite active")
                raise OSError("Manual overwrite active")
            foo = pickle.load(open("pickle_database/"+self.id+".pickle", "rb"))
            if printstatus:
                print("Found and loaded ID:", self.id, " - Created", foo.timestamp)
            self.location = foo.location
            self.person = foo.person
            self.type = foo.type
            self.file = foo.file
            self.sig = foo.sig
            self.cal_sig = foo.cal_sig
            self.calibrated_status = foo.calibrated_status
            self.Lp_rms = foo.Lp_rms
            self.LpA_rms = foo.LpA_rms
            self.timestamp = foo.timestamp
            self.subtype = foo.subtype
            self.LAFMax = foo.LAFMax
            self.LCFMax = foo.LCFMax

        except (OSError, IOError) as e:
            self.location = location
            self.person = person
            self.type = type
            self.file = file
            self.subtype = subtype
            if type == "Lydeffekt" or type == "Skala": #The subtype variable is used differently in this case
                self.sig = importwav_soundpower(self.person, self.location, self.type, self.file, self.subtype)
            else:
                self.sig = importwav(self.person, self.location, self.type, self.file)
                if type == "Bakgrunnsst y":
                    start, stop = 0, int((len(self.sig)-1)/fs)
                elif subtype == None:
                    start, stop = get_start_stop_times(person, location, type, df)
                else:
                    start, stop = get_start_stop_times(person, location, subtype, df)
                self.slice(starttime=start, stoptime=stop)

            if file in quarterinch_and_lavalier_list:
                self.cal_dB = 91.9
            elif file[0] == "M":
                self.cal_dB = 92.1
            else:
                print("Invalid file name for determining calibration value")

            self.cal_sig = importwav(self.person, self.location, "Kalibrering", self.file)
            self.calibrated_status = False
            self.calibrate()
            self.timestamp = datetime.now()
            self.Lp_rms = None
            self.LpA_rms = None
            self.LAFMax = None
            self.LCFMax = None

    def calibrate(self):
        self.sig = self.sig * get_scale_factor(self.cal_sig, self.cal_dB)
        self.calibrated_status = True

    def slice(self, starttime, stoptime):
        startidx = starttime * fs
        stopidx = stoptime * fs
        self.sig = self.sig[startidx:stopidx]
```

```
    def get_calibrated_status(self):
        return self.calibrated_status

    def get_Lp_rms(self):
        if not self.calibrated_status:
            print("Not calibrated")
            pass
        elif self.Lp_rms != None:
            if self.printstatus:
                print("Lp", self.file, ":", round(self.Lp_rms, 1), "dB re 20 \u03BCPa.")
            return self.Lp_rms
        else:
            p_rms = calculate_rms(self.sig)
            Lp_rms = amp2Lp(p_rms)
            if self.printstatus:
                print("Lp", self.file, ":", round(Lp_rms,1), "dB re 20 \u03BCPa.")
            self.Lp_rms = Lp_rms
            return Lp_rms

    def get_LpA_rms(self):
        if not self.calibrated_status:
            print("Not calibrated")
            pass
        elif self.LpA_rms != None:
            if self.printstatus:
                print("LpA", self.file, ":", round(self.LpA_rms, 1), "dB re 20 \u03BCPa."
                    )
            return self.LpA_rms
        else:
            a_weighted_levels = get_A_band_levels(self.sig)
            LpA_rms = 10 * m.log10(sum([10 ** (x / 10) for x in a_weighted_levels]))
            if self.printstatus:
                print("LpA", self.file, ":", round(LpA_rms, 1), "dB re 20 \u03BCPa.")
            self.LpA_rms = LpA_rms
            return LpA_rms

    def get_LAFMax(self):
        if not self.calibrated_status:
            print("Not calibrated")
            pass
        elif self.LAFMax != None:
            if self.printstatus:
                print("LAFMax", self.file, ":", round(self.LAFMax, 1), "dB re 20 \u03BCPa
                    .")
            return self.LAFMax
        else:
            LAFMax = []
            samples = 0.125 * fs
            count = 0
            tempsig = []
            for sample in self.sig:
                if count<samples:
                    tempsig.append(sample)
                    count +=1
                else:
                    a_weighted_levels = get_A_band_levels(tempsig)
                    LpA_window = 10 * m.log10(sum([10 ** (x / 10) for x in
                        a_weighted_levels]))
                    LAFMax.append(LpA_window)
                    count = 0
                    tempsig = []
            self.LAFMax = LAFMax
            if self.printstatus:
                print("LAFMax", self.file, ":", round(LAFMax, 1), "dB re 20 \u03BCPa.")
            return LAFMax

