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May 22 - 24 2024 Espoo, Finland

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WELCOME TO ESPOO

On behalf of the organizing committee for BNAM2024, I would like to welcome you all to Hanasaari, Espoo.

We have attempted to change the concept of the Baltic-Nordic Acoustics Meeting a bit compared to previous conferences. Traditionally, BNAM gathers many acoustics consultants and practitioners, so we have organized numerous workshop with practical, hands on session for different topics which we think will be interesting. Naturally, we also have a scientific program with interesting papers, covering nearly all areas of acoustics.

We are all looking forward to seeing you at the conference and hope that we will have three fruitful days of presentations, workshop, tradeshow and of course perhaps most important, meeting colleagues, discussion and networking.

On behalf of the organizing committee

Henrik Möller

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BALTIC-NORDIC ACOUSTICS MEETING 2024 – PROGRAM

BNAM 2024, May 22-24, Hanasaari, Espoo				
Wednesday 22.5.2024				
	Celsius		Kullager	
8:30	OPENING			
8:40	Steindór Gudmundsson (Keynote)	Acoustic regulations and quality classes -- Experience from over 35 years of collaboration in the nordic countries		
	Room Acoustics, Chair: Henrik Möller, Bård Støfringsdal, Tapio Lokki		Environmental noise measurements and simulations + industrial noise, Chairs: Laimonas Ratkevičius, Deniss Mironovs	
9:20	Marko Horvat and Kristian Jambrošić	Acoustic intervention in the French pavilion of the Student Centre in Zagreb as a protected cultural heritage building	Jarkko Keinänen, Olli Malmi, Kari Saine, Claus Paro, Aki Kinnunen and Pauli Valkjärvi	Dynamic properties of wire rope and rubber isolators at high frequency range in engine isolation
9:40	Yann Jurkiewicz, Vincent Berrier and Eckhard Kahle	Turku Fuuga – Acoustic Design of an Intimate and Immersive Concert Hall	Daudel Tchatat Ngaha and Kjell Eivind Frøysa	Effects of non-uniform flow on a sound field generated by a finite-size uniform piston
10:00		Company introductions, 2min / exhibitor, first 14 companies		
10:30	coffee	Tetra	coffee	Tetra
11:00		Company introductions, 2min / exhibitor, last 9 companies		
11:20	Octávio Inácio, Filipe Martins and Daniel José	Finite-difference time domain calculations of acoustic phenomena applied to concert hall design	Kari Saine, Antti Leskinen, Tero Korhonen, Roy Hjort and Zengxin Gao	Advanced method for workplace noise analysis
11:40	Ólafur Pjetursson, Finnur Pind and Cheol-Ho Jeong	New Opportunities in Room Acoustics Simulations Using Wave Based Technology	Christian Bergfjord Mørck, Nicolas Sogg, Helena Gabriella Axelssone and Ingunn Milford	Comparison of results from road noise measurements and road noise calculation methods CNOSSOS-EU, Nord2000Road and NBV96
12:00	Henrik Möller, Jukka Pätynen and Sami Reina	Renovating the Encore hall using electro-acoustic enhancement systems	Johan Hallimäe	A city designed using sound - A tool to make spatial decisions
12:20	Lukasz Blasinski and Jedrzej Kocinski	Perception of reverberation length in rooms with an Reverberation Enhancement System	Laimonas Ratkevičius and Steve Mitchell	Aircraft noise modelling with AEDT
12:40	Matias Remes and Perttu Korhonen	Renovation of Finnish Modern Theatres – Acousticians' Experiences from the Past 10 Years	Unto K. Laine	Magneto-acoustic triangulation method for electric discharge localization in the atmosphere
13:00	John O'Keefe	Applications of a Zone to Zone Reflector Optimisation Routine	Mikko Kylliäinen	Finnish acoustician Paavo Arni (1905–1969)
13:20	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
13:40	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
14:00	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
14:20	Jens Holger Rindel (Keynote)	Room acoustic measurement methods in the past, present and future, including the importance of the ISO 3382 series		
15:00	Henrik Möller and Łukasz Błasiński	Room acoustic measurements in halls with electro-acoustic enhancement systems	Jukka Pätynen, Python Workshop	
15:20	Petri Lehto, Henrik Möller, Jukka Pätynen, Javier Bolanos, Perttu Laukkanen, and Sara Vehviläinen	Fine tuning of the Sibelius hall stage acoustics		
15:40	Jorge Torres Gomez, Magne Skålevik, and Mattias Hill	Temperature effects on ice rink sports halls		
16:00	coffee	Tetra	coffee	Tetra
16:20	coffee	Tetra	coffee	Tetra
16:40	Anatoly Livshits, Aleksander Fadeev and Natalia Shirgina	Assessment of the influence of the method of fastening decorative and finishing panels on the fund of low-frequency sound absorption in a concert hall	Jukka Pätynen, Python Workshop continues	
17:00	Simen Helbæk Kjølberg	Sound Levels at a Clarinetist's Ears During Solitary Practice		
17:20	Andrzej Klosak, Bartlomiej Ziarko, Katarzyna Dusza and Dominika Woźniak	Polish National Television production studios: acoustic design and performance		

Thursday 23.5.2024					
	Celsius		Kullager		Blixtlås
	Building acoustic regulations and classification in the Nordic and Baltic countries, Chairs: Birgit Rasmussen, Tønnes A. Ognedal		Workshops		Workshops
8:30			Claus Lynge Christensen: ODEON		Torbjörn Kloow, Kari Pesonen: Dosimeter Workshop
8:50	Birgit Rasmussen and Claus Møller Petersen	Compliance procedures for sound insulation between dwellings in new housing – Rules according to Danish regulations & Experiences from practice			
9:10	Manu Rönkkö and Liisa Kilpilehto	Outdoor event noise levels and limits			
9:30	Birgit Rasmussen, Liisa Sell and Lars Sommer Søndergaard	Field tests of low noise levels from MVHR ventilation systems – Overview obstacles and pilot test of test procedure improvement			
9:50	Hassan Al-Ramadani	Acoustics in green buildings			
10:10	Mikko Kylliäinen, Simo Laitakari, Timo Huhtala, Matias Remes, Pekka Taina, Johannes Usano, Ville Veijanen, Janne Hautsalo and Oskar Lindfors	Revised Finnish standard SFS 5907:2022 on acoustical design and quality classes of buildings			
10:30	coffee	Tetra	coffee	Tetra	
10:50	coffee	Tetra	coffee	Tetra	
11:10	coffee	Tetra	Birgit Rasmusen, Tønnes A. Ognedal: Building acoustic regulations and classification workshop		
	Field measurements of Room acoustic parameters – Control measurements, Chairs: Mads Bollberg, Magne Skålevik				
11:30	Karin Norén-Cosgriff and Jörgen Johansson	Guideline limit values for vibration to avoid damage to structures and natural slopes. Revision of Norwegian Standard NS 8141			
11:50	Deniss Mironovs and Olivers Tarvids	Immersive sound system showroom acoustical design in existing industrial premises			
12:10	Maria Quinn and Anne Pollet	Acoustics in a modular operating theatre			
12:30	Mads Bolberg and Ingvar Jónsson	Effect of furniture in reverberation time measurements			
12:50	Mads Bolberg and Ingvar Jónsson	Repairs on rendered sound absorptive ceilings and the effect on their acoustic performance			
13:10	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)	
13:30	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)	
13:50	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)	
14:10	**Tønnes A. Ognedal (Keynote)**	**Acoustic properties to be included with new development of housing sales reports?**			
	Field measurements of Room acoustic parameters – Control measurements, Chairs: Mads Bollberg, Magne Skålevik		Workshops		
14:50	Christina Kjær, Christer Volk and Cheol-Ho Jeong	Perceptual Evaluation of Room Acoustic Simulation and Measurements	Patrick Grahn: COMSOL workshop		BUS TOUR, "CITY SOUNDSCAPE EXHIBITION"
15:10	Łukasz Błasiński and Jędrzej Kociński	Localisation of sound sources reproduced by immersive and stereo sound system			
15:30	Henrik Möller and Jukka Pätynen	Spatial acoustic measurements in concert halls with a reduced virtual orchestra			
15:50	coffee	Tetra	coffee	Tetra	
16:10	coffee	Tetra	coffee	Tetra	
16:30	Thomas Rittenschober and Mikko Halonen	Robust 3D Localisation of Anomalies in the Reverbaration Time Signal	Patrick Grahn: COMSOL workshop continues		
	Vibration in infrastructure, Chair: Mats Heikkinen				
16:50	Minna Santaholma, Timo Peltonen, Mats Heikkinen, Jukka Pätynen and Lauri Vapalahti	Optimising Railway Track Vibration Isolation by Matching Ground-Borne Noise Level Data to Train Location in a Tunnel			Torbjörn Kloow, Kari Pesonen: Dosimeter Workshop
17:10	Pekka Taina, Vesa Vähäkuopus and Jarkko Punnonen	Pile supported slab mitigating vibration under and next to a railway line			
18:00	**Dinner, arriving from 18.00, start at 19.00**	**Vanha Ylioppilastalo, Helsinki**	**Dinner**	**Vanha Ylioppilastalo, Helsinki**	

Friday 24.5.2024				
	Celsius		Kullager	
	Health effects of noise, Chair: Valtteri Hongisto, Maria Quinn		**Workshops**	
8:30	Magne Skålevik	Sound levels in symphony orchestra musicians		
8:50	Valtteri Hongisto, Jukka Keränen and Jenni Radun	Active noise-cancelling headphones: influence on performance, stress, and experience in work context	Finnur Pind: TREBLE, room acoustics modeling workshop	
9:10	Antti Kuusinen and Valtteri Hongisto	Predictors of noise annoyance and penalty of spectrally different wideband noises		
9:30	Lars Bramsløw	An auditory loudness model with hearing loss		
9:50	Valtteri Hongisto, Henna Maula and Jenni Radun	Stress effects of impulsive noise - a medical laboratory experiment		
10:10	Veronica Amodeo, Simone Secchi and Luca Marzi	Acoustic comfort assessment in hospital wards: measuring procedures and parameters		
10:30	Valtteri Hongisto, Reijo Alakoivu, Antti Kuusinen and Jukka Keränen	Psychoacoustic experiment in BNAM 2024 conference		
10:50	coffee	Tetra	coffee	Tetra
11:10	coffee	Tetra	coffee	Tetra
11:30	**Klas Hagberg (Keynote)**	**Sustainable wooden buildings -- opportunities for good indoor acoustics**		
	Building acoustics and noise from service equipment in buildings: modelling, calculations and measurements, Chair: Johannes Usano		**Workshops**	
12:10	Ville Kovalainen, Jesse Lietzén, Benjamin Oksanen and Giovanni Hawkins	Structure-borne vibration generated by a pallet jack exiting a service elevator	TBA: SoundPLAN workshop	
12:30	Jukka Keränen, Valtteri Hongisto and Giovanni Hawkins	Survey measurement of impact sound insulation of concrete walls		
12:50	Mikko Mantri Roininen, Oskar Lindfors and Mats Heikkinen	Gymnasium activity noise in residential and educational buildings		
13:10	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
13:30	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
13:50	lunch	Restaurant (2nd floor)	lunch	Restaurant (2nd floor)
14:10	Archil Cheghelidze, Nikoloz Tchegelidze and Henrik Möller	Acoustic design of Hotels - Comparison between different hotel brand acoustic requirements	Johan Hallimäe: Rhino/Grasshopper workshop	
14:30	Jose Cucharero, Kari Kammiovirta, Marko Makkonen, Tuomas Hänninen and Tapio Lokki	The end justifies the means – Sprayed sound-absorbing coating on non-acoustic materials.		
14:50	Johannes Usano and Joona Koskimäki	Structure borne noise emitted by building service equipment: laboratory measurements and modelling		
	Wooden buildings, Chair: Klas Hagberg, Alain Bradette			
15:10	Marina Rodrigues, Paulo Pinto and Reinhilde Lanoye	Harmonizing sustainability & acoustics: challenges in mass timber construction		
15:30	Paola Brugnara, Alice Speranza, Luca Barbaresi, Vincenzo Pettoni Possenti and Chiara Trucchi	Improving sound transmission in timber buildings: the role of flexible interlayers		
15:50	Jesse Lietzén, Ville Kovalainen, Mikko Kylliäinen and Sami Pajunen	FEM-based simulation procedure to predict impact sound insulation of a timber floor		
16:10	**Closing**			

BALTIC-NORDIC ACOUSTICS MEETING 2024 – PAPERS

KEYNOTES

ENVIRONMENTAL NOISE MEASUREMENTS AND SIMULATIONS + INDUSTRIAL NOISE

ROOM ACOUSTICS

VIBRATION IN INFRASTRUCTURE

HEALTH EFFECTS OF NOISE

Acoustic regulations and quality classes
Experience from >35 years of collaboration in the Nordic countries.

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In the sixties the Nordic countries started their collaboration for common requirements on sound insulation. And in the seventies the national Nordic building regulations regarding acoustics became almost identical. In the nineties the first Nordic national standards for sound classification of dwellings were drafted and a common Nordic classification standard was proposed, but not accepted. Now the Nordic countries have similar, but not identical classification standards and similar but not identical requirements regarding acoustics in their building regulations.

1 Introduction

In 2014 Jens Holger Rindel presented the paper The history of the Nordic Acoustic Association in the NAA 60 Years Anniversary book [1]. Information here about the early years of Nordic collaboration in acoustics stems partly from this paper. Already in the year 1960 the first proposal for common Nordic requirements on sound insulation was presented by NKB (The Nordic Committee on Building Regulations).

In 1973 a NKB acoustic working group was established, and in 1974 the group delivered recommendations that were adopted in the national Nordic building regulations, and in 1978 NKB presented the report: “Guidelines for building regulations concerning sound precautions" [2]. These guidelines were e.g. used when a new Icelandic building code was published in 1979, with acoustic demands included in the building code for the first time. At that point in time, all the Nordic building codes were almost identical regarding acoustics.

2 Development in the 80’s and in the 90’s

In the late 80’s and the early 90´s there was increasing awareness of the problem of insufficient insulation between dwellings. The acoustic demands in the Nordic countries had changed slightly in each country from the common values of 1978. It had also been pointed out that some European countries had somewhat stricter regulations than the Nordic countries. As a result, an acoustic working group was established again within NKB in 1993 with the task to suggest new harmonised sound insulation demands for residential buildings.

The NKB working group concluded that all residential buildings should have the same sound insulation demands, which would mean somewhat stricter demands than before for dwellings in apartments buildings. The first task of the group was to make a consequence analysis in each of the five Nordic countries to estimate how much the current building tradition would have to change to fulfil these proposed new demands, and how much it would cost.

The results were published in [3], “Lydbestemmelser i de nordiske lande. *NKB Report 1994:01,*1994”, and the main conclusion was the following: It is the conclusion of the working group that the recommended demands will stimulate the building industry to develop new and better building systems which typically will fulfil the slightly stricter demands without any extra costs.

It was also concluded to present a draft proposal for sound classification of dwellings with three quality classes, A, B and C, with C the minimum class in building regulations, and classes B and A noticeably better. The demands of the C-class would correspond to the proposed new and slightly stricter demands of new building regulations. It was also decided to propose that the dB-steps between the different quality classes A, B and C would be 5 dB.

In 1996 this draft proposal was sent to a new working group in INSTA (inter-Nordic standards common to all five Nordic countries). However, the proposed INSTA standard for sound classification was not accepted as a Nordic standard. Sweden had already introduced a Swedish standard for sound classification, and they were not prepared to change their new standard. So instead, the INSTA proposal has been used as a model for the national sound classification standards in the other Nordic countries.

3 Later development

Soon after 2000 the Nordic countries published different national classification standards. They have since developed and although they are similar in many ways, they are not identical. The latest updates of these standards are the following: see references [4], [5], [6], [7], [8] and [9].

[4] ÍST 45:2016 Sound Classification of various types of buildings (in Icelandic). The standard is not only for dwellings, but many different types of buildings. Sound classes A, B, C and (D)

[5] NS 8175:2019 Lydforhold i bygninger. Lydklasser for ulike bygningstyper (in Norwegian) The standard is not only for dwellings, but many different types of buildings. Sound classes A, B, C and (D)

[6] DS 490:2019 Lydklassifikation af boliger (in Danish). The standard is in principle intended for dwellings, but it is also used for hotels, student accommodations, nursing homes, residential institutions and similar. Sound classes A, B, C, D, E and F

[7] SS 25267:2024 Acoustics - Sound classification of spaces in buildings – Dwellings (in Swedish). Sound classes A, B, C and D

[8] SS 25268:2023 Building acoustics – Sound requirements for spaces in buildings – Healthcare premises, rooms for education, preschools and leisure-time centres, rooms for office work, hotels and restaurants (in Swedish) Sound classes A, B, C and D

[9] SFS 5907:2022, Acoustical Design and Quality Classes of Buildings. (In Finnish). The standard is not only for dwellings, but also for hotels and lodgings, facilities for the elderly, office buildings, schools, educational establishments, day-care centres, health care facilities and industrial workplaces. Sound classes A, B, C and D

4 European collaboration

In 2009 Birgit Rasmussen from Denmark together with María Machimbarrena from Spain managed to get funds for a project within COST – European Cooperation in Science and Technology. The Project was COST TU0901 "Integrating and Harmonizing Sound Insulation Aspects in Sustainable Housing Constructions". The project lasted for four years with close cooperation and discussion between experts from 29 European countries and 3 overseas countries A summary of the work can be found in [10] COST Action TU0901: Towards a common framework in building acoustics throughout Europe, published in 2013.

All the Nordic countries participated in this project, and one of the results was to prepare a proposal for a "European" sound classification scheme for dwellings. The proposal has six classes: A, B. C, D, E and F with generally 4 dB steps between the classes. The parameters or descriptors were somewhat different from those that have traditionally been used in the Nordic countries:

$D_{nT,50} = D_{nT,w} + C_{50\text{-}3150}$	Airborne sound insulation between rooms
$L'_{nT,50} = L'_{nT,w} + C_{I,50\text{-}2500}$	Impact sound pressure level
$D_{2m,nT,50} = D_{2m,nT} + C_{tr,50\text{-}3150}$	Airborne sound insulation of facades
resp. $D_{2m,nT,50} = D_{2m,nT} + C_{50\text{-}3150}$	depending on the type of outdoor noise and as defined in EN ISO 717-1
L_{eq}, $L_{max,F}$	Service equipment sound pressure level
resp. $L_{eq,nT,A}$, $L_{max,F,nT,A}$	as defined in EN ISO 16032 and EN ISO 10052
T	Reverberation time

This European collaboration has continued after the COST Action TU0901, and now a draft European/international EN ISO standard has been prepared, which to a certain degree is built on the work in the COST project: EN ISO 19488 Acoustics — Acoustic classification of dwellings. The standard proposal is called: ISO/DIS 19488(en) Acoustics — Acoustic classification of dwellings and this document is now under preparation for its final publication.

References

[1] Jens Holger Rindel, The history of the Nordic Acoustic Association, *NAA 60 Years Anniversary Book* 2014

[2] Nordic Committee on Building Regulations, Retningslinjer for bygningsbestemmelser vedrørende lydforhold. *NKB Report No 32,*1978.

[3] Nordic Committee on Building Regulations, Lydbestemmelser i de nordiske lande. *NKB Report 1994:01,*1994.

[4] ÍST 45:2016 Sound Classification of various types of buildings (in Icelandic)

[5] NS 8175:2019 Lydforhold i bygninger. Lydklasser for ulike bygningstyper (in Norwegian)

[6] DS 490:2019 Lydklassifikation af boliger (in Danish)

[7] SS 25267:2024 Acoustics - Sound classification of spaces in buildings – Dwellings (in Swedish)

[8] SS 25268:2023 Building acoustics – Sound requirements for spaces in buildings – Healthcare premises, rooms for education, preschools and leisure-time centres, rooms for office work, hotels and restaurants (in Swedish)

[9] SFS 5907:2022, Acoustical Design and Quality Classes of Buildings. (In Finnish)

[10] COST Action TU0901Towards a common framework in building acoustics throughout Europe, 2013

[11] ISO/DIS 19488(en) Acoustics — Acoustic classification of dwellings.

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Room acoustic measurement methods in the past, present and future, including the importance of the ISO 3382 series

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The measurement of reverberation time in a room was introduced by W.C. Sabine a few years before 1900 with the purpose to handle acoustic problems in a lecture theatre. The sound source was organ pipes at the seven octave frequencies from 63 Hz to 4000 Hz. Later, the measurements were greatly improved by using interrupted white noise emitted by a loudspeaker and a microphone connected to a level recorder that could display the decay curve. This led to the first international standard on reverberation time measurements in 1963, intended for laboratory measurements of the sound absorption coefficient of materials. In 1975 appeared another international standard intended for reverberation time measurements, primarily in rooms for speech and music. A revision in 1997 introduced the impulse response as a basis for room acoustic measurements. This opened up for derivation of other room acoustic parameters than the reverberation time, and this again led to a better understanding of how to design good rooms for music and speech. Some types of rooms are not sufficiently characterised by the reverberation time, and for that reason a new standard appeared in 2012 with a measurement method specifically intended for open-plan offices. Future development of room acoustic measurements may include faster and more reliable methods, better methods to overcome problems at low frequencies, a method to handle the influence the high-frequency problem of varying temperature and humidity of the air, and new ways to derive three-dimensional information on the sound reflections in a room.

1 Introduction

This paper presents a brief overview of room acoustic measurements and how the methods have developed during the last 125 years. The emphasis is on the major milestones that mark significant improvements. During the last 60 years, international standards on measurement methods have played an important role for promoting new and improved methods. The list of references is made with the intention to point at the origin of the measurement methods and the room acoustic parameters.

2 The early days of room acoustic measurements

Wallace C. Sabine (1868-1919) is known as founder of room acoustic measurements around 1900. He introduced the concept of reverberation time, defined as the time for the sound intensity in a room to decay to 1/1.000.000 of the initial intensity after a sound source is turned off [1]. The measurements were performed with specially constructed organ pipes as sound sources and a chronograph (stopwatch) to measure the audible decay time. The organ pipes were of the type *gemshorn*, which have a strong fundamental tone and relatively weak overtones. Sabine could calculate the reverberation time related to 60 dB decay by a very clever method. He used four identical sets of organ pipes, and could then measure the audible decay time using one, two, three or all four organ pipes as the sound source. This gave different results because the initial energy was different. For example, the initial sound level is 6 dB higher with four organ pipes compared to one organ pipe; thus, the reverberation time is ten times the difference in audible decay time for the two measurements [1]. The frequencies of the organ pipes were at seven octaves from 64 Hz to 4096 Hz, corresponding to the music tones C. Later, more organ pipes with the tones E and G were included, so measurements could be made in approximately one-third octave intervals. This was meant for laboratory measurements of sound absorption of materials [2].

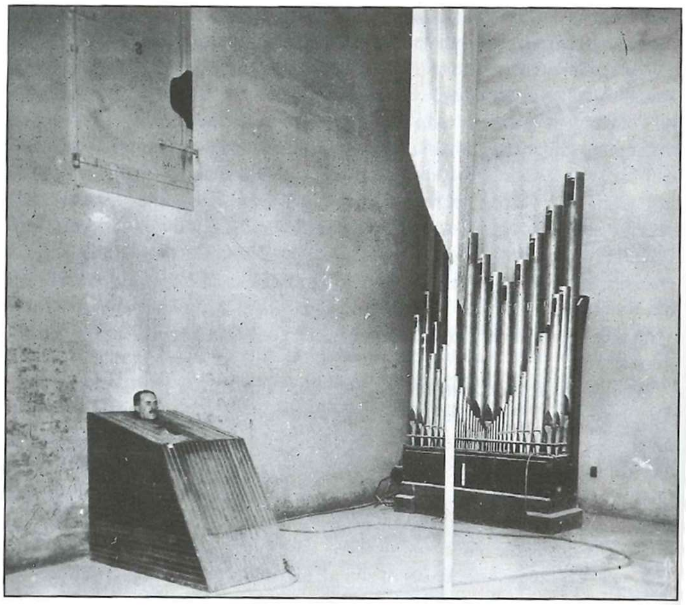

Figure 1: Conducting an absorption test in the reverberation chamber, 1919. From Kopec [2] p. 64.

In Figure 1 is seen the setup for an absorption test in the large reverberation chamber at the Riverbank laboratories. The operator is sitting in a box in order to minimize the absorption from clothing. The rotating fan seen in the photo was used to increase the diffusion. Paul E. Sabine continued the work with acoustic measurements after the death of Wallace C. Sabine. He explains: "*During the 1930s the stop watch and ear method was replaced by relay-controlled chronometers (acoustic clocks) operating in conjunction with frequency oscillators, amplifiers, loudspeakers, attenuators, and microphones*" [3]. The methods using a loudspeaker emitting periodic varying tones, a microphone and oscillograph are described in the book by Vern O. Knudsen [4, Chapter VII]. An example of a warble-tone signal and measured decay curves are seen in Figure 2.

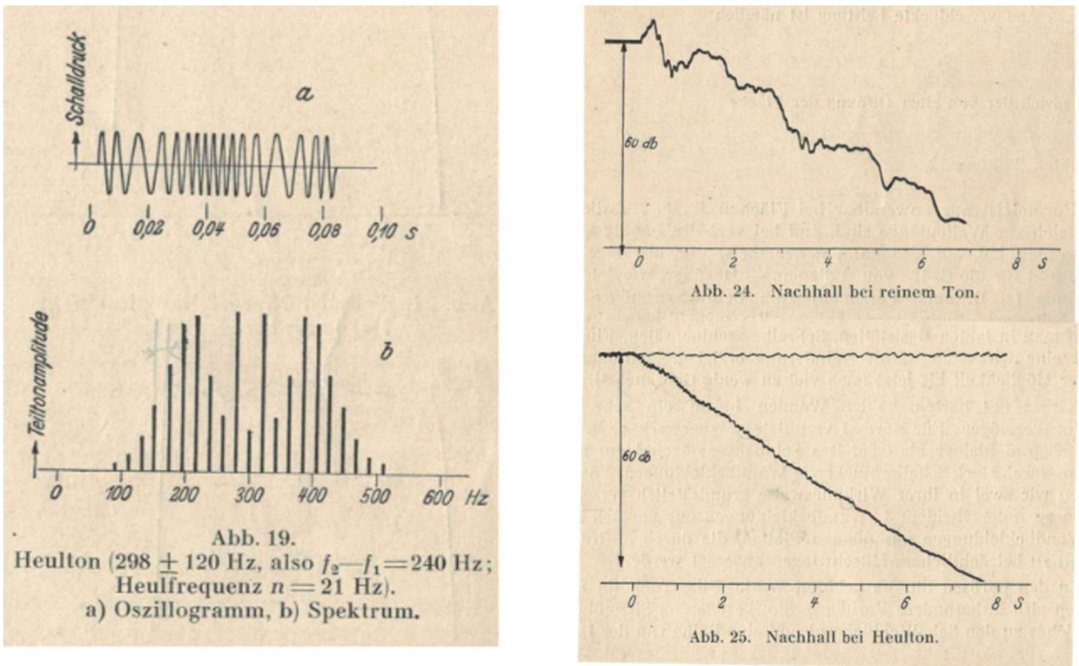

Figure 2: Measurement using a warble-tone. Left: The time- and frequency function of the signal. Right: Decay curves with a single frequency (upper part) or with the warble tone (lower part). From Schoch [5].

Field measurements of reverberation time were also made with (portable) organ pipes and stopwatch until around 1930. However, in 1934 measurements were made in the old Philharmonic Hall of Berlin (destroyed during the second world war) using for the first time a symphony orchestra as the sound source [6]. The music from the very first bars of the Beethoven *Coriolanus* overture was used as the sound signal. This particular music starts with some tutti cords in *fortissimo*, followed by pauses long enough to evaluate the reverberation time, see Figure 3. This implies, that the measurements could be made with the audience present. Reverberation times with and without an audience were reported in (approximate) octave bands from 125 Hz to 2 kHz. The 4 kHz band was measured, but no results were reported due to insufficient signal-to-noise level. Measurements with pistol shots (a start revolver) were also taken into use [6].

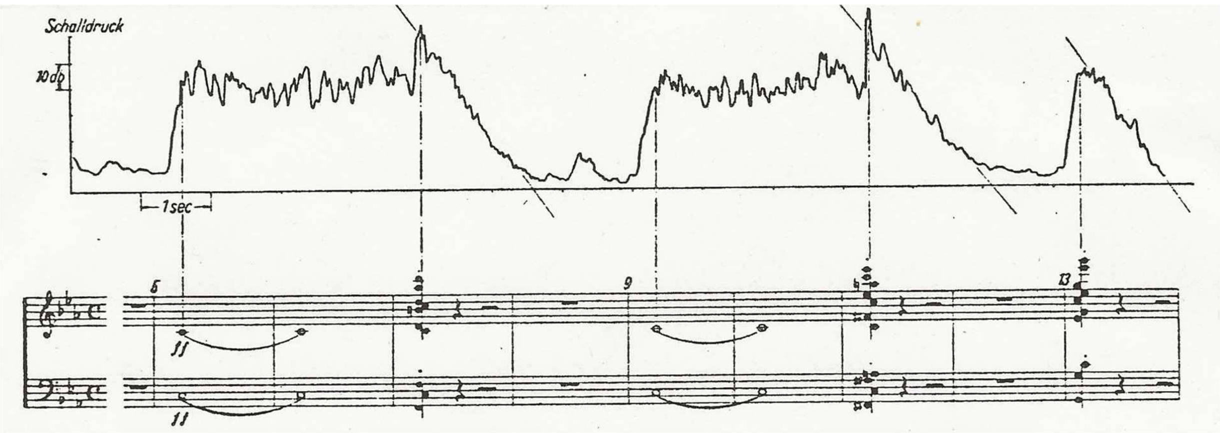

Figure 3: Decay curves in 500 Hz band measured 1934 in the old *Philharmonie*, Berlin, using a symphony orchestra as sound source. The music was Beethoven's *Coriolanus* overture, and the hall was with full audience. From Meyer & Jordan [6].

The registration of the decay curves was very difficult in the 1930s. For the concert hall measurements was used a technique, where the signal from the microphone was sent through a filter and a logarithmic amplifier to a movable mirror. A light beam was directed via the mirror towards a slowly rotating cylinder, which was covered with a phosphorescing layer. The curve that was made by the light beam was visible for long enough time to allow redrawing on paper.

A milestone in room acoustic measurements is the level recorder with a high-speed, logarithmic potentiometer. The first level recorder was made by Neumann around 1940, and from 1943 an improved model by Brüel & Kjær became widespread as an unavoidable part of the acoustic measurement equipment [7].

3 First measurement standards

ISO/R 354 recommendation was the first international standard for measuring the absorption coefficient of materials in a reverberation chamber [8]. It was published in 1963, five years after an American ASTM standard with similar contents [9]. A loudspeaker was used to emit the sound, either a *warble tone* or *white noise*. The sound was interrupter to get the decay. A microphone was connected to *cathode ray tube* or a *level recorder*. From this could be derived the decay rate (dB/s) or the reverberation time. The reading was taken from the part of the decay curve between -5 dB and -35 dB (or less) from the start of the decay curve. Measurements were made in octave bands or one-third octave bands from 100 Hz to 4000 Hz.

4 Measurement of reverberation time in auditoria

Since the first measurement standard ISO/R 354 was made for laboratory testing of materials, it soon became clear that there was a need for another standard directed towards field measurements, especially for auditoria. Such a standard was published in 1975 as ISO 3382 [10]. It described measurements made with loudspeakers and interrupted noise, and it also opened for other methods using organ pipes, blank pistol shots, or an orchestra as the sound source. The description of source and receiver positions were adapted to typical auditoria or concert halls. The derivation of reverberation time from the decay curve was the same as in ISO/R 354. The frequency range was one-third octave bands from 125 Hz to 4000 Hz. If the decay curve was not close to a straight line, two reverberation times were to be reported for the early and the late part of the decay, respectively.

There are several remarkable matters in this standard. One is the use of one-third octave bands. While this made sense for the measurement of absorption coefficients, it is more problematic for measurements of reverberation time in auditoria. Among acousticians working with concert halls, it was common practice to measure in octave bands and often to look at the average of two or three octave bands. The wider bandwidth was in order to obtain results that were not too sensible to variations in measurement positions.

Another remarkable matter in the standard is the possibility to use a pistol shot. It had become common to measure with pistol shots and derive the reverberation time directly from the squared impulse response, see Figure 4. However, already in 1965, Schroeder had shown how to derive a correct decay curve by backwards integration of the squared impulse response [12]. Without this integration, the results could be wrong, especially if the decay curve deviated much from an exponential decay. The example in Figure 5 shows the squared impulse response and the integrated impulse response measured in a concert hall.

The measurement method with the integrated squared impulse response is a milestone in room acoustic measurements. While the interrupted noise signal is stochastic and has to be repeated several times, the integrated impulse response remains the same if repeated. Schroeder had shown that the decay curve of the latter is the same as the ensemble average of an infinite number of interrupted noise signals [12]. While the interrupted noise method has remained a common method for laboratory measurements, the integrated impulse response has become the preferred method for field measurements because it is time-consuming and accurate.

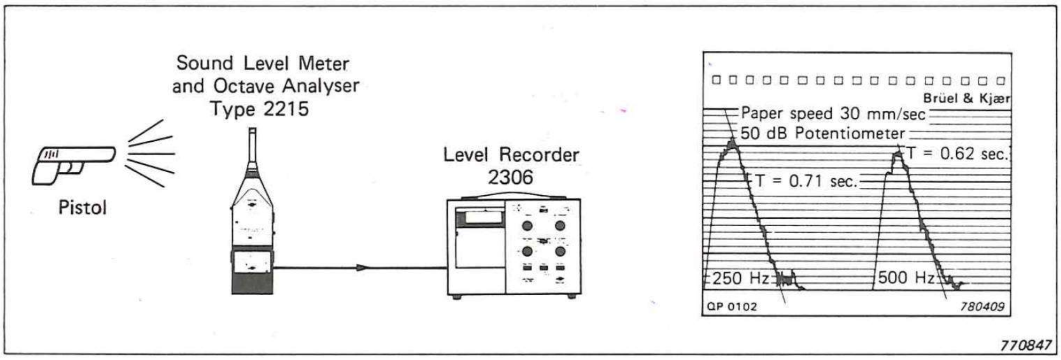

Figure 4: Set-up for measurement of reverberation time with the pistol shot method. From Fig. 6.1 in Ginn [11].

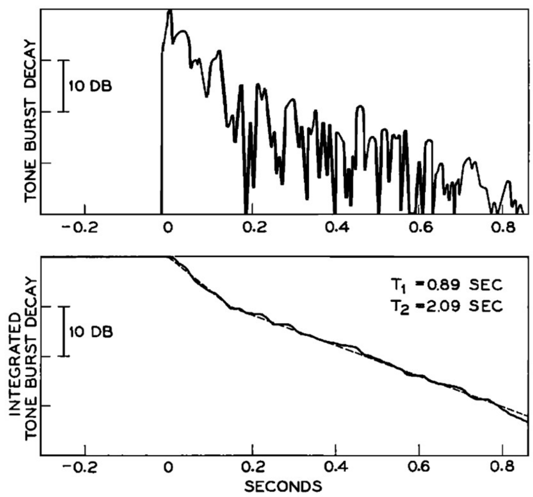

Figure 5: Measured squared impulse response (tone burst decay) and corresponding integrated impulse response (decay curve). A one-third octave band filter centred at 167 Hz was applied. Two different reverberation times have been derived, T_1 from the upper 10 dB and T_2 from the lower part of the curve. From Schroeder [12].

5 Improved measurement methods

5.1 Integrated impulse response method

The second edition of ISO 3382 was published in 1997 with several improvements [13]. The introduction states: "*The intention is to make it possible to compare reverberation time measurements with higher certainty, and to promote the use of and consensus in measurement of the newer measures.*" The scope includes this sentence: "*It is not restricted to auditoria or concert halls; it is also applicable to rooms intended for speech and music or where noise protection is a consideration.*"

The standard makes a clear distinction between two alternative methods for measuring the reverberation time: The interrupted noise method or the integrated impulse response method. The decay directly after excitation with a pistol shot or other impulse sources should only be used for survey purposes and is not recommend for accurate evaluation of the reverberation time. Impulse responses can be generated with loudspeakers using signals with Maximum Length Sequence (MLS), chirps or linear sweeps.

The measurement of impulse response opens up for the derivation of several other room acoustic parameters, and such parameters are included in informative annexes of the standard. Most of these new parameters had been developed since the 1950s, especially in relation to the use of scale models for acoustic design of auditoria, see Jordan chapters 4, 9, and 11 in [14]. The room acoustic parameters in this edition of the standard are listed in Table 1. The *lateral energy fraction* (LF) requires a *figure-of-eight* microphone, and the *inter-aural cross correlation* (IACC) requires measurement with a *dummy head*.

Table 1: Auditorium measures derived from impulse responses, ISO 3382 Annex A and B [13].

Name	Symbol	Unit	Description	Origin
Sound strength	G	dB	Total sound level relative to 10 m in free field	Lehmann [15]
Early decay time	EDT	s	Reverberation time derived from the first 10 dB	Jordan [16]
Early-to-late index	C_{50}, C_{80}	dB	Ratio of early to late sound energy in dB (*Clarity*)	Reichard [17]
Definition	D_{50}	-	Ratio of early to total sound energy (*Deutlichkeit*)	Thiele [18]
Centre time	T_S	ms	Time of gravity of the squared impulse response (*Schwerpunktzeit*)	Kürer [19]
Lateral energy fraction	LF	-	Ratio of early lateral energy to early total energy	Barron [20]
Inter-aural cross correlation coefficient	IACC	-	Maximum of normalised inter-aural cross correlation function	Damaske [21]

5.2 Linear regression replaces manual estimate of decay rate

The standard of 1997 states that the measurements of reverberation time in concert halls should be made in octave bands from 63 Hz to 4 kHz, while one-third octave bands from 100 Hz to 5 kHz can be used in other kinds of rooms. All reverberation parameters (EDT, T_{20}, and T_{30}) shall be determined from the slope of linear regression line within the evaluation range. For reverberation time this is a clear improvement from earlier manual methods, but for early decay time (EDT) this is more problematic as shown by Bradley [22]. For nearly 30 years, the EDT had been derived from the two points at 0 dB and -10 dB on the decay curve, neglecting the irregularities that are often seen in the first part of the decay curve, see Jordan [14, page 70]. While the difference between the two evaluation methods for EDT is negligible in long distances from the source, the difference can be very significant in positions close to the source. EDT is known to be a good measure of *perceived reverberance*, but the research behind this was based on the two-point EDT (before 1997).

Due to the fact that the air attenuation can influence the measurement results at frequencies above 500 Hz, the standard requires temperature and relative humidity in the room during the measurements to be included in the test report. These should be measured to an accuracy of ± 1 °C and ± 5 %, respectively.

5.3 Speech Transmission Index

In Annex A of the 1997 edition of the standard is mentioned in a note, that the Speech Transmission Index (STI) can also be used to determine the *speech intelligibility*, but this is a special measuring technique not covered in the standard. A simplified version of the method using only signals at 500 Hz and 2000 Hz is the Rapid Speech Transmission Index (RASTI). Brüel & Kjær produced portable equipment for these measurements, consisting of a speech transmitter (Type 4225) and a speech receiver (Type 4419). The method is based on a series of complicated modulation transfer functions, but the measurement result is simple and easy to understand. For many years, the RASTI measurements became quite popular as a supplement to other room acoustic measurements. However, in the 2011 edition of the measurements standard [23], the RASTI method was declared obsolete and was not be used. Also, the STI method has problems in relation to room acoustic measurements, mainly because the method is not sensible to echoes [24].

5.4 Sine sweep measurements

An early example of measuring with sine sweep signals was the series of concert hall measurements from 1984 by Gade & Rindel [25]. A linear sine sweep that covered one octave band was used and scaled in time and frequency for the six octave bands from 125 Hz to 4 kHz, see Figure 6. The response measured in the room was converted to an impulse response by *convolution* with the original sine sweep signal.

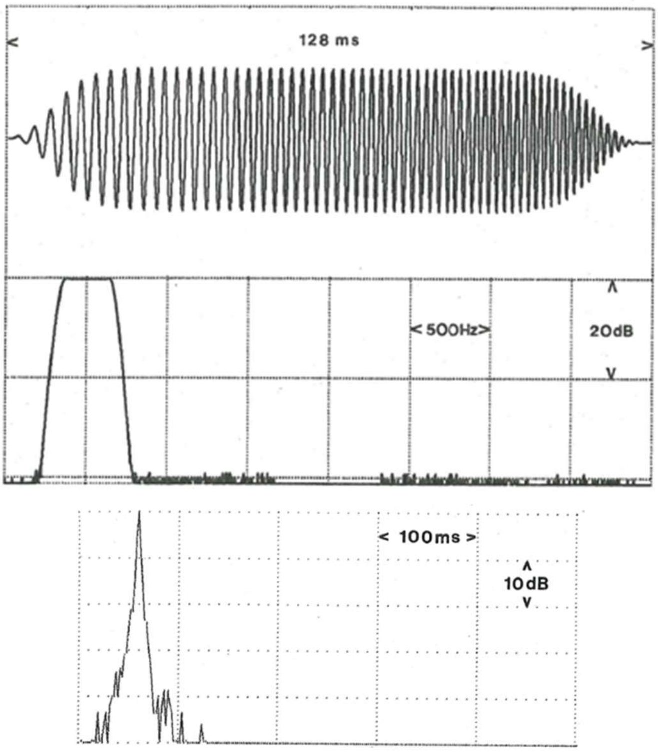

Figure 6: Sine sweep signal covering one octave, here shown for the 500 Hz octave band. Top: Signal in time domain. Middle: Signal in frequency domain. Bottom: Impulse response after convolution. After Gade & Rindel [25].

6 Recent developments in room acoustic measurements

6.1 Performance spaces in ISO 3382-1

For reasons to be explained in the next section 6.2, it was decided that the room acoustic measurement standard should be divided into two parts (and later three parts). Part 1 from 2009 [26] was in fact a 3rd edition of the previous ISO 3382 standard. This new edition had some minor changes, mainly in the annexes on various room acoustic parameters.

Annex A was extended with information on JND (just noticeable difference) [27] and the *late lateral energy level*, a new parameter related to the *perceived listener envelopment*. It is measured as the lateral energy level after 80 ms using a calibrated *figure-of-eight* microphone and is based on research by Bradley & Soulodre [28].

A new Annex C was added with the parameters *Early Support* (ST_{Early}) and *Late Support* (ST_{Late}) for the acoustic conditions at the orchestra platform. The former is related to the musicians' ability to *hear each other*, while the latter is related to the musicians' *perceived reverberance*. Both are measured in the distance of 1.0 m from the acoustic centre of an omnidirectional sound source. These parameters are based on research by Gade [29].

6.2 Ordinary rooms in ISO 3382-2

In 2001 the technical committee on acoustics ISO/TC 43 received a New Work Item Proposal (ISO/TC 43/SC 2 N 638) that recommend to divide ISO 3382 into two parts. Part 1 should be for performance spaces as the existing standard, while Part 2 should deal with "*the measurement of reverberation time in rooms in general. Examples are living rooms, classrooms, workshops, stairwells, and industrial halls*". ISO 3382-2 [30] is supposed to be a reference standard for building acoustic measurements and other standards where reverberation time is a part of the measurements. The part 2 of the standard deals with reverberation time, only. Both the interrupted noise method and the integrated squared impulse response method are described. The evaluation range for derivation of the reverberation time can be 20 dB or 30 dB, with a preference for 20 dB for various reasons, as explained in the introduction to the standard.

Concerning the number of source and receiver positions and other technical details, the part 2 of the standard distinguishes between three levels of measurement accuracy: *survey*, *engineering*, and *precision*. The precision method is meant for testing laboratories, especially for the measurement of the absorption coefficient of materials. Two annexes are included, one for the measurement uncertainty, and the other one with definition of "*measures that quantify the degree of non-linearity and the degree of curvature of the decay curve. These measures may be used to give warnings when the decay curve is not linear, and consequently the result should be marked as less reliable and not having a unique reverberation*" [30]. The origin of these quality measures is the work of Bodlund in 1984 for a Nordtest method [31]. He elaborated on the possibilities of using a computer for pression measurements in a laboratory with the interrupted noise method. He also introduced *ensemble averaging* as an attractive alternative to arithmetic averaging of reverberation times.

6.3 Open plan offices in ISO 3382-3

A second New Work Item Proposal (ISO/TC 43/SC 2 N 889) was proposed in 2007 that recommend establishing a new part 3 to the ISO 3382 series. The reason was acoustical problems in open plan spaces (offices, schools), and the recognition that the reverberation time was not sufficient or useful for characterizing open plan spaces. The proposal stated that "*there is reasonable agreement that other types of measurements such as rate of spatial decay of sound pressure levels, speech transmission index and background noise levels are needed for a more complete evaluation of the performance of open plan spaces*".

The measurement of *spatial decay* in large spaces had already been taken into use in industrial halls, see ISO 14257 [32]. The possibilities of using the speech transmission index (STI) in relation to open plan offices had been suggested by several researchers [33, 34]. Finally, the ISO 3382-3 [35] was published with the title limited to open plan offices.

Several new measures were defined, four of them being mandatory:

- distraction distance r_D
- spatial decay rate of A-weighted sound pressure level of speech, $D_{2,S}$
- A-weighted sound pressure level of speech at 4 m, $L_{p,A,S,4\,m}$
- average A-weighted background noise level, $L_{p,A,B}$.

The distraction distance r_D is derived from STI measurements performed on a line of receivers with increasing distances from an omni-directional source. The distance where STI is estimated to decrease below 0,5 is defined as the *distraction distance*. The sound source must emit a noise signal at specified sound power and spectrum that represents speech sound. This new part of the standard did not solve the acoustical problems in open plan offices. However, it did establish a very important common reference for research in the field of acoustics of open plan offices. The acoustical problems in open plan offices are much more complicated than ordinary noise control. It is more like an optimisation problem, and how use the new parameters is still under discussion [36].

A second edition of ISO 3382-3 was published in 2022 [37] with minor changes. Most important is the addition of another parameter, the comfort distance (r_C). This is defined as the distance from the omni-directional source, where the A-weighted sound pressure level of speech decreases below 45 dB.

7 Future measurement methods

7.1 Revision of ISO 3382-1

ISO 3382-1 is currently being revised and some of the ideas for future changes may be unveiled. Most importantly, it will be emphasised that the standard is not only for music rooms, but also includes rooms for speech and open-air performance spaces. It will be made clear which of the room acoustic measures are relevant in rooms for speech, especially classrooms. Some new room acoustic parameters may be suggested, such as an *echo-parameter* and the *acoustic efficiency*, both of them most relevant for open-air performance spaces [38].

A new method for *normalisation* of the measurements to a standard atmosphere (20 °C, 50 % RH) is also being discussed.

7.2 Measurement of modal reverberation times

The problem of measuring reverberation time at low frequencies (< 100 Hz) in small rooms (< 200 m^3) is still a challenge. One idea is to measure the reverberation time of single modes. Instead of a conventional statistical approach, the source and microphone positions can be chosen strategically in order to separate one low-frequency mode at a time [39]. This is best applied to rectangular rooms, where the modal pattern of the modes is well known.

7.3 Application of cepstrum analysis

Flutter echoes and the spectral *colouration* due to periodic sound reflections are easy to hear but difficult to measure objectively. A method that has proven useful, especially for colouration issues, is the application of the so-called *cepstrum* analysis [40]. This is the inverse Fourier transform of the logarithm of the spectrum. The spectrum is derived as the Fourier transform of the impulse response. Peaks in the *cepstrum* indicate a periodicity in the spectrum, which is typical for the colouration.

7.4 Measurements with a 3D field microphone

The 3D distribution of the sound reflections in a receiver position can measured by replacing the omnidirectional microphone with a number of cardioid capsules, three orthogonal intensity probes, or some other microphone array. The measured signal can be transformed into *first order ambisonics signals* (B-format) or higher order ambisonics signals with addition of more microphones in the array. This technique can be used to measure the *spatial room impulse response* (SRIR) and to visualise the directions of reflections in the measured impulse response as a *hedgehog pattern* [41-42].

However, the use of 3D field microphones opens up for several other applications. The measured signal can be transferred into a figure-of-eight signal and a simultaneous omni-directional signal for deriving the lateral energy parameters in ISO 3382-1. By application of a *head-related transfer function* (HRTF) the measured signal can also be transferred into a *binaural room impulse response* (BRIR), thus replacing a *dummy head* for the measurement of the IACC parameter.

7.5 Application of sound intensity and particle velocity

Further application of the above-mentioned technique is to measure two different impulse responses, the traditional one based on sound pressure and another one based on the 3D sound intensity. Thus, a *dynamic diffusion curve* (DDC) can be derived from the difference between the two decay curves [43]. The DDC is a measure of the amount of non-directional energy as a function of the decay level. This technique is very efficient for analysing the degree of *diffusivity* and for detecting echoes and flutter echoes in a room. It has also been suggested for evaluation of the acoustical quality of reverberation rooms [44].

The simultaneous measurement of sound pressure and *particle velocity* opens up for improvements of measurements in reverberation rooms, that are used for measuring the absorption coefficients of materials. With current technique, it is a significant problem that the *diffusivity* in the room is compromised when the test material is installed. With the possibility to measure both the *potential* energy density and the *kinetic* energy density, the *total energy density* can be achieved. This may offer more accurate and robust measurement results, because the total energy density varies very little throughout the room. This applies to the *steady-state* sound as well as to the *decaying* sound field. This is the basis for a detailed measurement method that has been described in a US patent by Hanyu [45].

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Acoustic properties to be included with new housing sale reports?

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Sales reports for dwellings normally lack information about the acoustic properties of the house for sale. A new law about safer house trading was brought into act in 2022. This law led to a revision of a Norwegian standard: NS 3600 "Technical condition analysis when selling dwellings" (AI-translated name). A possibility for including acoustic specifications was then opened. Such specifications, however, need to be easy obtainable and easy understandable to be accepted. A proposal that now is out for comments, is based on the following: Internal sound insulation will be described in relative terms compared to new houses, based on the age of the house. External noise level shall be reported from available noise contour maps for traffic noise, railway noise, tram noise, airport noise, industry noise etc. It must be noted that the standard is out for hearing at the moment. The final content of NS 3600 is therefore unknown.

1 Introduction

The purpose of informing buyers about sound conditions in homes during transactions is to ensure that their expectations align with the actual conditions both inside and outside the property. This can help to reduce conflicts after a sale and enable buyers with varying levels of sensitivity to noise, to find a property that suits their needs.

2 Acoustic properties of older buildings

2.1 Minimum requirement in regulations

Following the acoustic quality of buildings over the last decades, reveals that houses and dwellings have been built to fulfil legal requirement in regulations, and no more. Therefore, information about the regulations at various times tells a lot of what can be expected of airborne- and impact sound isolation in the house. The picture below shows regulations and specifications for acoustic properties in dwelling among others, at various times in Norway.

The regulatory papers include:
10 versions of building regulations and
5 version of our sound standard NS8175
(These are from private libraries. Not all are available on the web).
Some major requirements can be found by searching "byggeforskrifter" at: http://norskakustiskselskap.org (in Norwegian only)

2.2 Outdoor situation

During the later years, official requirements in Norway as well as from EU has lead to an increasing presentation of noise contour maps. Several noise maps can be found related to the spesific sources. An overview for Norwegian maps is found here: https://kartkatalog.geonorge.no/

3 General extend of acoustic information

3.1 Basic challenges

The housing standard has increased as long as houses have been built. Many homes are also exposed to increasing noise from outdoor sources such as road traffic, railways, tramways, and airplanes. Also industries, warehouses, ports and terminals, shooting ranges, motorsports tracks, etc. can create noise above existing limits.

The content of an eventual seperate acoustic report may easily grow to include measurement of airborne sound insulation, impact noise levels, noise from technical installations etc. The cost of such reports could then likely exceed the cost of the full technical condition report that are presented today. It is therefore a challenge to find an acceptable amount and precision of acoustic properties to be presented and an effective way of reporting them.

Information about acoustic properties for older buildings may be regarded as negative, especially for those that only have got a “surface-shine” before selling. This may of course also be regarded as negative for the sellers and the real estate agents. For the buyer it seems reasonable to have information of all important issues about the house. This may be a problem at the moment for those who bought without knowing anything about sound insulation or outdoor noise and selling with the information. This is similar to other new information and related problems will be overcome over time.

3.2 Internal sound insulation

Ideally indoor sound conditions should include values for airborne sound insulation, impact noise from neighbors, noise from technical installations (e.g. ventilation) and noise from outdoor sources. Building methods and technical solutions are as indicated in 2.1, closely related to requirements in planning and building regulations with associated building codes. However, this applies to a general level mainly for sound insulation. Improvements made to the partitions inside and/or towards other dwellings should therefore also be reported as “advantage information”.

Some information will obviously be lacking in a limited report like the one suggested in the new NS 3600. However, this does not reduce the value of the information that is easily available.

3.3 Outdoor noise situation

Outdoor noise sources affect the noise levels around the residence and outdoor recreational areas. If the sound insulation in exterior walls is insufficient, indoor noise levels can also be disturbing, affecting relaxation, rest, and sleep.

There may be minor outdoor sources that are not presented in the official noise contour maps. This challenge can be partly compensated by a note about noise in the self-declaration. Some sources of noise may also be visually observed or heard during inspection. However, the lack of full coverage should not prevent information about available information.

3.4 Self-declaration

The value of self-declaration on noise is discussable, as there may be large differences in the subjective perception as well as the willingness to reveal potential problems.

There may also be a big difference in what sellers and buyers want to focus on are re disturbed by. Another issue is that during a viewing, there are normally several potential buyers present. People will have less sense for the sound conditions than what they would be able to if they were alone in the dwelling.

Sometimes viewing also is done at times when noise from outdoor sources is low. These factors may lead to a “false” impression of the situation. Therefore, a statement from the seller regarding outdoor noise conditions and/or perceived sound insulation against potential neighbors may still be valuable.

At the moment the proposal includes a question about noise in the sheet for self-declaration.

4 Sound-insulation related to age

4.1 General

A rough assessment of internal sound conditions can be made based on the year of construction of the house, supplemented with information about measures taken and/or actual constructions. Measures will have additional value if they are documented with sound measurements or with other form of report from qualified individuals.

The table in Norwegian below, is from the reference in section 2.1.

Regelverk	År / endr.	Luftlyd	Trinnlyd	Kommentarer
Lov om bygningsvesenet	1924	-	-	Konstruksjonskrav tilsvarer ca. 39 dB
Byggeforkrifter	1949	D ≥ 50 dB*)	ΔL ≥ 12 dB*)	"Middelreduksjonstall" / trinnlyddemping. Anvisninger for konstruksjonstyper.
Byggeforskrifter	1969 / 69	$D_{0,5}$ ≥ 49 dB	L_s ≤ tabell verdier	Tilleggskrav; R ≥ 50 / 52 dB og tabellverdier. 3 dB strengere krav til luftlyd i rekkehus
Byggeforskrifter	1969 / 72	=	=	=
Byggeforskrifter	1969 / 76	=	=	=
Byggeforskrifter	1969 / 79	I_a ≥ 52 dB	I_i ≤ 63 dB	I_a 3 dB strengere for rekkehus I_i 5 dB strengere for rekkehus
Byggeforskrifter	1969 / 80	=	=	=
Byggeforskrifter	1969 / 81	=	=	=
Byggeforskrifter	1969 / 83	=	=	=
Byggeforskrifter	1985	=	=	=
Byggeforskrifter	1987 / 88	R'_w ≥ 52 dB	$L'_{n,w}$ ≤ 58 dB	R'_w 3 dB strengere for rekkehus $L'_{n,w}$ 5 dB strengere for rekkehus
NS 8175	1997	R'_w ≥ 55 dB	$L'_{n,w}$ ≤ 53 dB	Samme grenseverdi i blokk og rekkehus. **Noe om eksternstøy**
NS 8175	2005	=	=	L'_n: Volumbegrensning på100m³ i boliger
NS 8175	2008	=	=	=
NS 8175	2012	=	=	=
NS 8175	2019	R'_w+C≥54dB	R'_w+C≤54dB	Ikke referert i forskrifter pr. 05.11.2021

Statements for each category are given in 4.2 – 4.5.

Note: It has been decided by the non-acousticians in the working group to avoid dB in the descriptions. (Translation from Norwegian to English may confuse the expressions a bit.)

4.2 Houses built after 1997

Houses built after 1997 are assumed to meet today's regulations. It means that the property has sound conditions according to the limits in NS 8175 (1997-2012) class C. This also applies if improvements that provide equivalent sound conditions are made in older houses. The sound insulation will then be experienced as satisfactory by most people.

4.3 Houses built between 1967 and 1997

There is a change in specifications in 1987. This is however so small that one may choose to omit it in this coarse type of classification.

The following may then be stated as general information for the period: "The sound insulation is noticeably worse compared to newer homes, even if the sound insulation is weaker than current limits".

In acoustic terms the difference is 3 – 5 dB.

4.4 Houses built between 1949 and 1967

The sound insulation may be 5 – 10 dB worse than in new houses. The following may then be stated as general information for these houses:

"In dwellings built in the period from 1949 to 1969, sound insulation is usually significantly worse than today's standard."

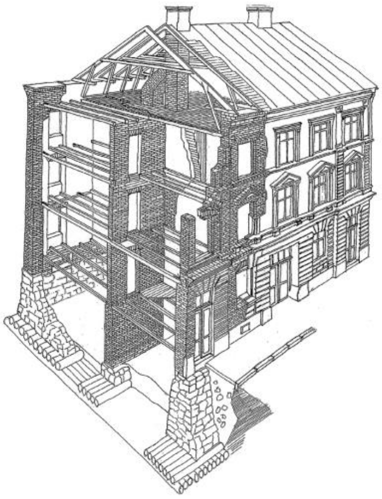
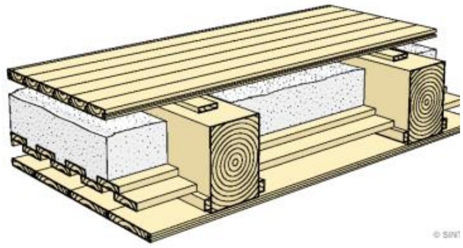
Examples of Norwegian houses built early in the period 1949 to 1967 are shows in the pictures below:



Some of these have concrete slabs and some have wooden slabs. Inform about slab type should be given, as the wooden slabs normally have lower performance that the concrete slabs.

4.5 Houses built before 1949

Old apartment buildings are probably similar over most of the Nordic and maybe also in the Baltic countries. They may be very nice looking when they have been remoulded and renewed, but there may be a big difference in sound insulation compared to new houses. Then picture below show the building system of houses from around 1900 and new facades.

Typical basic floor-slab in an old house

The following text applies to this situation: "In residential buildings constructed before 1949, the sound insulation is usually much worse than in homes with today's standard. If no measures have been taken, the sound insulation can be perceived as less than half as good as in new dwellings".

5 Noise contour maps

Noise contour maps will be used only as a tool for finding a number concerning the outdoor noise level, and only if these exceed the relevant limits for the source.

The method is well known by acousticians, thus further description here is not necessary.

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Official law, regulation, guide and standard:

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- Forskrift til avhendingslova (tryggere bolighandel) https://lovdata.no/dokument/SF/forskrift/2021-06-08-1850
- Guidance on use (DIBK) https://www.dibk.no/regelverk/forskrift-til-avhendingslova-tryggere-bolighandel
- NS 3600 Teknisk tilstandsanalyse ved omsetning av bolig (not available on web)

The slab figure and most pictures are from Byggforsk publications, others are private.

BNAM 2024
Baltic-Nordic Acoustics Meeting
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Sustainable Wooden buildings – opportunities for good indoor acoustics

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The number of wooden buildings is increasing in many countries mainly due to their contribution to a greener environment. The building development is also boosted in general by increased population, migration to cities and demography. As we construct more, both multifamily houses and essential facilities like schools and hospitals, embracing the wooden trend is therefore of great importance. However, building acoustics is crucial regarding final design in wooden structures, implying that detailed acoustic optimization to keep the cost limited can be a challenge. While numerous multi-storey wooden buildings stand tall, the understanding of acoustic solutions in floor and wall build-ups and junctions is still lagging compared to concrete counterparts. To enhance our knowledge, we must understand and refine criteria for sound insulation, tackle annoyance factors to a larger extent, and from that optimize structural details in various building systems to fit specific needs in each type of building. This keynote will explore different ongoing and completed projects and disclose their specific technical solutions providing sufficient sound insulation. It will shed light on the diverse opportunities offered by various wooden building systems in different types of multi storey buildings.

DYNAMIC PROPERTIES OF WIRE ROPE AND RUBBER ISOLATORS AT HIGH FREQUENCY RANGE IN ENGINE ISOLATION

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Abstract

Engines, equipment, and structures need to be protected from disturbing vibrations and shocks. The needed isolation may be achieved by installing vibration isolators between the base and the object. For the design of isolation systems finite element analysis is an effective tool. Unfortunately, accurate isolator models are not always available. Dynamic properties of vibration isolators are needed to create reliable mathematical models. This paper describes field measurements of wire rope (WRI) and rubber isolators in engine application. Measurement results can be used to create more accurate models of vibration isolators. Common knowledge in industry is that rubber isolators have better isolation capacity at high frequency range than WRIs. However, it is proven that WRIs have lower amplification at resonance than rubber isolators and when vibration excitation is relatively high their isolation capacity at high frequencies is adequate. The tests were done in VEBIC facilities in Vaasa Finland.

INTRODUCTION

The dynamic behavior of flexible elements has been studied in order to predict the vibration isolation capability of different isolators. Two studied flexible elements are rubber and wire rope isolators (WRI). Earlier the tests were conducted in the laboratory and for example engine excitation have been estimated from field measurements. The excitation has been created using electro-magnetic shaker or hydraulic test device. The results have shown that there are clear differences between rubber and WRI. [1-5]

The purpose of this study is to investigate the isolation capability of WRIs and rubber isolators at high frequency range using diesel engine as excitation source. The isolation capability is measured using accelerometers and strain gauges. The isolation capability at low frequencies is already known. The WRI has higher damping which leads to lower response vibration levels at resonances. The non-linearity of WRI is also well known and rubber isolator is known to be linear in relation to amplitude. The damping mechanisms are also different in these isolator types (viscous for rubber and Coulomb friction for WRI). [6-7]

METHODS

First the isolators were studied using a direct method (hydraulic test device). [8]. In direct method measurement force is measured from the force output side of the part and the displacement is measured from the force input side. Dynamic stiffness was calculated from time displacement-force figures using trend line of the graph. Loss factor was determined by the basis of dissipated energy related to maximum of potential energy. These results are already presented in other publications (see references 1-5). Based on the laboratory tests, flexible elements with closely matching stiffness were chosen. The rubber isolator is Trelleborg 17-0391 C2 and the wire rope isolator is WRI 8-8/58-75-148.

Experimental setup

The experiments were performed at the University of Vaasa's VEBIC (Vaasa Energy Business Innovation Centre) engine laboratory.

Engine

The investigated isolation elements were installed between AGCO Power 49 AWF engine and the engine base. The engine is a typical turbocharged and intercooled heavy-duty diesel engine with a common-rail injection system. It is intended primarily for off-road machinery applications. During the experiments, a Schenck W400 water-cooled eddy current dynamometer, controlled via Horiba SPARC controller, was used to load the engine. Main engine specifications are listed in

Table 1. Test engine specification

Type	AGCO Power 49 AWF
Nominal speed (r/min)	2100
Cylinder number	In-line 4
Displacement (liter)	4.9
Bore (mm)	108
Stroke (mm)	134
Air intake	turbocharged, intercooled
Fuel injection	common-rail

3D accelerometers were installed above and below the isolators and transmissibility was calculated between them. Amplification, which is the magnitude of transmissibility, is used for comparing the isolation capacity of isolators, because it represents how much excitation is transmitted through the isolator to the isolated target.

To measure forces directly, strain gauges were installed under the isolators. The added part was instrumented with strain gauges under the isolators. Both isolator types were measured in a similar way. The rubber isolator was the original isolator that was provided with the engine. The WRI was selected as presented above.

Two tests were performed, stepped rpm varied from low to maximum and sweep test from maximum to low rpm. The measurement results are shown from maximum rpm and sweep test.

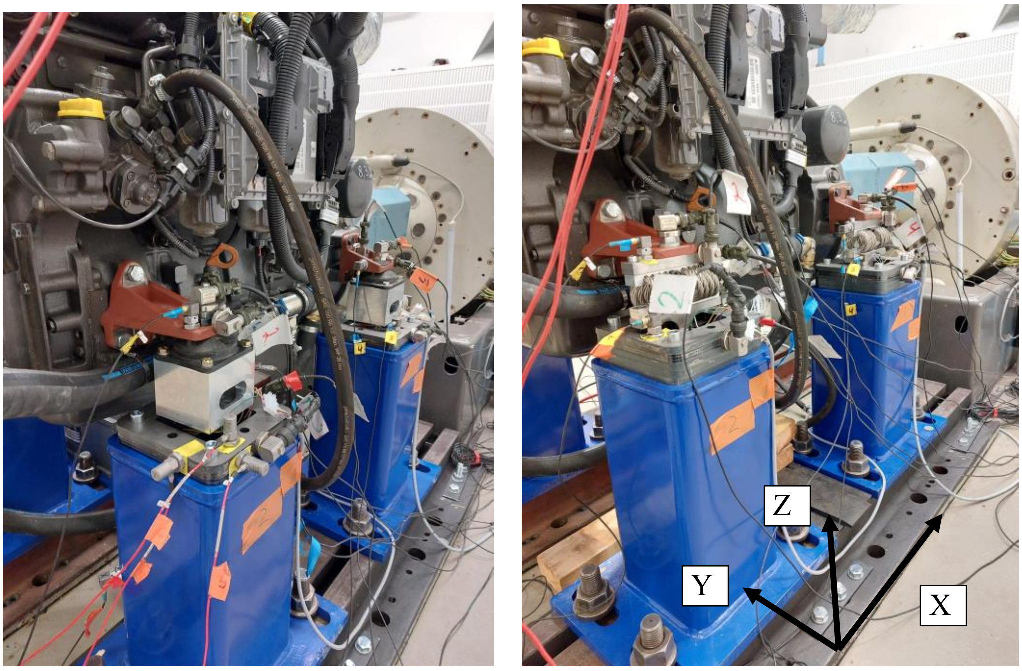

Figure 1. The studied isolators.
Left: rubber (Trelleborg 17-0391 C2) and right: WRI (8-8/58-75-148).
The strain gauge sensor is below the isolator (both types). The coordinate system is presented with black arrows.

RESULTS

In Figure 2 and Figure 3 the vibration peak-hold curves of the sweep tests are illustrated. At higher frequencies the accelerometer installation is not valid anymore. Therefore, results are presented only up to 2000 Hz. In the Figure 4 the comparison of rubber and WRI is presented from the sweep test. In Figure 5 the strain gauge measurement results are presented in vertical direction. The measurement test was steady at nominal 2100 rpm.

Figure 2. Vibration velocity peak-hold. Three directions – WRI. Sweep test from 2100 to 700 rpm.

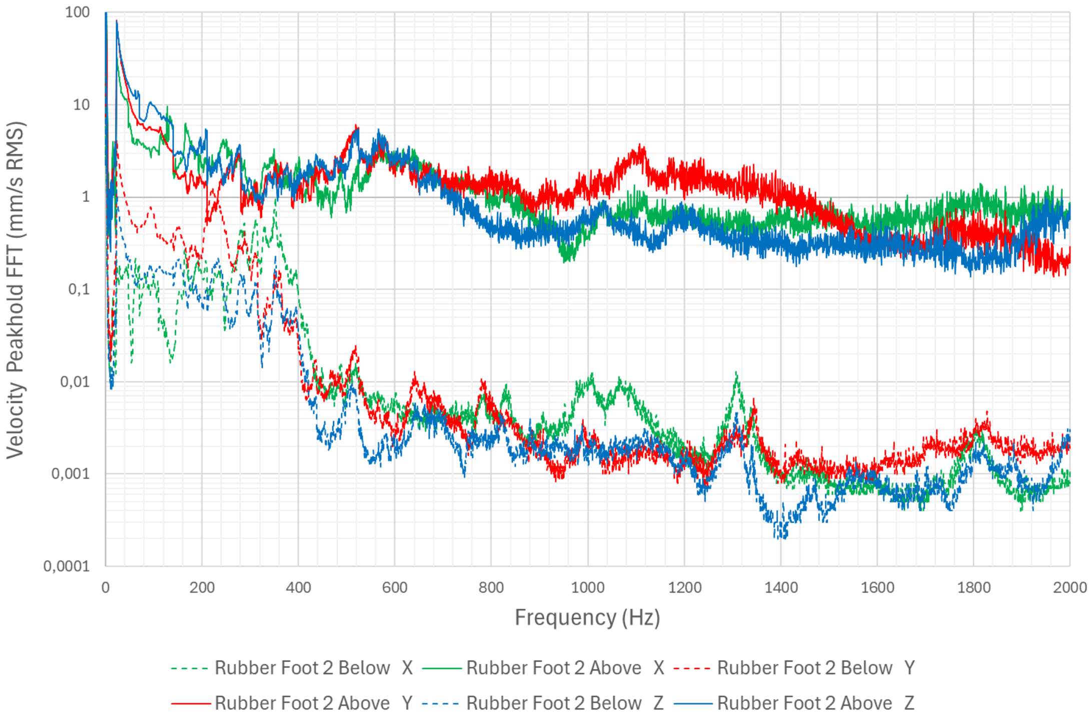

Figure 3. Vibration velocity peak-hold. Three directions – rubber. Sweep test from 2100 to 700 rpm.

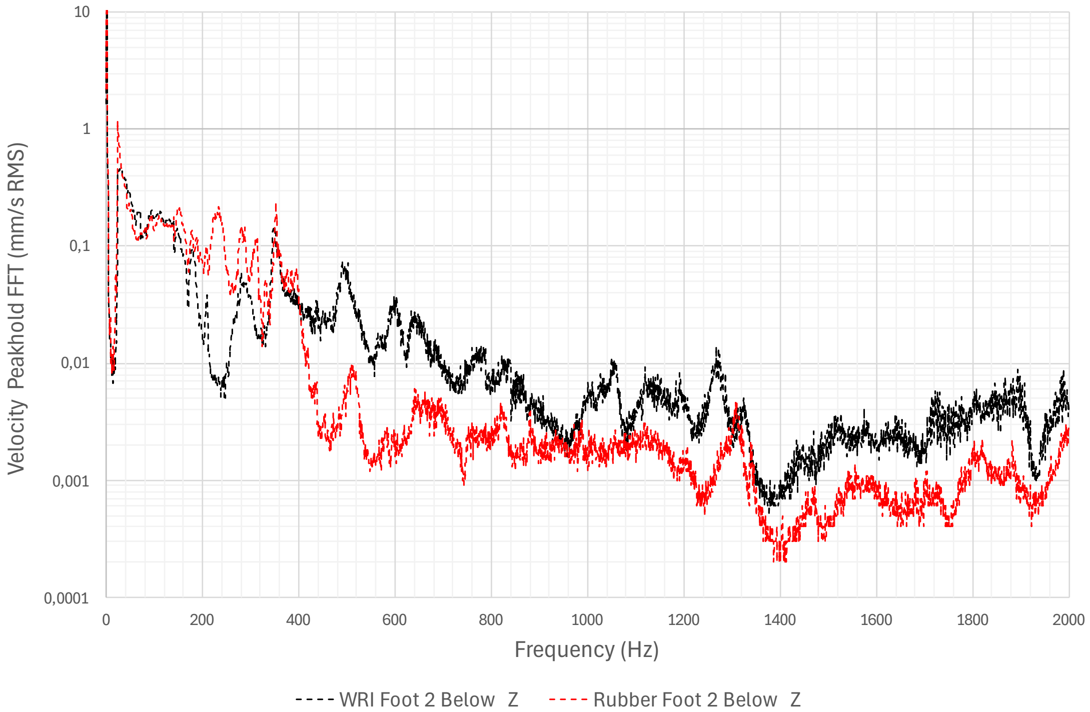

Figure 4. Vibration velocity peak-hold. Vertical direction. Red rubber black WRI. Sweep test from 2100 to 700 rpm.

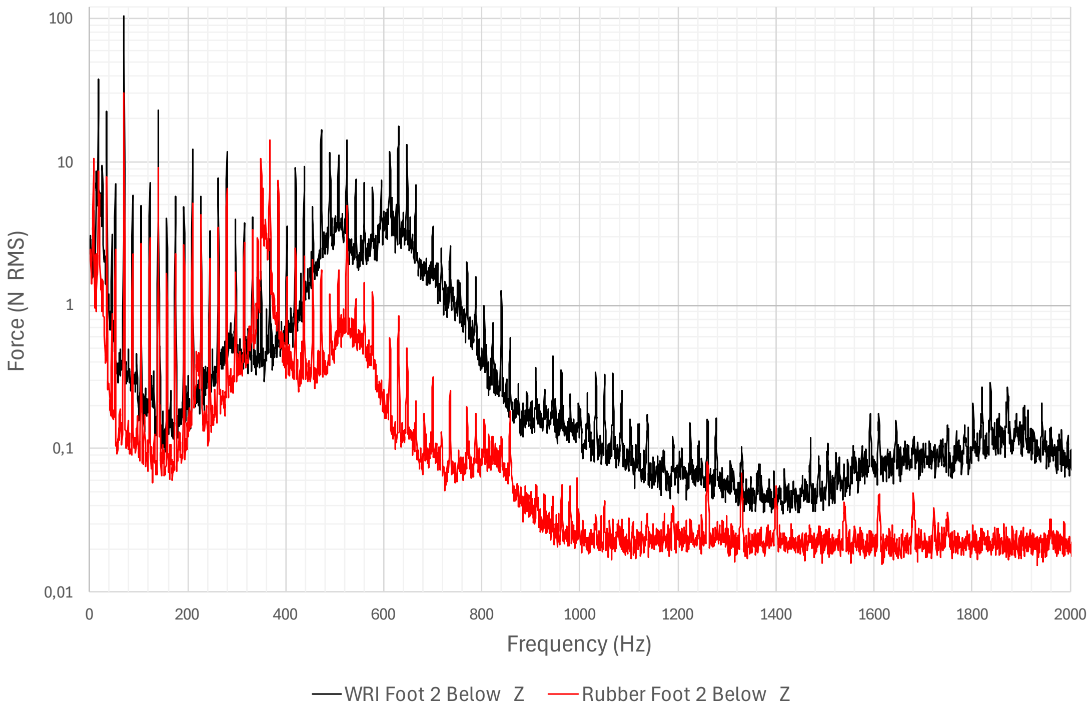

Figure 5. Strain gauge measurement results in vertical direction. Nominal 2100 rpm. Red: rubber and black: WRI

At 400 Hz to 800 Hz frequency bandwidth the isolation ability of rubber isolator is clearly better than WRI. At 10 Hz to 400 Hz the WRI isolation ability is better. These differences can be related to strain gauge measurement arrangement. That created a new spring like dynamic element to measurements. At lower frequencies the amplification of WRI is clearly lower than rubber.

DISCUSSIONS

First tests were done on January 16th and 17th 2024 and some unclear results were noticed. For example, the foot of the isolators had higher vibration than expected in transversal direction. Later it was discovered that the engine was not mounted to the floor properly. All the tests were repeated to get correct results. The engine was installed properly, and new tests took place on 7th and 8th of February.

The strain measurements were done using separate part that was attached below the isolators. The strain gauges need some flexibility in order to get a good signal. If the strain gauges are installed to very stiff component, for example the engine support foot then signal / noise ratio can be poor. It was decided to use this kind of method in order to get good results. There was a risk that the added part would create a spring that would interfere with the measurements. Unfortunately, this was the case. The flexible part created disturbances to higher frequencies which were of interest. In the beginning of the project there was the idea to do the measurements using commercial force sensor. A proper sensor was not available, and it was decided to do measurements with own strain gauges.

In addition, some further aspects need to be considered. In this research, only one type of WRI and one type of rubber isolator were compared. The isolation capability of other kinds of rubber can vary a lot compared to studied isolator.

Next vibration isolator measurement will be comparison test with WRI and original isolator of diesel engine in a ship. The WRI was designed to have similar stiffness to the original isolator of diesel engine. The biggest difference between the isolator types is the damping which is clearly higher with WRI. The tests will be conducted with accelerometers and strain gauges. The goal is to carry out the measurements in winter 2024-2025.

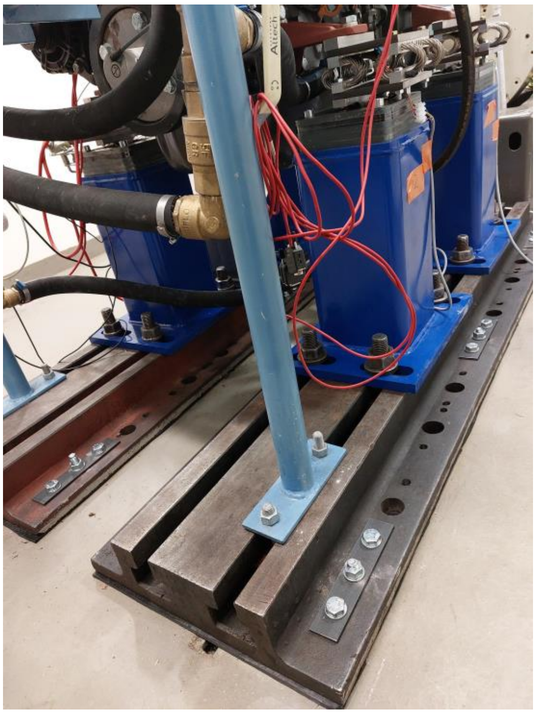

Figure 6. The floor installation with bolts.

CONCLUSIONS

The strain gauge measurement results show that the isolation of the rubber isolator was better than WRI at higher frequencies. Even though the strain gauge measurements were not reliable, the strain gauge installation also affects accelerometer measurement results. The difference was clearly lower with accelerometer results. At lower frequencies the amplification in resonance with rubber isolator is clearly higher compared to WRI. This means that isolation in robust environments where transient or sweep excitations occur the isolation is clearly better with WRI compared to rubber type isolator. Similar results were seen in laboratory measurements earlier using electro-magnetic shaker. Laboratory measurement results are typically more reliable than field measurements because the conditions are well known, and sensors are very accurate. It is recommended to repeat the measurements using commercial force sensor or install the strain gauges straight to base even though the measurement accuracy would be lower.

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Abstract

Title: Effects of non-uniform flow on a sound field generated by a finite-size uniform piston.

Ultrasonic gas flow meters have been increasingly used for fiscal metering of gas in recent years, and ultrasonic transit time difference flow meters are now considered as a realistic and competitive alternative to the use of more conventional technologies. In appropriate applications, such meters offer significant cost benefits and are planned to be used in subsea and remote operations, where traditional methods used topside cannot be used. In international trading of oil and gas that relies on high-precision, traceable, and accredited fiscal flow metering systems, the accuracy of transit time measurements may be affected by diffraction and side-flow effects.

Theoretical and computational studies of diffraction and non-uniform flow effects on an acoustic beam generated by a uniform piston source are needed to improve accuracy and are therefore investigated. The modelling of flow-acoustic interaction is based on a narrow-angle three-dimensional parabolic equation, where both magnitude and phase are presented to illustrate the mathematical results.

Keywords: Ultrasonic gas flow meter, diffraction effect, side-flow effect, subsea, parabolic equation.

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Advanced method for workplace noise analysis

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The awareness of the environmental noise impact has grown significantly, especially within workplaces. Many employers are striving to create a healthy and noise-free environment for their employees. However, heavy industries face challenges in adopting such practices. Worker complaints related to noise in factory settings are common, underscoring the ongoing importance of accurately assessing its impact. Presently, workplace noise regulations often rely on equivalent noise levels, but incorporating percentile distribution values such as P10, P50, and P90 offers a more comprehensive understanding of environmental noise. Additionally, frequency domain analysis provides crucial insights into noise source characteristics, aiding in the identification of root causes. Due to the diverse tasks and working conditions, various measurement methods for noise need to be implemented and compared. This approach enhances our understanding of noise characteristics, behaviour, and influencing factors. Worker involvement in the evaluation process, including feedback, is essential. A program developed by APL Systems and Wärtsilä enables the simultaneous analysis of personal noise dose and local noise for workplace assessments. The methodology presented in this paper enhances the accuracy of interpreting noise measurement results, facilitating effective actions for noise abatement.

1 Introduction

Year after year, noise retains its position as one of the most common problems in workplaces, particularly in the heavy industrial sector. While noise exposures have decreased across various fields of business, exposure levels are slightly rising in the construction and mining industries, as well as the metal industry. It has been assessed that approximately 480,000 workers are exposed to noise levels exceeding 80 dB (A), and around 190,000 workers are subjected to noise levels surpassing 85 dB (A) at their workplaces on a daily basis in Finland.

Annually, noise remains the primary cause of suspicions of occupational diseases. Each year, roughly 1,700 new cases of hearing losses or suspected losses are reported, with about 800 of them confirmed as occupational diseases initiated by noise exposure in Finland. The primary impacts of noise include incurable hearing losses and damages. Additional effects consist of tinnitus, heightened stress levels, sleep disturbances, psychological impacts, cardiovascular changes and issues, and delayed reaction times. Furthermore, noise impedes communication in workplaces and heightens the risk of accidents.

2 Wärtsilä factory noise history

In Wärtsilä, occupational safety and health have been taken seriously for many decades, particularly by blue-collar representatives who prioritized noise issues. This emphasis led to the first workplace noise measurements in the mid-1980s. At that time, simple handheld sound meters were used to assess overall noise levels in various areas of the Vaasa factory, with documented results identifying the noisiest locations.

In the mid-1990s, the first factory noise project was initiated with the goal of understanding workers' daily noise exposure (LEP,d) and reducing noise levels in different workplaces. Wärtsilä purchased three Bruel&Kjaer dose meters (type B&K 4436) for this purpose. By 1994, personal noise dose measurements were conducted, totalling around 200 over the next two years at the Vaasa factory. These results proved invaluable in identifying and addressing hearing damage among workers. Additionally, Wärtsilä collaborated with the Turun aluetyöterveys institute (TATL), led by Valtteri Hongisto and Vesa Viljanen, to reduce noise levels in its factories. This collaboration resulted in the adoption of the ODEON simulation program, which facilitated the reduction of noise levels in various departments.

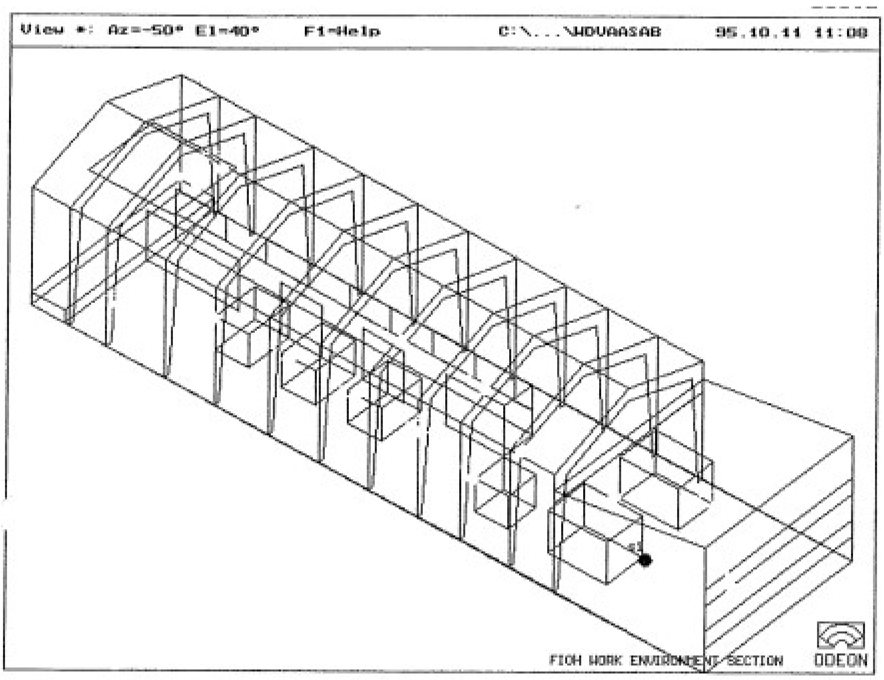


Figure 1: ODEON model for one Vaasa factory department, 1995

Despite the simplicity of the model compared to today's technology, it effectively guided efforts to measure sound power levels and reduce noise. After four years of work, noise levels in the department decreased by more than 5 dB, indicating the project's success. Similar strategies were replicated in other departments.

Around 2010, a subsequent factory noise project aimed to pinpoint noise sources from different working machines, such as compressor air tools. This endeavour highlighted the need for improved measurement instruments capable of capturing one-third-octave spectra and storing data at one-second intervals. Implementing these advancements would provide more comprehensive information about the surrounding noise environment. Similar concerns applied to noise dose meters, questioning whether a single L(Aeq) value adequately describes daily exposure noise over an 8-hour period. Despite the availability of 1-minute L(Aeq) values, much of the noise data remains underutilized, prompting further inquiry into why this is the case.

3 New advanced noise instruments

APL Systems and Wärtsilä have extensive experience in analysing indoor and environmental noises, with their collaboration dating back to 2008. The AuresSound® system offers continuous sound recording, live data streaming, and comprehensive reporting capabilities. AuresSound® measurement devices may be used as independent measurement stations or as part of a network of sound measurement devices. The user interface is intuitive, providing results in both the time and frequency domains.

Recently Wärtsilä has purchased Spartan 730 noise dosimeters from Larson Davis. The dosimeter can be used as stand-alone device or remotely controlled device by mobile phone or laptop via Bluetooth in terms of monitoring, making setups and controlling the unit. The measurement report will be generated by using the LD Atlas app. The app will also generate and download all data files. There is also option to record the audio files of events and overall octave band analysis. The time domain results are able to be saved with 1s interval as minimum.

Figure 2: Spartan 730 and AuresSound noise instrument.

4 Advanced analysing method

The ability to conduct long-term recording and analysis of surrounding noise in both time and frequency domains has introduced a new methodology for environmental noise investigation. The technology of frequency domain measurement and analysis enables a better understanding of the components of specific noise sources, leading to more reliable measurement data compared to calculations. In many cases of long-term recording, the values of P10 and P90 appear to be even more significant than L(Aeq).

Saara Seppänen initiated her M.Sc. thesis in 2014 to develop advanced analysis methods for indoor noise in factories. In her thesis, she outlined how results should be managed and analysed to derive the most useful information from workplace noise. One new parameter she introduced was the reverberation time of different departments. However, the analysis was time-consuming as all one-second measurement data had to be analysed in Excel. Additionally, obtaining personal noise dose measurement data in ASCII format with the new B&K Analyze 4445 proved challenging. It became evident that creating a custom-made program to analyse measurement input data would be crucial in the future. Following Saara's thesis, the noise project continued for another 2-3 years.

4.1 Aures Noise

Aures Noise is a sound analysis tool where you can input data straight from AuresSound devices but also from external systems while a certain formatting rule is followed for the input file. Aures Noise then categorizes these datasets to measurement periods which the user can give a description for easier usage see figure3/left

Using Aures Noise it is then possible to adjust the range from which time the data is wanted, select level for L(Aeq) and L(Cpeak) from where the program calculates the time these have been exceeded during the time range selected. After this you can select whether to display data in one graph or individual graphs for every device, figure 3/second.

After selecting the visualization type program starts processing the data. It will calculate 1 minute average for LAeq, P10%, P50% and P90% from the L(Aeq,1sec) data and also 1 minute average for L(Cpeak). Also, for every hour in the selected time range the program calculates the following, figure 3/third:

- Average value for L(Aeq).
- Average value, minimum value and maximum value for P10%, P50% and P90%.
- Maximum value for L(Cpeak)

If the data uploaded includes the 1/3 octave bands the program also calculates the average 1/3 octave for the measurement period, also automatically recalculates the graph if user zooms to a certain time period. Also, for the hourly values it is possible to display the 1/3 octave bands from that time. The indication for the graph can be seen in figure 3/right.

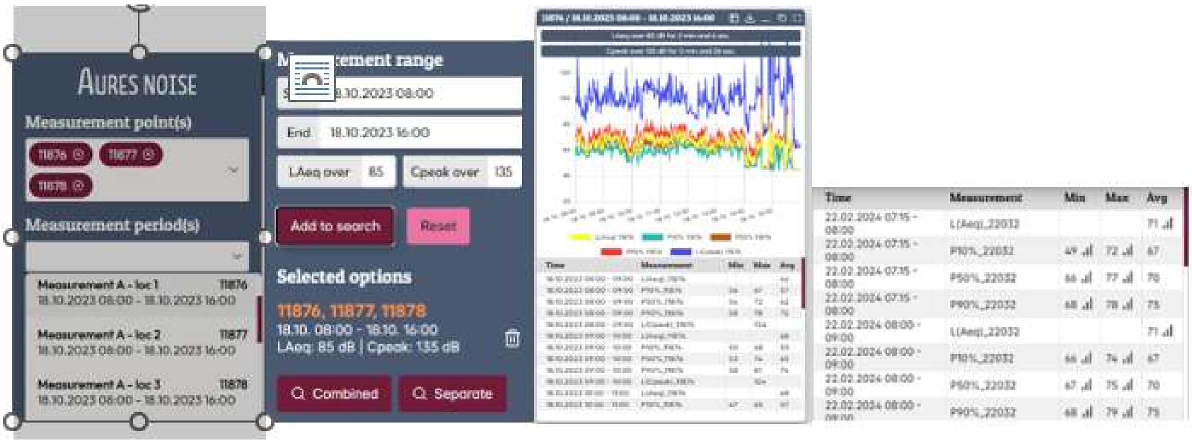


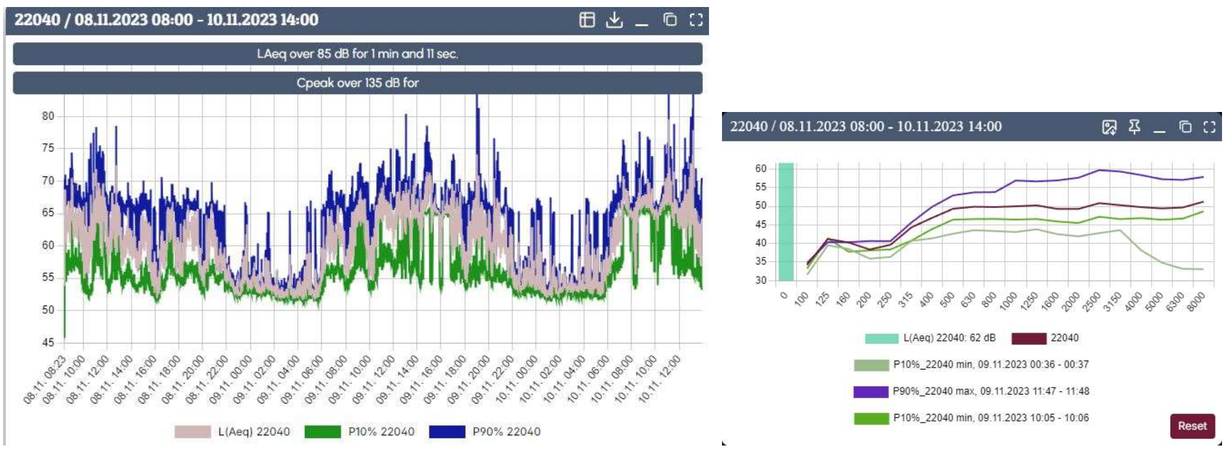
Figure 3: Data is uploaded from three different units (11876,11877 and 11878) and the system categorizes them according to the measurement time. Adjusting time range and select sound levels for LAeq and Cpeak. Time those levels are exceeded will be then calculated. Underlined is the button to press where you can open the certain 1/3rd octave band graph

4.2 STH – Wärtsilä new Sustainable Technology Hub

Wärtsilä commenced the construction of a new factory on Vaasa's Vaskiluoto in 2019, with the majority of employees relocating from our old urban facility to Vaskiluoto by 2022. This expansive new factory spans approximately 220 by 150 meters and stands at a height of 15 meters, aiming to centralize all production activities under one roof. During the design phase, noise considerations were paramount, with A-insinöörit primarily responsible for acoustical design, supported by the provision of historical noise reports by Wärtsilä.

The third factory noise project commenced in spring 2023, focusing on a custom noise program named Aures Noise, procured from APL System. Initial data input included one-second noise data from Aures data loggers and Larson&Davis noise dose meters. The program computed 1-minute L(Aeq) values, alongside P10, P50, P90, and L(Cpeak) values for each minute, leveraging the capabilities of Aures instruments to calculate 1/3 octave noise spectra. The program's development spanned three months.

Figure 4 displays two days of Aures measurement data from distinct locations, with instruments positioned approximately 4-5 meters from the work area but in different orientations. The upper curve illustrates P10, P90, and L(Aeq) values, alongside the overall 1/3 spectra on the upper right. Noise levels fluctuated by approximately 10-15dB in this workspace, with 80dB levels rarely exceeded. Nighttime factory noise levels hover around 55dB, reaching maximum levels of 75-80dB. Figure 5 depicts personal noise dose measurements, revealing fluctuating noise levels throughout the workday and similar noise levels among different individuals measured at the same workplace.



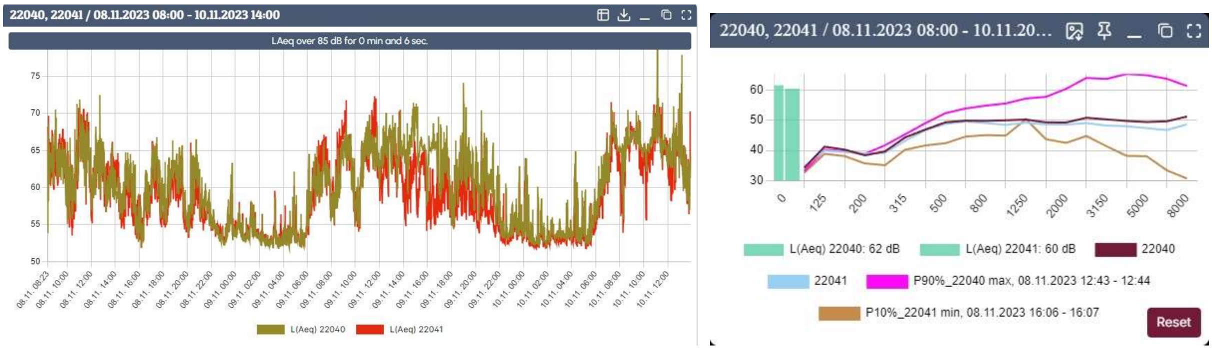


Figure 4: Two days noise L(Aeq) values of two different Aures instruments, left. On right one-third octave overall spectra with 1 min maximum and minimum 1/3 spectra.

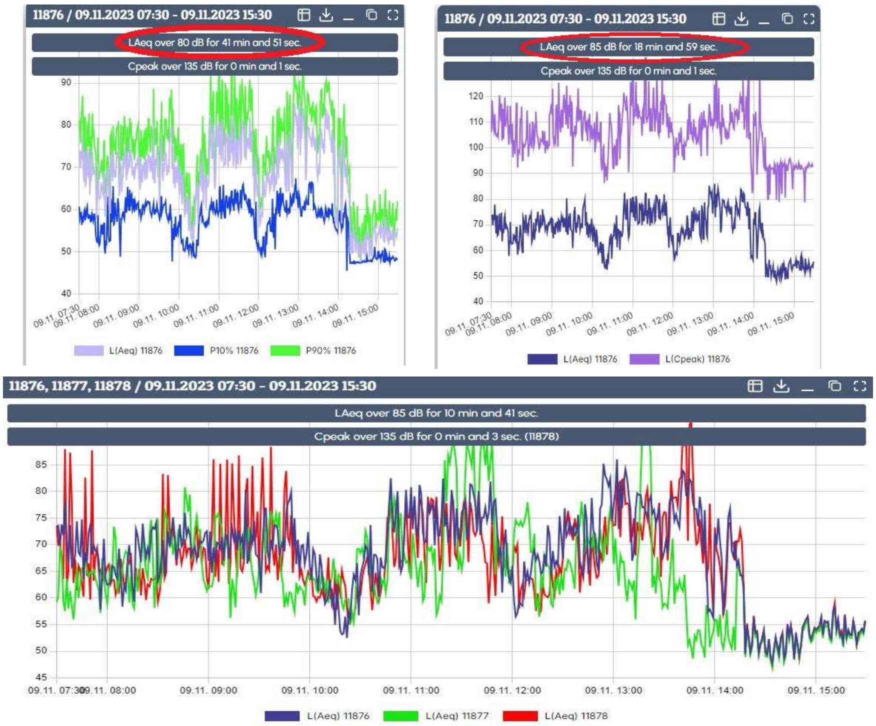


Figure 5: Upper left curve is presented a person P10, P90 and L(Aeq) and on upper right is presented L(Cpeak) and L(Aeq) values during measurement day. At same time is presented time what person is exceeded 80dB, time 42 minutes, and 85dB 19 minutes. On lower curve is presented three different person noise dose levels at same working area.

Most workers find the measurements very favourable because they accurately described their workdays.

When all measurement data are input into the program, the analysis of workplace noise takes about 2 days, whereas previously it took 1.5 weeks. Therefore, the custom-made program is a huge help in analysing noise at different workplaces. The manufacturer's own program is too simple for advanced analysis.

5 Summary

The possibility of long-term recording and analysing factory noise, both in time and frequency domains, has introduced a new methodology for environmental noise investigation. The technology of frequency domain measurement and analysis has enhanced our understanding of the components of specific noise sources. The values of P10 and P90 appear to be even more useful than L(Aeq) in many cases of long-term measurements. However, the most crucial aspect is a custom-made analysis program that aids in analysis. It would be beneficial to include a percentile distribution for L(Cpeak) values as well.

The Spartan730 proves to be a powerful noise dosimeter. The only thing missing is a spectrum option, such as automatically reporting 1–5-minute 1/3 octave values. With this addition, the instrument would become exceptionally powerful.

Preliminary conclusion showed that even reverberation time in factory is quite high, 2.1-2.3 seconds, the noise not seems to be big issues. The noise mainly concentrates on local workplaces and attenuation quickly, because big volume of factory and no reflective surface area.

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Comparison of results from road noise measurements and road noise calculation methods CNOSSOS-EU, Nord2000Road and NBV96

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To gain more experience with CNOSSOS-EU a series of road noise measurements was conducted to compare actual noise levels with calculated results. These measurements were taken along an urban four-lane motorway in Oslo with speed limit 70 km/h, during the summer of 2023, in accordance with Norwegian standard NS 8174-02 as far as practicable. The resulting road noise levels $L_{p,Aeq,\,24t}$ were compared to the calculated levels obtained using three different methods for road traffic: NBV96, Nord2000 and CNOSSOS-EU. The measurement locations were chosen to span from open areas to more complex situations. The key finding is that the calculated noise levels exceed the measured levels across all three calculation methods. The discrepancies between measured and calculated levels are highest in open areas and tend to diminish in more complex scenarios. Specifically, Nord2000 yields higher levels than NBV96 and CNOSSOS-EU, although the differences between Nord2000 and the other two methods become negligible in complex situations. These findings indicate a need for further investigation into source data for vehicles, and potentially road surfaces for Norwegian conditions. Additionally, the transition in software from Cadna/A to SoundPLAN influences the results to some degree.

1 Introduction

In connection with the adoption of the CNOSSOS-EU calculation method for road traffic noise assessments, Multiconsult sought to examine calculations performed using CNOSSOS-EU and compare the results with measurements of road traffic noise. The measurements were carried out as an internal project during the summer of 2023. On behalf of The Norwegian Public Roads Administration (Statens vegvesen), the internal project was expanded to include additional calculations with NBV96 and Nord2000.

2 Measurements

Road traffic noise measurements was conducted at 34 different locations along Rv. 4, between Sinsenkrysset and Linderud in Oslo, Norway in June through August of 2023. These measurements were carried out in accordance with NS8174-2 [1] as far as practicable. Independent measurements were taken on three different days at each of the 34 measurement locations. The measurement locations are distributed across three areas: Linderud, Årvoll, and Bjerke, as indicated in figure 1. The rationale for selecting the areas for road traffic noise measurements stems from Multiconsults earlier noise calculations along Rv. 4 in Oslo, Norway, on behalf of The Norwegian Public Roads Administration [2].

Measurement locations were strategically chosen to cover a range of scenarios, from open areas (relatively straightforward situations) to more sheltered and built-up areas with complex reflection conditions. Emphasis has been made to achieving variation in terms of height difference between the source and receiver, shielding conditions, urban density, and distance to large and reflective buildings, as shown in table 1. The measurements were taken at distances ranging from 17 meters to 91 meters from the centreline of the road at a height of 1,5 m above ground level. The speed limit was 70 km/h. Temperatures spanned from 16°C to 31°C, and wind speed was less than 6 m/s.

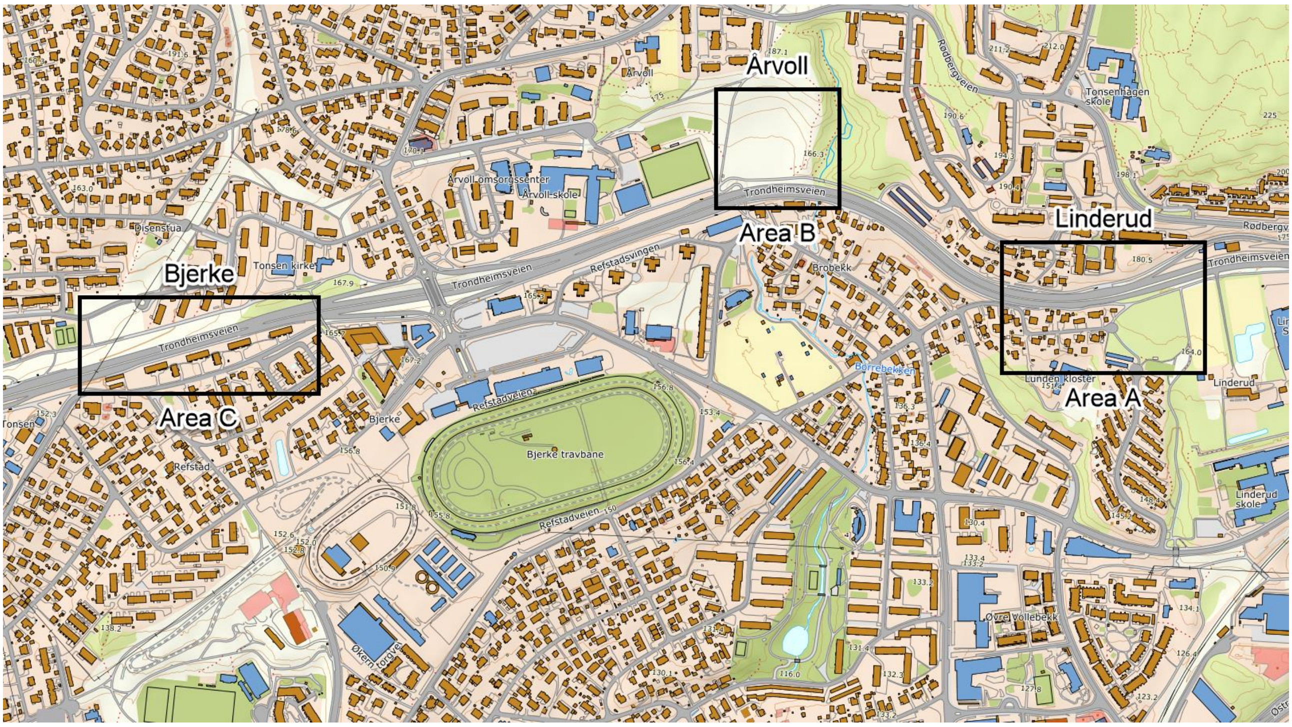

Figure 1: Overview of measurement location areas.

Table 1: Overview of parameters that can affect the noise level at the different measurement locations.

Measurement location		Height difference between source and receiver	Shielding conditions	Building conditions	Estimated geometrical complexity		
					Simple	Medium	Complex
A	1	None	None	Open	X		
	2				X		
	3				X		
	4	2–5 m below road	Noise barrier	Houses	X		
	5			Apartment buildings		X	
	6			Houses		X	
	7					X	
B	1	None	Noise barrier	Open		X	
	2					X	
	3					X	
	4					X	
	5		Partial noise barrier		X		
	6	5 m above			X		
C	1	None	Noise barrier	Open		X	
	2					X	
	3					X	
	4					X	
	5					X	

	6	5 m below road				X	
	7					X	
	8					X	
	9					X	
	10					X	
	11	None	Noise barrier and varying shielding from apartment buildings	Apartment buildings			X
	12						X
	13						X
	14						X
	15						X
	16						X
	17						X
	18						X
	19						X
	20						X
	21						X

3 Calculations

The calculations were performed using Cadna/A (Version 2023 MR2, build 201.5366) and SoundPLAN Noise 9.0. For the Cadna/A calculations the methods NBV96 [3] and CNOSSOS-EU [4], [5] were used. For the SoundPLAN calculations the methods CNOSSOS-EU and Nord2000 [6] were used. Calculation settings for CNOSSOS-EU were set according to the recommendations in Handbook for use of CNOSSOS-EU in Norway (Håndbok for bruk av CNOSSOS-EU i Norge) [7], and calculation settings for Nord2000 were set according to Handbook V717 – User guideline Nord2000 Road (Håndbok V717 – Brukerveiledning Nord2000 Road) [8]. The calculations settings are described in detail in Multiconsult report 10253612-01-RIA-RAP-001 [9].

3.1 Meteorology

According to Handbook for use of CNOSSOS-EU in Norway recommended settings for meteorology, based on Norwegian weather statistics, is favourable conditions in 50, 60 and 70 % for day (7-19), evening (19-23) and night (23-7) respectively. This is a conservative estimate for favourable propagation.

To compare the measured noise levels with the calculated noise levels the air and road temperature was set to an average of the measured temperature for the three measurement days.

3.2 Reflections

Noise levels were calculated at 1,5 m height above ground level including up to 2nd order of reflection. The impact of using 3rd order reflections has been tested.

3.3 Vehicle speed

The vehicle speed was set to 70 km/h in the calculations. Available road speed data indicates an average speed of 75 km/h during the measurement periods. If a speed of 75 km/h was considered the noise levels would increase by up to 0,6 dB.

3.4 Road surface

Road surfaces are in general noisier in Norway than most European countries due to the use of studded tyres. However, the calculations are performed using reference road surface for the CNOSSOS-EU method. Reference for Norwegian road surfaces are for the present not specified. In the Nord2000 calculations, data for a newly laid SMA11 is used, and not corrected as recommended in Handbook V717 – User guideline Nord2000 Road.

As the measurements were taken during summertime, studded tire fraction was not included in the calculations.

4 Results and evaluation

Road traffic noise was measured three times per measurement location, apart from measurement point A5, which at the has been measured twice (measurement no. 2 carried out in October 2023). The measured noise levels have been adjusted for YDT (yearly average daily traffic) and then averaged. The measured noise levels are stated as an expected equivalent A-weighted sound pressure level for each measuring point, with a corresponding confidence interval. The confidence intervals are calculated in accordance with NS 8174-2, Appendix B.

4.1 NBV96 and CNOSSOS-EU calculated in Cadna/A

The measured expected noise level $L_{p,Aeq,24h}$ for each measurement location is shown in figure 2. The figure also includes the calculated noise level for NBV 96 and CNOSSOS-EU. The measured expected noise level per measurement location is shown with corresponding confidence interval. The figure shows the calculation results where the road surface temperature and air temperature are set to an average of the measured temperature for the three different measurements days.

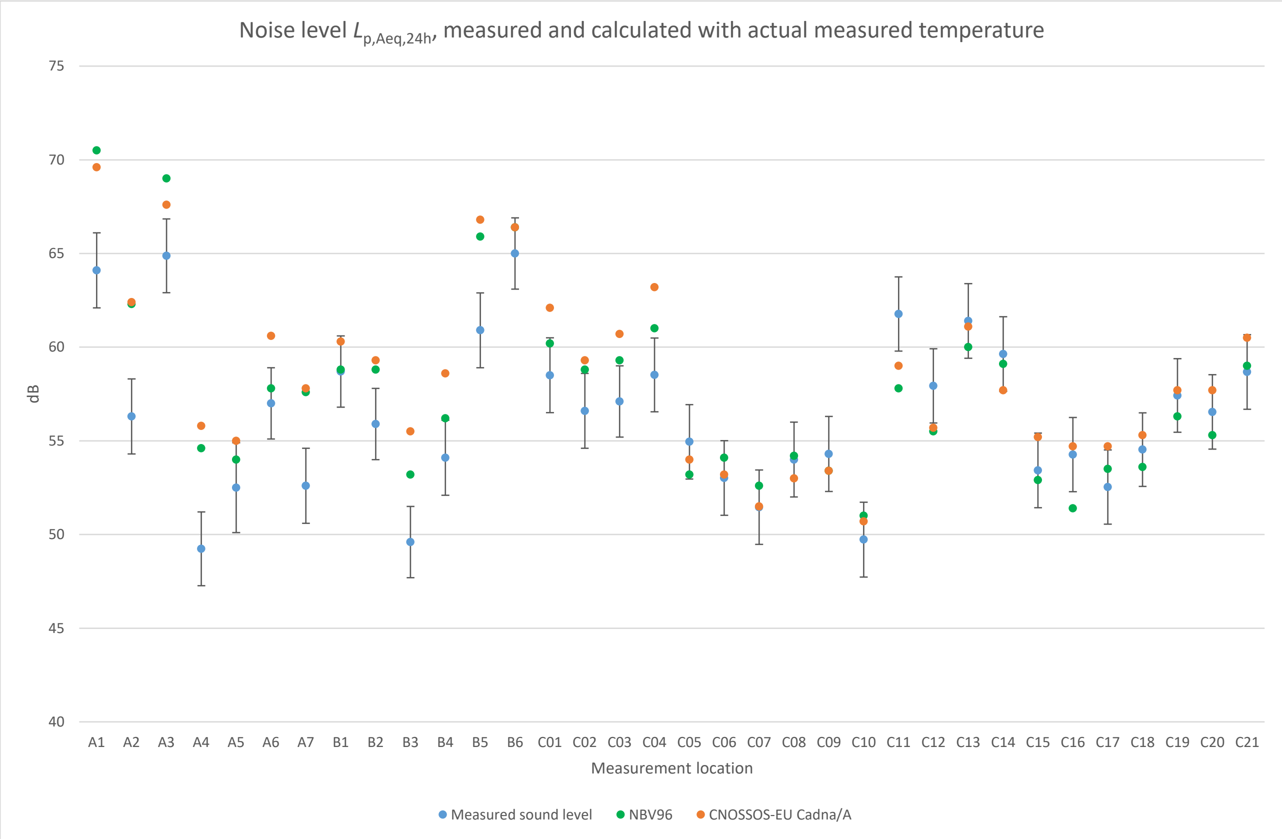

Figure 2: Measured expected noise levels $L_{p,Aeq,24h}$ with confidence intervals compared to calculated noise levels $L_{p,Aeq,24h}$ for NBV96 and CNOSSOS-EU with temperatures corresponding to actual measured temperatures. Calculations performed in Cadna/A.

4.2 CNOSSOS-EU and Nord2000 calculated in SoundPLAN

To compare measured noise levels to calculated noise levels for Nord2000, calculations have been performed in SoundPLAN. Noise levels have also been calculated for CNOSSOS-EU in SoundPLAN to gain more information about how the change in software affects the calculated noise level.

The calculated noise level for each measurement location for CNOSSOS-EU and Nord2000 are shown in figure 3. Calculated noise levels for CNOSSOS-EU performed in Cadna/A are also included to illustrate the effect of change in software. Noise levels calculated using CNOSSOS-EU in SoundPLAN are lower than the noise levels calculated in Cadna/A.

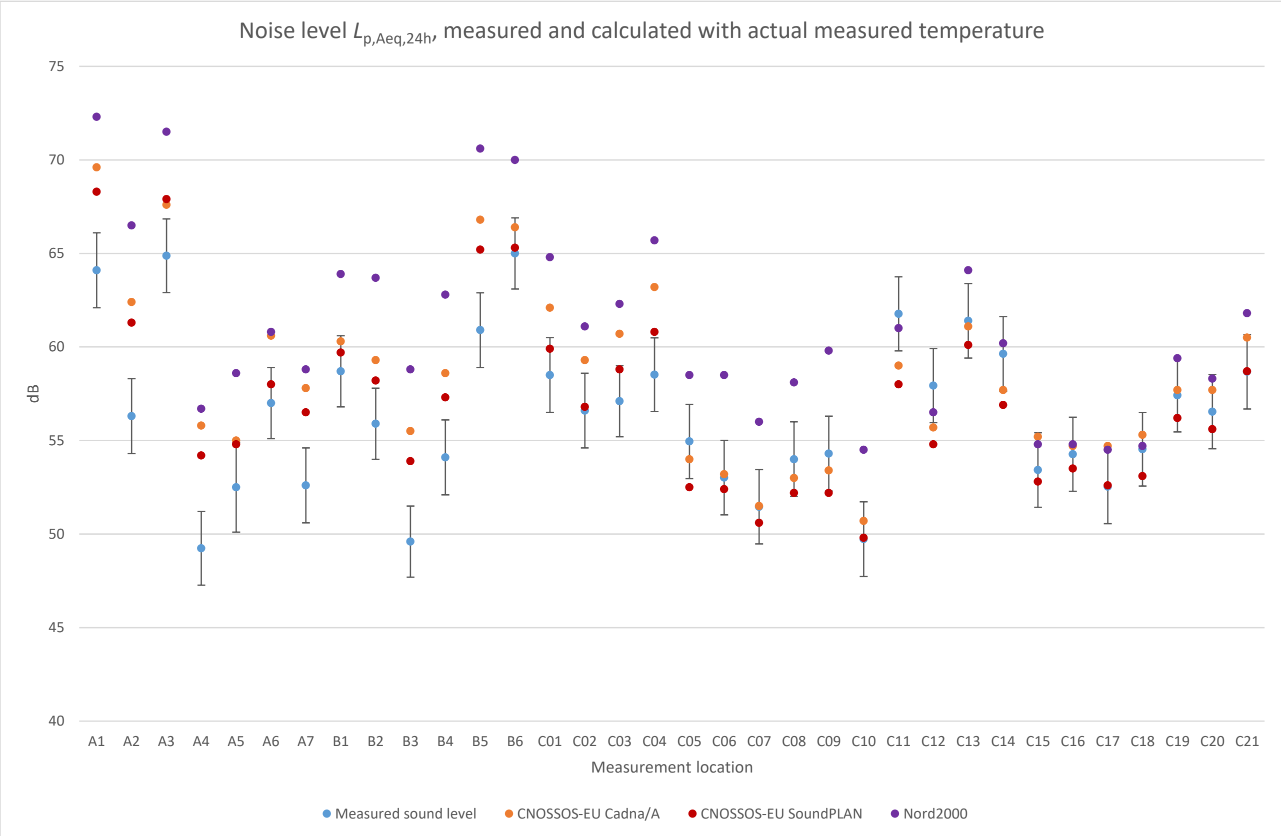

Figure 3: Measured expected noise levels $L_{p,Aeq,24h}$ with confidence intervals compared to calculated noise levels $L_{p,Aeq,24h}$ for CNOSSOS-EU and Nord2000 with temperatures corresponding to actual measured temperatures. Calculations performed in Cadna/A and SoundPLAN.

4.3 Effect of geometrical complexity

To better understand the reasons for discrepancies between measured noise levels and calculated noise levels, we have examined the results considering geometric complexity, as given in table 1. The table indicates whether there is a difference in elevation between the road surface and the measurement point, whether there is shielding (such as barriers or buildings), and whether the area is open or densely built. Based on this information, we have categorized the conditions as simple, moderate, or complex in terms of geometry. The deviation between measured and calculated noise levels $L_{p,Aeq,24h}$, sorted by geometric complexity is shown in figure 4. The confidence interval of ±2 dB is included. It's worth noting that for measurement location A5, the confidence interval is ±2.4, as only two measurements were taken for this location.

Figure 2 to figure 4 reveal relatively significant discrepancies between measured and calculated noise levels $L_{p,Aeq,24h}$ for all calculation methods at measurement points A1 to A4. These measurement locations share the common characteristic of being primarily situated in open terrain without shielding from noise barriers or buildings. Note that there are double Jersey barriers (height 0,8 m) between the lanes along the entire road. These barriers affect the sound transmission, so none of the situations can be considered as geometrically simple. It was initially expected that there would be smaller deviations between measured and calculated noise levels for these measurement locations due to the straightforward conditions. Unfortunately, almost none of the calculated points fall within the confidence interval of the measurements. While Nord2000 exhibits the largest deviations, there are also significant discrepancies for the other calculation methods.

For the measurement locations with medium complexity there are still large discrepancies between measured and calculated noise levels. Nord2000 still has the largest discrepancies and highest calculated noise levels, compares to the measured values.

For the complex measurement locations, several factors contribute to the discrepancies between measured and calculated noise levels. These factors include shielding (sometimes from noise barriers and buildings) and the presence of multiple buildings with large reflective surfaces that can impact noise conditions. Interestingly, the three calculation methods exhibit better alignment with the measured noise levels in these scenarios. A significant number of calculated points fall within the confidence interval, and the apparent difference between the Nord2000 method and the other two calculation methods diminishes.

It's worth noting that in these complex situations, calculations including a higher number of reflections become relevant. If this was to be considered, the calculated noise levels would be slightly higher for the complex scenarios.

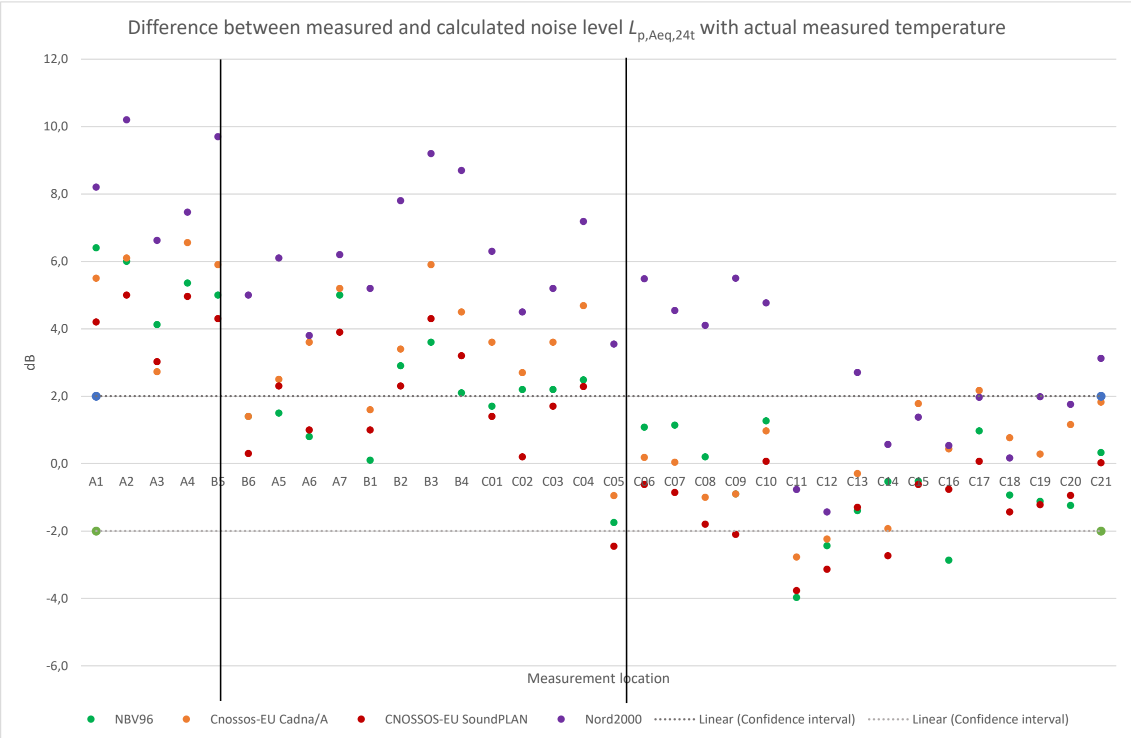

Figure 4: Deviation between measured and calculated noise level $L_{p,Aeq,24h}$ for different calculation methods, sorted according to geometrical complexity. The complexity is divided by the black vertical lines and is sorted from simple to complex. The confidence interval is shown as dotted lines (+/- 2 dB).

5 Summary

The results from the comparisons indicate that measured expected noise levels are lower than the calculated noise levels, regardless of whether the calculations were performed using NBV96, CNOSSOS-EU, or Nord2000. The largest discrepancies between measured expected noise levels and calculated levels occur in the situations with simple complexity, where initially better correlation was expected. This indicates that further investigations should be made into source data for vehicles and road surfaces for Norwegian conditions.

Note that there are double Jersey barriers (height 0,8 m) between the lanes along the entire road. These barriers affect the sound transmission, so none of the situations can be considered as geometrically simple. The noise reducing effect of the Jersey barriers, and how they are handled in the calculation methods should be further investigated.

It's important to note that this study is limited to a road with speed limit 70 km/h and locations for measurements in distances from the road ranging from 17 to 91 meters from the centreline.

6 Acknowledgements

This work was supported by The Norwegian Public Roads Administration (Statens vegvesen). The measurements were conducted by Hanna Beate Frøiland, Kristian Hallaråker Reigstad, Franz Dannenberg og Nicolas Sogge.

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A city designed using sound – A tool to make spatial decisions

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1 Abstract

This paper focuses on the spatial planning and architectural design for an area in Tallinn next to Linnahall by using noise mapping tools in such a way as to create an acoustically less stressful place to live and work in and to reduce unwanted sound in public areas. By doing so this creates a lively place to live and work in, while keeping the soundscape interesting and alive throughout the year.

People sense space in which they're in – if it's big or small. We can see dimensions and sense the scale of the city. Like we see the city, the buildings and the rooms, we also hear them and everything, which surrounds us. By hearing we can understand if the space around us is big or small, lively or dull, alive or damp.

In Tallinn the dominating noise is traffic noise. Areas like Vanalinn or Kadrioru park and Noblessner offer some relief from noise, but they're not planned as an acoustic space. Tallinn is a port city, but the seaside areas do not offer enough escape from the busy streets.

This project consists of different phases – methodology, analysis & creating a spatial tool for noise reduction in spatial planning. The analysis phase focuses on buildings, which purpose includes acoustics. Through the analysis a set of criteria is formed, which can be used to locate possible concert hall locations in Tallinn. The locations are then compared to each other using properties of sound, through which a final location is chosen.

A study on noise will be done for the final location. Possible noise sources will be mapped along with a noise map. Based on the map and possible sources of sound, a methodology is created, through which acoustic properties in relation to noise of the area can be impacted.

In the planning phase, the methodology is used. First the buildings will be used as noise barriers. After this locations with higher qualities in terms of acoustics and architectural design are created.

As a result, a masterplan design employing noise as a spatial tool creates a cohesive connection for the seaside areas and an interesting and quiet part of the city in which people can rest whilst living and working in the area.

2 Introduction

The world of sounds is nothing new. We hear sounds from an early stage in our lives and we are used to living in different soundscapes our whole lives. Different voices, sounds allow us to sense the space we are in, they allow us to recognize dangers and communicate. In our daily lives we tend to forget that we can hear the world, so we focus more on the visual contact with it.

Sounds are associated with noise – everything we do not want to hear, we filter out into something unimportant called background noise. As background noise differs from situation to situation, we tend to lose the ability to listen to the soundscapes around us. In the scope of a city, the moments of realization of a change in soundscape can be mostly be noticed only, when the change is sudden.

3 Soundscapes

Every city has its own soundscapes, which can be described with combining all of the sound sources present there. These can be birds, cars, technical equipment, trees, people etc. Every space is described by a different soundscape. Soundscapes change depending on the space the listener is in. Through this a city can be mapped through sound creating soundwalks. [1]

Soundwalks can be described through time, as the sounds of horses working on a field, the sound of a bell starting and ending the workday change to the first cars, tractors, the sound of a bell turns to a signal. [1] The noise coming from the fields, factories is being pushed further away from the cities, but the “horses” are still ever present and the sound of the start of the workday can be chosen “freely” on our phones.

Going into the future raises the question of where is the soundwalk of a city through time headed?

4 City planning

The soundscapes indoors can be changed through the means of building and room acoustics and outside it is only thought of through traffic and technological equipment. A key feature of city planning and geometry of buildings is missed – they can be used as noise barriers.

Placing a building parallel to a road will allow for higher noise levels on that side of the façade, but by the means of building acoustics the effects indoors can be minimized. On the quiet side of the building forms a space, which has a higher acoustic quality. Building geometries, which allow focusing of sound should be discouraged.

As an extra losing visual contact with traffic, the psychological impact of noise coming from cars can be decreased. [2]

Planning a building, which runs along the perimeter of an estate block, will form a quiet space inside it. This space can be designed to offer peace from traffic noise without the use of complex solutions for lowering noise.

On the scale of a city all three of these methods come into play. By using these three methods, a city which allows the use of public functions, is acoustically more diverse and interesting and lowers the impact of traffic noise to health can be designed. It’s key to understand that using distance to lower noise levels is not enough and new more precise methods need to be used. For example when planning the city of the future, trees which allow for songbirds to thrive should be chosen, in key noisy areas fountains should be placed to mask the sound of traffic.

5 Concert houses & multi-functional centers in a city

Concert houses are buildings, which have a clear function, they are usually separate buildings and they work as an anchor for the surrounding areas. They can be thought of as sacred sites, where entry is not easily granted. To get the experience of the features of the building or its surroundings a special event or a concert has to be held. This can lead to a concert house being like a office block, which fills during a concert/workday and empties afterwards. The only difference is architecture, which most probably is more luxurious for the concert house.

Multi-functional centers are planned with connectivity with the city in mind. Even though their architecture can be more expressive than a regular residential or office building, they are planned with functionality in mind. They can mostly operate 24/7 and the psychological effect of exclusiveness is more faded so they feel more open.

Within the limits of this paper, the proposed concert house has to have an exclusive yet open design, which would give the feeling of a culturally important location. The façade has to be open and take its surroundings into account. The area around the concert house has to planned in a way, which fades the borders between exclusiveness and openness.

6 Tallinn & noise

Tallinn is an uncomfortable place to live acoustically – large magistrals cut through the city and the amount of cars is high, which translates into noise levels. It is difficult to get some rest from traffic and the greenery is not noticeably present in the city. [3]

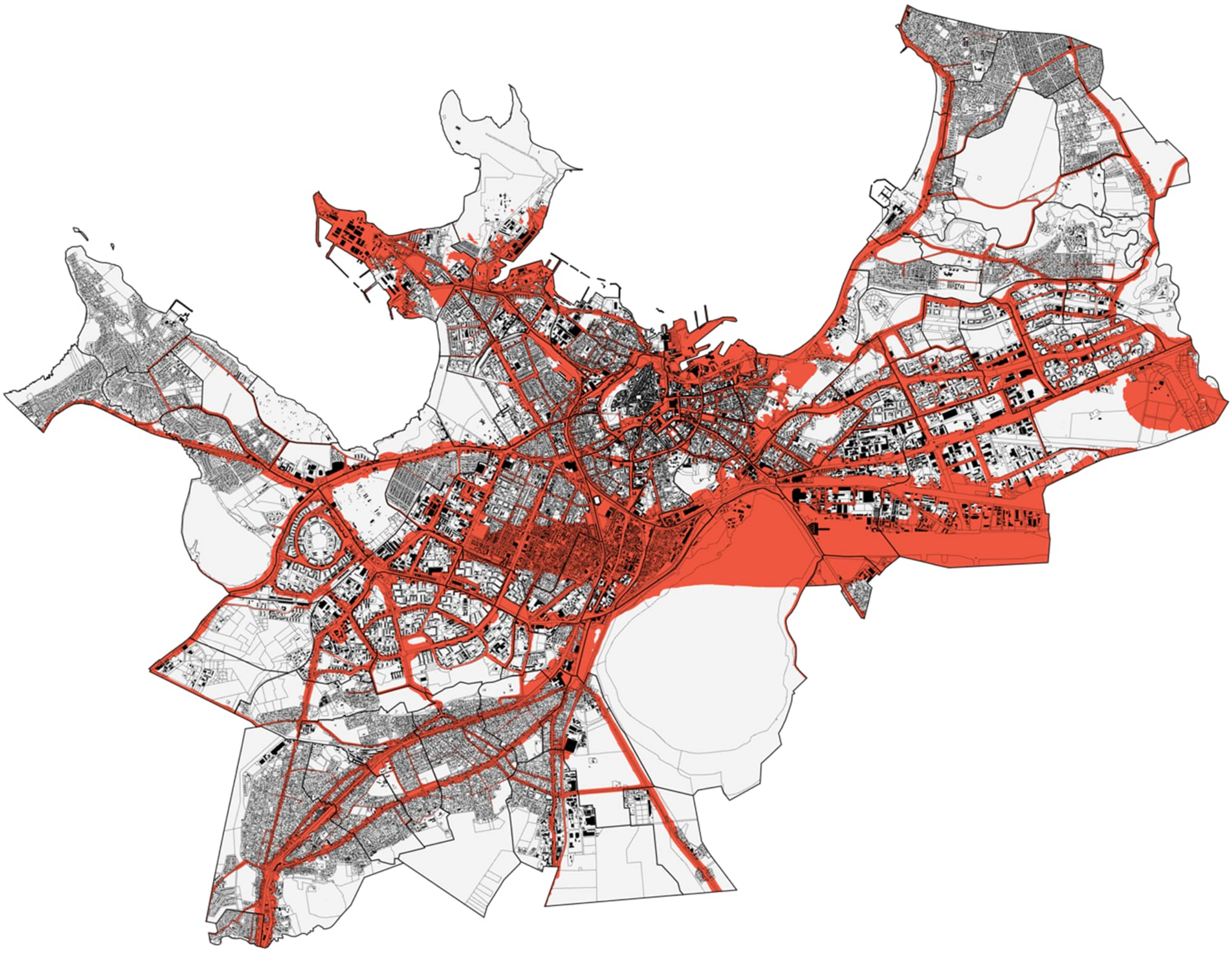

Figure 1: > 55 dBA noise zone in Tallinn. [4]

Based on the information from “Tallinn’s environmental noise reduction plans for years 2019-2023” the amount of people who are living in $L_{den} \geq$ 55dB noise zones is over 56,8% in the year of 2019. [5] In Tallinn the physical dangers are monitored by Terviseamet, who also provides reports of noise complaints. Comparing the traffic noise complaints between years from 2014 to 2021, it can be seen, that the amount of traffic noise complaints has risen by 14,6%. [6]

Taking into account the noise situation in Tallinn, the location for the concert hall can be chosen. The best area for it is between Linnahall and the Port of Tallinn.

7 Site analysis

Linnahall is an important place historically. It was one of the first structures, which “broke” its way to the sealine and allowed the people in Tallinn to reach the sea. Within the building lies a concert hall of which back walls can be rised into the ceiling allowing access to the room.

From the north the project area is cornered by the Baltic Sea, in the east lies the Port of Tallinn, in the south there is the North Boulevard, which connects the port to the old town and in the west lies the Linnahall building. The site is perfect for a concert house as it is familiar to the residents of Tallinn and the seaside area is being developed, which in turn connects together the residential area of Noblessner, Tallinn Seaplane Harbour, Patarei sea fortress, Linnahall, cruise terminal, Reidi road and Pirita beach.

A new tramline is being built for the area, which connects the city centre with the Port of Tallinn and Linnahall. As the area is empty there is not a lot of traffic noise to deal with, with the exception of the port. The empty space allows to form a new structure without heavily disturbing the current situation.

In terms of noise there are two main sides, which need to be addressed – traffic & ship noise coming from the Port of Tallinn and traffic noise coming from the North Boulevard.

7.1 Analysis of existing and proposed masterplans for the area

For this analysis I have chosen two projects from architectural competitions and composed a plan of the cities detail plannings for the area. I have created 3D acoustic models of the area, where I calculated traffic noise levels.

In all of the calculations it is estimated that hourly there are about 100-200 cars on each road, which is equivalent to filling a parking lot used to accommodate the buildings. The road connecting the port and the city will have about 950 cars, which is equivalent to twice the average capacity of a ship stopping at the Port of Tallinn. The height of the buildings is set to 18 meters.

From the analysis it can clearly be seen that in the planning phase environmental noise was not thought of.

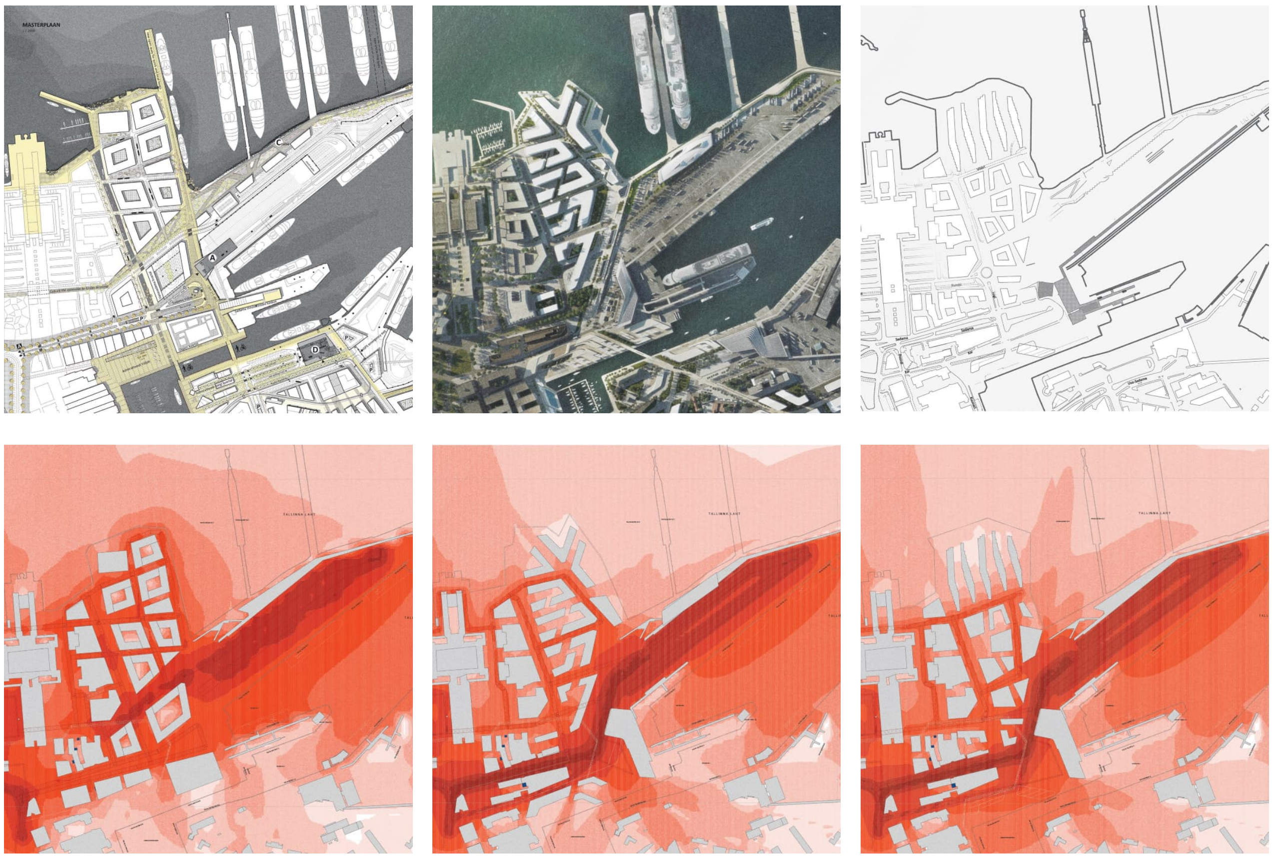

Figure 2. From left to right. Noise analysis of KavaKava [7] and Zaha Hadid [8] masterplans, detail plannings of Tallinn [9]

8 Project

First a noise map of the area is created to figure out the exact sources of noise and the amount of it. The next step aims to create architectural structure within the area, making it logical. Using CadnaA 2023 and data from Stratum OÜ and Akukon Eesti OÜ the architectural plan can be made. Going step by step and analyzing the effect of each added building a plan is formed. As the next step greenery is added to the area to reduce psychological effects of noise and minimize the effect of traffic and ship noise. The last step is to add sound absorbing construction to entry points of closed blocks to minimize reflected sound within the courts.

9 Masterplan

The masterplan focuses on planning an area of a city based on noise coming from traffic and ships. The plan connects the currently empty seaside area with the rest of Tallinn, it connects the culture kilometer with the Cruise terminal and Linnahall with the new concert hall. Within the area, the main focus is on pedestrians as cars can not enter the area, with the exception of police, ambulance etc.

In the quieter parts residential buildings are planned with the ground floor being dedicated to businesses to keep the area active throughout the day and night. In the middle of the area forms a park, which adds another sea connection to the area. The concert house is planned in a way, which allows the west side of the building be used as a stage and the park as an audience.

The main difference from a regular masterplan is seen through noise calculations. Based on the results of the calculations the noise levels in the cultural center are > 40 dBA. This means that trying to measure them physically would bring complications as the background noise from trees etc is higher than noise from traffic or ships.

10 Sound in noise

The planned park offers a changing soundscape, which changes throughout the year depending on weather conditions and seasons. In the early morning it is possible to hear ships leaving the port, when low frequency noise from ships joins with noise from waves morphing into warm sounding waves.

During lunch time it is possible to hear socializing of people, which is supported by songbirds and the rustling of leaves, making the park feel calm. During night time, when the wind rises, rustling of leaves becomes dominant offering the sensation of being in a forest. During rainy periods the sound of water and rustling of leaves race to be heard.

For winter time low water level installations are planned, which allow crackling ice to form giving children and adults alike a large playground to explore.

Throughout the year, the park can be influenced by music coming from the concert house creating a multitude of listening positions, where the audience can choose the balance between music, nature and sounds of the city.

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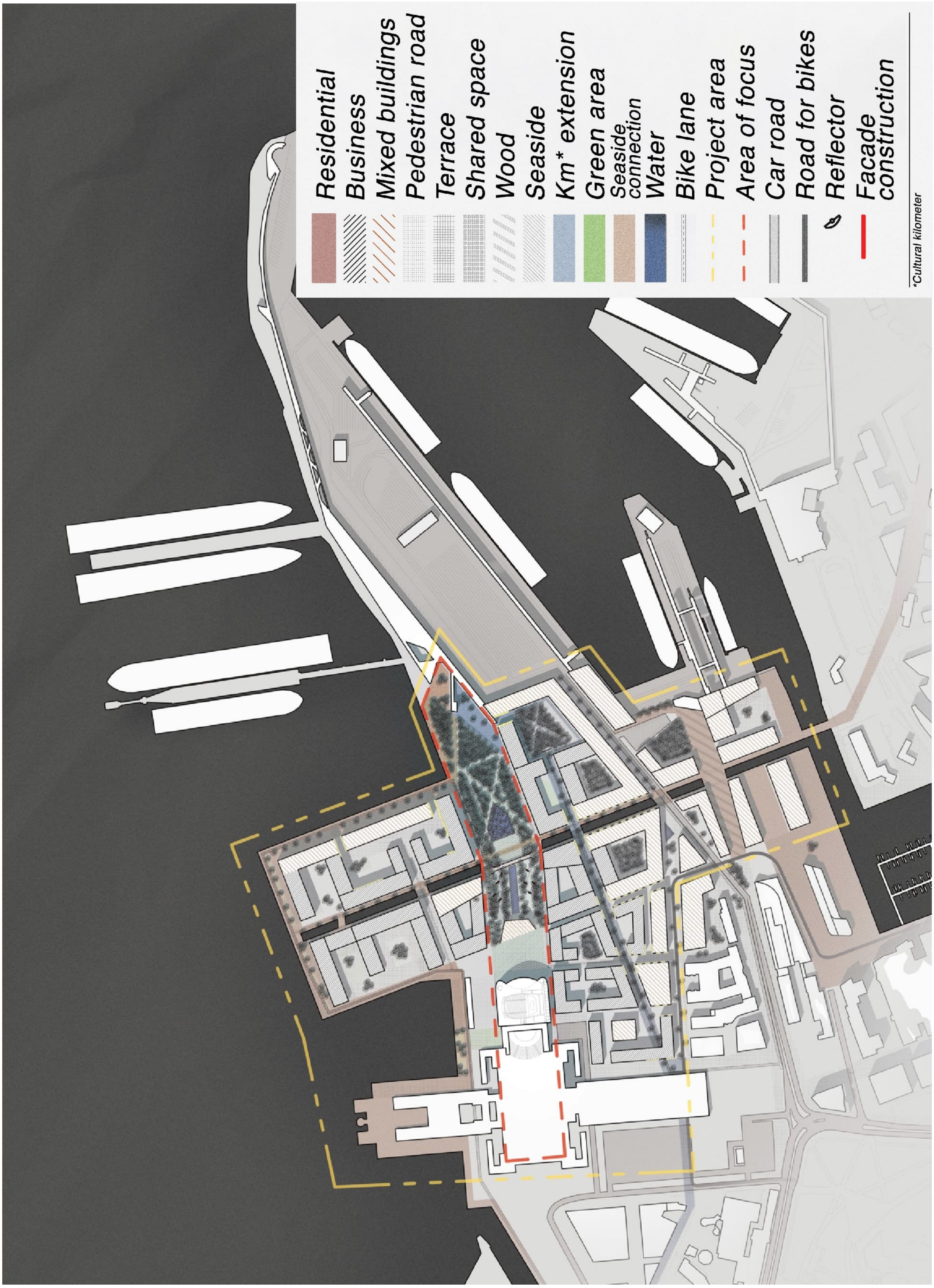

Figure 3. Masterplan

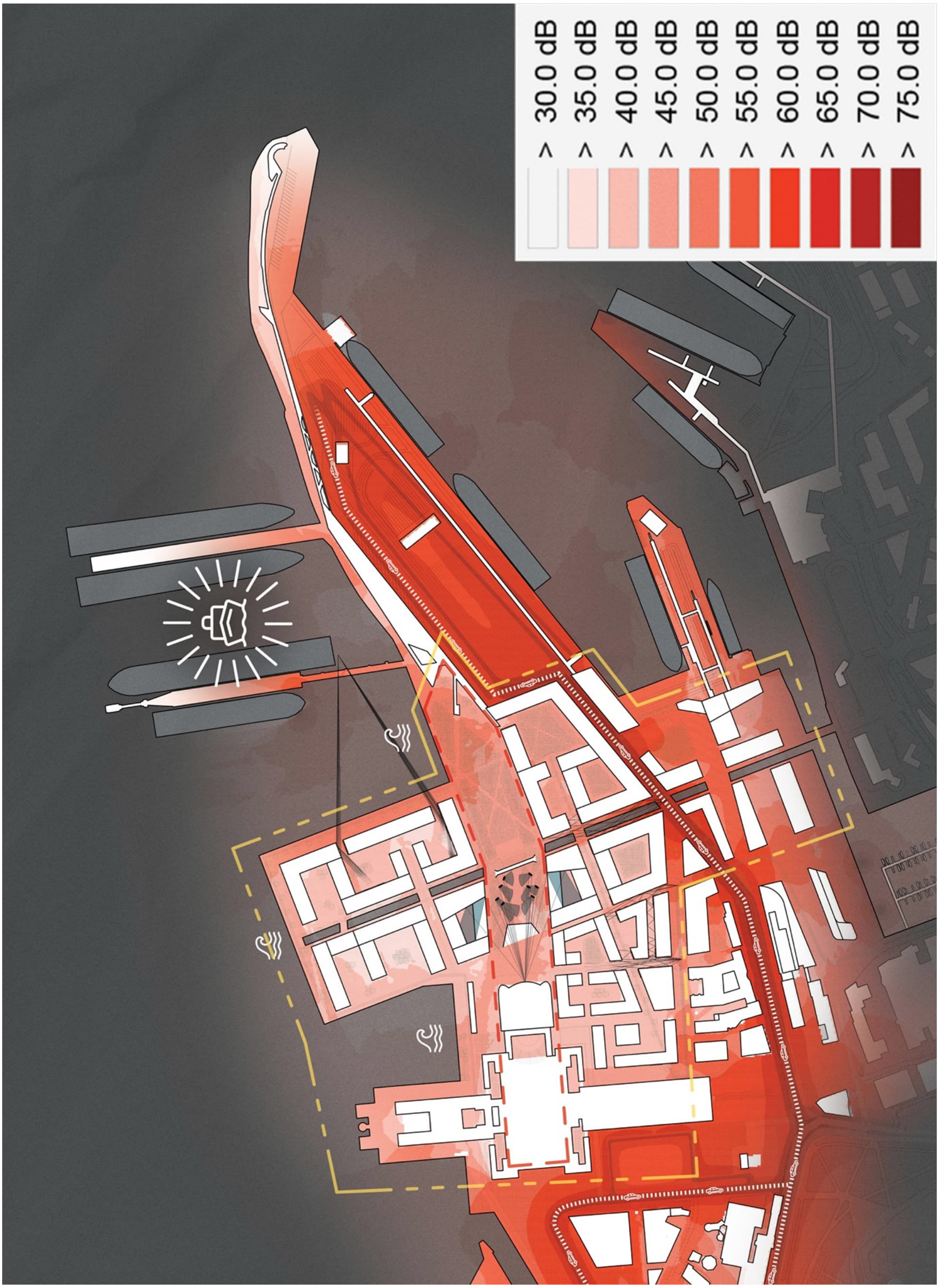

Figure 4. Masterplan noise map

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Aircraft noise modelling with AEDT

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Aircraft noise is a major environmental concern around many expanding airports around the world. With increasing global travelling trends for both business and leisure, we see airports' capacity demands increasing each year. Communities around the airports, however, do not always welcome the airports' expansions, because for those who do not feel the benefits it simply means a noisier living environment. Continuing research and development on the quieter engines and improving aerodynamics are reducing aircraft noise emission, and increasing capacity brings social-economic benefits to the surrounding communities. Using sophisticated aircraft noise modelling software, acoustic consultants can predict how noise levels will change in the future and can play an important role to help make aircraft operations quieter by carefully studying the way aircraft operate in the sky (their trajectories, speed, altitude, approach angles etc) and suggesting mitigation measures to reduce resulting noise footprints in populated areas on the ground. This paper discusses the ins and outs of one of such software – the Aviation Environmental Design Tool (AEDT) produced by the United States of America Federal Aviation Administration. We look into the software's capabilities, the strengths, the limitations, and show that properly utilized and calibrated noise models can realistically replicate real-world aircraft operations and enable consultants to forecast aircraft noise levels in affected communities and work with airport operators to develop operating procedure to help mitigate noise impacts.

1 Introduction

The Aviation Environmental Design Tool (AEDT) is an FAA and NEPA approved 3D aircraft modelling software that allows user to estimate fuel consumptions, emissions, noise, and air quality [1]. AEDT is designed to model varying scale studies, ranging from a single aircraft operation to a whole fleet of the airport's yearly operations at the airports all around the world. It is a highly sophisticated software developed and supported by the FAA over many years and used on licence to many thousands of uses in the US and around the world.

1.1 Library

AEDT built-in library offers over 33000 airports, over 3000 aircraft and helicopters and over 400 ground equipment that are essential for airport operation for emission and noise modelling. The software also allows for bespoke airport design, that can play a major role in planning stage of a newly built or expanding airports. Unlike most environmental noise models, AEDT not only predicts noise propagation but also simulates aircraft take-off and landing profiles and tracks dependent on local weather.

However, although over 3000 individual aircrafts (airframe and engine combinations) are available in the library, only 300 individual noise footprints are available, that are grouped under ANP IDs. These IDs describe aircraft's operations (approach, departure, circuit, touch and go) steps, flap and thrust settings, and noise emission data.

1.2 Basic settings

As with the majority of the noise modelling software, AEDT allows its user to enter a number of variable parameters that encompasses the whole study, for example: weather parameters (temperature, pressure, humidity and wind speed) and ground terrain model that all allow for more accurate noise propagation calculations. The user is also able to choose from 33 pre-defined noise and emissions prediction metrics or is allowed to define custom metrics.

2 Operational properties

Each aircraft operation within AEDT is defined by a number of properties that can vary with operational scenarios, airframe and engine models. Aircraft routes are defined by a series of coordinates that follow the horizontal vector of the aircraft's flying path as shown in Figure 1 A). These single vector paths can be dispersed in an attempt simulate aircraft real-world deviation from the centre route line as Figure 1 A) shows multiple lines dispersing from middle line.

The flight path is then assigned with speed, altitude and thrust profiles shown in Figure 1. B), C) and D) that consist of a series of aircraft's procedure steps defined by flap, thrust, altitude, speed and climb rate settings (changes between procedure steps are marked on the profiles with black arrows) that are individual for each aircraft. Figure 1. B), C) and D) show example default profiles of the Airbus A320NEO.

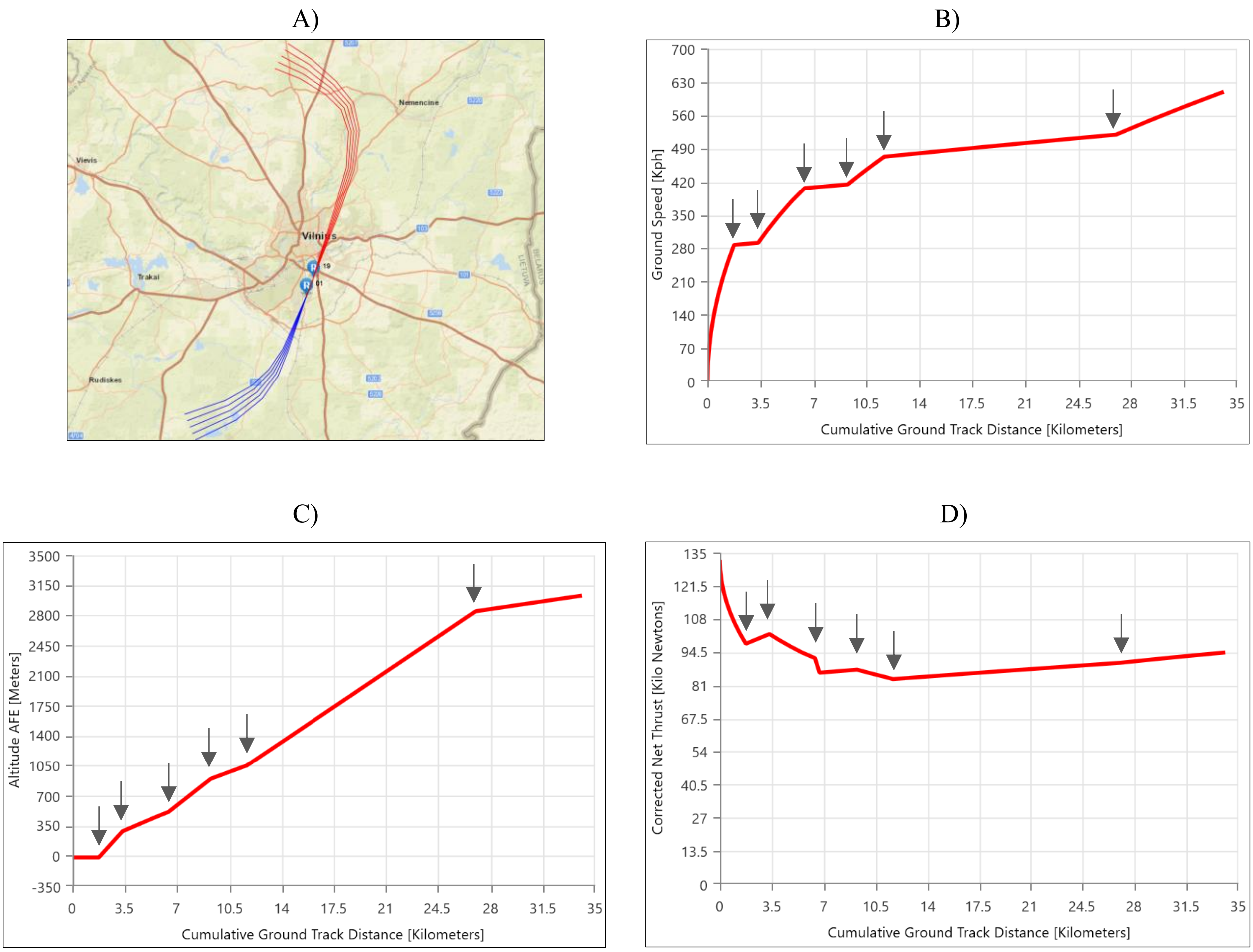

Figure 1: A) Dispersed arrival (red) and departure (blue) tracks; B) Ground speed profile; C) Altitude profile; D) Thrust profile

2.1 Profile settings

Once the user understands aircraft operating procedures principles in AEDT, more advanced modelling can start to take place. Aircraft profiles editing allow the user to manually change the procedure steps aircraft undertake during landing or take-off scenarios. For example, if comparison between measured real-world aircraft altitude profile and default one from AEDT show that aircraft in the software is gaining altitude too quickly – a number of settings can be altered to extend the horizontal distance at which aircraft gains altitude, for example:

- Aircraft's stage length (weight) can be increased; or
- Aircraft's thrust power at take-off can be reduced; or
- Aircraft's flap settings can be altered to give slower altitude gain, but quicker horizontal acceleration; and more.

As a result, any of the settings manipulation can majorly influence the noise footprint aircraft leave on the ground and thus allows the user to calibrate the model against the real-world measured data with the very high degree of accuracy.

The editing of the procedural steps however is heavily limited by certain rules, that must be strictly followed at all times when adjusting the procedural settings to ensure the procedure followed is realistic. For example, one cannot expect to force modelling the aircraft flying upside down!

2.2 NPD settings

The further advanced settings in AEDT allows the user to interact with is the noise-power-distance (NPD) curves. These are the very fundamental information that AEDT contains on aircraft's noise emission data. The NPD (decibel values at the distances) are derived from aircraft fly-over measurements taken for a range of aircraft configurations and engine power settings [2]. Along with the flight profiles, this data is key when calculating noise exposure contour maps. Similarly as with the profiles editing mentioned above, AEDT allows its users to adjust the NPD curves (e.g. L_{AMAX} and SEL) which together can result in both accurately calibrated aircraft flying procedure and closely matched noise footprint data. The example of default NPD L_{AMAX} and SEL curves for A320NEO are listed in the Table 1.

Table 1: Arrival NPD Curve of Airbus A320NEO

Distance, feet	Arrival NPD parameters, dB	
	L_{AMAX}	SEL
200	90.5	92.5
400	83.4	88.4
630	78.8	85.6
1000	73.8	82.8
2000	66.1	76.9
4000	60.0	74.0
6300	42.4	69.1
10000	46.3	63.8
16000	39.1	58.3
25000	32.2	53.1

3 Example case study

The example case study was set up at Vilnius Airport to show the noise contour differences between the regular and calibrated aircraft simulation models. Two simulations were run with a single Airbus 320 NEO aircraft operation. First the operation has default AEDT settings, and the second one has height/altitude profiles calibrated and NPD L_{AMAX} curve adjusted to closely match the real-world measured noise contour data measured by the UK Civil Aviation Authority for Gatwick Airport Limited.

Figure 2. A) and B) show comparison between the AEDT default Airbus 320 NEO operation profile in black and the edited one in red that is closely matching to measured data. Figure 2. C) shows the L_{AMAX} 65dB contours for both

operations. It can be seen, that with this particular aircraft and engine model AEDT default data simulates substantially greater L_{AMAX} 65dB contour, compared with the contour calibrate my measurement.

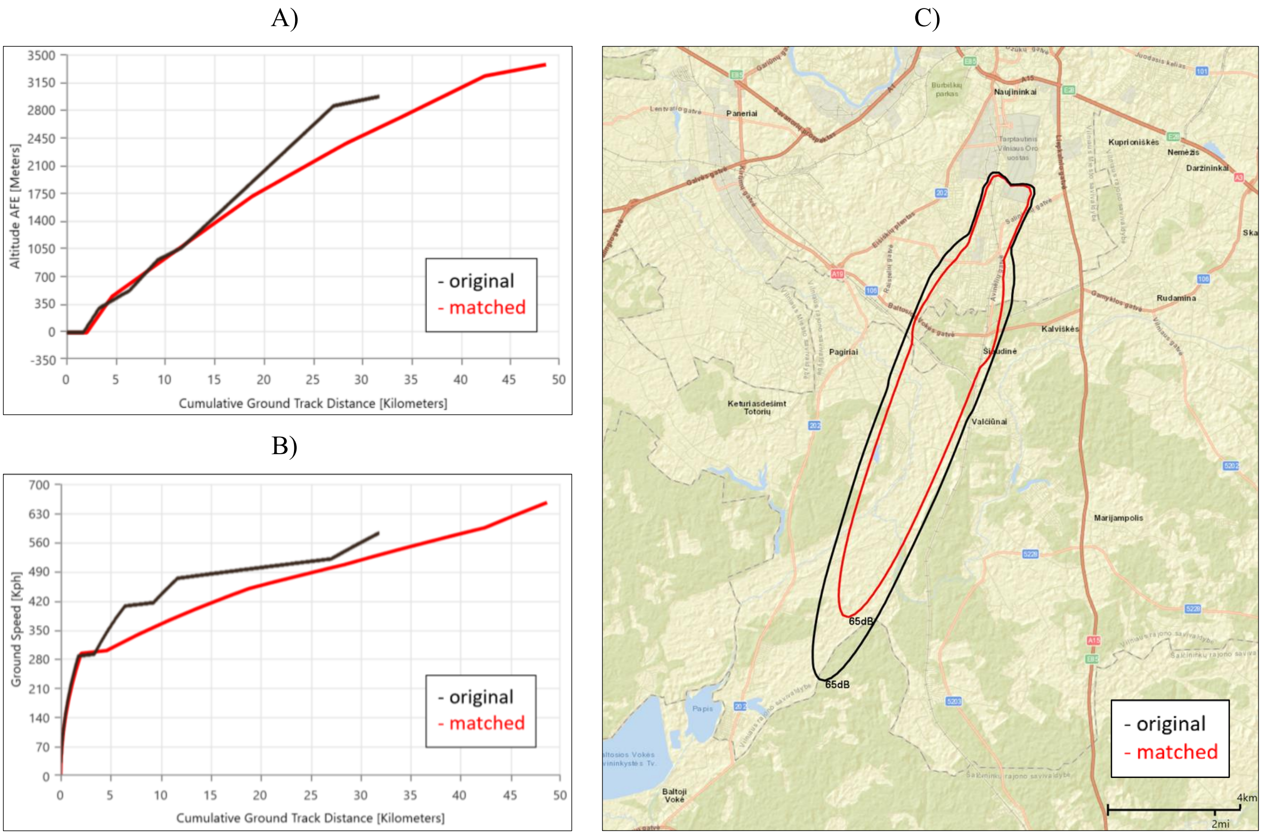

Figure 2: A) Original and edited altitude profile; B) Original and edited speed profile; C) L_{AMAX} 65dB noise contours for original (black) and calibrated (red) single EA320NEO operation

4 The benefits of calibrated AEDT modelling

This paper presented the basic and advanced technical capabilities of AEDT modelling in one area. Configuration and calibration of the aircraft profiles together with the NPD adjustments provide very accurate noise contours predictions, that can majorly help in the estimation of the true noise changes around the airports with both increasing airports capacity demands and modernising aircrafts. The software can also simulate relocation of flight tracks to displace and disperse noise to help avoid communities.

Using AEDT software and correct methodologies provides airports around the world with a substantial amount of information to understand the environmental noise change expected in the surrounding communities that airport expansions bring. This then allows for better and clearer communication between the airports and local authorities and to foresee and mitigate the change of the environmental noise.

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Magneto-acoustic triangulation method for electric discharge localization in the atmosphere

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The Auroral Acoustics project enters its 25th year exploring further the hypothesis linking electric discharges to auroral sounds. Over this period, instrumentation has undergone significant advancements, evolving from a 16-bit stereo DAT to a six-channel 32-bit float recorder. A pivotal breakthrough came with the development of magneto-acoustic triangulation, revolutionizing sound source localization and leading to the discovery of a previously unidentified mechanism in the lower atmosphere producing these sound phenomena. This methodological innovation laid the groundwork for the inception of the inversion layer hypothesis, offering a compelling explanation for both auroral and meteor sounds. This paper elucidates various formulations of the magneto-acoustic triangulation method employed throughout the project, exemplified by recent six-channel sound and magnetic field recording which provides a new explanation for so called frost crackling sounds.

1 Introduction

In ancient times, the enigmatic spectacle of the northern lights often intertwined with other natural phenomena such as comets and distant lightning, blending luminous displays with rumbling sound effects. Reports of similar sounds occasionally emerge alongside active auroras. Over a millennium later, descriptions of auroras likened them to slow, dry lightning or celestial conflagrations. Crackling and humming noises, occasionally audible on the ground, were readily associated with Aristotelian beliefs portraying auroras as heated material ascending from the earth to ignite in the sky. Accounts abound with sightings of angels, fiery dragons, and celestial armadas, evoking vivid imagery in the minds of observers. Sixteenth-century Scottish historian Hector Boece vividly described auroras as resembling "a burning fire, as if knights in armour and on foot fought with great force," while reports of heavenly hosts' noises and battles echoed on the ground.

The groundbreaking experiments of Benjamin Franklin, revealing the presence of electricity in the atmosphere, challenged prevailing theories linking auroras solely to sunlight reflections. Auroral crackling sounds emerged as evidence supporting the notion of an electric origin, fueling speculation that these celestial phenomena could indeed be manifestations of atmospheric electrical discharges.

The hypothesis attributing auroral sounds (AS) to electric discharges near observers dates back almost as far as the understanding of atmospheric electricity itself. Some of the earliest texts proposing this mechanism likely originate from the years 1831 and 1834 [1,2]. Notably, Professor William C. Baker of Queen's University, Kingston, recounted an AS event occurring around 1884–1885 in a letter addressed to C. A. Chant, editor of The Journal of the Royal Astronomical Society of Canada [3]. Describing the sounds as akin to "the crumpling of stiff paper," Baker suggested they were produced by the same electromagnetic (EM) source controlling the visible aurora. He drew parallels to the familiar noise of discharging a Layden jar, a type of capacitor. Baker observed that the sound source must be closer to the observer than the light source, based on the short delay between auroral movements and the accompanying sounds. Chant collected a large amount of observation reports and summarized that a *brush discharge* in the vicinity of the observer could explain these sounds [3].

In 1933, C. S. Beals proposed an intriguing hypothesis in Nature [4], suggesting that certain atmospheric conditions may lead to electrostatic discharges near the Earth's surface. He speculated that these discharges, if real, might be secondary phenomena induced by processes occurring at higher altitudes, which are responsible for the more conspicuous aspects of auroras studied by Størmer and his colleagues.

The contemporary focus remains on elucidating the mechanisms underlying the charging and discharging of capacitor-like structures, as addressed by the recent Temperature Inversion Layer (TIL) hypothesis [5,6]. This hypothesis posits that during calm, clear nights, warm air rises in the atmosphere, carrying negative ions from the ground. As this rising layer encounters colder air, it forms a stable temperature inversion layer, inhibiting vertical ion movements. Meanwhile, positive ions continuously precipitate from the ionosphere, charging the plate capacitor-like structure located within one hundred meters of the ground. Energy for the sounds accumulates in this layer until geomagnetic disturbances induce additional potentials, triggering the discharging process with audible sounds.

The presence of a "Schumann fingerprint" in the crackling sounds measured provides compelling evidence of the connection between these sounds and geomagnetic activity [7]. Additionally, magnetic field measurements conducted by the Finnish Meteorological Institute (FMI) corroborate the causal relationship between geomagnetism and the recorded auroral sounds [6].

Electric discharges occurring in the atmosphere manifest in various forms, generating not only sounds but also EM radiation, ultraviolet light, and ozone. These discharges induce electric currents that, in turn, produce magnetic field pulses (M-pulses). Recognizing the need for a method to localize and study the physical properties of these magneto-acoustic sources in the atmosphere, the Magneto-Acoustic Triangulation method (MAT) was developed.

The assumption that purely acoustic methods alone might not suffice for solving this problem was validated. While ground-based observations allow for relatively accurate estimation of the direction of sound sources in the sky, determining their altitude, particularly close to 100 meters, presents a more formidable challenge. MAT addresses this challenge by combining two key ideas: utilizing the magnetic field component to *estimate the distance* of the magneto-acoustic source in the atmosphere and providing evidence that the sounds are indeed caused by *electric discharges*.

This paper elucidates the methodological advancements made throughout the extensive Auroral Acoustics project, with a focus on the recent MAT method and its novel findings. Notably, the shape of the magnetic field pulse associated with the sound is of particular interest. Horizontal magnetic field components (Mx and My) offer a more accurate representation of the discharge pulse's waveform compared to the previous reliance on the Mx component alone. The resulting M-pulse exhibits a triangular or pyramid-like shape with a duration of approximately 30 milliseconds, aligning well with the envelopes of the produced sounds.

2 First experimentations

The Auroral Acoustics project commenced in the spring of 2000. At that time, the most advanced portable digital audio system available was a stereo DAT-recorder with 16-bit resolution. One channel of this recorder was dedicated to capturing data from a Brüel & Kjær measuring microphone positioned at the focal point of a parabolic reflector, while the other channel recorded signals directly from a 5.2-meter-long vertical VLF antenna [8]. This rudimentary setup, dubbed the "VLF recorder," was compared to a commercial device and demonstrated unexpectedly robust performance. Despite this promising start, data collected between 2000 and 2005 using the VLF antenna await thorough analysis. The initial enthusiasm waned as it became apparent that even if a correlation between the sounds and VLF measurements were established, it would offer limited insight into the underlying sound production mechanism or the precise location of the sound source. However, recent comparative studies involving the VLF antenna and a loop antenna have yielded similar results, also predicted by the basic electromagnetic theory, motivating for revisiting and reevaluating this early dataset.

The initial phase of the Auroral Acoustics project concluded in 2005 amidst several setbacks, including the rejection of multiple grant applications. It became increasingly evident that the methodologies employed were insufficiently sophisticated to address the complexities of the problem at hand. Nevertheless, a renewed phase of activity gradually took shape in 2011. The advent of new four-channel digital recorders with 24-bit resolution, coupled with the maturation of a novel approach for integrating audio recorders into VLF measurements, spurred this revival. A specialized type of VLF loop antenna was developed, featuring a symmetric output mirroring typical microphone signals. This innovation enabled direct connection of the loop antenna to a low-noise microphone preamplifier, marking the inception of the first iteration of the MAT method, which incorporated three microphones and one VLF loop antenna. This method yielded initial estimates for the location of the sound source [9], leading to the discovery of a previously unknown sound-producing mechanism in the lower atmosphere. The revelation of such a sound source beneath a clear sky in the lower atmosphere was met with surprise and it took the time to explain this novel observation.

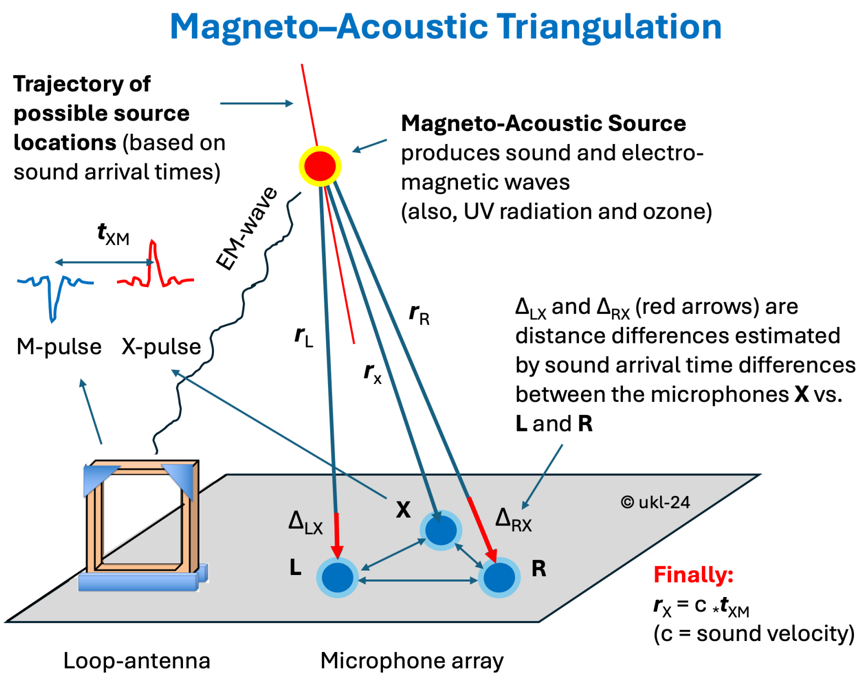

Figure 1. The MAT method realized by three microphones and a VLF loop antenna.

3 MAT method

The instrumentation utilized for sound source localization with the MAT method between 2012 and 2022 is illustrated in Figure 1. The X-microphone employed is the Brüel & Kjær 4179 with preamp 2660, while the L and R microphones are Sennheiser MHK8020 models. The Mx component is recorded by the VLF loop antenna.

The MAT method involves several phases, each necessitating digital signal processing:

1. *Estimation of Arrival Time Differences*: The arrival time differences of the audio pulses, represented as $t_L - t_X$ and $t_R - t_X$, are determined from the microphone signals. Utilizing the speed of sound at the relevant temperature, the distances Δ_{LX} and Δ_{RX} can be calculated.
2. *Geometric Solution for Source Location*: With the distances between the microphones known, a set of potential source location points can be geometrically derived. These points form a trajectory along which the source is expected to be situated. At this stage, the direction of the source is estimated, though its distance from X remains unknown.

3. *Determination of Source Distance*: The final step involves calculating the distance between the source and the X microphone ($\boldsymbol{r}_X$). The arrival time difference between the M and X signals provides an estimate of the sound pulse transit time. Subsequently, $\boldsymbol{r}_X$ is determined using the equation $\boldsymbol{r}_X = c\,\boldsymbol{t}_{XM}$, where c represents the speed of sound and $\boldsymbol{t}_{XM}$ signifies the arrival time difference between the sound at microphone X and the magnetic pulse M.

Recent tests employing four microphones on the ground have revealed that this configuration, relying solely on acoustical triangulation, fails to provide accurate estimates for the altitude of the sound source. The limitation arises from the absence of a microphone positioned high in the atmosphere—a logistical challenge yet to be overcome. Intuitively addressing this issue, the MAT method replaces one microphone with the M-field sensor which performs like a *virtual microphone* locating right next to the sound source. This strategic substitution enables accurate estimation of sound source distances, contingent upon the successful association of magnetic field pulses with their acoustic counterparts.

While conceptually straightforward, the practical implementation of the MAT method demands sophisticated signal processing due to frequently low signal-to-noise ratios of both the sounds and M-pulses. Arrival time differences between pulses at X and L, as well as X and R, are resolved through cross-correlations. A clean signal segment from X (the B&K model) facilitates the accurate identification of similar segments in the noisy L and R signals (microphones with higher internal noise). It is therefore essential that even one of the microphones is of the highest quality.

When localizing sound sources such as clapping, popping, or crackling, it's possible to attenuate environmental noises and improve the signal-to-noise ratio by focusing on the energy concentrated in the 0.7-2.5 kHz frequency band. Occasionally, the calculated time difference between acoustic pulses exceeds the time it would take for sound to travel between microphones, indicating erroneous data that must be discarded. Associating sound pulses with M-pulses benefits from the polarity of the M-pulse, which reveals the location of the magneto-acoustic source in a half-space. When multiple M-pulse candidates precede a sound, this polarity information aids in selecting the correct candidate. Sound sources are typically localized within a single layer situated in the temperature inversion structure, a fact that can inform the association of M-pulses with sound pulses. Furthermore, comparing the RMS values of M-pulses with those of sound pulses provides a means to assess result quality. Under optimal conditions, this correlation is typically high, serving as additional evidence supporting the *causal connection* between sounds and electric discharges [5].

The recent iteration of the MAT method utilizes four microphone signals and two magnetic field signals, all recorded at a sampling rate of 48 kHz and in 32-bit float mode. The subsequent chapter presents an example of sound source localization using the MAT method.

4 Example of recording at Fiskars village Jan 16, 2024 – Frost crackling?

The ambient temperature at Fiskars Village (Raasepori, Finland) on January 16, 2024, ranged from -17 °C in the afternoon to as low as -25 °C around local midnight, rising to -15 °C at 5 AM, and then returning to -17 °C by 9 AM. Throughout the night, winds remained calm at 2 m/s, with air pressure stabilizing at 1000 hPa and humidity reaching 80%. Despite the absence of significant geomagnetic activity, with the Kp index fluctuating around 1–2 and the Kyoto Dst index ranging from -5 to 4, ideal weather conditions prevailed for the formation of a robust inversion layer.

A decision was made to capture the tranquility of this night without active auroras. However, contrary to expectations, the night was far from silent. Over the course of the ten-hour recording, commencing at 8:23 PM local time, nearly three hundred distinct sound events were identified. These events exhibited an average sound pressure level (SPL) of 57 dB (measured over a 35 ms window), with the most intense reaching over 70 dB. The abundance of sound phenomena surprised us, prompting speculation that they might be attributed to "frost crackling" sounds generated by snow, ice, trees, buildings, or other mechanical structures nearby.

While frost crackling sounds have often been assumed to originate from natural environmental elements, such as trees or ice formations, our findings cast doubt on this explanation. Forestry specialists assert that frost cracks in trees are rare occurrences, suggesting that the sheer volume of observed sound events cannot be solely attributed to this phenomenon [11,12]. Consequently, a deeper investigation into the recordings was warranted.

Utilizing four microphones and two magnetic field sensors, we initially attempted to determine the locations of the sound sources through purely acoustic triangulation. However, this approach proved challenging. Subsequently, the MAT method was adapted and applied, enabling the localization of sound sources at nine different altitudes, ranging from 40 m to 120 m, with 10 m increments. To identify sound candidates, the ten largest M-pulses were extracted from the envelope (Env) of the M-field signals using the rule: Env=Abs[Mx+I*My], leveraging the orthogonal components of the M-field, Mx and My.

Interestingly, all obtained pulseforms exhibited an *isosceles triangle* shape, resembling a pyramid, with a pulse length close to 30 ms (Figure 3). In contrast to previous Mx based studies where M-pulse shapes varied widely, the utilization of both Mx and My signals provided a consistent and reliable method for defining the true M-pulse shape. The obtained M-pulse shape is consistent for all sound events which significantly enhances the localization accuracy. Furthermore, the correlation between M-pulse and sound envelope shapes was notable, with correlation coefficients (r) exceeding 0.95 in many instances. Only those M-pulses are accepted for the final phase which are coming from the direction of the associated sound source. Finally, the source *distanc*e is trimmed based on the associated M-pulse time difference (t_{MX}) keeping the *direction* unchanged.

Automatic clustering of the sound events found active layers at different altitudes revealing a stratified structure within the lower atmosphere. Particularly intriguing was the similarity between the sharp clapping sounds originating from the 70 m layer and the auroral sounds recorded in previous studies at the same location and almost from the same altitude. Approximately two hundred localized sound sources, reliably linked to corresponding M-pulses, were identified, shedding light on the nature of sound phenomena in the lower atmosphere. Figure 2 illustrates the distribution of these sound sources across altitude layers.

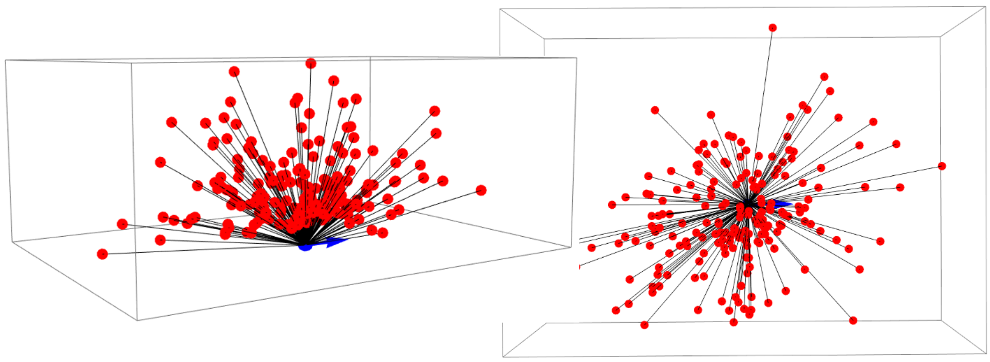

Figure 2. With the help of associated M-pulses localized sound sources. A hundred meter long blue arrow points north. The microphone array is in the middle. Left: side view. Right: Top view.

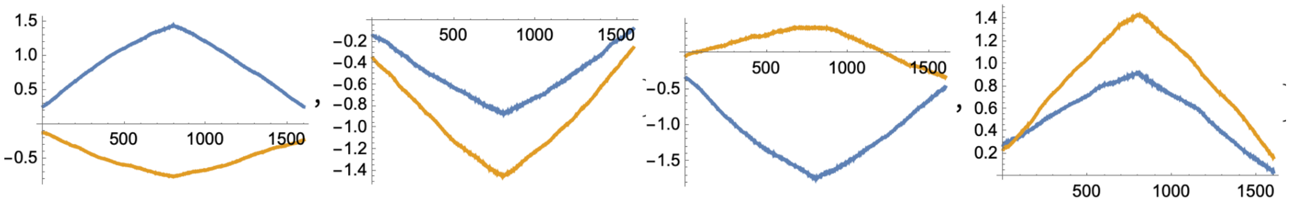

Figure 3. Example of cluster means of M-pulses. Over a hundred and fifty pulse pairs {Mx–blue, My–brown) were clusterd. The number of members in these classes varies between 20–40. The corrsponding average source location is in NW, SW, SE, and NE when interpreting the curves from left to right.

5 Sumary and discussion

The MAT method has undergone evolution across various phases of the Auroral Acoustics project. It addresses the challenge of estimating source altitude by providing a virtual microphone positioned proximately to the source. Additionally, it offers compelling evidence supporting electric discharge as the mechanism behind sound production. Given that all other potential sound sources in the environment are purely acoustic and not magneto-acoustical, this method serves as an excellent tool to differentiate them from those originating in the TIL.

The recorded horizontal components (Mx, My) of the magnetic field have provided a dependable solution for the shape of the M-pulse, resembling an *isosceles triangle*. In earlier studies, this shape was merely one among many forms obtained through the coherent sum method with varying criteria. Understanding the true form of the M-pulse may facilitate a more detailed model of the inversion layer discharge process in future research.

The calm and clear night of January 16, 2024, presented an exceptional opportunity to study processes within the TIL. The SPL of the recorded sounds varied around 60 dB and tens of the sound sources had SPL around 90–100 dB. Presently, it remains unclear whether the sounds are linked to geomagnetism, despite minimal variations in these parameters. Alternatively, the TIL may exhibit spontaneous charging and discharging without additional geomagnetic support under favorable weather conditions. Future studies aim to elucidate this aspect.

Previously, frost crackling was often used to dismiss auroral sounds, attributing them to natural elements such as forests, ice, or man-made structures. However, our study reveals that the majority of these crackling sounds originate directly from the clear, open sky above. While it's straightforward to associate sounds with auroras when they synchronize with visible auroral movements, our findings indicate that both auroral and frost crackling sounds are similar and created by electrical discharges in the TIL.

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Finnish Acoustician Paavo Arni (1905–1969)

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Mr Paavo Arni (1905–1969) was a central influencer in the field of acoustics in Finland since the 1940's until his death. Paavo Arni ran an engineering office specialized in acoustics, but he was also active in education, founding of the Acoustical Society of Finland, preparing regulation concerning sound insulation of apartment buildings as well as occupational noise. He also supported research on acoustics in Finland and internationally, too. In addition, he was the author of the first Finnish handbook on acoustics published in 1949. The base of the wide range of activities of Paavo Arni was possible because of his main work at the Finnish Broadcasting Company YLE. He began as a studio manager in 1931 and ended his career as a technical director of YLE. Construction of studios provoked a need for knowledge on acoustics. Paavo Arni started to form international connections with other broadcasting companies and with acousticians in the 1930s by making visits, attending conferences and hosting international experts' visits in Finland.

1 Introduction

Paavo Arni (1905–1969) graduated in 1930 as a Master of Science in Technology from the Department of Mechanical Engineering of the Helsinki University of Technology, but he only worked for a short time in the mechanical engineering industry. During the Great Depression, Arni could not find work in his own field, and he ended up working for the Finnish Broadcasting Company as a studio technician in 1931. The engineers of the Finnish Broadcasting Company had to familiarise themselves with acoustics when designing the studios. Knowledge of acoustics was thus focused in the Finnish Broadcasting Company, where an acoustics laboratory was built in the basement of the Radio House (1934) to measure the acoustic properties of structures and materials. [1]

Paavo Arni has been featured in research literature from the point of view of the Finnish Broadcasting Company, in which he held several significant positions. For example, in 1952, he was responsible for the radio broadcasting of the Helsinki Olympic Games around the world (Fig. 1) and, at the end of the decade, for the launch of the Finnish Broadcasting Company's television operations [2]. During his career at the company, he progressed to become Chief Engineer in 1952 and then Technical Director in 1964 [1]. Arni's work as an acoustics expert, on the other hand, is only briefly mentioned in research literature. His activities have been described a little more extensively in one area of acoustics: in research on the technology transfer that led to the creation of sound insulation regulations for residential buildings. Paavo Arni played a key role in the technology transfer that took place in the absence of domestic research [3]. Paavo Arni's work as an acoustics expert took place in an era when the advancement of industrialisation and urbanisation in Finland made noise caused by machines, traffic and housing a social problem that had to be solved in some way. At the same time, there was also a need for acoustic expertise because theatres and orchestras that had previously performed at town halls, clubhouses, community centres, workers' halls or community halls began to be municipalised, and performance facilities were built for them. In the 1940s, people also started paying attention to the acoustics of schools. [1]

The purpose of this article is to present Paavo Arni's work as an acoustics expert through his international connections. This article examines how Arni networked with international acoustics experts and how he acquired information about acoustics and conveyed it to Finland. The article focuses on Arni's activities especially in the fields of building and room acoustics and noise abatement. Electroacoustics is excluded from this article. Similarly, Arni's activities at the Finnish Broadcasting Company are not discussed except in relation to building and room acoustics. The article is based on a wider work on life's work of Paavo Arni [1]. Thus, apart from Arni's own articles, the Finnish references are mainly not cited in this article, but they can be found in reference [1].

Figure 1: Paavo Arni demonstrates his aquarium, a water model built to study the acoustics of the upcoming concert hall in 1946. The hall was never realized. Source: Finnish Broadcasting Company Yle Archives.

2 Formation of Arni's international networks

The construction of the Finnish Broadcasting Company's studios required knowledge of acoustics, but there was no research and teaching in acoustics in Finland in the 1930s [4]. Therefore, knowledge had to be acquired from elsewhere through technology transfer. One form of this is study trips abroad, which Paavo Arni frequently had in various positions at the Finnish Broadcasting Company from the 1930s onwards. The newspapers reported Arni's travels abroad in great detail and quoted his enthusiastic reports of the sound insulation, room acoustics and noise abatement solutions he had seen on the way. [1]

An example of how technology developed abroad was transferred to Finland is adjustable room acoustics. In 1946, a new music hall was planned for the Finnish Broadcasting Company in an old military building, the Riding Hall of the Guard in Helsinki. Arni told to the press that he had recently made a "quick expedition to similar buildings abroad" and continued: "I travelled to Copenhagen and Brussels." In the recently completed radio buildings in these cities, his attention was drawn to the fact that the concerts broadcast were not played in studios but in concert halls located in the radio buildings. In Brussels, he was interested in the fact that the studio could "automatically change the acoustics", i.e. use electric motors to rotate hexagonal columns covered with different sound-absorbing materials to the positions required to achieve the ideal reverberation time for the performance. [1] The project of converting the old Riding Hall into a 600-seat concert hall was soon dropped, but plans for a music studio continued. In 1949, the Finnish Broadcasting Company's "hall with changing acoustics" was introduced to the press, mentioning that it was unique in the Nordic countries. The hall featured structures that could be turned by hand to adjust the reverberation time to suit different purposes. Arni (Fig. 2) predicted that "construction plans for larger halls in the next few years will use the 'invention' now being tested." This did happen: in the early 1950s, the new buildings of the School of Business and Hanko City Hall were completed, and Arni had designed adjustable acoustics for both of them [6].

After developing the Finnish application of adjustable acoustics, Arni did not keep it to himself; in 1950, he published its principle in a Finnish architectural journal in his article on the acoustics of the new building of the School of Business. In the same year, he published an article on the subject in *The Journal of the Acoustical Society of America* [7]. In spring 1950, Arni gave a presentation on "changing acoustics" at an international acousticians' meeting in Marseille. As a result, a group of English acousticians visited Finland in the summer of that year to learn more about this technique [1].

The above shows how Arni networked with foreign acousticians. Thanks to the networking, in the 1940s and 1950s, the meetings of the Acoustical Society of Finland featured presentations by three Danes – Dr. Per Brüel, founder of an acoustic measuring equipment company and assistant professor at Chalmers University of Technology, Dr. V. L. Jordan, leading room acoustics expert in the Nordic countries, and Professor Fritz Ingerslev – as well as Swiss Professor Willi Furrer. During a trip to the United States to study acoustic laboratories in spring 1948, Arni met Leo Beranek, the American regarded as the most important acoustician of the 20th century. During the trip, he visited the acoustic laboratory at MIT, where Beranek was working at the time. At the International Congress of Acousticians in London in summer 1948, Arni met Beranek again. Late that same summer, Beranek made a visit to Finland, hosted by Arni, and mentioned it in his memoirs published 60 years later [8].

By the beginning of the 1950s, Paavo Arni had achieved international fame to the extent that, in autumn 1951, he was invited to London to listen to the problematic acoustics of the newly completed Royal Festival Hall at a test concert [9]. The building was one of the first large concert halls built after the Second World War, and its design had aimed to follow up-to-date guidelines on acoustics as closely as possible [10]. Of the 18 members of the international group of experts that evaluated the acoustics of the originally 3,404-seat concert hall, Arni was among the most critical [9]. The international appreciation for Arni is also shown by the fact that Professor Willi Furrer, after visiting Finland in 1953, published a spectacularly illustrated article on Arni's design work in the *Schweizerische Bauzeitung* [6].

3 Organisational duties and publications

A visit to Helsinki in 1942 by the German Dr. Hans Joachim von Braunmühl proved to be very important for the development of acoustics in Finland. The visit was related to the Finnish Broadcasting Company's project to construct a new radio building to replace the quickly overcrowded premises from 1934. An expert in acoustics and broadcasting technology, von Braunmühl gave a presentation for the company on the acoustic design of buildings, sound insulation, room acoustics and the acoustic properties of building materials. The presentation was published as an extensive two-part article in a Finnish professional journal, translated by Arni [1].

The radio building project was dropped during the war, but von Braunmühl's presentation generated so much interest and enthusiasm for acoustics that, on 25 August 1942, a meeting was held at the Finnish Broadcasting Company, attended by 11 experts from various fields of acoustics, from both business life and research institutions. Paavo Arni had prepared a proposal for the meeting that "a society should be established in Finland whose task would be to carry out research and measurements in the field of acoustics and to carry out educational work among certain circles." All participants supported the establishment of the society and, at the end of the meeting, an ad hoc committee was set up to draw up rules for the society. Paavo Arni was elected secretary and convener of the committee. [1]

After the committee had completed its duties, the founding meeting of the Acoustical Society of Finland was held on 29 March 1943, during which the rules of the society were approved. According to them, the society was to organise meetings, presentations and lectures, promote research in the field and the distribution of professional literature, engage in publishing and advisory activities and develop Finnish vocabulary in the field. Paavo Arni was elected a member of the Board of the society. He served as a member of the society's Board for three periods totalling 26 years, 15 of them as Chairman. [1]

The Acoustical Society of Finland became one of the channels for technology transfer. Although technology transfer relies to a large extent on the actions of individual people, the connections between them and their mobility, contact between individuals is not enough to make it happen. In order to adopt the technology, a more general interest is needed, and the activities must be well-organised [11]. The Acoustical Society of Finland provided the necessary framework for the promotion of acoustics. During its first ten years of operation, eight foreign experts lectured at its meetings. Arni himself gave nine presentations during the same period, at least four of which were based on his trips abroad. [1]

Arni's most important literary work is a textbook published in 1949, *Käytännöllisen akustiikan perusteet* (The Basics of Practical Acoustics) [12], which remained the only Finnish textbook in the field for over a decade. It was also the result of Arni's contacts abroad and visits by international experts to Finland: of the 61 titles in the book's bibliography and recommended literature, 20 were from Germany, 13 from the Nordic countries, 10 from the United States, and the rest from the United Kingdom, the Netherlands and Switzerland. There were only six Finnish references. Based on his book, Arni also lectured on an acoustics course at the Helsinki University of Technology in the 1950s [4]. The development of teaching was on the agenda of the Acoustical Society of Finland during the time that Arni was on its Board [1].

In the 1950s, Paavo Arni's organisational duties expanded outside Finland. He attended the first International Congress on Acoustics (ICA) in Delft in 1953 [1]. Towards the end of the congress, the representatives of the four Nordic countries gathered together and decided to establish a Nordic acoustical society under the name *Nordiska Akustiska Sällskapet* (NAS) [13]. The Acoustical Society of Finland appointed Paavo Arni as Finland's representative to the Board of the

Nordic society [3]. After the success of the NAS test conference held in Copenhagen in 1954, the first actual Nordic conference was held in Helsinki in 1956, as the Acoustical Society of Finland was the oldest national society in the field in the Nordic countries [13]. The conference was opened by the Chairman of the hosting society, Paavo Arni. When the ICA congress was held in Copenhagen in 1962, the NAS was responsible for organising it. The advisory committee of the congress was composed of members of the Boards of the national member societies. Finland was represented by Paavo Arni and two other Finns [14].

4 Committee and research work

At the initiative of the Acoustical Society of Finland, the Government appointed a committee at the beginning of 1945 to prepare legislation to reduce the harmful effects of noise at workplaces. Paavo Arni was invited to be a member of the committee but, after the death of the Chairman of the committee, he was appointed as the new Chairman in 1947 [1]. At this time, Arni's study trips abroad were to sites that were important in terms of hearing protection. For example, during a trip to Sweden for a presentation at a radio association meeting in Stockholm in 1945, he studied the "magnificent work done at the ASEA factory in Västerås to suppress the high levels of noise in the factory and offices". Between 1948 and 1949, he made three trips abroad in connection with the work of the noise abatement committee: in April 1948 to the United States to study acoustic laboratories and "the protection of workers against noise", in July 1948 to the International Congress of Acousticians in London, and again to London in September–October 1949. The experts that Arni met during his travels provided the Committee with research literature in the field [1].

Later, Arni was involved in an international committee relating to noise. Since the early 1950s, he had been involved in motor vehicle noise measurements and, when the NAS started developing a measurement standard in 1956, Arni was appointed as the leader of the standard workgroup. The workgroup's proposal later served as the basis for the international ISO standard [13].

In 1947, the Board of the Acoustical Society of Finland decided to approach the Ministry of the Interior with a letter on "the drafting of sound engineering standards for house builders". This had been influenced by the fact that, at the society's meeting in the spring of the same year, Dr. V. L. Jordan from Denmark had given a presentation on the regulations concerning sound insulation and noise abatement in various countries. The secretary of the society, Paavo Arni, submitted Jordan's presentation to the press, translated into Finnish, with the title *Current legal rules and standards concerning noise and sound insulation in different countries*. A proposal for a committee for the drafting of domestic standards was submitted to the Ministry of the Interior in spring 1948. [1]

Achieving sound insulation standards proved to be a long-lasting project, whose important milestones were the sound insulation research committee in 1952–1955 and a committee established in 1957, which drafted a proposal for sound insulation regulations published in 1960. Paavo Arni was invited to be a member of both committees. The achievement of sound insulation regulations is probably societally the most significant project initiated by the Acoustical Society of Finland, resulting in the improvement of sound insulation in homes and thereby affecting the daily lives of millions of Finns. The project was completed in 1975, when the National Building Code of Finland was published. [15]

Arni did not only use his networks to transfer information from abroad to Finland, but he was also able to support research elsewhere [9]. For example, he was involved in organising Leo Beranek's second visit to Finland in 1960, when Beranek was writing the first version of his famous book on concert halls [8, 16]. During his trip, Beranek visited the University of Helsinki's festival hall, which had been rebuilt and expanded in 1945, as well as the Helsinki House of Culture and the Turku Concert Hall. Arni was responsible for acoustics in all these projects. At the time, the Turku Concert Hall appears to have had a good international reputation, and thus it and the House of Culture were introduced in Beranek's book [16].

In the early 1950s, the diffusion of the sound field in a space was a subject of interest to researchers, since it had been observed that the smooth surfaces favored by the architecture of the era did not provide listeners with the same experience of acoustics as old concert halls did, even if the reverberation time was the same. Arni defined the matter as follows: "In old halls, the sound is reflected from broken surfaces, columns, recesses, balconies, etc. as irregularly as possible, diffusively, which again means as even a reflected sound field as possible throughout the hall, i.e. balanced and good audibility. In a modern hall, special diffusers are used in an attempt to achieve this diffusion […]". Convex spherical surfaces had been arranged on the surfaces of the festival hall of the Swedish School of Economics designed by Arni and finished in 1953, and vertical wavy and prismatic surfaces had been placed on the side and back walls. Arni's article in the architectural journal in 1955 is apparently the first Finnish text describing diffusion. [1]

Among the researchers Arni met in London in 1951 [9], Professor Willi Furrer and Professor Erwin Meyer from the University of Göttingen studied diffusion in the early 1950s. Furrer also gave a presentation on the subject at a conference in Delft in 1953 [17], attended by Arni. In an article in 1955, Arni said that a research group at the University of Göttingen was studying diffusion in acoustically interesting halls. The festival hall of the Swedish School of Economics was also measured in this context, and so was the Turku Concert Hall. Paavo Arni was involved in arranging access to both spaces for the group of researchers, for which Meyer and Thiele thanked him in a scientific article they published in 1956 [18].