    def get_LCFMax(self):
        if not self.calibrated_status:
            print("Not calibrated")
            pass
        elif self.LCFMax != None:
            if self.printstatus:
                print("LCFMax", self.file, ":", round(self.LCFMax, 1), "dB re 20 \u03BCPa
                    .")
            return self.LCFMax
        else:
            LCFMax = []
            samples = 0.125 * fs
            count = 0
            tempsig = []
            for sample in self.sig:
                if count<samples:
                    tempsig.append(sample)
                    count +=1
                else:
```

```python
                    c_weighted_levels = get_C_band_levels(tempsig)
                    LpC_window = 10 * m.log10(sum([10 ** (x / 10) for x in c_weighted_levels]))
                    LCFMax.append(LpC_window)
                    count = 0
                    tempsig = []
            self.LCFMax = LCFMax
            if self.printstatus:
                print("LCFMax", self.file, ":", round(LCFMax, 1), "dB re 20 \u03BCPa.")
            return LCFMax

    def get_signal(self):
        return self.sig
    def get_duration(self):
        return len(self.sig)/fs
    def plot_cal_sig(self):
        plt.plot(self.cal_sig)
        plt.show()
    def save(self):
        pickle.dump(self, open("pickle_database/"+self.id+".pickle", "wb"))
    def __del__(self):
        self.save()

class Measurements(): #Multiclass used for group of Measurement instances
    def __init__(self, measurements: list[Measurement]):
        self.measurements = measurements
    def get_avg_Lp(self):
        Lp_values = []
        for m in self.measurements:
            Lp_values.append(m.get_Lp_rms())
        Lp_avg = 10*np.log10(np.mean([10**(Lp/10) for Lp in Lp_values]))
        print("Average Lp:", round(Lp_avg,1))
        return Lp_avg

    def get_avg_LpA(self):
        LpA_values = []
        for m in self.measurements:
            LpA_values.append(m.get_LpA_rms())
        LpA_avg = 10*np.log10(np.mean([10**(LpA/10) for LpA in LpA_values]))
        print("Average LpA:", round(LpA_avg,1))
        return LpA_avg

    def get_measurements(self):
        return self.measurements
    def get_single_Lp_values(self):
        return [m.get_Lp_rms() for m in self.measurements]
    def get_single_LpA_values(self):
        return [m.get_LpA_rms() for m in self.measurements]
    def get_single_LAFMax_values(self):
        return [m.get_LAFMax() for m in self.measurements]
    def get_single_LCFMax_values(self):
        return [m.get_LCFMax() for m in self.measurements]
    def get_duration(self):
        duration_list = [m.get_duration() for m in self.measurements]
        if max(duration_list) == min(duration_list):
            return duration_list[0]
        else:
            print("Measurements have different durations")
            pass

# Global functions related to Measurements
def get_Lp_mic_measurements(person, location, type, subtype=None, printstatus=False, overwrite=False):
    global filelist
    filelist_Lpmics = filelist[-4:]
    Lpmic_list = []
    for file in filelist_Lpmics:
        Lpmic = Measurement(person=person, location=location, type=type, subtype=subtype, file=file, printstatus=printstatus, overwrite=overwrite)
        Lpmic_list.append(Lpmic)
    return Measurements(Lpmic_list)

def get_m_array_measurements(person, location, type, subtype=None, printstatus=False, overwrite=False):
    global filelist
    filelist_m_array = filelist[:-4]
    m_list = []
    for file in filelist_m_array:
        if file[0] == "M":
            m = Measurement(person=person, location=location, type=type, subtype=subtype, file=file, printstatus=printstatus, overwrite=overwrite)
            m_list.append(m)
    return Measurements(m_list)
```

H.10 max_peak_levels.py

This Python-script calculates the maximum, C-weighted peak levels from a given calibrated signal.