5 Arni as acoustical designer

Paavo Arni's design work includes concert halls and theatres, sacral buildings, educational institutions, office buildings and hotels. They were designed by leading Finnish architects of their time, and Arni seems to have had long-term cooperation especially Kaija and Heikki Siren. Of Alvar Aalto's projects, Arni was involved in designing the acoustics of the House of Culture, and he also started design work for the Finlandia Hall before his death. Abroad, Arni participated in the design of the Brucknerhaus Concert Hall in Linz and the studios built by Austrian Radio in Salzburg, Innsbruck, Dornbirn and Linz [19].

Paavo Arni described his design work methods in his book *Käytännöllisen akustiikan perusteet* (The Basics of Practical Acoustics) [12]. According to Architect Alpo Halme, who worked in Arni's office, the book "contained almost all the relevant information that existed at the time." [19] In terms of measurement methods, Arni refers in his book to the German DIN 4110 standard from 1938, and he presents a mass law formula for the calculation of airborne sound insulation. For room acoustics design, the book presents formulas by Sabine, Eyring and Millington for calculating reverberation time. In addition, there are tables and diagrams on both topics with recommendations for different spaces and material properties. Arni also introduces the use of geometric room acoustics in design and the light and water models used by the Finnish Broadcasting Company. Geometric room acoustics were used by him in the design of the rebuilding of the University of Helsinki's festival hall. [1]

The water model was used in the discontinued project of converting the Riding Hall of the Guard into a concert hall for the Finnish Broadcasting Company. Arni explained the water model to the press in 1946: "A metal basin has been made that closely follows the shape of a concert hall, and its bottom is made of glass. There is also a metal ball hanging at the end of a kind of lever. The metal ball corresponds to a sound source in tests made with this water model. When it touches the surface of the water, waves are formed on the surface that correspond to sound waves. By observing these, we can see how sound waves are reflected from the surfaces of the room." It was also also reported that the Finnish Broadcasting Company's personnel had named the water model "Arni's aquarium" (Fig. 1). [1]

When Arni started his work, there were yet no international standards for acoustic measurements, but he lived to see measurement standards published for most conventional measurements. The magnitude of the change is illustrated in an article published by Arni in 1944, in which he describes reverberation time measurements in concert halls. At the time when Arni wrote his article, an orchestra was used as the sound source, and works suitable for measurements included "for example, the end of Sibelius's 5th symphony, the first bars of Beethoven's Coriolan Overture, the beginning of Bach's Toccata and Fugue in D Minor." In all of these works, fortissimo sections are followed by rest, during which the reverberation time could be measured from the decrease in sound pressure. [1]

6 Conclusions

The above is an account of Paavo Arni's work as an acoustician and his international networks, created on the basis of written sources. In recent decades, the history of technology has been dominated by ideas about the social construction of technology [20], with little attention being paid to the influence of individuals on the development of technology. According to the sources available, Paavo Arni seems to have been an exceptionally influential person in his field. Arni's design work does not necessarily differ in significance, scope or difficulty from the work of other Finnish acousticians who worked at the same time. What sets him apart from his contemporaries is his broad scope and apparent desire to acquire and share information as well as to promote the development of the field of acoustics and the creation of better sound conditions in Finland. This also made him the first internationally known Finnish acoustician who, through his extensive contacts, was able to convey the latest research information to Finland, where the chair of acoustics was established only after his death [4].

Paavo Arni's main occupation in the fast-developing broadcasting industry provided opportunities to establish contacts with international acoustics experts at conferences and on study trips and by hosting visits by foreign experts in Finland. Young Arni's aim seems to have been to learn about different fields of acoustics and convey information about them to Finland. Subsequently, he seems to have sought to alleviate the social problems associated with sound conditions that were developing, as exemplified by his work in the noise abatement and sound insulation committees. His work thus ranged from practical design work to societal influence. Some of the things he advocated became reality only after his death, such as the issuing of sound insulation regulations [1, 3].

A key part of Paavo Arni's career and achievements was the international network of experts that he had formed since the 1930s, which enabled the transfer of technology to Finland, where there was not much teaching and research of acoustics in the absence of a chair. Mr Juhani Borenius, who worked under Arni at both the Finnish Broadcasting Company and Arni's engineering office, assumed that Arni's mobility in international circles was made easy by his diverse language skills and his childhood environment in the multicultural city of Vyborg. Although Arni managed to achieve a lot and was efficient in all his duties, his work was left unfinished: Paavo Arni died in spring 1969 at the age of 64, and his plan to focus on acoustic design during retirement did not materialize [1].

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Acoustic intervention in the French pavilion of the Student Centre in Zagreb as a protected cultural heritage building

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The French pavilion of the Student Centre in Zagreb was built in the 1930s as a part of what was known as the Zagreb Assembly, serving as the exhibition pavilion of the Republic of France. Designed by French architects and built by local contractors, it boasts some revolutionary and unique design solutions and construction techniques. It was fully restored in 2014 and is considered a protected cultural heritage building of immense architectural and cultural value. Today, the pavilion is used for a broad range of events of scientific, educational, and cultural nature, such as exhibitions, concerts, meetings, receptions, banquets, graduation ceremonies, etc. Unfortunately, the acoustics of the pavilion is unfavourable for any kind of speech- or music-based events due to its circular shape and hard finishing materials. This results not only in excessive reverberation, but also in very strong echo, both having severe negative impact on the intelligibility of the spoken word and on the clarity of the performed music. This paper presents the challenges in improving the acoustics of this space in the frame of the very limited range of interventions that are allowed by the conservation experts. The current state of room acoustics was diagnosed by means of room acoustic measurements, and the 3D model of the space was built for the purpose of simulations. Although suboptimal, a solution was found that conforms with all the imposed limitations and is expected to considerably improve the acoustics of the pavilion, thus significantly increasing its functional value.

1 Introduction

Although it is vital in the design and construction of a building, room acoustic design often proves to be a challenging part of this work due to numerous reasons, even though at the design stage a wide range of options is available and at the disposal of the multidisciplinary teams that include architects, acousticians, and other experts in their respective fields. For example, in 32 years of Croatian independence, no rule or regulation has existed that would enforce the obligation to include room acoustic design into design projects or the implementation of any kind of acoustic treatment in rooms, regardless of their use. Consequently, room acoustics design is often left to the conscience of architects, most of whom do not see it as a design issue, and the ones who do are often overruled by investors as there is no legal obligation to implement these design measures. The situation is currently changing for the better, as a new Technical Regulation on Acoustics in Buildings is being developed at this time that will set the room acoustics criteria according to the HRN DIN 18041:2024 standard [1], and the method of calculating the reverberation time as the relevant parameter is to be stipulated by HRN EN 12354-6:2005 [2].

In buildings that require restoration or refurbishment, the challenges prove to be even greater, as the range of options that are available and implementable in such buildings is often quite narrow, as it is defined by the existing appearance of the rooms that require acoustic treatment, and both the investors and the architects strive to preserve the original appearance of the building.

A special category of buildings are the ones that are under protection as cultural heritage. An architect is legally required to obtain a license to be able to work on buildings of this category. The requirements set by conservation bodies on any work done in or around such buildings (restoration or otherwise) are immensely detailed and strict and are to be followed to the letter. From an engineering perspective, the requirements are sometimes driven to the point of the absurd, as they are focused on preserving the original appearance of the building, while the functionality of the building for its intended purpose is often overlooked.

This paper presents a case study of the room acoustical properties of the French pavilion of the Student Centre in Zagreb [3], which falls into the category of cultural heritage of Croatia protected at the highest level. The original design of the building did not account for the acoustics of its interior, and the restoration work that was carried out was focused on preserving the constructional and aesthetic details. Thus, profoundly bad acoustic conditions remained one of its hallmarks. This case study was made on request of the University of Zagreb as the current manager of the building. It includes the diagnostics of the current room acoustical conditions in the building through measurement, and the proposal of an acoustic solution that would adhere to very strict requirements set by conservation bodies, the investor, and even the architect in charge of restoration. The view of the architect that "people will be swept off their feet by the sheer visual grandiosity of the building and will be prone to tolerating its bad acoustics" was an additional motivation to commence with this project, as the experiences from practice show otherwise.

2 The history of the French pavilion

The French pavilion was built in 1937 as a part of the Zagreb city fair known as the Zagreb Assembly. Serving as the exhibition pavilion of the Republic of France, it was designed by the French architect and urbanist Robert Camelot of the Jacques and Paul Herbé architectural bureau, and the structural engineer Bernard Lafaille. The construction of the building was entrusted to local contractors led by the Faltus Brothers construction company from Zagreb.

The pavilion boasts some unique engineering innovations regarding its extremely lightweight construction. Due to these innovations the building has earned the status of the protected cultural heritage of the highest level and was entered into the Registry of cultural heritage of the Republic of Croatia in 2003.

Despite its status, the history of the French pavilion has been quite turbulent. It retained its original purpose as an exhibition building only until 1956, when the Zagreb Fair was relocated to a more suitable location, and the present one was used to form the Student Centre as the central location for young academic population in the city. Unfortunately, from that point on, no sensible use has been found for this building other than the storage facility for timber, office equipment, furniture, and stage equipment of the theatre that was also a part of the Student Centre. In the 1990s, the building was used as a theatrical stage space for a short while, after which the use of the pavilion was forbidden for any purpose due to poor state of the building. The building suffered from several poorly designed and executed construction details from the start (faults in the roof construction, poorly designed water drainage within the building, poor quality of wooden panelling), which, accompanied with a lack of proper maintenance, ultimately led to the building falling into disrepair.

Over the years, several attempts were made to induce the restoration of the French pavilion in form of elaborate restoration projects written in 1963, 1982, and 1992. None of these projects ever reached the execution stage. The latest attempt to restore this piece of cultural heritage was initiated in the late 2000s, and the comprehensive restoration work was finally finished in 2014, thus marking the beginning of the new life for this architectural masterpiece. The restoration included numerous stages of construction restoration that were executed to improve the structural stability of the building and to meet the requirements set by modern standards in construction, but utmost care was taken to preserve the visual appearance of the building, as well as the essence of the engineering innovations that were built into it.

As a part of the Student Centre, the French pavilion is now used for different kinds of events under the auspices of the University of Zagreb, such as graduation ceremonies and similar events, exhibitions, meetings, concerts, etc. The use of the pavilion is permitted for other types of events, provided that it does not result in damage to the building as a cultural heritage.

During its existence, the acoustics of the French pavilion has never been investigated as an issue or even considered as a design goal that needs to be achieved. However, in 2023 an initiative came from the Chancellor's office at the University to address this issue, as it became clear that the acoustical conditions in the building are appropriate neither for any kind of event that relies on the intelligibility of the spoken word, nor for events that include musical performance.

3 The design of the French pavilion

The design of the French pavilion is based on an essentially circular footprint defined by twelve steel columns with a diameter of 80 cm regularly spaced along the perimeter of a circle that is 29.6 m in diameter. The lightweight self-supporting roof construction has a total mass of only 18 kg per square meter of covered space. It rests on these perimeter columns with no additional support inside the building and is shaped as a shallow inverted cone with a truncation that houses the central skylight. The inner shell of the building is somewhat more elaborate than the basic circular design it stems from, as shown in Figure 1 on the 3D model that was built from available blueprints and field measurements of the physical dimensions of the interior. The perimeter walls are 12.8 meters high. The bottom 3.75 meters are made of solid concrete as the load-bearing support. The next 7.50 meters are built as wooden construction frames with wooden boards as the finishing, also housing narrow window strips in full height that are 65 cm wide. The top part of the perimeter walls (below the ceiling) is also made of wooden frames with wooden boards as the finishing material. The entire floor of the pavilion is made of smooth reinforced concrete. A photograph of the current interior of the pavilion is shown in Figure 2.

The described geometry of the pavilion yields 575 m^2 of floor space, as well as 2480 m^2 of total surface area, and approximately 7000 m^3 of net room volume. The pavilion is to host a seating or standing audience of up to 350 persons, which yields the volume per person of 20 m^3 or more.

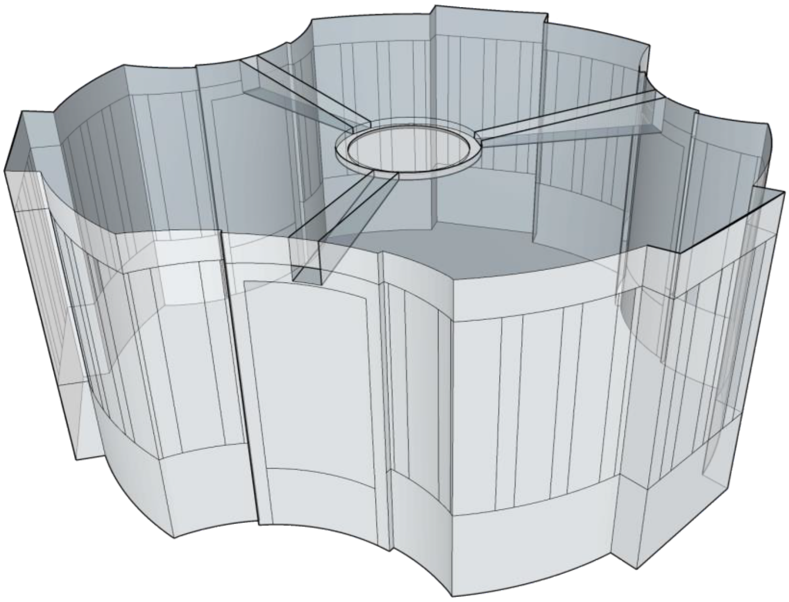

Figure 1: The 3D model of the French pavilion in X-ray view

Figure 2: The interior of the French pavilion at the time of room acoustical measurements

4 Diagnostic measurements of the current state

Acoustical properties of the French pavilion have been investigated through a series of measurements of room acoustical parameters performed according to HRN EN ISO 3382-1 [4] using the integrated impulse response method. Additionally, several impulse responses were recorded separately using simple hand claps to investigate the issues with extensive echo present in the room. Due to the described nature of the problem, it was expected that the principal task in acoustic design will be to control the amount of reverberation and to attempt to shorten it to acceptable length, so the principal parameter of interest was the reverberation time T_{30}. Additionally, the early decay tome *EDT* was measured as well, as the parameter that is strongly associated with perceived reverberance. As the main complaint received from the users of the building is bad speech intelligibility, the speech transmission index *STI* was measured as well.

The results of the acoustical measurements conducted according to HRN EN ISO 3382-1 are shown in Table 1 and on the chart in Figure 3.

Table 1: The reverberation time T_{30} and the early decay time *EDT* measured in the French pavilion in its present state

Frequency (Hz)	63	125	250	500	1000	2000	4000	8000
T_{30} (s)	1.78	4.32	5.50	7.04	7.01	5.48	3.04	1.67
EDT (s)	1.83	4.13	5.64	6.86	6.85	5.31	2.70	1.38

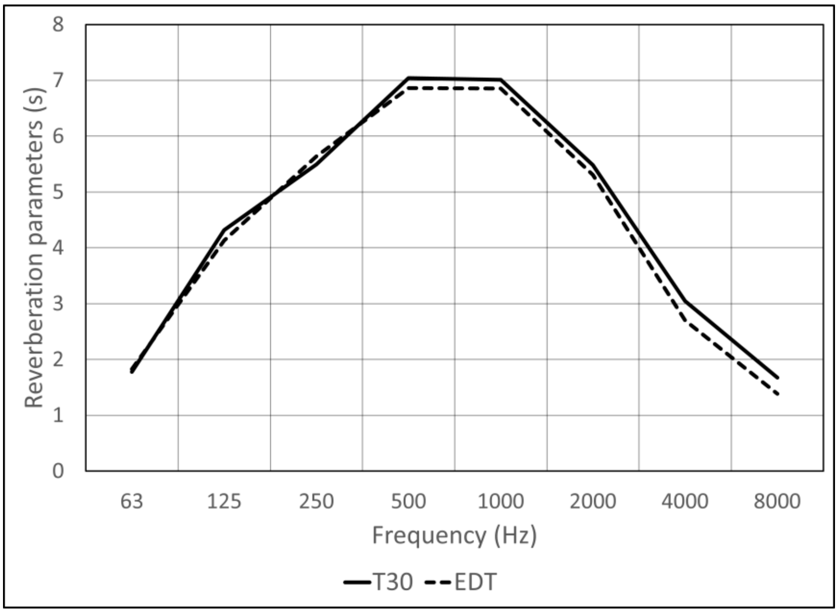

Figure 3: The reverberation time T_{30} and the early decay time *EDT* measured in the French pavilion in its present state

Both the reverberation time T_{30} and the early decay time *EDT* show a rather unique dependence on frequency. A roll-off towards the high frequencies is expected due to the large volume of the room and the sound absorption in the air that occurs. However, low-frequency reverberation is also controlled up to a point due to very large surfaces in the room that act as low-frequency absorbers. Virtually all the perimeter walls above 3.75 meters (wooden plating and windows above the concrete) and the entire ceiling are lightweight membrane-like constructions that are quite efficient at absorbing low frequencies. On the other hand, mid-frequency absorption is poor, as the described materials behave as hard, reflective surfaces in this frequency range. As the entire floor and the lower part of the perimeter walls are hard and reflective, being made of concrete, there is very little absorption in the mid-frequency range, resulting in an excessively long reverberation time that reaches the value of 7 seconds.

Figure 4 shows the impulse response recorded in the geometric centre of the room as the response of the room to a hand clap produced at the same position. A strong reflection arrives at this position 80 ms after direct sound, resulting in an audible echo. Another reflection, albeit somewhat more diffused, arrives 100 ms after direct sound. These arrival times correspond with the distances of the measurement position from the critical segments of the perimeter walls. Similar results were obtained at other measurement positions.

The diagnosed acoustic issues in the French pavilion regarding the excessive reverberation and strong echo are in accordance with the observations given by the users, who deem the acoustic conditions in the pavilion to be highly inappropriate for the events it hosts. Moreover, the values presented here were obtained for the worst possible case, i.e. for empty pavilion, whereas the observations that come from the users reflect the state when the pavilion is, in fact, occupied. The complaints about bad speech intelligibility were confirmed by measurements, as the average value of the speech transmission index *STI* is only 0.33, and all the individual values fall in the range from 0.30 to 0.39.

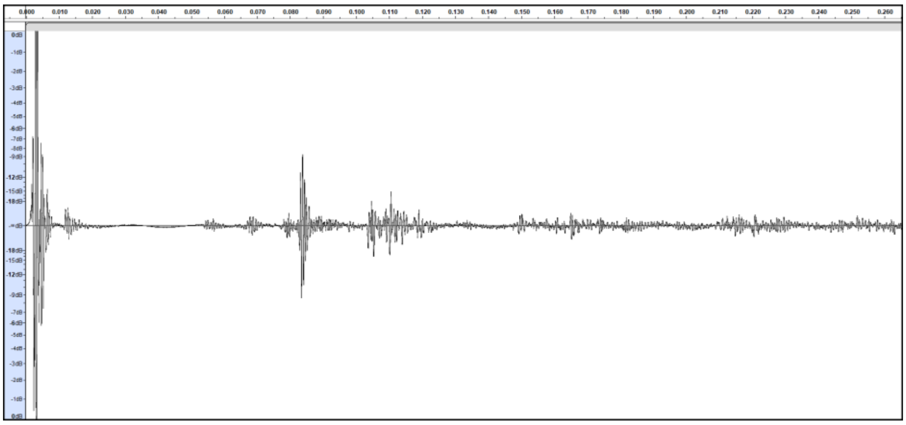

Figure 4: An impulse response recorded in the centre of the pavilion using a hand clap, showing strong, clearly audible echoes arriving 80 and 100 ms after direct sound

5 The proposed room acoustic solution

The design constraints in proposing a solution for improving the room acoustics of the French pavilion were considerable, mostly because the building is registered as a national cultural heritage. The fundamental, non-negotiable requirement was that no permanent installation of any kind is permitted, and that any proposed solution is to be portable, so that it can be moved around the room, or even taken out if necessary. To address the evident problem with echoes caused by the concrete part of the perimeter walls, and to put the absorption material as close to the audience area as possible, the proposed solution was based on heavy curtains put in front of all the concrete parts of the perimeter walls, as shown in red in Figure 5. The curtains would be divided into manageable pieces, and each of them would be placed on a movable and foldable stand in front of a given section of the perimeter wall. Apart from being a portable one, the proposed solution is designed to address the issue of excessive reverberation in the mid-frequency range.

The curtains used as the design solution are generic cotton curtains with the surface mass of 0.5 kg/m^2, draped to ¾ of their stretched surface area, and mounted at 130 mm from the concrete wall. Total surface covered by the installed curtains is 310 m^2, which turns out to be exactly 1/8 of the total surface area of the room.

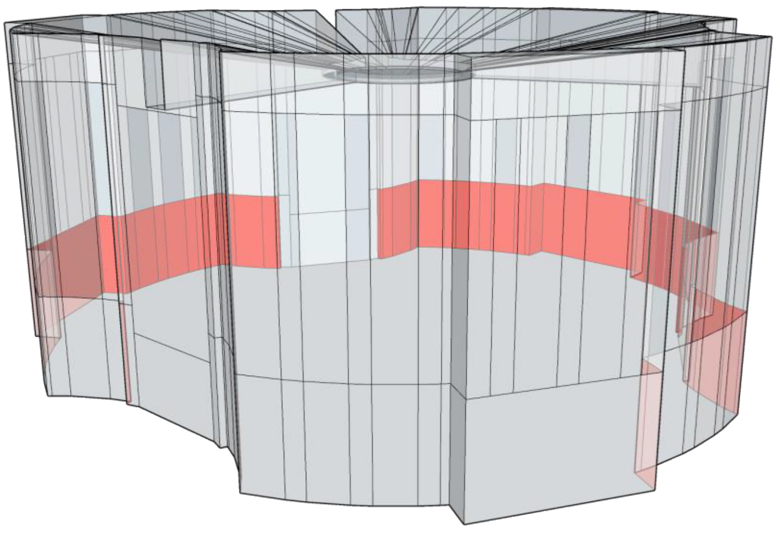

Figure 5: The locations of curtains as the sound-absorbing material (in red)

To obtain a solid starting point for the design of an appropriate acoustic solution, the measured values of the reverberation time were reverse engineered in the simulation model of the initial/current state, yielding a very good agreement between the measured and the simulated values. In the second stage, curtains were added into the model as the designed solution to investigate their influence on the acoustic conditions in the room. To simulate the presence of the audience, a conservative estimate was made by replacing 240 m^2 of the otherwise empty floor with an audience area that contains one person per square meter sitting on a wooden chair. The audience was added to the simulations of both the initial state and the final one obtained after acoustic treatment. The reverberation time for all the investigated cases is shown in Table 2 and on the chart in Figure 6.

Table 2: The reverberation time T_{30} for the initial state (measured and simulated) and the final state after treatment (simulated), in empty and occupied conditions

Frequency (Hz)	63	125	250	500	1000	2000	4000	8000
T_{30} (s), initial state (measured, empty)	1.78	4.32	5.50	7.04	7.01	5.48	3.04	1.67
T_{30} (s), initial state (simulated, empty)	1.93	4.20	5.40	7.06	6.91	5.39	3.21	1.38
T_{30} (s), initial state (simulated, occupied)	1.70	3.76	4.18	3.53	3.12	2.54	1.92	1.07
T_{30} (s), after treatment (simulated, empty)	1.50	3.13	3.26	3.19	3.47	3.00	2.08	1.13
T_{30} (s), after treatment (simulated, occupied)	1.44	2.88	2.75	2.10	2.05	1.76	1.40	0.90

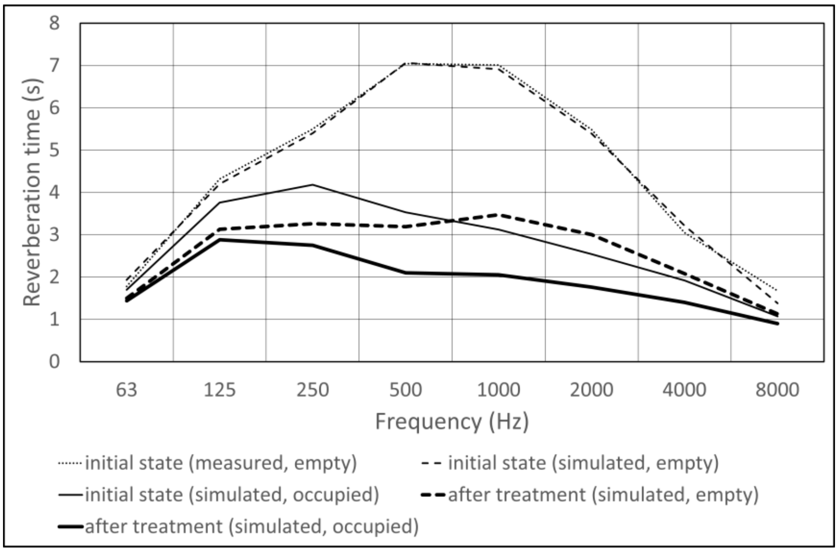

Figure 6: The reverberation time T_{30} for the initial state (measured and simulated) and the final state after treatment (simulated), in empty and occupied conditions

The results of the simulations shown in Table 2 and Figure 6 show that the introduction of the acoustic treatment in form of curtains on perimeter walls has significantly shortened the reverberation time in the room. In other words, the best possible acoustic conditions that can be obtained in the current state, i.e. in the room with no acoustic treatment, but occupied, are similar to the starting conditions that can be obtained in the empty room after treatment. Further improvement is gained by allowing the audience to enter the room, as intended in normal use.

The advantages of the proposed solution are its sound-absorbing properties targeted specifically at mid-frequency absorption, its full portability that allows the curtains and their stands to be deployed, moved around, and taken out of the room if necessary, and its positioning at the listening height from 0 to 3.75 m that automatically facilitates the removal of the echo. Most importantly, the absolute requirement set by the investor has been satisfied, i.e. permanent installation of acoustic treatment was avoided to maintain the protection of the building as a cultural heritage.

The possible drawbacks of the proposed solution stem from the fact that the curtains need to be placed against (in front of) a hard surface to be effective as sound absorbers. If the room host an event such as an exhibition, for which it is required to have the perimeter walls free and available, the proposed solution cannot be fully implemented as designed. Therefore, in this case the effect of shortening the reverberation time will be directly proportional with the surface area of the perimeter walls that remain available for acoustic treatment. As the segments of the curtains are supported by mobile stands, they can be placed elsewhere in the room, i.e. away from walls, but such positioning would diminish their sound absorbing properties. To address this issue, an alternative solution was proposed that would comprise free-standing mobile panels of manageable size, equipped with wheels that facilitate easy deployment, relocation, transport, and removal. The panels would need to be made of a solid upright plate, and absorbing material could be mounted on both sides of that plate to double the sound absorption capacity of such panels.

The proposed solution represents a step towards achieving good speech intelligibility in the room, as it helps shorten the reverberation time and removes the problematic echoes. The simulations show that the values of the speech transmission index *STI* obtained for the empty room in its current state match the ones obtained from measurements, both revealing that poor or bad speech intelligibility ($STI < 0.45$) is to be expected in the entire room. After treatment, the simulations predict that fair speech intelligibility ($STI > 0.45$) is to be expected for most of the floor area when the room is occupied as described above. Regarding the speech intelligibility, the simulations show the worst-case scenario, as they were made using an omnidirectional source. Further improvement is expected by using directional loudspeakers in sound reinforcement systems that are to be utilized in the room. At this point it was not possible to make exact predictions on how large this improvement will be, as there is no permanently installed sound system in the room, but the systems change from one event to the next and are fully mobile (just as the proposed room acoustic solution is).

6 Summary

The French pavilion of the Student Centre in Zagreb was built in 1937 and is now a protected cultural heritage building at the highest level. It has been given a new life upon a thorough and long overdue restoration that was completed in 2014. It boasts some construction techniques that were revolutionary for the period when it was built, thus having earned the status of a cultural heritage.

Unfortunately, neither the original architects nor the ones in charge of restoration paid any attention to the acoustics of the pavilion, despite its intended use as a place of gathering. The visually spectacular, but acoustically unfortunate choice of the overall shape of the building and the materials used in construction has resulted in acoustic conditions that are extremely unfavourable for any kind of event based on spoken word and/or music performance due to excessively long reverberation and strong echoes.

To address this issue, a solution for acoustic treatment was proposed that adheres to the severe limitations and strict requirements imposed by the investor and the public bodies in charge of conserving the cultural heritage. The calculations show that the implementation of the proposed solution will lead to significant shortening of the reverberation time and the complete removal of the detrimental echo. Although the resulting reverberation time is still longer than required for rooms used for the described purpose, mostly due to the excessively large room volume, considerable improvement of acoustical conditions in the pavilion is expected. Regarding the speech intelligibility, further improvement is expected in comparison with the calculated values using directional loudspeakers as an integral part of the sound reinforcement system.

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Turku Fuuga – Acoustic Design of an Intimate and Immersive Concert Hall

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A new music centre is currently under construction in Turku, Finland, containing a 1300-seat concert hall dedicated to symphonic concerts and a 300-seat multipurpose hall primarily designed for chamber music concerts and orchestra rehearsals. The centre will become the new home of the Turku Philharmonic Orchestra. Design work started in early 2021 and opening is planned for spring 2026. The architectural design is led by PES-Architects, with the acoustic design by Kahle Acoustics and Akukon Ltd.

This paper describes the acoustic intentions for the main symphony hall and the specific design process developed for the project. The typology developed for the hall aims above all at acoustical excellence, while also seeking to transform the traditional frontal shoebox shape into a new, more intimate paradigm. The design is entirely based on curved surfaces, requiring precise analysis of the acoustical behaviour of 3D curved shapes and an appreciation of their potential to convey early reflections with optimum delay, strength and direction of arrival. Excellent clarity and strong acoustic impact are expected from this concert hall, since the design optimisation – with architects and acousticians working within the same 3D parametric environment – has achieved multiple early reflections to all audience seats.

Particular consideration was also given to the spatial distribution of the acoustic volume, aiming at maximizing the audience's sensation of being immersed in and enveloped by the music.

Inspired by successful precedents and displaying very promising acoustic simulation results, anticipation for the forthcoming Turku Fuuga music centre is growing.

1 Introduction

The final stage of the architectural competition for the new Turku Fuuga music centre took place in the first half of 2021. The Turku Philharmonic Orchestra is currently based in Turku Konserttitalon. Built in 1952, this building is a remarkable example of Finnish modern architecture by the architect Risto-Veikko Luukkonen, and Finland's oldest concert hall. But it no longer meets the orchestra's needs. It was thus decided to build a new home for the orchestra.

Concert hall acoustics specialist Tapio Lokki was appointed as client advisor at the start of the project to ensure that acoustic excellence was regarded a primary goal even before the design process began. In discussions with representatives of the orchestra, he defined the sound that the new concert hall ought to have. The result can be summarized as follows: the main concert hall in the music centre has to be designed for symphonic music concerts, with no acoustic compromise related to other uses. The acoustics must be powerful and reverberant, with excellent clarity and a strong feeling of immersion in the sound. From an architectural point of view, a shoebox typology is explicitly recommended, and the audience must be organised with a relatively flat parterre accommodating the majority of seats, plus additional seats in one or several balconies.

Kahle Acoustics took part in the architectural competition with PES-Architects and Akukon Ltd, putting together a team that had already completed two performing arts projects together. The developed design aims at providing a concert hall that would sound like the best shoebox concert halls, with all the qualities described in the acoustic brief, but would also enhance the feeling of intimacy and visual proximity to the stage for all audience members, compared to what a traditional rectangular-shaped concert hall can offer.

This paper will describe the design process for the new concert hall, including the initial development of the general room shape and the more detailed geometrical optimizations achieved at a later stage. The final acoustic outcome – insofar as the acoustic predictions can approach reality given that the building is currently under construction – is discussed in the last chapter.

2 Early design stage: development of an intimate shoebox

2.1 A shoebox, a vineyard, or what else?

In the design of concert halls, the shoebox typology holds a special place. It is mostly defined by its rectangular plan shape, although some other features such as a relatively flat parterre, the existence of side balconies and a large empty volume of air in the upper part of the hall are also often cited as key factors. The acclaimed shoebox concert halls of the 19th century, such as Vienna Musikverein and Amsterdam Concertgebouw, have contributed to making this typology a long-established reference, recognized by most as the safest design choice when it comes to ensuring uncompromised acoustic quality [1], [2]. But with modern-day demands for comfort, safety and accessibility, and above all the need for classical music to reach younger audiences, building copies of the most successful 19th century concert halls is not a viable option. Other concert hall typologies such as the vineyard-terraced geometries have grown in influence in the last decades, even though opinions about the resulting acoustic quality are sometimes critical [3], [4]. Modern versions of the shoebox typology have proven to be a viable solution for ensuring acoustic excellence. The 1500-seat Fartein Valen hall in Stavanger, inaugurated in 2012, is one of the most successful examples of this approach [5]. As the required acoustic characteristics for the new concert hall in Turku closely matches the observed acoustic qualities of Fartein Valen hall, it became one of the major sources of inspiration for the design.

Another major source of inspiration came from the most recent concert hall designed together with PES-Architects: the 1000-seat symphony hall of the Fuzhou Strait Cultural Arts Centre in southern China, inaugurated in 2018. An innovative acoustic optimization process was developed for that project [6], which informed the architectural design and led to a much-appreciated result. The Fuzhou concert hall is however very clearly of a vineyard type.

Would it be possible to "modernise" the shoebox typology, much further than in the Stavanger concert hall, taking inspiration from other successful precedents such as the Fuzhou concert hall to improve sightlines and visual intimacy, without compromising the distinctive acoustic quality of a shoebox? The Turku project seeks to demonstrate this through facts. Hybridisation of the shoebox and vineyard typologies is not a new concept in acoustic design [7], [8], [9], but Turku concert hall is a new take on hybridisation, retaining more of the elements that underlie the acoustic success of shoebox rooms, resulting in a concept referred to as the intimate shoebox typology.

Figure 1: Fartein Valen concert hall in Stavanger is a 1500-seat modern shoebox concert hall inaugurated in 2012, and one of the main reference projects for Turku Fuuga new concert hall

Figure 2: Fuzhou concert hall is a 1000-seat vineyard concert hall inaugurated in 2018, in the other main reference project for Turku Fuuga new concert hall

2.2 The Stavanger and Fuzhou precedents

Several key aspects contributing to the acoustic success of modern shoebox concert halls are integrated to the geometry developed for the new concert hall in Turku. These aspects can be seen as a legacy of the Stavanger concert hall:

- The limited width of the hall, in conjunction with side balconies attached to the sidewalls, allowing for an efficient distribution of early lateral reflections to all audience members in the parterre. These reflections, generated by the cornice between a vertical wall and a horizontal balcony soffit, are known to be crucial in shoebox halls. They are responsible for enhanced spatial perception, acoustic presence and musical dynamics [10],[11],[12],[13]. To be fully efficient, these reflections need to reach the audience sufficiently early and from an adequate elevation angle [14].
- Special attention is paid to creating similar early lateral reflections towards audience members in the balcony levels, which is not typically achieved in purely orthogonal geometries and requires specific optimization. Balcony front surfaces are typically involved in this process, in addition to other appropriately located wall surfaces.
- A large "empty" space is provided in the upper part of the volume, where no audience or other sources of acoustic absorption is present. A rich reverberation can develop there and reach the audience from many directions, generating a sensation of being immersed in the music.
- Steeply sloped parterres and balconies of many modern concert halls shadow the sound reaching the audience from a direction behind them, and simultaneously reduce the upper reverberant volume towards the rear of the hall. As a consequence, reverberation becomes weaker and more frontal / "monophonic-like", with the consequence that the audience is no longer immersed in the music. Immersive reverberation is being considered as a critical advantage that needs to be regained, both by limiting the slopes and by implementing some changes to the classic shoebox typology. The ceiling first needs to be raised from about 16m above stage floor in historic shoeboxes to about 21m in modern ones. A slight projecting angle is also implemented above the stage and the first rows of the parterre. Suspended canopy reflectors above the stage, which did not exist in the historic shoeboxes, become necessary as the main ceiling is too far from the musicians.
- However, gentle audience slopes and additional ceiling height are not always sufficient. In some modern shoebox halls, reverberation is still heard as coming mostly from around and above the orchestra, rather than from all around the listeners as it should be. Stavanger concert hall [5] has proven the possibility to further enhance immersive spaciousness when additional volumes are created along the sidewalls of the concert hall, very much like with reverberation chambers but without the tuning complexity and with limited risks of

excessive residual absorption [15]. Smaller additional volumes with larger fixed openings towards the main volume do not create an ill-coupled reverberation and double-slope decays, but rather convey reverberant sound within the space. In the proposed design for Turku concert hall, four eye-shaped volumes are created behind the side balconies. Each of these is first fed by sound coming directly from the stage, through large portions of walls treated with a 66% open sound transparent lattice. These openings are located in portions of the sidewalls that do not generate any useful early reflections. The upper portion of these walls, responsible for the critical cornice reflections discussed earlier, are kept fully reflective and effective. The so-called "eye volumes" are then also fed by reverberant sound energy developed in the upper part of the hall, through large openings in the floor of each balcony level. This reverberant sound is channelled towards the lower part of the volume, providing the audience with late lateral sound that is known to be crucial to acoustic envelopment [16].

Several successful features of the Fuzhou concert hall also inspired Turku's design:

- Deviations from the rectangular shape are introduced in plan. Splayed walls are created around the stage, projecting sound towards the audience, and reversed-splayed walls are created at the rear of the auditorium to improve early lateral reflection coverage towards audience in the balconies. As observed in the Fuzhou concert hall [6], this also intensifies the lateral reflection coverage towards the audience more generally.
- Audience on the side balconies is subdivided into several smaller blocks, creating a more intimate setting, a better visual connection to the stage, and improved sightlines. The size of absorptive audience blocks is also limited. The subdivision of the side balconies in Turku is directly inspired from the terraced layout of Fuzhou concert hall, but the arrangement is adjusted to follow a more shoebox-like profile: limited width, two superposed balcony levels of 1 – 3 rows instead of one wider terrace level, not excessively reversed-fan orientation of the balcony fronts.
- As in Fuzhou, the design makes widespread use of convexly curved surfaces. All walls and balcony fronts are convex, except for the rear walls of the four eye-volumes that are concave but carefully checked to avoid harmful focussing effects. This gives the acoustic consultant the opportunity to precisely tune each early reflection, not only by adjusting the orientation of the surfaces to create sound reflections of appropriate delay and direction of arrival, but additionally by adjusting the curvature to control the acoustic strength and the reflection coverage [17]. Reflections off relatively narrow surfaces such as balcony fronts are inherently attenuated at low frequencies due to diffraction effects. They can therefore generate slightly harsh-sounding reflections towards limited zones of the audience (and – for a given listener – of the sound coming from only a limited portion of the orchestra [18]). Adjusting the radius of curvature is then a way of extending the reflection coverage spatially, and simultaneously ensuring a more balanced frequency content. Similarly, the right balance between reflection strength and coverage can be determined on a case-by-case basis for each individual surface in the concert hall.

Figure 3: Competition renderings of the Turku Fuuga new concert hall, in 2021 © PES-Architects

At the end of this very early design phase, the intimate shoebox typology developed from the Stavanger and Fuzhou precedents offered a very promising concept, with considerable scope for geometric optimisation and the potential to create a concert hall with truly outstanding acoustics.

3 Detailed geometrical optimisation process

3.1 Solid angle analysis – how far should the geometry be optimized?

With so much potential for optimising early reflection coverage, one of the first concerns at the beginning of the detailed design phase was whether there might be a risk of over-optimisation. Could excessive amounts of early reflections generate an overly loud sound, unable to cope with the power of a full symphony orchestra? Wouldn't extensive geometric optimisation "consume" a significant proportion of the emitted sound energy in the early reflections, at the expense of a late reverberant field that would become too weak? These concerns are generally analysed by means of predictions with geometrical acoustic software. An alternative approach was developed some years ago, precisely with the aim of answering this type of question [19]. It is based on a solid angle analysis of the geometry in which the total energy emitted by an omnidirectional sound source on stage is subdivided into 4 parts. A first part of the total emitted energy is directed towards the audience to produce a direct sound at each seat. Once received by the audience, the acoustic energy is mostly absorbed. A second part is directed towards absorptive surfaces other than the audience, and towards reflective surfaces that will send their reflections towards absorptive surfaces other than the audience. A third part of the energy is directed towards reflective surfaces that will generate early reflections towards the audience (of 1st or 2nd or higher order). These reflective surfaces are named "efficient surfaces". This part of the acoustic energy is once again mostly absorbed in the process. The remaining part of the emitted acoustic energy will contribute to the late energy. Obviously, this approach deliberately ignores what each audience member will experience at their specific location in the room to focus on the global room acoustic behaviour. In a given room, optimizing the orientation of a surface to make it efficient will increase the amount of early energy received by some audience members. Early efficiency then relates to the average over the entire audience of the early-reflected energy.

Each of the 4 parts of the total acoustic energy emitted by the source can then be expressed geometrically by subdividing the entire space around the source in 4 solid angles:

- The direct solid angle Ω_{dir} that the audience surfaces subtend at the point of the source.
- In case absorptive surfaces other than the audience exist in the hall, Ω_{abs} can be defined as the solid angle that these absorptive surfaces subtend at the point of the source. Reflective surfaces sending acoustic energy towards these absorptive surfaces are also to be included in the estimation of Ω_{abs}.
- The efficient solid angle Ω_{eff} is defined as the solid angle of all efficient surfaces generating early reflections from the source point towards some receiving plane(s). Under the assumptions of geometrical acoustics, it can be estimated using a simple raytracing algorithm specifically designed to analyse the geometry of the hall and identify the zones of each surface that effectively receive energy from the source and redirect it towards the audience after one or more reflections, with a delay inferior to 80ms.
- The solid angle Ω_l containing the energy that contributes to the late part of the room response is simply obtained as the remaining solid angle: $\Omega_l = 4\pi - \Omega_{dir} - \Omega_{abs} - \Omega_{eff.}$ (4π being the solid angle of the entire space seen from the source).

Simple formulas then provide estimates of the average values of early-reflected strength G_{em} and late strength G_{lm} over the entire audience (possibly also including the stage platform).

Early-reflected strength is obtained from three parameters characterizing the geometry of the hall: the efficient solid angle Ω_{eff}, the total surface area occupied by audience or musicians S_{aud}, and a specific average value of the angle of incidence of early reflections on audience planes θ_m:

$$G_{em} = 20 + 10.\log\left(\Omega_{eff}\right) - 10.\log\left(\cos\left(\theta_m\right)\right) - 10.\log\left(S_{aud}\right)$$

θ_m is defined from the individual angles of incidence θa_i (0° for normal incidence, 90° for grazing incidence) weighted by the individual efficient solid angle $d\Omega_i$ of each reflector:

$$\frac{1}{\cos\left(\theta_m\right)} = \sum \frac{d\Omega_i}{\cos\left(\theta a_i\right)} \Big/ \sum d\Omega_i$$

The formula for the average value of late strengh across the audience is derived from statistical acoustic theory:

$$G_{lm} = 10.\log\left(31200.(1-\beta)\frac{T}{V}\right)$$

With $\beta = \frac{\Omega_{dir} + \Omega_{abs} + \Omega_{eff}}{4\pi}\alpha_a$

α_a is the average absorption coefficient of the audience and other absorptive surfaces in the room, and T and V are respectively the reverberation time (in seconds) and volume (in cubic meters) of the room.

Once implemented, this geometrical analysis process can quickly be repeated for various source positions to identify possible inhomogeneity in the orchestra sound caused by the geometry, or for each audience zone separately to detect if some areas are less well served than others.

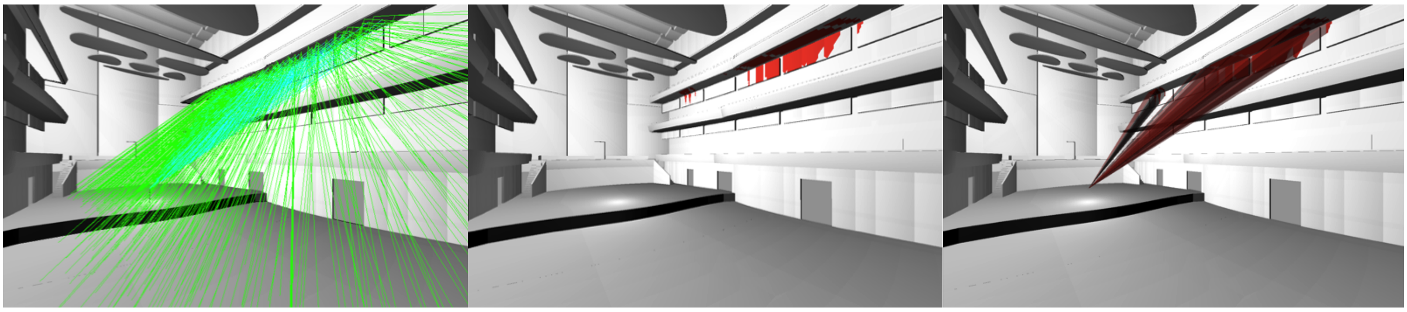

Figure 4: Example of early efficiency analysis in a 3D model of Stavanger concert hall. The picture on the left displays the results of a ray-tracing algorithm on the 2nd side balcony soffit (acoustic rays in green). The middle picture displays the corresponding efficient surfaces in red, and the picture on the right the corresponding cones representing the individual efficient solid angles for this cornice. When such an analysis is performed on the full geometry of a concert hall (which is not the case here) it needs to consider higher order reflections to properly identify all possible early reflection paths (typically up to 4th order). By definition, the corresponding efficient surfaces are those receiving direct sound from the source, even if other surfaces are involved in higher order reflections. This enables to estimate the proportion of the total energy emitted by the sound source that is used to generate early reflections as the solid angle of these efficient surfaces divided by the total solid angle for the entire space (4π)

This solid angle approach has a few advantages that proved very useful in this case. First, the effects of surface curvature are fully taken into account, as long as the raytracing algorithm is devised for that purpose (which can relatively easily be implemented in a software such as Rhino3D). Second, the geometrical analysis outcome can be visualised, providing useful explanations of the obtained results and how they could be improved by altering the geometry. Finally, the question of the extent to which a specific geometry is optimized for early reflections is directly answered through proportions of the total solid angle 4π, allowing the balance between early and late energy to be easily grasped.

Two main conclusions could be drawn from the application of this method to a preliminary version of the Turku concert hall: Overall, the geometry at that point was not overly optimised for early reflections with an efficient solid angle Ω_{eff} of 1.25 sr (just under 10% of the entire space seen from the source, sr = steradians, the SI unit for solid angles) while the late solid angle Ω_l is 4.15 sr (about one third of the entire space). The risk of excessively weakening the room's late response was then not very significant at this stage. It was also observed that the early reflection coverage was not sufficiently homogeneous, with the balconies receiving much less early energy than the parterre. Measures needed to be taken to improve the reflection coverage of balconies, while that of the parterre could be slightly reduced.

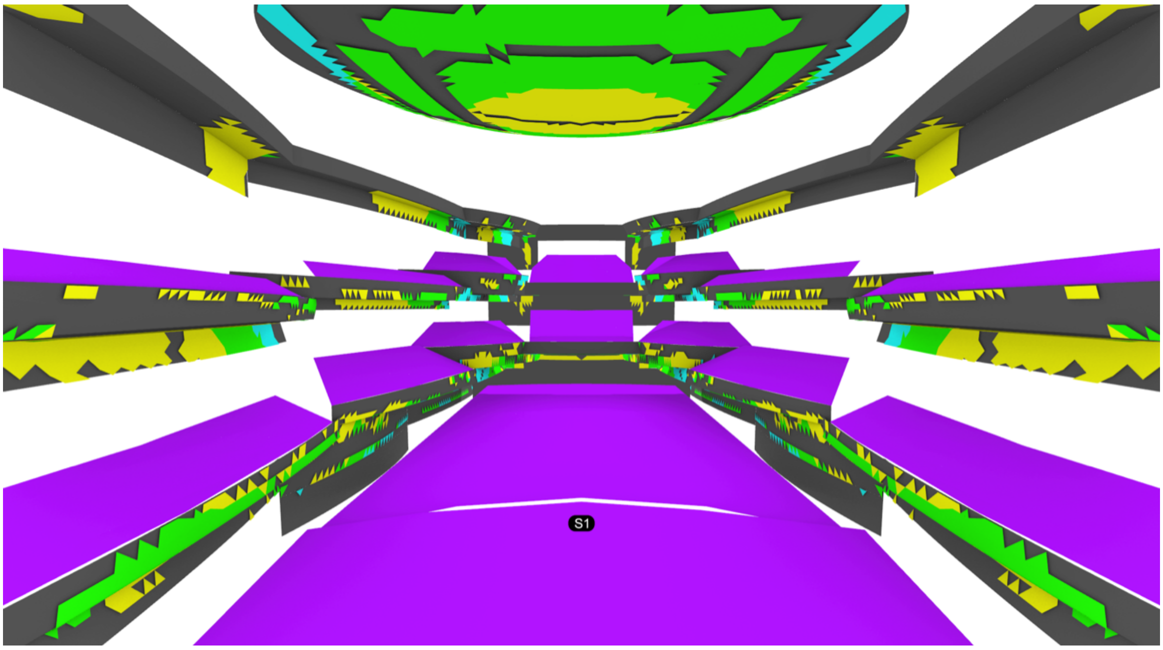

Figure 5: Visual output of a solid angle analysis performed in Rhino3D on an early version of Turku concert hall geometry, for a sound source S1 located near the conductor's podium. Receiving areas (audience and stage) are in purple, reflective surfaces in dark grey, efficient surfaces found by the algorithm are in 3 different colours depending on their delay: cyan (0 to 20ms), green (20 to 50ms) and yellow (50 to 80ms). Some of the reflective surfaces are not included in the analysis, either because they are not expected to take part in early reflections or because they were not yet well defined at this point of the project (such as the upper walls and the ceiling). Assumptions then had to be made on the additional amount of efficient surfaces and efficient solid angle that will be obtained from these missing surfaces.

3.2 Adjusting the detailed shape of potentially efficient acoustic surfaces

In contrast to the case of Fuzhou concert hall, it was very early on decided to limit the use of diffusive surface texture in the design of Turku concert hall. The only exception to this exclusion rule is the small concave surface areas connecting the convex balustrades of adjacent balconies (visible on figures 6, 9 and 14). The reason for excluding diffusive treatments is twofold: First, as previously discussed, the room concept allows precise tuning, with the possibility of reducing the energy of reflexions at high frequencies and increasing their spatial spread, when needed and in a very controlled manner, by adjusting surface curvature. Rather than relying on stochastic diffusion that would spread the high frequency content of incident sound in all directions with no distinction, it is possible to make decisions on a case-by-case basis. Second, the provision of early reflections with limited temporal smearing has been shown to offer decisive advantages in terms of acoustic clarity [20].

Going through the precise geometrical optimisation of each surface would go beyond the scope of this paper, but a few interesting examples can be discussed here. All raytracing figures in this paper were obtained with a Grasshopper script within Rhino3D environment. A differential raytracing algorithm [17] is used to estimate the strength of the reflections off curved surfaces based on the parameter ΔL_{curv}, quantifying the influence of surface curvature: a positive ΔL_{curv} represents amplification of sound energy due to the curved surface, whereas a negative ΔL_{curv} denotes attenuation of sound energy due to the curved surface.

Figures 6 and 7 illustrate how additional early lateral reflections were provided towards the seats on 1st balcony level. In Figure 6, a portion of the sidewalls is tilted and curved both in plan and section. The tilt angle and both radius of curvature in two directions were adjusted to ensure that most of the possible source positions on stage generate the intended reflections, and that late reflections are not excessively strong.

In Figure 7 the shape of a balcony front surface is adjusted to provide early reflections to the 1st balcony. In addition to a slight curvature in plan and section, the surface is tilted with an angle that varies along its length. This leads to a complex warped (non-developable) shape that will be precisely built by carving it out of massive wood.

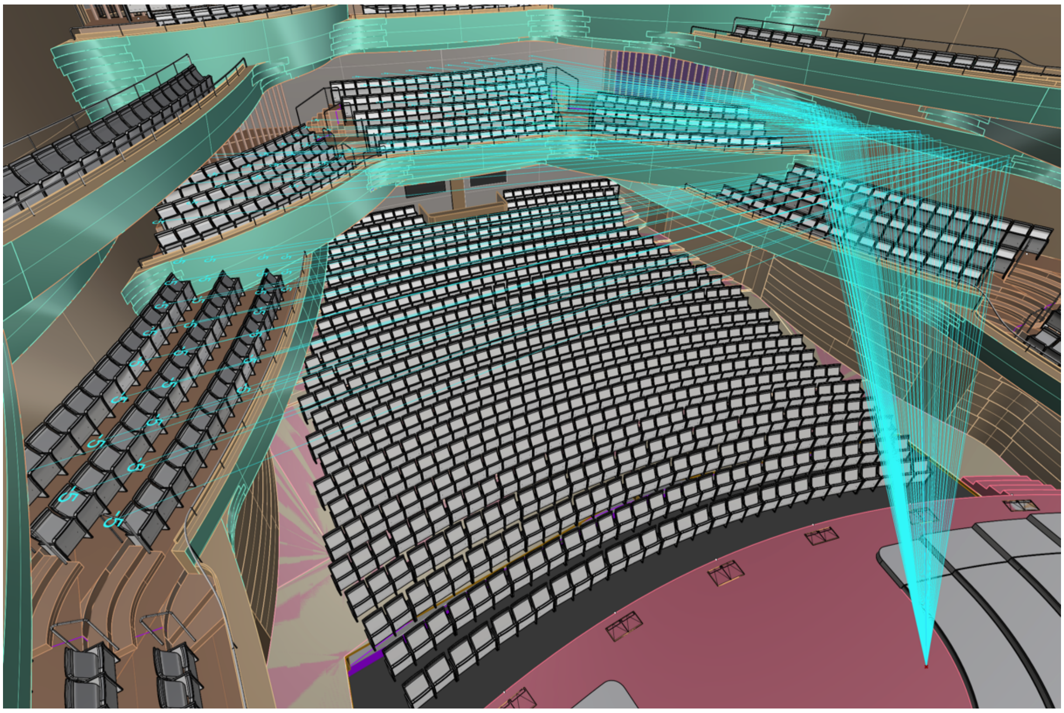

Figure 6: Raytracing on a convex and inclined wall section located on the sidewalls between 1st and 2nd balcony levels. These generate early lateral reflections towards the seats in the rear sections of 1st balcony level, and relatively late lateral reflections towards the seats in this same balcony level but located closer to the stage. ΔL_{curv} = -5dB.

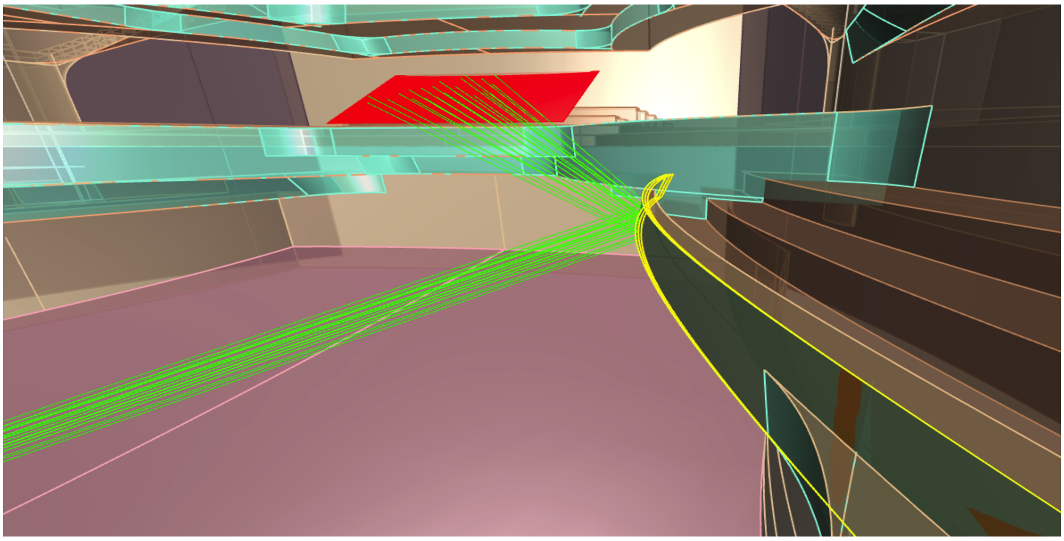

Figure 7: Adjusted shape of the 1st side balcony front to provide early lateral reflections towards the 1st level rear balcony. The balcony surface is intentionally warped.

Figure 8 shows a case of cornice reflection optimisation. Traditional cornice reflections in shoebox concert halls are 2nd order reflections generated by two adjacent surfaces: a vertical wall and a horizontal soffit or ceiling. The cornice reflection illustrated on Figure 8 is generated by the flat and perfectly horizontal soffit under the technical gallery level, together with a vertical wall portion (or downstand beam) whose shape can be freely adjusted. As all wall surfaces in this design, the downstand is convexly curved in plan. An additional convex curvature in section was introduced to provide more homogeneous coverage and to reduce reflection intensity by 3dB. The original cornice reflection with a single-curved downstand had ΔL_{curv} values ranging from -1 to -2dB, while the optimized cornice reflection has ΔL_{curv} values from -4 to -5dB.

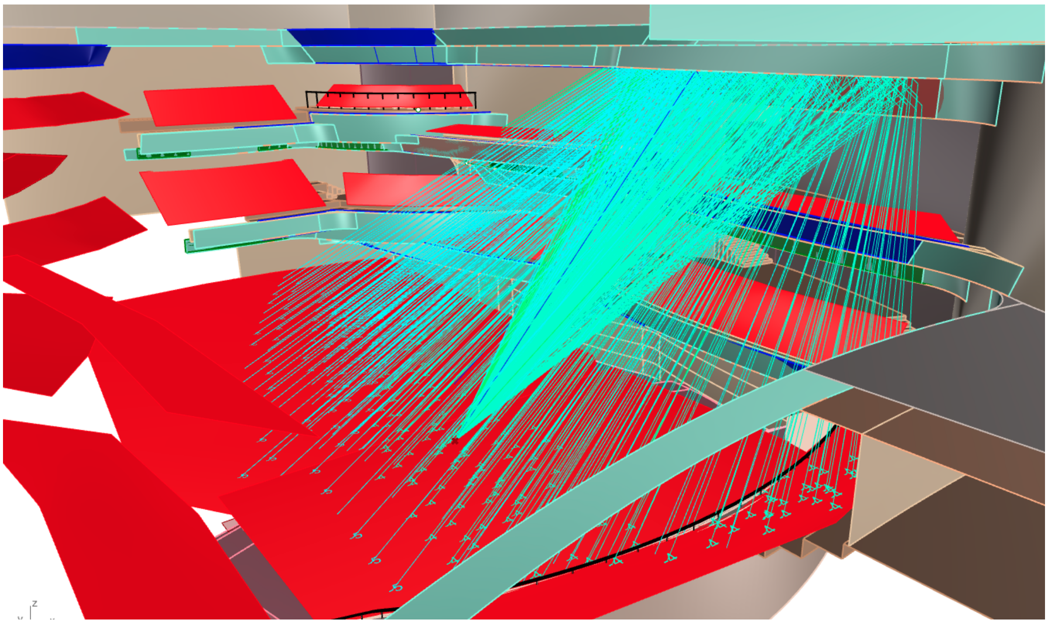

Figure 8: Adjusted shape of the downstand beams located on each side of the stage, under the technical gallery level. In addition to the convex curvature in plan, a slight curvature in section was introduced to provide more homogeneous coverage and reduce reflection intensity by 3dB.

3.3 Geometrical optimisations for stage acoustics

A significant part of the geometrical optimisation effort was dedicated to providing good listening conditions for the musicians on the stage platform.

The canopy reflector was initially designed as a monolithic warped disc, later split into 5 bands for the integration of stage lighting. The general disc shape is kept in plan view, but the surface is intentionally warped to vary the local radius of curvature both in long section and short section. This allows controlling the acoustic strength of the early reflections generated, with stronger reflections from instruments at the front of the stage and near the room axis (from parts of the canopy with a more flat shape), and weaker reflections from the sides and the rear of the stage (from parts of the canopy with a more pronounced convex curvature). The general tilt of the canopy projects sound from the stage towards the audience, while its rear part also allows reflections from the choir balcony towards the conductor. Each of the 5 bands is then once again very slightly curved in short section to avoid shadow zones in the reflection coverage for high frequencies.

The first round of acoustic predictions using Odeon software in the concept design phase gave fairly inhomogeneous results regarding the stage support ST1 (ST_{early}) parameter. Values varied from excessively high (> -12dB) to excessively low (< -18dB) depending on source position on stage. Reasons for such results had to be investigated. In the design, all surfaces generating early reflections back the orchestra are curved, some of them with a complex warped geometry. They all needed to be approximated and facetted to comply with requirements of an Odeon model. A specific verification process was developed in order to estimate the support parameter directly within Rhino3D, from the exact shape of the surfaces surrounding the stage.

In this process, a partial ST1 is estimated by summing the energy of all acoustic reflections (1^{st} order and 2^{nd} order) arriving between 20ms and 100ms after direct sound from a point source to receivers at 1m distance. As this method does not take into account edge diffraction and higher order reflections, the estimation is incomplete. The parameter ST1# (partial ST1) is then used instead of the standard ST1.

For each reflective surface, a colored contour corresponding to the acoustic coverage of this reflector is computed using a raytracing algorithm. If a receiver lies within this contour, a reflection path exists between the source and the receiver via the corresponding surface, as shown in figure 10. The algorithm will then provide a ΔL_{curv} (in dB) corresponding to the change of reflection strength due to the curvature of the surface [17], and a ΔL_{diffr} (in dB) corresponding to the attenuation of the reflection due to diffraction and the finite size of the surface [21].

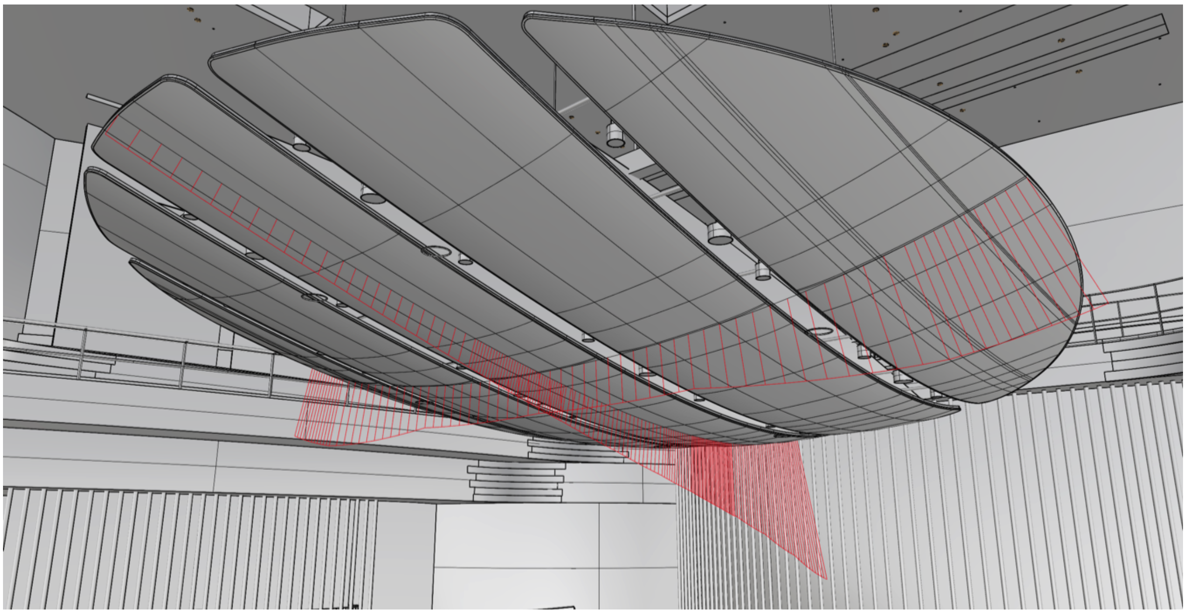

Figure 9: The canopy is shaped after a disk and convexly curved in two directions, with varying radii of curvature in both short and long sections. Lines in red are the curvature graphs of the general shape in each direction. Local surface normals are plotted, with a length inversely proportional to the local radius of curvature: longer surface normals on the sides and in the rear part of the canopy (in the direction of the organ) indicate shorter local radii. The resulting shape is warped and highly complex, but the absence of any locally concave zones needed to be precisely checked. Corresponding ΔL_{curv} values for reflections generated on stage and in the audience range between -8dB and -5dB.

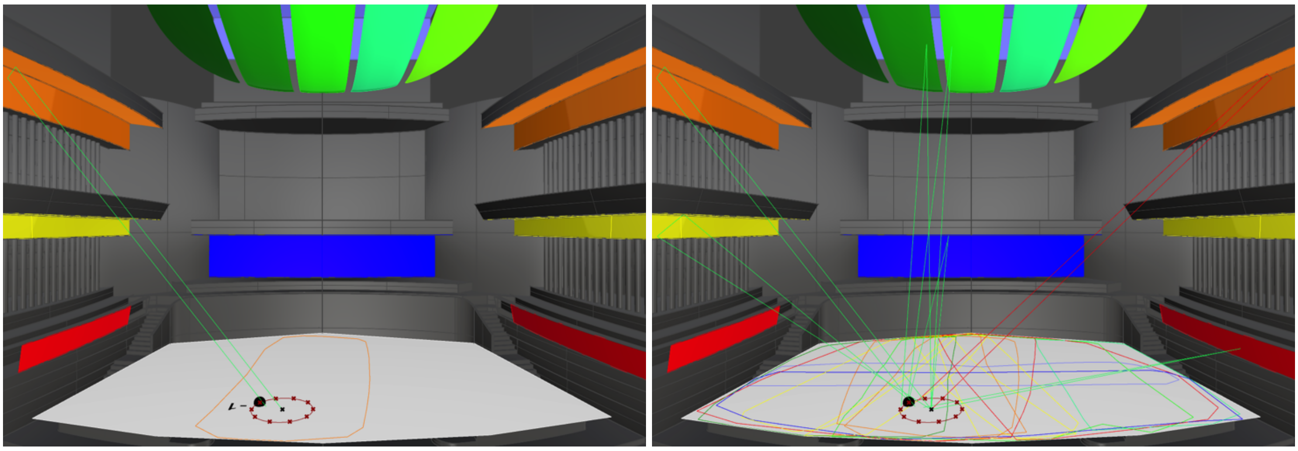

Figure 10: Left picture: acoustic reflection path via the upper left soffit (orange) from the source (center of the red circle) to one receiver at 1m (black dot). The coverage zone is indicated as an orange contour, and the calculated ΔL_{curv} is marked near the receiver (-1dB). Right picture: all possible acoustic reflection paths from the source to the same receiver, as used for ST1# calculation. Green rays have a 20ms to 100ms delay, and red rays have a delay > 100ms. In this example, reflections from the stage back wall and the organ box are not considered.

To finally obtain a spatial average value of ST1# around a given source position, this process is repeated for 10 receivers equally spaced on a 1m radius circle around the source. Results for 4 source positions are given in figure 11.

In the first simulation run, excessively high ST1# values were obtained for sources at the back of the stage. This result could be traced back to several surfaces providing early energy to this source location. Among others, the shape of the balcony front in red in figure 11 was modified to send more early energy to the front part of the stage and less energy to the rear part. This led to a twisted shape with a varying tilt angle, ending vertically towards the back of the stage. A second run of simulations was performed with adjusted reflectors and the obtained results can be compared in figure 11.

Interestingly, the inhomogeneity identified and solved using this ST1# analysis process differs from the one initially observed in Odeon simulation results. The subtle changes in curvature and tilt angle, decided on the basis of this ST1# analysis, have no significant implications for the faceted version of the geometry used by Odeon.

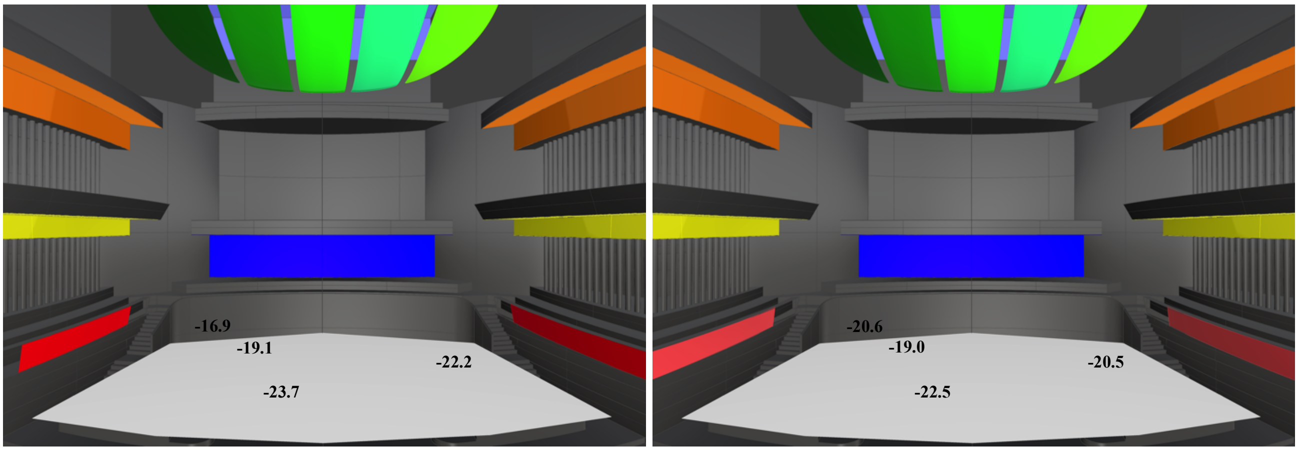

Figure 11: Estimated values of ST1# for 4 different source positions before (left) and after (right) surface optimisations. On the left, ST1# is found to be excessively inhomogeneous on stage (6.8 dB difference between extremes). On the right, after shape optimisation a more homogeneous ST1# is obtained (3.5 dB difference between extremes).

It is sometimes thought that late arriving acoustic reflexions towards the stage are dangerous and may cause disturbing echoes for the musicians. However several authors have proven that musicians need a sufficiently audible late acoustic feedback from the hall [22], [23], [24], which the present authors confirm on the basis of their experience. This desirable late acoustic feedback is ideally made up by several late reflections spread over time and arriving from different directions in space. If each individual late reflection is not excessively strong, it will not be perceived as a distinct event or "echo", but will rather play its part in building up a smooth halo of sound. In the acoustic design of the Turku concert hall, it was therefore necessary to make sure that such a desirable acoustic feedback was effectively provided, with sufficiently audible effect and without excessively isolated and distinct events that would be perceived as echoes. To do so, all relatively late reflections (from about 50ms) generated by the concert hall geometry were listed with their individual characteristics (delay, strength, direction of arrival). The level of each reflection was calculated, taking into account attenuation effects due to curvature and diffraction on finite size objects, and normalized to the level of direct sound at source-to-receiver distance of 1m. Graphs were built with this data, as displayed in figure 12.

In these graphs, three echo thresholds available from the literature ([10], [25]) are indicated as coloured lines in order to assess the risk of echo disturbance, as well as the risk that some reflections are too weak to contribute to a positive acoustic feedback to the musicians. It must be stressed that all these echo thresholds were obtained from listening tests in which a single reflection was added to an anechoic environment, which is a very different situation compared to that of a musician on a real concert hall stage. More recent studies [26] highlighted these limits, and also found that the echo threshold for rhythmical music from Dietsch and Kraak [25] (red curves in figure 12) is too high for some musical instruments such as the trumpet. Echo threshold curves for speech from Dietsch and Kraak (orange curves in figure 12) appear to better relate to what is perceived with such echo-critical music instruments.

This echo-analysis procedure was repeated for 3 different source positions on stage (soloist position, woodwind position and percussion position), and for the 1kHz and 2kHz octave bands (as diffraction effects depend on frequency). Two feedback reflections were detected as possibly being excessively distinct in a preliminary version of this analysis, which led to adjustments to the geometry at the rear of the parterre (concave wall) and in the upper wall-ceiling corner above the 2nd rear balcony (87° angle changed for a 90° angle between the rear wall and the ceiling, see figure 13). The analysis in figure 12, corresponding to the final, corrected version, gives confidence concerning the result that will be achieved. Reflections remain in the vicinity of the speech echo threshold by Dietsch and Kraak, and below the extended echo threshold from Barron, meaning they are likely to be audible without generating an excessively distinct echo.

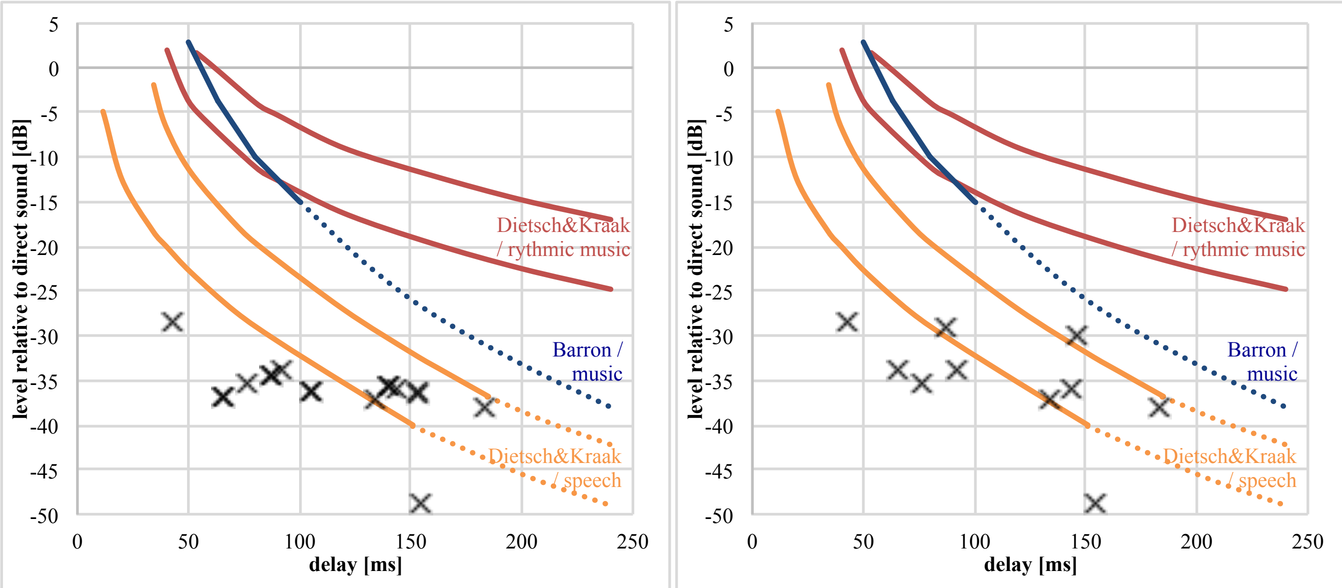

Figure 12: Echo graph for a source located at a soloist position on stage near the conductor, and for the 1kHz octave band. Each reflection is represented by a black cross corresponding to its relative level and delay with respect to direct sound at 1m distance to the source. Red and orange lines correspond to empirical echo thresholds found by Dietsch and Kraak respectively on a rhythmical music excerpt or a speech excerpt [25]. The blue line corresponds to the threshold for disturbing echo found by Barron with a music excerpt [10]. Dashed lines are regression lines extending the original experimental data to longer delays. On the left, a distinct cross represents each individual reflection, while on the right reflections arriving with very similar delays and directions are grouped and their level is aggregated.

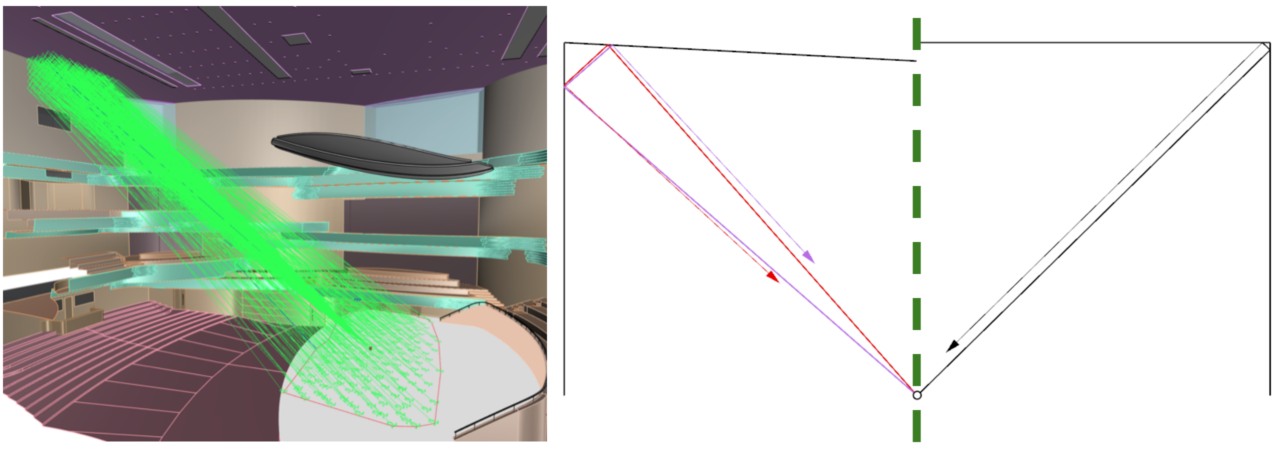

Figure 13: Left picture: Raytracing diagram showing the path for 2nd order reflections off the corner between the rear wall and the ceiling. Right picture: 3D diagram showing the difference between a 90° rear corner reflection (in black, to the right of the source point) and twin reflections from a 87° rear corner (in red and purple, to the left). The main ceiling was initially angled at 3° to the horizontal right up to the rear wall, generating an angle of 87° between the two surfaces instead of the usual 90° corner. As a consequence, twin feedback reflections were generated instead of a single one, as two reflection paths exist for rays originating from and ending at the source: a first path in which sound waves first hit the ceiling, then the wall before returning to the source, and a second one in which sound waves first hit the rear wall, then the ceiling before returning the source. A 3dB level increase can be estimated when such twin reflections are generated, compared to the typical case of a 90° rear corner for which only one reflection path exist. In the final design the rear part of the ceiling was finally set to perfectly horizontal to avoid that 3dB level increase.

4 Discussion on last acoustic predictions before completion

At the time of writing, the Turku Fuuga music centre is taking shape on its building site and it is too early to judge the acoustic success of the concert hall. The latest available version of the Odeon model is dating from October 2023, while a final version is still in progress. Small changes have been implemented in the meantime, but they are not expected to have a strong impact on the general average values of predicted acoustic parameters. The Akukon team performed all Odeon simulations for this project: Sara Vehviläinen, Perttu Laukkanen and Henrik Möller.

Main parameter values are given in table 1 and are compared to the measurement results in the built Stavanger concert hall, from [5].

As can be observed, obtained values are generally very similar. It seems that Turku concert hall will sound even slightly stronger (with a G value that is 0.8 dB higher on average) and clearer (with a C80 value that is 0.9 dB higher on average) than the Stavanger concert hall. This corresponds perfectly to the wishes of the representatives of Turku Philharmonic Orchestra. It is worth highlighting that this increase in C80 compared to Stavanger is obtained while keeping the same high value of occupied reverberation time. The two halls also stand out for their very high average LF values.

Tapio Lokki and his team at Aalto University also used this Odeon model to generate auralizations of Turku concert hall as predicted by the model. Comparison with a calibrated Odeon model of Vienna Musikverein was available. The listening tests results are very promising, with a general preference for Turku that – in the context of these auralizations – was found to provide both clearer and more enveloping sound when compared to a similar seat in Vienna Musikverein.

In addition, the solid angle analysis was updated based on the latest version of the architectural 3D model (with non-faceted curved surfaces). A significant increase of efficient solid angle was obtained in comparison to the initial design discussed in paragraph 3.1, with a Ω_{eff} of 1.61 sr (13% of the entire space seen from the source, and +29% compared to the initial design). This increase of efficient solid angle implies a slight reduction of the late solid angle Ω_l that is now 3.96 sr (31% of the entire space seen from the source, and -5% compared to the initial design). This results in a predicted increase of average early-reflected strength (G_{em}) of 1.1dB and a corresponding decrease of average late strength (G_{lm}) of 0.2dB. Homogeneity was also drastically improved compared to the initial state, with an average increase of G_{em} in balcony seats of 3.8dB. These results provide a good quantitative summary of the overall impact of several months of geometrical optimisation for the Turku concert hall, and illustrate the possibility of simultaneously achieving very clear acoustics with generous reverberation.

Table 1: Provisional Odeon simulation results in Turku concert hall compared to measurements results in the built Stavanger concert hall. Each parameter value is the average of all receiver position located more than 10m to the source, distributed evenly in each room. The acoustic configuration with maximum RT is considered in both cases (symphony orchestra setting, as both halls have variable acoustics). Parameters are defined, measured and frequency averaged according to ISO 3382.

Occupancy	**Acoustic parameter**	**Turku concert hall Odeon simulation results**	**Stavanger concert hall Measurements**
Full audience	T30 (s)	2.2	2.2
Unoccupied	G (dB)	5.0	4,2
	G80-∞ (dB)	2.3	1.9
	G0-80 (dB)	1.5	0.3
	C80 (dB)	-0.7	-1.6
	LF (%)	0.27	0.30

Figure 14: Architectural rendering of the main concert hall of Turku Fuuga music centre, in its latest version at the time of writing. © PES-Architects

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Finite-difference time domain calculations of acoustic phenomena applied to concert hall design

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The use of wave-based methods for the simulation of the acoustical field inside rooms was until recently far away from the widespread use of ray-tracing/image source algorithms. Although new commercial software has recently started to use this approach for room acoustic simulations, full spectrum analysis still faces difficulties. Nevertheless, the interest in obtaining a clear understanding of diffraction and diffusion phenomena, pushed the development of more efficient computational implementations. Precisely because diffraction is hardly simulated using ray-tracing algorithms, the study of the interaction of sound waves with coupled volumes, rigid suspended or "floating" objects in a room benefits from the use of wave-based methods. In a concert hall, theatre, or auditorium these objects include fixed or suspended balconies, reflectors or canopies, or even volume diffusers. During the design of a 1200 seat multi-purpose auditorium, the problem of how to distribute and orientate a series of objects that create a simple type of volumetric diffuser close to a wall in a concert hall required the implementation of a finite-difference time-domain algorithm. This study focused on parametric changes in the characteristic dimensions and shape of these elements and in their relationship with the boundaries of the room. The results show how variations on these geometrical characteristics influence the temporal and spatial distribution of sound wave reflections. Apart from the simulation of impulse responses that could enlighten how reflections occurred it was important to implement animations that helped visually convince the architect of the need for relevant changes.

1 Introduction

The use of wave-based methods for the simulation of the acoustical field inside rooms has only recently started to be implemented in commercial software aiming for such a widespread use as ray-tracing/image source algorithms. Some of these new wave-based software implementations (see [1]), still need hybridization with geometrical methods to ensure feasible computation times. Solving the wave-equation on a 2D and particularly 3D closed domain is a challenging process, but the interest in obtaining a clear understanding of diffraction and diffusion phenomena, without using an overly simplified approach as in ray-based methods, pushed the development of more efficient computational implementations using parallel high-performance computing clusters with graphics processing units (GPUs), as in [2] and [3]. Additional difficulties result from lack of data on material acoustical characteristics that can be used with this approach, amongst other factors. However, it is undeniable that current research on this field, as in [4] or [5], has allowed this methodology to become more accessible, not only to researchers but also to practitioners like acoustical consultants (see [2], for example).

Precisely because diffraction is hardly simulated using ray-tracing algorithms, the study of the interaction of sound waves with rigid suspended or "floating" objects in a room benefits from the use of wave-based methods. In a concert hall, theatre, or auditorium these objects include fixed or suspended balconies, reflectors, canopies, or even volume diffusers [6]. In some of the author's room acoustics projects, the problem of how to distribute and orientate surfaces or objects that create useful reflections required the implementation of a 2D finite-difference time-domain algorithm [7]. Apart from the simulation of impulse responses that could enlighten how reflections occurred it was important to implement animations that could help visually convince the architect of the need for relevant changes. This paper shows some of the results obtained from these simulations, discussing the interesting aspects of a 2D detailed approach, not intended to calculate the full room acoustics parameters, only possible with 3D calculations, but highlighting the characteristics of the resulting wave interference phenomena.

2 Methodology

The Finite Difference Time Domain (FDTD) method is a wave-based numerical technique for solving the wave equation. This technique is suitable for the simulation of wave propagation in a sound field, particularly due to the fact that it is a meshed based algorithm, in which the computational domain is discretized into a grid of finite cells. When applied to acoustic field simulation, the FDTD method offers several advantages such as:

- Computationally efficient for acoustic simulations, especially when compared to some other methods like finite elements or boundary elements, and its straightforward implementation.
- It can handle complex geometries and boundary conditions, and, like other methods, it inherently models acoustic phenomena such as wave interference and diffraction.
- Gives accurate solutions to the acoustic wave equation, particularly near the specular reflection angle.
- FDTD allows for the real-time visualization and analysis of the simulated sound field, as it provides detailed temporal and spatial information.

One of the main disadvantages of the FDTD method is that the grid resolution limits the geometry of the reflective surfaces and the maximum frequency that can be represented. As the grid resolution of the domain increases, so does the solution time, in a way that can be computational unfeasible to accurately extend the accurate frequency range of the solution. Additionally, like many other numerical methods, FDTD requires appropriate boundary conditions to accurately model wave propagation at domain boundaries, which can be difficult to implement [8].

2.1 Numerical implementation

To model the resulting reflections of an incoming wave on a vertical surface, anechoic boundary conditions for the domain were implemented, so that no interference would arise from the propagating wave being reflected on the boundaries of the mesh, altering the resultant sound field. As such, these reflected waves near the domain's boundary were eliminated through the use of perfectly matched layers, or PMLs, which gradually absorb and dissipate the outgoing waves, effectively preventing reflections from the boundaries. Due to complexity of the boundary conditions, an existing Matlab toolbox (k-Wave) was used for the time-domain simulation of acoustics fields, as in [9].

The reflective surfaces or objects are defined by a set of rigid elements, introduced in the domain by the creation of a mask of points at which the speed sound is close to zero ($1\text{x}10^{-9}$ m/s). The density of the non-reflective points in the domain is set to that of air.

The model assumes a two-dimensional scheme of FDTD, in which the central finite difference approximations to the pressure and particle velocity are implemented using the following equations:

$$p_{i,j}^{n+\frac{1}{2}} = p_{i,j}^{n-\frac{1}{2}} - \rho_0 c^2 \Delta t \left(\frac{ux_{i+\frac{1}{2},j}^{n} - ux_{i-\frac{1}{2},j}^{n}}{\Delta x} + \frac{uy_{i,j+\frac{1}{2}}^{n} - uy_{i,j-\frac{1}{2}}^{n}}{\Delta y} \right) \tag{1}$$

$$ux_{i+\frac{1}{2},j}^{n+1} = ux_{i+\frac{1}{2},j}^{n} - \frac{\Delta t}{\rho_0} \left(\frac{p_{i+1,j}^{n+\frac{1}{2}} - p_{i,j}^{n+\frac{1}{2}}}{\Delta x} \right) \tag{2}$$

$$uy_{i,j+\frac{1}{2}}^{n+1} = uy_{i,j+\frac{1}{2}}^{n} - \frac{\Delta t}{\rho_0} \left(\frac{p_{i,j+1}^{n+\frac{1}{2}} - p_{i,j}^{n+\frac{1}{2}}}{\Delta y} \right) \tag{3}$$

The chosen method of pulse excitation was the Ricker wavelet (or the second derivative of the Gaussian function), that is characterized by its bell-shaped curve with a single peak, and has a bandwidth inversely proportional to its central frequency, f. The Ricker wavelet is mathematically defined as:

$$W(t) = -\sqrt{2}\pi f_{cent} \left[\left(\sqrt{2}\pi f_{cent} t \right)^2 - 1 \right] . \exp \left[-\frac{1}{2} \left(\sqrt{2}\pi f_{cent} t \right)^2 \right] \tag{4}$$

The maximum frequency that can be accurately simulated is $f \leq c/10\Delta x$, in which Δx is the mesh step size and c the speed of sound. This study assumed a grid step of 1 cm, that results in a maximum accurate frequency of 3,4 kHz, assuming c = 340 m/s.

For this study, the objective was to study reflections close to a set of surfaces or objects. For this purpose, a simulation domain of approximately 5m x 8m was implemented. The impulse source was placed at 1,5m above the lower limit of the spatial domain (whether it included a floor or not) and an array of horizontally distributed virtual sensors were placed at 1,2m from the same limit. The source impulse excitation frequency range was limited to 1,7 kHz.

3 Simulations

Simulations were performed to analyse the effect of a stage wall with a balcony underside with different lengths, orientations, and scattering characteristics. The lateral wall is 3m high and the balcony underside varies between 0,5m, 1m and 2m, with an orientation varying from 0º to 60º in 20º steps. Figure 1 shows a snapshot of the wave propagation at the same instant for all situations for a detailed comparison of the consequences of the referred geometrical changes. The surfaces in this case are completely specular.

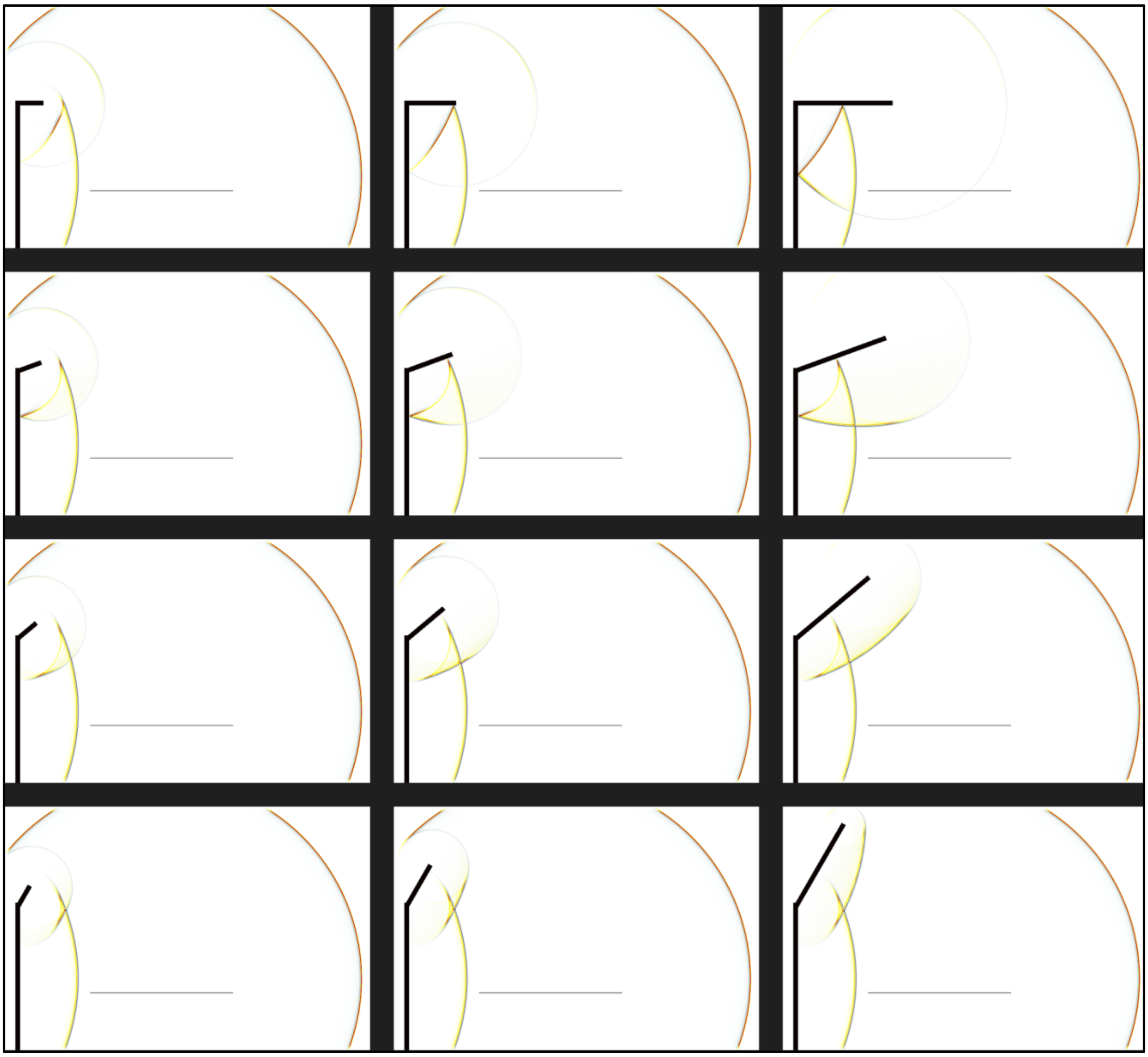

Figure 1: Specular reflections from a wall and varying lengths and orientations of a balcony underside. The horizontal thin line represents the sensor array.

In the previous figure, the inefficiency of the 0,5m depth balcony is evident with only a small fraction of the wave being reflected to the sensor array. Horizontal orientations (0º) of the balcony capture most of the useful sound energy although the 40º and 60º orientation was the most useful for the application for which the simulations were performed: to choose the right orientation of lateral reflectors for stage support and communication between different places in the orchestra. Without any other reflection surfaces in these calculations, and for purely specular reflections it was expected that a comb filter effect was created as can be seen in Figure 2, for each of the previous set of simulations presented in Figure 1.

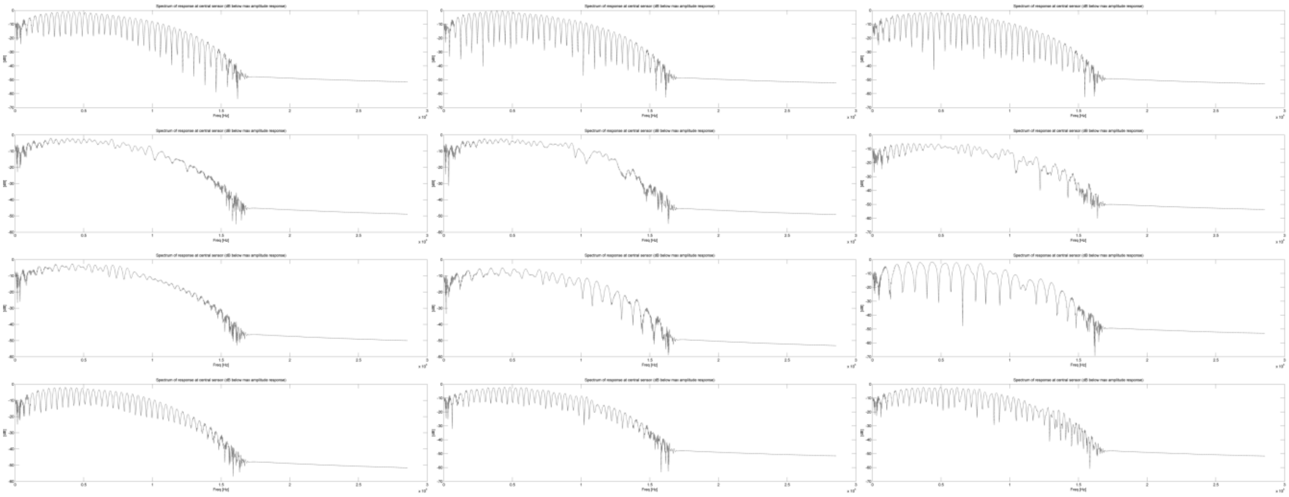

Figure 2: Sound pressure spectra at the central sensor of the horizontal array, corresponding to the wall/balcony arrangement in Figure 1.

Although the comb-filter effect is present due to the absence of further sound reflections from other surfaces, the introduction of a scattering surface was studied, by applying a Shroeder type diffusor to the vertical wall. Figure 3 presents the sound pressure spectra corresponding to the simulations presented in Figure 5.

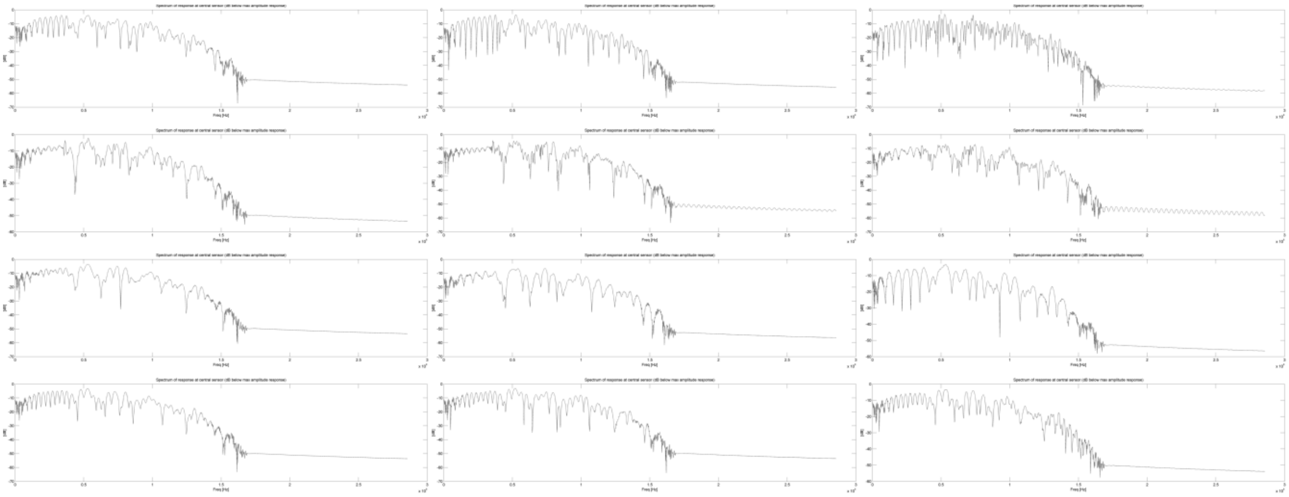

Figure 3: Sound pressure spectrum of the sound at the central sensor of the horizontal array, corresponding to the wall/balcony arrangement in Figure 1.

As expected, the efficiency of the diffusor depends on the depth and width of the size varying wells. In this case, the wells were 4cm wide, with 1, 2, and 5cm depths with a 9cm waveguide depth, as represented in Figure 4.

Figure 4: Representation of the diffusor geometry used for the vertical wall.

The diffusor is able to prevent the comb-filter effect above its cut-off frequency, distributing the reflected energy through a longer time interval, even if the balcony underside is not a diffusive surface. Further simulations were performed with additional surfaces (such as a reflecting floor) which, as expected, significantly reduced the referred comb-filter effect.

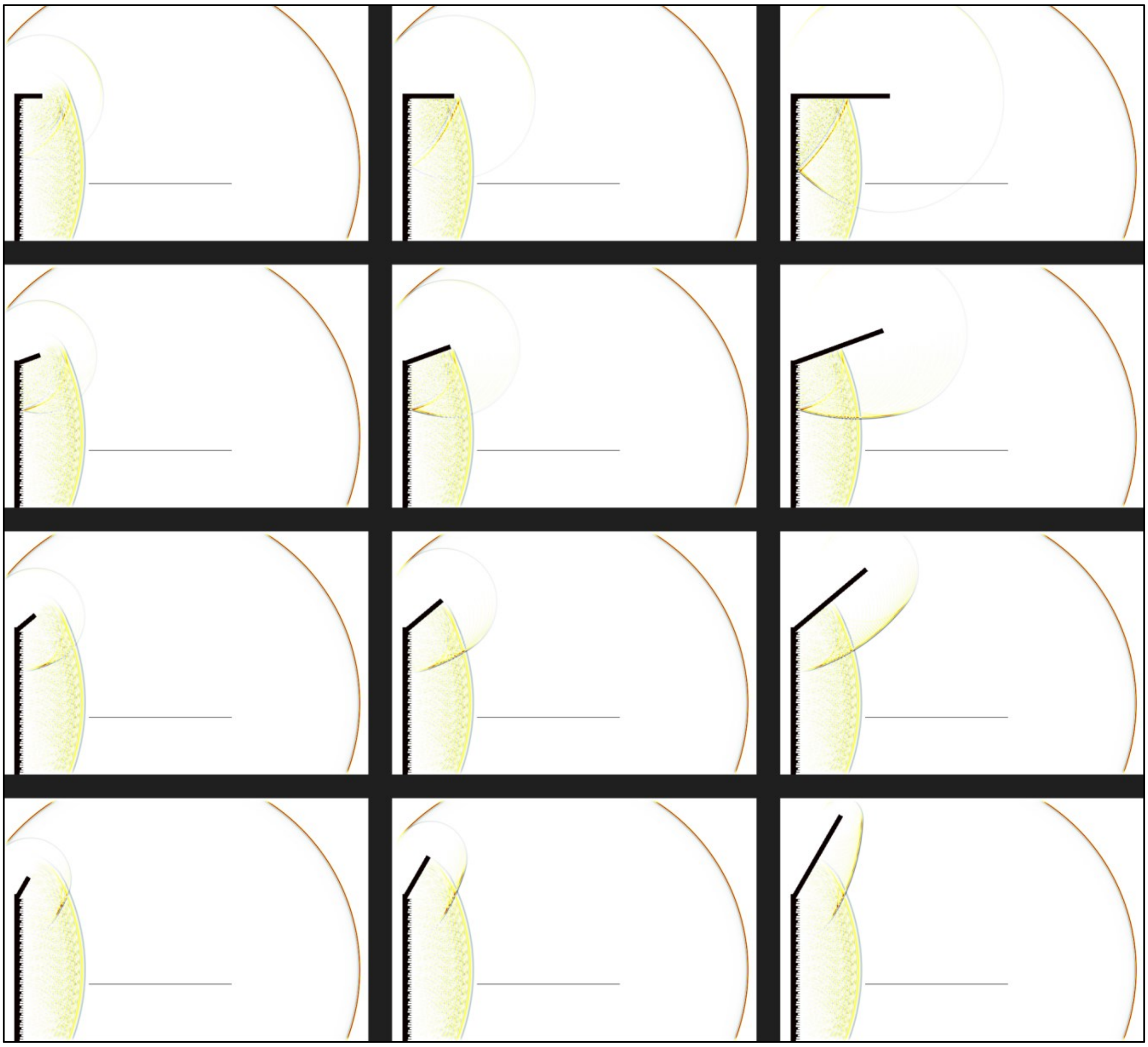

Figure 5: Diffuse reflections from a wall and varying lengths and orientations of a balcony underside. The horizontal thin line represents the sensor array.

Analysing the time-space variation of the sound pressure allows more information to choose the most optimized solution. Considering only the third column of Figure 1 and Figure 5, which considers the 2m balcony depth, the representation of the spatial and temporal impulse response depicted in Figure 6 shows how the sensor array is irradiated by the direct and reflected waves, both for the specular and diffuse reflections.

While the horizontal balcony gathers most of the energy and redirects it to the stage with an even distribution to most of the sensors, the 40º orientation demonstrated, in this case, to also be very effective returning sound to the stage both with specular reflecting surfaces and with the scattering vertical surface. The 20º orientation is ineffective except for the very first set of sensors closer to the vertical wall, and the 60º orientation was only useful for the more distant set of sensors. If the horizontal balcony is not an option from the design point of view, a set of differently oriented surfaces can allow for a more optimized energy distribution.

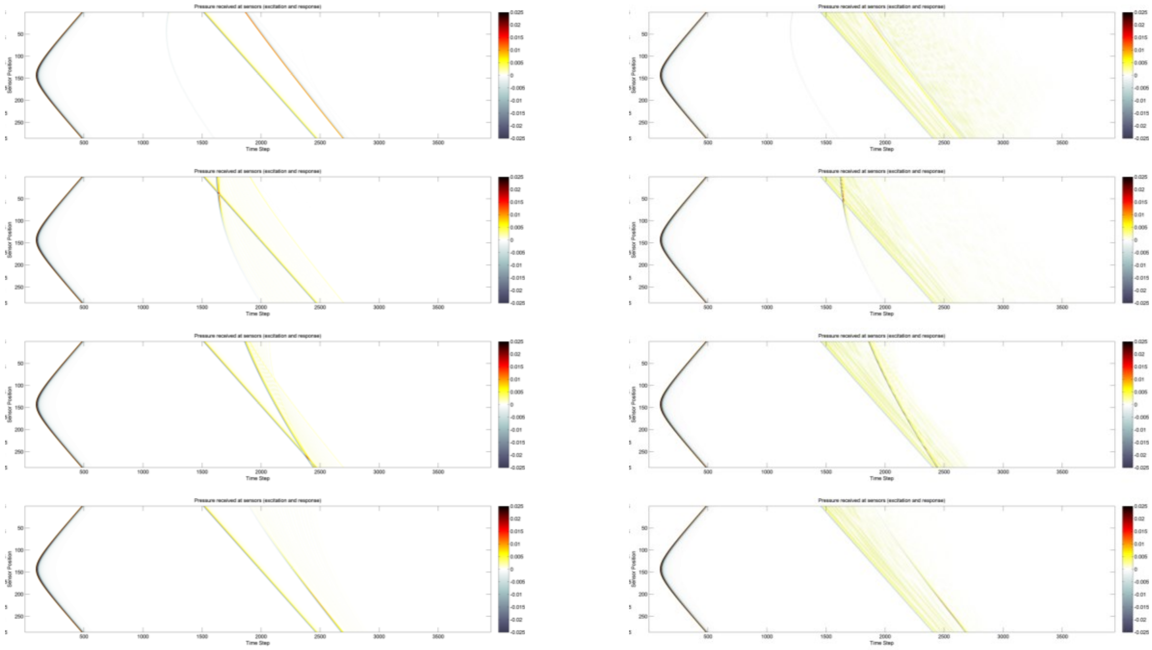

Figure 6: Spatial-temporal variation of sound pressure along the sensor array (left – no scattering; right – scattering on the vertical surface. The darker line depicts the direct sound arriving at each sensor.

4 Conclusions

In this paper the FTDT method was used to study simple cases of surfaces geometry changes for a concert hall stage. The detail that the method allows is useful to understand the distribution of sound energy both in time and space and how small variations can impact design decisions. Apart from the information presented in this paper, the animations that derive from the calculations were very important to illustrate to the design team, the importance of surface orientations.

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New Opportunities in Room Acoustics Simulations Using Wave Based Technology

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New technical developments have enabled the use of computational cloud solutions in the field of acoustics. It is therefore possible to run state of the art wave-based solvers in an efficient and cost-effective manner. In this paper we present the advantages and opportunities of using Treble's cloud-based computing service for acoustic simulations. A handful of selected validation cases will be presented with special focus on cases that are relevant to acousticians working in the building industry. They include both direct comparisons with measurements on site as well as comparisons with reference scenes from the BRAS database (Benchmark for Room Acoustical Simulation).

1 Introduction

Treble is a hybrid room acoustic simulation tool that combines accurate wave-based (WB) simulations at low frequencies, to capture the important wave and modal behaviour, and geometrical acoustic simulations (GA) at high frequencies, to take advantage of the fast calculations. Treble is the first room acoustic software that hybridizes the WB and GA methods, which has been long awaited development in the acoustics community.

The aim of this paper is to highlight the fundamentals and strengths of Treble's room acoustical simulation tool. This is achieved by comparison with a reference case from the BRAS (Benchmark for Room Acoustical Simulation) database as well as a comparison with on site measurements. The paper will also serve to highlight some of the most important aspects when simulating acoustics in a hybrid solver.

2 RS 5 – Diffraction

2.1 Setup and background

When there is an impenetrable obstacle between a sound source and a receiver, sound diffracts around the obstacle. Examples include diffraction around a noise barrier outdoors and diffraction through half-open doors. Diffraction is an important phenomenon in room acoustics that traditional energy-based geometrical acoustics methods struggle to simulate. There are some ways to add diffraction in geometrical acoustics, such geometrical theory of diffraction [1] and analytical secondary source modelling [2] etc. As previously stated Treble's wave-based solver however solves the wave equation directly which allows for more accurate simulations. In this work, a diffraction scenario is simulated in Treble, which is RS5 case in the BRAS database [3], as shown in Figure 1.

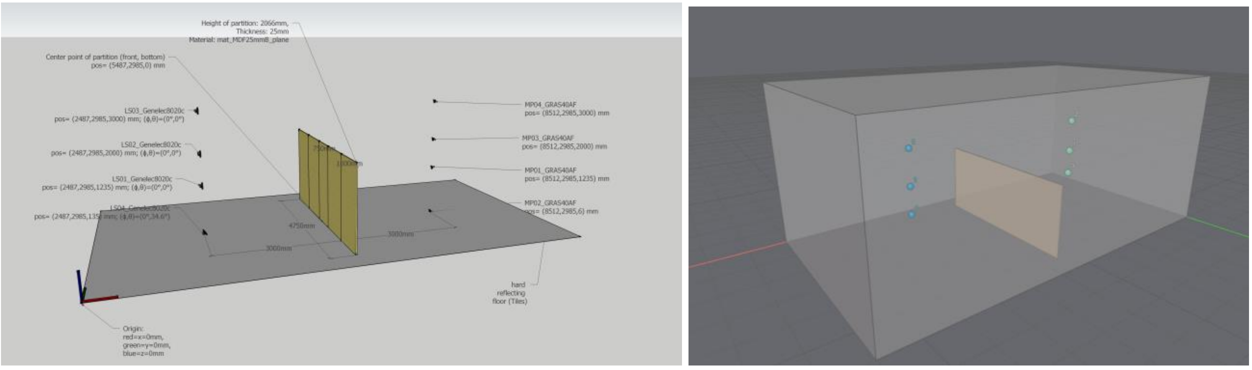

Figure 1: Setup for diffraction modelling. Top – Measurement setup. Bottom – Setup in Treble

2.2 Results

All source/receiver configurations were tested in Treble and compared to measurement results. For the LS02 and MP01 (without any line of sight), Treble and another commercial GA simulation with built in diffraction modelling are compared to the measurement as shown in Figure 2. It is clear that Treble shows a better agreement in the trough locations, which actually shows destructive interference due to diffracted wave components from the edges.

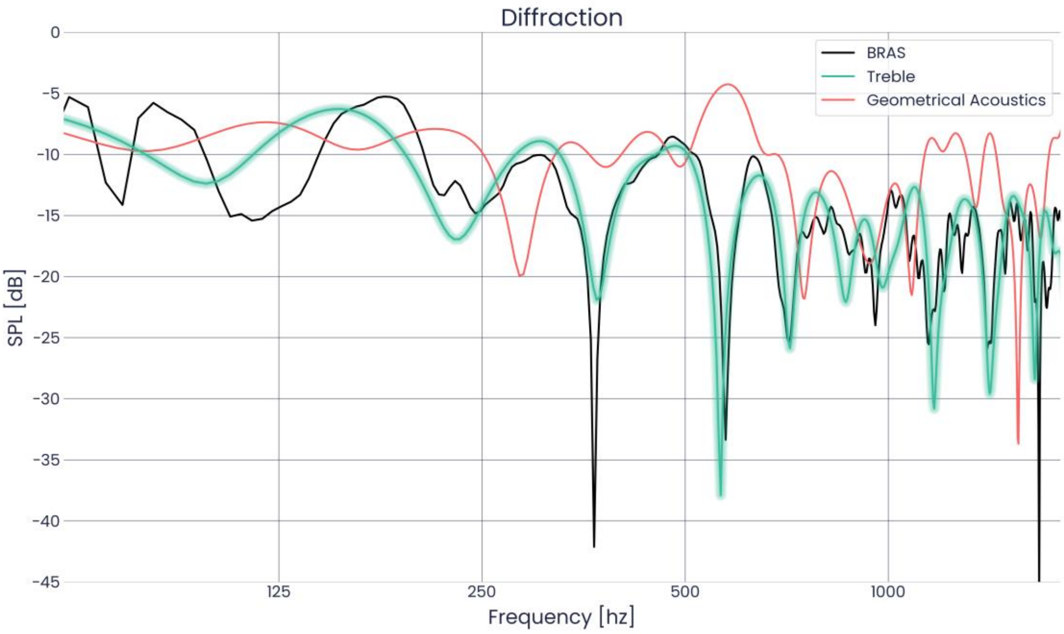

Figure 2: Comparison between a simulation in Treble, another GA solver software and measurements.

In Figure 3, the transfer functions at all three receiver locations for source LS01 are compared between measurements and Treble simulations. This particular source was chosen for display purposes as it's out of sight from all receivers. In all cases, there are good agreements observed.

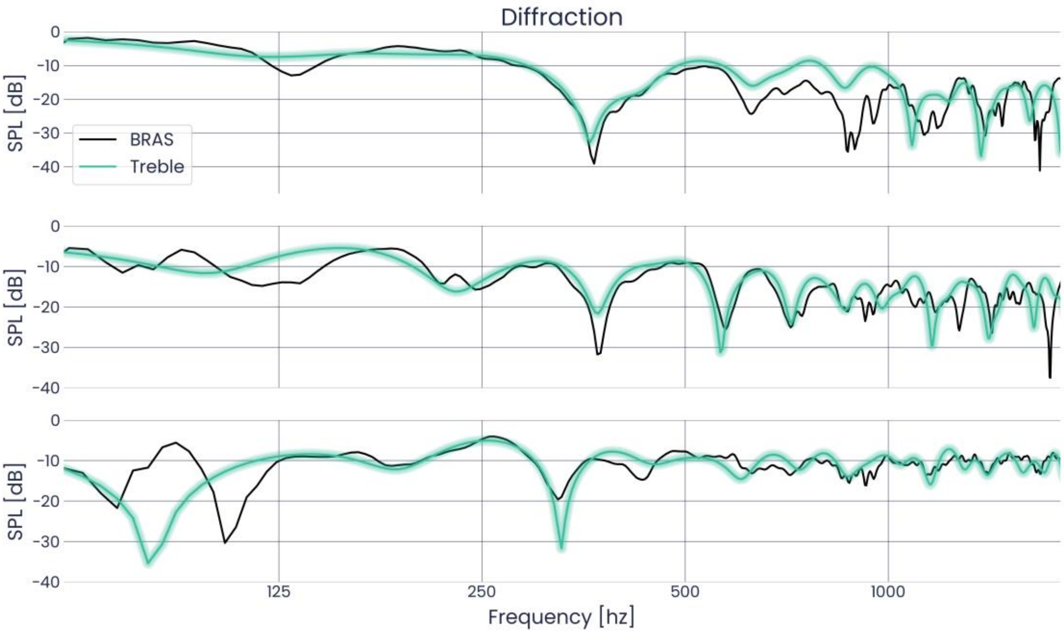

Figure 3: Comparison of transfer functions between Treble and measurements for LS1

The main sources of errors are slight uncertainties in the source and receiver locations, an omni-directional source modelling in Treble, and spurious reflection possibly from the room surfaces in Treble as the random incidence absorption coefficients of the boundary walls are 95% and not fully absorptive. Despite these uncertainties, the Treble simulation is convincing.

3 On Site Measurement Comparisons

3.1 Small Office Space

A study was conducted in a small office space to validate the accuracy of Treble's simulation algorithm on site. Figure 4 shows a photo of the space in question and the associated 3d model that was created to simulate in Treble. The room has a volume of approx. 48 m^3 and a Schroeders frequency of 190 Hz.

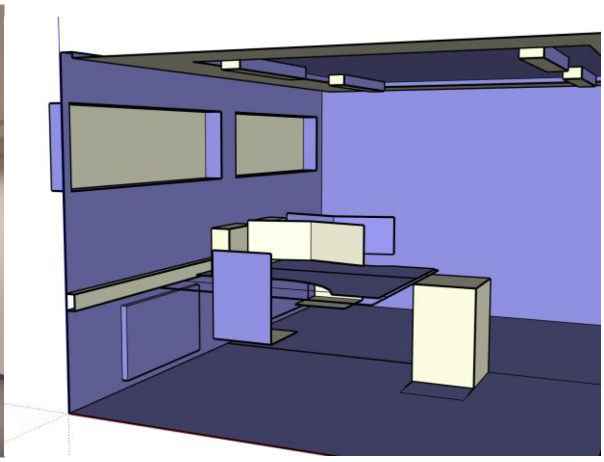

Figure 4: Left- The room in question. Right – The corresponding 3d model in Sketchup

Two main reasons influenced the choice of the room for this experiment. Firstly, that it's a small office with a high Schroeder's frequency. The modal behaviour of the sound field is therefore dominant up to the mid-frequency range which is difficult for traditional GA algorithms to simulate. Secondly the acoustic treatment in the room consists of 50 mm rockwool panels that have been lined along the perimeter of the ceiling while the middle of the ceiling is left untreated. The location along the perimeter is strategic as it is where the pressure peaks of the room modes occur. It should therefore lead to lower reverberation times at low frequencies when compared with the same amount of absorption located elsewhere. This effect on the low frequency response is not possible to simulate using traditional GA algorithms or other statistical methods such as Sabine's or Eyring's estimations.

To accurately predict the impulse response of a space using Treble's wave solver, the amount of detail in the associated 3d model is critical. The higher the transition frequency between the wave and geometrical solvers, the higher the level of detail necessary to model. A good rule of thumb is to model detail down to approximately a quarter wavelength of the transition frequency between the wave- and geometrical solvers, which in this case was chosen as the octave band centred around 1000 Hz.

To begin with the simulated frequency response was compared with the measured one, up to the Schroeder's frequency, where modal behaviour is expected to dominate. The measurements were conducted using Nor276 omnidirectional loudspeaker, Nor145 sound level meter and Nor280 amplifier. The source and receivers in the Treble simulation were set as omnidirectional. Figure 5 shows the results at the measurement position pictured in Figure 4. Note that there were some slight spatial discrepancies between the position of the source in Treble and on site that can affect the comparison:

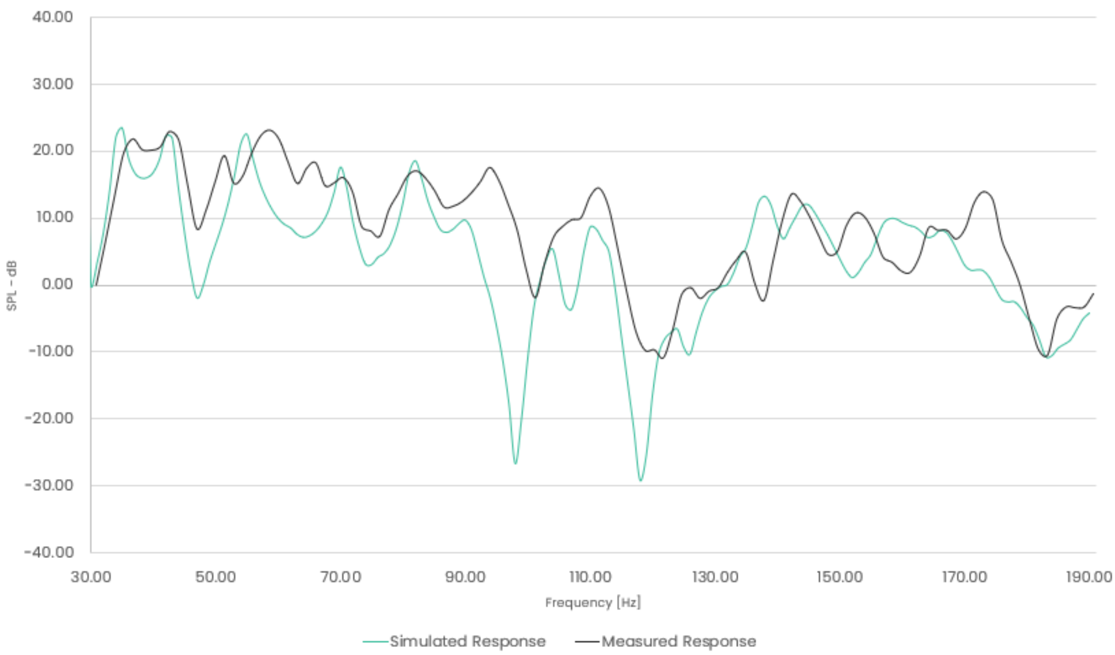

Figure 5:Comparison of the measured and simulated frequency responses

When setting up the simulation in Treble a conscious decision was taken only to use Treble's database of materials and not create any custom materials to avoid the problem of tailoring a simulation exactly to measurements. The materials and the associated absorption coefficients chosen for the simulation were chosen as follows:

Table 1: Absorption coefficients and scattering for the surfaces in the Treble model

Material	α -125 Hz	α -250 Hz	α -500 Hz	α -1000 Hz	α -2000 Hz	α -4000 Hz	Scattering
Gypsum walls	0.09	0.08	0.06	0.04	0.04	0.03	0.15
Concrete walls	0.01	0.01	0.01	0.01	0.01	0.01	0.15
50 mm rockwool Absorption	0.35	0.69	0.83	0.84	0.87	0.91	0.15
Wooden furniture	0.02	0.03	0.04	0.04	0.04	0.04	0.5
Windows	0.08	0.05	0.04	0.03	0.03	0.03	0.25
Linoleum flooring on concrete	0.02	0.02	0.02	0.02	0.02	0.02	0.15
Office chairs	0.35	0.47	0.63	0.72	0.78	0.86	0.6
Papers and books in the shelves	0.04	0.09	0.2	0.32	0.39	0.41	0.5
Monitors and printer	0.07	0.09	0.14	0.19	0.22	0.22	0.5
PC	0.1	0.1	0.1	0.1	0.1	0.1	0.5

It’s important to note that to achieve accurate simulation results above the transition frequency a good scattering input is also important. It’s generally more detrimental to underestimate scattering rather than overestimating it, when comparing with measured values. The more geometrical detail omitted in the 3d model, the more important it becomes to input sufficiently high scattering coefficients. E.g. when omitting modelling small objects on a table it’s important to increase the scattering coefficient of said table when compared with an empty one. Small objects and edges are also generally more diffusive than larger surfaces.

A comparison of the measured T20 to the one simulated in Treble as well as the calculated using Sabine’s estimate can be seen in Figure 6. The measurement and simulations were conducted using three source positions and 15 receiver positions for each source position. As can be seen the fit is excellent, not in the least in the low frequency range which is generally the most difficult to simulate.

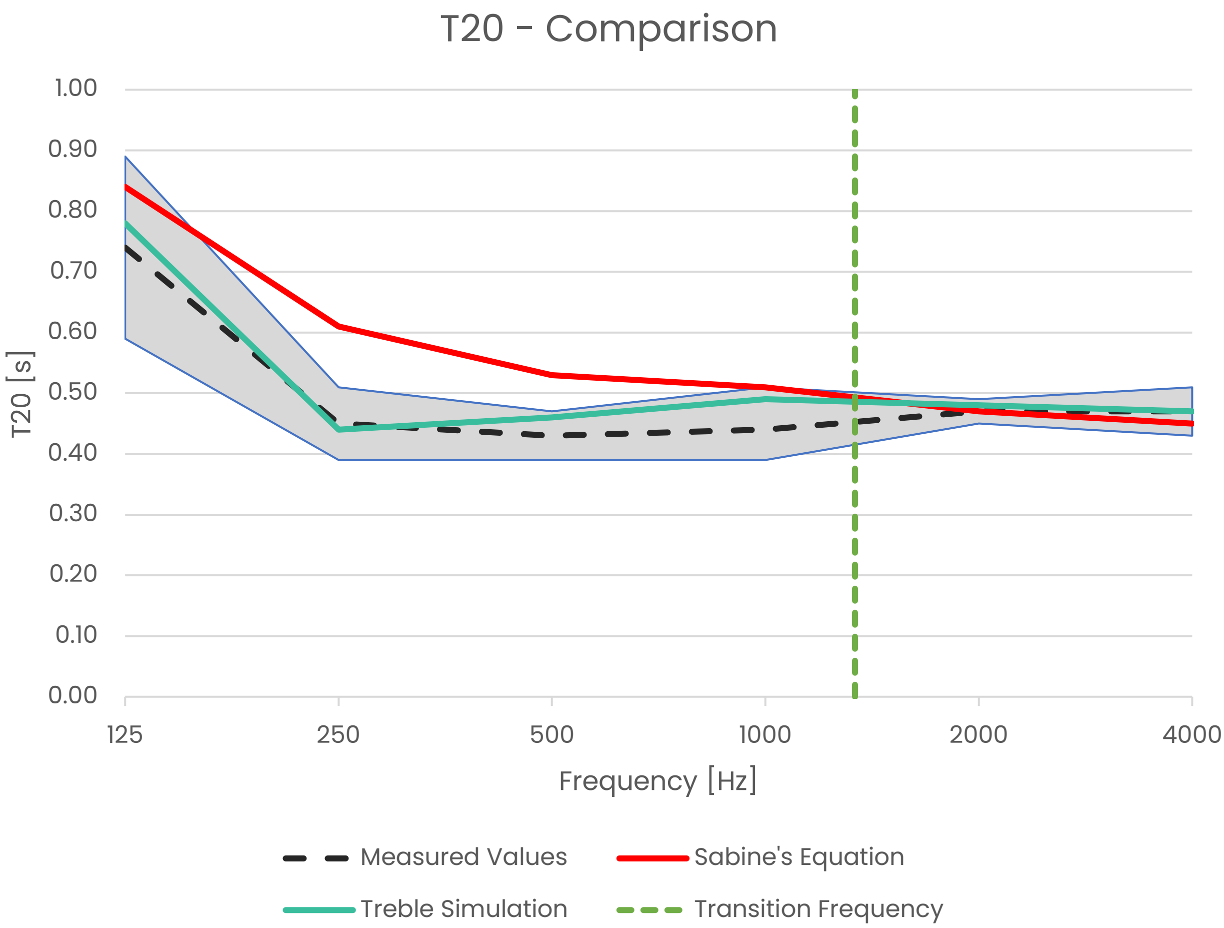

Figure 6: Comparison of the average of measured and simulated T20 values as well as Sabine results. The shaded area represents the standard deviation of the measurements

3.2 CR4 - Simulation of a Large Auditorium

To validate a Treble simulation of a large room, BRAS scene CR4 was chosen as a reference [4]. The room in question is a huge auditorium with a volume of 8700 m^3. The absorption coefficients for all surfaces are predefined in the BRAS database as shown in Table 2:

Table 2: Absorption coefficients for the surfaces in the Treble model

Material	α -63 Hz	α -125 Hz	α -250 Hz	α -500 Hz	α -1000 Hz	α -2000 Hz	α -4000 Hz
Bricks	0.16	0.19	0.16	0.20	0.08	0.21	0.16
Concrete	0.04	0.08	0.07	0.07	0.07	0.03	0.05
Parquet	0.08	0.06	0.04	0.03	0.05	0.07	0.10
Seating	0.16	0.09	0.24	0.29	0.30	0.27	0.35
Windows	0.12	0.14	0.07	0.05	0.05	0.08	0.05

A 3-way dodecahedron sound source was used during the measurement and an omni-directional microphones B&K 4134 was used in the seminar room. The Treble simulation uses an omni-directional source and omni-directional microphone.

The transition frequency between the wave- and geometrical solvers is 355 Hz, meaning that the wave-solver is used up to the 250 Hz octave band. 5 microphones are spread over the audience plan as can be seen in Figure 7.

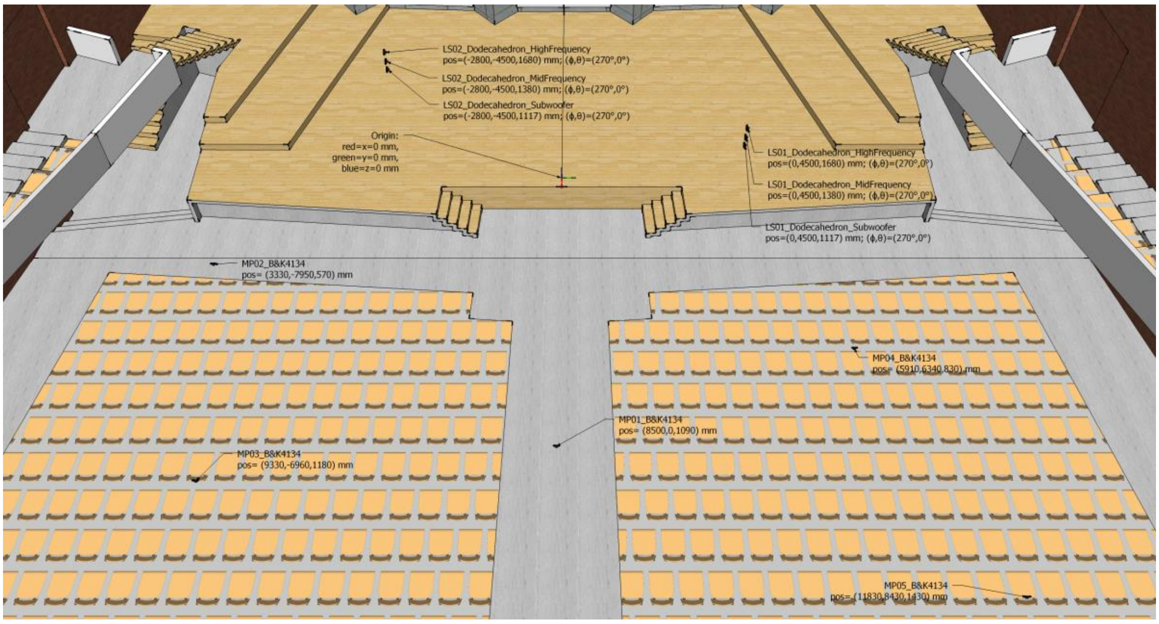

Figure 7: Measurement setup with the sources on the stage and receivers in the audience area

The simulated and measured ISO 3382 parameters are compared in Figure 8. The parameters are calculated from 2 sources and 5 receiver locations (thus 10 combinations) and then their averages and standard deviations are plotted. The resulting impulse response was analysed in Dirac to get the results in one-third octave bands. Note that in Treble the seating area was modelled as a flat box to conform with the predefined BRAS model. For accurate wave-based simulations it is however recommended to model the seating area in more detail. Since this case is mainly a test case for Treble's geometrical solver this simplification was deemed acceptable.

EDT is a reverberation parameter which is sensitive to early reflections and spatial deviations. It is a relatively difficult parameter to fit reliably between simulations and measurements. Typically, the spatial deviation of EDT is larger at lower frequencies. In the low frequency bands up to 125 Hz, a slight overestimation in the simulated results is observed. However Treble results shown in Figure 8 agree well with the measurement from 250 Hz and up. Centre time, Ts, is also a sensitive parameter to the temporal structure of the impulse responses. The average values of Ts from Treble are quite reasonable as well as the standard deviation. C50 also agrees well with measurements, particularly the average values. D50 is another clarity parameter that uses 50 ms as the transition time from early to late reflections. Good agreements of D50 parameter are observed between the simulation and BRAS measurement.

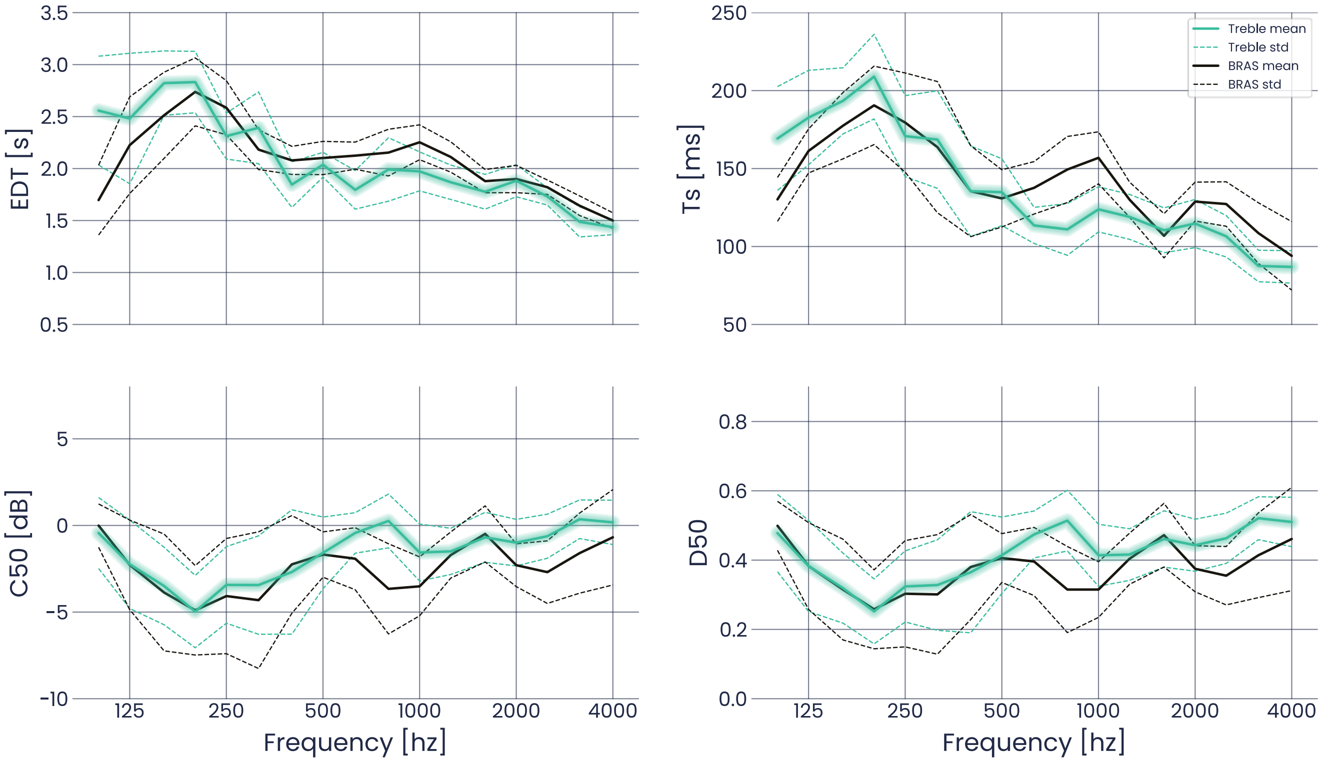

Figure 8:Parameter comparison between Treble simulations and measurements

4 Summary

The paper has highlighted the validity of Treble's simulation algorithm for some of the hitherto hardest cases to simulate in traditional room acoustics. Namely the effect of diffraction as well as low frequency behaviour in small rooms. This has been achieved both by simulating predefined and controlled reference cases as well as more traditional on site measurements. It has also been highlighted that Treble's geometric solver is capable of accurate simulations in larger rooms and in the mid-high frequency range in small rooms. The results show that Treble's platform can outperform traditional GA simulation tools in the aforementioned difficult cases and therefore provide more accurate simulations of the sound field than has been possible before.

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Renovating the Encore hall using electro-acoustic enhancement systems

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The Paviljonki exhibition centre in Jyväskylä was built in the 2002 and included the Wilhelm Auditorium. In 2020 it was decided that the hall should be the new home for the Jyväskylä Sinfonia.

Very quickly, it became clear that, while speech acoustics were excellent, the hall did not have room acoustics appropriate for symphonic music. It was also very clear, that to physically change to hall to have appropriate acoustics, would essentially mean rebuilding the whole hall from scratch.

It was therefor decided to design for an electro-acoustic enhancement system in the hall and to redesign the stage for use with a symphonic orchestra. Also, rehearsal rooms for the orchestra were made in one of the exhibition halls.

The paper will describe both the design and the final result.

1 Introduction

The Jyväskylä Sinfonia is a gem of Central Finland's classical music, based in Jyväskylä. The orchestra's almost 120 events reach more than 35,000 listeners every year. Since there has been no concert hall designed for acoustic music in Jyväskylä, the Jyväskylä City Theatre has been the home hall of the Jyväskylä Symphony.

When the renovation of the city theatre was approaching, preparations began to take over Jyväskylä Paviljonki Wilhelm Hall for conference use as well as for the Jyväskylä Sinfonia. Initial acoustic measurements were done in the hall, using the reduced virtual orchestra system [1],[2]. These clearly showed that even though the Wilhelm Hall had excellent speech intelligibility, neither the acoustic conditions, nor the stage or backstage arrangements, were anywhere close to what would be needed to host regular acoustic concerts.

Also, it was clear that achieving sufficiently good acoustic conditions, using traditional methods, would not be possible so it was decided from the very start that the only feasible solution would be to install an Electro Acoustic Enhancement system.

2 Acoustic design

The acoustic design for the original hall was done by Raimo Parjo.

The Wilhelm Hall is actually 3 separate halls: a traditional, very wide auditorium and two smaller halls that can be rotated 180 degrees and used separately, see figure 1.

The original ceiling in the original halls were gypsum board, profiled to optimize natural speech intelligibility, and it worked very well for this purpose. For the future use of the hall, the ceiling proposed two major concerns: its shape was too projective (optimized for speech intelligibility) and it was quite heavy. This was a problem because there was no official information about the loadbearing capacity of the ceiling, so by removing the heavy gypsum board ceiling and replacing it with a more lightweight metal mesh ceiling, at least some load capacity was achieved.

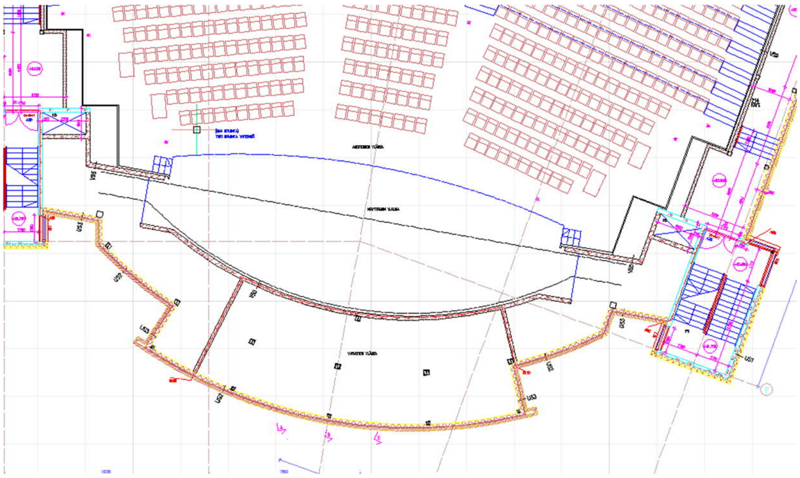

Figure 1: Original stage

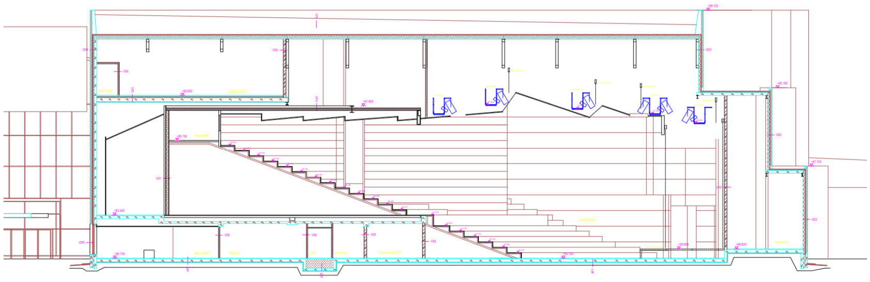

Figure 2: Length section before renovation

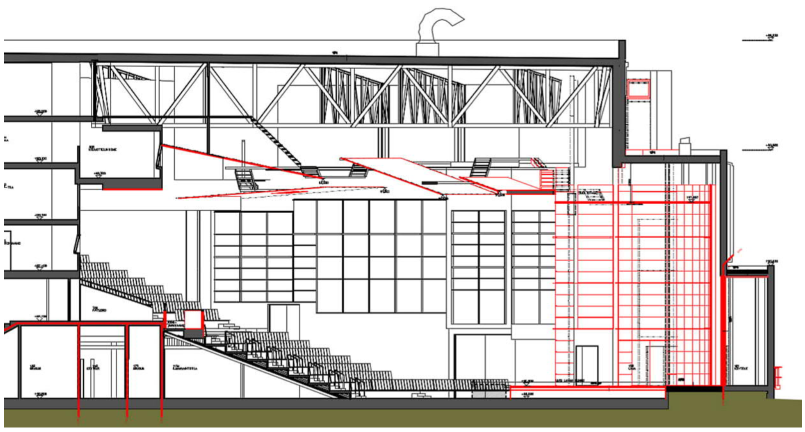

Figure 3: Section of the hall after renovation

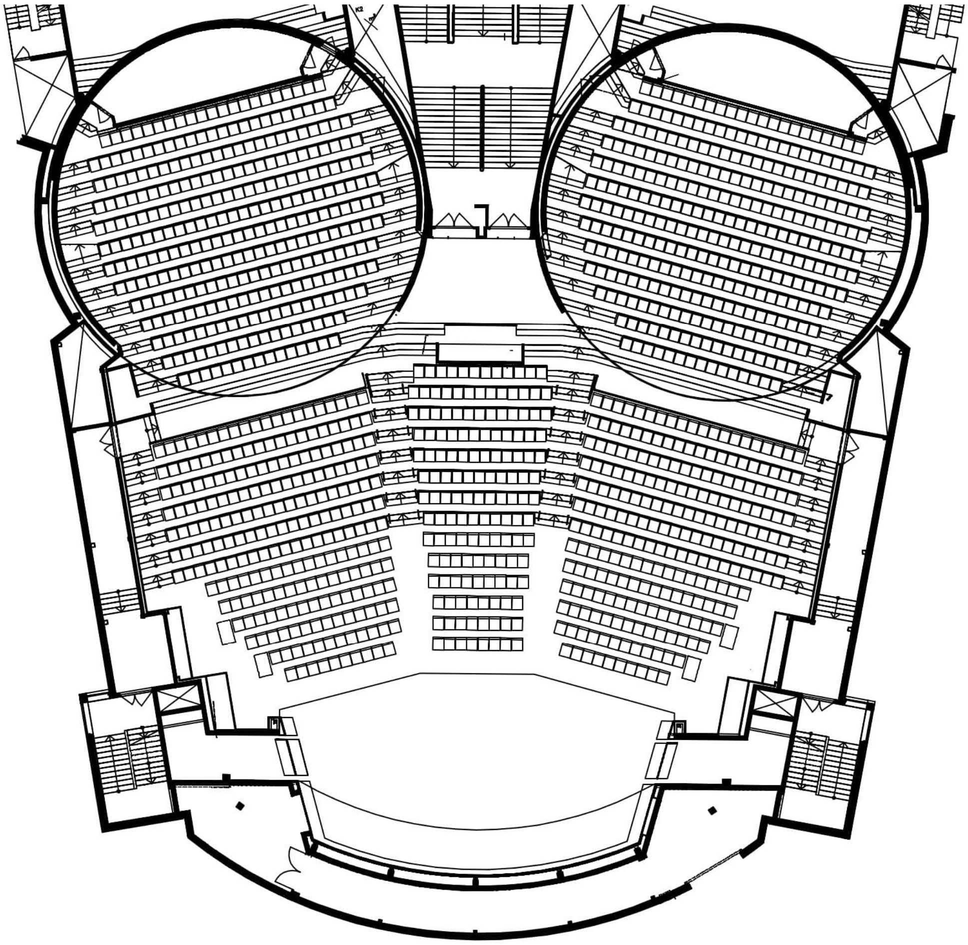

Figure 4: Plan of the hall after renovation

Also, it was clear that the original stage was neither sufficient large nor appropriately for a symphony orchestra. All stage walls were curtains, and there was very little storage space behind the

The original stage was made from podiums and not sufficient deep for a symphony orchestra. Furthermore, the orchestras demand was for a stage with as good as possible natural acoustics, so fixed stage walls had to be designed.

The solution was to remove the backwall and extend the stage about 2 m back and to extend the back-stage area by extending the curved outer walls to the staircase and to make a loading ramp in front. Also, this meant that the rear part of the stage, see figure 3, is built on top of the old backstage concrete floor, where as the rest of the stage is a traditional wood on joist construction. Furthermore, there is an option to extend the stage by 1,5 m for large orchestras, see figure 6.

The overhead reflectors are made from bended plywood.

The stage wall is constructed from gypsum boards with a veneered plywood board on top. The shelves are veneered plywood.

The original ceiling was removed and replaced with an acoustically transparent meal mesh. This increased the volume of the hall but as the area above the ceiling is mainly ventilation installations and thus very absorbent, the increase in volume did not increase the reverberation time in the hall.

It was also clear that the electro acoustic enhancement system had to work from the beginning and as the orchestra had experience with an old, not very well working system in their old hall, it was decided not to take any chances with the new system. Therefore, the main requirement for the new system was:

- Reference installation in halls with a resident orchestra
- Well documented tuning
- Fast service/repair

For this reason, the Meyer Constalation system was chosen.

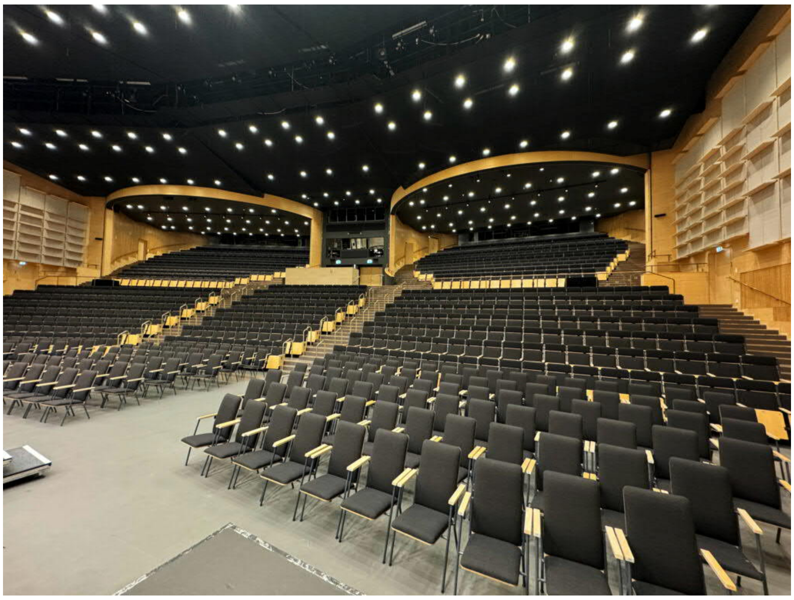

Figure 5: Hall after renovation

Figure 6: Enlarged stage

3 Achieved acoustic conditions.

As the acoustics of the hall is based on the electro acoustic enhancement system, the hall has several different setups:

- PA
- Drama
- Musical Theatre
- Chamber Music
- Symphony
- Choir

Also, all settings are done of both occupied and unoccupied hall and with the rear pods connected to the hall or closed. However, it was decided that for concert, the rear pods will always be connected to the hall, so all 1000 seats can be sold.

The reverberation time for the different setting is shown in figure 7. As can be expected, the variation is far above what one can achieve with traditional variable acoustics. The same can be seen in figure 8 for the Early Decay Time.

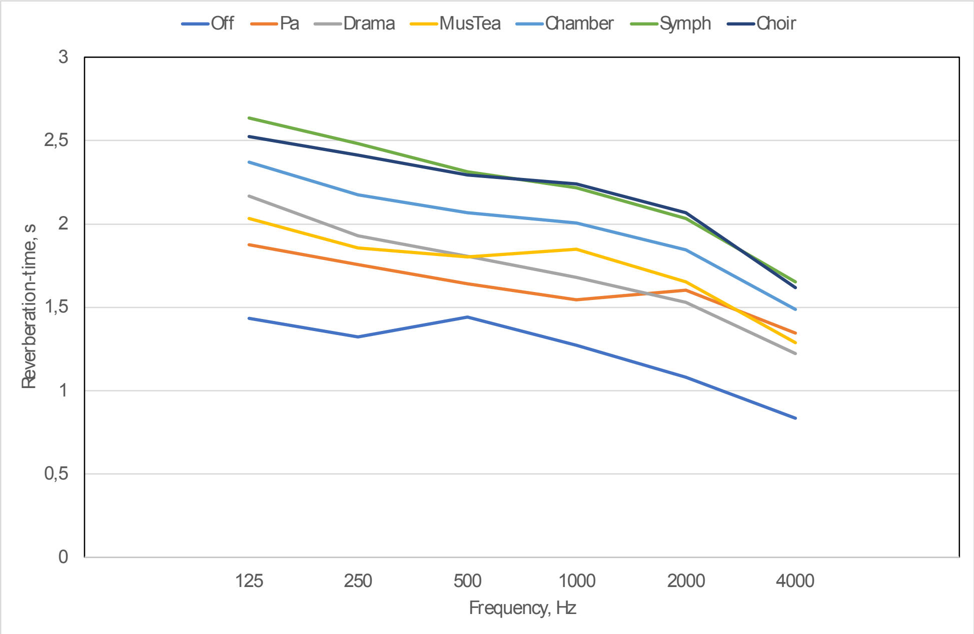

Figure 7: Reverberation time for the different setting

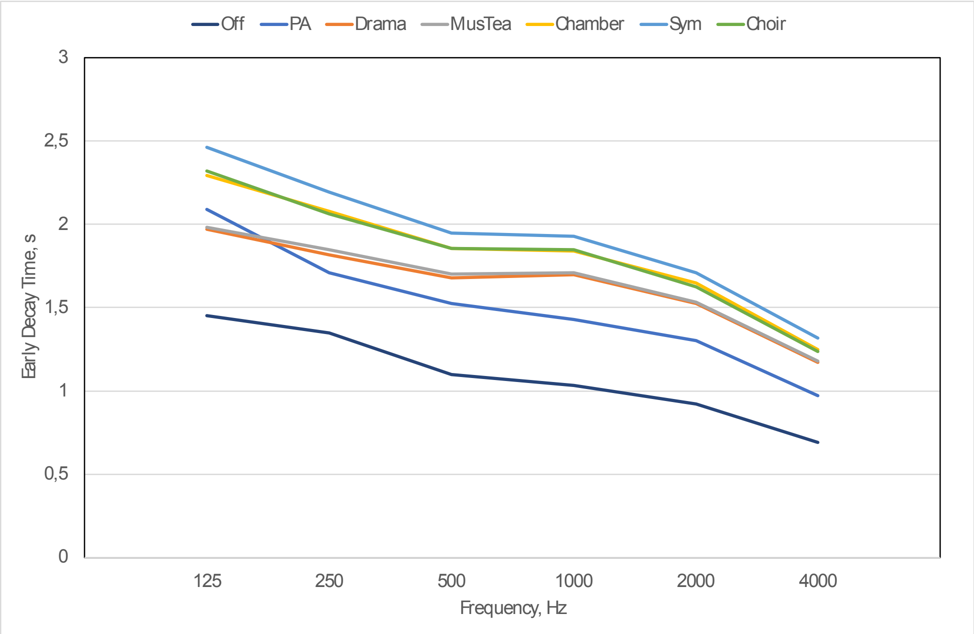

Figure 8: Early Decay time for the different setting

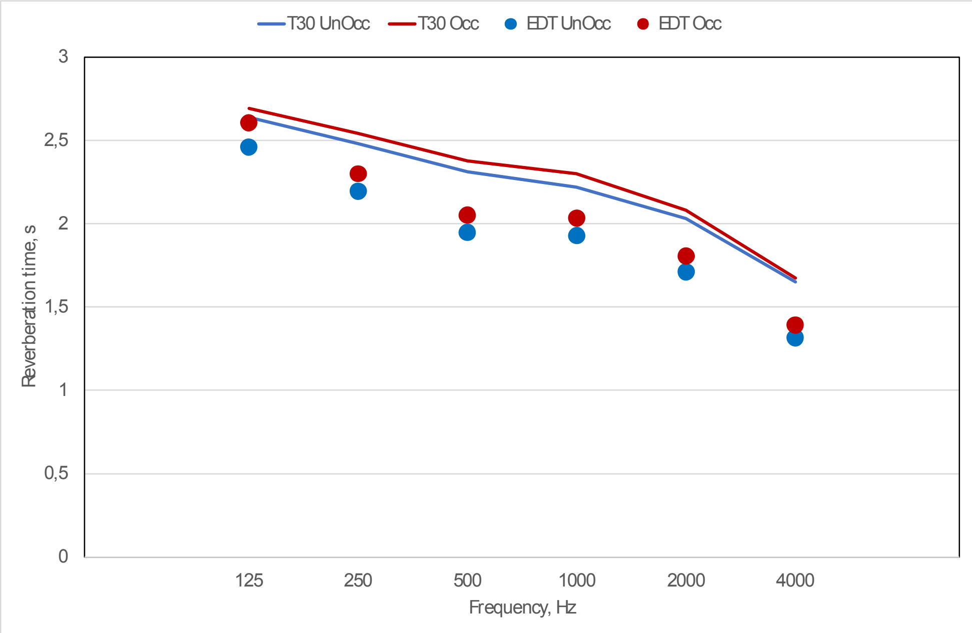

Figure 9: Comparison for Occupied and Unoccupied settings for Symphonic music, T30 and EDT

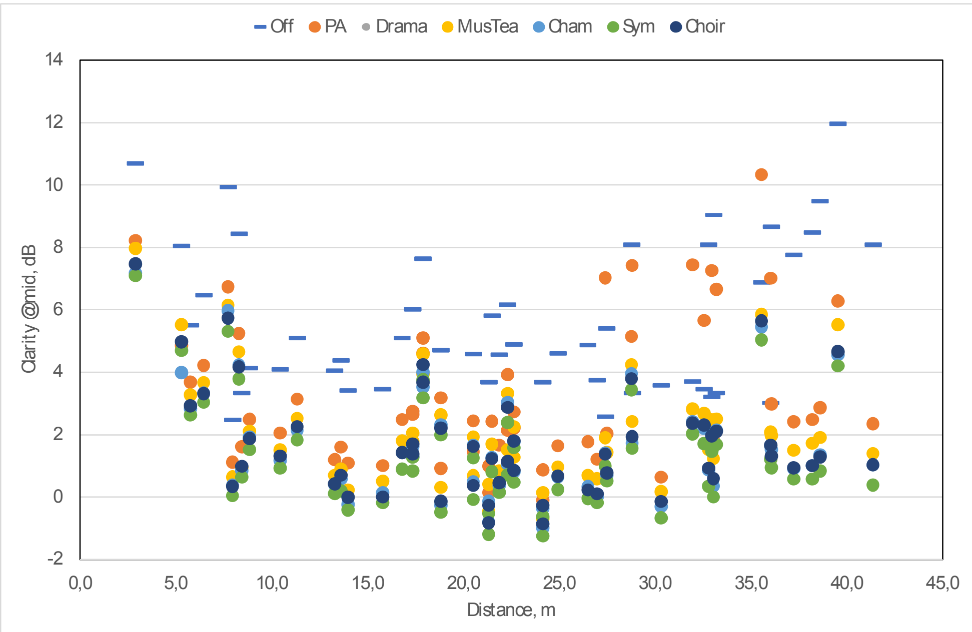

Figure 10: C_{80} @Midfequencies as a function of distance

As can be seen from figure 10, the C_{80} is greatly lowered with the system on, however is can also be seen that the change with distance is much smaller. In particular this is evident receiver's 30+ m from the stage which is the receiver located in the rear pods.

4 Conclusion

The comments from most reviewers have been “Well, this is now a concert hall” which can be seen as a confirmation that the design has been successful. The conditions on the stage are still being optimized, both for the electronic enhancement system as well as for the “natural” reflectors. Also, some extra reflectors covering the entrances to the stage are planned.

But overall, the end-result proves that it is possible to convert a hall, which by traditional acoustic measures would be “impossible” for acoustic symphonic music, it to a venue where it is possible for the orchestra to play without having to fight the acoustic conditions.

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Perception of Reverberation Length in rooms with Reverberation Enhancement Systems

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This study examines the subjective perception of reverberation length in rooms equipped with Reverberation Enhancement Systems (RES) through laboratory listening tests. While many acousticians have explored the correlation between subjective judgments of reverberation and objective criteria in regular concert halls, the influence of RES on these evaluations remains inadequately investigated. The decay of acoustic energy in rooms with RES exhibits a double-slope characteristic, distinguishing it from typical impulse responses recorded in rooms lacking this system. The study reveals a discrepancy in the assessment of perceived length between "stop-chord" and "running" reverberation. Additionally, the results suggest that both the initial and later segments of the impulse response should be considered when evaluating reverberation length perception. These findings enhance our understanding of the relationship between subjective perception and objective measures of reverberation length, offering insights into the perceptual evaluation of acoustic environments.

Introduction

Reverberation time (RT), originally defined by W. C. Sabine, serves as the fundamental parameter for assessing interior acoustics [1]. Despite the existence of supplementary parameters, RT retains its prominence, often being the sole metric reported. Numerous room acoustics parameters are founded on energy dissipation and exhibit correlation with RT, with certain metrics such as Bass Ratio or Treble Ratio are directly calculated from RT [2]. Extensive research has been conducted on the relationship between subjective perception of reverberation and objective criteria. Atal and Schroeder [3] identified the initial 160 ms of the impulse response (IR) as crucial for reverberation perception. Jordan [4] introduced the concept of Early Decay Time (EDT) as the measure of the first 10 dB of decay, positing it as the most indicative of reverberation length perception. In 1974, Hungarian acousticians conducted a test employing 24 audio sample pairs with 25 listeners in an anechoic chamber and 136 listeners using headphones [5]. However, conclusive results were not obtained, leading the researchers to emphasize the intricate nature of reverberation perception and the necessity for further investigation. Kahle and Jullien [6] proposed an extension of the effective duration of the Early Decay Time (EDT) window and advocated for the utilization of a 15 dB decay as the optimal indicator of reverberation time, based on empirical inquiries. Soulodre and Bradley's study observed a higher correlation between EDT and subjectively perceived reverberation in laboratory tests compared to RT20 [7]. Barron, through an extensive investigation of 17 British concert halls, explored the relationship between EDT and RT [8]. While he did not directly link the objective measure of RT to subjective evaluations of reverberation length, Barron acknowledged the validity of previous studies' results and highlighted new research directions in reverberation perception.

Nowadays, the demand for tailored acoustic solutions for various events has led to the construction of venues with variable acoustic conditions. Reverberation Enhancement Systems (RES) using signal processing techniques are gaining popularity for this purpose. These systems enable the adjustment of acoustic parameters as needed for specific performances. The earliest installations of RES, utilizing microphone-speaker loops primarily to introduce additional resonances and reflections, emerged in the 1960s [9,10]. Generally, active enhancement of room acoustics can be accomplished via two main methods: regenerative and in-line approaches. Regenerative systems employ omnidirectional microphones positioned beyond the critical distance from the sound source, reproducing natural reflections within loops comprising microphones, delay lines, amplifiers, and speakers. Conversely, in-line systems utilize cardioid microphones placed relatively close to the sound source, with artificial reverberation engines integrated into each loop, augmenting the original sound captured by the microphones [11,12,13]. It is noteworthy that the reverberations generated by RES often exhibit non-linear impulse response

(IR) slopes, distinguishing them from most IRs recorded in venues lacking such systems. This discrepancy is evident in the differences between Early Decay Time (EDT) and reverberation time (RT) values, which would typically approximate each other in the case of linear decay.
Moreover, in accordance with ISO 3382 [14] standard, the challenge of achieving a 60 dB drop during measurements has led to the introduction of RT20 and RT30 parameters, denoting decays of 20 dB and 30 dB (measured from -5 dB below stable state), respectively. Additionally, RT10, representing a 10 dB drop, serves as an intermediary measure between Early Decay Time (EDT) and RT20. Regarding reverberation perception, research suggests that the Just Noticeable Difference (JND) for EDT and RT is approximately 5% [15], although recent findings indicate an EDT JND value of 18% [16]. The discourse on standards pertaining to objective measurements and result interpretation remains ongoing [17], and further investigations are likely to influence existing parameters. Nonetheless, this study adheres to currently established standards. Employing such a methodology facilitates the comparison of findings with those of analogous experiments conducted previously.

Aim

The primary objective of this experiment was to determin the most suitable objective measure—namely, EDT, RT10, RT20, or RT30—that aligns with the subjective perception of reverberation length experienced by listeners, particularly in environments where reverberation is generated by RES. Additionally, the study aimed to evaluate the consistency of this perception in spaces both with and without RES.

Materials and Methods

The present experiment aims to explore the perceived reverberation length elicited by reverberations generated through RES. A listening test was administered to examine the influence of individual parameters: EDT, RT10, RT20, RT30, on the perceived reverberation length. This test entailed the presentation of stereo sound sample pairs differing solely in one of the aforementioned objective parameter values utilized for quantifying reverberation length. Subsequently, listeners were tasked with comparing the samples and indicating which one they perceived to possess a longer reverberation length.

1.1 Impulse Responses (IRs)

Given the spatial nuances inherent in reverberation, stereophonic audio samples were selected for the experiment to accurately replicate conditions observed within concert halls. Utilizing AKG 414C B-ULS microphones configured in Mid-Side (M-S) arrangement, comprising a cardioid microphone and a figure-of-eight microphone, over 200 impulse responses (IRs) were captured from five distinct concert halls outfitted RES. Among these venues, four are multipurpose halls equipped with RES, while one serves as a demonstration hall with RES installed for illustrative purposes. Each auditorium within these halls shares a similar seating capacity, accommodating audiences ranging from 320 to 408 spectators.
Parameter values presented in this study were derived by averaging measurements across bands spanning from 125 Hz to 8 kHz at each measurement point. The EASERA Pro v. 1.2 software facilitated the collection of IRs and the computation of parameter values. EASERA Pro v. 1.2 calculates parameter values by extrapolating from level decay patterns specific to each parameter, aligning with the requirements outlined in measurement standard regulations [14].

1.2 Stimuli

The IRs utilized in generating audio samples for the listening test were meticulously chosen to maintain close alignment among three out of four parameters (EDT, RT10, RT20, RT30) and the meaningful difference for the rest of them. Specifically, the discrepancy between parameter values was constrained to be less than half of the Just Noticeable Difference (JND) threshold for each respective parameter. Furthermore, the value of the parameter under investigation was deliberately selected to exhibit a difference between values of at least one and a half times the JND for that parameter.
The anechoic recordings employed in this experimental setup comprised brief instrument excerpts spanning durations of 15 to 20 seconds. These excerpts encompassed a solo flugelhorn performance, representing "running" reverberation, and rhythmic beats from a snare drum, representing "stop-chord" reverberation. These anechoic recordings underwent convolution with the IRs to yield stimuli for the listening test.

For the snare drum sound samples, participants were able to discern both the buildup and decay of reverberation. Conversely, for the flugelhorn sound samples, the beginning part and final part of the excerpts were gradually faded in and out, respectively, to obviate the perception of reverberation buildup and decay by the listeners. The flugelhorn melody was played continuously, without of pauses.

1.1.1 IRs with different EDT

Table 1 presents the objective parameter values of one of pairs used in the study, corresponding to the sound samples employed in the experiment, which aimed to examine the influence of altering the Early Decay Time (EDT) value on reverberation length perception. The disparity in EDT values among the samples exceeds 0.15 seconds, whereas the difference between the values of other parameters—namely, RT10, RT20, and RT30—are less than 0.03 seconds.

Table 1: Measured values for different EDT

pair/ sample	**EDT [s]**	RT10 [s]	RT20[s]	RT30 [s]
1/A	**1,08**	1,19	1,20	1,20
1/B	**0,91**	1,20	1,20	1,19

1.1.2 IRs with different RT10

Table 2 delineates the objective parameter values of one of pairs used in the study corresponding to the sound samples employed in the experiment, which aimed to explore the influence of modifying the Reverberation Time at 10 dB (RT10) value on reverberation length perception. The divergence in RT10 values among the samples exceeds 0.15 seconds, while the disparities between the values of other parameters—namely, Early Decay Time (EDT), Reverberation Time at 20 dB (RT20), and Reverberation Time at 30 dB (RT30)—are less than 0.06 seconds.

Table 2: Measured values for different RT10

pair/ sample	EDT [s]	**RT10 [s]**	RT20 [s]	RT30 [s]
1/A	0,98	**1,14**	1,27	1,31
1/B	1,02	**1,30**	1,32	1,31

1.1.3 IRs with different RT20

Table 3 presents the objective parameter values of one of pairs used in the study corresponding to the sound samples utilized in the experiment, which aimed to assess the effects of modifying the Reverberation Time at 20 dB (RT20) value on reverberation length perception. The disparity in RT20 values among the samples exceeds 0.15 seconds, while the differences between the values of other parameters—namely, Early Decay Time (EDT) and Reverberation Time at 10 dB (RT10)—are less than 0.02 seconds. Notably, discrepancies in the values of Reverberation Time at 30 dB (RT30) exceed the predefined threshold. This discrepancy is attributed to the non-linear characteristics of the impulse responses (IRs). When RT20 surpasses both EDT and RT10 values, RT30 is correspondingly elevated.

Table 3: Measured values for different RT20

pair/ sample	EDT [s]	RT10 [s]	**RT20 [s]**	RT30 [s]
1/A	0,67	0,79	**1,08**	1,19
1/B	0,67	0,80	**0,85**	0,87

1.1.4 IRs with different RT30

Table 4 presents the objective parameter values of one of pairs used in the study corresponding to the sound samples utilized in the experiment, which aimed to explore the impact of altering the Reverberation Time at 30 dB (RT30) value on reverberation length perception. The discrepancy in RT30 values among the samples exceeds 0.15 seconds, while the differences between the values of other parameters—namely, Early Decay Time (EDT), Reverberation Time at 10 dB (RT10), and Reverberation Time at 20 dB (RT20)—are less than 0.06 seconds.

Table 4: Measured values for different RT30

pair/ sample	EDT [s]	RT10 [s]	RT20[s]	**RT30 [s]**
1/A	0,21	0,48	0,67	**0,77**
1/B	0,20	0,48	0,67	**0,96**

1.3 Participants

The experiment was conducted with 105 participants with normal hearing. Among them, eight individuals were below 20 years of age, 45 fell within the age range of 20 to 29 years, 19 were aged between 30 and 39 years, 25 were aged between 40 and 49 years, and five were above 50 years of age. Experiment was conducted during sound engineering training sessions at an industry conference prepared for sound engineers and music producers, all participants possessed experience in assessing reverberation length and demonstrated proficiency in discerning subtle nuances within reverberant structures.

1.4 Procedure

The experiment was executed with an internet-based platform for stimulus presentation to participants. Sound samples were presented through a MOTU M2 interface with hardware volume control. Participants employing Focal Listen Pro headphones for auditory perception; however, a minority of participants (less than 10%) utilized their personal headphones during the test. Prior to commencement, participants were briefed that inquiries pertained solely to reverberation length, excluding other acoustic parameters from consideration. Throughout the experiment, participants encountered pairs of sound samples where three reverberation time parameters remained constant while the parameter under investigation varied. For each parameter—namely EDT, RT10, RT20, and RT30 - five pairs of "stop-chord" and five pairs of "running" sound samples were presented. Participants could listen to each sample pair repeatedly before furnishing their responses. Initial sound level was standardized to 65 dB. Participants were tasked with identifying the sample within each pair perceived to exhibit a longer reverberation time. Evaluation of participants' responses involved on the consistency between their perceived alteration in reverberation length and the corresponding numerical variation in the specific reverberation time parameter under examination. Correctness of responses was adjudged based on the congruence between participants' perceived alterations in reverberation length and the actual numerical shift in the tested parameter. Conversely, if a participant identified the sample with shorter reverberation when the tested parameter was augmented, the response was considered as incorrect.

2 Results

Figure 1 illustrates the outcomes for running reverberation samples, while Figure 2 portrays the results for stop-chord reverberation samples. The depicted values signify the percentage of correct responses for the parameter under scrutiny across all presentations administered to the test participants. To evaluate whether participants' responses were statistically significant or merely random, a binomial test was conducted, contrasting the proportion of correct responses against a hypothetical value of 0.5. The findings revealed statistical significance in nearly all instances ($p < 0.05$), except for the "RT30 running" condition, where both responses were equally prevalent (263 and 262). The data indicate that for running reverberation samples, correct responses were observed in 61%, 58%, 72%, and 50% of instances for the EDT, RT10, RT20, and RT30 parameters, respectively. Conversely, for stop-chord reverberation samples, the corresponding figures were 27%, 55%, 90%, and 77% for the EDT, RT10, RT20, and RT30 parameters, respectively. Of note, the parameter exhibiting the highest concordance between its numerical variation and the perceived reverberation length was RT20, evident in both running and stop-chord reverberation samples. Moreover, RT20 obtained the highest proportion of correct responses across both types of reverberation samples.

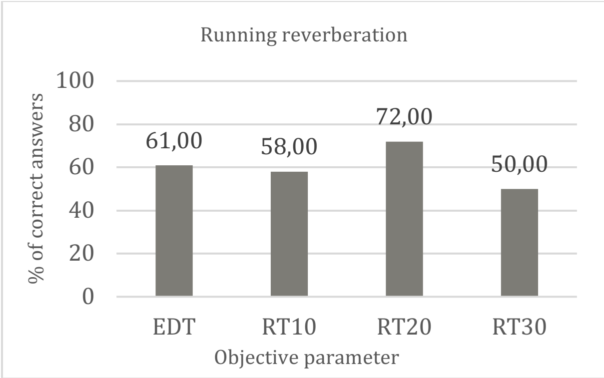

Figure 1: Correct answers for "running” reverberation

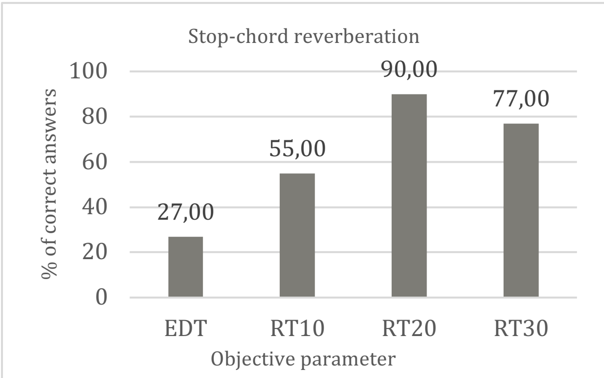

Figure 2: Correct answers for "stop-chord” reverberation

Discussion

In the assessment of "running" reverberation, the RT20 parameter exhibited the highest level of agreement between its numerical variation and the subjectively perceived reverberation length. Specifically, 72% of participants provided responses consistent with the anticipated outcome based on changes in RT20. The remaining parameters, ranked in descending order of performance, were EDT with 61% correct responses, RT10 with 58% correct responses, and RT30 with 50% correct responses. Despite achieving statistical significance for responses pertaining to the RT10 and RT30 parameters, response rates slightly exceeding 50% are deemed insufficient for reverberation length evaluation. Consequently, the findings suggest that RT20 emerges as the most pertinent parameter in assessing reverberation length for "running" reverberation.

For "stop-chord" reverberation, the RT20 parameter similarly demonstrated the highest level of concordance between its numerical variation and the subjectively perceived reverberation length, attaining a correctness rate of 90%. Subsequent parameters ranked in descending order of performance were RT30, RT10, and EDT, with correctness rates of 77%, 55%, and 27%, respectively. Once more, the findings underscore RT20 as the optimal predictor of reverberation length. It is noteworthy that the number of correct responses for RT30 is notably high, surpassing that of any parameter in the "running" reverberation segment of the study. However, RT10, with a correctness rate of 55%, cannot be considered a dependable predictor of reverberation length. Additionally, a negative correlation is observed between the increase in parameter value and subjective evaluation of reverberation extension for EDT, evidenced by a correctness rate of 27%. Nevertheless, this still highlights the significant influence of parameter value on reverberation length assessment. Furthermore, the discrepancy in relevance assessment between EDT and RT10 when appraising reverberation length is noteworthy, despite both parameters delineating the same decay slope and occupying the early segment of the impulse response.

Statistical significance assessments were conducted for all decay evaluation parameters across both "running" and "stop-chord" reverberation modes, revealing statistically significant findings. However, the RT30 parameter, commonly utilized in accordance with global measurement standards, displayed the lowest significance in the "running" reverberation mode, while the RT10 parameter exhibited the lowest significance across both "running" and "stop-chord" reverberation modes. Notably, a negative correlation was identified among impulse samples characterized by variations in their Early Decay Time (EDT) value, an observation of particular interest. This correlation mirrors findings reported in a previous study [5], albeit the present study confirms the relationship across a larger sample size. Moreover, an escalation in EDT value while maintaining consistent values for other parameters fails to align with subjective perceptions of increased reverberation length.

It is imperative to emphasize that the findings obtained in this study deviate from established research on reverberation length perception in environments devoid of RES [4, 7]. This incongruity suggests a potential divergence in the perception of reverberation characterized by double-slope impulse responses (IRs) compared to reverberations featuring more linear decay slopes. Further investigations are warranted to conclusively ascertain the influence of RES on reverberation perception. The present study predominantly centered on double-slope decay IRs generated by RES, shedding light on the unresolved inquiry surrounding the relationship between diverse objective measures of reverberation and its subjective perception by human observers.

Moreover, the survey outcomes corroborate the contrasting perception of reverberation length between impulse and continuous audio samples, aligning with findings in the existing literature [18].

Conclusions

The findings underscore the intricate nature of reverberation perception, highlighting its multifaceted nature that cannot be distilled into a straightforward relationship between objective parameters and subjective evaluation. Consequently, prevalent standards for objective parameter measurements may inadequately encapsulate the perception process and fail to accommodate individual differences in perception. In view of these revelations, it is recommended that the acoustic design of enclosures should consider the perceptual aspects of human listeners, rather than relying exclusively on objective metrics. Integrating subjective evaluations into the acoustic design process holds the potential to yield more precise assessments of acoustic quality, thereby enhancing the overall acoustic experience for listeners. Below enumerated the detailed findings from the study:

- The correlation between subjective perception of reverberation and objectively measured parameters is not a straightforward one-to-one relationship.
- The study did not conclusively establish the dominance of EDT values in assessing reverberation length for double-slope reverberations generated by RES, contrary to findings in prior research on reverberation with a more linear impulse response shape.
- The presence of double-slope IRs in reverberations suggests the potential existence of a similar relationship for other types of double-slope IRs, such as those resulting from the utilization of coupled chambers.
- Further investigation is warranted to ascertain the generalizability of the findings to other forms of double-slope IRs.
- It is essential to acknowledge that reverberation length perception is multifaceted and is not solely contingent upon changes in a single parameter value. Parameters including RT10, RT20, and RT30 were deemed pertinent in reverberation length assessment.
- The highest level of agreement was observed for the RT20 parameter across both "running" and "stop-chord" reverberations, indicating its superior correlation with the subjective perception of reverberation length across all signals.

It is pertinent to underscore that while the perception of reverberation length may differ between halls with and without Reverberation Enhancement Systems, listeners are distinctly able to perceive changes in reverberation length. Thus, it can be inferred that the implementation of RES to modify reverberation length in concert halls is effective.

Given the confirmed differential perception of "running" and "stop-chord" reverberations, further investigation is warranted in this domain to identify potential objective measures of this phenomenon. Comparative analysis of results will aid in elucidating the predominant evaluation mechanism: whether reverberation length is assessed during musical passages or during intervals between passages, when reverberation decay is distinctly audible.

Comments

The research results presented in this paper were presented also during Forum Acusticum 2023 conference and were published in post conference proceedings. Thus, there is no new founding in this article, which was written to present the results of the same research at the BNAM 2024 conference.

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Renovation of Finnish Modern Theatres – Acousticians' Experiences from the Past 10 Years

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The era from the 1960s to the 1980s saw a great boom in the construction of city theatres in Finland. These theatre buildings – the status symbols of their cities – were designed by the most renowned modern Finnish architects alongside the top experts of acoustic design and theatre technology of the time. 50 years later, however, the buildings were reaching the end of their technical lifecycles, all the while there had been developments in the demands of the theatre production process as well. Since the 2010s, many modern era theatre buildings have seen extensive renovations to meet current requirements.

This paper presents five Finnish theatre buildings built during the 1960-80s' heydays and renovated in the past 10 years with the authors as acousticians: Seinäjoki city theatre (completed 1987, renovation 2014-20), Lahti city theatre (completed 1983, renovation 2019-2022), Jyväskylä city theatre (completed 1982, renovation 2022-25), Helsinki city theatre (completed 1967, renovation 2013-17), Rovaniemi city theatre / Lappia-talo (completed 1975, renovation 2011-15).

The paper compares the theatres' architectural and technical features that affected the acoustical design of the renovations. The paper presents an overview of the acoustical conditions before the renovations, the encountered issues and their remedying principles and the post-renovation acoustical conditions. Discussing the room acoustics of the auditoria, studio design, sound insulation and noise control the authors share some of the acousticians' experiences gained from these unique buildings, wishing that the insights might by helpful to similar renovation projects in the future.

1 Introduction

From the 1960s to 1980s a significant number of city theatres were built in Finland [1], which by the 2010s had reached renovation age. This paper discusses five theatres of the period – Seinäjoki, Lahti, Jyväskylä, Helsinki and Rovaniemi city theatre – renovated between 2011-25. The buildings have much in common: all being theatres the uses and functions of the spaces are largely similar. All are modern landmarks of their cities designed by renowned architects [1], resulting in strict building conservational requirements. Furthermore, all theatres shared the common main reason for the renovation need: outdated technical systems. Despite the many similarities, however, the unique characteristics of each building became clear during the renovation projects, presenting unique challenges to the design team – not the least acoustically.

The paper is intended as a case study of the five theatre renovations from an acoustician's point of view. The authors' aim is to present the properties of the theatres that have affected their acoustical design, highlighting the similarities and dissimilarities, as well as to share insights into the encountered acoustical issues and their remedying principles. The main source material for the paper consists of the architectural- and other plans and design material of the theatres, to which the authors have had access having been involved in their renovation.

The purpose is to give the reader a general overview of the theatres and their properties, rather than immersing in too much technical details. The renovations have largely been extensive projects, thus the presentation in the paper is inevitably cursory. While the main emphasis is on the auditoria and their room acoustics, factors related to sound insulation are also discussed. Finally, some insights are given into the acoustical design of studios and HVAC-systems.

2 Case studies

2.1 Overview of the theatres

The basic information of the five theatres is given in Table 1. All the theatres were built between the 1960s and 1980s, with possible later extensions noted in the table. The size of the buildings varies drastically, with Helsinki city theatre being the largest and Seinäjoki the smallest. All the theatres contain several auditoria, of which the most significant ones are presented. The theatres were built in an era during which acoustical consultancy, albeit still in its infancy, already existed as a specialized design field in Finland. Consequently, most of the theatres were originally designed with the help of acoustical expertise. Table 1 gives the original acoustician in the cases that the authors were able to trace the information from literature sources.

Table 1: Basic information of the five theatres. Sources: architectural plans, [1, 2, 3].

	Seinäjoki	Lahti	Jyväskylä	Helsinki	Rovaniemi
Built	1987	1983	1982	1967/89 1)	1961/72/75 2)
Renovation	2014-20	2019-22	2022-25	2013-17	2011-15 3)
Size [brm^2]	4560	11 720	10 100	27 260	12 300
Architect (orig.)	Alvar and Elissa Aalto	Pekka Salminen	Alvar and Elissa Aalto	Timo Penttilä	Alvar and Elissa Aalto
Architect (renov.)	Talli Architects	Sitowise	UKI Arkkitehdit	LPR Architects	A-Konsultit
Acoustician (orig.)	Alpo Halme	Alpo Halme	Unknown	Paavo Arni & Co, Alpo Halme 4)	Unknown
Acoustician (renov.)	Sitowise	Sitowise	Sitowise 5)	Sitowise (Helimäki Acoustics) 5)	Sitowise (Helimäki Acoustics)
Auditoria (no of seats)	Alvar (429), Elissa (110), Verstas (60)	Juhani (650), Eero (250), Aino (81)	Suuri näyttämö (550), Studionäyttämö (177)	Suuri näyttämö (920), Pieni näyttämö (360)	Tieva (445), Kero/studio (145)

1) The original theatre was finished in 1967 and the extension Studio Elsa in 1989. A small extension was also added in the 2017 renovation.
2) The first phase was finished in 1961 and the second phase, consisting of the theatre and congress halls, in 1972 and 1975.
3) A smaller indoor air quality renovation was carried out in 2018, in which Sitowise was not involved.
4) Paavo Arni & Co was responsible for the acoustics of the original theatre and Alpo Halme for the later extension Studio Elsa.
5) Sitowise was not involved in the preliminary design phase (hankesuunnittelu), but from there onwards.

2.2 Auditoria

2.2.1 Basic properties

Table 2 presents the basic properties of the theatres' two largest auditoria regarding, e.g., their size and seat capacity. The information is mainly based on the architectural drawings. The volumes given in the table are approximate values including the auditorium only, without the flytower. The reported reverberation times are mostly calculated values based on room acoustical modelling, and in a few cases measured values are provided.

To give a general picture of the differences in size, geometry and layout of the five theatres, Figure 1 presents a simplified section cut and floor plan of each of the theatre's main auditorium. The main hall in Helsinki is the largest, Lahti, Jyväskylä and Rovaniemi come as second with roughly the same size, while Seinäjoki is the smallest. All are essentially proscenium -type theatres (proscenium opening as dotted line in Figure 1), with the front stage slightly protruding into the auditorium in some cases. The main halls in Seinäjoki, Jyväskylä and Rovaniemi are fan-shaped, Lahti is rectangular, while Helsinki is a mix of the two with splayed side walls. The stage configuration and layout vary.

Table 2: Basic properties of each theatre's two largest auditoria. Sources: architectural plans, authors' calculations, [1].

	Seinäjoki	Lahti	Jyväskylä	Helsinki	Rovaniemi
Large auditorium					
Name	Alvar	Juhani	Suuri näyttämö	Suuri näyttämö	Tieva
Volume [m^3] 1)	2300	3400	3200	4900	3400
Seats	429	650	550	920	445
Volume/seat [m^3]	5,4	5,2	5,8	5,3	7,6
Proscenium W x H [m]	12,8 x 4,6	11,9-23,6 x 8,4	13,3 x 6,1	22,5 x 9,3	12,8 x 5,7
Flytower height [m] 2)	17	23	20,7	28	15

Stage configuration	Main + back + 1 side stage	Main + back + 2 side stages		Main + back + 1 side stage	Main + back + 2 side stages	Main + 2 side stages
RT (before/after) [s] 3)	1,3 / 1,3 (calc./calc.)	n/a		1,1 4) / 0,5 (meas./calc.)	1,2 / 1,1 (calc./calc.)	1,5 / 0,7 (meas./calc.)
Small auditorium						
Name	Elissa	Aino	Eero	Studionäyttämö	Pieni näyttämö	Kero
Volume [m^3]	700	780	3700	1560	5300	750
Seats	110	81	250	177	360	145
Volume/seat [m^3]	6,4	9,6	14,8	8,8	14,7	5,2
RT (before/after) [s] 3)	n/a	0,7 / 0,5 (calc./meas.)	1,1 / 1,1 (calc./calc.)	- / 0,5 (calc.)	0,8 / 0,8 (meas./calc.)	1,5 / - (meas.)

1) Auditorium only, excluding the flytower.
2) Free height from the stage floor to the highest point of the flytower ceiling.
3) Reverberation time, average at mid frequencies 500-2000 Hz in unoccupied theatre before and after renovation. Room acoustic models adjusted to match a typical theatre setting (stage curtains etc.) and to ensure the comparability of results.
4) Reverberation time T20 measured in a typical orchestra setting with the stage partially covered with props/curtains, sound source on the front stage and the original virtual acoustics system turned off [4].

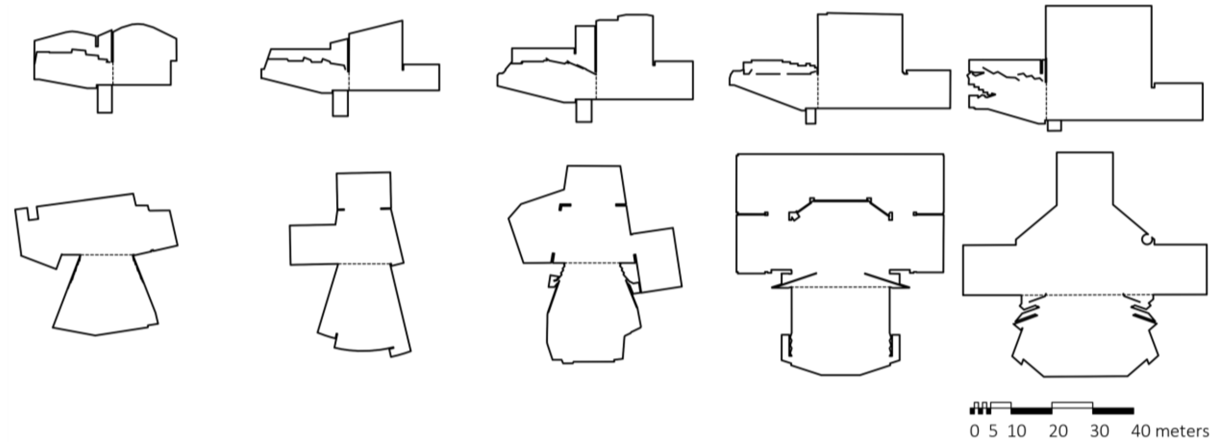

Figure 1: Section and plan of the main auditorium from left to right: Rovaniemi, Seinäjoki, Jyväskylä, Lahti, Helsinki.

2.2.2 Room acoustics

A significant emphasis in all the five renovation projects was in updating and improving the functionality of the primary and secondary theatre halls. The key acoustical issues and applicable solutions were investigated using room acoustical modelling of the auditoria, utilizing measurement data whenever available. Feedback from the in-house audio technicians and other theatre staff served as crucial background information to acoustical design, pinpointing the auditoria's user-experienced acoustical problems to be addressed in the renovation.

According to user feedback, the auditoria in all the five theatres relied rather heavily on live (unamplified) sound with reinforcement utilized only as additional support when needed. Room acoustically, the halls did not much resemble a traditional black box -type space with dead acoustics but, instead, the acoustics was generally more live, enabling also unamplified sound to carry in the hall (not without problems, however, as later discussed). This is seen in the pre-renovation reverberation times as well (Table 2), which are closer to a typical multipurpose hall than a traditional speech theatre, the former having a recommended reverberation time of 1,2…1,5 s at 500-1000 Hz and the latter, respectively, 0,9…1,1 s [5]. Indeed, most halls were used for other purposes also, such as congresses and concerts, despite the main use being theatre. Concerning the auditorium volume, most of the main auditoria fall in the range typically recommended for a speech theatre, 4…6 m^3/seat [5], except for the somewhat larger Rovaniemi's Tieva hall which also had the highest reverberation time before the renovation.

#1 Rovaniemi

One of the well-known aspects of Alvar Aalto's performance spaces is that, while their architectural value is undisputable, their acoustical quality is often not at the same level. This goes for the three theatre halls of this study as well – all had acoustical challenges related to factors such as poor audibility and stage support. The main hall, Tieva, in Rovaniemi was originally designed as a multipurpose space suited to – not only conventional theatre – but also concert and congress use. The reverberation time (1,5 s at mid frequencies), while suitable to concerts, was too high for speech theatre use. Indeed, the main acoustical issue of the hall had to do with poor speech audibility and clarity in the audience. In addition to too high reverberation time, this was due to insufficient amount of early reflections from the wall and ceiling surfaces, caused by the problematic geometry of the hall. Pre-renovation speech intelligibility was further worsened by the old ventilation system, which produced excessive background noise ($L_{A,eq}$ measured at 36…38 dB). The hall has a wide, asymmetrical stage, a relatively low and narrow proscenium, and a steeply fan-shaped auditorium. The side walls are angled in such a way that useful reflective surfaces are only visible from the front half of the stage – as a result, the rear of the stage is acoustically in a separate room from the audience.

The acoustic issues were tackled with by adding absorptive surfaces in the auditorium, bringing the reverberation time down to 0,7 s, and lowering the background noise level. Additional ceiling reflectors were also designed, although – due to architectural and theatre technical constraints – only three reflectors were ultimately realized. All in all, the speech intelligibility in Tieva did indeed improve, which was demonstrated by a notable rise in the modelled speech transmission index (STI) in the auditorium: from 0,43…0,55 before to 0,62…0,66 after the renovation. At the same time, the changes did also mean that the original live and reverberant nature of Tieva's acoustics was lost, and performing in the hall became more dependent on sound amplification.

#2 Seinäjoki

The main hall, Alvar, in Seinäjoki bears a strong resemblance to Tieva in Rovaniemi, except the proscenium is even narrower and the stage is even deeper. However, consensus at the time was that the hall works extremely well and no improvements to room acoustics were requested by the users. The only item on the users' wish list for the renovation was a new mixing booth to the rear of the auditorium. This required building a diffusive/absorptive structure behind the booth, to remove any unwanted reflections caused by the curved rear wall. Additionally, all mineral wool-based absorbers were replaced using synthetic materials due to indoor climate reasons. The synthetic absorption material selected for the hall required an air gap behind the material to retain adequate mid-range absorption near 500 Hz – this was incorporated into architectural plans bearing in mind the visual impact on partially protected ceiling and wall surfaces. Beyond this, the room acoustics of neither the primary hall Alvar nor the secondary hall Elissa, saw any major changes during the renovation. Both halls also retained their multifunctional characteristic following the renovation.

#3 Jyväskylä

The third theatre hall, also by Aalto, is in Jyväskylä. The pear-shaped main hall suffered from large variations in sound quality between different areas of the auditorium. Especially poor were the seats near the mid part of the auditorium where live sound from the stage receives minimal support from the side walls – partly due to the fan-shape geometry of the auditorium, but also because the front parts of the walls are covered with a wooden lath structure typical to Aalto, which breaks up any useful reflections. The original idea in the renovation was to preserve the live acoustics of the hall and improve upon it using ceiling reflectors etc. – this approach was, however, ultimately rejected by the client. Instead, it was decided to base the functionality of the hall on a new virtual acoustics system used, among other things, to amplify the actors' voices and create a sense of reverberation for concert uses. Acoustically, this requires a rather dead acoustics in the auditorium – this is to be achieved re-coating the walls with seamless acoustic plaster, while the original particle board dropdown ceiling will be replaced with an absorptive one to add necessary mid to low frequency absorption. This is a delicate task indeed, considering that the hall's original visual appearance is to be preserved as well as possible. A separate bass trap will also be built above the main stage close to the subwoofers.

Overall, the changes (if realized, the project still being in progress at the time of this paper) will drastically change the acoustics of the hall, bringing the reverberation time down to about 0,5 s from the original 1,1 s. The idea of a virtual acoustics is nothing new in Jyväskylä theatre: such a system was installed in the main hall during the 1990s to increase the reverberation time for classical concerts [4]. However, the system was mainly left unused, as the users did not much like the resulting acoustics – hopefully the new system will have better success.

The secondary hall (Studionäyttämö) will also be completely redesigned, with the total volume extended and room acoustics updated. The hall will house a similar virtual acoustic system as the main theatre hall and the reverberation time will be in the same ballpark as well.

#4 Lahti

The renovation project in Lahti, begun in 2019 with upgrades for the smaller Aino and Eero auditoria. The in-house sound technician specifically requested that both halls remain usable without amplification. This meant that instead of an acoustically dead black box design, there was need to ensure enough sound reflecting support surfaces remain for the performers as well as the audience. Acoustical design was also strongly affected by architectural considerations – although lacking an official listed building status, the modernistic concrete-brutalistic interior architecture of the halls was to be preserved, hence all the visible acoustical alterations needed to be carefully studied with the architect.

For the smallest auditorium, Aino, the plan was to leave parts of the side walls reflective and only add absorption required by the loudspeaker system. Acoustical modelling was used to make sure the first reflections from loudspeakers were hitting either absorptive or diffusive surfaces in order to minimize unwanted colorization, while leaving enough reflective surfaces in between to provide support for stage sound. Some diffusive surfaces were added as well – for example, wood beam lattices were mounted to the crevasses next to the stage and angled plasterboards added to the sides of the stage itself. The purpose of these diffusor structures was to break any flutter echoes between the opposing concrete walls.

This was the plan. However, at some point during the construction, the plans were changed – without consulting the acoustician – and as a result, the side and stage walls had been all but covered with absorption material. Moreover, the rear wall was left completely untreated. Reverberation time in the finished hall was measured at 0,5 seconds at mid frequencies, which was less than recommended. Instructions to remedy the situation were given, but to the knowledge of the authors no changes have been made as user feedback has been positive. It seems that the main reason why the hall works well enough, despite the room acoustical shortcomings, is it's small size.

The second smallest auditorium, Eero, saw major structural changes: seats were replaced, stage and floors rebuilt, an air intake chamber constructed and parts of the technical rooms and walkways extended to better utilize the available space. Room acoustics overhaul followed the same formula as in the Aino hall: reflective support surfaces were added to side walls where possible, while making sure amplified sound works by adding absorption in the areas hit by the first reflections from the speakers. The reverberation time was kept at the same level as before the renovation (1,1 s). No final measurements were made, but user feedback has been positive. The largest hall, Juhani, was not included in the 2019-22 renovation project – however, with its boxy and flutter-echoey auditorium and unusual stage configuration, it appears to have some challenges in store for the acoustician in the coming renovations.