```
from functions_constants import *
from scipy.signal import find_peaks
from import_scale import *
from numpy import pi, polymul
from scipy.signal import bilinear
from scipy.signal import lfilter

def C_weighting_filter(fs): #Returns filter coefficients as given in IEC 61672-1:2013
    C1 = 20.6
    C2 = 12194
    C3 = 0.0619

    D = polymul([1,4*pi*C2,(2*pi*C2)**2.0],[1,4*pi*C1,(2*pi*C1)**2])
    N = [(2 * pi * C2) ** 2 * (10 ** (C3 / 20.0)), 0, 0]
    return bilinear(N,D,fs)

for location in ["SemiAnechoic", " vingsrom "]:
    subtype_list = []
    if location == "SemiAnechoic":
        type_ = "Lydeffekt"
        for deg in [0, 45, 90, 135, 180, 225, 270, 315]:
            subtype_list.append(str(deg)+ " Opp")
            subtype_list.append(str(deg) + " Ned")

    else:
        type_ = "Skala"
        for take in takelist:
            subtype_list.append(take+ " Opp")
            subtype_list.append(take + " Ned")

    for person in ["Person1", "Person2", "Person3", "Person4"]:
        for subtype in subtype_list:
            Lp_measurements = get_Lp_mic_measurements(person=person, location=location,
                type=type_, subtype=subtype)

            indent()
            print(location, person, type_, subtype)

            # Left Ear
            leftsig = Lp_measurements.get_measurements()[0].get_signal()
            b, a = C_weighting_filter(fs)
            leftsig = lfilter(b, a, leftsig) # Applying c-weighting filter
            peak_indices_pos = find_peaks(leftsig, distance=48000)[0]
            peak_indices_neg = find_peaks(-leftsig, distance=48000)[0]
            peak_values = [leftsig[i] for i in peak_indices_pos]
            for i in peak_indices_neg:
                peak_values.append(-leftsig[i])
            max_peak_left = max(peak_values)
            max_peak_SPL_left = amp2Lp(max_peak_left)

            # Right Ear
            rightsig = Lp_measurements.get_measurements()[1].get_signal()
            b, a = C_weighting_filter(fs)
            rightsig = lfilter(b, a, rightsig) # Applying c-weighting filter
            peak_indices_pos = find_peaks(rightsig, distance=48000)[0]
            peak_indices_neg = find_peaks(-rightsig, distance=48000)[0]
            peak_values = [rightsig[i] for i in peak_indices_pos]
            for i in peak_indices_neg:
                peak_values.append(-rightsig[i])
            max_peak_right = max(peak_values)
            max_peak_SPL_right = amp2Lp(max_peak_right)

            #Evaluating maximum level among ears
            if max_peak_left > max_peak_right:
                max_peak_SPL = amp2Lp(max_peak_left)
                max_ear = "Left"
            else:
                max_peak_SPL = amp2Lp(max_peak_right)
                max_ear = "Right"

            print("L_Cpeak = ", round(max_peak_SPL, 1), "dB -", max_ear)
```

H.11 soundpower_measurements.py

This Python-script calculates the SWL, SPL and power-pressure difference for test signals in octave bands, for all rotations of the source.

```
import numpy as np
from functions_constants import *
from matplotlib import pyplot as plt
import pandas as pd
from matplotlib import ticker
import scipy.stats as st
from scipy.stats import t

persons = ["Person1", "Person2", "Person3", "Person4"]

S = 2*np.pi*1.670**2 #Measurement surface
S0 = 1

df = pd.read_csv("Lydeffekt resultater 1_1oktavb nd.csv") #Retrieving data

all_LwA_band_values = []
all_LpA_left_band_values = []
all_LpA_right_band_values = []
all_LpA_ears_avg_band_values = []
all_LwA_minus_LpA_band_values = []

def CI_95percent(vals): #95% confidence intervals
    limits = t.interval(0.95, df=len(vals)-1, loc=np.mean(vals), scale=st.sem(vals))
    CI = (limits[1]-limits[0])/2
    return CI

for person in persons:
    for angle in anglelist:
        LpA_band_values = []
        LpA_left_values = []
        LpA_right_values = []
        LpA_ears_values = []

        for f in f_m_oct:
            colstring = "LpA " + str(f)
            m_values = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] != "Left") & (df["File"] != "Right") & (df["File"] != "Clips") & (df["File"] != "1.5m ref"), colstring].tolist()
            LpA_band_values.append(avg_db(m_values))

            LpA_left = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] == "Left"), colstring]
            LpA_right = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] == "Right"), colstring]