#5 Helsinki

The renovation of Helsinki theatre saw extensive acoustical improvements to both the main and small theatre halls. These were, however, executed within strict architectural constraints, as the halls constitute a valuable part of the original interior architecture by Timo Penttilä. In the main hall, the users had experienced unwanted sound reflections on stage as well as in the mixing booth at the back of the auditorium. For the stage, the problem was mainly caused by late reflections from the hard surfaces on stage walls, doors and the flytower. This was remedied by adding absorption material to some of the wall and ceiling surfaces of the main stage, which were determined using acoustic modelling. Additionally, a diffusive dropdown ceiling was built to both side stages to improve their diffusivity while still keeping stage support at a reasonable level. For the mixing booth, absorption was added to some of the untreated surfaces above and behind the work station. The overall absorption area added during the renovation was kept small enough to not much affect the reverberation time of the auditorium, which was already considered to be at a desirable level. Overall, the changes did indeed improve upon the user experience and feedback has been positive. [6]

The small theatre hall's main issue was with focusing caused by the steeply curved rear wall, combined with the ceilings and walkways. This produced surprisingly strong localized reflections near the middle of the stage (see Figure 2). The focusing was reduced by adding absorption to the rear wall and adjacent ceiling surfaces where possible and tilting the windows (shown in purple in Figure 2) in such a way that sound reflections became more scattered. Some focusing still remains but objectively the situation has improved significantly and user feedback has been positive here as well. [6]

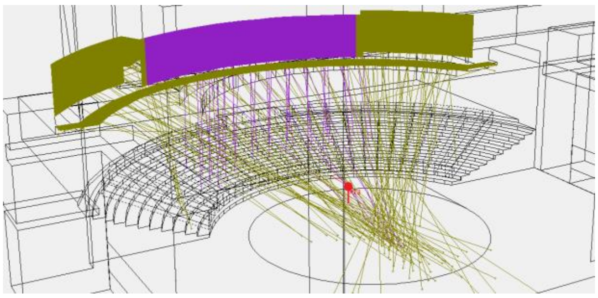

Figure 2. Illustration of the focusing effect on the stage of the small auditorium in Helsinki [6].

2.3 Sound Insulation

In addition to room acoustics, factors related to sound insulation played an important role in the renovations. Table 3 summarizes the sound insulation deficiencies reported by the users before the renovation, "x" indicating that sound insulation between the corresponding spaces was found poor or inadequate. In most of the cases sound insulation measurements were carried out at the beginning of the project to verify the starting point and to yield more detailed information concerning the possible reasons for the deficiencies. The sound insulation problems mostly had to do with airborne sound insulation, although some structure-borne and impact sound insulation issues were also found.

It is noteworthy that in all the theatres the user had reported problems with inadequate sound insulation in the auditoria – although one could guess these to be the spaces to which the most acoustical effort was originally invested. In many cases the reported problems had to do with doors or openable walls, which is not surprising as the stage in all the investigated main auditoria has access to surrounding (noisy) workspaces – or even another auditorium – via large sliding or ascending doors/walls. Issues were also partly related to inadequate airborne or structure-borne sound insulation of old wall and floor structures – these were improved upon within structural, architectural and economic constraints.

Table 3: User reported sound insulation problems before the renovation.

Sound insulation deficiencies	Seinäjoki	Lahti	Jyväskylä	Helsinki	Rovaniemi
Halls:					
- large – small auditorium	x	- 1)	x	x	x
- auditorium – foyer spaces	x	- 1)	-	x	x
- auditorium – workshop spaces	x	- 1)	x	x	-
- auditorium – outside	-	- 1)	-	x	-
Other spaces:					
- Studios (recording and control rooms)	x 1)	- 3)	x 1)	x 1)	x 1)
- Rehearsal rooms	x	- 3)	x	x	x
- Offices (actors' rooms, management etc.)	-	- 3)	-	x	x
1) Large auditorium was outside the renovation scope, thus the current situation is unknown.					
2) New studios were built during the renovation replacing the old ones.					
3) Spaces outside the scope of the renovation.					

2.4 Studios

The studios in Seinäjoki, Jyväskylä, Helsinki and Rovaniemi were largely redesigned, because the old spaces were technically outdated and had numerous acoustical problems. There was also need to reconsider the layout of the studios and their placement in the building to improve functionality. An important part of the design process was finding out the true needs and requirements of the users and client to avoid over-doing the acoustics and, ultimately, save costs and resources – as an example, whether the studio space in question really needs heavy duty box in box structures and highly optimized room acoustics, or whether a more modest approach to acoustical design is sudfficient.

Implementing acoustically and technically modern studio facilities in the old buildings was challenging in many ways, and the acoustical solutions needed to be designed case by case within rather strict constraints. One of the main issues was related to space, or rather to the lack of it – the existing spaces where the new studios were to be fitted were rather small to begin with, and there were serious height constraints dictated by the existing load-bearing floor structures. This meant that the size and geometry of the studios were rather fixed, while also setting constraints to the thickness of the acoustical structures, affecting the achievable level of sound insulation and room acoustics (absorption, diffusion).

The general design principle of the sound recording studios, requiring the highest level of insulation, was to use box in box double structures with as high thickness and mass as possible, as well as heavy-duty double windows and doors. The target level of sound insulation for such studios was typically set at $R'_{w} \geq 65$ dB / $L'_{n,w} \leq 43$ dB. An essential criterion in room acoustical design was to ensure a reflection-free-zone to the recording area, while the requirements for reverberation time and early reflections were mostly set based on commercial audio production standards [7].

2.5 Building Service Systems

Most of the renovations were initially motivated by the necessity to upgrade the building service systems (HVAC) to meet modern requirements. In practice this meant, for example, equipping the buildings with mechanical ventilation systems having considerably higher airflow rates than in the original situation – all the while ensuring that the acoustically critical spaces remain silent. Moreover, a maze of new ductwork, pipes, electrical wiring etc. had to be added to the existing spaces while meeting high sound insulation requirements. Adding up the general constraints of an old building – confined spaces for ventilation ducts and machinery, poorly sound insulating structures and strict building conservational demands – meant that the sound insulation and noise control of HVAC-systems was not without challenges.

The requirements for HVAC-noise levels were typically set at $L_{A,eq,T} \leq 25$ or 28 dB for the auditoria and $L_{A,eq,T} \leq 20$ or 25 dB for the studios. In the auditoria, ≤ 25 dB was found to be achievable, but required careful consideration of all the noise sources in the hall (not just the ventilation) as well as sound insulation improvements to the surrounding noisy technical spaces. As for the studios, the experience was that ≤ 25 dB represents a good-enough level in most cases, although the experience from Helsinki showed ≤ 20 dB to be feasible as well, should better quality be desired. All in all, achieving the HVAC noise requirements required a considerable effort from the acoustician, not only in the design but during construction too, from providing acoustical consultation to the HVAC contractors to doing quality-control measurements.

In most of the five theatres, MagiCAD for AutoCAD -software was used to aid the calculation of ventilation sound levels in acoustically critical spaces. Although mainly used by HVAC-designers, in an acoustically demanding renovation such as these the software was found to be a useful tool for the acoustician as well. However, the acoustic calculation in the software was found to be rather simplified, not taking account of all the factors properly that affect noise in a ventilation system – thus, acoustician's expertise and complementary calculations were required in the design [6].

3 Conclusions and Discussion

This paper discussed five Finnish modern theatre renovations which, while sharing many similar characteristics, also presented unique challenges to the acoustician. All in all, perhaps the most important common lesson from the projects was to listen to the users – their experiences, feedback and requests – and adjust the acoustical requirements and solutions accordingly. This was found to be especially important for the room acoustics of the theatre halls but applied to sound insulation and studio design as well.

An important realization concerning the auditoria was that they are, after all, not just theatre halls per se but, rather, multifunctional venues (often the only ones in the municipality) that house all types of performances from conventional theatre to musicals, conferences and town hall meetings. From an acoustical and audio-technical standpoint, this requires a somewhat hybrid approach in which the loudspeakers provide the necessary support, all the while the acoustics also enable natural unamplified sound to carry in the space with ample room for the sound to breathe above and around the audience. As the case with the Jyväskylä auditoria indicates, however, this hybrid functionality is being phased out with multichannel soundscape implemented by virtual acoustic systems, which require an acoustically dead hall to work properly. Whether favoring such loudspeaker-dependent virtual acoustics over natural unamplified acoustics is a good thing or not, remains to be seen – in an architecturally protected theater hall with limited means for acoustical improvements, it is nevertheless a tempting option.

Regarding sound insulation, a lesson to learn is that it should not be forgotten in theatre renovations – on the contrary, while theatres are often considered mainly from the room acoustical perspective, the experience showed that improving the sound insulation is in many cases as important to the users. The most effective means for improvement is, however, not so much acoustical as it is architectural: laying out the spaces in such a way as to minimize noises from noisy spaces reaching the spaces requiring silence. As an example, any noise from the workspaces adjacent to the theatre hall should always pass thru multiple well-damped hallways and airtight doors before reaching the stage or audience. While the means for such improvements in an existing theatre are naturally limited, they should be utilized to the largest extent possible.

Incorporating studios into an old theatre building is inevitably a compromise between space requirements and economic viability. Finding out the true uses of the studio spaces and the needs of the user becomes essential: an expensive box in box structure is overkill, if all the client needs is a better-than-average office room for light monitoring and editing purposes. The heavy-duty box in box structures are best reserved for sound recording studios and similar spaces, where truly needed. As for the noise control of building service systems, the experience showed that, in a theatre renovation, it is a field which cannot be left in the hands of the HVAC-engineer alone, but requires careful attention – and, indeed, design effort – from a skillful acoustician as well.

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Applications of a Zone to Zone Reflector Optimisation Routine

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Traditional reflector optimisation is based on point to point or point to zone transmission. A new optimisation routine has been developed to shape a reflector for zone to zone transmission. The routine includes the NSGA-III genetic algorithm, a so-called "many" objective algorithm, i.e. it can optimise more than 3 objectives at once. The objectives that can be optimised include: an even distribution of reflections in the receiving zone, an even distribution of sources in the source zone, reflector efficiency, Lateral Energy, Gain and Speech Intelligibility. Optimisation runs were performed with combinations of 2, 3 or 4 of these 6 objectives, but never more. Computer speed considerations limit the algorithm to 1st and 2nd order reflections. Similar to previous studies, calculation for the Lateral Energy objective employed the Single Lateral Fraction (sLF) parameter. A new single reflection speech intelligibility fitness function is introduced, using Lochner and Burger's energy integration curves. Applications of the new routine are presented for two experimental studies: a 1840 seat thrust stage theatre and a side wall balcony soffit addition to a 626 seat multi-purpose venue.

1 Introduction

The importance of early reflected energy, that had been established in the academic world in the mid to late 20th century led to a paradigm shift in the acoustical design industry, essentially from the 1980s onwards. It became apparent that if, for example, one wanted to design a room for good speech intelligibility, the tools were no longer absorption coefficients and reverberation time equations but, rather, geometric studies that would deliver reflected sound within the first 50 ms [1,2]. If the room was for music, the lateral energy thesis [3,4], established in the early 1970s, dictated a design that would cast reflections to arrive at the listeners from the sides. By the end of the 20th century, reflector design had replaced reverberation times as the acoustical consultant's primary stratagem of choice.

Since that time many efforts have been made to optimise acoustic reflectors. However, few of these attempts, if any, have embraced the modern multi-objective optimisation techniques currently being applied in so many other disciplines. This is somewhat perplexing given the reality that the perception of sound in a room is a multi-dimensional experience. Central to the application of multi-objective optimisation is the concept of Pareto Optimisation. The first part of this paper will present a brief history of modern multi-objective optimisation then introduce this concept of Pareto Optimisation and how it is applied in Genetic Algorithms. The second part will then demonstrate the application of these techniques and how they are used to optimise surfaces casting reflections, not from a single point source to a zone of receivers but, rather, from a zone of sources to a zone of receivers.

2 Multi-objective Optimisation Methods

2.1 Design Space and Objective Space

Multi-objective Optimisation studies work in two different mathematical settings, often known as the Design Space and the Objective Space. In acoustic reflector optimisation, the Design Space is the 3-D geometric space inside which the reflector is built. The performance of that reflector is then assessed with mathematical formulae designed to encourage the goals of the optimisation. In Genetic Algorithms, these are often referred to as the Fitness Functions. The results of these calculations are expressed graphically in what is known as the Objective Space. An example is shown in Figure 1.

This graph comes from a classic challenge in concert hall design, the desire to optimise Reverberance and Clarity at the same time. Note how the axes in this graph do not quantify Early Decay Times (EDT) and Clarity (C80) in the traditional sense but, rather, as Fitness Functions. In this case, the Fitness Functions are designed to be minimised. So, for example, if the desired goal for a room is an EDT of 2.25 seconds and one of the designs actually returns a value of 2.25 seconds, it would get a perfect score. Its Reverberance Fitness would be 0.0. Likewise, if the EDT was either higher or lower than the design goal of 2.25 seconds, it would get a Reverberance Fitness closer to 1.0.

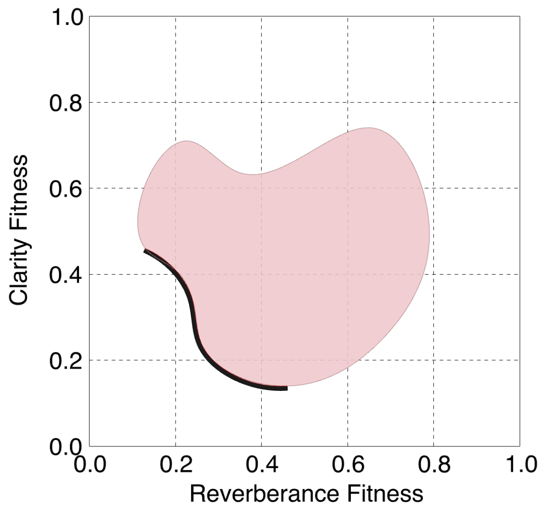

Figure 1 Objective space with a shaded zone indicating the extent of the Fitness Function calculation data points. The Pareto Front shown as a thick black line.

In Figure 1 the Objective Space is the full range of the graph, from (0, 0) to (1, 1). The shaded area is what is known as the Feasible Objective Space for this particular collection of concert hall designs. The Fitness Functions in this demonstration have been designed such that their "best" values are the ones closest to the origin. Any Fitness Function points on the thick black curve are said to be non-dominated. Which is to say that none of the other points in the set shown in shaded area are dominate over these points in both dimensions. The points on the thick black line form what is known as the Pareto Optimal Front or, more conveniently, just the Pareto Front.

2.2 Weighted-sums Optimisation

One of the earliest modern methods of Multi-objective Optimisation is the system of weighted sums, also known as Linear Aggregates or the Linear Combination of Weights. The major limitation of the Linear Combination of Weights method is that it cannot find the optimum values on what is known as a non-convex Pareto Front. This is a serious weakness when one is trying to perform a Multi-objective Optimisation. A non-convex Pareto Front is demonstrated in Figure 2. The front in this image has a non-convex region between points A and D with a nadir near the coordinates (0.22, 0.38).

The iterative process of finding a 2-Dimensional minimum using the Linear Combination of Weights method is equivalent to moving a line of slope $-w_1/w_2$ downwards across the Objective Space, as shown in Figure 2. The line starts out from the upper right and moves downwards diagonally until no feasible fitness point is below it and at least one fitness point is on it. In this example, the search is successful when points A and D are found. However, Points B and C, the points inside the non-convex part of the Pareto Front, cannot be found in this manner. As the $-w_1/w_2$ line moves past points B and C, there will always be feasible points below both of them and therefore they are not detected as part of the Pareto Front.

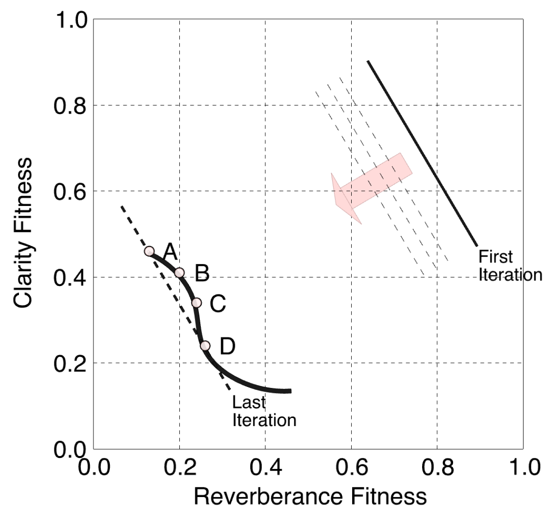

Figure 2 Iterative search for minima using the weighted sums method. A line of slope -w1/w2 is iteratively moved downwards. It can find the minima at points A and D but not and B and C.

2.3 Pareto Optimisation

Pareto Optimisation addresses this problem. The core of Pareto Optimisation is the concept of dominance. In a two-objective optimisation problem, dominance is essentially a pair-wise comparison of the points in the Objective Space.

The concept will be explained graphically, using the Reverberance/Clarity optimisation problem, remembering that the goal is to minimise the two Fitness Functions. A set of three points will be considered, Points A, B and C, as shown in Figure 3.

Consider first Points A and B and ask which one is better. Point A is better than Point B insofar are Reverberance Fitness is concerned. That is to say, it is closer to the origin on the horizontal axis. But Point B is better than Point A for Clarity Fitness. It's closer to the origin on the vertical axis. It turns out that neither one of the points is better than the other when both objectives are considered. Or, put another way, neither one of the points dominates the other.

Consider now Points B and C. Point C is closer to the origin than Point B on both axes. Point C has better Clarity Fitness and better Reverberance Fitness, when compared to Point B. As a result, Point C is said to be dominate over Point B. In the set that has been circled, neither Points A or B dominate over Point C so, for this small set, Point C is said to be the non-dominated point.

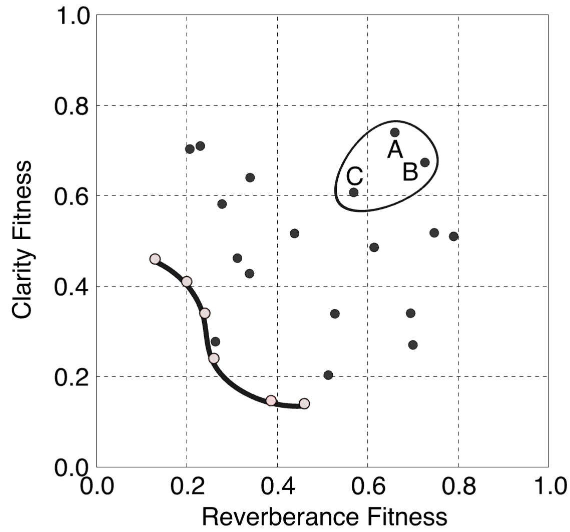

Figure 3 Demonstration of dominance. Point C dominates Point A and B. The grey points on the thick black curve dominate the entire set and are called Pareto Optimal.

If this procedure is expanded to the entire set of points shown in Figure 3, the non-dominated front will be found. If, indeed, these are the best points that are physically possible given the restraints, it is referred to as the Pareto Optimal Front. The Pareto Optimal Points for the complete set shown in Figure 3 are indicated by shaded grey dots on the thick black line.

A Pareto Front may be thought of as the barrier between the feasible and the infeasible. On one side are the data points representing all the possible solutions to the multi-objective question that has been posed. On the other side are solutions that cannot be built within the limitations of the question. In reflector design, the "limitation" is the geometric space available inside the room. Along the curve of the Pareto Front are the points that represent the best possible compromise between the competing objectives, sometimes referred to as the trade-off solutions.

3 Zone to Zone Reflector Optimisation

The author has recently developed a routine to optimise an acoustical reflector based not on a traditional point to point or point to zone design paradigm but, rather, a zone to zone optimisation. The method is explained in ref. [5] and briefly summarised here. The routine is based on the assumption that a listener located in a receiver zone wants to "see" as much of the source zone as possible looking at the acoustical reflector as if it were an optical mirror. Reflections are cast from a receiver point towards a given reflector geometry, hoping that it will generate an intersection with the source zone. This is done for several receiver points in the receiver zone. Then the reflector geometry is perturbed and the procedure is repeated several times. A Genetic Algorithm (GA) is used to develop an optimised geometry for the reflector. In this case the GA employed was the Non-dominated Sorting Genetic Algorithm (NSGA-III) [6]. The code was developed by the author based on the C# version published by Chan [7].

Optimisations in the GA runs that will be presented here were governed by four different Fitness Functions:

(i) **Area Fitness**. Quantifies how much of the source zone a receiver can see looking at the reflector as it if it were a mirror. The objective being to increase the listener's view of the source zone as much as possible. Please see ref.[5].
(ii) **Spreading Fitness**. Quantifies the distribution of the landing points in the source zone for reflections cast from a point in the receiver zone towards the reflector and from there towards the source zone. The objective being to create a uniform view of the source zone from the listener's position in the receiver zone. This too is described further in ref. [5].

(iii) **Single Reflection Lateral Fraction Fitness**. By necessity, all the optimisations presented here are based on 1st and 2nd order reflections. Protheroe and Day have found a reasonable correlation between a single reflection Lateral Fraction (sLF) in free field [8] and a standard ISO 3383 Lateral Fraction [9]. The sLF Fitness Function has been developed accordingly. The objective being to increase the Lateral Fraction in the receiver zone. Please see ref. [10].

(iv) **Single Reflection Lochner Burger Fitness**. A new fitness function to encourage good speech intelligibility is introduced here and will be described below.

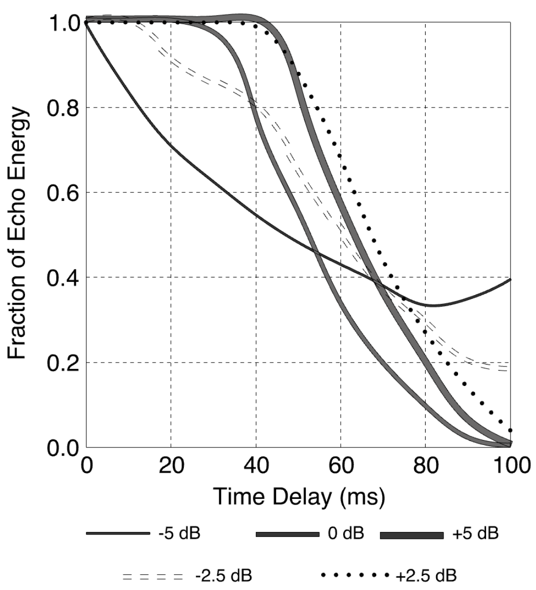

Figure 4 Fraction of reflected energy (α) integrated with direct energy for various reflection levels (L), as developed by Lochner and Burger [11].

In 1961, Lochner and Burger presented a family of curves describing how reflected energy is integrated with the direct sound to improve speech intelligibility [11]. They developed what some considered to be a more refined version of Theile's 50 ms Distinctness function (D50) [2]; the so-called Lochner Burger Ratio (LBR). Where the D50 ratio integrates energy according to a step function – on either side of 50 ms a reflection is either good or bad with nothing in between – the LBR integrates the reflected energy gradually with curves based on the actual measured psycho-acoustic behaviour. Analog measurement technology at the time, however, could not easily mimic Lochner and Burger's integration curves so the LBR quantifier never gained the popularity of the much simpler D50 ratio. The family of curves that they developed, shown in Figure 4, are still useful however. They have been employed here to develop what will be called the single reflection Lochner Burger Fitness Function, referred to more simply as the sLB Fitness. Its calculation is performed as follows.

Although the reflector is optimised using reflections cast from the seats in the receiver zone towards and looking for the source zone, it is assessed in the normal fashion, i.e. sound generated in the source zone, directed towards the receiver zone. For each reflection cast from a source point that intersects a receiver surface, its Level (L) with respect to the direct sound is calculated. It is then weighted by the Lochner Burger curves, shown in Figure 4. These curves have been fitted to a family of 5th order polynomial regressions, as shown in Equation (1) and Table 1. So, for example, if a reflection cast from a point in the source zone successfully intersects a receiver surface, and its Level (L) with respect to the direct sound is -1 dB, the coefficients in the middle column of Table 1 would be inserted into Equation (1) to determine its integration weighting (α).

$$\propto = at^5 + bt^4 + ct^3 + dt^2 + et + f \qquad (1)$$

where: a, b, c, d, e & f are the coefficients shown in Table 1

t is the time delay of the reflection with respect to the direct sound (in seconds)

Table 1
Coefficients for Equation (1)

	Reflection Level (L) with respect to direct sound (dB)				
	$L \le -3.5$	$-3.5 < L \le -1.25$	$-1.25 < L \le 1.25$	$1.25 < L \le 3.75$	$L > 3.75$
a	80128	-808494	-2584722	-1579327	-1545009
b	2316	210588	610630	425790	449919
c	-2640	-17370	-46415	-37307	-43735
d	290.41	465.81	1068.68	1034.06	1461.51
e	-18.98	-10.64	-6.53	-8.25	-14.52
f	1.0007	1.0000	0.9989	1.0000	1.0100

This is then used to calculate the weighted relative energy for that particular reflection, which will be referred to as the Lochner Burger "Weighted Energy" (W):

$$Weighted\ Energy = \alpha \left(\frac{r_o^2}{(r_i + r_r)^2} \right) \quad (2)$$

where: α is the integration weighting from Eqn. (1)
r_o is the direct path length
r_i is the incident path length
r_r is the reflected path length

This process is repeated for each successful reflection cast from a given source zone point to the receiver zone, then the entire procedure is repeated for every other source point in the source zone. (In the examples to be shown below, there were 12 point sources representing the source zone in the Stratford model, 16 in the P.C. Ho model.). The entire procedure is summarised in Equation (3).

$$sLB\ Fitness = 1 - \frac{1}{N_{sources}} \sum_{i=0}^{N_{sources}} \frac{1}{N_{reflections}} \sum_{j=0}^{N_{reflections}} W_{i,j} \quad (3)$$

where: $N_{sources}$ is the number of point sources in the source zone
$N_{reflections}$ is the number of reflections that have been cast from the i^{th} source and have successfully intersected the receiver surface(s) in the receiver zone.
$W_{i,j}$ is the Lochner Burger Weighted Energy as calculated in Eqn. (2)

Finally, it should be noted that, unlike the sLF parameter developed by Protheroe and Day, there have been no laboratory measurements performed to qualify the single reflection Lochner Burger Weighted Energy (W) parameter. It is merely proposed here as a heuristic to guide the GA towards better levels of speech intelligibility.

4 Applications

4.1 Stratford Festival Theatre Experiment

The thrust stage format, which can be quite exciting theatrically, is acoustically problematic. The audience is laid out in a broad fan shape around the stage and there are typically little or no significant wall surfaces available for early reflecting surfaces. The progenitor of the thrust stage format is Canada's Stratford Festival Theatre. A recent study by O'Keefe et al. [12] tried to implement a Genetic Algorithm optimisation of the room's ceiling and back walls using large arrays of tilted, flat triangles. The intention of the experiment was to see if the computer could improve on the manually generated optimisation that the author had produced in a 2008 feasibility study. The results of the computer optimisation were mixed, at best. The experiment has been revisited, this time using curved Nurb surfaces for the reflectors on the ceiling and back walls instead of the flat triangles. The optimisation also takes advantage of the new Zone to Zone optimisation procedure, the (many-objective) NSGA-III algorithm and the newly proposed Lochner Burger (sLB) Fitness Function. Results of the optimisation at the time of writing are preliminary. Although the ceiling and back walls were optimised for multiple source locations, only one source location has been quantified at this time – a position at downstage centre. It shows improvements in both 50 ms Distinctness and Centre Times [9]. The former is shown in Figure 5. It shows improvements on the stalls level, where it is required, but not on the balcony, where it is not.

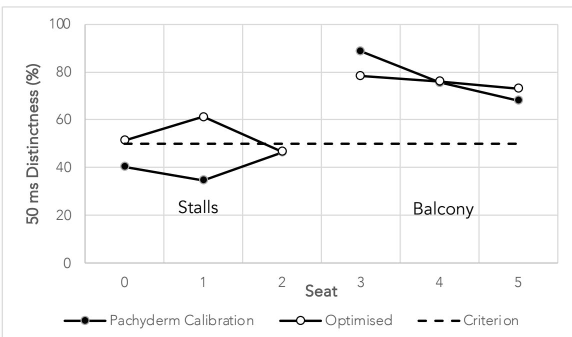

Figure 5 Calculated D50 values. Solid circles indicate the calibration of the Pachyderm model to measurements in the existing room. Open circles indicate the optimised values. Dashed line indicates minimum acceptable values for D50.

4.2 Balcony Soffit Experiment

Side wall balconies in a concert hall are thought to be beneficial in that they can provide laterally reflected sound to the audience that is not influenced by the low frequency attenuation of grazing incidence "seat dip" effects [14,15]. Recently, a two dimensional Boundary Element Model (BEM) study, carried out by Green et al. [16], found that low frequency reflections are significantly weaker inside the zone suggested by a specular reflection study. The spatial distribution of the lower frequency reflections, however, was much broader than suggested by the specular reflection calculations. Green et al. suggest that multiple bass reflecting surfaces might help to overcome the grazing incidence effects of seat dip. An undulating, Nurb based balcony soffit might be capable of providing the kind of multiple reflections that are required. With this in mind, a new version of the Zone to Zone optimiser was developed. One that is capable of calculating 2nd order reflections, i.e. off the side wall then the balcony soffit and eventually towards the audience.

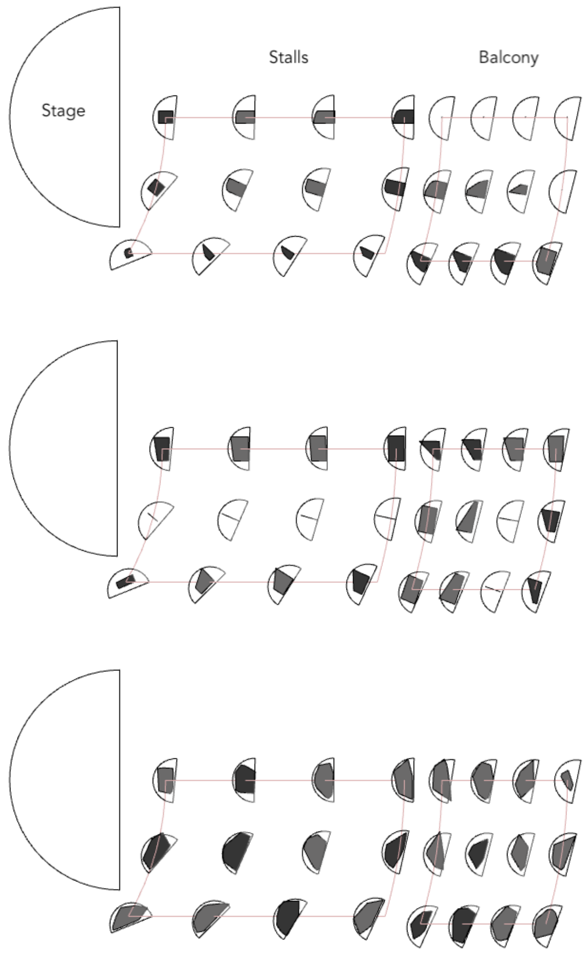

Figure 6 Views of the stage through the soffit reflector. Flat soffit reflector at top, cylindrical reflector in the middle, optimised reflector at the bottom.

An experiment was performed on a model of the 626 seat P.C. Ho Auditorium in Toronto, Canada. A room designed by the author in 2004 that does not currently have side wall balconies but has had one introduced into the experimental model. The 2nd Order Zone to Zone optimiser was used to study potential geometries for a new balcony soffit. Some of the results are shown in Figure 6 – which requires some explanation. The goal of the optimisation is for the listeners in all 24 of the chosen seats to "see" as much of the stage as possible through a 2nd order reflection off the soffit reflector. The open half circle at the left of each image is the stage source zone that the listeners are looking for. (Shaped in a traditional orchestral layout.) A scaled down version of the half circle source zone is superimposed on each of the 24 seating locations. The shaded areas inside of each of these is the zone of the stage that the listener in the given location can "see" through the reflector. The room is symmetrical about its centre-line so the images show only half of the house. Note also that although the end balcony overhangs a small part of the stalls in the actual room, the image has separated them for visual clarity.

The image at the top of Figure 6 is for a side wall balcony with a flat soffit underneath of it. There is no tilt, it is positioned at a 90º angle to the side wall. Most of the seats have only partial views of the stage and five of the seats on the balcony have no view at all. The middle image is for a semi-cylindrical soffit underneath the side wall balcony. The views have been improved in some seats but four seats in the stalls and 2 on the balcony have essentially no view at all. The optimised soffit, shown at the bottom of Figure 6, shows almost complete views of the stage in 23 of the 24 seating locations.

All of these calculations were, of course, based on specular reflections. They will, eventually, need to be confirmed with a BEM model or, perhaps, a scale model.

5 Acknowledgements

The author would like to thank Henrik Möller and Johan Hallimäe for their work on the previous attempt to optimise the Stratford Festival Theatre acoustics. Many, many thanks also go out to Arthur van der Harten for his help with the Pachyderm Acoustic calculations of the Stratford model.

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Room acoustic measurements in halls with electro-acoustic enhancement systems

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Electro acoustic enhancement systems are becoming more and more acceptable in particular for multipurpose halls. Throughout the last years, several systems have been implemented, both InLine- , Feedback- as well as combination-systems.

As part of the PhD studies of the authors, several halls with electro acoustic enhancement systems will be measured, using a scaled down version of the Virtual Orchestra, described by Pätynen et all. The halls measured will represents all 3 types of systems and are all found in norther/eastern Europe.

The object of the measurements is to get data for listening tests, however the measurement data has also been analyzed to examine the traditional room acoustic parameters.

In the paper we will present the preliminary measurement results for the halls measured and present an outline for the further work.

1 Introduction

So far 4 halls have been measured. The halls measured are:

- 500-seat multipurpose hall
- 500-seat opera hall
- 1800 seat multipurpose hall
- 1000 seat multipurpose hall

It is the intention to measure at least on hall for each of the currently available systems (within some geographical limitations). Furthermore, at least 2 halls are known to have proprietary system, essentially a modified MCR system based on a standard sound system processor.

2 Measurement setup

The measurements are done with the reduced Virtual orchestra, described in [1].

A swept-sine method is used and recorded in the hall using an A-format microphone and an omnidirectional microphone on top of it. The recorded spatial impulse responses are analyzed with Spatial Decomposition Method (SDM) [5] and visualized using mainly the spatiotemporal and time-frequency visualizations [4]. Traditional, acoustic parameters are calculated from the omni response.

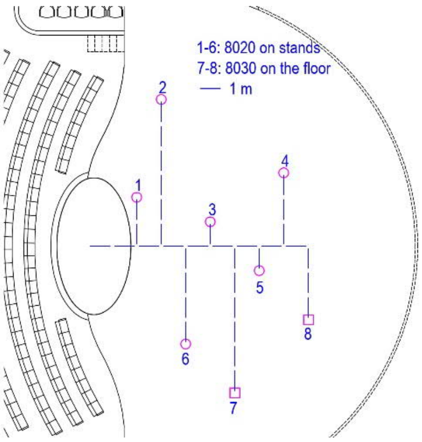


Figure 1: Loudspeaker setup

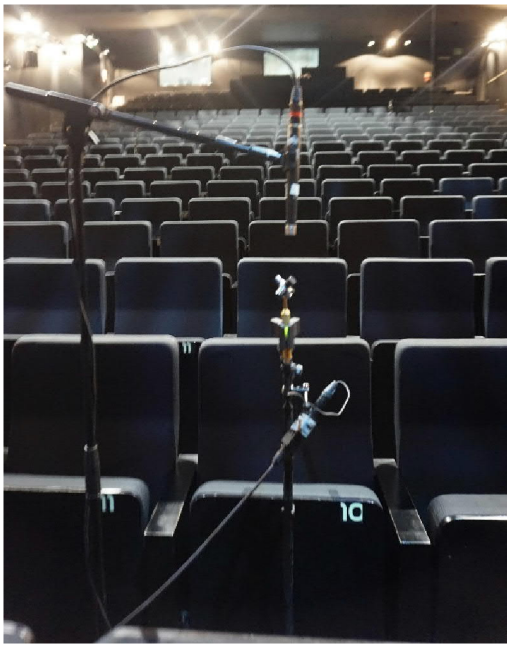


Figure 2: Microphone setup

As some of the acoustic enhancement systems are time-variant, it is clear that the measured impulse response is a “time-slice” and as the measurements are done with 8 separate sweeps, the impulse response will change between the different sweeps. However, this is same as the situation for the audience in the hall and therefore the method is considered to be appropriate. From purely technical perspective, an average of several repeated measurements could counteract the time-variant character of the electro-acoustic system, but with the expense of other time-variant natural properties.

In all halls, the receiver positions are chosen in accordance with ISO3382-1, and all audience areas of the hall is sampled.

3 Measurement results

The measured reverberation time for the halls for the different settings are shown in figures 3 -6.

In all cases, the change in both reverberation and clarity, are by far larger than what would be achievable with traditional variable acoustic means.

When looking at the different settings in the halls, it seems that the setups reflect a certain preference, not necessarily what would be expected from an acoustic design point of view. For instance, the Opera setting in the opera hall, has a significantly longer reverberation and lower clarity than the Concert setting. Also in this hall, the differences between the different setting are not as large as in the multipurpose halls.

In both multipurpose halls, all settings produced a quite significant bass boost, however, as can be seen figure 6, the change in clarity is not as pronounced.

One issue could be that in all the halls, the natural reverberation time at bass frequencies is higher than at mid frequencies.

The rise of the reverberation time at low frequencies (125 Hz and 250 Hz) without enhancement in the opera, and 500 seat and 1800 seat multipurpose halls, could be regarded as somewhat illogic design decision if the hall was designed for an electro acoustic enhancement system, however all the halls in this investigation are renovated which might explain the natural acoustics.

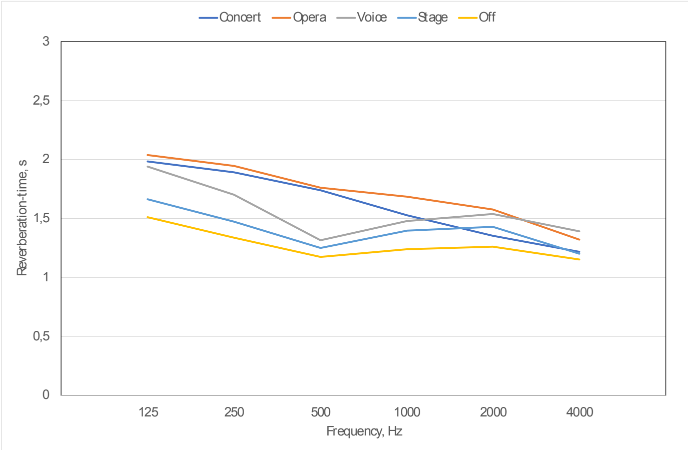


Figure 3: 500 seat opera hall

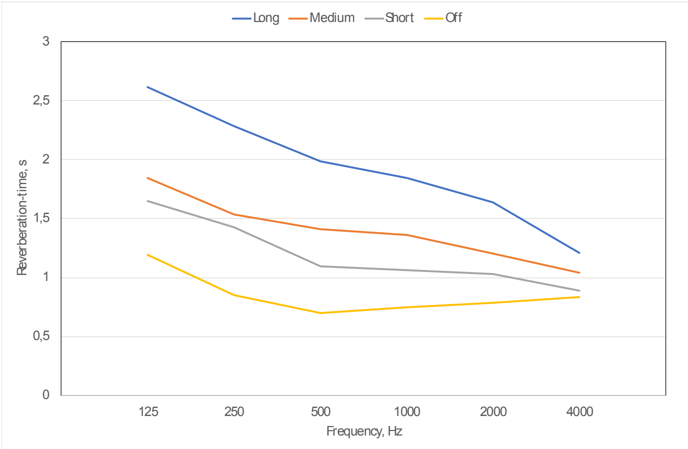


Figure 4: 500 seat multipurpose hall

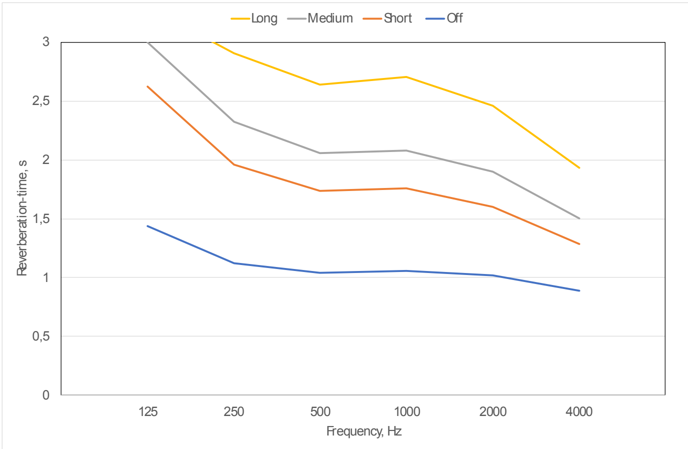


Figure 5: 1800 seat multipurpose hall

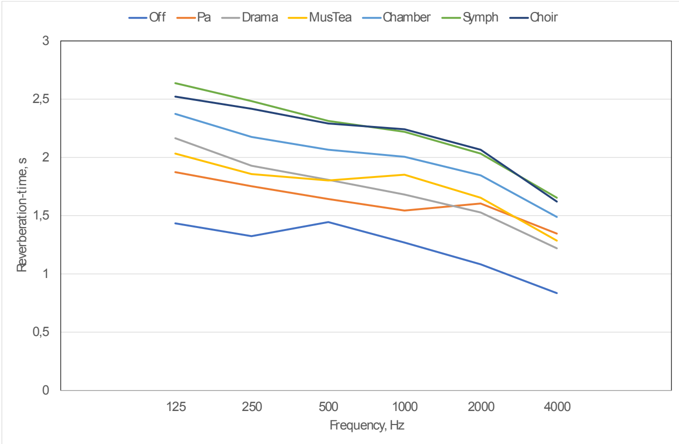


Figure 6: 1000 seat multipurpose hall

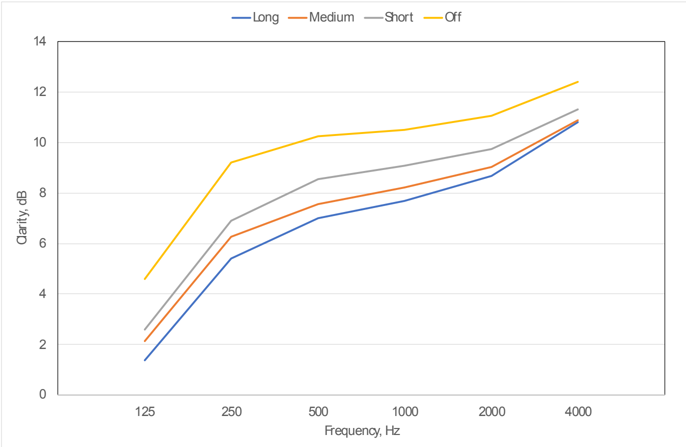


Figure 7: Clarity in the 500 seat multipurpose hall

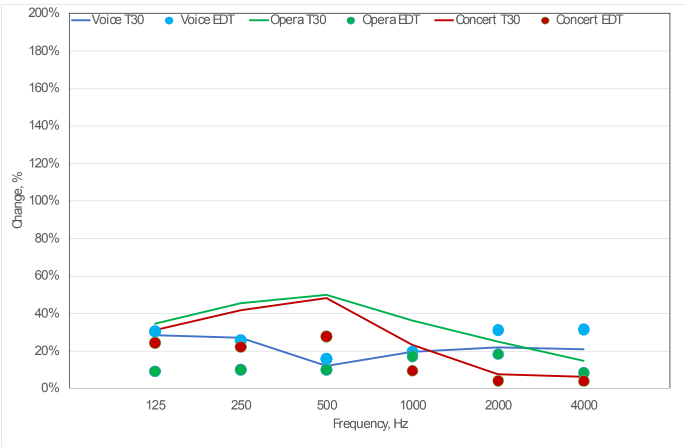


Figure 8: Relative percentage change of reverberation time and EDT in the opera hall

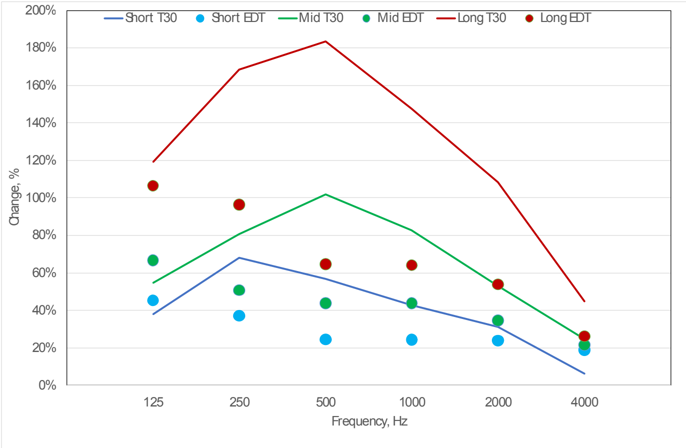
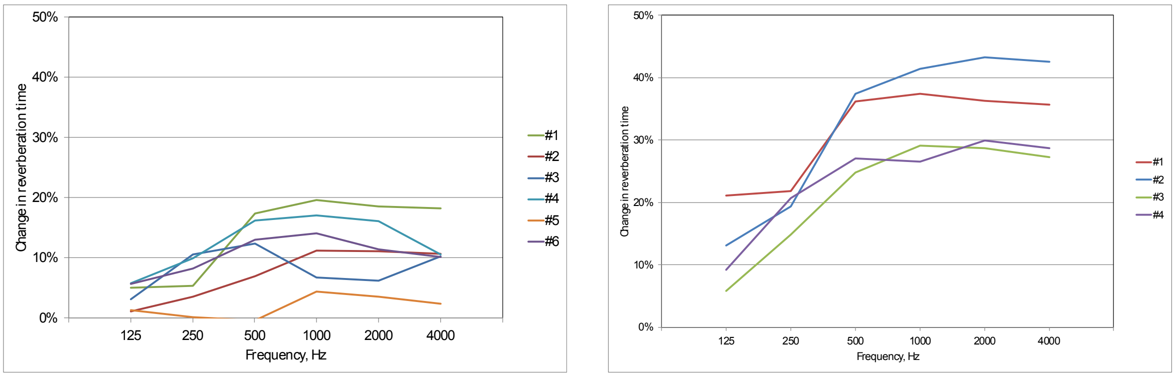


Figure 9: Relative percentage change of reverberation time and EDT in the 500 seat multipurpose hall

When comparing the variability in the to 500 seat halls measured for the investigation the variability of Finnish multipurpose halls presented in [8], some clear differences can be noticed. First of all, the overall dynamics are much larger with electronic enhancement systems. But what is perhaps more interesting is that is possible to achieve very large variability at bass frequencies with electronic enhancement systems and that, to some extent, the different acoustic characteristics can be independently adjusted for the different settings. With variable acoustic surfaces, this only possible to a limited extent, with variability to control some distinct areas of reflections.

As Cees Mulder states in [9], every mic-amp-loudspeaker channel, can produce a 2% percent naturally sounding increase in the reverberation time. It would seem that this may be part of the reason for the large difference of the performance of the systems in the halls, the amount of change implemented corresponds to the since of the size of the system, or at least the number of channels in the feedback part of the system.



Just surfaces in the auditorium Auditorium surfaces and stage curtains

Figure 10: Variability of reverberation time in multipurpose halls with variable acoustic surfaces

4 DISCUSSION AND FUTURE WORK

As stated earlier, the project is stilling on going and it is intended to measure more halls. The main limitation of the data presented in this paper is that the data is only from 3 halls, representing 2 of the currently available systems. It is clear that currently there are at least 4-5 commercially available systems with similar functionality and furthermore there some installations with proprietary systems based on standard audio processors. It is the intention to gather a comprehensive database for all these systems.

The main issue, of course which have not been discussed in this paper, is how do these systems sound or rather do they sound natural. One argument or question, should they sound natural ? The sound in a cinema hall is far from “natural” and still enjoyable. It is clear that unnatural sound becomes a problem if in interferes with musicians’ ability to play on stage. Playing should like playing in natural acoustics, not playing with sound reinforcement monitors. But again: it is problem if it doesn't sound natural if it sounds good?

In the next phase of the work, first of all more halls will be measured and spatial analysis will be performed on the data to present better metrics for the acoustic performance of the halls.

Finally, a listening test based on both the measurements in halls with electro acoustic enhancement systems as well as halls with traditional acoustic solutions will be made.

5 CONCLUSIONS

Electro acoustic enhancement system has become a feasible tool to increase the acoustic variability of halls, in particular multipurpose halls. Judging by the measurement done so far, the actual settings implemented, are more a result of user preferences, than strict acoustic theory.

However, it is the clear impression of the authors that electro-acoustic enhancement system will be one of the primary tools for creating multipurpose halls in the future.

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FINE TUNING OF THE SIBELIUS HALL STAGE ACOUSTICS

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1 INTRODUCTION

The Sibelius Hall, completed in February 2000, has a versatile adjustable acoustics. On both sides of the actual hall there are full height coupled volume chambers [1] through which the audience enters the hall. Between the hall and the chambers, on the sides of the hall and in the rear corners of the stage, there are a total of 188 continuously adjustable acoustic doors, which are one floor in height. In addition, the coupled volumes have a total of 2.7 km of continuously releasing two-layer wool curtains covering the area of the acoustic doors. With the curtain, the amount of reverberation can be varied when the acoustic doors are open. In addition, the canopy above the stage is adjustable in three parts.

The Lahti Symphony Orchestra moved to Sibelius Hall in early 2000 and has been playing on the stage ever since, basically with the acoustic configuration tested and instructed at the first Sibelius Festival in autumn 2000 by Artec Consultants Inc. team lead by Russell Johnson.

Over the past twenty years, the orchestra has expressed hopes at times for exploring and optimizing acoustics for different ensembles. Deficiencies in stage acoustics, i.e., problems with mutual hearing and hearing within the orchestra, have been particularly highlighted. Previous attempts at testing and possible improvements of stage acoustics had usually been hampered by budgetary constraints and, above all, a lack of final activity. Part of the reason for the latter is the fact that the acoustics in the hall have already been quite favorable as such.

Scattered experiments with adjustments have been made over the past couple of decades, but due to a lack of expertise and systematization, the attempts have mainly been inconclusive.

In spring 2020, the Lahti Symphony Orchestra decided to take advantage of the expertise of Akukon Ltd and have the tests executed. The auditorium measurements were carried out in early 2021, and when in autumn 2021 the pandemic was easing and the orchestra was back on stage in full numbers, the actual tests could be scheduled in the normal setting.

Compared to the earlier published procedures of adjusting and improving stage acoustics for a resident orchestra [2], this study had the advantage of utilizing original and versatile means for acoustic regulation, enabling precise systematization. This paper describes the investigations by objective measurements and subjective assessments by orchestra members for exploring and evaluating the improved acoustic settings for symphony orchestra.

2 MEASUREMENTS IN THE SIBELIUS HALL

To better understand the overall conditions and the variability of hall acoustics, we conducted an objective investigation by spatial impulse response measurements and spatiotemporal analyses. In addition, supplementary data was collected with a binaural head (B&K KEMAR). Measurements were repeated with a wide set of hall setups across parameters of the variable acoustics. The explored settings included several combinations of door panel openings to the coupled volume chambers, chamber curtains, and canopy heights.

2.1 Measurement positions and analysis

Measurement positions for sources and receivers used in the hall measurements are shown in Fig. 1. Room impulse responses were primarily measured spatially with a tetrahedral open microphone array (CoreSound Tetra) with logarithmic swept sinusoidal excitation signals from each source channel individually.

Measured spatial impulse responses were analyzed using the Spatial Decomposition Method [3] and with the spatiotemporal visualization approach [4].

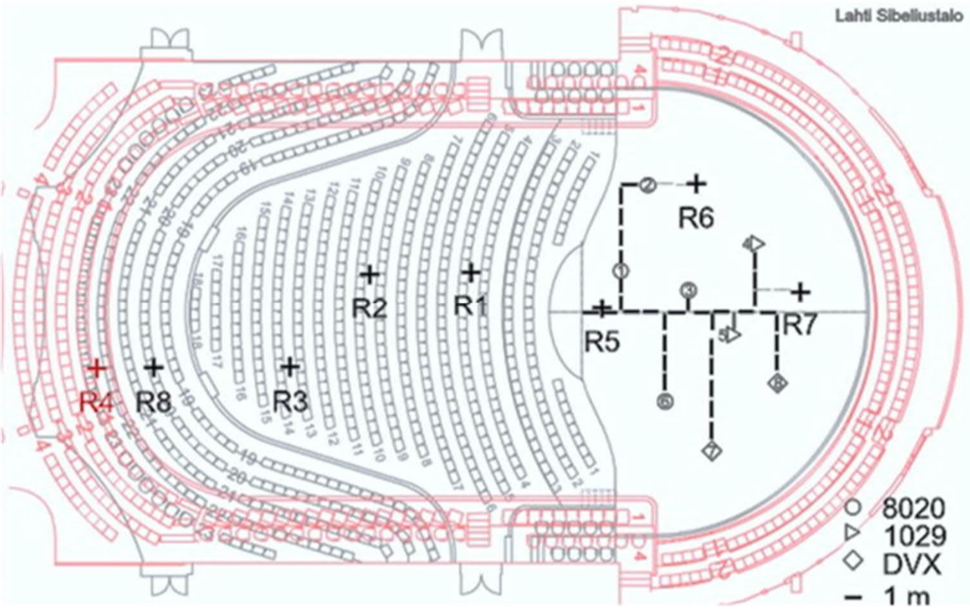

Figure 1: Measurement source (1…8) and receiver positions (R1…R8) in the hall. First balcony geometry with balcony receiver position R4 are shaded in red. Sources on stage are indicated with respective symbols for Genelec 8020 and 1029, and dB Technologies DVX speaker types.

2.2 Spatial measurement analysis and result application

The measurements were analyzed in two manners. First, the spatiotemporal analyses were evaluated separately for assessing the overall reflection patterns in different receiver positions. Then, analyses from various acoustic configurations were compared to the results from the initial acoustic setting for assessing the effect to reflected energy and directions. An example of the comparative principle is shown in Fig. 2, where the receiver position at the conductor podium is compared between settings with chamber doors open around the stage and in the hall and chamber curtains deployed, and initial setting with all doors closed. The red curve shows the cumulative energy up to 1 s with doors opened, and the green curve shows the respective result with all doors closed. The reverberant energy is higher in most directions with the doors closed, which is expected with more solid reflective surfaces towards the hall.

The corresponding approach was employed for several various comparisons between the measured range of acoustic settings. The observations from the results were subsequently applied to planning of suggested improvements to the new base setting for orchestral use.

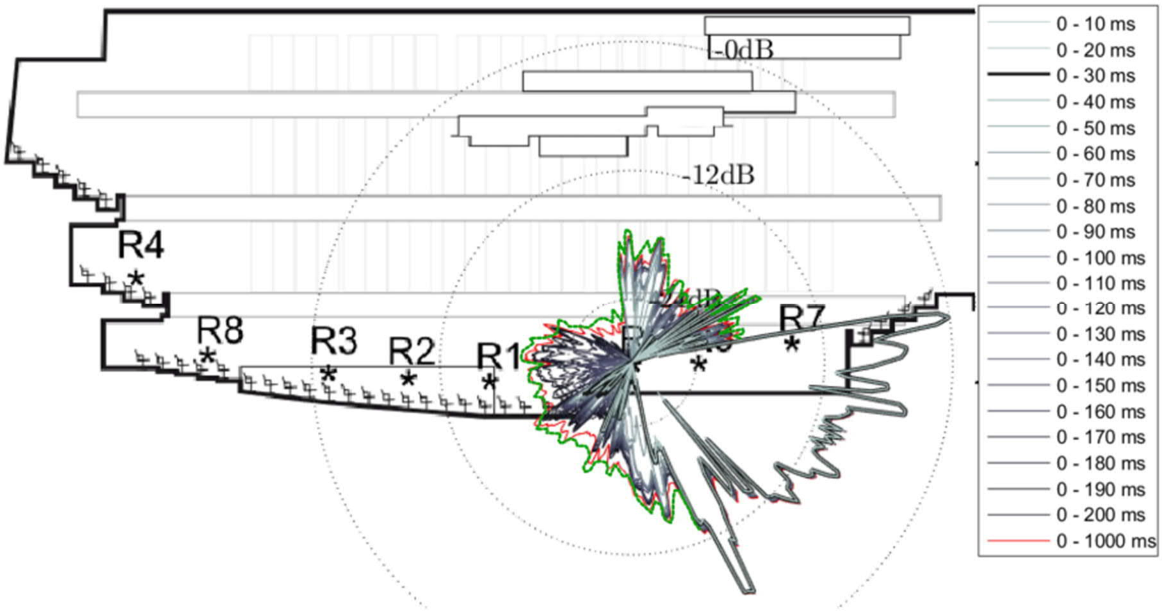

Figure 2: Example of overlaid spatiotemporal visual analysis between two hall settings. Grey and red: stage and coupled volume chamber doors open and curtains deployed; Green: 0-1000 ms window from all doors closed setting (directly comparable to red curve).

3 ORCHESTRA TESTS

3.1 Test routine

Subjective tests were conducted with the full orchestra for gathering perceptual evidence to reflect objectively measured and compared data. The procedure of the orchestra tests was based on music excerpts of 5–7-minute in total that were chosen from the concert programme of the week. This music was first played at the orchestra's then standard acoustic setting (A). After that, the orchestra left the stage to avoid direct visual cues on the changes being made. The acoustic adjustment proposal (B) was then set up. The orchestra then returned to the stage and played the musical excerpt with the new setting. The orchestra left the stage once again while the original setting (A) was restored, after which the orchestra played the musical example one more time. After the procedure, the orchestra members filled out a questionnaire form and gave oral feedback.

Four separate tests were organized, and the music excerpts were from the following works: *Cello concerto* by Antonin Dvořak; *Stabat mater* by A. Dvořak; *Sinfonia Brevis* by Helvi Leiviskä; and Suite from opera *"Les Indes Galantes"* by Jean-Philippe Rameau (in the chamber orchestra test).

3.2 Auditorium control

During each orchestral test, Akukon team, Chief conductor Dalia Stasevska and General manager Teemu Kirjonen formed a control group distributed to the auditorium to monitor the changes of the sound to the audience area. The test excerpts were conducted by an assisting conductor. Author PL acted as a "double agent" working both in Akukon team and playing in the orchestra double bass section.

3.3 Questionnaire and subjective evaluation

The acoustic impressions experienced by the orchestra on stage were investigated with subjective evaluation from musicians. After orchestra tests with modified acoustics, responses from the musicians were collected with a questionnaire which included the following five main questions:

1. Where is your seat on the stage?
2. In your experience, how did the timbre to your seat change with the new setting? ("Change in timbre")
3. How did you consider the acoustic feedback in the hall (spatial impression from the hall to the stage) with the new setting? ("Change in acoustic feedback")
4. Was it easier to hear certain instruments or groups of instruments with the new setting than with the original setting? ("Change in orchestra balance")
5. Your overall impression of the acoustic conditions in the hall with the new setting ("Overall change")

Each question was answered using a 7-point Likert-scale. The structure of the questionnaire was based on the studies by Barron and Dammerud [5].

4 PROPOSED ACOUSTIC MODIFICATIONS AND EVALUATION

The propositions for the orchestra tests' acoustic modifications were drafted according to the findings from the measurements as well as the conductors' and the orchestra members' remarks.

Prior to the testing sessions the Akukon team had several discussions with the chief conductors Dima Slobodeniouk (former), Dalia Stasevska (present) and the orchestra members to collect the comments about the stage acoustics.

The conductors appeared moderately confused about the difference of the orchestra sound to the conductor's podium and to the auditorium. This contrast can be attributed to the difference between relatively absorbing curved rear wall of the stage and typically reflecting surfaces in the auditorium. Furthermore, the conductor's position receives slightly focusing reflections locally, creating a characteristic acoustic impression.

For the musicians, the acoustic feedback on stage appeared not accurate enough and the sound rather boomy in fortissimo. The measurement results showed rather continuous increase in the reflected energy on stage from the upper hemisphere in the approximate time-interval of 30…50 ms from the canopy region as well as from surrounding upper surfaces. Based the perceived boominess and lack of accuracy in the acoustic response, the possibility of excessive reflections was considered. Therefore, Akukon team aimed for particularizing the diffusion of acoustic response on stage and letting the excess sound mass out, which can be described as a concept of "ventilating the stage". However, the already good general acoustic quality and the overall gain to the audience was seen important to retain.

4.1 Symphony Orchestra Test I

The acoustic doors in the rear corners of the platform were 100% open on the 2nd – 4th floor (4th floor above the canopy and service loft) and the curtains were completely deployed. In the auditorium, all doors were closed. The canopy was at Artec original height; front 13.7 m, middle 14.2 m, and rear 14.7 m.

4.1.1 Evaluation

This acoustic setting was received remarkably positively by woodwinds and string players sitting in the middle of the orchestra. It was easier to hear and control one's own playing. In the large violin groups, satisfaction decreased towards the end of the sections. The brass players and timpani sitting on the third riser at the rear of the stage did not notice much difference. The original slight sense of acoustic separation from the rest of the orchestra remained. The overall distribution of responses to each question is shown in Fig. 3.

The impressions of the auditorium control group were slightly differing by the seat. To some places, the new setting slightly zoomed out the overall sound, for others it zoomed in. However, the negative effects were minimal.

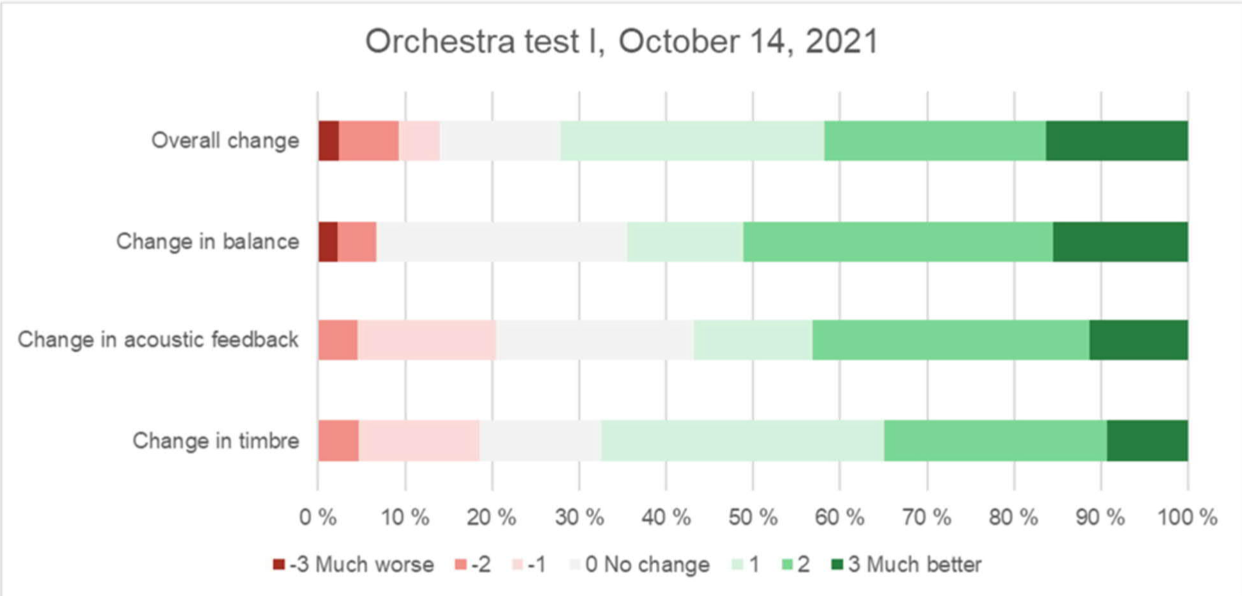

Figure 3: Results from subjective evaluation by musicians who observed difference between compared original and modified acoustic configurations (total N=66, 12 responses with no perceived difference omitted).

4.2 Symphony Orchestra Test II

Given the largely positive feedback in the previous test setup, especially in terms of ensemble control, Akukon team decided to use it as the new initial start setting (A) for subsequent tests. It was necessary to follow the direction for acoustic tuning the first test clearly indicated.

In the further adjusted setting (B), the doors at the rear corners of the platform were open 100% on the 2nd – 4th floors and on the sides of the auditorium on the 4th floor. The curtains were fully deployed. The plan to lower the canopy was thwarted by technical failure, so the test was partly incomplete. The orchestra was naturally not informed about this shortcoming.

4.2.1 Evaluation

The starting point for the test was poor as new canopy adjustments were impossible. The feedback from the orchestra was also clearly scarcer in numbers and, in practice, also worse than received from the first test. The overall results are shown in Fig. 4.

The auditorium panel also considered this stub option to be worse than the first.

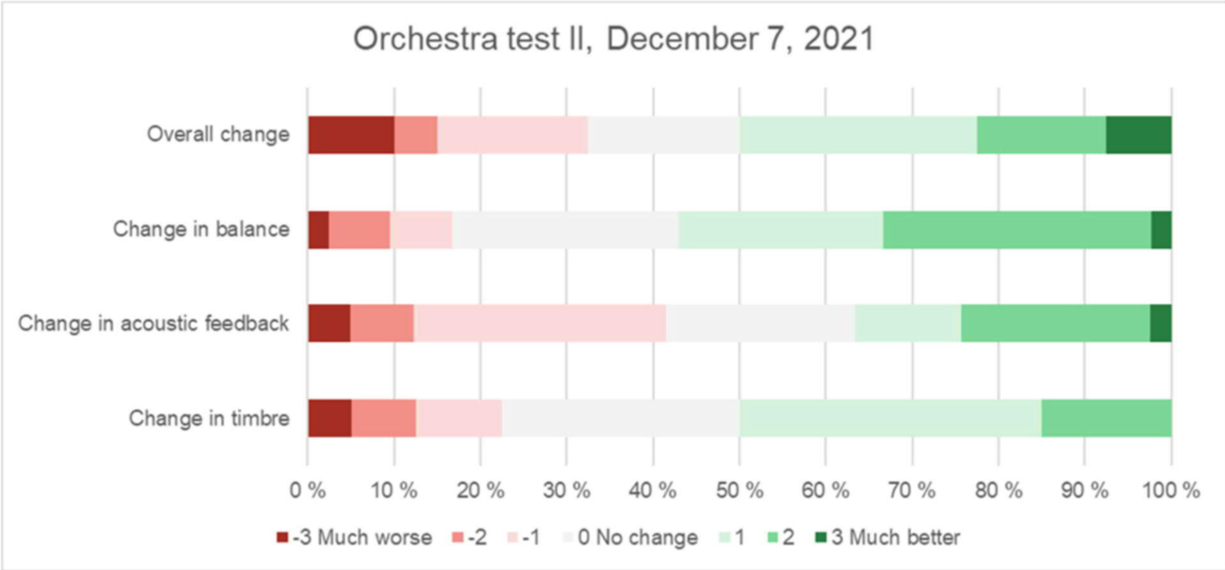

Figure 4: Results from subjective evaluation by musicians who observed difference between compared original and modified acoustic configurations (total N=64, 21 responses with no perceived difference omitted).

4.3 Symphony Orchestra Test III

The acoustic doors in the rear corners of the stage on the 2nd and 3rd floors were partially open allowing the sound transmission to the coupled volume chamber as well as reflecting sound towards the audience and the stage with increased diffusion. The aperture of the doors was adjusted following a geometrical projection from the centre of the stage to induce a uniform distribution of the sound reflections across the audience. The curtains were fully deployed and the canopy in Artec original setting.

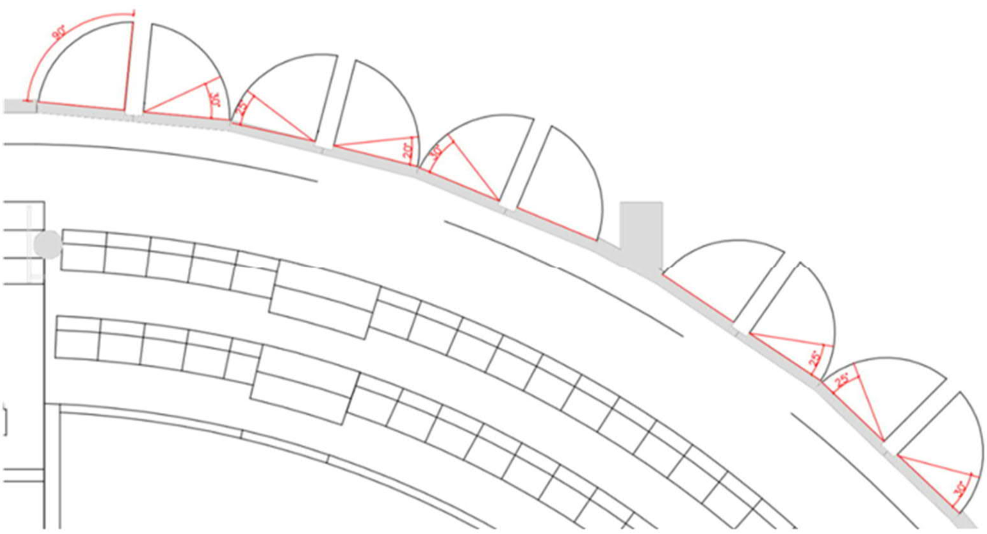

Figure 5: Door positions during test 3.

4.3.1 Evaluation

The test results now highlighted remarkable evenness of acoustic feedback in the orchestra's seating area. The players liked the ease of controlling the playing and the pleasantness of the general sound. For the first time timpani and brass on the third riser also experienced a clear improvement in their playing touch and mutual hearing. Their feeling of acoustic unity with the rest of the orchestra also improved markedly. Overall results are shown in Fig 6. It should be reminded that this comparison shows the evaluation between the originally suggested improved condition and the further optimized setting.

The panel seated in the auditorium considered this option to be the absolute best, and especially the most balanced and unanimously assessed configuration at the audience area.

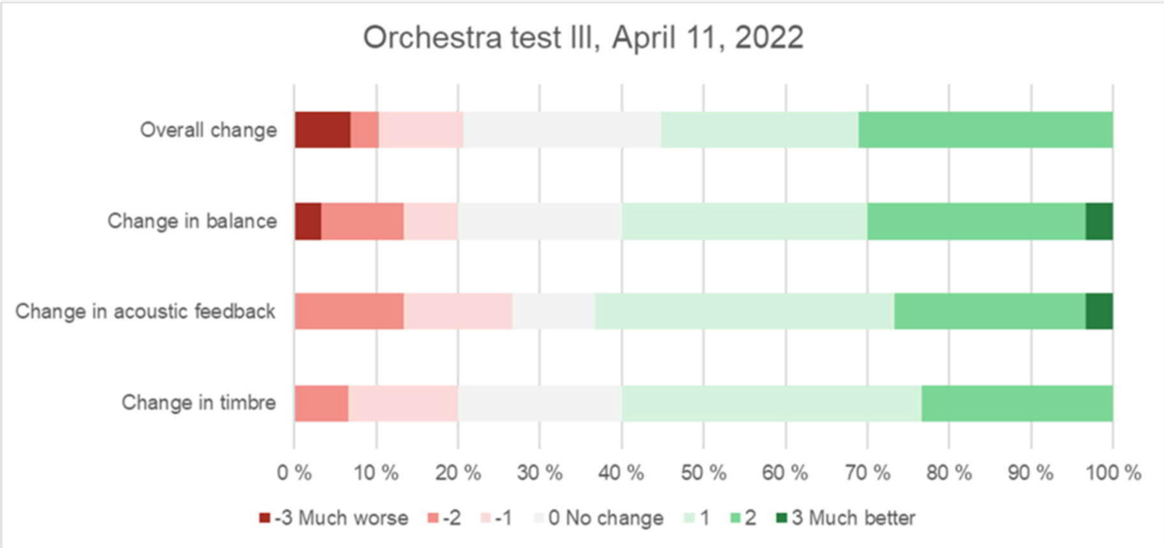

Figure 6: Results from subjective evaluation by musicians who observed difference between the originally improved and the further modified acoustic configurations (total N=53, 23 responses with no perceived difference omitted).

4.4 Chamber Orchestra Test

One test with the same procedure was conducted with a chamber orchestra ensemble of 28 musicians. The doors in the back corners of the stage were open 100% on the floors 2nd – 4th and on the 4th floor also in the auditorium. The curtains were fully deployed. The canopy assembly was set 0.5 m lower from Artec original height.

4.4.1 Evaluation

The results of the test were positive, but not very conclusive. Opinions were divided. Lowering the canopy was not a good solution for this relatively large chamber ensemble. It caused a vertical compressing effect, a "widescreen" sound picture. The overall results are shown in Fig. 7.

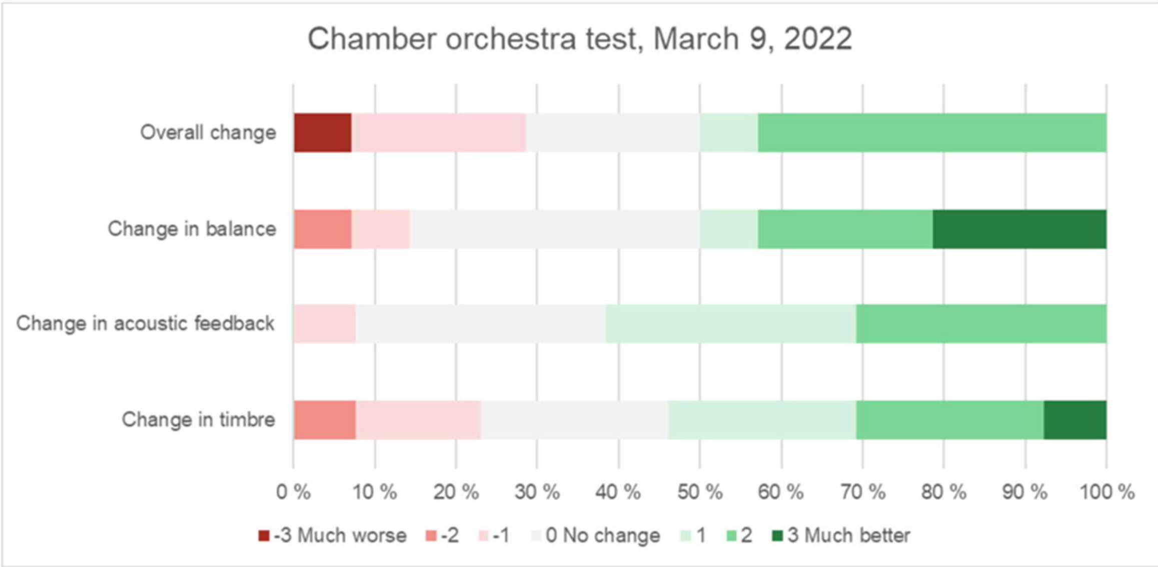

Figure 7: Results from subjective evaluation by musicians in the chamber orchestra ensemble who observed difference between the originally improved and the further modified acoustic configurations (total N=26, 12 responses with no perceived difference omitted).

5 CONCLUSIONS

Already after the first orchestra test, the acoustic adjustment possibilities enabled by the hall appeared adequate for reaching the desirable enhancement of stage acoustics.

The musicians' preferences of the proposed settings seemed to be, naturally not unanimous, but rather clear from the beginning. However, only the third setup proposition, being already more detailed fine tune, was able to make an obvious improvement to most of the players. In the first test, approximately 65% of the respondents who observed some overall change in the acoustic quality from the initial hall condition found the first proposed setting at least slightly favourable. Subsequently, the third orchestra test with further developed setting resulted in an additional approximately 60% at least slightly favourable opinions. In both these development stages the proportion of at least slightly displeased responses was substantially lower than those for the opinion of improved stage acoustics.

Based on the author’s personal experiences, the acoustic conditions at the rear percussion and brass instrument risers are typically felt challenging. However, the responses received from this area also indicated a positive development during the process. Other select comments from the musicians reported improvement in the touch and responsiveness of their own instrument, as well as in the mutual hearing for most sections. In summary, the subjective evaluation on acoustics both on the stage and in the auditorium suggests that the adjustments accomplished by the existing tools for variable acoustics provided a clear improvement for the overall acoustic quality.

An interesting concluding work would be to repeat the spatial impulse response measurements, now comparing the initial setting and the specific final preferred configuration. For more comprehensive results, a larger number of sources and receiver positions could be beneficial for more detailed analysis. However, this phase is left for future work.

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APPENDIX: FINAL SUGGESTIONS FOR ACOUSTIC SETUPS

Proposed setup for symphony orchestra

Akukon team proposed the setting of the 3rd test as such for the Lahti Symphony Orchestra:
Acoustic doors in the rear corners of the stage on the 2nd and 3rd floors open according to the presented irregular, but symmetrical scheme. Curtains fully deployed and canopy in the original Artec setting.

Proposed setup for chamber orchestra

Acoustic doors in the back corners of the stage on floors 2nd – 4th, and on the 4th floor of the hall open 100%. Curtains fully deployed. However, the canopy in Artec original setting.

Proposed setup for small chamber ensemble

Due to the Covid-dictated repertoire changes during the concert season, Akukon team was not able to test the stage acoustics with a small chamber ensemble. However, based on the tests executed and previous experiences in the hall, for an ensemble max. 7 players the team proposed:

Acoustic doors in the back corners of the stage on the 2nd – 4th floors, as well as on the 4th floor of the hall open 100%. Acoustic curtains fully deployed. The original canopy assembly 1.5 m below basic Artec setting. Therefore, the audience would use only the parterre.

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Temperature effects on ice rink sports halls

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Reverberation time is one acoustic parameter used to design "good" acoustics in sports halls. The reason for this is because the reverberation time is strongly related to speech intelligibility and bass ratio. A reverberation time over 3 seconds can negatively affect speech intelligibility for players and speakers.

Regardless of the height of the hall, to achieve a good acoustical environment in a sports hall acoustic materials must be added to multiple surface areas, not only on the ceiling. Usually, absorption is added to audience areas, chairs and on the walls. However, in an ice hall the reverberation time should be treated carefully. In a typical ice rink, the floor is a hard surface, with or without ice. The rink is surrounded by hard boards/glass to protect the spectators from line-drive puck. This means that it is not possible to add absorptions materials inside the rink. To make matters worse, rinks commonly have parallel walls, leading to large numbers of resonant modes and flutter echoes.

This paper shows that reverberation time is affected due to cold ice, when the rink has ice. We show that certain temperature and ice conditions can strongly reinforce certain frequencies, leading to extremely high reverberation time. It is found that sound waves bend down and circulates in the rink, resulting in a tripling of reverberation time from expected calculated values. It is also reviewed the use of computer programs, which are used to predict room acoustic parameters in planning before construction. Acousticians must have alternative methods to predict these types of acoustical phenomena. There is also discussed some methods to avoid "surprises" when these types of sports halls are built. It is proposed models to predict the effect of temperature and humidity for computing of reverberation time.

1 Introduction

For safety reasons, it is necessary to set up an appropriate shield to avoid injury to spectators. In addition, it is important to ensure that player/ spectators can see into the ice rink. According to the ASTM Standard "*Guide for Skating and Ice Hockey Playing Facilities*" F1703-04, the spectator protection must be a resistant tempered "impact" glass or acrylic extending a minimum of 182 cm above the dasher boards at rink ends, and a minimum of 127 cm about the dasher boards along rink sides, except the bench areas, where the shield is placed alongside and behind the benches. In bench areas the tempered "impact" glass or acrylic extending 122 cm to 182 cm above dasher board [1]. Another obvious consideration is that the shield must be built without air leakage because this has an economic impact in cooling demand.

Due to, the shield that surrounds the rink are hard materials, and they are parallel walls alongside, the reverberation time is increased significantly, and consequently the speech intelligibility is reduced. This environment can affect players performance and communication with the coaches[2, 3]. Some investigators have reported bad acoustic in ice halls. Flutter echoes and consequently high reverberation time can occur because inside the rink the sound is reflected many times before the sound go out the ice rink and gets absorbed for acoustic materials placed on the publicum and the ceiling [4].

Some previous works pointed out that it is necessary to include another acoustic parameter in ice halls due to bad acoustic inside the ice rink. Nevertheless, acoustic analysis with thermo-hygrometric conditions is infrequently done for those halls. Only a few standards take into consideration thermo-hygrometric conditions in acoustic analysis. For example, the standard ISO 9613-1 [5] considers the effect of air absorption in the propagation of sound waves. Even though, standard is aimed on the sound propagations through the atmosphere outdoors, the standard gives analytical methods for calculation of sound propagation with direct correlation with temperature, relative humidity, and static pressure. Another example is the standard ISO 354 [6]. This standard refers to the impact of temperature and relative humidity in an enclosed space. The standard provides acoustic methods to predict sound absorption in temperature range of 15 ºC to 30 ºC, and humidity from 30% to 90%.

The present study attempts to determinate the impact of air temperature and relative humidity on the reverberation time. It was measured, the reverberation time of four ice hockey halls. The results shows that the reverberation time is clearly affected in frequency range 250 Hz to 500 Hz, when there is ice in the rink reaching a reverberation of until 7-8 seconds. Without ice, i.e., with concrete on the floor, the reverberation time is reduced with a factor between 2-3. It was measured temperature at several heights. The results with ration of temperature confirm that sound waves have a tends to downward refraction. As result of this research, it is suspect that there is temperature inversion inside the ice rink.

2 Theory

Some researchers have reported changes in reverberation time as consequence of thermos-hygrometric changes. For temperature, it is possible to develop a mathematical model for sound waves propagation in the air. Due to the air does not support shear or bending, it can be assumed at the sound waves follow longitudinal propagation. Thus, under adiabatic condition the velocity of the sound (c) in the air can be calculated from the equation [7]:

$$c = 331.4\sqrt{1 + \frac{T}{273}}\ [m/s] \qquad (1)$$

Where “T” is the temperature in Celsius degree.

Modifying the Sabine equation for reverberation time (T_{60}) as function of air temperatures changes, results the following equation:

$$T_{60} = 55.3\frac{V}{c(A+4mV)}\ [s] \qquad (2)$$

Where the term “A+4mV” represents the total absorption of the room, the term “4mV” indicates the attenuation of the sound by the air, where “m” is the power attenuation coefficient, in reciprocal meters, calculated according to ISO 9613-1, and “V” is the volume of the room.

For air humidity, it was difficult to find a mathematical model. However, to determinate the sound absorption with changes in air humidity the equation (2) can be used. Hence, some investigators have reported tables / curves for air sound attenuation as function of air humidity. Nowoświat [8] concluded that reverberation time is strongly affected by changes in temperature and air humidity, and it is different for each frequency band. For low frequencies, the reverberation time tends to decrease as function of temperature, and for high frequencies trends to increase. Nowoświat found also that the reverberation time changes with relative changes of air humidity as function of the temperature for the same frequency band. It was found a particular peak value of reverberation time at 250 Hz and 500 Hz when air humidity is between 20% - 40 % and temperature is relatively low (~ 5 -15 ºC).

3 Case studies

The reverberation time (RT) was measured in four ice hockey halls, three of them in Norway and the other one in Sweden. Those cases are named here: case study 1, case study 2, case study 3 and case study 4. The results show that the reverberation time was strong affected by thermos-hygrometric conditions. The reverberation time when there was ice inside the rink was characterized by high values at the central frequencies. It was measured between 6 -10 seconds at 250 Hz – 500 Hz. It was observed that a little difference of 2 ºC could increase the reverberation time in 4 seconds. On the other hand, without ice in the rink, the reverberation time was measured below 3 seconds with a “flat” responds.

The measurements were conducted using the integrated impulse response method according to the ISO 3382-2 [9] where the impulse source was a start pistol. With the propose of comparison, five microphones were placed in a bar at different highs, both inside the ice rink and on the tribune, see Figure 1. Hence, the reverberation time was measured at the same time at various highs. Measurement positions (MP) in the ice rink were 1.5 m, 2 m, 2.5 m, 3 m and 5 m respectively. In addition, five devices to register both temperature and humidity were placed on the bar as well at 0.32 m, 0.64 m, 1.5m, 2.5 m and 5 m respectively.

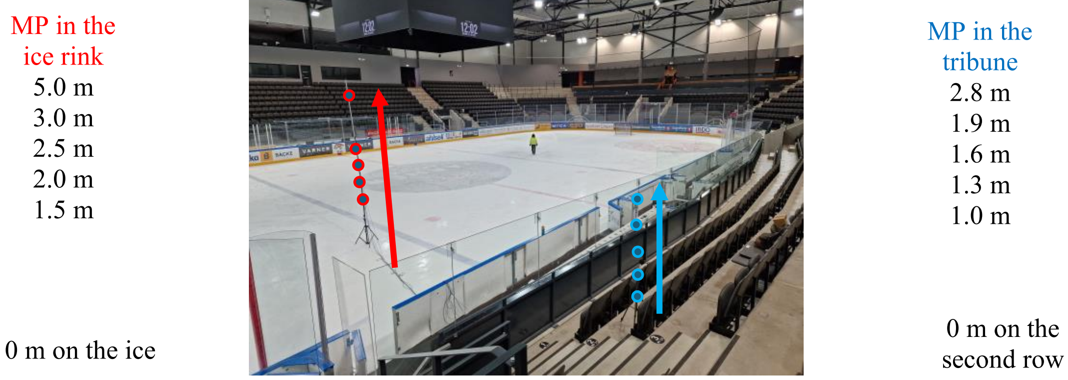

Figure 1: Measurements positions (MP) at the same time, five microphones in the ice rink and five microphone on the tribune.

All case studies are “traditional” ice hockey halls, with a curved ceiling. With exception for case study 2, where the ceiling is flat, but it has an inclination. In all the halls, there were used perforated plates with sound absorption class A in the ceiling.
Due to advertisements boards, the area on the wall where one can locate sound acoustic plates was limited. Thus, it was used chairs on the tribune with sound absorption for three of the halls. Table 1 shows a resume of case studies.

Table 1: Summary of case studies

Case study	Nr of Seats	Volume	Heigh m	Comment
Case study 1	3600	30 000 m^3	~17 m	Sound absorption on ceiling (sound class A), and on the chairs - tribune (sound class B).
Case study 2	5300	63 300 m^3	~ 19-20 m	Sound absorption on ceiling (sound class A), on the chairs - tribune (sound class B) and 20 % of the walls covered with acoustic plates sound class C.
Case study 3	7600	74 000 m^3	~ 18 m	Sound absorption on ceiling (sound class A) and on the chairs - tribune (sound class B).
Case study 4	2450	24 000 m^3	~ 12 m	Sound absorption only on the ceiling (sound class A).

4 Results

4.1 Parallel walls

To study the effect of parallel walls inside the ice rink, it was created a simple 3D model in COMSOL Multiphysics®. The model was created in reel scale, so the ice rink is 60 m long and 26 m wide, and the shield that surrounds the rink was modelled with hard materials with a height of 3.0 m. The Ray tracing method was used to compute the reverberation time. It was used a sound source (power Po = 0.3 W) located at corner of the ice rink, and one receiver 1.5 m over the ice surface, see Figure 2. Figure 2 also shows the results for two simulations, one with parallel walls alongside the rink and one with slightly tilted walls. As expected, it is confirmed that parallel walls alongside the rink can significantly increase the reverberation time. In general, the reverberation time with parallel walls increases more than 100 % at frequencies 250 Hz – 1000 Hz. The reverberation time increases with a factor of 2.2 and clearly shows a peak value at 500 Hz with parallel walls.

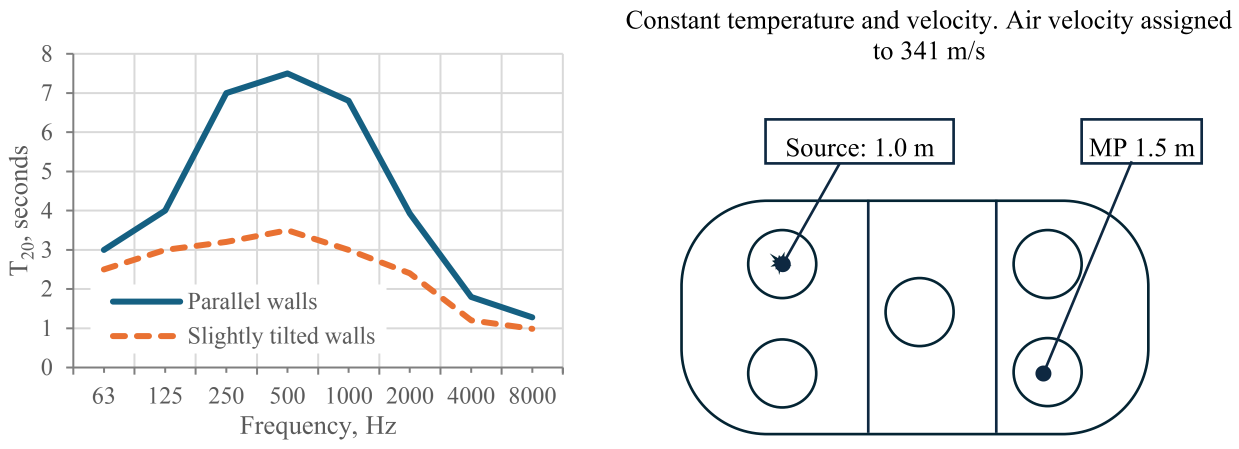

Figure 2: Results of preliminary studies. Effect of parallel walls.

4.2 Temperature gradient

As it was mentioned before, five thermometers were placed at various heights over the ice surface. Figure 3 shows the measured temperature in three case studies. It is observed slight differences between each case study. However, the temperature gradient has the same tendency. In general, the temperature close to ice (0 m) was measured between 2 - 4 ºC and the temperature over the ice rink (5 m over the ice surface) was measured between 9-12 ºC.

Figure 3 also shows a picture taken with a thermostatic camera. The measured temperature agrees with the results obtained from the thermostatic camera.

As result from this investigation, it was found an equation that could be used to predict the temperature gradient in future work:

$$T = 5 \text{ x } h^{0.5} \ [^{o}C] \quad (3)$$

Where "T" is the temperature in Celsius degree and "h" is the height measured from the ice surface.

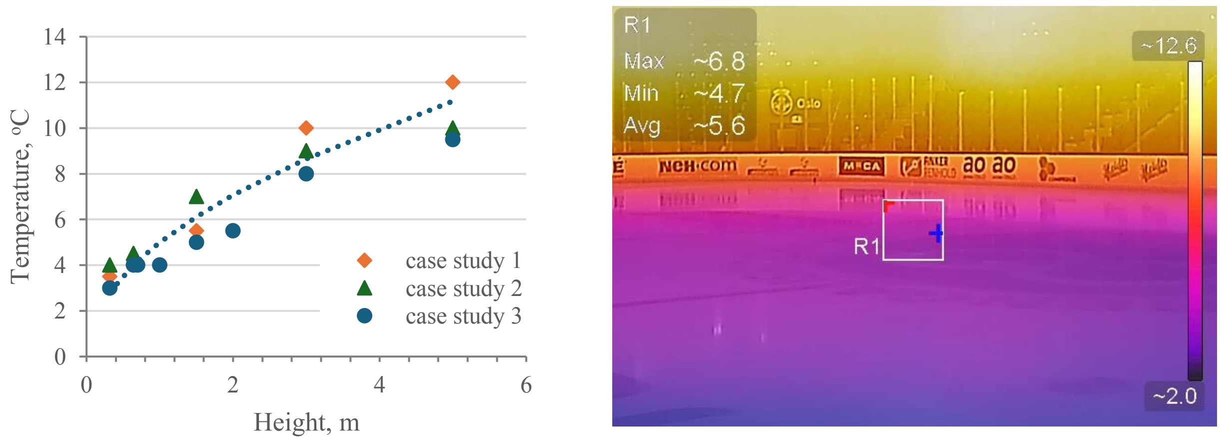

Figure 3: Measured temperatures in three case studies.

One particular case is the ice hall number 4. This case study has the lower ceiling from all studied cases (12 m), and the spectators reported cold temperatures. After a deep study, it was added a thermic isolation layer under the ceiling, see Figure 4. In this case study was measured another gradient temperature with a variation of 3 ºC between measure point at 1.5 m and measure point at 5 m. Therefore, this case study is not included in the model to predict variation of temperature with the high, equation (3).

Figure 4: Themis isolation layer, case study 4.

4.3 Humidity gradient

For the humidity gradient it was also found an equation that it could be used in futures analyses to predict the reverberation time in an ice hockey hall, see equation (4). It is observed that humidity in the ice rink is around 50%, and outside of ice rink about 40%.

$$\mathrm{Hr} = 50 \times \mathrm{h}^{-0.05}\ [\%] \tag{4}$$

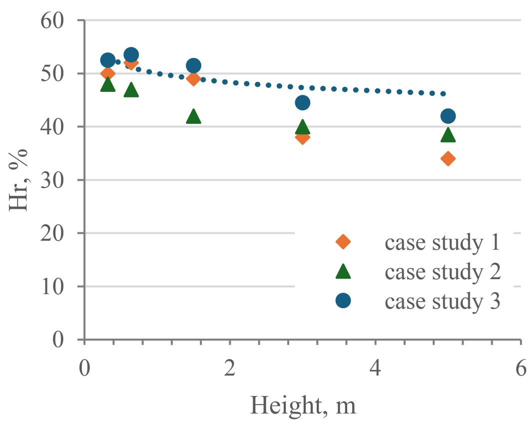

Figure 5: Measured humidity in three case studies.

4.4 Reverberation time

It was measured the reverberation time (T_{20}) at different points distributed both inside the ice rink and tribune. Figure 6 shows approximate location for measure points (MP) and source positions (red start) in case study 1. Measure points MP1 - MP3 were inside the ice rink and MP4 – MP6 on the tribune. For the other ice halls, it was also used a set of 6 measures points distributed, 3 in the ice rink and 3 on the tribune.

In case study 1 and case study 2, it was also computed the reverberation time with the software ODEON Room acoustic®, version 14.

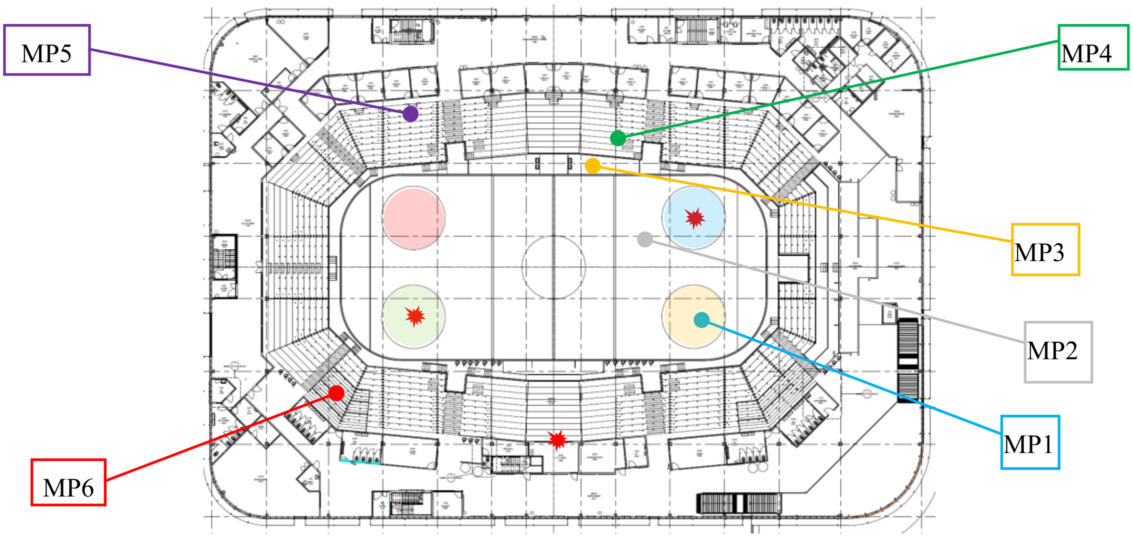

Figure 6: Measurements positions (MP) and source position, preliminary study.

Figure 7 shows the measured and computed reverberation time for case study 1 and case study 2. The figure shows the average of 6 measured points. The measurements reveal that the reverberation time is affected in frequency range 250 Hz to 500 Hz, where the measured reverberation time has extremely "high" values. Same trend is observed in all case studies.

For comparison propose, the results from computation are also plotted in the Figure 7 with split line. Numeral simulation shows lower values than measured values for reverberation time. In the software Odeon, it is possible to change air conditions, both temperature and humidity, but this tool is to compute air absorption in large rooms. Unfortunately, it was not possible to reproduce the measured results in the software Odeon Room acoustic®.