            LpA_left_values.append(LpA_left)
            LpA_right_values.append(LpA_right)
            LpA_ears_values.append(avg_db([LpA_left, LpA_right]))

        LwA_band_values = LpA_band_values + 10 * np.log10(S / S0) #Calculating LWA from LpA
        all_LwA_band_values.append([x for x in LwA_band_values])
        LwA_minus_LpA_values = [LwA - LpA for LwA, LpA in zip(LwA_band_values, LpA_ears_values)]
        all_LpA_left_band_values.append([x for x in LpA_left_values])
        all_LpA_right_band_values.append([x for x in LpA_right_values])
        all_LpA_ears_avg_band_values.append([x for x in LpA_ears_values])
        all_LwA_minus_LpA_band_values.append([x for x in LwA_minus_LpA_values])

        indent()
        print(person, angle)
        print("Freq:", f_m_oct)
        print("LwA:", LwA_band_values)
        print("LpA ears:", LpA_ears_values)
        print("LwA - LpA:", LwA_minus_LpA_values)

indent()
print("Freq:", f_m_oct)
print("LwA:", all_LwA_band_values)
print("LpA ears:", all_LpA_ears_avg_band_values)
print("LwA - LpA:", all_LwA_minus_LpA_band_values)

LwA_T = np.transpose(all_LwA_band_values)
LpA_ears_T = np.transpose(all_LpA_ears_avg_band_values)
LwA_minus_LpA_T = np.transpose(all_LwA_minus_LpA_band_values)

#Averaging
average_LwA = [avg_db(band) for band in LwA_T]
average_LpA_ears = [avg_db(band) for band in LpA_ears_T]
```

```python
average_LwA_minus_LpA = [avg_db(band) for band in LwA_minus_LpA_T]

band_SD_LwA = [np.std(band) for band in LwA_T]
band_SD_LpA_ears = [np.std(band) for band in LpA_ears_T]
band_SD_LwA_minus_LpA = [np.std(band) for band in LwA_minus_LpA_T]


for i, f_m in enumerate(f_m_oct):
    if i > 5: #Retrieving the CI values for relevant octave bands
        print(f_m, "Hz")
        print("LwA-LpA:", round(average_LwA_minus_LpA[i],1))
        CI = CI_95percent(LwA_minus_LpA_T[i].tolist())
        print("CI LwA-LpA 95%:", round(CI,1))

        print("LwA:", round(average_LwA[i]))
        CI = CI_95percent(LwA_T[i].tolist())
        print("CI LwA 95%:", round(CI,1))

        print("LpA:", round(average_LpA_ears[i]))
        CI = CI_95percent(LpA_ears_T[i].tolist())
        print("CI LpA 95%:", round(CI,1))
        indent()

#Plotting LWA
fig, ax = plt.subplots()
fig.set_figwidth(10)
fig.set_figheight(6)
linewidth = 4
ax.plot(f_m_oct[8:13],average_LwA[8:13], linewidth=linewidth)
w = 0.1
width = lambda p, w: 10**(np.log10(p)+w/2.)-10**(np.log10(p)-w/2.)
ax.boxplot(LwA_T.tolist(), positions=f_m_oct, widths=width(f_m_oct,w))
ax.set_xlabel("Frequency [Hz]")
ax.set_ylabel("dB re 1pW")
ax.legend(["$L_{WA}$ (m=32)"])
ax.set_xscale("log", base=2)
ax.set_xticks(f_m_oct)
ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
ax.set_ylim([65,100])
ax.set_xlim([200, 5000])
ax.grid(True)
plt.savefig('Plots/LWA_octave.png', dpi=600)

#Plotting LpA (ears avg)
fig, ax = plt.subplots()
fig.set_figwidth(10)
fig.set_figheight(6)
linewidth = 4
ax.plot(f_m_oct[8:13],average_LpA_ears[8:13], linewidth=linewidth)
w = 0.1
width = lambda p, w: 10**(np.log10(p)+w/2.)-10**(np.log10(p)-w/2.)
ax.boxplot(LpA_ears_T.tolist(), positions=f_m_oct, widths=width(f_m_oct,w))
ax.set_xlabel("Frequency [Hz]")
ax.set_ylabel("dB re 20 \u03BCPa")
ax.legend(["$L_{A,\mathrm{eq25s,ears}}$ (m=32)"])
ax.set_xscale("log", base=2)
ax.set_xticks(f_m_oct)
ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
ax.set_ylim([55,95])
ax.set_xlim([200, 5000])
ax.grid(True)
plt.savefig('Plots/LPA_ears_octave.png', dpi=600)