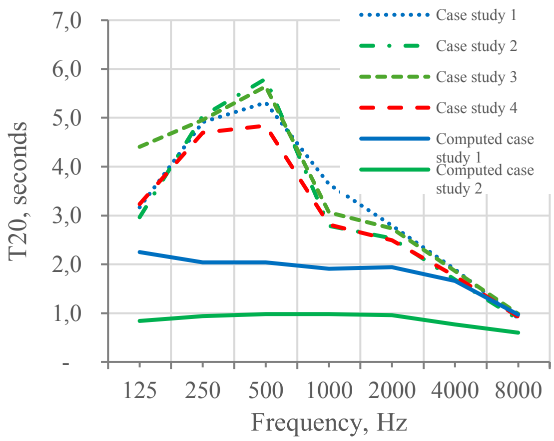

Figure 7: Measured reverberation time (T_{20}), and computed with Software Odeon

The case study 4 shows lower reverberation time. A qualitative explanation for lower values in the case study 4 could be understood with the phenomenon "*temperature inversion*". In case study 1-3 was measured higher variation of temperature from 1.5 meter to 5 meter, compare with variation measured in case study 4. The temperature in case study 1-3 was measured to 10 - 12 °C at 5 meters, meanwhile the temperature at 5 meters in case study 4 was measured below 8 °C. Thus, forming an "inversion layer" could be located at some given altitude that allows the acoustic energy spreads out of the ice rink and thereby it is absorbed, se Figure 8b. Conversely, for case 1-3, the energy may be redirected back towards the ice rink, increasing the reverberation time, see Figure 8a.

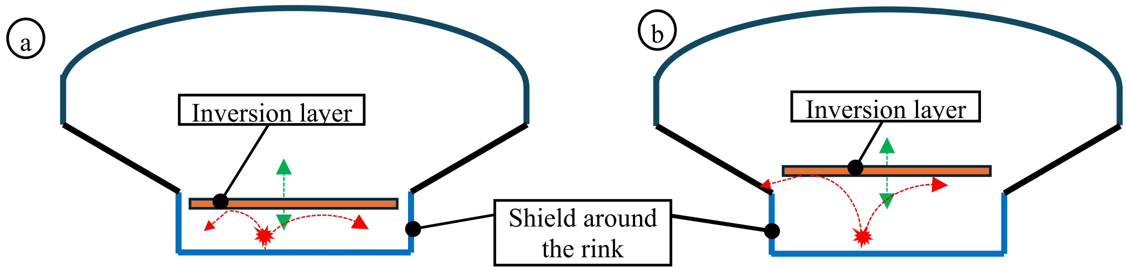

Figure 8: Measured reverberation time (T_{20}), and computed with Software Odeon

To understand the phenomenon of high reverberation time, it was measured the reverberation time in the same ice hockey hall with and without ice in the rink. Figure 9 shows the results. It is confirmed that thermos-hygrometric conditions affect the reverberation time. With a temperature of 10 ºC and humidity around 53 %, the reverberation time has peak values of reverberation time at 250 Hz and 500 Hz. Similar results were also reported by Nowoświat [8]. On the other hand, results without ice the reverberation time follow a "flat" respons. The same trend is computed with numerical results computed with Odeon Room acoustic®.

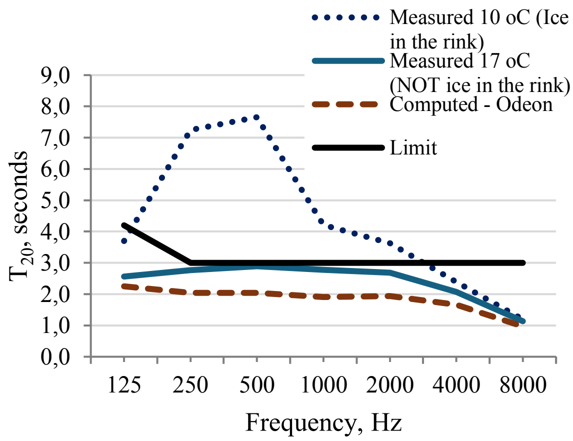

Figure 9: Measured reverberation time (T_{20}), and computed with Software Odeon Room acoustic®, comparison with and without ice in the rink.

Figure 10 shows measured reverberation with five sound level meters placed in a bar at different highs, as it was presented in section 3. For these measurements, the source (start pistol) was placed in the green circle inside the ice rink. The humidity below 3 meters (inside the ice rink) was measured by around same values. Therefore, Figure 10Figure 8 shows humidity by around 48 % – 53 %, and 40 %.

In general, all case studies have peak values of reverberation time at 500 Hz. Figure 10 evidence that the reverberation time is gradually reduced increasing the temperature. As demonstrative, Figure 10a depicts the reverberation time measured in case study 1. In this case study, the temperature goes from 5.5 ºC (1.5m) to 12 ºC (5m), and the reverberation time is reduced from 6.4 seconds to 4.6 seconds at 500 Hz.

As it was expected, the reverberation time in case study 4 was measured lower than another case studies. It was registered lower values for both temperature and humidity.

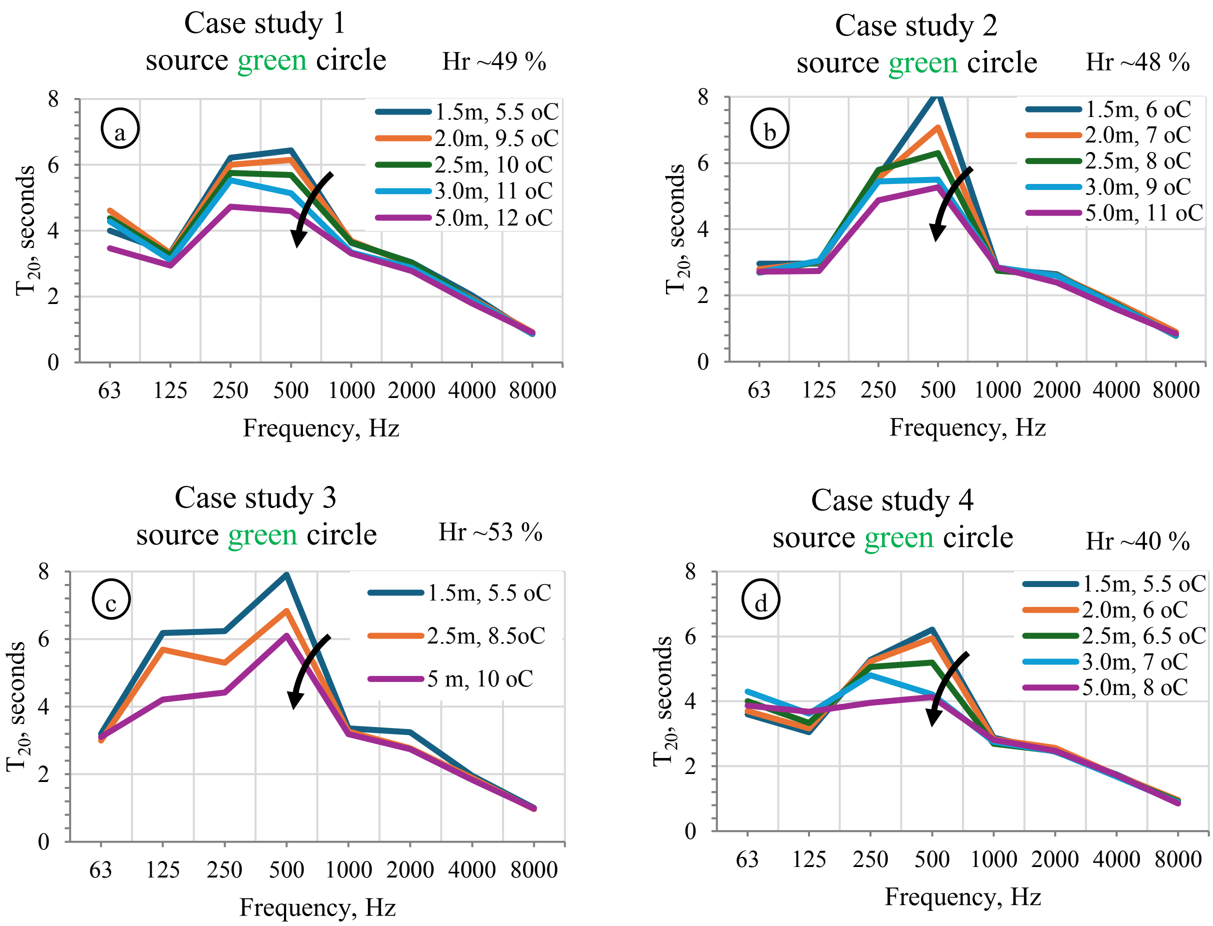

Figure 10: Measured reverberation time (T_{20}) at different highs.

5 Numerical simulations

It was not possible to get the phenomenon of high reverberation time with the software Odeon Room acoustic®. Thus, it was created a numerical model in COMSOL Multiphysics ®. A preliminary study was conducted with a 2D axisymmetric model. The purpose was to compare the distribution of acoustic energy by viewing ray paths in 2D. Using the model for temperature gradient obtained in equation (3), it is computed the air sound speed by the equation (1).

For reference, it was computed the sound pressure level for two cases. First case, it is computed with air sound speed constant of 341 m/s. The second model simulates graded air sound speed calculated by equation (1). This means that air sound speed is highest at 5 meter (~338 m/s) and lowest at the bottom, close to ice surface (~332 m/s).

Figure 11 compares the distribution of acoustic energy with constant temperature (Figure 11a) and a model with temperature gradient (Figure 11b). The model represents part of a section along the length of the ice hall. The shield around the ice rink is represented by a short wall. The inclined line is the tribune area. The left wall is used as axis of symmetry inside the ice rink. Sound absorption and scattering coefficients assigned to the wall are the same used in previous numerical computations with software Odeon®. For the case with constant temperature, the acoustic energy spreads out of the ice rink. With graded temperature however, this same energy has a tendency of downward refraction, enhancing the reverberation time. The same patron is found for phenomenon called "*temperature inversion*".

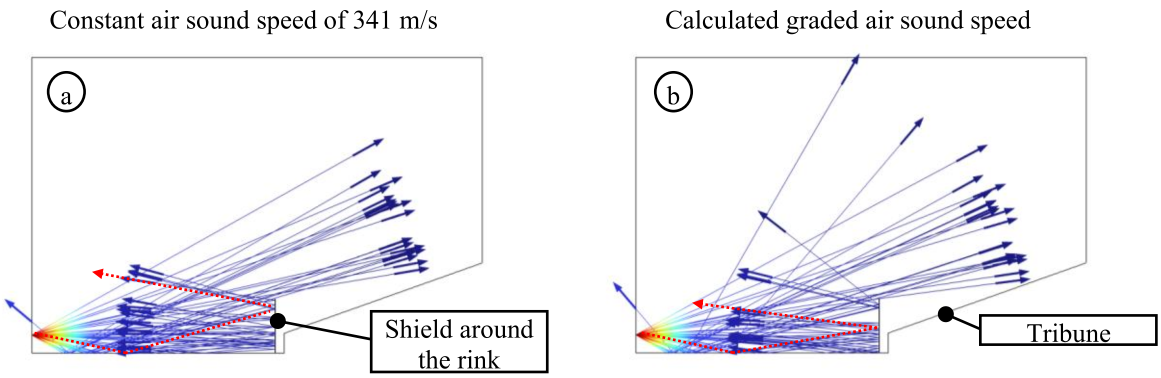

Figure 11: Ray paths in in 2D after 0.3 seconds for: a) uniform air sound speed of 341 m/s b) graded air sound speed calculated by equation (1).

Further work is being developed to validate measurements of reverberation time. It is also studied different alternative to avoid peak values of reverberation time. A first preliminary study in a 3D model was created. The main objective with the 3D model was to investigate how the reverberation time is affected due to temperature gradient. Figure 12 depicts computed reverberation time at different heights in the ice hall. The results suggest that software COMSOL can be considered to study the phenomenon found in all case studies.

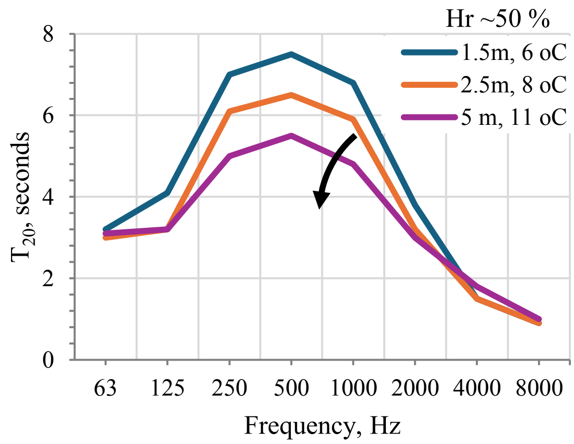

Figure 12: Computed reverberation time, T20, with a 3D model in COMSOL Multiphysics ®.

6 Summary

Several investigators have reported "bad" acoustic in ice hockey halls. In general, it was found that the reverberation time is affected in frequency range 250 Hz to 500 Hz, where the measurements in four ice hockey halls show peak values in those frequencies.

This paper shows that the phenomenon called "t*emperature inversion*" can explain why measured reverberation times are long in the case studies.

A new model to estimate changes of thermos-hygrometric conditions in an ice hall is advanced. Since, it is not possible to add sound absorbing materials inside the rink, it is difficult to reduce reverberation time. The authors are currently working with numerical simulations to propose alternative to reduce the reverberation time. A preliminary study with software COMSOL ® confirms that this software can be used to understand propagation of sound energy in an ice hockey hall.

Further work will aim to improve prediction methods and to develop solutions to avoid excess reverberation times.

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Assessment of the influence of the method of fastening decorative and finishing panels on the fund of low-frequency sound absorption in a concert hall

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On June 1, 2022, a concert hall was put into operation inside one of the Packhouse built in Nizhny Novgorod more than a hundred years ago. The hall is designed for concerts and opera productions without sound reinforcement. During the finishing work, deviations from design solutions were allowed: the presence of air cavities behind flexible sheaths, an understated surface mass of the ceiling. This resulted in a reduced measured reverberation time at frequencies below 1000 Hz compared to the calculated values. Measurements of the sound absorption coefficient of the wall structure according to the project and actually executed were carried out in laboratory conditions. These data are used to estimate the reverberation time in the hall, in case of implementation of design decisions. The data of computer modeling of the acoustics of the hall at the design stage, the results of measurements in the built hall and the predicted values for rigid fastening of wall panels are presented.

A special feature of the hall is the back wall of the stage, which is a translucent glass structure that can be closed with a sectional sliding partition. An analysis was carried out of the influence of the design of the rear wall of the stage on the acoustic parameters of the hall for various positions of the sliding structure.

Data on the subjective assessment of the acoustics of the hall are presented.

1 Introduction

Since the end of the nineteenth century, the largest art (not only in Russia) and industrial exhibitions have been held in Nizhny Novgorod, on the banks of the Oka River at its confluence with the Volga (Strelka). Some of the pavilions (Packhouse) for the exposition were built according to the project of the famous Russian engineer V. Shukhov. In Soviet times, the pavilions were converted into production and storage facilities. During the preparation of Strelka for construction and landscaping within the framework of the 2018 FIFA World Cup, the two remaining Shukhov pavilions (openwork metal frames) were freed from late layers (brickwork, slate coverings, auxiliary structures). In 2019, the Packhouse hosted several cultural events: the Intervals Audiovisual Art Festival and the Strelka International Art Festival. In 2020, work was carried out to preserve Packhouse as cultural heritage sites of regional significance for the 800th anniversary of Nizhny Novgorod. The metal structures were cleaned of old layers of paint and dust, and the historical color was returned. In the summer of 2022, a concert hall with 426 seats was opened in one of the Packhouse. The concept of preservation was developed by the architectural bureau "SPICh" under the guidance of architect Sergei Choban. The concert hall in Packhouse is a unique project in Russia. Its original architectural feature is a panoramic glass wall with a view of the Strelka and the historical part of the city.

In November 2022, this project became the winner of the international Trezzini Award.

2 Acoustic characteristics of the hall

The peculiarity of the concert hall, in addition to its elongated shape (the hall was fitted by the architect into the space of the Packhouse frame), were restrictions on the weight of the enclosing structures of the hall. In this regard, the implementation of design acoustic solutions in full was not possible in terms of the use of heavy and rigid reflective decorative linings on the surface of walls and especially on the ceiling.

The hall has a rectangular shape in plan, the main geometric characteristics are presented in Table 1

Table 1: Geometric characteristics of the hall

The length of the hall (up to the stage)	22.5 m
The width of the hall	13.6 m
The height of the hall	9.2 m
Capacity	320 seats
The air volume of the hall	3910 m^3

The hall is mainly focused on musical and theatrical productions in the mode of natural acoustics (without the use of sound reinforcement systems).

An additional architectural feature of the hall is the back wall of the stage, which is a translucent glass structure that can be closed with a sectional sliding partition.

The project provided for wall cladding with a surface density of ~ 85 kg/m2, on a flat section of the ceiling, cladding with a surface density of at least 30 kg/m2 was assumed. The ceiling and walls are finished with veneered gypsum panels, 8 mm thick, which were glued to the aerated concrete wall base and gypsum (drywall) ceiling (along the inclined perimeter). The main central flat part of the ceiling is made of two sheets of drywall. Due to the fact that the limitation of the bearing capacity of the new trusses bearing the ceiling of the hall was revealed at the stage of completion of finishing works, the ceiling in the hall was mounted with a surface density lower than that laid down in the calculation of acoustic parameters.

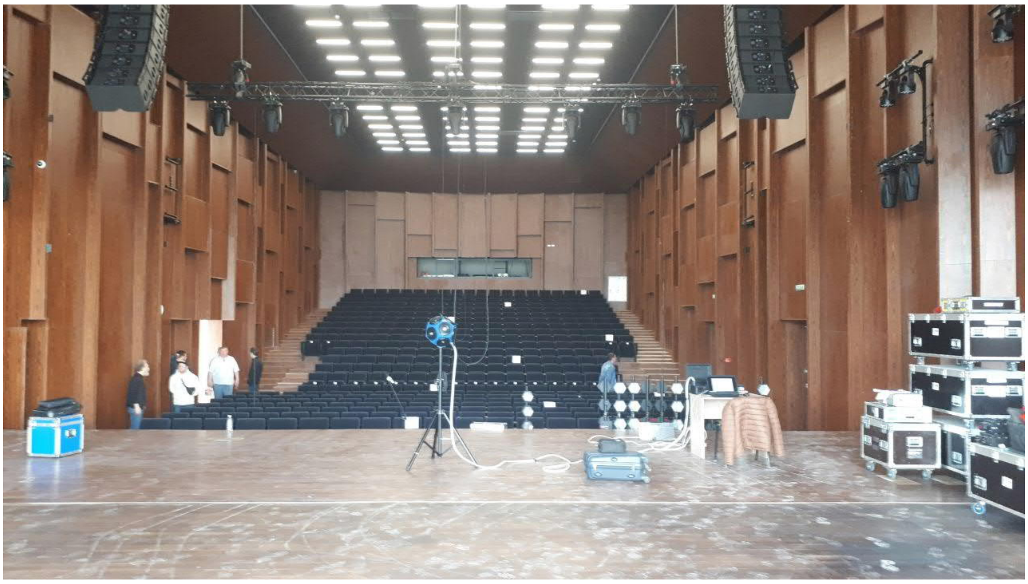

Figure 1: The hall at the end of construction works – 06.01.2022.

To assess the acoustic characteristics of the hall during the design process, a computer model was built in the EASE software package. Figure 2 compares the design values of the reverberation time and the measurement results at the end of construction.

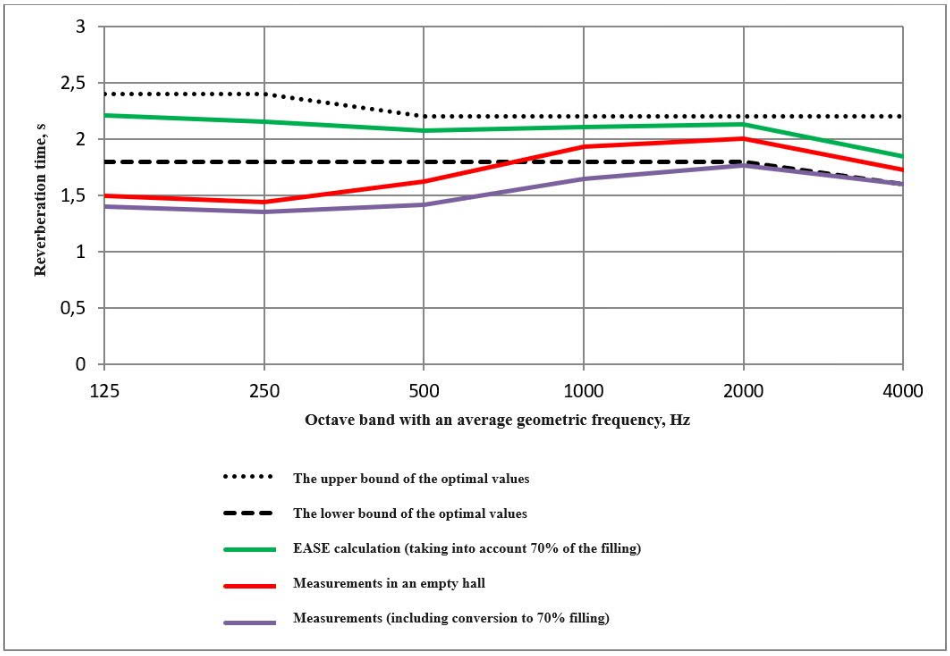

Figure 2. The reverberation time in the concert hall.

Based on the measurement results, it was found that the expected reverberation at high frequencies corresponds to the design values, the values at low and medium frequencies are slightly lower than the design values, which can be explained by the following factors established during the acoustic survey:

- The ceiling design is much lighter than the design solution;
- The presence of air gaps between the walls of the aerated concrete hall and decorative panels;
- Construction dust that was not completely removed during the acoustic examination;
- Possible hidden defects in the floor structure.

3 An experiment to assess the effect of the method of fastening decorative acoustic panels

For a qualitative assessment of the sound-absorbing properties of decorative acoustic panels mounted in various ways, acoustic measurements were carried out in a small acoustic chamber "Acoustic Group", the scheme and dimensions of which are shown in Figure 3.

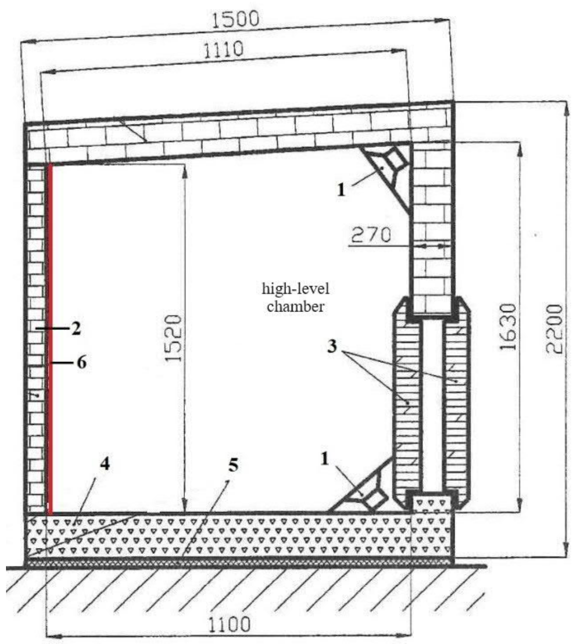

Figure 3. Small sound measuring chamber

1 – acoustic system; 2 – aerated concrete wall of 200 mm; 3 – doors; 4 – independent foundation; 5 – vibration–insulating layer; 6 - moisture-proof gypsum fiber sheet.

A moisture-resistant gypsum fiber sheet of 8 mm was selected as the finishing decorative panel. Figure 4 shows a photo of a partition made of aerated concrete and various ways of gluing a decorative panel.

a b c d

Figure 4. Photo of the back wall of a small sound measuring chamber. A wall made of aerated concrete blocks (a), point fastening of the moisture-proof gypsum fiber sheet (b), solid fastening of the moisture-proof gypsum fiber sheet (c), moisture-proof gypsum fiber sheet glued to the wall of aerated concrete (d)

To eliminate edge effects, the gap between the moisture-proof gypsum fiber sheet and between the moisture-proof gypsum fiber sheet and the side walls of the sound measuring chamber was sealed with a sealant. The reverberation time was determined by the intermittent noise method. A modulated sound signal ("pink" noise) was applied to the speaker system and recorded on a microphone located in the central part of the volume of the sound measuring chamber. The recorded pulse responses were processed in the DIRAC 3.0 software package.

The measured values of acoustic parameters, averaged over 10 measurements for each mounting method, are shown in Table 2.

Table 2: Reverberation time in a small measuring chamber

Experiment	The method of fastening the moisture-proof gypsum fiber sheet to the aerated concrete wall	Reverberation time, s, in the octave band with an average geometric frequency, Hz					
		125	250	500	1000	2000	4000
1	Spot glue attachment	1.11	1.21	1.24	0.93	0.86	0.80
2	glue over the entire surface	1.32	1.65	1.51	0.95	0.83	0.79

The reverberation time in the sound measuring chamber T, s, according to [1] can be recorded as a ratio:

$$T=0{,}16\,\frac{V}{\alpha_0 S_0+\alpha_i S_i^{:}} \qquad (1)$$

where V – the volume of the sound measuring chamber, m^3, α_0 is the reverberation coefficient of sound absorption of the test sample, S0 – the area of the test sample, m^2, αi is the reverberation coefficient of sound absorption of the inner surface of the chamber without taking into account the test sample, Si – the area of the inner surface of the chamber without taking into account the test sample, m2.

The change in the sound absorption coefficient due to a decrease in the adhesive area between the wall and the moisture-proof gypsum fiber sheet is determined by the difference:

$$\Delta\alpha_0 = \alpha_{02} - \alpha_{01} \qquad (2)$$

where α_{02} and α_{02} are the reverberation coefficient of the test sample during experiment 2 and 1, respectively.

The volume of the sound measuring chamber V = 1.69 m^3, the area of the test sample S_0 = 1.53 m^2. Substituting ratio (1) into formula (2) together with the results given in Table 1, we obtain the results of Table 3.

Table 3: Changing the sound absorption coefficient

Acoustic parameter	Octave band with an average geometric frequency, Hz					
	125	250	500	1000	2000	4000
A change in the sound absorption coefficient due to a decrease in the adhesive area between the wall and moisture-proof gypsum fiber sheet $\Delta\alpha_0$	-0.03	-0.04	-0.03	0.00	0.00	0.00

4 Assessment of the potential for increasing the reverberation time of the hall

It is of interest to assess the potential for increasing the reverberation time in the hall. An increase in the surface mass of the ceiling is not possible due to the limitation of the bearing capacity of the new internal metal trusses inside the Shukhov frame. For this reason, the scenario of increasing the reverberation of the hall due to the growth of the reflective properties of the wall surfaces associated with a change in the method of fastening the finishing panels is being considered. In the case of dismantling and re-mounting of panels without a gap with an aerated concrete base, the change in the sound absorption coefficient can be taken close to the values given in Table 2. The wall area is about 700 m2. Figure 5 shows the reverberation time measured at the end of construction and the reverberation time recalculated according to formula (1) taking into account the experimentally obtained values of Table 2.

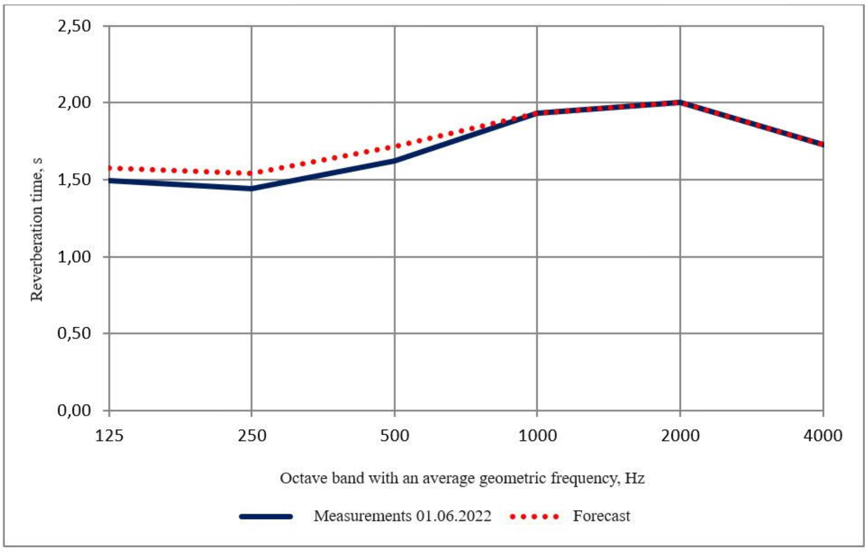

Figure 5. Reverberation time in an empty hall.

Figure 5 shows that changing the panel mounting method makes it possible to increase the reverberation time in octave bands with mean geometric frequencies of 125-500 Hz by about 0.1 s. It should be noted, however, that this estimate is qualitative and subject to clarification. Laboratory measurements are needed, taking into account the effect of attaching real large-format panels to the reflective base of the wall in a large sound measuring chamber.

4 The effect of a translucent wall on the acoustic characteristics of the hall

A special feature of the hall is the back wall of the stage, which is a glass translucent structure.

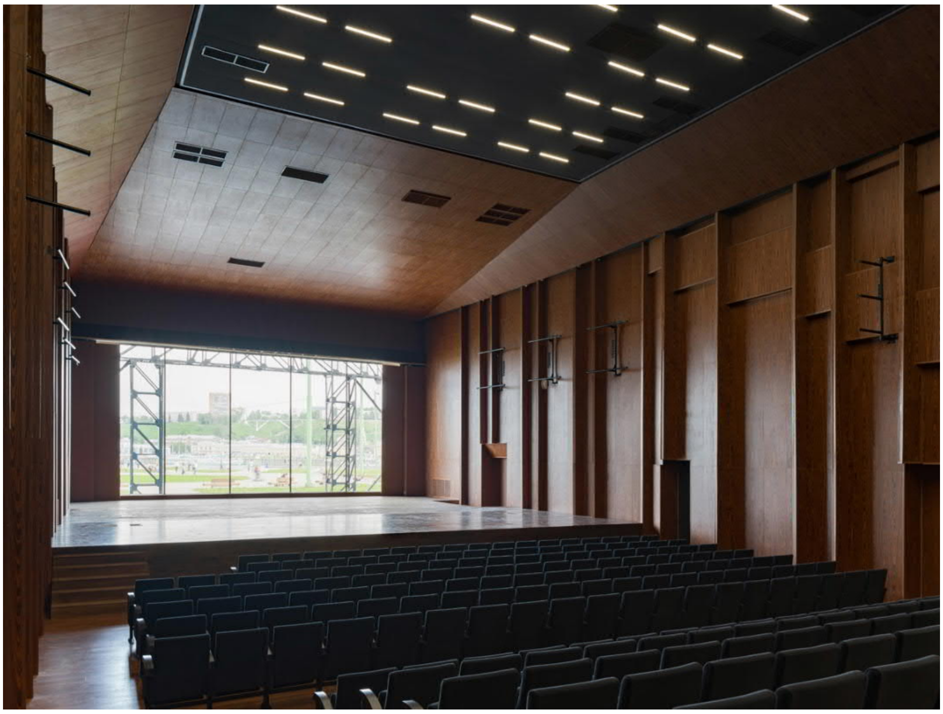

Figure 6. Translucent wall of the stage.

Depending on the artistic intent of the production, this wall can be closed with a sectional sliding partition (with a surface weight of no more than 10 kg / m2) or remain open. The analysis of the influence of the design of the back wall of the stage on the acoustic parameters of the hall for various positions of the sliding structure is carried out, the results are shown in Figure 7.

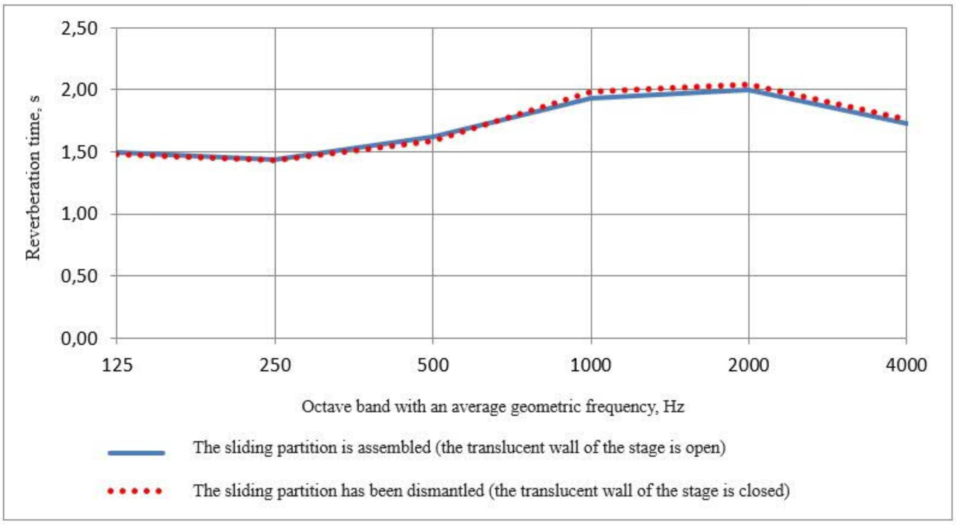

Figure 7. Reverberation time in an empty hall.

As can be seen, closing or opening the glass wall with a sliding partition does not have a significant effect.

5 Subjective assessment of the acoustics of the hall

On June 1, 2022, the rehearsal and opening concert of the hall took place. Theodor Currentzis (artistic director and chief conductor of the MusicAeterna orchestra), after the rehearsal and concert, gave a good assessment of the acoustics of the hall and, on the offer to play with or without sound reinforcement, chose to play the orchestra without sound reinforcement (in natural acoustics mode).

On June 1, 2022, there was a rehearsal and a gala concert opening of the hall. The Symphony Orchestra of the Nizhny Novgorod Opera and Ballet Theatre performed. The orchestra's chief conductor Dmitry Sinkovsky conducted. He praised the acoustics of the hall. Soloists, singers and musicians also spoke extremely positively about the acoustic conditions in the hall. The singers did not notice any negative effects (fluttering echo, theatrical echo – a large delay in reflections from the back wall coming back to the stage). The feedback from the audience was only positive.

It should be noted that Dmitry Sinkovsky stressed that a hall filled with spectators has a more preferable sound character than an empty one during rehearsal. This can be explained by the fact that in the filled state, the reverberation time in the hall at medium and high frequencies (over 250 Hz) is noticeably reduced, while at low frequencies it is practically not. That is, the frequency response of the reverberation time becomes more aligned and closer in nature to the design value. This means that if the proposals to increase the reverberation time at low frequencies are implemented, the hall will have acoustic characteristics close to the design ones (high reverberation time in the entire range of normalized frequencies).

6 Conclusion

Based on the results of the work, the following conclusions can be drawn:

1. A unique cultural facility has been designed and built, the central part of which is a concert hall with natural acoustics.
2. Despite the significant difficulties of the project implementation, the acoustic parameters lie in the corridor of optimal values, which is confirmed by the already established high acoustic reputation of the hall. The reduced reverberation time in the hall at frequencies below 500 Hz is explained by a deviation from the design values towards a decrease in the surface density of the walls and ceiling of the hall.
3. A qualitative experiment has been conducted confirming the potential of raising the reverberation time by changing the way decorative panels are attached to an aerated concrete wall. A quantitative assessment can be performed based on the results of additional studies and mock-up experiments in a large sound measuring chamber.

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Sound Levels at a Clarinetist's Ears During Solitary Practice

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This NTNU master's thesis presents a case study investigating the sound pressure levels (SPL) experienced by 4 clarinetists during individual practice sessions in both a semi-anechoic chamber and a regular practice room. The study involved both professional and amateur musicians. The measured difference between radiated sound power level ($L_{WA,instrument}$) and the direct SPL at ears ($L_{A,ears}$) have been found for test signals in the dynamic "forte", measured in the semi-anechoic chamber. For greater insight, the sound level contributions of the direct and the reflected sound at the musicians' ears have been determined for the practice room. The measured power-pressure difference ($L_{WA,instrument} - L_{A,ears}$) at "*forte*" was found to be 3.9 ± 0.3 dB (m = 32). The octave band level difference was however found to be lowest at 1000 Hz but most stable at 250 and 500 Hz. Results indicate that low values of $L_{WA,instrument} - L_{A,ears}$ correspond with the musicians' decreased ability to hear the room response. This is verified through determination of the direct and reflected sound contributions at ears in the practice room, where the 1000 Hz band alone showed a slight increase of direct sound compared to reflected sound. The findings provide new insights into SPL contributions at the ears of clarinetists during solitary practice, and the perceived sound field, while highlighting uncertainties for accurate assessments of sound power and pressure levels.

1 Introduction

Musicians in orchestras and woodwind bands constitute a unique group regarding daily exposure of sound, both due to levels and signal properties. Identifying the signal properties of musical instruments like the Bb clarinet is of high interest, as it can produce a great range of variety in properties like sound pressure level, directivity, frequency range and harmonic partials. Musicians spend much time practicing in solitude, in which the sound levels reaching their ears will include multiple varying contributions. For solitary practice, this is mainly the direct sound and reflected sound from the room. Separating the direct and reflected signals at the musicians' ears can help yield a greater understanding of the instrument. In previous research by O'Brien *et al.*[1], sound levels at musicians' ears during solitary practice in non-anechoic conditions. Moreover, findings by J. Meyer provide typical power levels (at the dynamic *forte*) radiated by various instruments [2]. However, little data is published regarding the corresponding ear levels from the direct sound contributions.

The Bb clarinet has been chosen as an isolated case study for investigating these concerns. The goal of this study 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 have been measured in a practice room, where a quantification has been made of the contributions from the direct and reflected sound. The conditions for the dynamic *forte* are of primary interest, relating to reference data by Meyer [2]. Specifically, a supplementing value for the expected A-weighted power-pressure difference $L_{WA,instrument} - L_{A,ears}$ between the radiated power (SWL) and the direct sound pressure level (SPL) contribution at the ears has been obtained, with assessments of the uncertainty.

2 Method

2.1 Measurement setup

The three main objectives of the test were as follows:

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, $L_{WA,instrument} - L_{A,ears}$, 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, $L_{A,ears,room,dir}$ and $L_{A,ears,room,refl}$, in 1/1 octave bands in a practice room.

The test was conducted in two parts; one in a semi-anechoic chamber[1] and one in a regular practice room. For both locations, measurements were made for a test signal in the dynamic *forte,* and for a structured practice session of 25 minutes. A total of 4 clarinetists participated in the study; 2 professional musicians and 2 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 findings by Meyer [2]. In addition, SWL was determined in the semi-anechoic chamber, both for test tones in *forte* and during practice sessions. Table 1 gives an overview of the measured parameters for the various measurement signals, in each location.

Table 1: Measured parameters in semi-anechoic room and in practice room.

Measurement	Parameter(s)	Semi-anechoic chamber	Practice room
Test signal (scale runs in *forte*)	L_{WA}, L_{WA}-$L_{A,ears}$	X	-
	$L_{Aeq25s,ears}$	X	X
Practice session (Warm-up, Music repertoire)	L_{WA}, L_{WA}-$L_{A,ears}$	X	-
	$L_{AeqT,ears}$	X	X
	$L_{CF,Max}$	X	X
Room acoustic parameters	G, T_{20}	-	X

The test signal indicated in Table 1 consisted of one upwards run and one downwards run of a major scale in two octaves, played at 70 beats per minute with the visual aid of a metronome. Measurements were averaged for 8 repetitions with the musician facing different angles, with increments of 45°.

For the practice session lasting a total of 25 minutes, a subset of three sections were used; one 5-minute free warm-up section, and two 10 minute freely structured practice sections of selected musical repertoire (J. S. Bach: "Gavotte" and "Menuett").

Ear-mounted microphones were used for measuring SPL at ears in both measurement locations, as well as a reference microphone position at 1.5m distance. An array was additionally used in the semi-anechoic laboratory, using a setup of 11 positions.

[1] A semi-anehoic room was chosen for determining the radiated sound power level according to the standard ISO 3744:2010 [3]. The method was deemed relevant since musicians usually perform or practice positioned above a reflective floor.

2.2 Determining SPL contributions in a non-anechoic room

In the non-anechoic practice 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, $L_{p,ears,room,refl}$:

$$L_{p,ears,room} = 10\log_{10}\left(10^{\frac{L_{p,ears,room,dir}}{10}} + 10^{\frac{L_{p,ears,room,refl}}{10}}\right) \quad (1)$$

It was desirable to identify these contributions in the practice room. This was done by combining level measurements, directivity reference data, simulation data and room acoustic measurements (including strength) with the classic formula:

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

where r_{ac} is an estimated distance between the ears and the acoustic center of the clarinet, determined from further reference/simulation data and semi-anechoic measurements. A full description of the calculations and estimation methods is available in the published master's thesis [4].

3 Results and further work

3.1 Test signal – scales 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. Previous research by T. Halmrast has shown that musicians tend to compensate unconsciously in absorbing environments [5], however the musicians were asked to perform the scale runs with as much dynamic consistency as possible.

Table 2 shows the arithmetic averages and standard deviations 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 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 $L_{A,eq25s,ears}$. The background noise was measured to $L_{A,background\ noise}$ = 17 dB, meaning that no corrections were necessary for determining the SWL according to the ISO 3744 method. Data have been added for SWL from Meyer [2], and for SPL at the musicians' ears from O'Brien [1] for comparison.

Table 2: Measured parameters for scale run in *forte*, in semi-anechoic room and in practice room, with 95% confidence intervals. Standard deviations in parentheses.

Measurement location/origin	L_{WA} [dB]	Left ear $L_{A,eq25s,ears}$ [dB]	Right ear $L_{A,eq25s,ears}$ [dB]	$L_{WA} - L_{A,eq25s,ears}$ [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*) [2]	93			
O'Brien (*ff*) [1]		97	96	

The measured SPL, SWL and power-pressure differences were additionally determined in 1/1 octave bands. Bands below 250 Hz are ignored, as the lowest note played during the test signals corresponds to a fundamental tone f = 233 Hz. The ear level $L_{Aeq25s,ears}$ was 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 inter-aural difference 0.9 dB). Figure 2 shows the corresponding arithmetic average of power-pressure differences, $L_{WA} - L_{Aeq25s,ears}$, in 1/1 octave bands.

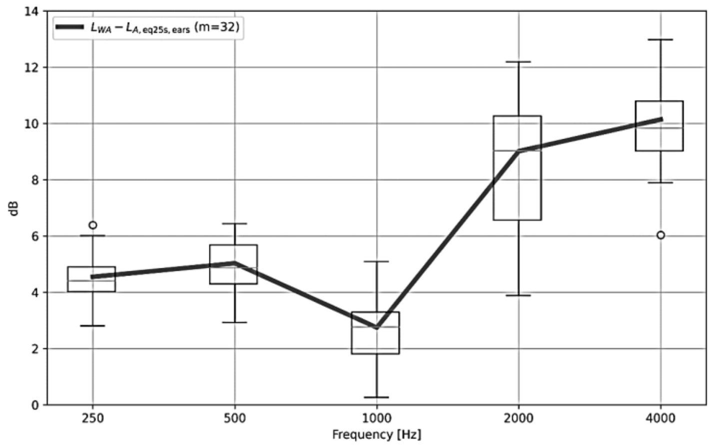

Figure 2: Arithmetic average of difference between power and ear pressure levels ($L_{WA} - L_{Aeq25s,ears}$) in 1/1 octave bands, for scale runs in forte, in semi-anechoic laboratory (m=32). Box plots show middle (boxes) and outer (whiskers) quartiles, along with medians (horizontal line) and outliers (circles).

The above figure 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.

The SPL contributions from the room and the direct sound were determined for the practice room. Figure 3 shows the measured $L_{A,ears,room}$ with the determined $L_{A,ears,room,dir}$ and $L_{A,ears,room,refl}$ in the practice room, in 1/1 octave bands, for test signal scale runs in forte.

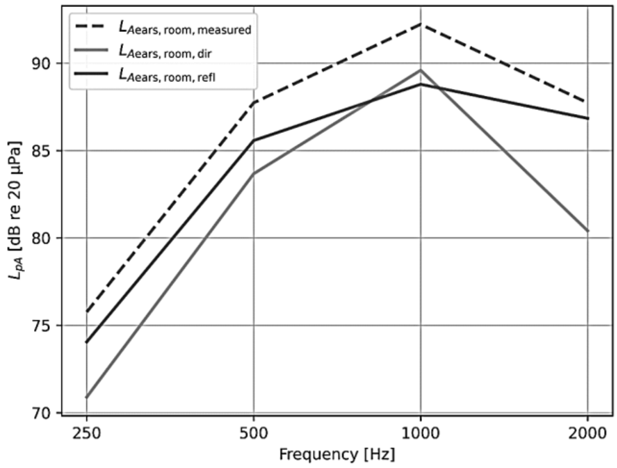

Figure 3: Sound pressure level contributions at ears for 1/1 octave bands, in decibels. Dashed line shows measured, total SPL at ears, while black and grey lines indicate the determined contributions of the reflected and the direct sound, respectively.

The calculated sound level contributions in the practice room presented in Figure 3, show that the reflected sound level was found to be higher than the direct signal for most evaluated octave bands, except for 1000 Hz. Hence, the musician hears “less of the room” for this band, implying a possible increase in directivity towards the musicians’ ears. As the curve in Figure 2 is otherwise increasing for higher frequencies, this indicates an expected directivity pattern “away” from the musician in higher frequency bands.

3.2 Technical measurement observations during practice sessions

Investigations of the power-pressure difference were done during practice sessions in the semi-anechoic chamber. The SPL measurements were analyzed over time, to investigate the variation of sound levels for the purpose of removing unwanted noise during rests and pauses. This includes reflections and resonances in combination with non-stationary noise from breathing, cloth rustling and similar. Figure 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 occurrences where L_{WAF} is in the immediate area below 65 dB re 1pW, but a 16% occurrence peak in the level range around 40 dB re 1pW. The occurrences in this lower-level range correspond to rests and breathing pauses of the musician, which is why the levels below the occurrence dip at 65 dB re 1 pW were gated out for this part of the study.

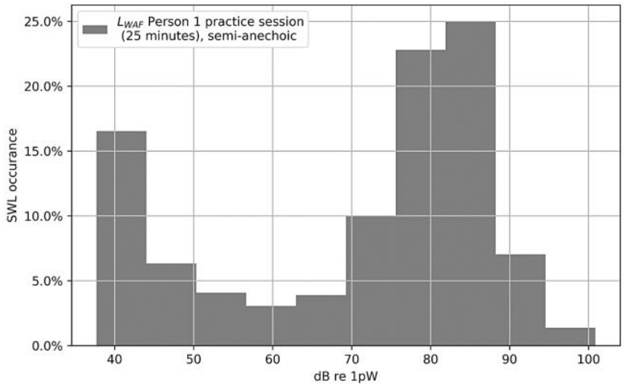

Figure 4: Example histogram for L_{WAF} values from entire 25-minute practice session in semi-anechoic chamber, from participant "*Person 1*". Occurrences approximately below 65 dB re 1 pW originate from rests and breathing pauses.

Filtering low levels due to rests and pauses was in addition found to have a stabilizing effect in determining the power-pressure difference. Figure 5 shows an example from one of the musicians during the "Gavotte" practice section, of how the $L_{WAF} - L_{AF,ears}$ difference value over time is affected by becoming less stable for lower values of L_{WAF}. This is especially visible in the right rectangular window. An equivalent power level $L_{WA,background}$ have additionally been calculated from the measured background noise in the semi-anechoic chamber, indicated in the figure.

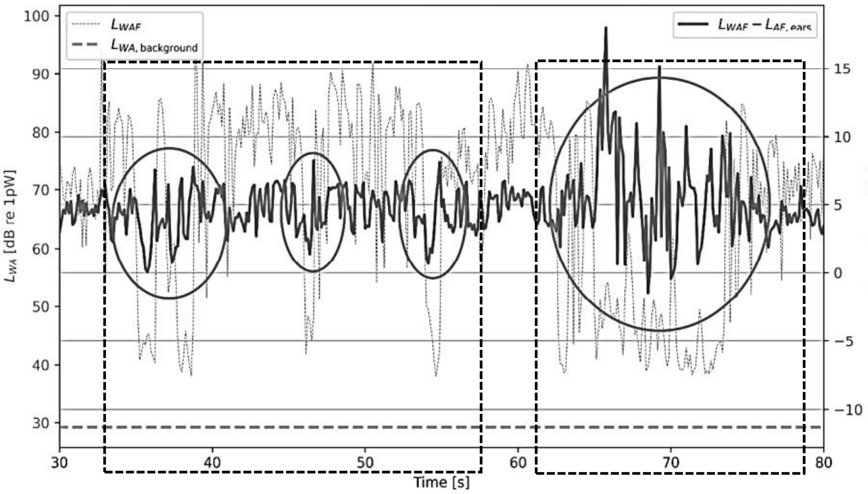

Figure 5: Example from Person 1, during practice of musical piece "Gavotte". Solid line/right y-axis shows $L_{WAF} - L_{AF,ears}$ over time, with the corresponding L_{WAF} value indicated with dotted lines/left y-axis. Background noise equivalent SWL $L_{WA,background}$ shown with dashed line. Occurrences where $L_{WAF} - L_{AF,ears}$ rapidly changes during rests and breathing pauses is indicated with circles.

3.3 Time envelope of power-pressure difference

Figure 5 shows that the most variation of $L_{WAF} - L_{AF,ears}$ occur in the moments where the instrument is silent, between the musician's playing. A trend can however be seen in the left rectangular window, in which the power-pressure difference displays sharp peaks whenever the clarinetist starts or stops playing. This suggests that large variations occur in the short time windows from the immediate starting/stopping of the direct signal, to when the effect from the contribution from the reverberant signal becomes relevant.

3.4 Possibilities for a simplified method

The method described should be further developed and if possible, simplified, with a verification of the agreement between the original and revised method. It is desirable to be able to extract similar information for an instrument by only measuring in a non-anechoic room, given a known and appropriate reverberant field. An energetic subtraction between ear levels and measurements in the reverberant field in the practice room gave a roughly estimated direct sound level of $L_{A,ears,direct}$ 92 dB, assuming an evenly distributed reverberant field. This value matches the measured sound pressure level at ears in the semi-anechoic chamber, for scales in *forte*. However, using this information to further determine the power-pressure difference requires information about the distance to the acoustic centroid. The underlying assumption about an evenly distributed reverberant field also places larger demands to the practice rooms.

A verification of the findings in the semi-anechoic chamber should additionally be done by comparing with measurements in a fully anechoic environment. Using a semi-anechoic environment was initially chosen due to being more representative of a realistic setting, though the resulting method proved to be unnecessary complex, requiring corrections from simulations of floor reflections, and thus increasing the uncertainty of results. Further work systematically investigating the agreement between methods with various degrees of simplification would be of high interest.

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Polish National Television production studios: acoustic design and performance

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This paper discusses the design, realization and acoustical performance of a newly opened large TV production studios in Polish National Television complex, in Warsaw, Poland. Several design aspects are discussed including location of smaller studios one over another, design of inner walls with both wide-band frequency absorption and impact resistance requirements as well as sound insulation of roof penetrations. Designs with computer models as well as the acoustical performance of the finished rooms are discussed. Construction details of the interior surfaces and of the technical installations and their influence on the acoustics are also discussed.

1 Introduction

In 2021, the results of a public tender for design&build of a new 7,000 m^2 complex of the Polish National Television production studios, inclusive of two large (~46 x 30 x 20m, 25,000 m^3 each) and four smaller studios (~28 x 21 x 8m, 5,000 m^3 each) was declared in Warsaw, Poland. The victors were PIG Architects, a Warsaw-based architectural firm and DORACO as General Contractor. The first author was enlisted to aid the architects in acoustically designing the complex. As of May 2024, the complex is finished, with a opening planned for June 2024. The initial building cost was estimated at 50 million Euros (net). Below are the images of the building exterior (Fig.1, left) and interior of the largest studio (Fig.1, right).

Figure 1: TVP television studio complex (left:exterior, right:interior of a large TV studio)

2 Building design

Design of the building was following a conceptual design provided by the Client. In this conceptual design and due to site size limitations, all four smaller tv studios were located one over another, without any buffer level between. Also, as all requirements given in Client specification were only related to laboratory values (R_W and not R'_W), it was not possible to convince General Contractor to invest into box-in-box design. Due to structural requirements and cost savings, no proper box-in-box design was implemented, making it difficult to secure high sound insulation between tv studios and forcing acoustical design to rely mostly on floating floors and thick monolithic reinforce concrete structures. Structural walls around tv studios were constructed as 30-35cm thick monolithic reinforced concrete, while floors between smaller studios were made as 60cm pre-stressed reinforced concrete plate with 20cm thick reinforced concrete floating floor on elastomeric springs. Walls around small tv studios were separated horizontally, to avoid vertical material sound and flanking transmission, however this elastic joint was located (due to structural requirements) below structural floor, and not above, which would be safer from acoustical point of view. All lift shafts around studios were separated from building structure to avoid structural noise transfer. As all HVAC installation were located at the roof, additional silencers and multilayer encasements were design around ductwork at roof penetrations. Most entrances and exits from tv studios were designed as acoustically dampened sound/light locks with double doors. Final building plan (Fig.3) and section (Fig.4) are shown below.

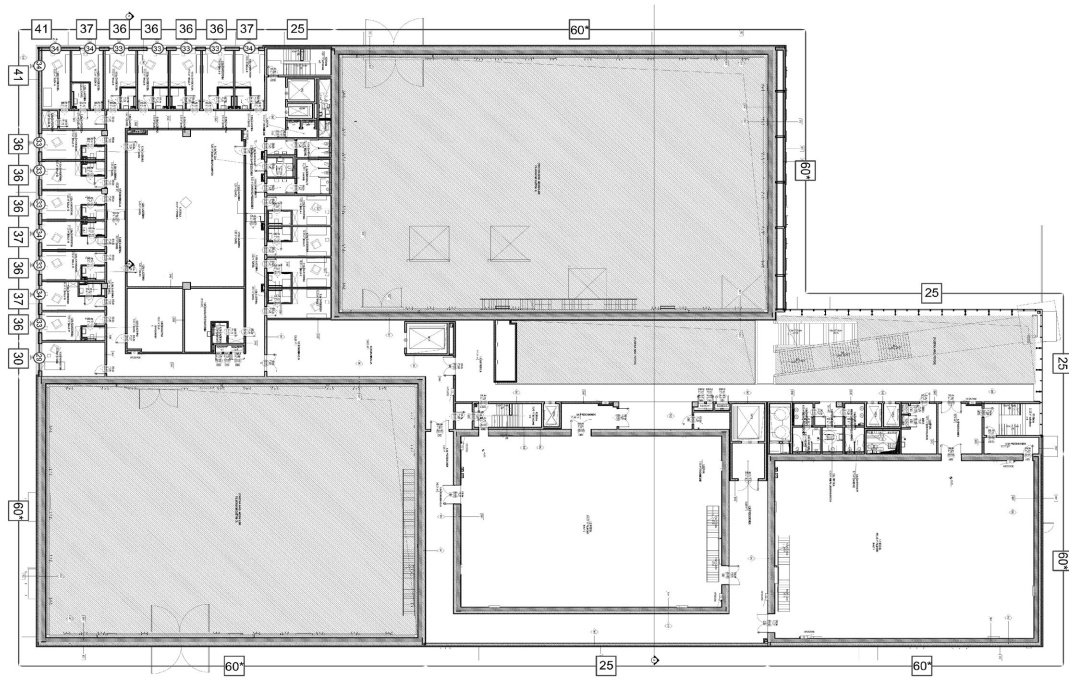


Figure 2: Plan, 1st floor

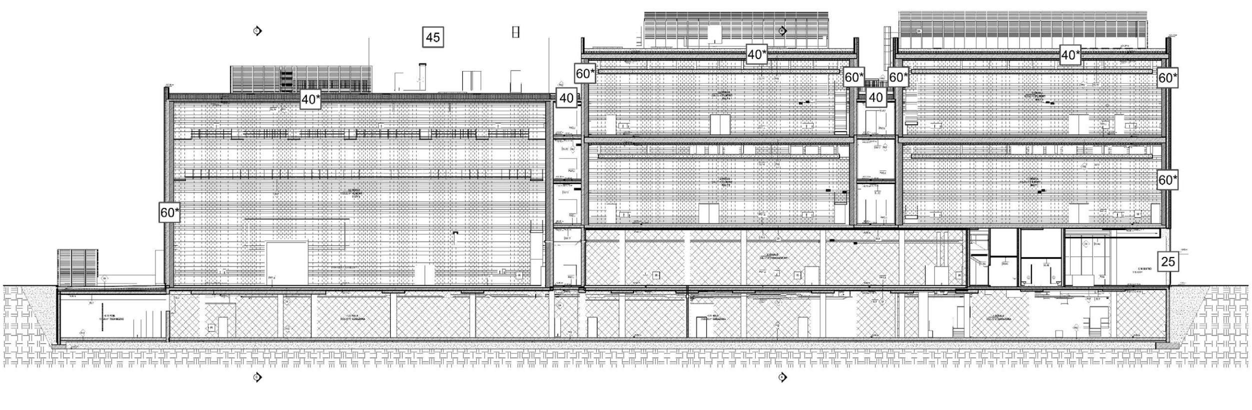
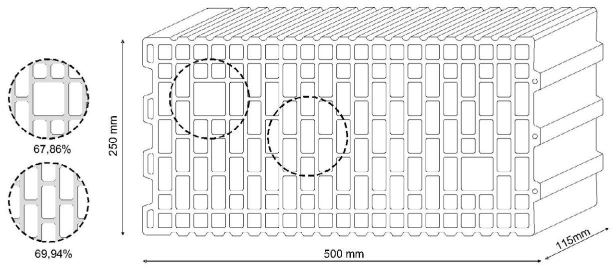


Figure 3: Long section through building

3 Room acoustical design

Room acoustical design of the tv studios interiors was following design specification provided by the Client. In large tv studios, required reverberation time was 0,9 seconds (+/- 0,2) and in small studios it was 0,6 seconds (+/- 0,2). To reach that level of reverberation time for such large rooms, almost every surface need to be sound absorbing. As there is no area left for dedicated low frequency absorbers, a special broadband sound absorbing cladding was designed. It was composed of a perforated brick, used as an aesthetically finishing and impact resistant element, with carefully selected mineral wool and airspace behind. For perforated bricks a hollow ceramic block, Thermoton 50-25 P+W, was used, measuring 500x250mm wide and 235mm deep, which was cut in half, to reduce it's depth to 115mm. Perforation ratio of brick elements was very high, between 67,8 to 70,0%, composed on several different square or rectangular openings with width between 18 and 40mm, but with relatively large depth of those holes (115mm). Bricks were installed on shallow concrete shelfs, 350mm from concrete wall behind. Behind brick, a single layer of 200mm glass wool (~15kg/m^3 and ~10kPa*s/m^2) was installed with additional 150mm of airspace behind. To choose best possible mineral wool, sound absorption of brick wall was first simulated in Winflag [1] and later measured in typical ISO354 laboratory test [2]. Decrease in high-frequency sound absorption, visible in simulation (due to large depth of perforations), was not confirmed in lab test. To save construction costs, upper parts of all walls were finished without perforated bricks, just with 200mm mineral wool with 150mm of airspace behind. Details of brick wall and results of it's sound absorption test are shown in Fig.4 and 5.



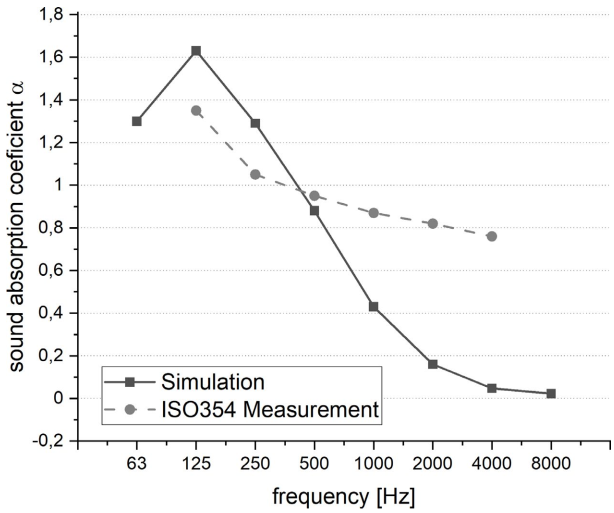
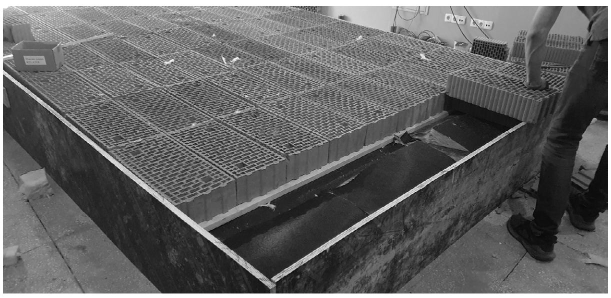


Figure 4: Brick cladding (left: brick detail, right: sound absorption simulation/measurement)



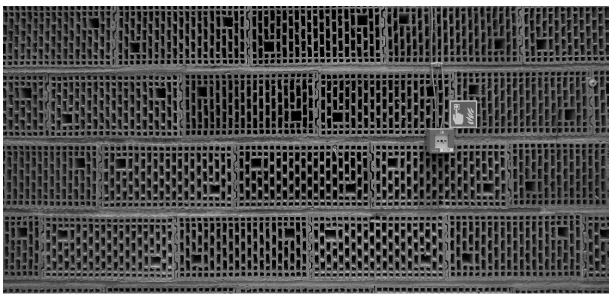


Figure 5: Brick cladding used on walls in all television studios (left: during ISO354 test, right: finished in hall)

4 Measurements results

In April 2024 authors had a chance to perform first few measurement session inside finished TV studios. We measured noise levels with HVAC system turned on, reverberation time in all studios as well as some basic airborne and impact sound insulation. Results from RT measurements from one large and two small TV studios are shown on Fig.6. During RT measurements, we detected some flutter echoes between flat mineral wool ductworks (installed under ceiling in small studios) and flat floor below, which needs to be eliminated. We also detected some reflections between vertical ductwork placed symmetrically on two parallel walls in one of large studios, as well as vibrating metal perforated grilles covering supply end of air ducts. Noise level, with all ventilation systems turns on, was measured at around 32dB(A), much below Client requirement of ≤ 37dBA. Airborne sound insulation between small modules (both vertically and horizontally) was measured at around $D_{n,tW}$=72dB, with sound insulation at 63Hz reaching around 48dB. Impact level between two modules (one over another) was measured at $L'_{n,tW}$=19dB, but additional test showed that floating floor resonance frequency seems to be much higher than what was calculated in design stage (~10Hz) so there is a risk that low frequency impact sounds could be transmitted from the upper studio to the lower one – more measurements are needed to analyse this.

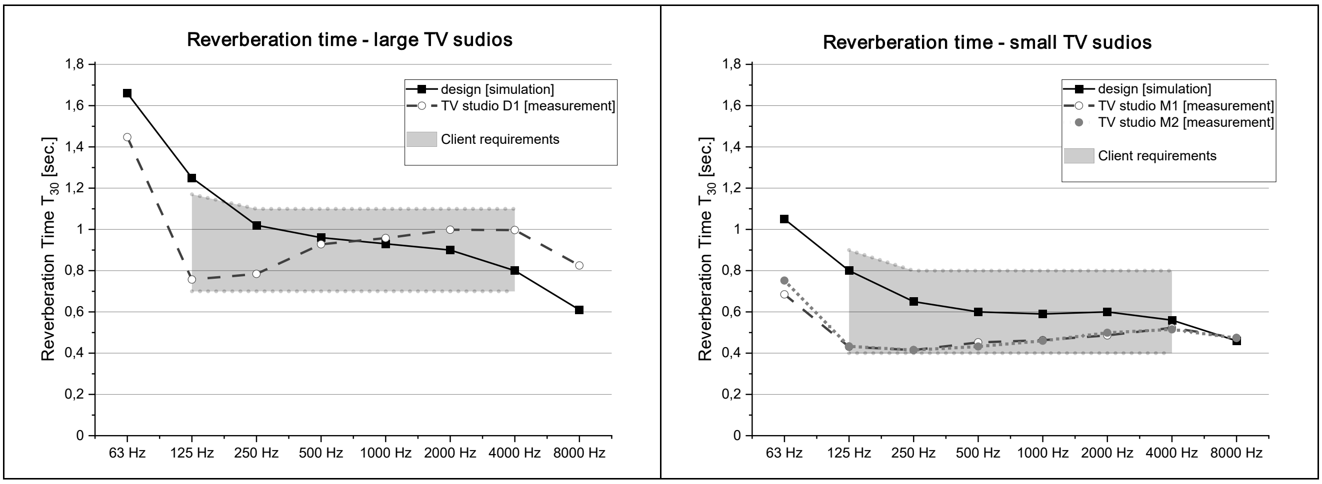


Figure 6: Design (Odeon simulation) and measured reverberation time in finished studios
(left: large TV studio D1, right: small TV studios M1 and M2)

5 Discussion

Acoustical design of such a challenging building, as TV studios, in the Design&Build formula can be difficult, when Client specification is not very specific (in acoustical terms) and conceptual design prepared by the Client is created only based on size requirements and site restrictions, without much of acoustical considerations. As almost all acoustical requirements result typically in increase of thickness of building elements and/or in increase of construction costs, well written Client specification in Design&Build formula is the only way to force General Contractor to introduce those changes into the design and into the final building construction. In this project, only reverberation time requirements were specifically written (at least to some extent), so implementation of room acoustical features was accepted by the General Contractor. However all those elements had to be acoustically designed with simple and cheap materials, not to increase the final construction cost. In respect to sound insulation specification, requirements defined with laboratory values are not very helpful in achieving proper acoustical separation in finished building. Also, restrictions in site size, should not be solved with placing two studios one over the other, without reserving extra space for proper box-in-box solution, or at least separating them with a buffer zone. As one can see from measurement results shown in previous chapter, main goals in that field were achieved, mostly in reverberation control, however simultaneous use of all studios, and transmission garage below and rooms above, has yet to be fully tested in real scenarios, to fully confirm it’s acoustical character.

6 Summary

It's an exciting and complex project. We're keenly awaiting the chance to perform final set of sound level and insulation measurements in the completed studios, as well as first TV recordings!

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Compliance procedures for sound insulation between dwellings in new housing – Rules according to Danish regulations & Experiences from practice

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National building regulations have existed in Denmark since 1961 and have included acoustic regulations for housing. The housing stock in Denmark consists currently (2023) of almost 2.8 mio dwellings, of these are ~40% dwellings in multi-storey (MS) housing, ~40% one-family houses and ~15% terraced houses (row houses). This paper focuses on compliance with limits for airborne and impact sound insulation between dwellings in new-build and apply to both MS housing and terraced houses (row houses).
Issuing of building permits and permits allowing use of buildings after completion are administered by local building authorities in the municipalities, and for decades they were involved in technical details related to the building permits. However, administration of building regulations and proof of compliance changed over time – with some of the changes unfortunately implying reduced options for check of compliance. In practice, compliance with acoustic regulations suffers from various shortcomings in the building process, some related to the builders' lack of understanding of acoustic regulations or lack of inspection of the construction work, others due to lack of compliance test or building authorities' lack of expertise with how to check the validity/invalidity of field test reports. The consideration behind the building regulations' requirement for documentation is partly to ensure that buildings comply with the requirements, and partly to ensure that later users of the building have a valid documentation basis. The paper describes details of the shortcomings and provides examples of severe construction defects being noticed mainly due to field tests following user complaints. Furthermore, indications of options for improvement of documentation procedures will be described.

1 Introduction

Acoustic regulations exist in most countries in Europe for different categories of buildings. Regulations and enforcement vary considerably between countries. This paper focuses on housing and specifically on compliance with limits for airborne and impact sound insulation between dwellings. In Europe, limit values are included in building regulations in more than 30 countries, cf. [1]. The paper provides brief information about the limit values and housing stock in Denmark and the compliance rules according to the current Danish regulations. The paper includes examples of new housing with sound insulation performance far below the requirements.

2 Sound insulation regulations in Denmark for new housing

2.1 Sound insulation regulations and classification

The current sound insulation regulations in Denmark are found in [2] and [3]. The requirements for housing have not changed since 2008. In [2] is referred to [3] for limit values. Additional guidelines related to field tests in Denmark are found in [4]. Since 2008, the regulations have referred to Class C in DS 490 [5]. Table 1 shows the sound insulation limit values from 1961 until now. An overview of the acoustic classes A-F in DS 490 is found in Table 2. Test methods for check of field performance are ISO methods, see [6] and [7], which are also implemented as EN standards and as national standards in CEN member countries. The methods also include requirements for the instrumentation, e.g. [8].

Table 1: Sound insulation main requirements [(1)] in the Danish building regulations for walls/floors between dwellings constructed in the period from 1961 until now [2].

Period	Housing type	Airborne sound insulation [(1)]	Impact sound insulation [(1)]
1961[(2)]-2008	Multi-storey [(3)]	$R'_w \geq 52$ dB (horizontal) $R'_w \geq 53$ dB (vertical)	$L'_{n,w} \leq 58$ dB
Since 2008	Multi-storey and row housing	$R'_w \geq 55$ dB	$L'_{n,w} \leq 53$ dB

Note 1: Limit values until 1982 are estimated by converting to the descriptors, R'_w and $L'_{n,w}$ applied in the current Danish building regulations.

Note 2: Before 1961, there were no general national building regulations.

Note 3: For terraced housing (row housing), the limit values were: $R'_w \geq 55$ dB from 1966 and $L'_{n,w} \leq 53$ dB from 1977.

Table 2: Occupants' expected satisfaction for different sound classes according to DS 490:2018 [5]. Summary based on information in DS 490.

Sound insulation between dwellings Main class criteria A-F in DS 490:2018			**Characteristics of DS 490 sound classes for dwellings and occupants' expected evaluation Information from DS 490:2018**		
Class	**Airborne**	**Impact**	**Sound class descriptions**	**Good or very good**	**Poor**
A	$R'_w + C_{50-3150} \geq 63$ dB	$L'_{n,w} \leq 43$ dB and $L'_{n,w} + C_{I,50-2500} \leq 43$ dB	Excellent acoustic conditions. Occupants will be disturbed only occasionally by sound or noise.	**> 90 %**	
B	$R'_w + C_{50-3150} \geq 58$ dB	$L'_{n,w} \leq 48$ dB and $L'_{n,w} + C_{I,50-2500} \leq 48$ dB	Significant improvement compared to minimum in class C. Occupants may be disturbed sometimes.	**70-85 %**	**< 10 %**
C	$R'_w \geq 55$ dB	$L'_{n,w} \leq 53$ dB	Sound class intended as the minimum for new buildings.	**50-65 %**	**< 20 %**
D	$R'_w \geq 50$ dB	$L'_{n,w} \leq 58$ dB	Sound class intended for older buildings with less satisfactory acoustic conditions, e.g. for renovated dwellings.	**30-45 %**	**25-40 %**
E	$R'_w \geq 45$ dB	$L'_{n,w} \leq 63$ dB	Sound class intended for older buildings with unsatisfactory acoustic conditions.	**10-25 %**	**45-60 %**
F	$R'_w \geq 40$ dB	$L'_{n,w} \leq 68$ dB	Sound class intended for older buildings with clearly unsatisfactory acoustic conditions.	**< 5 %**	**65-80 %**
Reference: DS 490:2018 "Lydklassifikation af boliger" (Sound classification of dwellings)			**Note:** Within each sound class, the percentage of satisfied or dissatisfied occupants may depend on the type of criterion. The grouping is mainly based on the subjective assessments of airborne and impact sound from adjacent dwellings.		

The first versions of DS 490 were published in 2001 and 2007, respectively, and with four classes A-D as in the other Nordic countries. DS 490:2001 was not formally related to the building code. DS 490:2007 was linked to the building code in 2008, since the acoustic requirements for housing were defined as fulfilment of Class C. In 2018, a revised version with two new classes E and F was published. The purpose was to have acoustic classes corresponding to older dwellings built before 1961, see Table 1 and Figure 1. For more information about the background, see [9].

2.2 Housing stock in DK

The housing stock in Denmark consists of about 2.8 mio dwellings [10], of these almost 1.2 mio dwellings in multi-storey (MS) housing, 450.000 row houses and most others single-family houses. In Figure 1 is found a diagram with number of MS dwellings according to construction year, and with the estimated acoustic class F, E, D, C according to DS 490:2018 [5], shown for various time periods.

The preferred type of dwellings changes over time. In the 25-year period 1990-2015, the number of dwellings in row housing and in multi-storey (MS) housing were almost identical. In the preceding 10 years (1980-1990), row houses were dominant, and since approx. 2016, MS housing has dominated.

From Figure 1, it is seen that in 2023, only about 10% of the MS dwellings can be expected to fulfil the current requirements. More information about constructions in various time periods is found in [9] and [11].

In the COST TU0901 books [12] and [13], more information is found about the housing stock in Europe (status 2013), focused on issues related to acoustic harmonization. An overview is given in [12] and information about individual countries (29 in Europe and 2 overseas) in [13].

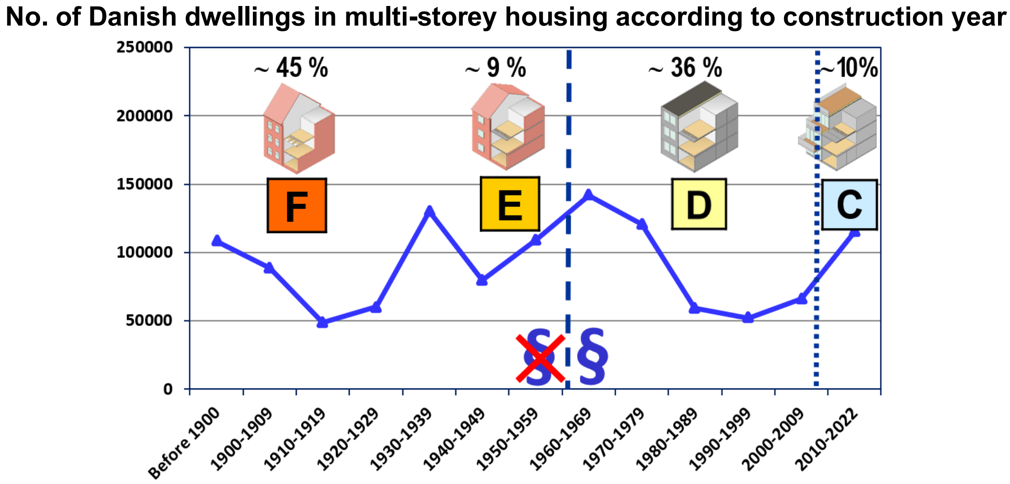

Figure 1: No. of Danish dwellings 1900-2022 in multi-storey housing according to construction year [10].
Estimated acoustic classes F, E, D, C according to DS 490:2018 are indicated.
More information about constructions in various time periods is found in [9] and [11].
Note: Time periods 10 years, except the first & last column. The dashed line indicates year 1961 with the first national building regulations. The dotted line indicates year 2008 with stricter sound insulation limit values.

3 Sound insulation compliance procedures according to BR18 and shortcomings

The current administrative procedures are explained in BR2018, Ch. 1 [2] and the related guideline [14]. To obtain a permit for starting construction, the builder must define the type of building and which Building Code Chapters (technical performances) apply. This applies to both MS housing and row housing. At the end of the construction process, the builder must upload documentation according to the Ch. 1 guideline, i.e. the chapters indicated in the application for the building permit. According to the introduction in [14] the purposes of the guideline are:

- To make sure that the requirements in the building regulations are fulfilled.
- To ensure that valid documentation is available for later users of the building.

According to experiences from many acousticians in Denmark, the reality is that the two above-mentioned purposes are far from being fulfilled in practice. Many cases about poor sound insulation appear due to complaints about neighbour noise at a point in time, where permit for use has already been provided, although compliance with the sound insulation requirements is not documented.

The procedure and main rules in the current BR2018, Ch. 1 [2], are as follows – with some comments added in Italic:

- A random sample of 10% of building cases is selected by building authorities for check of documentation. – *Is 10% enough?*
- Terraced houses (row houses) are exempted from check of sound insulation by authorities. – *However, in practice severe faults are not uncommon, and the number of row houses is large as mentioned in section 2.2.*
- Calculations considered sufficient for documentation [14]. – *However, practical experiences point to the necessity of field tests, since various mistakes and misunderstandings cannot be taken into account in calculations.*

Consequences of no field check or poor-quality test reports

The importance of regulations may be obvious, but proper design and quality control are necessary to obtain the desired quality, since uncertainties related to design and errors in workmanship are unavoidable in real life, not least when applying new solutions and materials. Learning from mistakes is important for the later building projects.

If severe deficiencies in sound insulation performance are not found or found too late, after people have moved in, people must suffer from poor acoustic conditions as long as they stay, if the problems are not solved, which may take years or might never happen. – After completion of buildings, modifications are typically very expensive in time and money. In addition, the process often implies a high mental load for the people involved.

An important issue when preparing valid documentation is also the quality of the test reports. The authors have seen test reports prepared by organizations or people without competences in acoustic regulations, test methods and equipment, and the result may be test reports misleading to both builders and building authorities or even “nonsense”. In one case, we observed that the building authorities considered such a “nonsense” report to be an accredited test report.

In [5], the clause about verification tests recommends that such tests are made by organizations with accreditation for building acoustic field tests or a person having a personal certification (does not yet exist in Denmark) for such field tests.

4 Field cases

Experience from practice have provided several examples of dwellings with sound insulation performance far below the limits in regulations. Thus, it is found important to initiate attention to the situation by informing about such cases from various parts of Denmark. The cases can hopefully lead to better rules and improved check of documentation.

Below is found information from four field cases:
Case 1: Row housing, airborne sound insulation.
Case 2: Multi-storey housing, impact sound insulation.
Case 3: Multi-storey housing with 240 dwellings – renovation.
Case 4: Row housing - 8 dwellings established in a former factory hall.

Case 1: Row housing with insufficient airborne sound insulation

In row-houses, double aerated concrete walls can provide very high airborne sound insulation provided they are constructed correctly. If not, in this case with 25 buildings, approximately 90 % of the dwellings did not fulfil the current requirements $R'_w \geq 55$ dB. One of the separating walls showed a 7 dB shortfall, see the lower curve in Figure 2. These results - found during control measurements between all the dwellings – would be unsatisfactory for the residents; since they would be able to hear the neighbours' conversations, thus causing complaints.

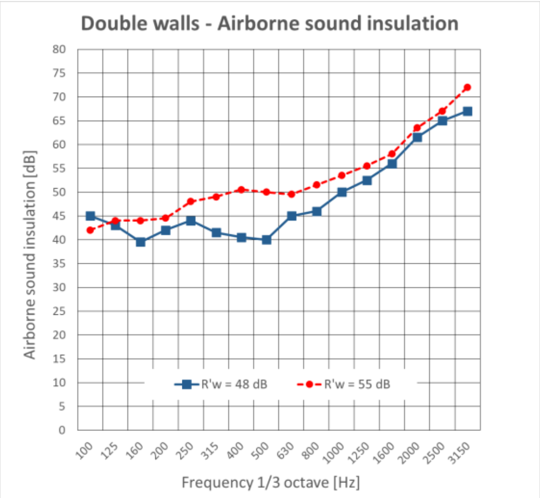

Figure 2: Diagram showing a good ($R'_w = 55$ dB) and a poor ($R'_w = 48$ dB) example from same building site

There are several possibilities for failures in double-wall constructions between dwellings, even if an elastic joint is made in the outer façade wall (to reduce flanking transmission through the facade). In this case, it was especially the foundations that messed with casting and bracing angle profiles that short-circuited the double walls, which should have otherwise been independent. After cutting the potential sound bridges, there were unfortunately still a few dB missing, which turned out to be caused by a too low separation in the foundation under the double walls (200 mm instead of 400 mm), see Figure 3.

The final solution included lining with two layers of plasterboard (one normal and one heavy weight plasterboard) directly mounted/screwed on each side of the double wall.

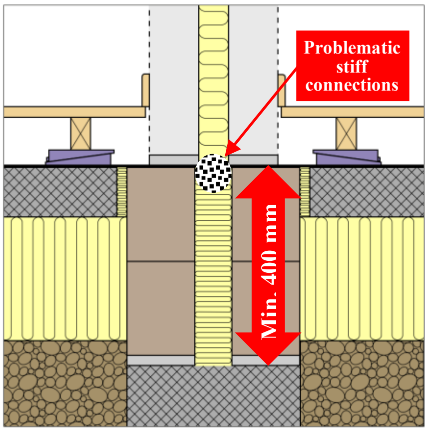

Figure 3: Vertical section showing a foundation/terrain floor assembly.

The remedy-cost was around 10.000 Euro per dwelling and the case was solved in two years. Even though the case is from before 2010 this one and several other cases show that is still highly relevant for the builders/investors of own initiative to perform control measurements, especially because it is now no longer permitted for the authorities to ask for such tests. Thereby it becomes the residents' own task to investigate and solve the problems – which more residents abstain from due to costs and lack of insight. Consequently, some residents may therefore live with unsatisfactory sound insulation, hearing noise and conversations from the neighbours.

Case 2: Multi-storey housing with too high impact noise levels

In an expensive 14-storey high-end building containing 25 dwellings with a sea view, high impact noise levels were measured in 2015 in living rooms. This resulted in resident complaints. A very extensive measurement program with 131 impact noise measurements showed that as many as 67% of the living spaces did not comply with the applicable minimum requirements in DS490 ($L'_{n,w} \leq 53$ dB) and had up to 9 dB's exceedances. Large variations in execution quality of the floors were revealed: Most exceedances were found in the dwellings in the right-hand stairwell, where 76% did not comply with limits, while it was 59% in the left-hand stairwell that did not comply. There were no impact noise problems from the very top floors. The problem turned out to be the "Foam-concrete" used (Thermotec/Thermowhite), see Figure 4, with a very variable stiffness – due to wrong mixing at the building site, i.e. elastic in the very top floors and quite hard in the other floors. This showed up as varying resonance peaks in the impact noise curves, see Figure 5.

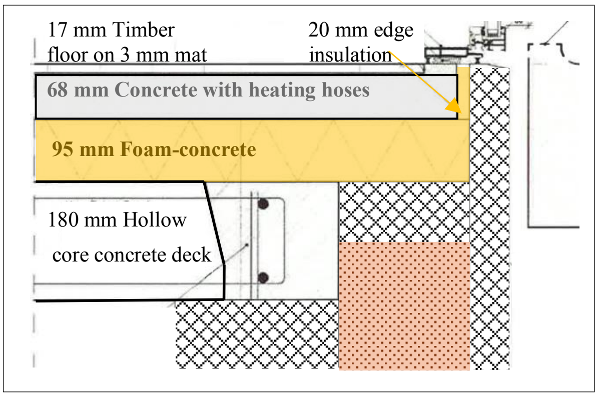

Figure 4: Vertical section at the facade showing the floating concrete floor on foam concrete.

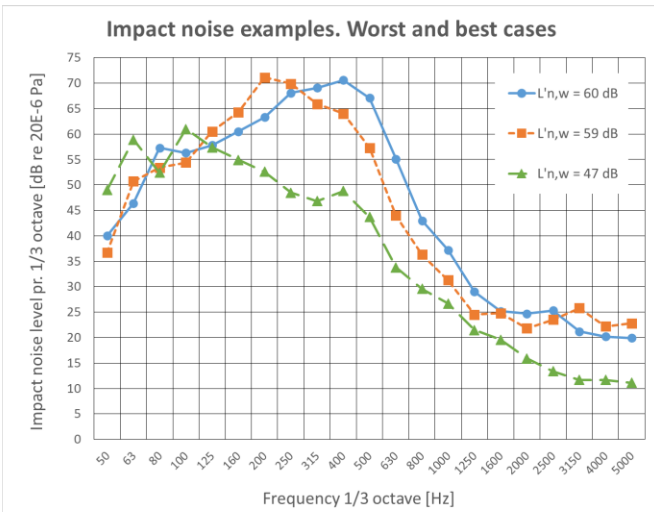

Figure 5: Diagram showing impact noise from floating floors in living rooms. One good ($L'_{n,w}$ = 47 dB from the very top floor) and two poor examples ($L'_{n,w}$ = 59-60 dB from lower floors).

The case was submitted to the opinion of an expert appointed by the court and a consequent very costly remedy (estimated to more than 100.000 Euro per dwelling): Rehousing, storage of fixtures, removal of wooden floors, concrete casting with underfloor heating hoses and Foam-concrete as well as restoration based on pressure-resistant/elastic mineral wool as a base for concrete casting with heating hoses and finished floors. A very long-lasting case that took more than 5 years. Consequently, control measurements are (still) very important and educational for both entrepreneurs and acousticians.

Case 3: Multi-storey housing with 240 dwellings – impact sound in renovated building

In a residential complex (public housing) with 240 apartments in 5-storey housing blocks, construction year 1980, a major renovation project was completed in 2022. The reasons for the renovation were a general need for maintenance, upgrading (new kitchen etc.) and a wish to have less maintenance costs in the future. Among other things, it was a goal to have maintenance-free floors that did not have to be sanded every time new residents moved into an apartment.

Original building: Walls of concrete, hollow core concrete slabs with wooden parquet on joists on underlay wedges. There was no information about neighbour noise complaints, and no sound insulation tests were made before renovation.

Goal for sound insulation in the renovated building: DS 490 Class C as for new-build, i.e. $R'_w \geq 55$ dB; $L'_{n,w} \leq 53$ dB. The old floors were removed, and new floor constructions were installed consisting of foam concrete (~50 mm), impact sound insulation mat (~6-8 mm), concrete (~40-60 mm), wooden click floor (easy to replace in the future).

After renovation people moved in. Soon after there were many neighbour noise complaints, mostly about impact sound. Consequently, vertical airborne and impact sound insulation tests were made and large deviations from the targets found:

- Impact limit $L'_{n,w} \leq 53$ dB – exceeded by more than 10-15 dB (i.e. impact levels far too high).
- Airborne limit $R'_w \geq 55$ dB – results low, more than 5 dB below limit.

From the inspection of the building, it was concluded that the concrete layer was connected to the walls, implying a strong vertical flanking transmission of both airborne and impact sound. This type of construction fault is typical and has been known for decades, see illustration in COST TU0901 book 2 [13], Ch. 5, Figure 5.12. The reminder could be that knowledge about design and details, visual inspection and early pre-completion testing are all important issues, so the same, severe construction fault could be detected and avoided, before being made in 240 apartments.

Estimated costs for improvement to comply with Class C limits were huge: ~40.000 EUR/apt, i.e. ~10 mio. EUR in total.

Conclusion: Currently (April 2024) a decision is pending about what could be done and when – and who should pay.

Case 4: Row housing - 8 dwellings established in a large, former factory hall

Case from 2017: Old brick building (factory hall) has been converted into dwellings (8 row houses). Thus, the building code requires that the sound insulation between the dwellings must fulfil DS 490 Class C.

The row houses have been constructed in the former factory building, each with basement, ground floor, first and second floor. Price around 1 mio. EUR each (2017).

All owners found the sound insulation unacceptable, and they ordered a consultant to do tests (2019). In total 16 tests of airborne and impact sound insulation were made. A few tests between rooms in the basement and in the upper floor fulfilled the required Class C. But the sound insulation tests between living rooms, ground floor and first floor, did not fulfil Class C (and also not Class D). The test results for airborne sound insulation were R'_w 40-49 dB (requirement $R'_w \geq 55$ dB) and for impact sound $L'_{n,w}$ 64-62 dB (requirement $L'_{n,w} \leq 53$ dB). Thus, the results correspond to the lower classes E and F, which are definitely not intended for new dwellings, cf. Table 2.

For a long time, there seemed to be no solution, since the builder was bankrupt (but had a large sum of money in another company), and the insurance company refused to pay for "repair" of the sound insulation. But then the house owners had contact to a lawyer, who initiated negotiations between the builder and the insurance company, and after two more years, some sound insulation improvements were made for walls and floors. It is not known exactly, which changes were made. A new sound insulation test was made, but the test report is not made available for this paper.

Experiences from DK and other countries about monitoring of compliance with building regulations

In [15] (1994) is found information about acoustic conditions in housing in Denmark, including a brief summary of legislation over time and enforcement. It is explained that the acoustic quality of dwellings depends strongly on the number of control tests, and it is stated that enforcement varies across the country. A figure from [16] shows how increased enforcement in Jutland implied a significant positive increase in compliance rate from 20% to 70% in the period 1975-1987. In [17] it is described how simple construction errors unfortunately keep reducing the sound insulation in housing significantly. International information is also included in [16], and related to the subject of this paper, there is interesting information about enforcement in Austria in 1994, where survey results for different regions showed various strategies and policies.

In [18] are found very interesting, recent survey results for Spain about compliance procedures and testing in different areas in the country. Of special interest are that testing is mandatory in some communities and that a "registry of competent entities for the performance of acoustic tests" is made by quality departments in some communities or local governments.

In UK, there is a long tradition for doing research and surveys about neighbour noise and for registering noise complaints. In [19] (from 1997) are included both information about noise sources, sound insulation field test results, how poor sound insulation affects life and emotions related to neighbour noise. A coordinated approach for improving sound insulation in new housing is found in "Robust Details" [20], which includes construction designs, acoustic site inspection, checklists, sample field testing etc. In UK, building acoustic performance compliance levels increased from 40% (floors) and 60% (walls) before Robust Details approach to 98% and 99%, respectively, by using the robust details approach. Noise complaints have reduced by a factor of 3 for new build attached housing, see [21] and [22].

In the COST TU0901 books [12] and [13], more information is found related to the issues of this paper. In book 1 [12] see especially Ch. 9 (Monitoring & Testing Sound Insulation…) and 10 (Common Errors and Good Practice…). Book 2 [13] provides information about individual countries (29 in Europe and 2 overseas) about housing stock, typical constructions and typical construction faults in design and workmanship.

In the Danish standard DS 490:2018 [5] for acoustic classification of dwellings, the clause about verification tests (Appendix B) recommends that such tests are performed by organizations with accreditation for building acoustic field tests or a person having a personal certification for such tests. Design guidelines (in Danish) for new housing and improvement of existing housing are found in [23] and[24], respectively.

5 Conclusions and recommendations

Based on Danish field cases described in Section 4 and experiences from several countries, severe faults in building constructions are still common, even faults being well-known for decades. Several examples are found in [13]. Improved enforcement of regulations would encourage higher awareness of the regulations.

Important issues in the planning and construction process are:

- Planning of constructions
- Workmanship
- Visual inspection
- Pre-completion testing in case of new construction types, e.g. wooden constructions.
- Pre-occupancy testing
- Permit for use only obtained if requirements are complied with.

Recommendations for administration of acoustic regulations for housing in Denmark:

- Building regulations must continue to include clear requirements, including test methods and limit values.
- Row housing to be included in check of documentation for acoustic performance.
- Percentage of check by building authorities to be increased? (currently 10%).
- Field tests must be carried out as measurements. Calculations cannot replace field tests.
- Preparation of a guideline stating minimum competences for people performing acoustic field tests. A sort of approval or certification of such people would be very useful and the feasibility should be investigated.
- Preparation of a guideline for building authorities on how to check validity of acoustic field test reports.
- Enforcement of regulations to be performed by building authorities in all municipalities.
- Acoustic quality of dwellings to be included in documentation for performance of dwellings as "open access" for potential users, both tenants and buyers – in line with the intention of the guideline for documentation.

The background for the above-mentioned issues and recommendations are that residents need privacy and opportunities for own activities without disturbing neighbours. A large part of the existing housing stock built before 1961 in Denmark does not offer privacy or such opportunities and could be characterized as *acoustic slum*, cf. [24], which should be avoided both when renovating housing and especially in new and future housing. – At last, it is important to emphasize that leaving the acoustic quality check to the residents creates a very stressful situation often lasting years. Instead, the responsibility belongs to the builders and administration procedures developed by building authorities.

Acknowledgements

The cases presented in Section 4 are prepared by Sweco (Cases 1 and 2) and BUILD (Cases 3 and 4). The authors thank Lasse Møller Pedersen, Municipality of Aarhus, for discussions on administration of BR2018 compliance procedures.

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Note: Ch. 2. Existing Sound Insulation Performance Requirements and Classification Schemes For Housing Across Europe
Note: Ch. 9 Monitoring & Testing Sound Insulation Performance in New Homes in Europe.
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Note: The book has in total 31 chapters with individual chapters for 29 countries in Europe and 2 overseas countries, each chapter explaining the national housing stock, acoustic regulations and classes, typical constructions for new-build and older housing and typical errors in design and workmanship. – Ch. 5 is about Denmark.

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Paper to be presented at the BNAM 2024 conference

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Outdoor music event noise levels and limits

Abstract

Outdoor events in Finland, do not have a uniform noise limit for compliance. When organising outdoor events, such as music festivals, which continue after 22.00, organisers are required to submit a *notification of noise* (meluilmoitus) to inform the local environmental authorities, who then derives a decision with possible noise limits and mitigation measures.

The purpose of limit values for noise from the environmental authorities' point of view is primarily to ensure that the noise emitted during an event does not exceed legal limits for noise toward residential buildings.

Traditionally, Finnish authorities have set an A-weighted 5-minute equivalent level limit value ($L_{\mathrm{Aeq,5min}}$) for the FOH and/or nearby residential buildings' facades and require that the organisers monitor, record and report noise measurement results within a specified time frame. The process and resulting decision making varies greatly between regions and cities.

The A-weighting, which adjusts the sound pressure levels to reflect the sensitivity of the human ear, largely undermines the loudness of the noise emitted by live music acts, where middle and low frequency sounds are prominent. Low frequency noise also travels further, penetrates façade structures, and is often considered more annoying than other frequencies.

We will discuss the limitations of the current decision-making process and infer the implications of different frequency weightings for the noise limits of outdoor music events in an assessment to potential improve the current methods.

1 Introduction

Outdoor events in Finland do not have a uniform noise limit for compliance. When organising outdoor events, such as music festivals which continue after 22.00, organisers are required to submit a *notification of noise* (meluilmoitus) to inform the local environmental authorities, who then derives a formal decision with possible noise limits and mitigation measures.

The purpose of limit values for noise from the environmental authorities' point of view is primarily to ensure that the noise emitted during an event does not exceed what could be considered "unreasonable noise exposure" for nearest habitants.

Typically Finnish authorities set an A-weighted 5-minute equivalent level limit value ($L_{\mathrm{Aeq,5min}}$) for the FOH (Front of House/mixing booth) and/or nearby residential buildings' facades and require that the organisers monitor, record and report noise measurement results within a specified time frame. The process and resulting decisions vary greatly between regions and cities.

The A-weighting, which adjusts the sound pressure levels to reflect the sensitivity of the human ear, largely undermines the experienced loudness of the noise emitted by live music acts, where middle and low frequency sounds are prominent.

Low frequency noise also travels further, penetrates façade structures, and is often considered more annoying than other frequencies.

We will discuss the limitations of the current decision-making process and infer the implications of different frequency weightings for the noise limits of outdoor music events in an assessment to potential improve the current methods.

2 The Process

A local environmental authority is responsible for the handling of notifications of noise, as well as setting any potential noise limits or other requirements for an event. There is no standard, nationwide approach to any part of the process, which is loosely described in the environmental protection law [1]. This chapter will describe the different parts of the process around managing noise at an outdoor music event, common practices, and brief anecdotal examples.

The typical timeline of the noise-related process in organising an outdoor music event is shown in *figure 1*.

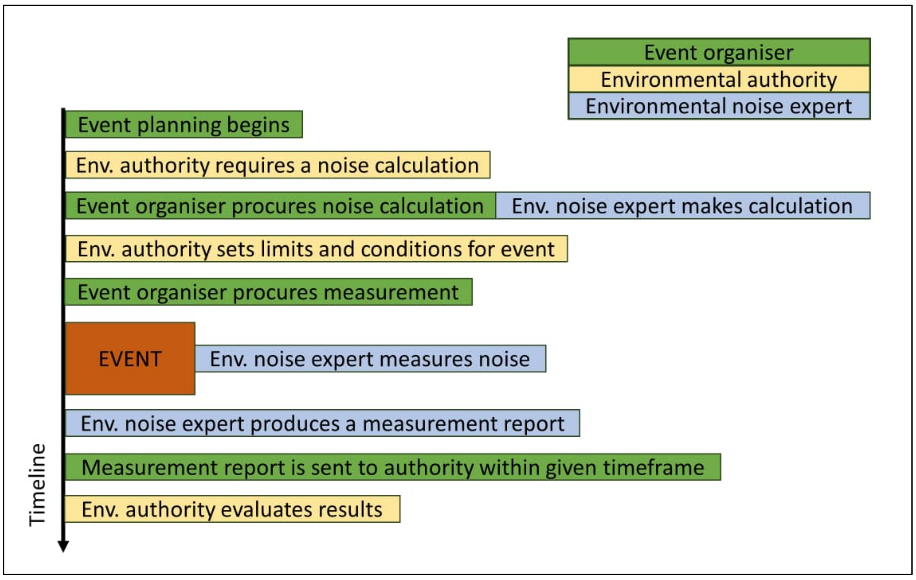

Figure 1. Typical timeline of environmental noise related things for an outdoor music event.