#Plotting LWA - LpA (ears avg)
fig, ax = plt.subplots()
fig.set_figwidth(10)
fig.set_figheight(6)
linewidth = 4
ax.plot(f_m_oct[8:13],average_LwA_minus_LpA[8:13], linewidth=linewidth)
w = 0.1
width = lambda p, w: 10**(np.log10(p)+w/2.)-10**(np.log10(p)-w/2.)
ax.boxplot(LwA_minus_LpA_T.tolist(), positions=f_m_oct, widths=width(f_m_oct,w))
ax.set_xlabel("Frequency [Hz]")
ax.set_ylabel("dB")
ax.legend(["$L_{WA} - L_{A,\mathrm{eq25s,ears}}$ (m=32)"])
ax.set_xscale("log", base=2)
ax.set_xticks(f_m_oct)
ax.xaxis.set_major_formatter(ticker.FormatStrFormatter("%0d"))
ax.set_ylim([0,14])
ax.set_xlim([200, 5000])
ax.grid(True)
plt.savefig('Plots/LWA_minus_LPA_octave.png', dpi=600)
plt.show()
```

H.12 soundpower_0degree_comparison.py

This Python-script calculates the SWL, SPL and power-pressure difference for test signals in semi-anechoic chamber, in octave bands, for all rotations of the source. The script then compares estimations from measurements in the single front angle (m = 4) and from all rotations (m = 32).

```
import numpy as np
from functions_constants import *
import pandas as pd
import scipy.stats as st
from scipy.stats import t

persons = ["Person1", "Person2", "Person3", "Person4"]

def conf_interval_95_student_t(vals): #Get 95% confidence intervals
    limits = t.interval(0.95, df=len(vals) - 1, loc=np.mean(vals), scale=st.sem(vals))
    CI = (limits[1] - limits[0]) / 2
    return CI

S = 2*np.pi*1.670**2 #Measurement surface
S0 = 1

df = pd.read_csv("Lydeffekt resultater 1_1oktavb nd.csv") #Retrieve data

all_LwA_band_values = []
all_LpA_left_band_values = []
all_LpA_right_band_values = []
all_LpA_ears_avg_band_values = []
all_LwA_minus_LpA_band_values = []

zerodeg_LwA_minus_LpA_band_values = []

for person in persons:
    for angle in anglelist:
        LpA_band_values = []
        LpA_left_values = []
        LpA_right_values = []
        LpA_ears_values = []

        for f in f_m_oct:
            colstring = "LpA " + str(f)
            m_values = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] != "Left") & (df["File"] != "Right") & (df["File"] != "Clips") & (df["File"] != "1.5m ref"), colstring].tolist()
            LpA_band_values.append(avg_db(m_values))

            LpA_left = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] == "Left"), colstring]
            LpA_right = df.loc[(df["Person"] == person) & (df["Degrees"] == int(angle)) & (df["File"] == "Right"), colstring]

            LpA_left_values.append(LpA_left)
            LpA_right_values.append(LpA_right)
            LpA_ears_values.append(avg_db([LpA_left, LpA_right]))

        LwA_band_values = LpA_band_values + 10 * np.log10(S / S0) #Calculating LWA from LpA
        all_LwA_band_values.append([x for x in LwA_band_values])
        LwA_minus_LpA_values = [LwA - LpA for LwA, LpA in zip(LwA_band_values, LpA_ears_values)]

        all_LpA_left_band_values.append([x for x in LpA_left_values])
        all_LpA_right_band_values.append([x for x in LpA_right_values])
        all_LpA_ears_avg_band_values.append([x for x in LpA_ears_values])
        all_LwA_minus_LpA_band_values.append([x for x in LwA_minus_LpA_values])

        if int(angle) == 0:  #Collecting the data only from the front source rotation angle
            zerodeg_LwA_minus_LpA_band_values.append([x for x in LwA_minus_LpA_values])

LwA_T = np.transpose(all_LwA_band_values)
LpA_ears_T = np.transpose(all_LpA_ears_avg_band_values)
LwA_minus_LpA_T = np.transpose(all_LwA_minus_LpA_band_values)
zerodeg_LwA_minus_LpA_T = np.transpose(zerodeg_LwA_minus_LpA_band_values)