It should be noted, that while even cities' own websites refer to the decision as a *permit*, it is in fact only a decision made by the local environmental authority, which is published as a section of the meeting minutes which regards the event in question. Still, this decision sets legal requirements for the limits and conditions, which event organisers must follow. Failing to do so could be considered an environmental crime. Applying for an actual environmental permit is a lengthy process, which would be too cumbersome for event organisers to follow through (and for the local environmental authority to review) for events lasting only a few days at most, where an actual noise permit is meant for longstanding, more permanent operations, such as industry.

For the sake of simplicity, the *formal decision* will in this paper be referred to as a *permit*, despite it not being entirely accurate.

2.1 Notification of Noise

When organising an outdoor music event where music will be played after 22.00, the organiser must submit a notification of noise to the local environmental authority. There is currently no nationwide standard approach or template for a notification of noise submission. The information which is required to be presented in a notification of noise varies by the city or municipality where the notification of noise is submitted and handled. The only distinct rule in the process is that the notification must be submitted at least 30 days prior to the event [1].

In Helsinki and some other major Finnish cities, the standard approach for large outdoor music events has been to submit report showing an estimate (noise model) of the noise emitted into the environment, as well as a measurement plan.

A noise model aims to calculate how the environmental noise will spread into the environment, and what the noise levels would be in the affected areas. A model incorporates a 3-D terrain model of the environment (incl. buildings) with the best available data or estimation of the noise source (SPL, directivity, spectrum).

A common process for noise modelling is to utilise a file prepared by sound engineers using a simulation tool (e.g. ArrayCalc from d&b audiotechnik) and import the settings into an environmental noise modelling software like NoizCalc or CadnaA. Some modelling software employ different calculation methods, some of which allow for a more precise parametrisation of the environmental conditions. However, as the precise weather conditions cannot be predicted several months before the event, there is little benefit to using more complicated calculation methods particularly in event noise. The comparison between noise calculation models has been ongoing discussion [2], and given the uncertainties regarding event noise and repeatable weather conditions, a more detailed input does not always add value to the final result.

It is commonplace for noise calculations to be made under the assumption, that L_{Aeq} 75…80 dB should not be exceeded at the facades of nearby residential buildings. The basis of this (mostly) unspoken rule will be discussed further in chapter 2.3. With L_{Aeq} 75…80 dB at facades being the limit in planning, noise models will be adjusted so that the SPL at FOH will correspond to a level below L_{Aeq} 75…80 dB at facades. This gives FOH staff an upper limit to work with. It may also be possible to change stage positioning or plan mitigative efforts beforehand.

A measurement plan is based on the noise modelling results, known noise-sensitive buildings (eg. residential, healthcare, educational), previous events or measurements for similar events or the same venue/area, and guidelines from the local environmental authority. Noise may also be measured in places, which represent an important location with adequate accuracy, if the specific location proves logistically challenging. As an example of this, event noise from the highly popular outdoor venue and former industrial plot Suvilahti has on the west side been measured on the roof of an office building standing closer to the event area than the residential building, that the measurement result should represent. The measurement point is more easily accessible and is technically a simpler measurement point (direct view of festival area, few major reflections). A coarse level correction is calculated for the result to represent noise levels on a façade further away, and a reasonable margin of error is considered.

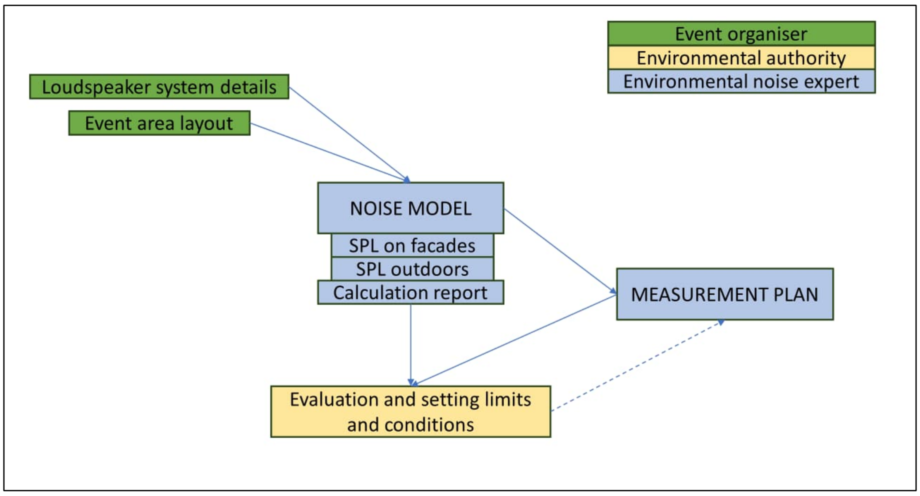

Figure 2. The noise modelling and measurement planning process.

2.2 Permitting

Currently there is no nationwide standard approach to the permitting of outdoor events, or limits and requirements set to the event organiser. Therefore, the permitting and setting limits depends entirely on the local authorities, and their collective or individual knowledge on the matter of environmental noise.

The document by the authority contains the limits and conditions set for an event, and acts as permission to arrange an event, when following the given conditions and limits. The permit also contains other conditions such as waste handling, which concern an environmental authority, but only the noise-related attributes are presented in this paper.

Permission from a local environmental authority to organise an outdoor music event is, unsurprisingly, an environmental permit. As such, failing to comply with any of the conditions of an environmental permit could be considered an environmental crime. In the authors' experience, event organisers have not been penalised for exceedances of set limits or delays in required documentation. There are however cases, where some event organisers have faced trial for seemingly failing to comply with limits and conditions set for an event. Despite the rigorous process of a court case, actual proven exceedance has very rarely been proved to occur in the judges decision. In some cases the event organiser has simply gotten away with an apology, slap on the wrist, or at most cancelling some future events. Out of discretion to the event organisers, the authors will not publish the names involved in such cases.

The city of Helsinki has set a limit on the number of days, when outdoor music events may continue until 24 or 01. These days are typically budgeted for known areas or venues and, in some cases, with specific events in mind. Nonetheless, each event must fulfil the process of notification of noise in due course. Events in areas or venues outside of those shown in *figure 3*, are considered at a case-by-case basis.

	Event days ending at 22–01	Event days ending at 22–24
Kansalaistori and Töölönlahti Park	0	10
Suvilahti	2	10
Olympic Stadium*	0	16
Kaisaniemi	0	7
Former Malmi Airport	2	7

Figure 3. Helsinki has a set number of event days, where the show may continue until 24 or 01 by location (Hel.fi).
**the number for the Olympic stadium is for music events only, not including sports events.*

2.3 Limits and Conditions

Currently there is no nationwide standard approach to limits and conditions set for outdoor music events. The approach may even vary depending on the specific location of the event, or the individual within the local authority handling the permitting. Two very common outdoor music event locations in Helsinki, The Olympic stadium and the Suvilahti area, have in the past been treated differently in terms of limits and conditions set for noise.

The Helsinki Olympic stadium, opened in 1938, hosts a major share of large concerts in Helsinki. The venue is situated near some residential buildings to the west, as well as a collection of healthcare facilities to the north.

Rammstein played two concerts on consecutive days at the Olympic stadium in Helsinki, in 2023. The concert layout was very typical for the Olympic stadium, with one main stage in the stadiums south end. In 2023, the following limits and conditions (paraphrased) were set for the Rammstein concerts [3]:

1. *Set times for using the PA-system, including one day for testing between 09.00 and 22.00, as well as two concert days for testing and performing between 09.00 and 24.00.*
2. *Concert noise must be measured daily from 22.00 until the shows end. Measurements should be continuous and supervised. A total of four measurement points (specified in document) in the nearby area were required, as well as one measuring point at FOH. Measurements should be conducted according to YM 1/1995. Measurements should be conducted using calibrated instruments, preferably class 1, but class 2 is acceptable.*
3. *The measuring party must be deemed competent by the Finnish Accreditation Service FINAS or certified by the Finnish environment institute.*
4. *A measurement report should be submitted to the environmental authority within two weeks of the event. The report should include the A-weighted 5-minute average levels LAeq,5min and spectrum of noise on the facades from 22.00 until the show ends. The report should also include a list of measuring equipment used, exact measuring locations and a detailed explanation of the weather, background noise, and the potential effects of those on measurement results.*
5. *Noise mitigation efforts should be discussed with nearby healthcare and childcare/educational facilities' managing staff.*
6. *Residents in nearby residential areas must be informed five days prior to the event, of the testing and show schedule, and given a number to which feedback can be given during the event.*

In recent years events at the Helsinki Olympic stadium have not been subject to any formal limits for noise, as shown in the case of Rammstein 2023. Earlier, however, there has even been a limit of $L_{\mathrm{Aeq,5min}}$ 80 dB for noise on facades, the last one being set in 2015 [4].

The Suvilahti area in eastern central Helsinki hosts some of the largest Finnish festivals annually. The festival area is surrounded by residential buildings in nearly every direction, including some residential towers reaching above 100 metres in height.

Flow festival is a three-day music festival with four outdoor stages, organised in the Suvilahti area. In 2023, the limits and conditions were like those given for the Rammstein concert at the Olympic stadium, with an exception in the number of days for testing and performances. However in 2022 and 2019, a limit of $L_{\mathrm{Aeq,5min}}$ 65dB was set for the time between 20.00 and 01.00, and in 2018 there was also a 75 dB limit for $L_{\mathrm{Aeq,5min}}$ on facades [5][6][7].

Measurement points specified in the noise permits are generally chosen based on the best available information regarding the environmental noise propagation from the event area. The best available information would ideally be a measurement plan, based on calculation results from an environmental noise model, as described in 2.1.

2.4 Measurements

Measurements are commonly required to be supervised. In addition to measurements where someone is physically present in the measurement location, a commonly accepted method is setting up measurements for remote control and live value reporting. Sound engineers may be given a direct link to view sound pressure levels, thus shifting the responsibility of monitoring directly to a person with the ability to intervene if a noise limit is exceeded.

Some events do not impose limits or require supervision at all.

2.5 Reporting and Evaluation

The permit often includes some specific values to be presented in the measurement report. Regardless of these requirements, the following values are commonly reported by companies producing accredited measurements:

- A-weighted 5-minute average level $L_{\mathrm{Aeq,5min}}$ for the entire duration of the measurements
- Unweighted frequency spectrum during concerts
- Weather conditions and a reasonable estimate of audible event noise vs background noise in the measurement result

Depending on the conditions set in the permit, these values may be presented as measured, or with corrections to account for any reflections or distance-related offsets between a measurement point and a noise sensitive receptor.

Measurement reports are typically the only objective source of information for environmental authorities to evaluate whether conditions have been met and limits have been respected.

The city of Helsinki also collects feedback for outdoor music events in a residents' survey, the results from which are used to evaluate the subjective experiences of residents. The survey has been completed in both 2015 and 2018. The results of this vast undertaking are used as feedback in the future decision making regarding events in the area, where noise is one of the many aspects to be considered [8][9].

3 Legal Basis for Noise Limits

Noise limits exist primarily to protect individuals and communities from noise, which can cause disturbance or even have negative health effects. There are various limits set for different types of noise, some of which specify a source, others the type of noise heard, and others set only a general limit. Noise permits for outdoor music events do typically not reference any limits, as such events are generally infrequent and may as such be considered outside of the context of some noise legislation.

4 Shortcomings and Challenges

A lack of expertise within some environmental authorities is a major cause of the varying approaches to outdoor music event noise and permitting for it. As there is no standard nationwide approach to any of it, the decision-making depends entirely upon the individual working on the case. In the experience of the authors, variances in decision-making often reflect the experience of the individual behind the decision and their possibly limited understanding of environmental acoustics. This is understandable, as their expertise often lie in other fields of environmental assessment.

Shortcomings in knowledge about environmental noise my also lead to environmental authorities not knowing what to expect from a company doing noise measurements. While the Helsinki environmental authority typically sets requirements for the competence of measurement staff (accredited measurements), some of the other large cities in Finland do not yet set a similar condition or have only recently started to adopt the practice.

In order to make a well educated decision, it is important to understand event noise immission as a whole, including the frequency spectrum. Measurement results (shown in chapter 5, *figures 4 and 5*) show that different musical performances have some variance in the frequency spectrum. Some acts may be characterised as having a more prominent "low-end" sound than others, and the trend is reinforced across some genres. The commonly used A-weighted limits do not make a significant distinction between acts even when they vary greatly in the prominence low-frequency sounds. Thus, some live acts may be unequally treated when noise limits are set with the A-weighting. This fundamental short-coming of the current limit setting process can easily be overlooked by an unexperienced authority.

International touring crews are oftentimes used to widely varying practices, which may reflect a different approach to environmental noise from outdoor events elsewhere in Europe or the world. Looser limits and little enforcement may also lead to problematic attitudes and downplaying the importance of local rules and potential consequences to organisers. The writers have first-hand experience of the following situations:

- Sound engineer for an internationally touring band insisting on monitoring $L_{\mathrm{Aeq,15min}}$ as opposed to the typical $L_{\mathrm{Aeq,5min}}$. Fundamentally the difference was irrelevant, as the noise level of the entire concert (t=2 h) was to be monitored.
- Sound tech from a foreign PA and stage rental company struggling to believe they should keep the SPL at under 98 dB at FOH and 75 dB at facades (remote access noise monitoring at facades)
- Finnish sound engineers printing large "max SPL at FOH" labels in mixing booths, to ensure that international crews are aware of limits imposed for noise. Finnish production teams often set their own limits at FOH, even if one isn't set in the noise permit. The limit often relates to a $L_{\mathrm{Aeq,4h}} \leq 100$ dB [10], but is usually more stringent at for example $L_{\mathrm{Aeq,5min}} \leq 99$ dB.

On the other hand, responsible sound technicians and production teams must be also acknowledged, whom understand and respect the limitations set by environmental noise, and the possible repercussions of exceeding them. In recent years they represent the majority.

5 Measurement results and experiences from the field

This chapter will present some measurement results from outdoor music events at the Helsinki Olympic stadium and the Suvilahti area, as well as some further experiences from measurements, including personal subjective experiences, as well as feedback heard from bystanders and residents of areas in the vicinity of concert and festival venues.

5.1 Olympic Stadium

Figure 4 shows a trend of a growing difference between measured SPL at FOH and measuring point: every data point is the SPL measured at FOH deducted from the SPL at a measuring point near the Helsinki Olympic stadium. The data is gathered from 29 different outdoor music events between 2007-2023. Measurement points 1-3 are located to the west of the Stadium near the residential buildings, and points 4 and 5 are located to the north of the stadium, effectively in the same direction where the PA is directed. Measurement points are shown in *figure 5*.

The figure displays a clear trend in the sound pressure level across the years becoming smaller in the northern measurement points, where the PA is directed at. Some increase can be seen at measurement points 1 and 2, which are slightly to the side of the PA but nonetheless closer to the PA.

Figure 6 shows a negative trend in measured SPL at FOH from outdoor music events at the Helsinki Olympic stadium between 2007 and 2023.

The given values are all expressed in A-weighted results. It can be seen that over the years the A-weighted sound from the concert has reduced in the direct line of the PA, meaning the music is "kept" where it's wanted, i.e. the audience area. It can also be seen that the A-weighted sound pressure level at FOH has generally reduced over the years. This can also be interpreted to show that the PA systems are becoming more effective, in that the same audience area may also be covered with a smaller sound pressure level at a given point. The noise leak which indicates an increase in emitted noise levels show, however, that noise remains an issue in close proximity to the PA-system; a small gap in the outer wall of the Stadium creates a clear view of the PA-system to the closest residential buildings, where noise levels have increased slightly across the years.

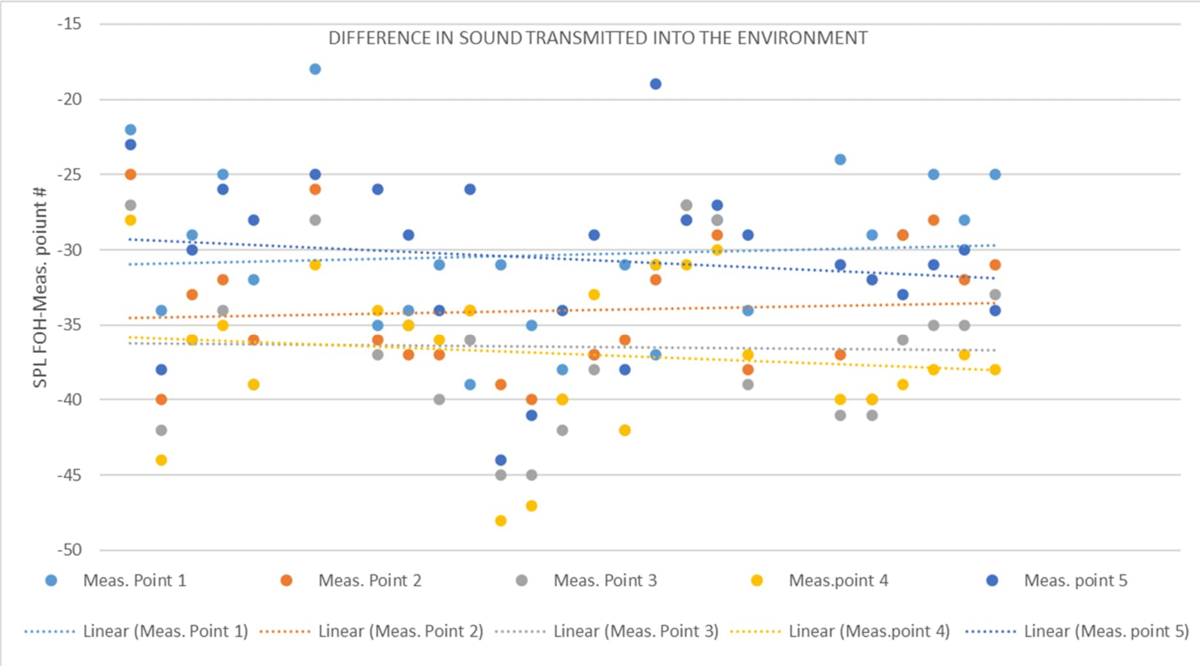

Figure 4. Difference in SPL between FOH and measuring point. Events are in chronological order from left (oldest) to right (newest).

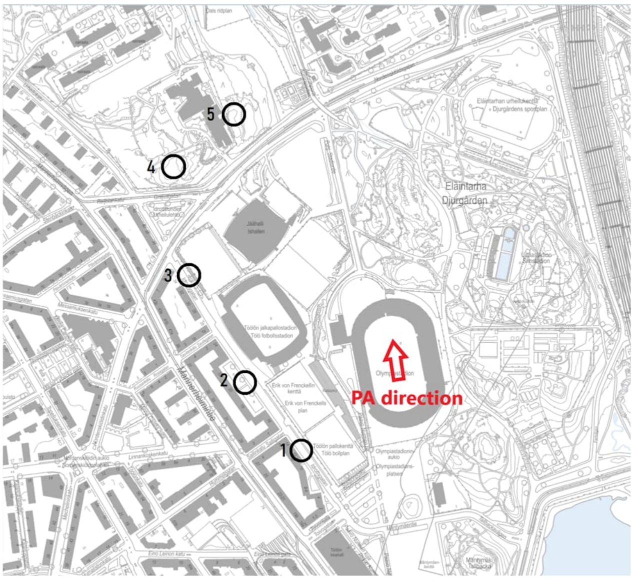

Figure 5. Location of measurement points around the Olympic Stadium.

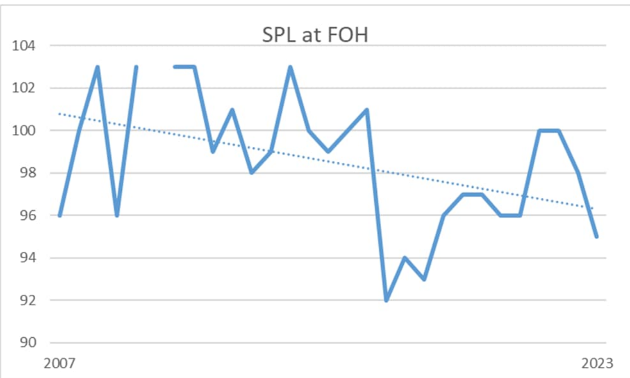

Figure 6. Measured sound pressure levels at FOH at the Helsinki Olympic stadium for 28 events between 2007 and 2023.

5.2 Suvilahti

The previous figures were shown as A-weighted results, as is the norm in Finland.

Figures 7 shows the measured frequency spectrums of two concerts from different festivals in the Suvilahti area. The levels are measured on a façade in the direction of the main stage and have been corrected for a reflection (-3 dB) from the wall behind.

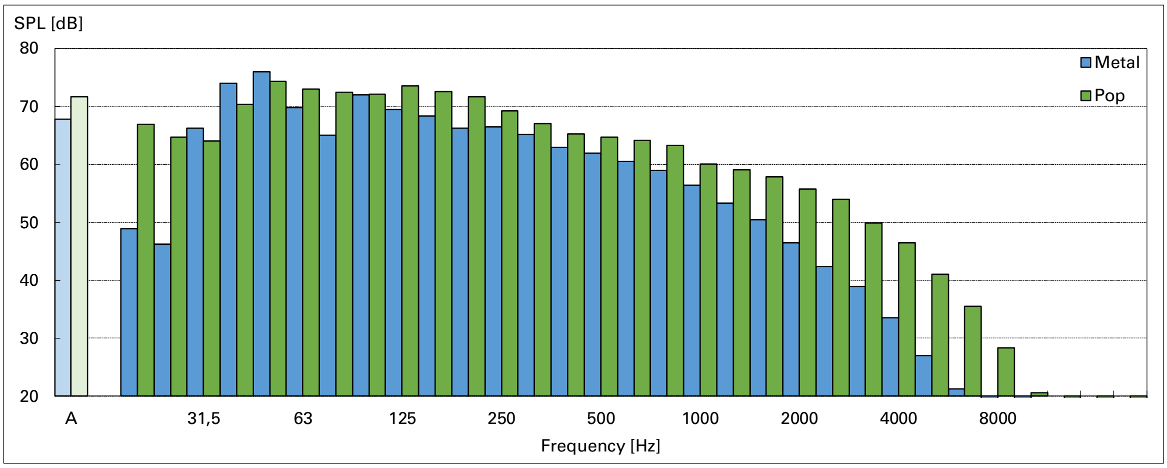

Figure 7. Measured frequency spectrum from a concert at a Metal festival in 2023, from 22.00 until 24.00 and a Pop/Alternative festival in 2023 from 22.00 until 01.00.

The frequency spectrum of music genres can vary greatly. As the noise immission levels are in nearly all cases presented and assessed as A-weighted levels, this proves the need for the conversation to extend beyond the A-weighted results. The fact is, that A-weighting does not treat all music genres equally.

In event noise the base frequencies of 50 Hz and 63 Hz are often experienced as the most intrusive ones, as they penetrate facades with relative ease, and room modes may even enhance them. With the case of Metal music the 50 Hz and 63 Hz third octave bands are several decibels higher than they are in Pop music, however the A-weighted result is 3 dB smaller.

One of the major short comings of the current noise limit process is the rigidity of using A-weighted levels only in assessing the noise levels in the environment. However, approaching a different or additional limiting value, such as C-weighting, which would include the lower frequency to a much higher accuracy, should be approached with caution. As has been previously stated, exceeding noise limits may even be considered an act of environmental criminality, so new adjustments to the limits should be approached with care and caution in order to provide fair treatment for event producers of all genres.

5.3 Subjective Experience, Feedback from the Field

In the authors' experience, there are many other attributes to the sound of a musical act, which greatly affect the subjective experience and annoyance in nearby residents. While EDM (Electronic Dance Music) is commonly spoken of as a very controversial genre, surprisingly the percussive prominence by artists such as Ed Sheeran has been cited as a very annoying attribute by residents living near the Helsinki Olympic stadium. Similarly, many metal acts have been well accepted by residents living near the Helsinki Olympic Stadium and Suvilahti area, while some acts stand out due to percussive prominence in their sound. This experience seems to correlate with acts that emulate a "power metal sound" as well as prominent bass guitar and drum coordination.

6 Conclusions

The Finnish Environmental Protection Law lays out a loosely defined process for assessing and controlling temporary, notable noise immission levels in the environment. Large outdoor events are considered as such. The process and the resulting decisions made from it are suspect to the expertise and experience of the individuals involved in the decision making – or lack thereof. There is no nationwide process of limit that should be adhered to; whereas this enables the decisions to be made case-by-case, it also leaves some smaller municipalities to struggle with decision making regarding event noise.

The process of the handling of notifications of noise should be taken more seriously and a more unified approach across the nation should be considered. The process should not be too rigid as to not take into account the nuances between locations and the types of events.

Measurement data displays evidence, that noise immission levels are due to change in accordance with the technological advancements; sound levels of the concert are now better aimed at the audience, and the noise immission levels are showing a decrease outside the near field of the PA.

The process and noise limits adhere to A-weighted levels solely, which can be shown not to treat all genres equally. Music genres with more elements of base may result in lower A-weighted levels, but still be considered more adverse by the habitants in the area. However, any changes in the decision-making process and the resulting limitation should be considered carefully and with sufficient data, in order to allow for fair treatment to all genres of event noise.

References

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[3] Rammstein 2023, HEL 2022-010410 T 11 01 00 02

[4] Jari Sillanpää 2015, HEL 2015-004432 T 11 01 00 02

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Field tests of low noise levels from MVHR ventilation systems – Overview obstacles and pilot test of test procedure improvement

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The Danish building regulations specify $L_{Aeq} \leq 30$ dB for ventilation noise in dwellings, both for new-build and new systems in renovated housing. The test method referred to is based on the survey method ISO 10052. In practice, many people get annoyed or disturbed by ventilation noise during night or when having quiet activities. Several people complain about the noise, and some people change the operation of the MVHR system or even destroy it. Due to dissatisfaction from occupants, it is considered reducing the regulatory limit to $L_{Aeq} \leq 25$ dB in living and sleeping rooms, which is currently being tested in a voluntary sustainability class (in Danish abbreviated FBK), including use of the engineering method ISO 16032 for background noise correction as prescribed in the FBK guidelines.
However, even with a background noise correction specified, for both limits $L_{Aeq} \leq 30$ dB and $L_{Aeq} \leq 25$ dB, it's often a challenge to quantify the background noise from various activities or traffic noise in the rooms to be tested. For that main reason, amended test procedures compensating for the shortcomings of the basic test procedures are needed. One potential tool might be increasing temporarily the airflow during test, implying much higher noise levels and thus less sensitivity to background noise, and then subsequently adjust the noise level to fit the nominal air flow, before reporting.
The paper describes test results from a preliminary field test with increased air flow, summarizes conclusions and indicates suggestions for a wider, future field study.

1 Introduction

During the latest decade, sustainability has got an increased focus and high ranking at the national agenda in most countries in Europe. In Denmark, the National Strategy for Sustainable Construction was published in 2021 [1], and one of the tools was a voluntary sustainability class, abbreviated FBK, published in 2020 [2]. FBK has nine topics, one of them being FBK8 about ventilation noise in living and sleeping rooms in dwellings. The reasons for including FBK8 was an increased focus on energy savings combined with a higher awareness on indoor air quality, and that MVHR ventilation systems have been installed in a high number of dwellings, but unfortunately with many complaints about ventilation noise.

The Danish building regulations are found in [3] with acoustic requirements in Ch. 17 and acoustic limit values in the related guideline [4]. For ventilation noise, the current limit is $L_{Aeq} \leq 30$ dB in dwellings, and the FBK8 limit is $L_{Aeq} \leq 25$ dB.

This paper focuses on field measurement of ventilation noise and especially on test of low noise levels.
The paper can be considered as a follow-up of conference papers from InterNoise2022 [5] and FA2023 [6].

2 Ventilation noise limits and test methods in building codes

Service equipment noise limits have existed for decades in building codes in many European countries. An overview of current limits for selected countries in Europe is found in Section 2.1. Service equipment consists of many different sources, e.g. heating, water supply, waste water, toilets, lifts and ventilation systems. Since this paper is initiated due to discussions in Denmark about field tests of ventilation noise, especially for cases with limits below regulation, Section 2.2 specifies details about ventilation noise limits in Denmark according to Building Regulations and to FBK8.

2.1 Ventilation noise limits in selected countries in Europe

Limit values for service equipment noise in selected countries in Europe are found in Table 1. References to measurement methods are found in the building regulations or in the guidelines referred to in the regulations or in other national guidelines. The ISO standards referred to for ventilation noise are typically ISO 10052 [7] and/or ISO 16032 [8], [9]. In several countries additional methods apply for low-frequency noise and correction for pure tones, impulses and intermittent noise. Some countries apply different limits and procedures for continuous sources, e.g. ventilation systems, and other sources with changing noise emission during the operating cycle. For reverberation time measurements, the method [10] might be used or, if ISO 10052 [7] is applied, according to the tables in [7]. It should be noted that the ISO standards are implemented as EN standards and subsequently as national standards in CEN member countries.

Table 1: Overview service equipment noise limits for dwellings (habitable rooms) in selected countries in Europe. Table copied from InterNoise2022 paper [5], Table 2. Note: Reference numbers in the table refer to the list of references in [5].

Acoustic regulations for HOUSING (1) – Service equipment noise – April 2022					
Country	**BR**	**Test method**	**Require-ment (2) [dB]**	**Fur-nished**	**Comments**
Denmark	[12]	ISO 10052 and DK guideline [29]	$L_{A,eq} \le 30$	–	If measured in a furnished room, +3 dB is added to the measured value. BR [12] refers to Class C in ACS [22] as the requirement for habitable rooms.
England	[13], [14]	National procedure and Guideline [31]	($L_{Aeq,T} \le 30$) ($L_{Aeq,T} \le 45$)	Not specified	Levels for living rooms and bedrooms, for ventilation systems. Levels for kitchens and bathrooms. Limits are recommendations just for ventilation noise.
France	[15]	ISO 10052 and FR guideline [32]	$L_{nAT} \le 35$ $L_{nAT} \le 30$ $L_{nAT} \le 30$ $L_{nAT} \le 30$	Not specified	Noise produced by individual heating or cooling systems. Noise produced by mechanical ventilation systems. Noise produced by other equipment belonging to another dwelling. Noise produced by collective building equipment, such as lifts, water pumps, boilers, etc. $L_{nAT} = L_{ASmax,nT}$. standardized to the reference reverberation time (average of the RT values for 500 Hz, 1000 Hz et 2000 Hz). Note: Noise limits for living rooms and bedrooms. Limit for kitchens specified in [15].
Italy (public)	[16], [17]	ISO 10052 or ISO 16032 and national procedure [17]	$L_{ic} \le 28$ $L_{id} \le 33$	+	Equivalent SPL from service equipment with continuous operation. Maximum SPL from service equipment with discontinuous operation. Note: The descriptors are explained in [23].
Italy (private)	[16]	ISO 10052 or ISO 16032	$L_{Aeq} \le 25$ $L_{ASmax} \le 35$	Not specified	Equivalent SPL from service equipment with continuous operation. Maximum SPL from service equipment with discontinuous operation.
Norway	[18]	ISO 16032	$L_{p,A,T} \le 30$ $L_{p,AF,max} \le 32$	+	BR [18] refers to Class C in ACS [24]. 5 dB higher sound levels are ok in kitchens, WC, bathrooms, entrances etc. In addition to the limits indicated, there are also LF-limits for octaves 31,5-125 Hz.
Portugal	[19], [33]	National procedure [33]	$L_{Ar,nT} \le 27$ $L_{Ar,nT} \le 32$ $L_{Ar,nT} \le 40$	Not specified	For building services producing a continuous noise. For building services that works intermittently For an emergency power unit. $L_{Ar,nT} = L_{A,eq}$ + corrections for background noise, tonal noise..
Spain	[20], [34]	National procedure [34]	$L_{k,n} \le 25$ $L_{k,n} \le 30$	Not specified	For bedrooms in dwellings. Limits for the night period, 23:00 – 7:00h. For the day and evening periods, limits are: $L_{k,d}$; $L_{k,e} \le 35$ For living rooms in dwellings. Limits for the night period, 23:00 – 7:00h. For the day and evening periods, limits are: $L_{k,d}$; $L_{k,e} \le 40$ Limit value $L_k = L_{A,eq,T}$ + corrections for background noise, tonal, impulsive and LF noise
Turkey	[21]	ISO 10052 or ISO 16032	$L_{A,eq,nT} \le 30$ $L_{A,eq,nT} \le 35$	+	Limit for continuous noise in bedrooms during night-time, 23:00 – 07:00. Limit for continuous noise in living areas, kitchen - during 24 hours.
			$L_{AF,max,nT} \le 34$		Limit for intermittent noise.
(1) Overview information only. Detailed criteria and conditions are found in references. (2) Limits in (brackets) = Recommendation.					

Since it is important to get reliable results, the test methods ISO 10052 and ISO 16032 have requirements for instrumentation [11], [12], [13].

2.2 Ventilation noise limits in Denmark & test procedure

In DK, the building code guideline [4] refers to ISO 10052 [7] for measurement of service equipment noise, but for ventilation noise (noise source in the room, where the measurement is made) is prescribed measurement in just one microphone position for each source, cf. [14]. The microphone is placed as indicated in Figure 1, implying that noise from the ventilation system is dominant – and noise from other parts of the room less dominant.

The Danish classification standard [15] for dwellings has six classes A-F for ventilation noise with limits starting with 20 dB in Class A and 5 dB steps between classes up to Class E. Class F has no limit. Minimum Class C with limit LAeq ≤ 30 dB is required by the Danish regulations. However, since many people are disturbed by ventilation noise during sleep and quiet activities, even if the building code requirement 30 dB is fulfilled, it has been considered to make the ventilation noise limit 5 dB stricter to get quieter living rooms and bedrooms and thus create a healthier indoor climate.

As a starting point, a voluntary sustainability class for ventilation noise in dwellings has been introduced in [2] with an upper limit 25 dB, i.e. 5 dB stricter than regulations. The purpose is to test the feasibility of measuring and implementing such 5 dB quieter ventilation systems and then later decide to make the limit mandatory if practice supports such step.

Since correction for background noise is not included in the ISO 10052 procedure [7], it is often difficult or impossible to measure low levels of ventilation noise, even if the microphone position is quite close to the source and does not include higher noise levels more far away from the source. In DK, it is considered either to switch to ISO 16032 [8], [9] (allowing background noise correction) or update the national guideline [14], so correction for background noise can be made, which the authors have previously tested, cf. [6]. However, it would be more optimal to revise the ISO standards aiming at optimizing the procedure for ventilation noise measurements, including also specific microphone positions for such tests.

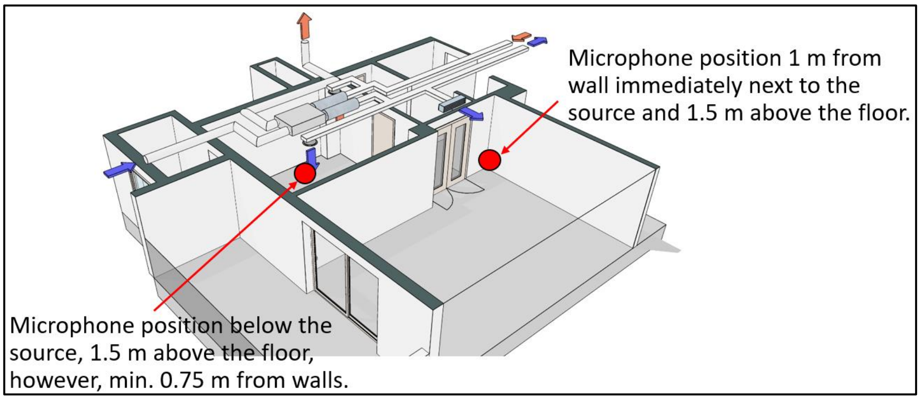

Figure 1: Microphone positions for measurement of ventilation noise according to the national guideline SBi 217 [14] in DK. Illustration copied from from InterNoise2022 paper [5] Fig. 2.

3 Overview obstacles for field tests of low levels of ventilation noise

According to consultants' experiences from field measurements of ventilation noise, the two major obstacles are about too high or disturbing background noise and about adjustment of operating conditions to be well-defined and corresponding to the building code requirements for ventilation. The two test methods, ISO 10052 and ISO 16032, both have clauses (8 and 10, respectively) specifying the required contents of test reports. Both methods request information about operating conditions and background noise (in ISO 10052 just information, since correction is not done) – in addition to the usual requested contents of field test reports about test object, rooms, address, operator etc. Field tests of ventilation noise have become more complicated with automated control of the ventilation systems because description and adjustment of the operating conditions might require a ventilation expert, who may need to be present during tests.

Several other issues are important as reflected in the requirements for the test reports. Correct measurements of ventilation noise have several pitfalls, which less experienced operators may not be aware of, e.g.:

- Does the instrumentation meet the IEC criteria included in the normative references??
- Instrumentation capable of measuring low SPL and all relevant frequency bands with sufficient accuracy?
- Expertise of operator (test method, ventilation systems)
- Calibration of test instruments.
- Quality of test reports.

The challenges are relevant for test of ventilation noise for systems meeting the current noise limit – DS 490 Class C – but become even more important for lower noise limits $L_{Aeq} \leq 25$ dB like in FBK8. The FBK8 limit is the same as in DS 490 [15] Class B. The challenges do also appear in other countries, and acoustic classes for ventilation noise are also included in the international specification ISO/TS 19488 [16], which has ventilation noise class limits similar to DS 490.

4 Pilot test procedure for field test of ventilation noise

Test procedures compensating for the shortcomings of the basic test procedures for measurements of low sound pressure levels are called for. One potential procedure might be to increase temporarily the airflow during the test, implying much higher nose levels and thus less sensitivity to background noise, and then subsequently adjust the noise level to fit the nominal air flow, before reporting. Below is described a potential new procedure, primarily supposed to be applied in combination with ISO 10052 [7] and SBi 217 [14]. The procedure is ideal for a measurement system that enables measurements in two microphone positions simultaneously, but the method can also be used serially if the noise is stationary. In that case, it is important to precisely mark the microphone positions so these can easily be found again. For example, two microphone stands can be used, cf. Figure 2. The method requires the possibility of adjusting the ventilation setting to increased air flow that provides a substantially higher SPL in the microphone positions. In the following, the procedure is referred to as Pilot Test Procedure (PTP).

- **Step 1:** The measuring point defined in SBi 217 is marked so it can be found again – hereinafter referred to as MP1.
- **Step 2:** A new measuring point is marked approx. 25 cm from the ventilation exhaust opening in the direction towards MP1. This new point is referred to as MP2 hereafter.
- **Step 3:** Adjust the ventilation system to operate in the setting with maximum air flow.
- **Step 4:** Perform a measurement in both microphone positions MP1 and MP2 and find the arithmetic difference between the two spectra.
- **Step 5:** Perform a measurement in MP2 with the ventilation system being adjusted to a setting that fulfils the ventilation requirements in the national regulations, in Denmark building regulations [3], Ch. 2 Ventilation.
- **Step 6:** The calculated difference in step 4 is subtracted from the noise level measured in step 5 which yields the new measurement result.

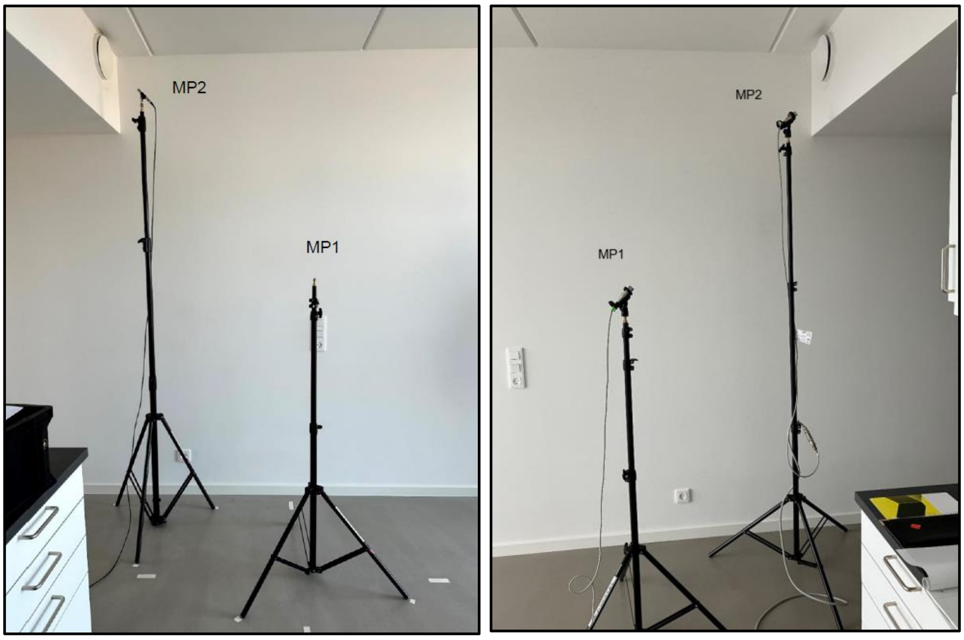

Figure2: Microphone positions for pilot test 1 (25/1-2024) and 2 (21/3-2024).

5 Preliminary pilot study of field test procedure with increased air flow

In a pilot study, ventilation noise tests were conducted according to the procedure described in Section 4 with measurements in two apartments over two days (25/1-2024 and (21/3-2024). The apartments were in a new multi-storey housing (Akademivej 1E, 2800 Kongens Lyngby) known by KAB (building association). The upper limit for ventilation noise was 30 dB as required in new-build (and not the lower FBK8 limit, since that was not the goal). In the first pilot test the noise level was measured with 3 different ventilation settings having different air flow rates. Unfortunately, no background noise could be measured due to frequent external noise disturbances from various construction activities and a time limited access to the apartment. Pilot test 1 was performed with a one-microphone system using a B&K type 4189 ½" transducer (only one microphone was used due to technical issues with cables and connections). In the second pilot test the noise level was measured with 2 different ventilation settings having different airflow rates, and background noise was also measured. Pilot test 2 was performed with a two-microphone system using B&K type 4144 1" transducers. Both pilot tests were performed in the same building in apartments with identical layout (mirrored) and Dantherm HCC 360P2 ventilation units. All measurements were performed with a B&K type 2270 sound level meter and calibration made with B&K Type 4231 calibrator. No reverberation time measurements were performed in either of the pilot tests since the rooms were unfurnished corresponding to the condition for the Danish limit value. The setting with the highest air flow (and highest noise level) is called Setting 4.

Information about different ventilation settings is presented in Table 2. The results of the measurements are shown in Figure 3, Figure 4 and Table 3.

In Figure 3, the directly measured results are shown. As can be seen, for pilot test 2 there were challenges with background noise during the measurements, both as a general high level of background noise and as interrupting short term background noise. For both pilot tests the difference between Setting 4 and Setting 2 is generally between 10-30 considering 1/3 frequency bands and more than > 25 dB for the total level. For pilot test 1 there is basically no difference between the noise level for Setting 2 and 3 for frequencies higher than 800 Hz, and it is assumed that for these frequencies what is measured is the background noise level. This hypothesis is supported by pilot test 2 where the background noise level is identical with the noise level measured in Setting 2 for frequencies above 630 Hz.

Table 2: Ventilation settings of Dantherm HCC 360P2 unit.

Ventilation setting (VS)	Description	Other info
VS1	Lowest airflow rate	Noise levels were not measured
VS2	Automatic setting	
VS3	Medium airflow rate	
VS4	Maximum airflow rate	

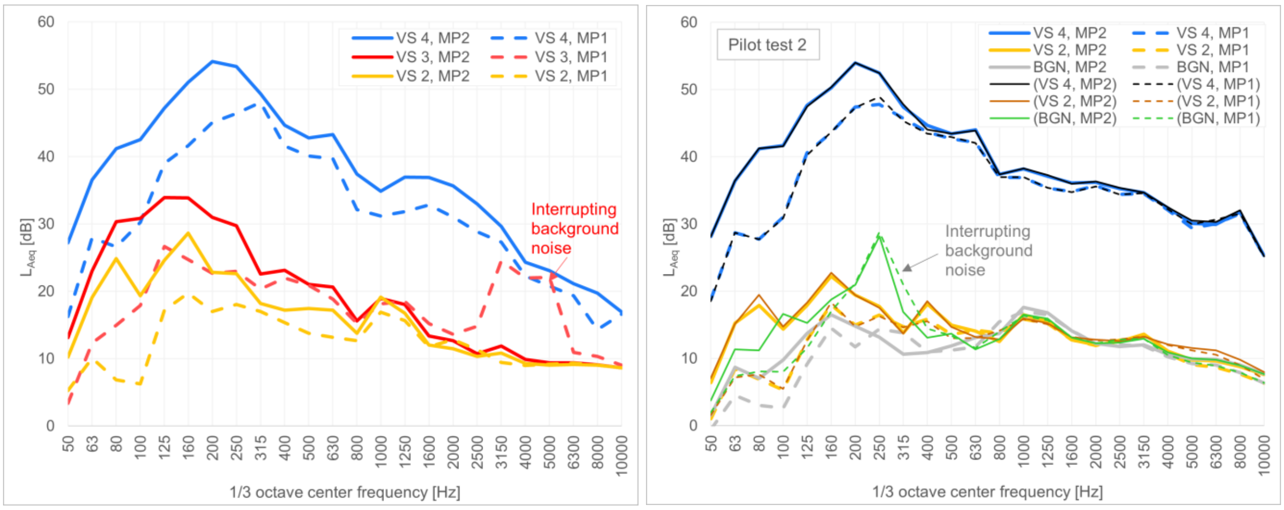

Figure 3: Measured background (BGN) and ventilation noise levels for different ventilation settings (VS) for pilot test 1 (on the left) and 2 (on the right) in microphone positions MP1 and MP2. Brackets indicate repeated measurements.

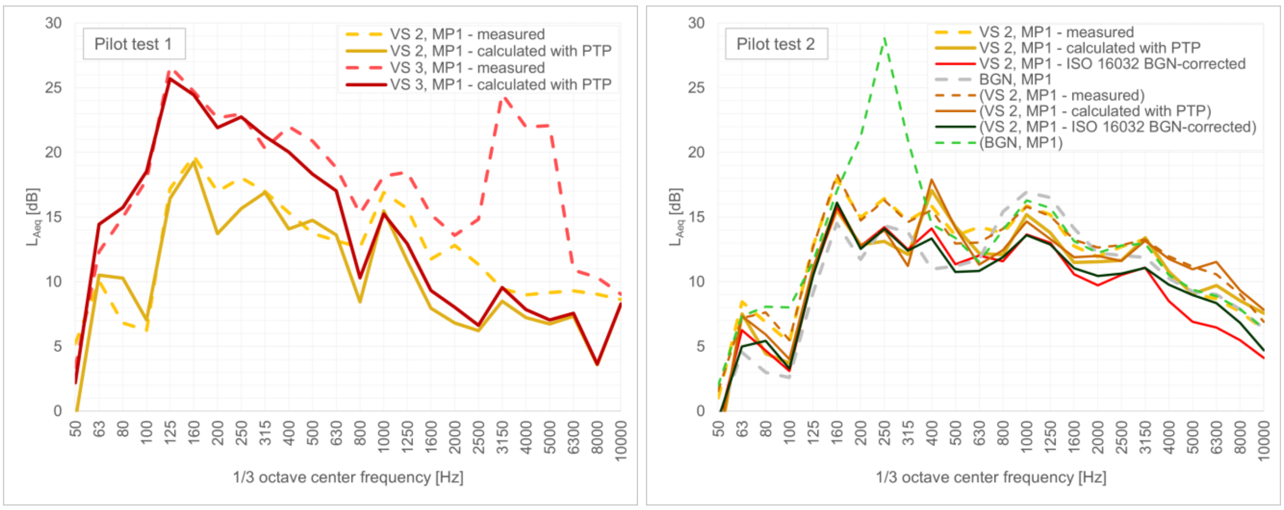

Figure 4: Ventilation noise levels in ventilation setting “VS 2” in microphone position MP1 – measurements and calculations according to the pilot test procedure (PTP) for pilot test 1 (on the left) and 2 (on the right). Brackets indicate repeated measurements.

Figure 4 shows both the direct measurements and the calculated results according to the pilot test procedure, all for MP1. The left side of the figure shows the results for pilot test 1, where a very good correlation between direct measurement and calculated result with the pilot test procedure is seen below 400 Hz for the Setting 3, which is probably due to low or no influence from background noise. A decent correlation between direct measurement and calculated result for Setting 2 is seen below 800 Hz, which is probably due to some influence from background noise. Above 630 Hz there is not much correlation between direct measurement and calculated result, but this is also the frequency area where it is expected that what is measured is primarily background noise.

The right side of the figure shows the results for pilot test 2, where for this measurement there is not much difference between background noise and measured noise, and where the measured noise for frequencies below 800 Hz is lower for pilot test 2 than for pilot test 1. Included on this figure is also the ISO 16032 background noise corrected level, where in general the ISO 16032 background noise corrected level is lower than the calculated level with the pilot test procedure.

In Table 3 the total ventilation noise levels for the two case studies are listed, both as direct measurements and calculated with the pilot test procedure (PTP), either based on 1/3 octave values or single number values. The results corrected for background noise according to ISO 16032 are shown as well. When comparing the calculated results with the pilot test method based on 1/3 octave band or based on single number, a difference between -1.1 dB to +2 dB is seen.

From the table, the following can be concluded:

- The calculated values are always lower than the measured values.
- The results corrected for background noise in 1/3 octave bands according to ISO 16032 are on the same level or slightly lower than the results calculated with the pilot test procedure.
- The variation between the calculated results with the pilot test procedure based on 1/3 octave bands and single numbers is too large, and the pilot test procedure based on 1/3 octave bands should be the preferred one.

Table 3: Total ventilation noise for frequency range 50 to 10 000 Hz in microphone position MP1 – measured, calculated according to the PTP method in 1/3 octave bands, calculated according to the PTP method in single number values. The results to be compared with limits are those in the column with heading Setting 2.

Ventilation settings in columns to the right **Description of tests below:**	Setting 4, L_{Aeq} [dB]	Setting 3, L_{Aeq} [dB]	**Setting 2, L_{Aeq} [dB]**	BGN noise L_{Aeq} [dB]
Pilot test 1 (25/1-2024)				
Measured	53.2	34.2	**27.9**	
Calculated (PTP – 1/3 octave)		32.0	**26.5**	
Calculated (PTP – single-no.)		34.0	**27.5**	
Pilot test 2 (21/3-2024)				
Measured	54.4		**27.0**	25.9
Calculated (PTP – 1/3 octave)			**25.9**	
Calculated (PTP – single-no.)			**24.8**	
Calculated (BGN-corr. 1/3 oct. ISO 16032 method)			**24.8**	
Pilot test 2 repeated – See brackets below:				
(Measured)	54.6		**27.1**	31.4
(Calculated (PTP – 1/3 octave))			**26.2**	
(Calculated (PTP – single-no.))			**25.3**	
(Calculated (BGN-corr. 1/3 oct. ISO 16032 method))			**24.9**	

The results from tests with the suggested procedure seem promising, but further studies are necessary since only two pilot tests were possible in the pilot study. More investigations need to be done, including tests of other types of ventilation systems.

6 Summary, conclusions and recommendations

A pilot study consisting of ventilation noise tests in two new apartments have been carried out to evaluate the feasibility of a suggested new procedure for measuring low noise levels from ventilation systems. The results from the pilot tests with the suggested procedure seem promising, but further studies are necessary since there were just two pilot tests. More investigations need to be done, including tests of other types of ventilation systems. The suggested procedure applies an additional microphone position and the ventilation system adjusted to higher air flow levels, thus creating considerably higher noise levels, and a subsequent adjustment to the normal air flow is made to get the ventilation noise level to be compared with the limit value in the regulations or even lower limits, aiming at establishing less noisy living rooms.

According to consultants' experiences from field measurements of ventilation noise, the two major obstacles are about background noise and about operating conditions to be well-defined and corresponding to the building code requirements for ventilation. Background noise correction is necessary for measurement of ventilation noise, especially for low levels.

Improved/innovative test procedures are needed for measurements of ventilation noise, especially for low noise levels limits ~ 25 dB, but also for higher limits like e.g. 30 dB. In the pilot study was tested one new procedure with increased air flow, which resulted in a ventilation noise level increased considerably, > 25 dB (broadband).

In the future, wider research studies could also include issues like instrumentation and applicability of both simple and complex objective methods for adjustments for tonality, impulsivity and low frequency contents.

For more discussion of ventilation noise tests according to the ISO methods, see results from a previous case study in [6], which also includes reverberation time measurements.

Other issues that could become relevant in practice are applicability and shortcomings of noise measurements with smartphones, which some people consider acceptable for the field tests – probably because they are not aware of that requirements for procedures and instrumentation in the ISO and IEC standards are not fulfilled.

Furthermore, it could be useful to make recordings of ventilation noise stimuli for demo purposes and future laboratory listening tests.

Acknowledgements

We appreciate the discussion about measuring low noise levels initiated by the Danish Authority of Social Services and Housing, when introducing the test period for FBK. FORCE Technology wishes to thank the Danish Agency for Institutions and Educational Grants for financial support of the case study measurements. The authors thank Birger Bech Jessen for suggesting and discussing a potential new test procedure for measuring low noise levels from ventilation systems. The authors also want to thank Magnus Jørck, FORCE Technology, for assisting with measurements in pilot test 2. The authors are especially grateful to Josef Wahdan, Energy and heating specialist, KAB, for help finding test sites for the preliminary pilot tests and for being present during both tests and assist with control of the ventilation systems for the measurements in the pilot study. Without Josef's help, we would not have been able to make the tests!

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Acoustics in Green Buildings

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The past two decades have witnessed a dramatic rise in the number of green buildings in the construction industry; green buildings are certified through rating schemes, such as BREEAM and LEED, which consider acoustics part of Indoor Environmental Quality (IEQ). Although some studies have examined IEQ, there has been no research conducted in this field in Finland.

This study evaluates the acoustics comfort of green office buildings by conducting a Post Occupancy Evaluation (POE) survey for the acoustics comfort on seven office green buildings and two office conventional buildings in Finland. Based on a literature review of earlier research, an online and paper-form survey was distributed to the users of green buildings and conventional buildings. In addition, acoustics measurements were collected from the investigated buildings. An analysis of the responses demonstrated that users' satisfaction with acoustics comfort fluctuated within building categories regardless of the building being either green or conventional and it acquiring or missing acoustics credits. The results correspond to previous studies' claims that acoustics comfort in green and conventional buildings is approximately the same.

The survey indicates that personal and contextual variables influence acoustics comfort inside buildings (such as age, gender, building tenure, work desk conditions, and visual privacy); specifically, gender and visual privacy significantly impact acoustics comfort. Another finding is that acoustics measurements do not correlate with the survey results. This demonstrates that there are other aspects (such as personal and contextual variables) to consider for acoustics performance together with specified acoustics parameters by authorities or organizations.

1 Introduction

This paper is based on the thesis study "Acoustics in green buildings" that was done in 2022 in the school of electrical engineering at Aalto University [1].

Over the last twenty years, there has been a remarkable surge in the construction industry's adoption of green buildings. This has been due to public awareness of adopting sustainability and sustainable practices for various reasons, such as global warming, immigration and the resulting rapid growth of population, as well as eliminating environmental impacts [2, 3]. Additionally, several studies have proposed that occupants of green buildings have a higher level of productivity and satisfaction than those of conventional buildings [4]. The construction industry is considered a major energy consumer contributing approximately 40% of the global total energy consumption, producing high carbon emissions, greenhouse gases, and massive waste. Embracing sustainable practices will reduce these effects [1, 3].Green buildings are certified using green buildings rating schemes, such as Leadership in Energy and Environmental Design (LEED) and Building Research Establishment's Environmental Assessment Method (BREEAM), which can be defined as tools to evaluate and encourage sustainable development. They also provide guidance and information on achieving sustainability [5].

Indoor Environmental Quality (IEQ) is one of the essential criteria of the green buildings rating schemes, IEQ includes indoor air quality, lighting and visual comfort, thermal comfort, acoustics, functionality, and aesthetics [3]. Recently, considerable research has addressed the IEQ of green buildings, focusing on green buildings in the US and Asia. However, few studies have focused on the acoustics performance of these buildings. Despite Finland being a global pioneer in sustainability and green buildings [6], no studies have addressed the IEQ of green buildings in Finland. Therefore, this study aimed to evaluate the acoustics performance of green buildings in Finland and compare their acoustics performance with conventional buildings in Finland. Additionally, this study investigated the impact of personal and contextual variables on acoustics comfort as proposed in previous research [7]. It is accomplished by conducting Post Occupancy Evaluation (POE) survey of acoustics comfort of green and conventional buildings in Finland, as well as collecting acoustical measurements. The POE survey of IEQ is restricted to the acoustics performance of the buildings alone, omitting other IEQ aspects, such as lighting, temperature, and others.

2 Post Occupancy Evaluation survey

Green buildings aim to seize building impacts on occupants' health and that is evaluated using IEQ criteria. IEQ has been investigated for several decades using POE surveys. Office buildings were specifically investigated more than other buildings as they highly influence employee production [7].

There is a debate about green buildings, as to whether they provide a high level of wellbeing and satisfaction for their occupants. POE surveys conducted by Montazami [8] showed that BREEAM green buildings have better acoustics comfort, especially when considering acoustics in the early design stage. Conversely, Altomonte survey [9] showed a lower level of acoustics comfort in BREEAM green buildings compared with conventional buildings. Additionally, other POE surveys [9, 10, 11, 12, 13] reported that acoustics comfort is not improved or barely improved in green buildings. Newsham [14] has supported the conclusion that acoustics comfort in conventional and green buildings are approximately the same by conducting a POE survey supported with on-site measurements for IEQ categories in LEED green buildings and conventional buildings.

However, one study focusing on IEQ perception and acoustics comfort suggested that acoustic comfort in office spaces is influenced by acoustics parameters (such as sound insulation, noise levels, speech intelligibility, etc.) as well as many other variables, such as personal variables and contextual variables [7]. Contextual variables include building automation, office layout, workstation location, privacy, tasks, and duration of working hours. Personal variables include occupant age, gender, social conditions, lifestyle, worker position, social status, etc.

3 POE survey in Finland

A POE survey [1] regarding the acoustics comfort of office users has been performed in selected green-certified office buildings (BREEAM and LEED certification) and conventional office buildings in Finland. After studying green buildings and the acoustics criteria in LEED and BREEAM buildings, the scope of this study was to investigate acoustics comfort in five different buildings categories:

- BREEAM-certified buildings that have acoustics credits (BREEAM-AKU)
- BREEAM-certified buildings that have no acoustics credits (BREEAM-NO)
- LEED-certified buildings that have acoustics credits (LEED-AKU)
- LEED-certified buildings that have no acoustics credits (LEED-NO)
- Conventional Buildings, buildings with no green certification (CB)

Two buildings of each category have been investigated. Except for LEED-NO; only one building has been investigated.

The survey had 18 questions, including multiple choice questions and one open-ended question. The questions about acoustics comfort are derived from previous research [14, 15], in addition to the input from the researcher. Some questions were included in the survey to examine personal and contextual variables' influence on acoustics comfort, as Fasano proposed [7]. The rating scale for the acoustics comfort questions was based on their level of satisfaction,

The survey was created using Webropol in English and Finnish. An online link and paper-form of the survey has been published to the building users. The total number of responses to the survey was 297. 70 responses for BREEAM-AKU, 86 for BREEAM-NO, 50 responses for LEED-AKU, 18 responses for LEED-NO, and 73 responses for CB.

Acoustics measurements of the background noise level (L_{Aeq}), reverberation time (T), and sound insulation of each office space have been collected to determine whether the buildings achieve the applicable acoustics criteria. Measurement results were analysed for meeting rooms and open offices. Acoustics measurements results of the meeting rooms and open offices were averaged for each building category. Table 1 shows background noise level, reverberation time, and sound insulation measurement results for each building category.

Table 1: Acoustics measurement results for background noise level, reverberation time, and sound insulation for each building category. The results marked in green fulfil their criteria, the results marked in yellow do not fulfil their criteria.

Building Category	**Background noise level (L_{Aeq}) dB**		**Reverberation Time (T) s**		**Sound insulation dB**	
	Meeting room	Open office	Meeting room	Open office	Between meeting rooms	Between meeting room and corridor
BREEAM-AKU	32	28	0,3	0,8	53	37
BREEAM-NO	35	28	0,6	0,4	55	38
LEED-AKU	28	33	0,7	0,7	44	37
LEED-NO	40	-	0,3	-	42	34
CB	35	-	0,4	-	44	-

4 Analysis

the survey responses have been used to compare acoustics comfort in green and conventional buildings in three different ways. The complete sets of the results are found in the thesis [1].

The first comparison demonstrated the occupants' responses of the acoustics satisfaction for the five main building categories. The first thing to notice from the first analysis is that most of the users of all buildings categories are satisfied with levels of noise from external sources.The results of this comparison revealed that satisfaction level varies between questions within each category of buildings, occupants didn't consistently respond the same way. For example, 71% of the users of BREEAM-NO buildings are satisfied with background noise levels compared to other building categories users (see Figure 1). But when asking about acoustics quality and overall acoustics comfort, the users of BREEAM-NO buildings were not as satisfied as other building categories.

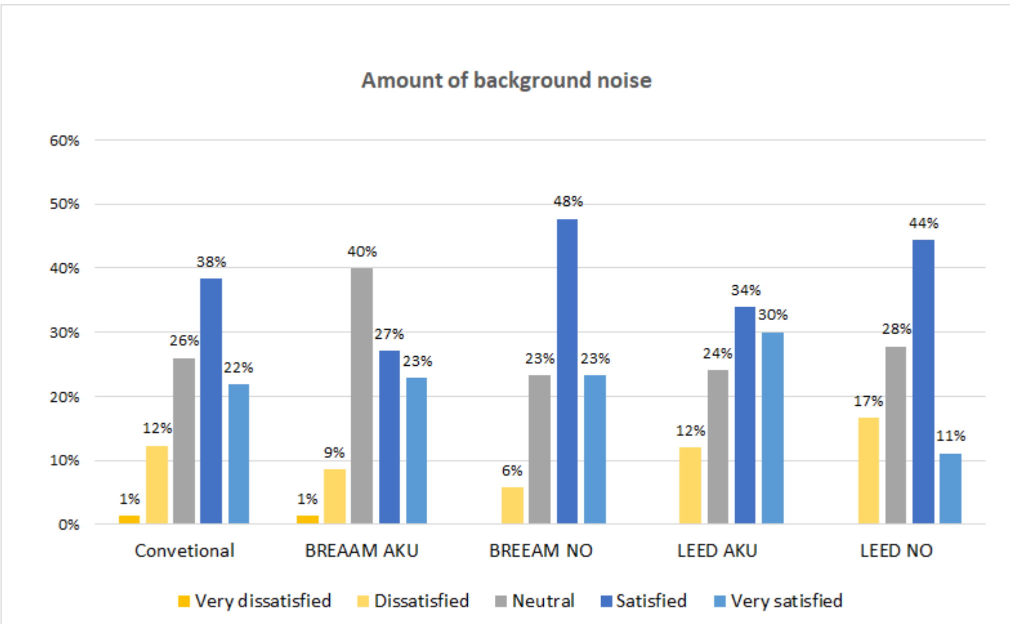

Figure 1: Background noise level satisfaction for building categories (first comparison).

The second comparison categorized buildings in three building types: conventional buildings, green buildings with no acoustics credits (Green-NO), and green buildings with acoustics credits (Green-AKU). Results showed that conventional buildings users' overall satisfaction with speech noise is 40%, which is higher than green buildings, regardless of whether the buildings have acoustic credits (28%) or not (33%). Office building users reported a high level of dissatisfaction with acoustics privacy for all building types: over 66% of all buildings categories are dissatisfied or very dissatisfied, see Figure 2.

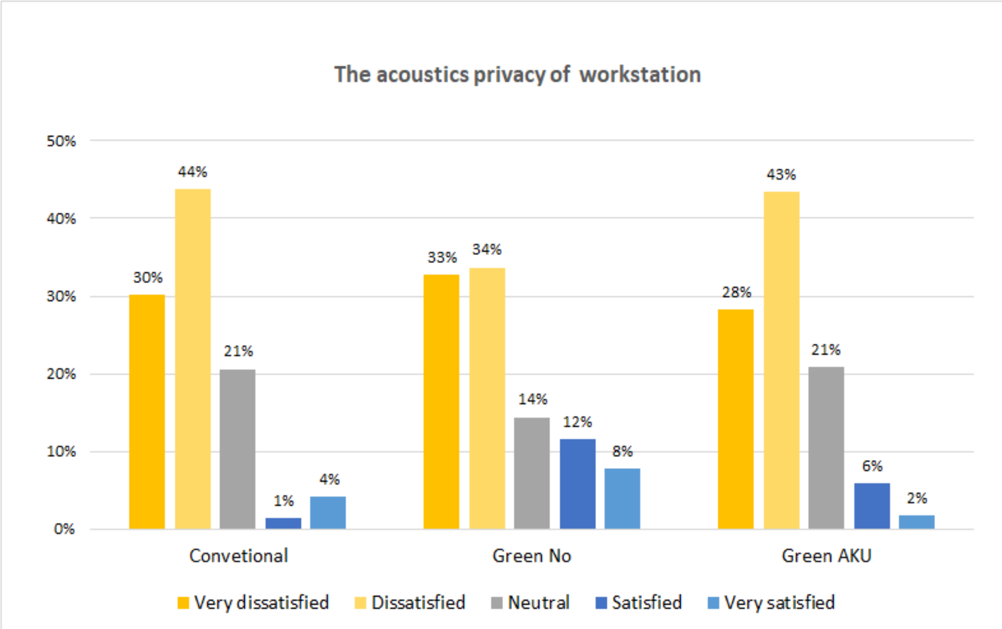

Figure 2: Acoustics privacy according to the second comparison

The third comparison was also between three building categories: conventional buildings, all BREEAM-certified buildings and all LEED-certified buildings. The results of this comparison demonstrated the overall acoustics comfort in Figure 3, users of the conventional buildings are less dissatisfied (22%) than the users of BREEAM (30%) or LEED (36%).

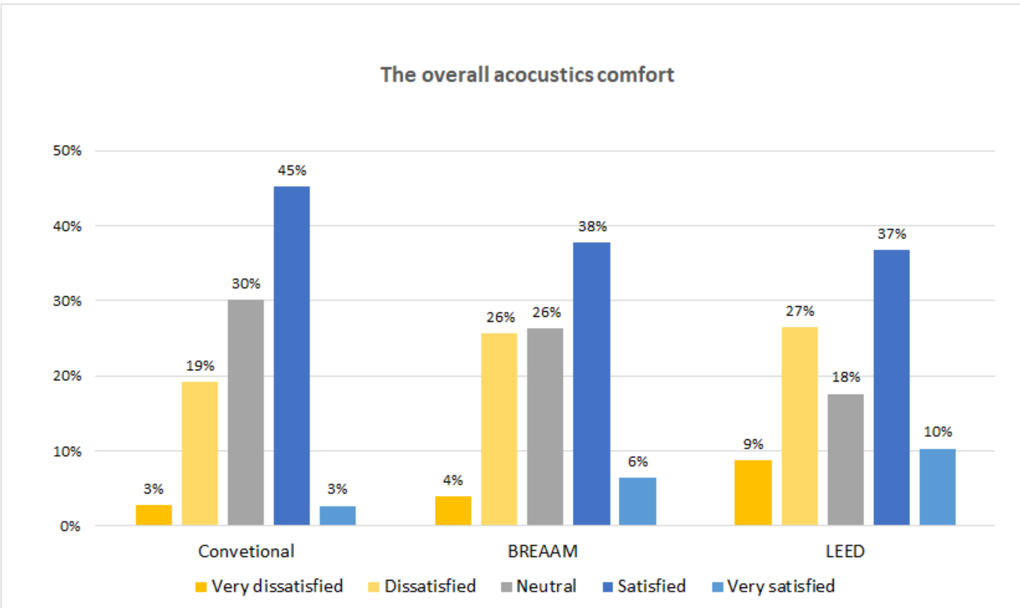

Figure 3: Overall acoustics comfort according to the third comparison; conventional buildings' users are more satisfied than BREEAM and LEED buildings.

The analysis of responses to survey questions about acoustics comfort based on building categories demonstrated that users' satisfaction with acoustics comfort varies among building category, regardless of whether the building is green or conventional and its acquiring of acoustics credits or not. Also, there is no consistency in users' responses within same building category.

The survey included questions about building users' background and contextual variables to investigate Fasano's suggestion [7] that personal and contextual variables have influence on acoustics comfort. This analysis combined responses for all building types and analysed them according to personal and contextual variables. The analysis results found that age somewhat impact how building users perceive acoustics comfort, the youngest group was more satisfied than the other age groups. A novel finding is that gender significantly influences acoustics comfort; male users were more satisfied than female users. For example, 75% of male users are satisfied with background noise and only 53% of female users are satisfied with background noise.

The results of the analysis of contextual variables indicated that in addition to gender, building tenure (period of using the office premises) influences acoustics comfort: users with one to six months of tenure were more satisfied with overall acoustics comfort compared to the users with over 2 years of tenure. Also, contextual variables related to workstation conditions seem to influence the results. The users with designated working desks reported to be more satisfied with acoustics quality than those who lacked designated desks.

The survey results indicates that visual privacy also has a significant impact on acoustics comfort. Responses demonstrated that users who were pleased with their visual privacy were also satisfied with the overall acoustics comfort (Figure 4).

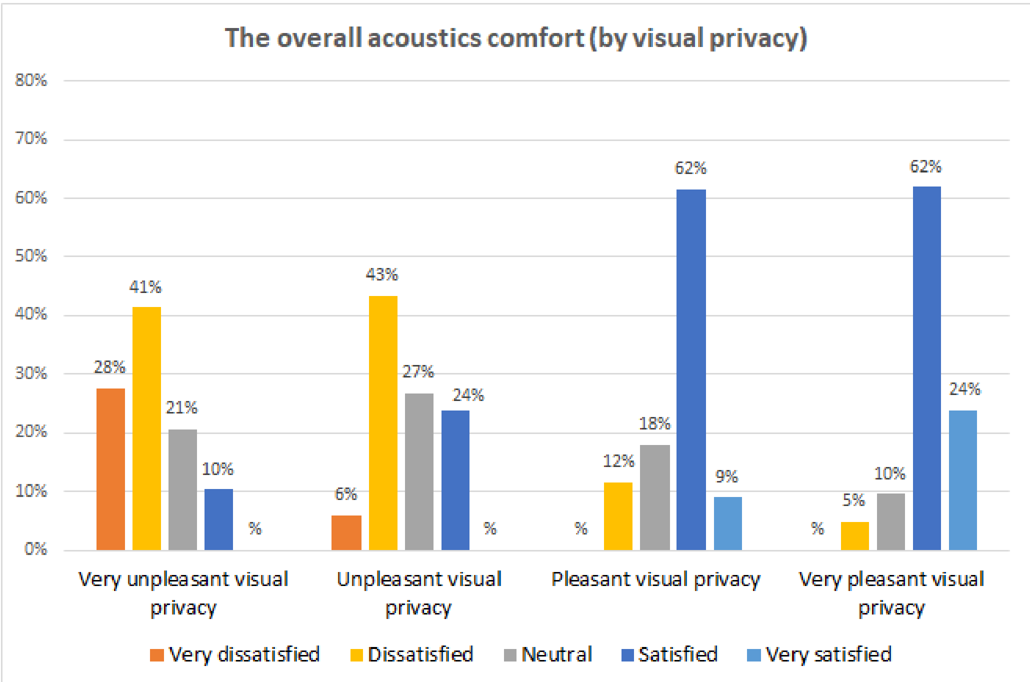

Figure 4: The influence of visual privacy on overall acoustics comfort; high peaks for satisfaction with overall acoustics comfort for the users who are pleased with their visual privacy

The data from this analysis contributes a clearer understanding of contextual and personal variables' impact on acoustics comfort.

The survey also included an open-ended question for further observations, 124 responses out of 297 commented about the acoustics performance, the most reported issues for all building types were related to speech privacy issues followed by room acoustics issues (Figure 5).

The survey considered the recent change in the working culture as a result of the COVID-19 pandemic; thus, the survey included a question about users' acoustics comfort in their remote working place. 66% of the respondents are satisfied with the acoustics comfort in their remote working place, whereas only 45% of respondents are satisfied with the acoustics comfort of their office.

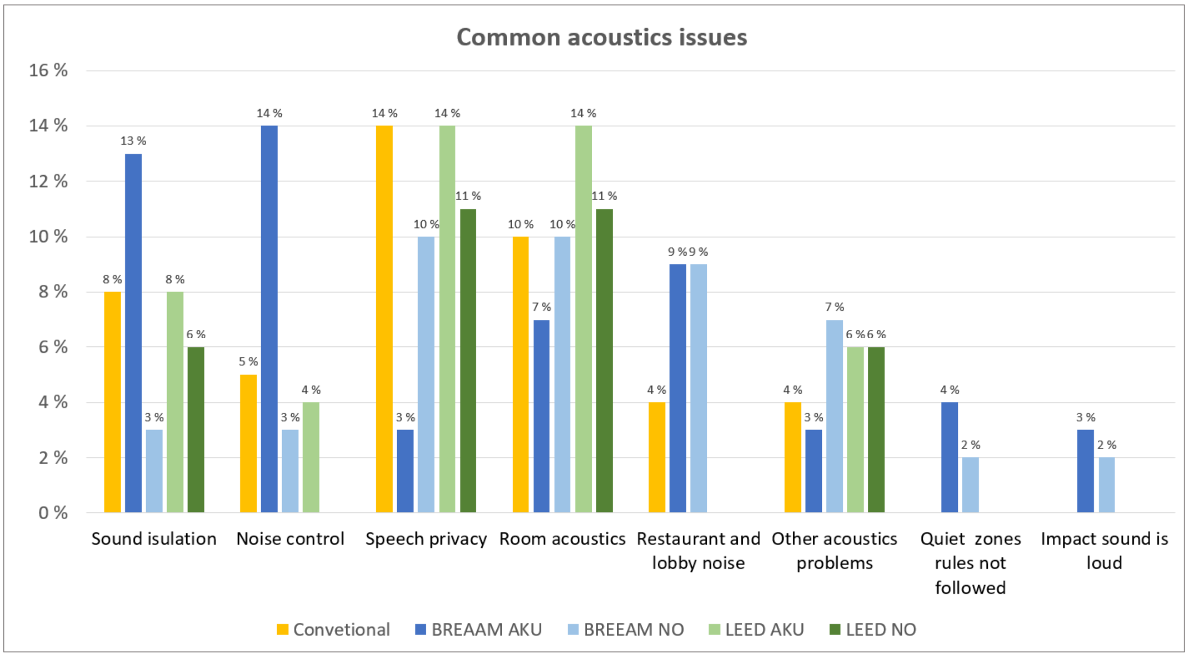

Figure 5: Buildings users commented about the acoustics performance, the most reported issues for all building types were related to speech privacy issues followed by room acoustics issues.

The goal of collecting acoustics measurement data for the buildings was to investigate whether certified green buildings fulfil their acoustic criteria and whether the measurement results correlate with the survey results. The survey included one question about satisfaction with background noise, the results from the question about satisfaction with background noise were converted into a satisfaction rate by method described in detail in the thesis (see p. 65) [1] and compared to the results collected from acoustics measurements in Figure 1. The finding is that acoustics measurements do not correlate with the survey results as shown in Figure 6.

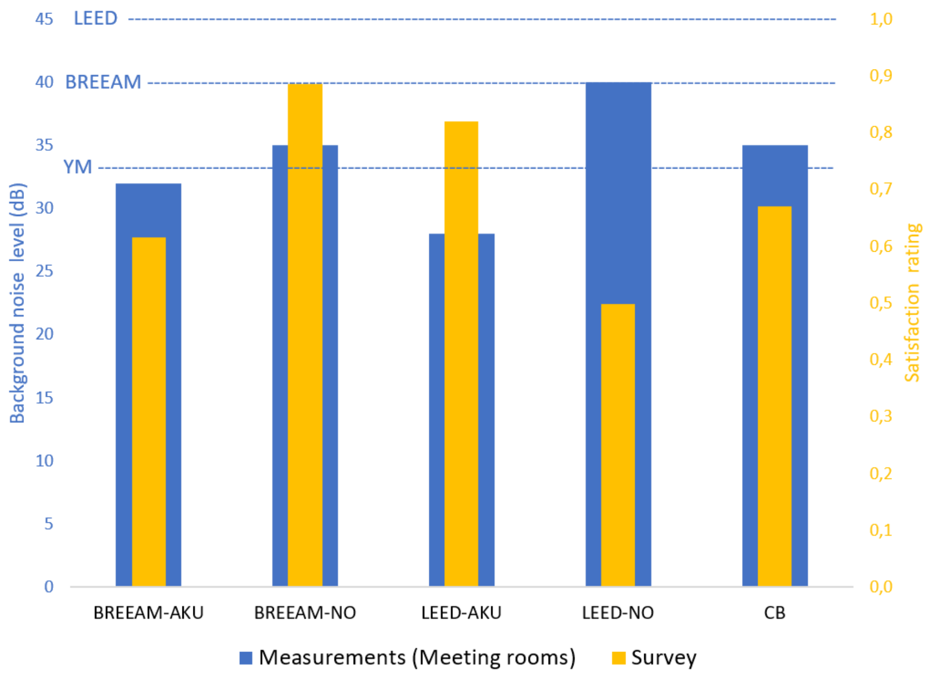

Figure 6: Measured background noise level (in blue) and acoustics satisfaction rating (in yellow) for meeting rooms in each building categories. The blue dashed lines refer to acoustic criteria of BREEAM, LEED, and YM (Ympäristöministeriön ohje rakennuksen ääniympäristöstä).

5 Conclusions

In this study, a POE survey results demonstrated that users' satisfaction with acoustics comfort fluctuated within building categories and questions regardless of the building being either green or conventional (within one building category, occupants did not consistently respond the same way; hence no robust pattern or correlation was revealed). This corresponds to previous studies' claims [9, 10, 11, 12, 13, 14] that acoustics comfort in green and conventional buildings are approximately the same. However, previous studies did not consider if the buildings investigated has acquired acoustics credits, whereas this study considered acoustics credits when investigating green buildings.

Additionally, survey results (for a question about speech privacy and open-ended question) demonstrated a high level of dissatisfaction with speech privacy in all building types.

This study suggested that personal and contextual variables influence users' acoustic comfort. Specifically gender as this study found that male users are significantly more satisfied with the acoustics comfort than female users. Additionally, the study indicated a direct connection between visual privacy and acoustics comfort: users who are pleased with visual privacy are more satisfied with the acoustics comfort.

A further finding of this study is that acoustics measurements results do not correlate with survey results. Accordingly, it demonstrates that there are other considerations when evaluating acoustics performance in office buildings in addition to the criteria for acoustics parameters specified by authorities or organizations. Other considerations include personal and contextual variables, which shall be considered to achieve the best acoustics comfort for building users. This means that designers need to consider office layout, workstation conditions, visual privacy, and working culture, which may affect acoustics comfort and acoustics conditions when complying with BREEAM and LEED criteria.

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Revised Finnish standard SFS 5907:2022 on acoustical design and quality classes of buildings

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The second edition of the Finnish standard SFS 5907 on acoustical design and quality classes of buildings was published in December 2022. Since the decree of the Ministry of the Environment on the Acoustic Environment of Buildings was given in 2017, several sections and definitions of the first edition (2004) of the standard had gone out of date. The classification system introduced in 2004 consisting of four acoustical classes had also turned out to be too detailed. Thus, the second edition of the standard introduces a new classification system consisting of three acoustical classes. Another major change is updating the single-number quantities for sound insulation so that both the requirements for airborne and impact sound insulation are now given as standardized values $D_{nT,w}$ and $L'_{nT,w} + C_{I,50\text{-}2500}$. In the second edition, special attention has been paid to the limiting values for railway induced vibration and ground-borne noise in different buildings, like apartments, hospitals, schools, and office buildings.

1 Introduction

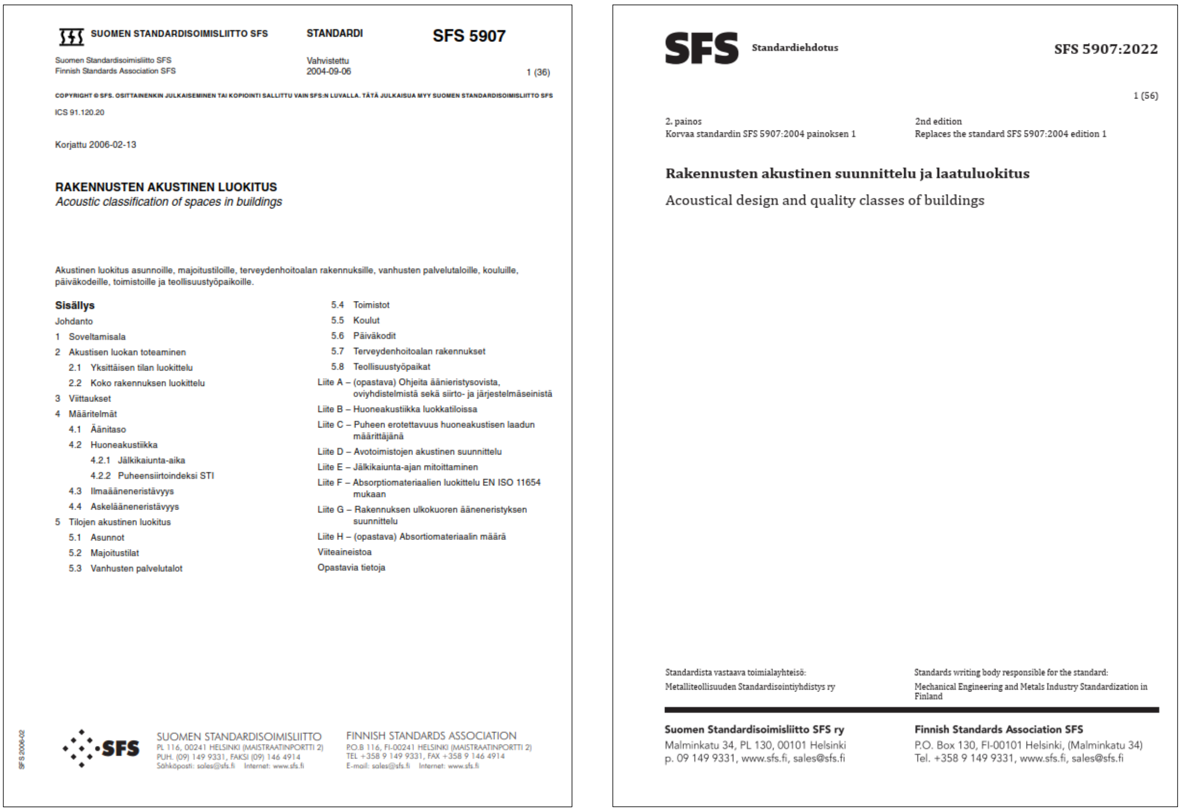
In Finland, the drawing up of the sound insulation requirements was first suggested in 1948, and some drafts were done in the 1950's and 1960's [1]. However, the first official Finnish sound insulation regulations came into force not earlier than in 1976. The regulations have been renewed in the years 1984 and 1998. During this period, the measurement methods have remained the same excluding few differences in the reference curve methods. The requirements for airborne sound insulation between rooms were given as weighted apparent sound reduction index R'_w and for impact sound insulation as weighted normalized impact sound pressure levels $L'_{n,w}$ [2].

The revision of the regulation in 1998 was significant: earlier, regulation concerned basically all building types except theaters, cinemas and concert halls. Thus, there were requirements given as single-number quantities for apartments, hotels, schools, offices, hospitals etc. [3]. In 1998, the regulation became functional. This meant that only requirements for apartment buildings were given as single-number quantities. For other buildings, sufficiently good acoustic environment was required. The acoustic environment had to correspond to the function of the building and room.

In couple of years after the revision had entered into force in the beginning of the year 2000, problems occurred especially in schools and day-care centers [4]. A solution to these problems was publication of standard SFS 5907 *Acoustic Classification of Spaces in Buildings* (2004) which complemented the regulations by presenting guidelines for schools, day-care centers, hospitals, offices and industrial workplaces (Fig. 1) [4–5].

New building acoustic regulation took effect in Finland in 2017 when the *Decree of the Ministry of the Environment on the Acoustic Environment of Buildings* was given [3, 6]. Based on scientific results [7–9], the single-number quantities were changed: the regulation is now based on weighted standardized level difference $D_{nT,w}$ concerning airborne sound insulation and sum of weighted standardized impact sound pressure level and spectrum adaptation term $L'_{nT,w} + C_{I,50\text{-}2500}$ for impact sound insulation.

The revision of the Finnish regulation made necessary to revise the standard SFS 5907, too. A working group consisting of the representatives of leading Finnish expert companies working in the field of acoustics was formed to carry out the revision. The group led by Dr Mikko Kylliäinen as a chairman started its work in the beginning of 2021 and the revised standard was published in December 2022 with a new title *Acoustical Design and Quality Classes of Buildings* (Fig. 1) [10].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ICS 91.120.20
Korjattu 2006-02-13

RAKENNUSTEN AKUSTINEN LUOKITUS
Acoustic classification of spaces in buildings

Akustinen luokitus asunnoille, majoitustiloille, terveydenhoitoalan rakennuksille, vanhusten palvelutaloille, kouluille, päiväkodeille, toimistoille ja teollisuustyöpaikoille.

Sisällys
Johdanto
1 Soveltamisala
2 Akustisen luokan toteaminen
2.1 Yksittäisen tilan luokittelu
2.2 Koko rakennuksen luokittelu
3 Viittaukset
4 Määritelmät
4.1 Äänitaso
4.2 Huoneakustiikka
4.2.1 Jälkikaiunta-aika
4.2.2 Puheensiirtoindeksi STI
4.3 Ilmaääneneristävyys
4.4 Askelääneneristävyys
5 Tilojen akustinen luokitus
5.1 Asunnot
5.2 Majoitustilat
5.3 Vanhusten palvelutalot
5.4 Toimistot
5.5 Koulut
5.6 Päiväkodit
5.7 Terveydenhoitoalan rakennukset
5.8 Teollisuustyöpaikat
Liite A – (opastava) Ohjeita äänieristysovista, oviyhdistelmistä sekä siirto- ja järjestelmäseinistä
Liite B – Huoneakustiikka luokkatiloissa
Liite C – Puheen erotettavuus huoneakustisen laadun määrittäjänä
Liite D – Avotoimistojen akustinen suunnittelu
Liite E – Jälkikaiunta-ajan mitoittaminen
Liite F – Absorptiomateriaalien luokittelu EN ISO 11654 mukaan
Liite G – Rakennuksen ulkokuoren ääneneristyksen suunnittelu
Liite H – (opastava) Absortiomateriaalin määrä
Viiteaineistoa
Opastavia tietoja

SUOMEN STANDARDISOIMISLIITTO FINNISH STANDARDS ASSOCIATION

SFS Standardiehdotus SFS 5907:2022
1 (56)
2. painos
Korvaa standardin SFS 5907:2004 painoksen 1
2nd edition
Replaces the standard SFS 5907:2004 edition 1

Rakennusten akustinen suunnittelu ja laatuluokitus
Acoustical design and quality classes of buildings

Standardista vastaava toimialayhteisö:
Metalliteollisuuden Standardisointiyhdistys ry
Standards writing body responsible for the standard:
Mechanical Engineering and Metals Industry Standardization in Finland

Suomen Standardisoimisliitto SFS ry
Malminkatu 34, PL 130, 00101 Helsinki
p. 09 149 9331, www.sfs.fi, sales@sfs.fi
Finnish Standards Association SFS
P.O. Box 130, FI-00101 Helsinki, (Malminkatu 34)
Tel. +358 9 149 9331, www.sfs.fi, sales@sfs.fi

Figure 1: The cover pages of the first (left) and second (right) editions of the standard SFS 5907.

2 Major revisions of the standard

2.1 Overview of the revised standard

The major changes in the scope and coverage between the first and second edition of SFS 5907 are shown in Fig. 2. Since the publication of the first edition, also new room types have occurred which has been taken into account in the revision. The revised standard covers all the conventional building and room types, like apartments, hotels, office buildings, daycare centers, schools, hospitals and industrial workplaces. For each type of buildings and spaces, recommendations are given for airborne and impact sound insulation, room acoustics, allowable sound levels from HVAC systems, allowable sound levels from outdoor traffic noise sources and railway induced vibration and ground-borne noise. Concert halls, cinemas, theaters, libraries and museums are excluded as more or less unique buildings.

One notable change in the revision is the amount of the references. The first edition was mostly based on regulation, other standards, either international or Nordic. The revised standard and its guidelines are mostly based on research presented in scientific articles.

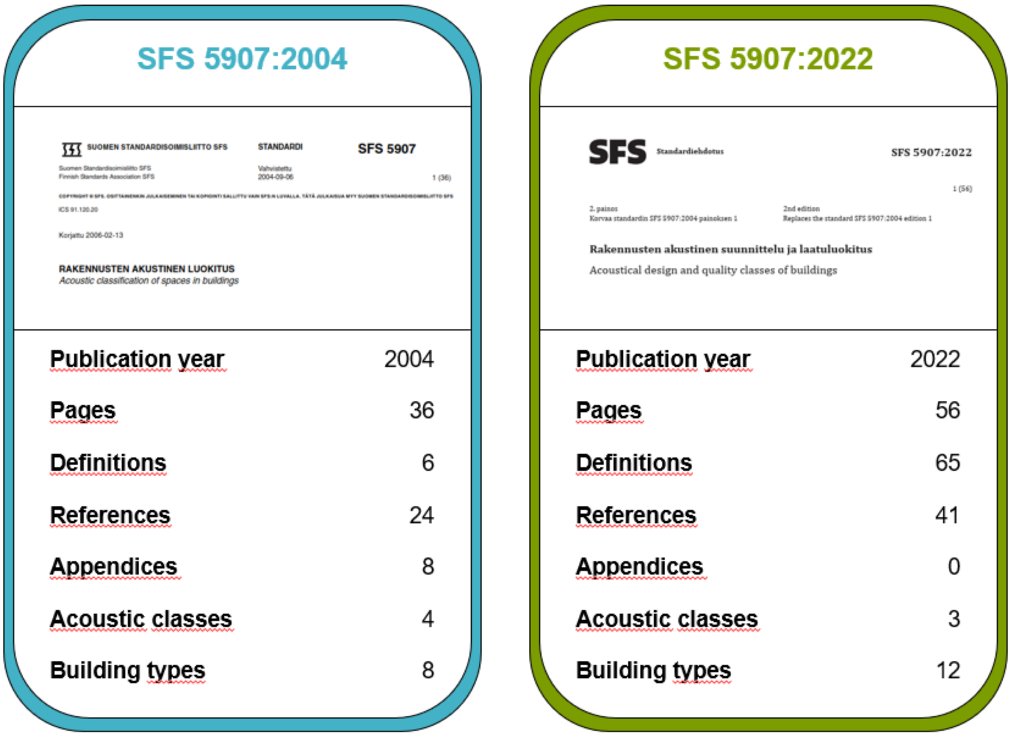

Figure 2: Dimensional inspection of the first and second editions of the standard SFS 5907.

2.2 Classification

Probably most notable change between the first and second edition of the standard is the amount of acoustic classes. Just a few years before the publication of the first edition, Nordic countries had co-operated in preparation of an acoustic classification scheme for dwellings [11]. The classification was soon adopted in other Nordic countries and in Baltic countries, too. In Finland, the classification was introduced in the first edition of SFS 5907 [4–5]. The classification consisted of four classes. Class C corresponded to the building regulation, classes A and B were believed to produce better acoustic environment and class D was meant for classification of old buildings only [5].

In the revised standard, the number of acoustic classes has been dropped from four to three. The reason for this is that buildings fulfilling the previous class A have not been designed or built. On the other hand, it is not at all clear how well people can experience the changes between the acoustic classes. In order to avoid confusion between the previous four-level and the new three-level classification, new notation is used. In the present classification, class A2 corresponds to the building regulation, class A1 is meant for better acoustic environment and class A3 is used in classification of old buildings [10]. The sound insulation level of class A3 corresponds approximately to the level given in Finnish building regulations in 1984 [2].

2.3 Railway induced vibration and ground-borne noise

During the last decade, railway induced vibration and ground-borne noise have become an important engineering problem in Finland because of new railway and tram projects and supplementary construction of the urban areas. In Finland, there has not been building regulation giving limiting values for vibration and ground-borne noise of all building types. The standard gives such values for each building and room type. According to the authors' knowledge, classification for railway induced vibration and ground-borne noise has not been adopted in the corresponding standards, at least not to this extent.

The limiting values for railway induced vibration are given as maximum allowed weighted vibration velocity level of 95 % confidence interval $v_{w,95}$ [mm/s]. For ground-borne noise, the limiting values are given as maximum allowed ground borne-noise level of 95 % confidence interval L_{prm} [dB] in buildings.

2.4 Omitting the low-frequency procedure in sound insulation measurements

Standard ISO 16283-2 for field measurements of impact sound insulation introduced a low-frequency procedure (LF-procedure) for measuring impact sound pressure levels and reverberation times at three 1/3-octave bands below 100 Hz since 2015 [12–13]. The LF-procedure is followed when the volume of the receiving room is smaller than 25 m^3. The LF-procedure is based on studies concerning field measurements of airborne sound insulation [14]. Its aim was to improve measurement uncertainty of low-frequency sound insulation measurements.

The LF-procedure has not been assessed for use with impact sound insulation measurements. In an assessment of the procedure in 2021 [15], it was shown that there are several problems concerning the application of the LF-procedure in field measurements of impact sound insulation. The application of the LF-procedure changes the rating of impact sound insulation in small rooms significantly, which means that that impact sound insulation and floors of small rooms should be designed and constructed differently from the larger rooms, i.e. with floor structures having better impact sound insulation. This leads to rising building costs. The conclusion of the assessment presented in [15] is that the use of the LF-procedure is so far not justified in impact sound insulation.

In Finland, the *Decree 796/2017 of the Ministry of the Environment on the Acoustic Environment of Buildings* [6] concerns airborne sound insulation in frequency range from 100 to 3150 Hz. In rating the impact sound insulation between spaces, the sum of $L'_{\mathrm{nT,w}} + C_{\mathrm{I50,2500}}$ is used. As the LF-procedure is shown to be unsuitable for impact sound insulation measurements and low frequencies below 100 Hz do not affect the rating of airborne sound insulation, the LF-procedure is not applied in the standard SFS 5907 [10].

2.5 Example of classification of apartments

As an example of acoustic quantities to be classified according to the revised standard SFS 5907 [10], the limiting values in the three classes for apartment buildings are given in Table 1. The limiting values for airborne and impact sound insulation are given as standardized single-number quantities $D_{\mathrm{n}T,\mathrm{w}}$ and $L'_{\mathrm{n}T,\mathrm{w}} + C_{\mathrm{I,50\text{-}2500}}$. The sound insulation values are given for insulation between dwellings. More situations for sound insulation are found in the standard.

Table 1: Limiting values for different acoustic quantities for apartment buildings in acoustic classes according to the revised standard SFS 5907 [10].

Situation Quantity	Acoustic class		
	A1	A2	A3
Airborne sound $D_{\mathrm{n}T,\mathrm{w}}$ [dB]	≥ 60	≥ 55	≥ 53
Impact sound $L'_{\mathrm{n}T,\mathrm{w}} + C_{\mathrm{I,50\text{-}2500}}$ [dB]	≤ 48	≤ 53	≤ 58
HVAC noise $L_{\mathrm{A,eq},T}$ [dB]	≤ 24	≤ 28	≤ 30
Outdoor noise inside $L_{\mathrm{A,eq,07\text{-}22}}$ [dB] $L_{\mathrm{A,eq,22\text{-}07}}$ [dB]	≤ 30 ≤ 25	≤ 35 ≤ 30	≤ 35 ≤ 30
Ground-borne noise L_{prm} [dB], from tunnel L_{prm} [dB], form track	≤ 25 ≤ 30	≤ 30 ≤ 35	≤ 35 ≤ 35
Railway vibration $v_{\mathrm{w,95}}$ [mm/s]	≤ 0,15	≤ 0,30	≤ 0,60

3 Summary

The revision of the Finnish standard SFS 5907 *Acoustical Design and Quality Classes of Buildings* [10] has brought the standard and its scientific basis to the state-of-the-art level of present knowledge on building acoustics. The working group believes that its impact on the construction industry in Finland and acoustic design of buildings will be significant as was in the case of its predecessor published in 2004 [4–5]. During the year and a half since the publication of the standard, it has been used as a guideline for setting the acoustical goals of the projects, in preliminary design phases, as a reference material in design competitions of different project types.