#Averaging values
average_LwA = [avg_db(band) for band in LwA_T]
average_LpA_ears = [avg_db(band) for band in LpA_ears_T]
average_LwA_minus_LpA = [avg_db(band) for band in LwA_minus_LpA_T]
average_zerodeg_LwA_minus_LpA = [avg_db(band) for band in zerodeg_LwA_minus_LpA_T]

band_SD_LwA = [np.std(band) for band in LwA_T]
band_SD_LpA_ears = [np.std(band) for band in LpA_ears_T]
band_SD_LwA_minus_LpA = [np.std(band) for band in LwA_minus_LpA_T]
```

```
band_SD_zerodeg_LwA_minus_LpA = [np.std(band) for band in zerodeg_LwA_minus_LpA_T]

start, stop = 8,14  #Only displaying octave bands of interest
indent()
percentage_deviations = [round(((zero/all_)-1)*100,1) for zero, all_ in zip(average_zerodeg_LwA_minus_LpA, average_LwA_minus_LpA)]
print("Bands:", f_m_oct[start:stop])
print("LwA-LpA all:", [round(r,1) for r in average_LwA_minus_LpA[start:stop]])
print("LwA-LpA 0deg:", [round(r,1) for r in average_zerodeg_LwA_minus_LpA[start:stop]])
print("Percentage deviation:", [round(r,1) for r in percentage_deviations[start:stop]])
indent()

print("SD all:", band_SD_LwA_minus_LpA[start:stop])
print("SD 0deg:",band_SD_zerodeg_LwA_minus_LpA[start:stop])

CI_all_LpA_minus_LwA = [conf_interval_95_student_t(band) for band in LwA_minus_LpA_T]
CI_zerodeg_LpA_minus_LwA = [conf_interval_95_student_t(band) for band in zerodeg_LwA_minus_LpA_T]

print("95% CI (+/-) all:", [round(r,1) for r in CI_all_LpA_minus_LwA[start:stop]])
print("95% CI (+/-) 0deg:", [round(r,1) for r in CI_zerodeg_LpA_minus_LwA[start:stop]])
```

H.13 test_signal_practiceroom.py

This Python-script imports and averages the SPL for test signals measured in the practice room (m = 12).

```
import numpy as np
from import_scale import *
from functions_constants import *
import pandas as pd

df = pd.read_csv(' vingsrom  skala resultater - oktavb nd.csv') #Open file for writing

#Retrieve info for the given location and person
person = "Person4"
location = " vingsrom "

printstatus=False
overwrite=True

for take in takelist:
    # Close-mics
    Lp_up = get_Lp_mic_measurements(person=person, location=location, type="Skala", subtype=take + " Opp", printstatus=printstatus, overwrite=overwrite)
    Lp_down = get_Lp_mic_measurements(person=person, location=location, type="Skala", subtype=take + " Ned", printstatus=printstatus, overwrite=overwrite)

    sig_up = [Lp.get_signal() for Lp in Lp_up.get_measurements()]
    sig_down = [Lp.get_signal() for Lp in Lp_down.get_measurements()]

    LpA_closemic_levels_1_1oct_up = [get_A_band_levels_1_1_oct(sig) for sig in sig_up]
    LpA_closemic_levels_1_1oct_down = [get_A_band_levels_1_1_oct(sig) for sig in sig_down]

    up_T = np.transpose(LpA_closemic_levels_1_1oct_up)
    down_T = np.transpose(LpA_closemic_levels_1_1oct_down)

    #Average levels
    LpA_closemic_bandlevels_scale_run = [[avg_db([upval, downval]) for upval, downval in zip(up_T_Lp, down_T_Lp)] for
                                          up_T_Lp, down_T_Lp in zip(up_T, down_T)]
    LpA_closemic_bandlevels_scale_run = np.transpose(LpA_closemic_bandlevels_scale_run)

    for Lp, Lp_name in zip(LpA_closemic_bandlevels_scale_run, quarterinch_and_lavalier_list):
        for val, f in zip(Lp, f_m_oct):
            colstring = "LpA " + str(f)
            df.loc[(df["Person"] == str(person)) & (df["File"] == str(Lp_name)) & (df["Take"] == str(take)), colstring] = val

#Write results to file
df.to_csv(' vingsrom  skala resultater - oktavb nd.csv', index=False)
```

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