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Guideline limit values for vibration to avoid damage to structures and natural slopes. Revision of Norwegian Standard NS 8141

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Construction activities such as blasting, piling, compaction, excavation, and construction traffic can produce vibrations of sufficient strength to cause damage to neighbouring buildings and structures. In Norway guideline limit values for construction vibrations are given in Norwegian Standard 8141. Part 1 of this series of standards has recently been revised and a new edition was published in 2022. Based on results from a research study, the determination of limit values has been simplified and the limit values have been adjusted. Vibration may also be a trigger for landslides in vibration sensitive ground. NS 8141-3 gives guideline limit values for vibrations from blasting triggering landslides in quick clay. However, these limit values have in many cases been considered too strict, as they do not take into consideration site specific conditions, i.e. safety factor of slopes. There is also a need for a better description of how the measurements are to be carried out, e.g. number of measurement positions and location of the sensors in the clay. Therefore, part 3 is currently under revision and a research project is ongoing to collect necessary background material. In this paper we describe the outcome of the research projects and changes made to NS 8141.

1 Introduction

Construction activities such as blasting, piling, compaction, excavation, and construction traffic can produce vibrations of sufficient strength to cause damage to neighbouring buildings and structures. In Norway guideline limit values for construction vibrations are given in the Norwegian Standard, NS 8141 *Vibration and shock. Guideline limit values for construction work, open-pit and pit mining and traffic*. NS 8141 was first issued in 1993 and revised in 2001, 2012-2014 and from 2021 to today.

The current Standard consists of three parts. NS 8141 – 1 [1] concerns effect of vibration and air blast on constructions, NS 8141 – 3 [2] concerns effect of vibration from blasting on triggering landslide in quick clay, and NS 8141 – 4 [3] gives guidance for surveying of construction works. Part 1 has newly been reviewed and a new edition was published in 2022. In connection with this, the previous Part 2 was withdrawn, and the contents were included in the revised Part 1. Part 3 is currently under revision and a new edition is planned to be published in 2025. The first edition of Part 4 was published in 2021. In addition, a guidance to NS 8141 was published in 2023 [4]. The Standard and the guidance are in Norwegian but a translation to English is planned.

2 Part 1 - effect of vibration and air blast on construction

The guideline limit values calculated according to the method described in Part 1 are values that buildings are supposed to withstand through repeated exposures and are intended to prevent damage. The guideline limit values should therefore not be considered as damage limits. Building damages assumed to originate from vibrations have seldom been observed. This may indicate that the limit values have been unnecessarily strict.

Vibrations at low frequencies are expected to be more damaging to structures than vibrations at higher frequencies, especially frequencies close to the building’s fundamental frequencies. In e.g. British and American standards this is handled by using frequency dependent limit values. In Norwegian and Swedish standards, the limit values are frequency independent, but the frequency is to some extent considered since the limit value depends, in addition to the properties of the structure, on ground conditions, type of foundation, distance from building and type of vibration source, all of which can affect the frequency. Using this approach, it is not necessary to determine the frequency when measured values are to be compared to the limit values. However, it is unclear how well the factors reflect the frequency, and they do not take into account factors such as design of the charge, which are known to affect the frequency. In addition, it is not straight forward to calculate the limit values since the ground conditions and foundation type are often unknown and the distance may vary.

In the previous revision of the Standard (issued 2012 – 2014), frequency-weighted limit values were therefore introduced to simplify limit value determination and to make the limit value more directly dependent on the frequency, as in the American and British Standards. The frequency weighting filter placed more emphasis on vibrations at low frequencies and less emphasis on vibrations at higher frequencies. By using the frequency filter, a limit value can be determined which depends only on the vibration source and the characteristics of the building, and not on ground conditions, foundation method and distance. Unfortunately, this frequency weighting had to be abandoned because the deviation from the industry's previous experiences was too great, which resulted in complaints and demands for withdrawal. The 2001 edition of the standard [5] was therefore resumed, but a revision was still considered necessary.

The latest revision of Part 1 started in 2021 and the revised NS 8141 – 1 was published in 2022. The vibration measure used in NS 8141 – 1:2022, is again unweighted peak particle velocity (PPV). Based on results from a research study with full scale blast tests described below, the determination of limit values has been simplified and the limit values have been adjusted.

The major changes in NS 8141 – 1:2022 compared to NS 8141:2001 are:

- Construction activities other than blasting and traffic have been included (this was covered by Part 2 in the 2012-2014 edition of the standard).
- The nominal frequency range has been expanded to 2 Hz – 315 Hz. This was made to incorporate e.g. vibration from traffic and piling at great soil depth which could have very low frequency components.
- Limit values are given for tunnels and rock caverns.
- Limit values are given for air blast.
- Triaxial measurements are mandatory for blasting closer than 10 meters from construction works.
- Simplification and adjustment of the Ground condition factor. The factor for buildings on soil has been increased, especially for soft soil, resulting in higher limit values. The factor for hard rock with high wave velocity has been removed, reducing the maximum limit value for a normal dwelling on rock to 50 mm/s.
- Simplification of the Foundation Factor from four different foundation methods to two alternatives “on Soil" and “on Rock".
- Regulated use between Building type factor and Material and Building detail factor to avoid an unfortunate combination of several reduction factors.
- Simplification of the distance factor. The distance factor is equal to one for all ground conditions for distances between 10-100 m.

2.1 Full scale blast test

Two instrumented blast tests were performed in Spulsåsen rock quarry in Våler municipality in Hedmark, Norway, in November 2018 and November 2020 [6]. In the first blast test, two test buildings were erected at the site, one in cast-in-place concrete and one made of lightweight construction (Leca) blocks. Both were founded on an approximately 500 mm levelled and compacted layer of gravel, over rock. In the second blast test, one building made of Leca was constructed on top of an approximately 4 m thick filling, established at the same location as the buildings in the first test series. The buildings were instrumented with velocity sensors (geophones) and strain sensors. The blasts were designed to give increased vibration strength, starting at a low value for the longest distance and increasing progressively as the blasts came closer to the test structures. The maximum amount of explosives detonated in one blast round was 1485 kg in total and 47.8 kg per delay. The minimum distance between the blast and the buildings was 7 m.

The first blast tests produced vibration values above PPV = 260 mm/s and the second blast tests a maximum PPV = 180 mm/s. These results are well above the vibration limit values calculated according to NS 8141:2001, which were 50 mm/s for the buildings on rock and 16-23 mm/s for the building on filling. Despite this, no visible damage could be detected on any of the test buildings. The results of the two blast tests gave confidence that the limit values included a large safety margin and could be adjusted.

2.2 Calculation of limit values according to NS 8141-1

The limit value in NS 8141-1:2022 is calculated according to equation (1).

$$v = v_0 \cdot F_G \cdot F_b \cdot F_m \cdot F_f \cdot F_d \cdot F_k \tag{1}$$

Where v_0 is uncorrected PPV, and the following factors: F_G ground condition, F_b building type F_m, material and building details F_f foundation type, $\cdot F_d$ distance, $\cdot F_k$ vibration source. Example of guideline limit values calculated according to NS 8141-1:2022 and NS 8141:2001 are shown in Table 1.

Table 1: Example of limit values PPV (mm/s) calculated according to NS 8141

Situation	NS 8141-1:2022	NS 8141:2001
Housing made of blocks of aggregated clay. Slab on clay. Blasting in 100 m distance	21 mm/s	7 mm/s
Wooden house. Strip foundation on moraine. Blasting in 50 m distance	32 mm/s	16 mm/s
Wooden house on hard rock. Blasting in 10 m distance	50 mm/s	70 mm/s

3 Part 3 - effect of vibration from blasting on triggering landslide in quick clay

Blast vibration may also trigger landslides in vibration sensitive ground. An example of potential triggering of a landslide by blasting for a tunnel construction under a slope is sketched in Figure 1. Guideline limit values to prevent triggering of landslides was first introduced in NS 8141 – 3:2014 as a reaction to the landslide at Namsos in 2009, where it was concluded that blasting caused the landslide. This limit value was based on a research study, [7][9], proposing an unweighted value of v = 25 mm/s measured in or on top of the quick clay. However, to harmonise Part 3 with the other parts of NS 8141, the proposed limit value was converted to a frequency weighted limit value of v_f = 45 mm/s, based on an assumption of the typical frequency content of vibration in soft soil. The limit value in the standard was set to ensure that vibrations from rock blasting do not trigger slides in quick clay, where conditions are such that an initial failure of clay material may develop into a landslide. There is a good safety margin in the limit value for safety factors for undrained slope stability above 1.4. In some recent construction projects, blast vibrations with peak values of up to 50 mm/s have induced only small amounts of pore pressure increase, with expected little or no effect on the overall stability of the slopes. The vibration tolerance is of course dependent on the site-specific ground conditions and slope's stability prior to blasting.

This limit value has in many cases been considered too strict/conservative, as it does not take into consideration site-specific conditions, e.g. the safety factor for stability of slopes that are exposed to blast vibrations. However, in a few cases it could potentially be non-conservative, if the slope stability is lower than computed, e.g. due to non-detected weak areas or layers in the slope. In addition, there is also a need for a better description of how the measurements are to be carried out, e.g. number of measurement positions and location of the sensors in the clay, especially for tunnel blasting. For typical tunnel blasting there is likely no need to measure vibrations with respect to slope stability at distances further away than 50 m. Therefore, part 3 is currently under revision and a research project is ongoing to collect necessary background material.

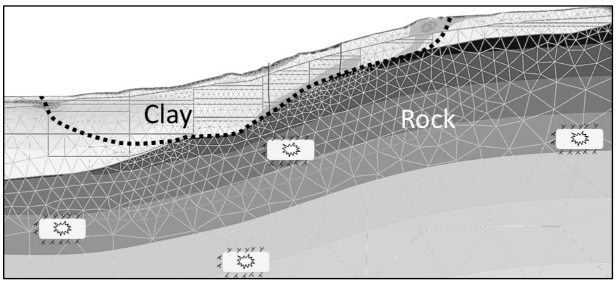

Figure 1: Potential triggering of landslide by blasting for rock tunnel beneath slope with vibration sensitive material for different locations of the tunnel with respect to the critical potential sliding surface marked with dotted line.

3.1 Previous blast related landslides in quick clay

There seems to be a common opinion in the geotechnical environment, especially in Norway, that vibrations from blasting do not have a high probability of triggering landslides in sensitive clays. However, several historical landslides in sensitive clays in Canada, Norway and Sweden are related to rock blasting activity [8]. It is emphasized that blasting itself is often not the only cause of a landslide. As part of the research project a literature review of previous landslides in quick clay was performed and for some of the events the vibration values were estimated. The study shows that landslides are usually triggered by a combination of destabilizing conditions such as low stability before blasting, unfavourable groundwater conditions (heavy precipitation or snow melting, artesian pressure), erosion, filling, etc. For all previous landslides in quick clay where (explosive) vibration may have contributed to the release, the landslide had a run out down towards water. This is most likely connected to the fact that loosely deposited material is often found around water bodies.

3.2 Adjustment of the limit value

In the research project, a basis has been established for including the safety factor in the calculation of the limit value. This is, among other things, based on laboratory tests in NGI's laboratory, field measurements of pore pressure and vibration. A database of vibration measurements from blasting in areas with quick clay was built up. The database consists of 1586 timeseries from three different locations with quick clay. Tunnel blasting was carried out at two of the locations and bench blasting at the third location. To assess a possible adjustment of the limit value, pore pressure measurements were also performed at the locations with tunnel blasting. In addition, several static and cyclic tests were carried out on quick clay from the locations with tunnel blasting. To simulate the shear stress acting on a soil element in a slope, the quick clay samples were subjected to different levels of static shear stress before they were loaded cyclically to mimic the blast vibrations.

Based on the results, a new procedure for calculating a vibration limit value, vel_{lim} is proposed. The procedure considers the slope's static (undrained) safety factor, Fs, the plasticity index, I_p, and the shear wave velocity, V_s. A starting point for the vibration limit evaluation is a relation between the cyclic shear strain in the soil and the peak particle vibration velocity and the shear wave velocity in the soil, which has been verified with numerical 2D and 3D equivalent linear analysis e.g. [10][11] and is valid for blast induced cyclic shear strains levels.

$$\gamma_{cy} = \frac{PPV}{V_s} \tag{2}$$

To compute a vibration limit, vel_{lim}, we use the concept of a threshold shear strain for volumetric compaction of a soil $\gamma_{cy,tv}$ [13]. For shear strains larger than this threshold, pore pressure can build up during cyclic loading and reduce the soil's strength. In all cases where blast vibrations are suspected to have caused quick clay landslides, there has been silt and sand involved. This together with the results from our laboratory tests, indicate that loose silt and sand are much more vibration sensitive than quick clay. Therefore, a threshold strain determined from several case histories of earthquake liquefaction in sandy/silty soils of 0.03% [12] has been adopted. However, this threshold strain is for relatively flat ground. For a sloping ground, or lower safety factor, cyclic lab tests indicate that the pore pressure and permanent strain increase. We have therefore introduced a linear reduction of the threshold shear strain from 0.03% for a safety factor, Fs = 2, to 0.01% for $Fs = 1$, giving the following threshold strain in percent.

$$\gamma_{cy,tv} = [0.01 + 0.02(Fs - 1)] \times f(I_p) \tag{3}$$

Where $f(I_p)$ is introduced to take into account to some extent that soils with higher plasticity index, I_p [%] withstand larger vibrations and more cycles due to viscous effects [15].

$$f(I_p) = 1 + 0.1 \times I_p \tag{4}$$

The undrained safety factor, Fs, is intended to be used in the calculation of limit value, but if the drained safety is lower, this one should be used instead.

The shear modulus decreases with increasing cyclic shear strain. Therefore, a factor of 0.8 is introduced in the limit value calculation to account for reduction of shear wave velocity with increased strain. Based on (2) and (3), the proposed equation for calculation of limit value then becomes:

$$vel_{lim} = 0.8 \times V_s \times \gamma_{cy,tv} \tag{5}$$

In cases where the shear wave velocity is not known, it may be estimated based on the depth, D, in meters below the terrain where the quick clay is located. The following equation, which is based on several field measurements performed in Norwegian soft clays [14], gives a relatively low estimate of the shear wave velocity.

$$V_s = 100 \times D^{0,25} \tag{6}$$

The threshold strain reduces with increasing number of vibration cycles. The 0.03% threshold strain for earthquake liquefaction is based on an average of 10 equivalent load cycles [12]. Our analyses of blast vibrations measured on or in clay at some depth, show an average of 7 equivalent vibration cycles, indicating that the above choice of threshold strain is reasonable and likely conservative. However, for large blasts, such as in quarries, the number of cycles could be larger depending on blast design.

One of the difficulties with quick clay slopes is the potential presence of thin loose silty or sandy layers within the quick clay. Such layers are not easily detected in the field and difficult to test in the laboratory. Therefore, the effect of these layers on a slope's sensitivity to blast vibrations needs further study. This uncertainty may be counteracted by the viscous strain rate effects in clayey materials, due to the relatively high frequencies of blast vibrations. Our study shows that the frequencies are mainly in the range from 50 to 150 Hz, which is considerably higher than in typical laboratory tests with 1 and 10 second periods.

3.3 Simplification in measurement set-up

The database was also used as a basis to propose changes and simplifications of the measurement set-up in the Standard. Differences between measurements performed into the clay, on the surface and on buildings were studied. In Figure 2 The PPV in the highest axis direction for each measurement point is against scaled distance calculated according to (7).

$$d_{scal} = \frac{d}{\sqrt{Q}} \tag{7}$$

Where d is the distance from blast to measurement point in meters and Q is the maximum charge per delay in kg. Measured PPV was compared to the value of a reference curve, and a factor F, which is measured PPV/PPV for the reference curve at the same scaled distance was determined for all measurements. The reference curve was established as the best fit to all measurement data. The average F for the three studied measurement positions are shown in the legend in Figure 2.

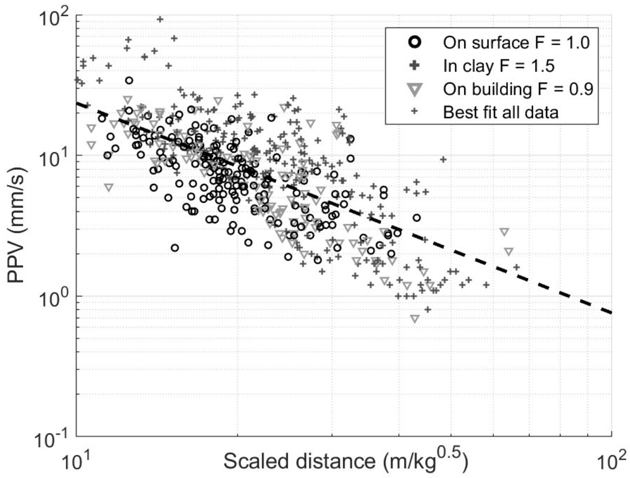

Figure 2: Measured PPV vs scaled distance for measurement points on the surface, into the clay and on buildings. Average F (measured PPV/PPV for the reference curve at the same scaled distance) are shown in the legend.

Based on the comparison of measurements deep into the clay, on the surface and on buildings, the following recommendations are given about changes in the measurement set-up:

- Figure 2 shows that measurements on the surface give somewhat higher values than measurements into the clay and can be considered a conservative approach. It is therefore probably not necessary to carry out measurements with sensors deep into the clay. Measurement on buildings seem to give somewhat lower values but may be allowed together with a correction factor on the limit value.
- As shown in Figure 2, the spread in data is large. It is therefore desirable that the requirement for a minimum of three measurement points in the present NS8141 – 3 is kept. Alternatively, it may be considered to use a correction factor on the limit value which depends on the number of measuring points.

There is a large spread in measurement data also for measurement positions located close to each other. It is doubtful whether the requirement in the present NS8141 – 3, that the measurement points shall be placed on a line, with the first point placed where the rock meets the clay, improves the situation. As this requirement has been experienced as difficult to achieve by the users, the requirement may be replaced with a requirement that the measurement points shall be spread over the area with quick clay, with at least one point as close to the blast site as possible.

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Immersive sound system showroom acoustical design in existing industrial premises

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Modern immersive sound audio systems manufacturers promote their products via showrooms, where customers can experience features of an audio system, its sound quality and technical possibilities. These showrooms require low reverberation time, similar to surround audio mixing rooms. The problem arises, when showrooms must be fitted in existing premises, especially if these are industrial gentrified premises, which is a common trend in Europe. Industrial premises, like production halls and storage rooms, are made using hard reflective materials – concrete, steel and brick. To fit room acoustics for showrooms function, in most cases extensive acoustical treatment is needed. This paper shows a practical case of acoustical treatment for L-acoustics L-ISA system showroom in Riga, Latvia. First, measurement results are presented for the untreated room, showing average T30 around 1.5 seconds. Next, acoustical design is shown, with emphasis on low frequency scattering and absorbing wall panels for bass control, based on Odeon modelling. Finally, post-treatment measurement results are presented, with T30 around 0.55 seconds, and conclusions are made about acoustical modelling techniques precision. Practical challenges and uncertainties due to various impact factors are addressed and discussed.

1 Introduction

L-acoustics showroom located at 8 Unijas street, block 2, Riga, Latvia, is a demonstration hall for L-acoustics products, including immersive sound system L-ISA. The hall is equipped with 7 loudspeakers and 2 subwoofers at the front wall, 8 loudspeakers suspended from the ceiling around the central listening area for sound effects, and 7 loudspeakers further and around the central listening area at a lower height. The sound system is purposed to reproduce all music genres with high fidelity, clarity and spaciousness.

Immersive sound systems for speech intelligibility purposes require the room acoustics to be with sufficiently low reverberation time RT [1] for wide frequency range. The showroom operator according to L-acoustics recommendations has established RT target of 0.9 s. As an additional requirement it was established that the sound field of the room must remain diffusive.

2 Initial measurements

2.1 Description

Measurements were performed in accordance with ISO 3382-1:2009 Acoustics - Measurement of room acoustic parameters - Part 1: Performance spaces [2]. Empty hall is approximately 120 m^2 and has a ceiling height of 3.7 m, volume 444 m^3. There are concrete and metal structural ribs on the ceiling of different dimensions. The floor is polished concrete. The façade is a brick wall mixed with glass façade structure with entry doors. The amount of glass relatively to the full area of the façade is approximately 70%. There is a semi-transparent folded curtain in front of the façade wall and windows. The second wall is a massive brick wall. The surface parallel to the façade is both a light-weight wall, a glass window and an opening that leads to another room. This wall is covered with the same curtain as the façade wall. The fourth wall has an integrated wardrobe, the rest of the wall area is a hard surface, presumably painted concrete. Sound system was not yet installed in the room. During the measurements, the temperature was 18°C, humidity 45%.

A sine e-sweep signal was applied to the measurements and the resulting impulse response was back-wards integrated to obtain a time curve of the sound energy decay. Sound source height was 1.5 m from the floor level, microphone heights 1.2 m. In the hall, the sound source was placed in 3 different positions and measurements were taken at 5 points for each source location thus creating 15 source-receiver combinations.

Equipment used: audio interface *Presonus AudioBox 22 VSL*, omnidirectional speaker *NTI DS-1*, subwoofer *Thomann TheBox Pro*, omnidirectional microphone *NTI M2230*, software *Odeon Auditorium*.

The measured average reverberation time is given in Table 1.

Figure 1. View on the showroom from two corners. An opening leading to administration office is behind the curtain on the photo to the left.

Table 1. T30 (s) values in octave frequency bands.

f (Hz)	63	125	250	500	1000	2000	4000	8000
Min	1.89	1.67	1.59	1.47	1.39	1.23	1	0.63
Max	2.85	2.81	1.93	1.66	1.67	1.53	1.27	0.82
Average	2.17	1.92	1.76	1.54	1.53	1.39	1.12	0.74

2.2 Discussion

The manufacturer's recommended reverberation time for demonstration rooms is 0.8 seconds, while the mean T30 measured is 1.54 seconds (500-1000 Hz). To achieve the recommended reverberation time, absorption materials were applied in the room. The hall has a relatively high average clarity C80 of 4.7 dB, Latvian construction standards recommend C80 >0 dB in rooms where electroacoustical amplification is applied, and up to +3 dB in concert halls. Early decay time EDT measured in the hall was lower than the reverberation time T30 (1.27 s versus 1.54 s accordingly at 500 Hz octave band), so the reverberation time subjectively is perceived to be lower, and the overall subjective impression is that of a clear, informative soundcape.

3 Modelling and acoustical treatment

A 3D acoustical model of the hall was developed for *Odeon Auditorium*. Calculations are based on geometrical acoustics, namely the Reflection Based Scattering method [3]. The frequency of modal overlap, proposed by Schroeder [4], is

$$f_g = 2000\sqrt{\frac{T}{V}} = 117 \text{ Hz} \quad (1)$$

Therefore, calculations based on geometrical acoustics for the modelled hall can be considered reliable for octave bands 250 Hz and higher and partially reliable for 125 Hz octave band. That also means that the results of 63 Hz octave band calculations cannot be considered reliable.

Variable parameters of the model were absorption coefficient and scattering coefficient. The shape of the model represented the real room together with the adjacent administration rooms through an opening, which were not subjects of the study, but impacted the acoustics of the showroom. Walls and floor were modelled flat, the ceiling included structural bars, two supporting columns were also modelled. The model was calibrated using measured reverberation time T30. The calibration ensured that simulated and measured data had only minor differences, both absolute and relative (Table 2). Absorption and scattering coefficients for the model are given in Table 3.

Acoustical treatment, e.g. the applied solutions, were following.

To reduce sound energy in low frequencies custom designed membrane absorbers were proposed. It was more advantageous to place these membrane absorbers onto the wall opposite to the loudspeakers. That wall is a flat concrete wall, so membrane absorbers were designed in a random manner, not only to introduce low frequency absorption, but also to introduce mid-low and mid frequency scattering, while also keeping enough reflective surfaces for the high frequency saturation. Panels have varying depth from 100 mm to 250 mm, varying dimensions from 50 cm to 120 cm, front panel – plywood 5 mm, thick (at least 15 mm) plywood side panels filled with mineral wool.

The second solution was to treat the ceiling – large reflective surface, to bring reverberation time to overall lower values. Mineral wool suspended ceiling panels *Ecophon Industry™ Modus TAL* were used, 40 mm thickness, suspended at 300 mm from the ceiling. Total area – approximately 65 m^2.

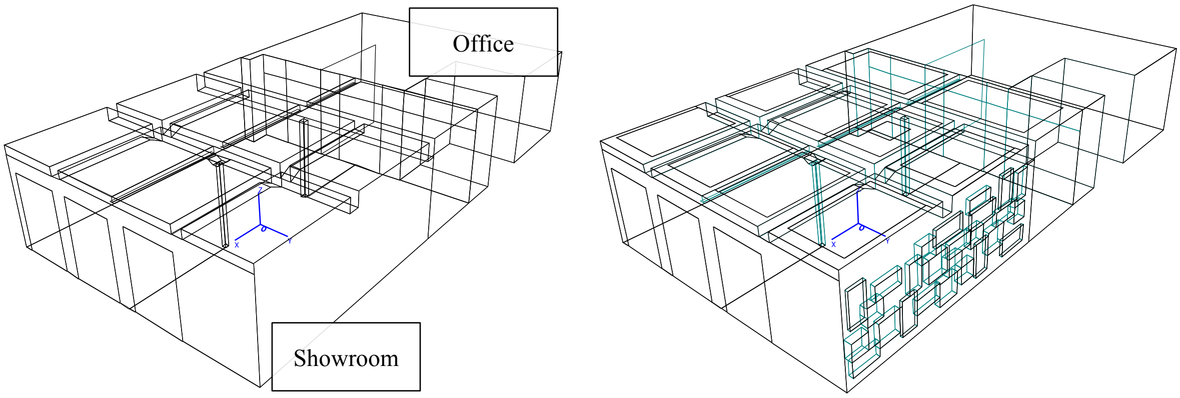

Figure 2. Acoustical model before (to the left) and after (to the right) the implementation of solutions.

Table 3. Absorption and scattering coefficients in octave frequency bands [f, Hz].

Surface	**63**	**125**	**250**	**500**	**1000**	**2000**	**4000**	**8000**	**Scatter**
Ceiling	0.01	0.01	0.01	0.02	0.02	0.02	0.05	0.05	0.05
Floor	0.01	0.01	0.01	0.01	0.02	0.02	0.02	0.02	0.05
Structural beam	0.02	0.02	0.03	0.03	0.03	0.04	0.07	0.07	0.10
Glass wall	0.35	0.35	0.25	0.18	0.12	0.07	0.04	0.04	0.05
Brick wall	0.08	0.08	0.09	0.12	0.16	0.22	0.24	0.24	0.20
Concrete wall	0.11	0.11	0.08	0.07	0.06	0.05	0.05	0.05	0.10
Gypsum board wall	0.30	0.3	0.12	0.08	0.06	0.06	0.05	0.05	0.10
Wardrobe	0.28	0.28	0.22	0.17	0.09	0.10	0.11	0.11	0.15
Ceiling panels	0.40	0.65	0.90	1.00	1.00	1.00	1.00	1.00	0.25
Membrane absorber	0.50	0.5	0.30	0.10	0.05	0.05	0.05	0.05	0.40
Curtains on HVAC	0.07	0.07	0.31	0.49	0.81	0.66	0.54	0.54	0.40
Curtains on wall and glass	0.07	0.07	0.31	0.49	0.81	0.66	0.54	0.54	0.30
Curtains on openening	0.05	0.05	0.06	0.39	0.63	0.70	0.73	0.73	0.15

Table 2. Reverberation time T30 measured and modelled before solutions in octave frequency bands [f, Hz] and respective difference percentage.

T30, s	63	125	250	500	1000	2000	4000	8000
Measured	2.18	1.93	1.88	1.56	1.53	1.39	1.13	0.74
Model	2.23	1.95	1.8	1.57	1.52	1.35	1.04	0.68
Abs. difference, s	0.05	0.02	0.08	0.01	0.01	0.04	0.09	0.06
Rel. difference, %	2.3%	1.0%	4.3%	0.6%	0.7%	2.9%	8.0%	8.1%

4 Measurement after treatment and results

Measurements were repeated after the installation of the acoustical treatment at the same locations as previously. During these measurements, the temperature was 21°C, humidity 37%. Minor changes had been introduced – the sound system (as described in the Section 1) was installed in the room, a large wooden table and a leather chair were placed in front of membrane absorbers (Figure 3).

The new environment resulted in a noticeable increase in difference between modelled and measured T30 (Table 4). Difference between 8-10 % is observed at 125, 250 and 8000 Hz octave bands. Larger differences of 18-33 % are observed for low frequency 63 Hz, mid frequencies 500 – 2000 Hz and high frequency band 4000 Hz. An underestimation of total absorption area of acoustical solutions may be in place.

Figure 3. View on two opposite walls. To the left – wall membrane absorbers and furniture, to the right – L-ISA system loudspeakers.

Table 4. Reverberation time T30 measured and modelled after integrating solutions in octave frequency bands [f, Hz] and respective difference percentage.

T30, s	63	125	250	500	1000	2000	4000	8000
Measured	1.52	0.98	0.74	0.57	0.54	0.54	0.49	0.39
Model	1.13	0.9	0.8	0.74	0.72	0.66	0.58	0.43
Abs. difference, s	0.39	0.08	0.06	0.17	0.18	0.12	0.09	0.04
Rel. difference, %	25.7%	8.2%	8.1%	29.8%	33.3%	22.2%	18.4%	10.3%

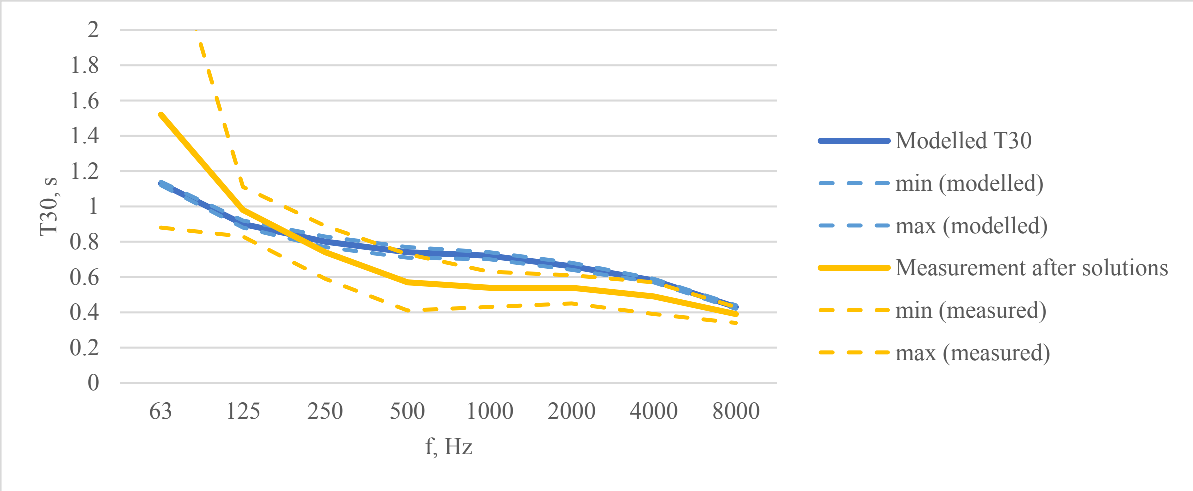

Figure 4. T30, modelled and measured.

There is no significant difference in amount of absorption materials (i.e. absorption area *A*) between the numerical model and real situation. However, additional furniture, loudspeakers and membrane panels add to overall scattering of the sound waves in the room, which 1) creates additional absorption due to scattering; 2) scatters the sound field in vertical direction, which results in more sound energy absorbed by the ceiling panels. It is assumed that these factors are the main reason for discrepancy in reverberation time between the model and the real situation.

Interesting observation is done for the reverberation time T30 values at 63 octave band. The difference between the modelled and real situation has the highest absolute value of 0.39 s across the frequency spectrum, which partly shows how geometrical methods are not valid for low frequencies, as anticipated (Eq. 1). It is assumed that due to pronounced modal nature of the sound field at 63 Hz frequency band, the limited area of membrane absorbers is not as effective as the geometrical model predicts. This is especially relevant for relatively small spaces.

One other factor to explain worse than expected performance of membrane absorbers is that the contractor has filled the entire inner volume of membrane panels with mineral wool, by mistake. This has introduced damping to the front panel, which reduces the absorption resonance peak, making it wider and lower.

Figure 5 shows the musical clarity C80 (average between 500-2000 Hz octave bands) as a function of distance from a sound source. The slopes of C80 functions of distance in each condition are similar, which shows that the diffusion of the sound field has remained mostly unaltered. On average, C80 has increased by 10 dB compared to the condition without acoustical treatment. The slopes shown in Figure 5 are linear, with an average R-squared prediction score of 0.71. The exponential model fitted to the C80 data shows an average R-squared prediction score of 0.77. This means, that the exponential model describes the data more precisely that a linear model, although not significantly.

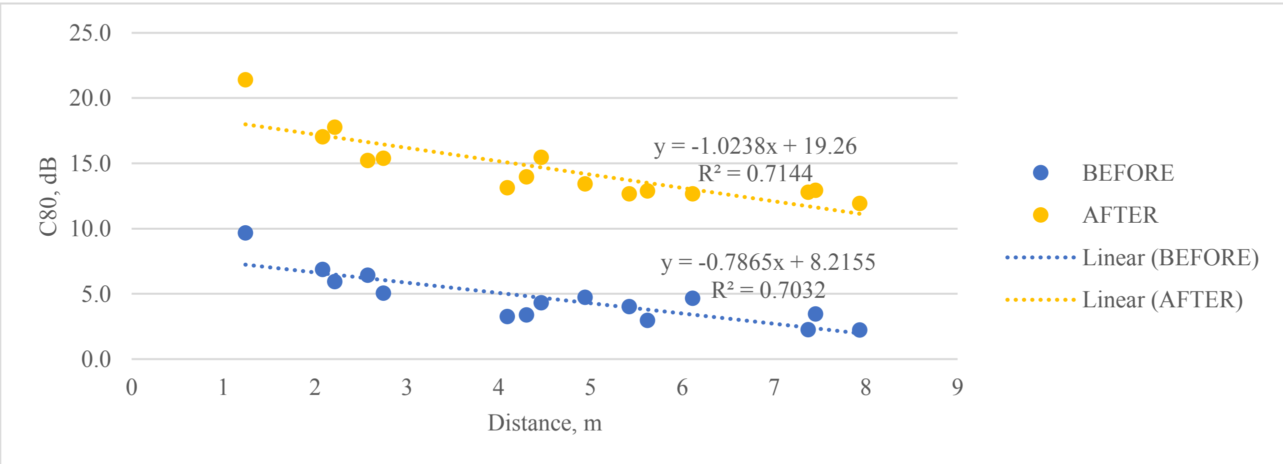

Figure 5. Musical clarity C80 (500-2000 Hz average) as a function of distance from the sound source.

5 Conclusions

A case study of L-ISA sound system showroom acoustics has been performed. Initial room acoustics measurements in a gentrified empty hall have been made, which have shown unsatisfactory reverberation time of approximately 1.5 seconds, unapproriate for the showroom purposes. Acoustical solutions have been designed to reduce RT down to 0.9 s or more and increase musical clarity C80.

Acoustical solutions were custom designed separate membrane absorbers, installed on the wall opposite to the main system speakers in a random fashion and mineral wool suspended panels on the ceiling. Membrane absorbers were intended not only for low frequency absorption but also for introduction of low frequency scattering. Solutions were calculated in the *Odeon* model of the showroom.

The second round of measurements in the showroom after the acoustical treatment have shown that there was overestimation of absorption effect for the 63 Hz octave band and underestimation for mid and high frequencies from 0.06 to 0.18 seconds and 0.1 seconds on average between 125 Hz and 8000 Hz. Nevertheless, the resulting acoustical conditions turned out to be satisfactory and comply with the RT target set by the client.

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Acoustics in a modular operating theatre

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Construction practices change. Modular construction is increasing, and we see it in the healthcare sector. Several factors contribute to this: reduced building time, convenience, reduction of waste, and less pollution on-site. It can also serve as a temporary solution with all the technical and hygienic demands met as in a traditionally built operating theatre.
Working conditions in Healthcare can include challenging acoustical issues. For modular constructions, either being used for laboratories or operating theatres, acoustic comfort is not always addressed or considered by modular suppliers. In the traditional construction way, acoustic comfort and its impact on staff and patients have been well studied. Nevertheless, comfort and work quality are equally affected, especially in sensitive specialist areas in new modular buildings, whether they are temporary or permanent modular constructions.
The objective is to present feedback on the modular construction of an operating theatre in Trollhättan, Sweden, which was constructed in merely 6 months following a technical issue with existing operating rooms. These modular rooms are temporary. The customer required the same level of acoustic comfort as in the existing operating rooms so that the medical staff could have the same supportive work conditions with a well-functioning sound environment.
Acoustic measurements were made in the modules to assess whether they comply with the room acoustic regulations in SS 25268. A survey was also conducted with the staff to evaluate their experience with these temporary rooms.

1 Introduction

Noise levels have steadily increased in Specialist care in hospitals over the last 40 years, both daytime and nighttime [1]. In operating rooms, sound levels can be very high. It is important to differentiate noise generated by equipment, alarms, background noise, and conversational noise to make informed decisions on how to improve the sound environment.

The term operating theatre in this paper refers to the complete operating department including the adjacent rooms for setting up medical instruments and preparing patients. An operating room is the room where the operation takes place.

1.1 Sound levels and peaks

In France, Marie-Hélène Landreau analysed in her report [2] the noise in operating rooms especially in an orthopaedic sector in a Parisian hospital. The two rooms were used for the same type of surgery. In conclusion, in the first room, the sound pressure level was between 75 and 90 dB(A) with peak levels at 100 decibels when using a saw and drill. For the second room, the values registered were between 58 and 77 dB(A) with peak levels at 95 decibels.

Kracht, Busch-Vishniac and West's study [3] investigated noise in the operating rooms at Johns Hopkins Hospital. This study monitored the sound pressure level before, during, and after operations. Orthopaedic surgery had the highest sound pressure level, measured in LAeq at approximately 66 dB(A). Neurosurgery, urology, cardiology, and gastrointestinal surgeries followed closely ranging from 62-65 dB(A). For neurosurgery and orthopaedic surgery, peak levels exceeded

100 decibels over 40% of the time they occurred, and peak levels were over 90 decibels during 90% of the time. The minimum noise without equipment and activity was at least 50 dB(A), mainly due to the ventilation system.

1.2 Consequences and cognitive ability

According to the INRS, a Health & Safety French institute, noise affects our cognitive ability. A sound pressure level for focused work should be below 55 dB(A) [4]. Sound pressure levels in operating rooms may affect the ability of the staff to perceive proper oral instructions.

High sound-pressure levels also affect the staff's perceived stress and well-being. One study from Dholakia [5] has specifically analysed the association between noise and surgical-site infection in day-case hernia repairs. In this study, the background noise was 47,6 dB(A) and the sound level pressure during the operation could increase by up to 10 dB(A). Among 64 patients, 5 patients have developed an infection and it was found that it corresponds to the operations with the highest noise level.

Medical errors due to noise are a safety concern that needs to be prevented in all possible ways. Staff report that they find it difficult to hear what is being said properly. Misunderstandings due to reduced intelligibility are a common problem [6]. It can also be difficult to determine the origin of alarm signals from the technical devices due to the reflection of the multiple soundwaves. Assessment of distance and direction is very well developed for the human ear in a setting without disturbing reflections, but it will be negatively affected in a confined space with multiple sound sources.

1.3 Reducing noise

Reducing noise requires a combination of actions: the use of sound-absorbing materials to deal with excess sound energy and unwanted reflections, alteration of human behaviour, setting of alarms, and functional sounds by med-tech devices and equipment [7]. Sound-absorbing material can improve the sound environment and the effect is valuable on both staff work environment, wellbeing and stress as well as the patient experience of received care [8].

It is important to differentiate between useful sound and unwanted noise. Even if operation rooms are considered noisy areas, it does not mean that all sounds must be deleted. Sounds and conversations are important and a natural part of the work in the operating room and alarms alert the medical team in case of an emergency or change in condition. However, some sounds are disturbing the work or can affect the quality and the intelligibility of the conversations. Unwanted sounds could be avoided by having good acoustic treatment. A sound-absorbing solution will capture and lower the sound-pressure levels, shorten the reverberation time and improve speech intelligibility.

2 Acoustic Standard

The acoustic standard in place at the time of building was SS 25268:2007+T1:2017 Acoustics – Sound classification of spaces in buildings – Institutional premises, rooms for education, preschools and leisure-time centres, rooms for office work and hotels [9].

It states that the reverberation time in the operating room should be ≤ 0,6 seconds. The arithmetic mean value of octave bands 250-4000 Hz may be exceeded by no more than 0.1 second in individual octave bands. At 125 Hz it should not exceed 0,8 seconds. The maximum noise level from technical installations (those over which the user does not control) in an operating room is 35 dB(A) and 55 dB(C) [9].

3 Case study

3.1 Context

The Northern Älvsborg County Hospital (NÄL) in Sweden is an emergency hospital that conducts almost 9000 planned and emergency operations every year. In 2021, a new modular constructed operation department was built.

On the existing premises, which have been in use for about 30 years, there have been repeated roof leaks in recent years, which eventually became an acute problem. This meant closing one of the operating departments for a long time since the renovation process takes several years. Therefore, they wanted to solve the leakage but also avoid closing too many operating rooms. A quick alternative was needed to keep operations up and running. The solution involved temporary replacement facilities in the form of modular operating rooms. The new modules needed to fulfil all the requirements used for the old rooms including acoustic treatment, hygiene requirements, advanced ventilation, medical technologies, and provide good working conditions for the staff.

The existing operating rooms had a very good sound environment, complying with the Swedish standard 25268:2007+T1:2017 [9] where the reverberation time is lower than 0,6 seconds.

One of the advantages of building with a modular solution is that the building and construction time is heavily shortened, and a fully functioning solution can be up and running in less than a year. For this specific project, it took 6 months from start to finish. As much of the construction work is done in the factory the disturbances towards ongoing hospital work can be reduced, both regarding the length of the time but also regarding disturbing noise and construction dust. Very often refurbishment or extensions of operating facilities are close to the ongoing daily work at the existing operating rooms and effects the work within them for a long time.

When the original operating rooms have been renovated there will be two options for the future of modular operating rooms: either the hospital keeps them, or the full module will be moved and installed in another hospital. With the many advantages and speed of delivering a fully functional temporary solution, we can expect modular facilities to increase in presence and it is important to gain knowledge of the acoustical consequences of this sort of construction.

The image below shows the layout of the operating rooms and adjacent supporting areas within the modular construction (Figure 1). Acoustic measurements were conducted in the upper right blue operating room.

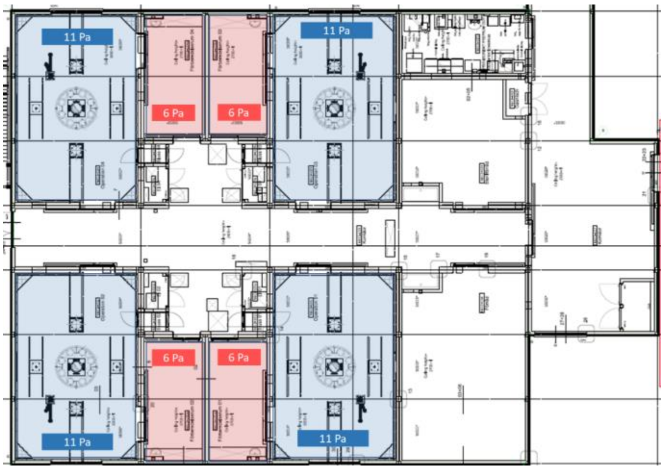

Figure 1: Operating theatre layout

3.2 Design of the rooms and material choices

General aspects

In this project, the customer chooses a complete package solution for the operating rooms with a modular solution that enables great flexibility in future setups. All the technologies and displays on the wall surface are integrated into the module unit. This gives maximum use of the surface and is functional and hygienic.

The operating room has a floor area of 50 m^2, the outline of the room is rectangular but the corners have been designed to make room for ventilation creating eight corners instead (see Figure 1). The wall partitions are made of High-Pressure Laminate (HPL) with 0.2mm lead integrated. The floor is linoleum, and the ceiling is by default a modular suspended HPL ceiling. All operating rooms are equipped with sliding doors, there are no windows. A separate airlocked transfer for equipment has been installed to avoid door opening and air contamination. The visual impression of the room is beautiful and perceived as pure and aesthetic.

Acoustic design

However, there is a disadvantage of common modular solutions: floors, walls, and HPL ceilings adversely affect the sound environment as the absorption of sound is more or less non-existent. According to Swedish Standard 25268:2007+T1:2017, the maximum reverberation time in operating rooms is ≤0,6 seconds [9]. To create a good sound environment and meet the standard, it was decided to use a ceiling with sound-absorbing qualities corresponding to Absorption Class A [10]. In acoustic performances, absorption Class A is the highest class and has a very good ability to absorb sound energy. In addition, to acoustic qualities, the ceiling also had to meet the hygienic requirements. Air particle levels should comply with ISO 5 according to ISO 14644-1:2015 [11]. The maximum level of Colony Forming Units (CFU) was <10 in all areas of the room.

The product chosen was a glass wool mineral absorbent with a thickness of 40mm. This is a comprehensive ceiling solution except for the advanced ventilation, pendants, and lighting, giving coverage of approximately 80-85% of the total ceiling area. No absorbents were mounted on the wall. The absorbers were assembled in suspended grids with clips securing the tiles from above. All operating rooms and preparation rooms were treated with sound-absorbing tiles and in the adjacent areas and corridors the same ceiling tile was used but with 20mm thickness instead of 40mm. The tiles fulfil all cleaning requirements (methods, cleaning agents, and disinfection according to national practice of Specialist areas).

3.3 TcAF ventilation technology

The technology behind Temperature controlled Air Flow (TcAF) is based on the ventilation system delivering HEPA (high-efficiency particulate air)-filtered and slightly cooled air into a zone around the operating table. By taking advantage of the fundamental laws of nature, TcAF breaks the convection currents in an effective and energy-efficient manner. Since cool air is denser than the surrounding warmer air, it drops toward the floor. The airspeed is dictated by the temperature difference in the room and a temperature difference (ΔT value) of -1.5 to -3°C is required between the ultra-clean air and the ambient room air at the operating table to guarantee a fall speed of about 0.25 m/s at the operating table. The system enables reliable and stable control of air movements, and thereby also the airflow's fall speed over the patient and the sterile-clad staff. The technology reduces the presence of bacteria-carrying particles in the operative zone.

3.4 Acoustic measurements

3.4.1. Reverberation time

For operating rooms, the reverberation time is set to a maximum of 0,6 seconds and this requirement refers to furnished rooms. The arithmetic mean value of octave bands 250-4000 Hz may be exceeded by no more than 0.1 second in individual octave bands. At 125 Hz it should not exceed 0,8 seconds [9].

The measurement in modular operating rooms showed a mean of 0,5 seconds and 0,6 seconds at 125 Hz (T20) (Figure 2). According to the measurement, the room complies with the Swedish standard. The mean of 0,5 seconds is 0,1 less than the maximum limit and it represents a decrease of 16%. A decrease of 5% in reverberation could be perceived in terms of comfort [12]. In our case, the staff was accustomed to good acoustics in the previous operating rooms and one of the goals of the acoustic treatment was to create the same high standard in the modules. The new modules are furnished to the same extent as the old operating rooms.

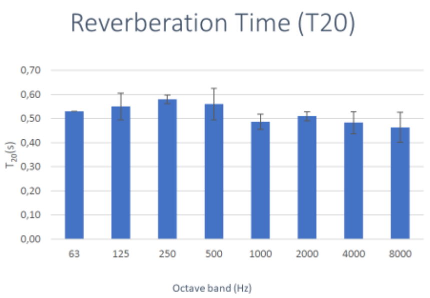

Figure 2. Reverberation Time, operating room

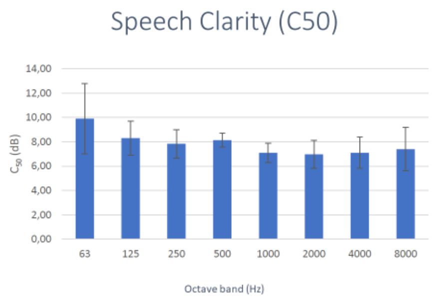

Figure 3. Speech Clarity, operating room

3.4.2. Speech clarity

Speech Clarity, C50, is a measure of the distribution of sound energy between early and late sound reflections. The reflections that reach the listener less than 50ms after the direct sound are counted as early, others as late. C50 is presented as a decibel value, and it is not included in any regulation but is of interest to better understand how staff will experience the sound environment and how well they will be able to understand spoken words.

A higher value means better speech clarity. The mean value for C50 (63 - 8000 Hz) was 8 dB (Figure 3). As there are no recommendations for C50 in operating rooms we compare the values to the findings of J. Harvie-Clark et.al in classrooms, where an 8 dB value falls between Good and Excellent [13].

In the operation room, the staff is wearing a facemask which is a barrier to good communication [14]. It is therefore even more essential to have a very good intelligibility. The staff is also restricted in how to move around between the three zones in the operating room and cannot always adjust the position to hear spoken communication better.

3.4.3 Speech Transmission Index

The Speech Transmission Index (STI) measures how well speech information is transmitted from speaker to listener. Unlike C50, STI considers the background noise in the room and the distance between the speaker and the listener. As there are different zones in the operating room, it is of interest to measure how well spoken words will travel in the room between the furthest away groups of staff. Measurements were made at 2,9 m.

Measurements show an STI value of 0,53. This is a relatively low number, possibly due to high background noise. In these rooms, a reasonable expectation could be that speech intelligibility be at minimum Good (0,60<STI<0,75) or Excellent (STI>0,75) [15] to achieve good quality and comprehension during a conversation.

The background level, LAeq & LCeq, is sound from, for example, permanent technical installations and ventilation. The results showed 41,1 dB(A) and 55,2 dB(C). According to SS 25268:2007+T1:2017 [9], the maximum noise level from technical installations (those over which the user does not control) in an operating room is 35 dB(A) and 55 dB(C). The measurements reported were not carried out to accurately evaluate sound from technical installation but can be seen as indicative. If we refer to the Swedish standard, the LAeq is too high and the LCeq is just beyond the maximum value.

The acoustic ceiling has a good impact on the acoustics of the room as a first step of acoustic design. Additional wall absorption can be considered. It is of interest to know how to lower the background noise level, and the ventilation system could be investigated further to see how much impact it has on the overall background noise.

3.5 Staff survey

Questionnaires were carried out and valuable feedback was received from the staff, in total of 21 answers. It cannot be interpreted as statistical figures but as a tendency regarding the topic. The survey includes anaesthetic-, surgical- and other medical staff in the operating room where 84% have more than 2 years of experience.

80% of the staff believe the sound environment can affect the ability of the staff to do a good job and 90% estimate that it also affects their well-being. However, if the shift is stressful with many staff in the room, 65% of the staff think the sound environment could benefit from further improvement.

48% of the staff say it is easy to understand speech in the operating room and say that disturbing sounds come from functional noise (e.g. engine, fan) and operating noise (e.g. suction, saw) rather than alarm functions. The intelligibility could be improved by further acoustic treatment on the walls to complement the acoustic ceilings. The speech intelligibility could also improve if the background noise was lowered.

4. Summary

Creating a good sound environment in the operation theatre requires a combination of actions: usage of sound-absorbing materials to deal with excess sound energy and unwanted reflections, awareness and sometimes alteration of human behaviour, as well as settings and functional sounds on the med-tech devices. The advanced ventilation systems available on the market all contribute to the background noise even when only run in basic mode. Hopefully, they will develop further in this aspect.

Sound-absorbing material can improve the sound environment and has a positive impact on the staff work environment, well-being, and stress. Acoustic comfort should be addressed by modular suppliers as building regulations apply and the comfort and workplace quality need as much attention in these modules as in traditionally built operating theatres. The modules at NÄL comply with the building regulations regarding reverberation time and are perceived as equal to the well-functioning traditional operating rooms by the staff.

Acknowledgements

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Effect of furniture in reverberation time measurements

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The most common measure for room acoustical performance in ordinary rooms is reverberation time. To perform satisfactory measurements, it is often emphasized that the rooms should be furnished. However, rooms are often measured with sparse or no furniture since they are tested in mock-ups or during commissioning. This paper presents results from a case study of reverberation time in which a room changed systematically between furnished and unfurnished situations for a range of monolithic ceiling installations. It is shown that the unfurnished room had similar results independent of ceiling performance, whereas the furnished room showed a clear difference in performance. The results were replicated in a room acoustical model using the Treble software. Comparisons are made to other prediction methods using Sabine's formula and EN 12354-6 including Annex D.

1 Introduction

In the past decades, it has become more accessible to make room acoustical measurements, i.e. reverberation time measurements. There can be multiple reasons for making room acoustical measurements in connection to a building project, but they often lead to the same problem: The rooms are not furnished. When the measurements are not as expected, the producers of acoustical solution are sometimes blamed. Especially in cases of rendered ceiling applications, this has become a common claim. It is common practice to specify that the rooms tested for reverberation time need to be furnished appropriately [1]-[6]. Rasmussen investigate the regulatory requirements for classrooms in the Nordic countries and states the reason for furniture during reverberation time testing is to introduce scattering objects, that will force sound waves toward the ceiling [7]. Since reverberation time measurements are still conducted without furniture, it seems that this common knowledge has been lost.

In this case study a room was investigated with and without furniture for a number of cases where the acoustical behaviour of the ceiling was altered. The motivation has been to show the importance of considering measurement conditions including furniture. The measurement results are backed up with simulations of the room conditions.

2 Why is furniture needed when making reverberation time measurements?

The authors do not know origin of demanding furniture, but the need for furniture seem obvious from an acoustical point of view. Acoustics is a 3D behaviour stretched in physical range from mm-scale to tens of meters. This means that the acoustics of a room is heavily influenced by anything place in the room. Secondly the furniture also add needed diffusivity to the room by interfering with sound only bouncing between the walls. This leads to an increase of exposure of the absorbers in the room. Simple calculations models like Sabine's formula [8] relies on the sound field to be absolute diffuse, which means that such models cannot satisfactory be used when rooms are sparsely furnished. Also, this leads to an overestimation of the impact of significant absorbers like porous ceiling absorbers, since in a sparse furnished room the exposure of the absorber will be far less.

Annex D of EN 12354-6 [8] presents an interpretation where a room is split up in reverberation planes. A visual representation is given in Figure 1. This represents a tangible way of considering a rooms interior by letting the designer consider if there are obstacles and/or absorbers in every plane. For the case of a room with a sound absorbing ceiling

Figure 1 indicates, that planes x and y will interact with the ceiling, whereas plane z is not sharing any edges with the ceiling and therefore is not influenced by the ceiling absorber. If the ceiling absorber is the only one in the room and the furniture is sparse, this leads to the reverberation time of the plane z will dominate the overall reverberation time of the room. In principle, this leads to cases where the efficiency of a sound absorber is not detectable from a reverberation time measurements.

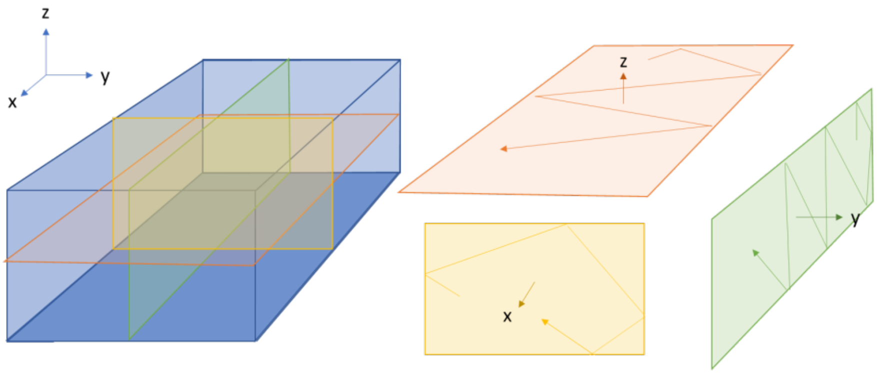

Figure 1: Visual presentation of reverberation planes as defined in EN 12354-6 [8]. The blue box represents the room surfaces, and the red, yellow and green surfaces represent the planes.

2.1 EN 12354-6 (Sabine's formula)

Sabine's formula is probably the most well-known equation in room acoustics. In the standard EN 12354-6, the use of the formula have been standardized with details on how to treat partial connected rooms and more. The formulation is repeated here for the sake of easy overview [8]:

$$T = \frac{55.3}{c_o} \cdot \frac{V \cdot (1 - \psi)}{A}, \qquad (1)$$

where T is the reverberation time, c_o is the speed of sound, V is the volume of the room, ψ is the object fraction and A is the sound absorption area. T and A are frequency dependent.

2.2 EN 12354-6 with Annex D

Annex D of EN 12354 [8] presents an interpretation where a room is split up in reverberation planes. In Figure 2 an overview of the surfaces is sketched with respect to the planes sketched in Figure 1. The approach considers first off only surfaces effect on each plane separately, e.g. plane x only considers surfaces parallel to the plane x, i.e. surfaces 'y = 0',' y = 1', 'z = 0', and ' z = 1' in Figure 2.

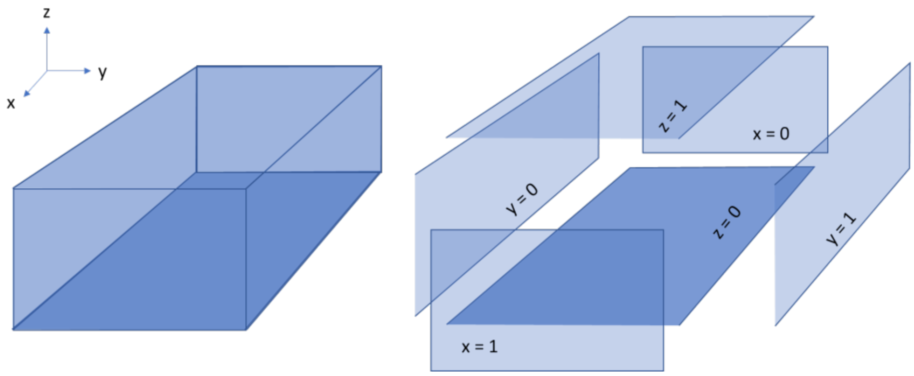

Figure 2: Visual presentation of surfaces as defined in [8]. The blue box represents the room surfaces 'x = 0',' x = 1', etc. in respect to the reverberation planes defined in Figure 1. I.e. the 'x = 0' surface is parallel to the plane x.

Secondly, the reverberation planes consider furniture either placed at a surface or in the "center" of a room. When furniture is positioned at a surface it only interacts with reverberation planes that interact with that surface, whereas furniture positioned in the "center" of a room interacts with all planes. In addition, the size of the furniture can be consider by removing part of the volume V.

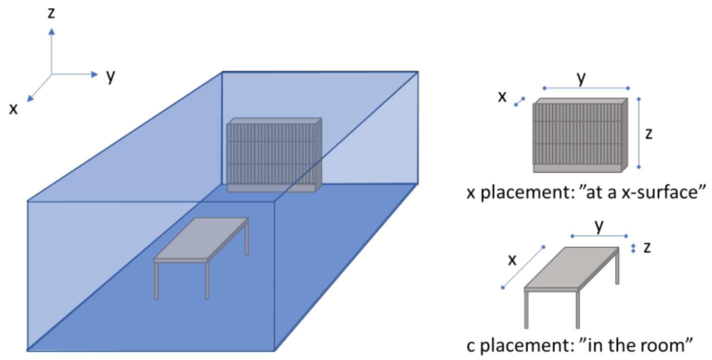

Figure 3: Visual presentation of a room with furniture as defined in [8]. The blue box represents the room surfaces and the grey drawing represent furniture with respect to the reverberation planes defined in Figure 1. I.e. the 'x placement' is placed at a surface parallel to plane x and 'c placement' is placed in the room and interact with all reverberation planes.

Thirdly, scattering and a transition frequency is considered. The scattering included in the calculation of this paper are set to the same for all surfaces, since it indirectly is included in the absorption coefficients. The values of the scattering and the associated equivalent absorption area they lead to can be seen in Table 1.

Table 1: Scattering coefficients and assosiated equivalent sound absorption area A used for all surfaces in the calculations

Frequency band in Hz	125	250	500	1000	2000	4000
Scattering coefficient δ	0.02	0.06	0.12	0.20	0.34	0.82
Equivilant Absorption area A in m^2	0.0	0.2	0.3	0.9	1.7	3.5

The transition frequency determines when the room "transitions" from low to high frequency. When calculating reverberation time for low frequencies all surfaces are consider and the reverberation planes considerations are ignored. Basically, T is found using Sabine's Formula in Equation (1) for the low frequencies. The transition frequency f_t of EN 12354-6 is found from the volume using the following equation [8]:

$$f_t = \frac{8.7c_o}{\sqrt[3]{V}} \tag{2}$$

The calculation included in this paper uses a f_t four times lower than what Equation (2) suggests. The change was made based of the measurement results.

2.3 Complex calculation models (Treble)

The second approach for acoustical calculations was using simulations using the Treble software. The software uses a hybrid between a wave-based solver based of the work by F. Pind [9] and geometrical acoustics solvers using ray-tracing.

For the low frequencies, Treble's acoustic simulation engine consists of directly solving the wave equation in the time domain numerically. This has the advantage of not introducing any further physical approximation to sound propagation, and inherently accounts for all wave phenomena relevant to room acoustics, such as wave interference, scattering, diffraction, modal behavior, etc. [10]

The geometrical raytracing method is a well-known procedure. Treble uses the image source method for modelling early reflections combined with the ray-radiosity method for modelling the diffused field. This method is well-known and is considered the industry standard approach for room acoustics modelling [11].

Accurate input data is paramount for effective simulations, particularly concerning furniture and other room details, which significantly influence reverberation time and sound wave propagation within a room. In Treble, attention to geometry details is crucial in achieving realistic predictions, especially when using the wave solver. For the wave-based solver, the level of detail in the geometrical 3D model is critical, with higher transition frequencies leading to a need for more geometrical detail. Additionally, boundary conditions for solving the wave equation relate to both geometry and material coefficients. On the other hand, for the geometrical acoustics solver, the diffusivity of surfaces and objects is included in the model by means of a scattering coefficient of 0.12 for the room's outer shell but more scattering for furniture for increased diffusion, which is standard practice since small interior objects are more scattering than smooth walls.

It turned out to be challenging to find results similar to the measured reverberation times using the same surface properties for both the unfurnished and furnished situations. The main uncertainty being the sound absorption values of the ceiling and estimating accurate scattering values. Treble's material database contains absorption coefficients based on impedance measurements. This ensures maximum accuracy due to the pressure-based nature of the wave-based algorithm. In this case neither the flow resistivity nor impedance of the material was known and the absorption values therefore based on reverberation chamber measurements which are not as accurate and can be influenced by the measurement setup. The absorptive effect of the wall absorbers of the furnished case also had significant effect on the horizontal sound field which made it challenging to replicate the scattering effect of the chairs and table in detail.

3 Test room

The test room used for the case study is a basement room at the Rockfon office in Denmark. The room has no windows, three masonry walls, one gypsum wall and only one door. There were four cases of comparative situations where only the ceiling was altered besides having furniture or no furniture. The volume of the room was $V = 4.60 \times 4.65 \times 2.35 = 50.3$ m^3.

The furniture were two large wall absorbers, a table and 4 chairs. The wall absorbers were a 50 mm thick with a 40 mm stone wool core and textile finish. They were 1.8 m × 1.2 m and 2.4 m × 1.2 m with a total surfaces area of approx. 5 m^2. The table was a round table of painted wood with a diameter of 1.3 m. The chairs were of plastic and steel.

A sketch of the room including measurement and source positions are shown in Figure 4. The total absorption area from absorbers, table and chairs are summarized in Table 1.

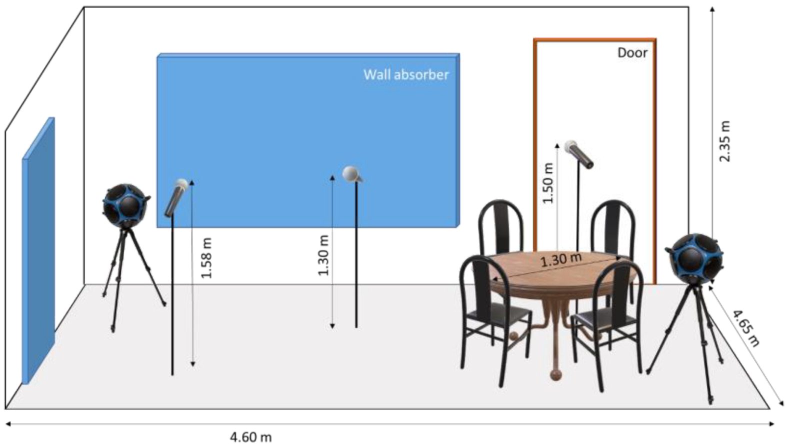

Figure 4: Sketch of test room including furniture and indications of source and measurement positions.

Table 1: Total equivalent sound absorption area, A, for wall absorbers, table, and chairs.

Frequency band in Hz	125	250	500	1000	2000	4000
EN 12354 calculations, A in m^2	2.1	4.3	5.9	5.9	5.9	5.8
Treble simulaitons, A in m^2	1.4	2.3	3.4	4.9	5.1	5.0

4 Measurement results

Measurements were conducted according to ISO 3382-2 [12] with the same source and measurement positions every time. The four comparative cases are summarized in Figure 5. The cases consist of the situations where stone wool slabs with rendered finish or unfinished (e.g. '40 mm Mono slabs + filler + render std.' or 'Only 40 mm Mono slabs', respectively). The effect of rendering is treated in conference paper [13] and is therefore exclude from this paper.

The measurements show a large effect of the furniture on reverberation time. In the midrange frequencies from 500 Hz to 2 kHz, where flutter echoes are expected to be possible, the reverberation times are more than 2 times longer in the non-furnished case than in the furnished case. This indicates that flutter echo effects in the horizontal plane are dominating the overall results when the room is not furnished. It can be concluded, that the furniture removes the flutter echoes by breaking up the dominating horizontal plane significantly.

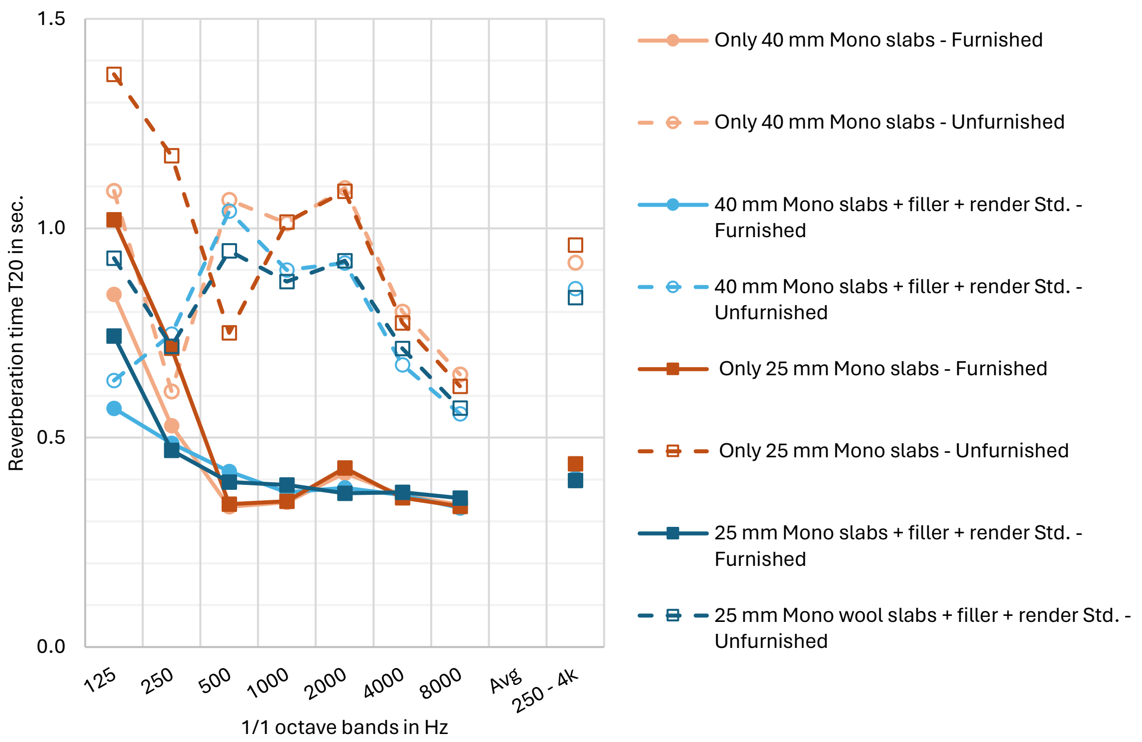

Figure 5: Measurement results for four comparative cases – meaning eight measurements results in total.

5 Calculation results

Calculations using the EN 12354-6 with and without Annex D were made along with simulation using the Treble software. Figure 6 show a comparison between the calculation methods and one case of the measurement comparative cases.

The calculation results for the furnished case show that both the Treble software and the EN 12354-6 incl. Annex D approach can replicate the measurement results within 0.1 second. The Treble software show the closest correlation with the measurement results. Sabine's Formula underestimates the reverberation time in all bands, but most significantly at lower frequencies.

For the non-furnished cases the effect of the missing furniture is significant in the results from the Treble software and the EN 12354-6 incl. Annex D approach, whereas it is not seen in the Sabine's Formula calculations. The difficulty of finding transition frequency is also clear from the Treble software where the 500 Hz band is significantly underestimated compared to the measurement results. It should also be noted that the sound absorption of ceiling is significantly different in the two models. The effect of the furniture seems to be more significant in the Treble model, which could be due to sound absorption values in general are set too high. Details on the sound absorption coefficients used in the models can be found in [13].

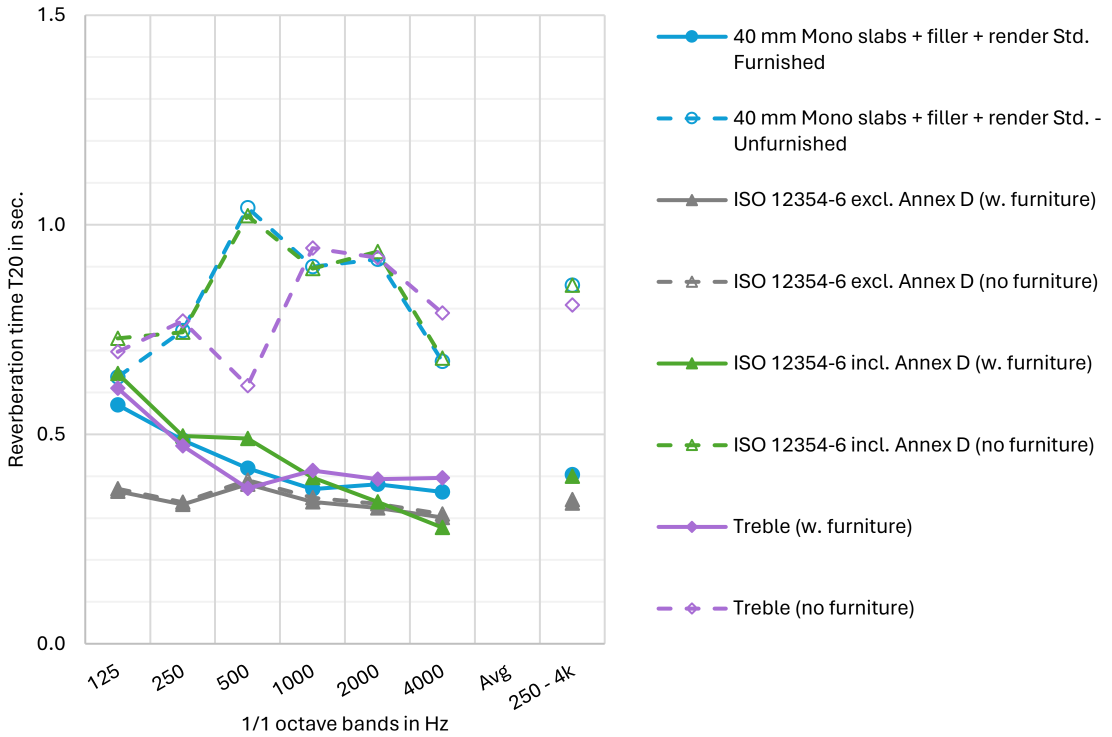

Figure 6: Measurement and calculation results for one comparative case with and without furniture. Calculations are made using the Treble software, the EN 12354-6 incl. or excl. Annex D, where excl. Annex D is using Sabine's Formula, cf. Equation (1).

6 Conclusions

Measurements of reverberation time were performed in a room with and without furniture. The measurement showed more than 2 times longer reverberation times in some frequency bands between the two situations. The calculations to replicate the measurement results were made using Sabine's Formula, EN 12354-6 extended calculation method using Annex D, and a wave-base and geometrical acoustics solver using the Treble software. Calculations results using Sabine's Formula was found to be uninfluenced by the added furniture, whereas the EN 12354-6 with Annex D and the Treble software could replicate the effect of furniture. It was also noticed, that it was not trivial to replicate results, especially for the non-furnished case, where the transition frequencies between modal and non-modal behaviour of the room for both elaborate models was found tricky to set right.

7 Acknowledgements

The authors kindly thanks Rockfon for the support during this project. Especially Per-Henrik Andreasen for performing the reverberation time measurements and Carsten Christensen for organising the investigations at the Rockfon office.

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Repairs on rendered sound absorptive ceilings and the effect on their acoustical performance

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Monolithic rendered ceilings with sound absorption properties have been around for more than a decade. Since their introduction to the market, questions have been asked about the sound absorption abilities, if they need repair or a "freshening up". In this paper, a case study is presented, where a freshening up can be achieved by adding additional layers of render. Investigations on the effect of this on sound absorption coefficients are done based on reverberation time measurements and backwards deducted to the best of the authors abilities using a spread sheet implementation of EN 12354-6 including Annex D and the Treble software. Complications when doing this are outlined and a discussion of alternative approaches to determine the absorption coefficient of installed rendered ceilings are included in the paper.

1 Introduction

For many years, high levels of sound absorbing ceiling and smooth monolithic visual was impossible to combine. First, efforts were made to make ceiling tiles more monolithic looking by adding fleeces before painting and creating grooved edges to conceal the supporting grids.

Then came the rendered solutions. At its very basics, the solution is a mineral wool slab, that is sprayed with plaster. But to make it work, the importance of the plaster type and skill it takes to instal in the right way must be emphasized. A second issue is the finish, that needs to be very smooth to ensure no shadowing from gracing light.

After this comes the issue of use. A ceiling is over time used for hanging light and other things. Maybe something hits it and make a discolouring or similar. All in all, this leads to need for freshening up by adding more plaster on the ceiling. This paper will look into the effect of this by estimating the sound absorption coefficients indirectly in a case study. This approach for estimating sound absorption of a ceiling is often used badly with unfurnished rooms, to claim that the plastered ceilings are not working. A closer look into the unfurnished situations can be found from [1]. For this reason, this paper share a few paragraphs with [1].

The case study included in this paper only considers furnished situations. Three situations are included:

- Normal finished solution
- More render than usual (around 80 % more)
- A lot more render than usual (around 120 % more)

All situations are on 40 mm mineral wool base, for which joints and fixations point have been plastered to ensure a smooth finish. Some stone wool slabs have airtight membrane on the back towards the ceiling called HPM. The slabs in this case study do not have this membrane.

2 The approach for estimating sound absorption

Two ways of estimating sound absorption indirectly were used: EN 12354-6 incl. Annex D [2] and using the Treble software. The generally approach in both cases was to create a model and chance only the sound absorption properties of the ceiling to fit the reverberation times of the models to the measured reverberation times. Both calculation tools are described in more details in [1].

2.1 Alternative approaches to estimating sound absorption

In reason years more in-situ methods to measure sound absorption of installed solutions have been introduced. These include Microflown and SonoCat applications, that can be used to scan a surface and give an indication of the sound absorption coefficient in the normal directions. Both of these applications are still new in market and usage applications have not yet been standardized, but they can with success be use determine whether a surface is sound absorptive or reflective, whereas the direct comparisons to sound absorption coefficients properly should be avoided. C. Nocke have looked into using a transfer function method to measure the surface impedance in free-field [3]. This method is closely related to the measurement method in the road surface standard ISO 13472-1 [4], but a commercial setup for indoor use is still not available. The authors have also seen attempts to use the road surface method ISO 13472-2 with questionable success [5].

3 Test room

The test room used for the case study is a basement room at the Rockfon office in Denmark. The room has no windows, three masonry walls, one gypsum wall and only one door. The volume of the room was $V = 4.60 \times 4.65 \times 2.35 = 50.3$ m^3.

The furniture were two wall large absorbers, a table and 4 chairs. The wall absorbers were a 50 mm thick with a 40 mm stone wool core and textile finish. They were 1.8 m × 1.2 m and 2.4 m × 1.2 m with a total surfaces area of approx. 5 m^2. The table was a round table of painted wood with a diameter of 1.3 m. The chairs were of plastic and steel.

A sketch of the room including measurement and source positions are shown in Figure 1. The total absorption area from absorbers, table and chairs are summarized in Table 1.

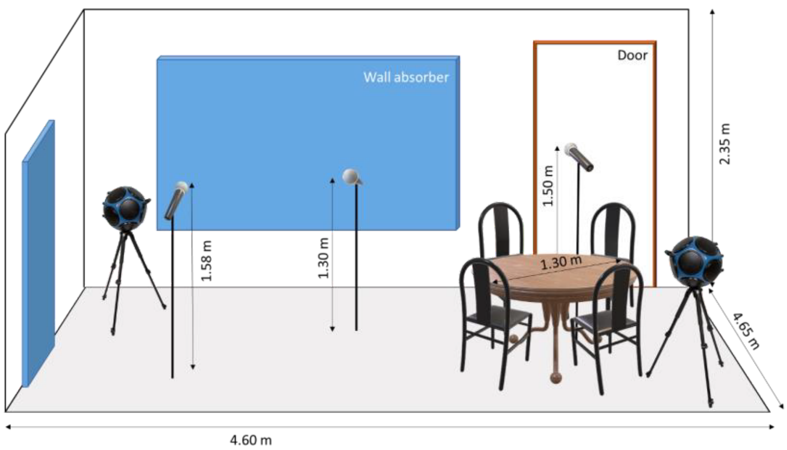

Figure 1: Sketch of test room including furniture and indications of source and measurement positions.

Table 1: Total equivalent sound absorption area, A, for wall absorbers, table, and chairs.

Frequency band in Hz	125	250	500	1000	2000	4000
EN 12354 calculations, A in m^2	2.1	4.3	5.9	5.9	5.9	5.8
Treble simulaitons, A in m^2	1.4	2.3	3.4	4.9	5.1	5.0

4 Measurement results

Measurements were conducted according to ISO 3382-2 [6] with the same source and measurement positions every time. The reverberation times found in the three cases are summarized in Figure 2. The cases consist of the situations of normal rendering, more than normal and a lot more than normal called ’40 mm Mono slabs + filler + render Std. Furnished’, ‘40 mm Mono slabs + filler + render Std.+ Furnished’, 40 mm Mono slabs + filler + render Std.++ Furnished’, respectively. The effect of furniture is treated in [1] and is therefore exclude from this paper.

Also included in Figure 2 is confidence intervals across the source and receiver positions at every frequency for each case. The uncertainty intervals were found using the standard error and a covering factor of k = 1.96 to find the uncertainty using 95% confidence interval. To create the intervals the uncertainty was then either subtracted from the average or add to the average. The calculation details for this procedure are summarized below in Equations (1)-(3).

$$s_e = \frac{u}{\sqrt{N}}, \tag{1}$$

where s_e is the standard error, u the standard deviation and N is the of source and microphone combinations, which were six in this case study.

$$k = 1.96 => U = 1.96 \cdot s_e\,, \tag{2}$$

where k is the covering factor associated with a 95% confidence interval and U is the uncertainty of the measurement.

$$T_{min,i} = T_i - U\ \&\ T_{max,i} = T_i + U\,, \tag{3}$$

where T_i is the reverberation time in the i’th octave band and $T_{min,i}$ and $T_{max,i}$ are the minimum and maximum reverberation times for the “true value” of the reverberation time.

The uncertainty intervals indicate how well the cases can be separated. E.g. even though the differences at the 250 Hz band is larger between the cases than at 4 kHz band, there is a larger uncertainty around the results 250 Hz. This means, that with an increase in source and receiver combinations the difference in the 250 Hz could change significantly compared to the 4 kHz results.

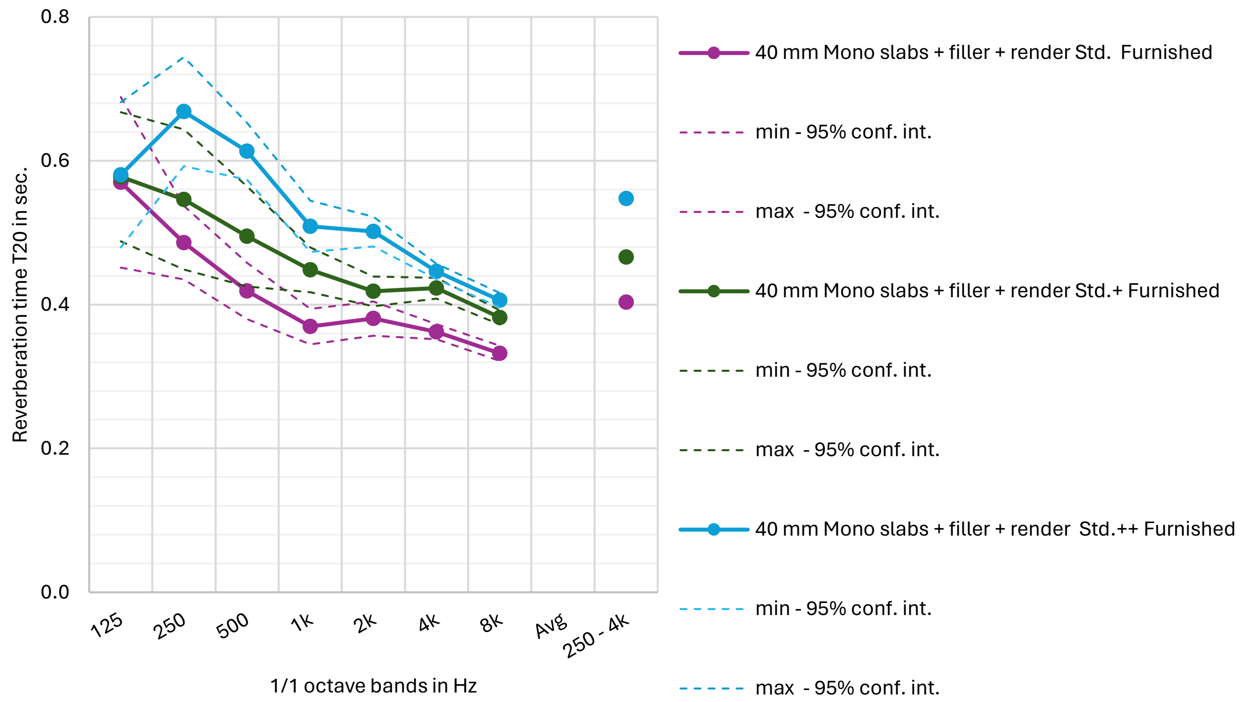

Figure 2: Measurement results for three cases of normal rendering (purple), more than normal rendering (green) and a lot more rendering (blue). Also include is the 95 % uncertainty intervals for each frequency found across all source and receiver combinations.

5 Calculations of reverberation time and estimations of sound absorption

This section includes the calculations of reverberation time using two difference calculation tools. First the calculation using the EN 12354-6 [2], that is based of the Sabine's formula. The implementation includes additions of Annex D of the standard, that considers that a room can have planar reverberation behaviours. In short, this means that the reverberation time can be dominated by the e.g. the reverberation between the walls only.

Secondly, calculations using the Treble software are presented in similar way. Treble calculates impulse responses and results in acoustic parameters like reverberation time and other acoustic parameters. For this paper, the methodology involves a reverse engineering approach, where the model is adjusted to match measurements. Including model geometry, material acoustic properties like absorption coefficient and scattering, and transfer frequency. It was found challenging to fit the same model for the range of setups, ensuring simulation results accuracy averaging within the measurement standard deviation. This was to some extent relate to the size of the room, since the transition frequency between the wave-base solver and the geometric acoustics solver moves up in frequency, when the room is small.

The calculation results can be found in Figure 3.

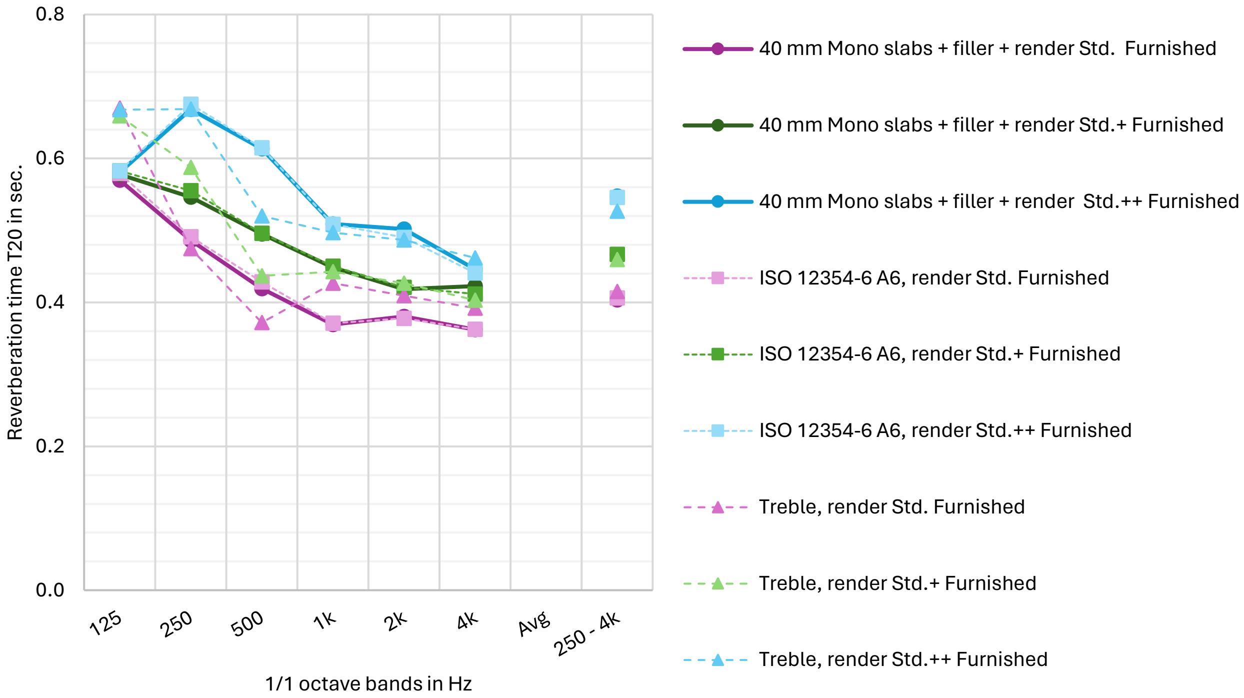

Figure 3

5.1 Estimation of sound absorption using the EN 12354-6 Annex D calculator

The EN 12354-6 including the Annex D **Error! Reference source not found.** approach have been implemented in a spread sheet, including a library of product values. The sound absorption coefficients used in the calculation are summarized in Table 2: Random incidence sound absorption coefficients, α_s, for wall, floors and celiling *Values have been altered to fit suspension height.Table 2 (excl. the furniture). The estimations of the rendered cases show that the sound absorption of the ceiling drops to about half of the initial values at the highest frequencies, whereas the lowest frequencies (125 Hz) seem almost unaffected. This could be partly down to the ceiling absorber acting a bit like a membrane absorber at the lowest frequencies. Also, empirical models like Delany-Bazley-Miki model predicts a slight increase in lower frequencies with an increase airflow resistivity [7], which is the expect result from adding more render.

Table 2: Random incidence sound absorption coefficients, α_s, for wall, floors and celiling *Values have been altered to fit suspension height. Values for *Ceiling rendered standard* + and *Ceiling rendered standard* ++ have been fitted to match reverberation time results.

		Sound absorption coefficient[1]					
Surface	Area m^2	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Concrete floor	21.4	0.02	0.02	0.02	0.02	0.02	0.02
Concrete walls	32.7	0.05	0.02	0.02	0.03	0.02	0.02
Door	2.0	0.25	0.20	0.15	0.08	0.08	0.07
Gypsum wall 2 x 13 mm w. insulation	8.8	0.12	0.10	0.08	0.06	0.03	0.03
Ceiling rendered standard	21.4	0.85*	1.00*	0.85	0.95	1.00	1.00
Ceiling rendered standard +	21.4	*0.80*	*0.80*	*0.65*	*0.65*	*0.70*	*0.70*
Ceiling rendered standard ++	21.4	*0.80*	*0.55*	*0.40*	*0.50*	*0.50*	*0.55*

Table 3: Sound absorption coefficients for wall, floors and ceiling using the Treble simulations. Values for *Ceiling rendered standard*, *Ceiling rendered standard* +, and *Ceiling rendered standard* ++ have been fitted to match reverberation time results.

		Sound absorption coefficient					
Surface	Area m^2	125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz
Concrete floor	21.4	0.02	0.02	0.02	0.02	0.02	0.02
Concrete walls	32.7	0.05	0.02	0.02	0.03	0.02	0.02
Door	2.0	0.24	0.20	0.14	0.09	0.08	0.07
Gypsum wall 2 x 13 mm w. insulation	8.8	0.12	0.10	0.08	0.06	0.03	0.03
Radiator	0.7	0.08	0.06	0.05	0.05	0.06	0.06
Ceiling rendered standard	*21.4*	*0.29*	*0.44*	*0.52*	*0.58*	*0.60*	*0.50*
Ceiling rendered standard +	21.4	*0.31*	*0.38*	*0.40*	*0.46*	*0.47*	*0.50*
Ceiling rendered standard ++	21.4	*0.35*	*0.31*	*0.30*	*0.29*	*0.29*	*0.30*

5.2 Estimation of sound absorption using Treble software

The absorption values used for the Treble simulations can be seen in Table 3, along with the estimations of the ceilings. The simulations show that adding more render generally makes the ceiling less absorbent.

The accuracy of the reverse engineering approach can be questioned since the absorption coefficients of other surfaces in the room are also estimated. E.g. it can be argued that if the other surfaces have too high absorption, this will lead to the rendered ceilings looking less absorbent. It is though worth noting, that the total amount of sound absorption area in the Treble model is about half of what is found in the EN 12354-6 with Annex D calculations. The EN 12354-6 with Annex D does not take into account the modal behaviour of the room which the Treble algorithm does. This can explain some of the differences at low-mid frequencies where the placement of absorption plays a great part in its overall effectiveness.

[1] Sound absorption coefficients as measured using ISO 354:2003

With that said, the general conclusion is similar to the EN 12354-6 Annex D calculations: The sound absorption of the ceilings seem to drop to about the half of the initial values for the highest frequencies, whereas the lowest frequencies seem unaffected.

6 Conclusions

The paper presents reverberation time measurements for three comparative situations: Ceilings rendered normally, more than usual, and far more than usual. The measurements show a clear effect on the reverberation time when additional render is added to the ceiling. The measurements have been replicated using the calculation method in EN 12354-6 including Annex D and the Treble software. The goal was to indirectly estimate the sound absorption properties of the ceiling, which is often done in cases of claims. The two calculation approaches are shown to be significantly different in the estimations. On the other hand, both methods agree on the propagation of the sound coefficients when adding more render. It can also be noted that small rooms, like the one investigated in this project, can be challenging to handle due to the needed accuracy in input values. This was especially true for the wave-based solver in the Treble software, that should excel in accurately simulating low-frequency phenomena in small rooms.

7 Acknowledgements

The authors kindly thanks Rockfon for the support during this project. Especially Per-Henrik Andreasen for performing the reverberation time measurements and Carsten Christensen for organising the investigations at the Rockfon office.

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Perceptual Evaluation of Room Acoustic Simulations and Measurements

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Acoustic simulation and auralization have proven useful for recreating characteristics of an acoustic space. Objective and perceptual measurements are crucial to validate and gain insights in the accuracy of the auralizations. While previous studies have performed perceptual evaluations of simulations based on geometric acoustics (GA), the development of a new software based on numerical methods (NM) makes it possible to investigate perceptual differences between these two types of algorithms. In this study two listening experiments are conducted to assess the perceptual differences between measurements and auralizations from three software based on either GA or a hybrid of NM and GA.
In the first listening experiment, 21 participants evaluated the similarities between a measured reference and the simulated auralizations. Neither simulation was found to be indistinguishable from the measurement. A second listening experiment, conducted with 10 trained expert listeners, evaluated the auralizations on six selected attributes: Treble Strength, Bass Strength, Clarity, Level of Reverberation, Localizability, and Width. All software introduced perceptual differences between the reference and simulations in multiple attributes. The results indicate that increasing the volume of the geometry will decrease the perceptual accuracy of the Level of Reverberation.
A multiple linear regression, incorporating data from both experiments, shows that Treble Strength, Bass Strength, Level of Reverberation, and Localizability introduce the most significant influence on the perceptual evaluation of similarity.
This research underscores the importance of considering both objective and subjective measures in assessing the effectiveness of auralizations in diverse applications.

1 Introduction

The application of acoustic simulation is expanding beyond its traditional domain, finding utility in architectural design processes, virtual reconstructions of historic buildings [1,2], as well as immersive gaming experiences and virtual reality applications [3]. These simulations enable the ability to recreate information about acoustic characteristics within a space, with the perceptual representation of sound known as auralization [4].
The two primary methods used in acoustic simulation are Geometric Acoustics (GA) and Numerical Methods (NM), the latter typically being named as wave-based methods. The choice of one over the other will essentially be a trade-off between calculation accuracy and computational efficiency. The emergence of hybrid acoustic simulation techniques, alongside advancing viability of NM with compelling results, shows the potential for physically accurate simulations while minimizing computational overhead [5].
Previous studies have evaluated the perceptual accuracy of different simulation methods based on GA [3], or GA compared to a hybrid of NM and GA [6]. The latter however acknowledge the necessity for more refined tests.

This study aims to evaluate the perceptual differences between measurements and simulations generated by the three different software applications: Odeon, Treble, and RAVEN, based on GA and a hybrid approach integrating NM and GA. The comparison entails assessing perceptual differences in the auralizations produced by the software applications and measurements through a similarity test employing the MUSHRA (MUlti Stimulus test with Hidden Reference and Anchor) framework [7]. Furthermore, an attribute test is utilized to characterize the perceptual differences.

By combining the outcomes of the MUSHRA and attribute test, the study explores how each attribute contribute to and impact the perceptual similarity between the measurements and simulated auralizations.

2 Method

2.1 Room Acoustic Simulation

Today, GA stands as the industry standard algorithm for acoustic simulation and is widely used for consultancy [8] due to its efficiency. GA is an approximate method using straight rays to predict sound reflection statistically, which is valid in spaces intended for music and speech, where the space dimensions are large compared to the wavelengths, above the Schroeder frequency, calculated by $f_S \approx 2000\sqrt{T/V}$ where T is the reverberation time (RT) in seconds, and V is the volume in m^3 [9].

In commercial software, GA simulation is typically based upon two common algorithms: image source (IS) and ray tracing method [8]. The approximate nature of GA neglects diffraction in the geometry [4,5], although some applications incorporate geometric diffraction as a statistical feature [10,11]. Nevertheless, diffraction modelling is still noted as a lacking feature in all GA methods [3].

For this study, two software based on GA are employed: the first being the commercial software Odeon Combined (version 17.15) developed by Odeon A/S, and the second being the open-source application RAVEN (Room Acoustics for Virtual ENvironments) developed at the Institute of Technical Acoustics, RWTH Aachen University [12].

In small spaces, the Schroeder frequency is higher, and thus the frequency contribution below this threshold is more significant, where energy-based GA gets less accurate [13]. This is where NM is a more accurate method [8]. NM approaches give more reliable results by accurately including acoustic phenomena such as diffraction [14], and interference [15]. NM methods numerically solve the wave-equation, yielding impulse responses (IRs) closely resembling measurement results [4], at the expense of heavy computational load.

The new software, Treble, developed by Treble Technologies, introduced an Acoustic Simulation Software offering three simulation models: using GA only, NM only (utilizing a discontinuous Galerkin Finite Element Method (dG-FEM)), or a hybrid combining NM for lower frequencies and GA for higher frequencies. Treble is the first commercial software hybridizing NM and GA [16], and a beta version of the software developer kit (SDK) was employed in this study.

2.1.1 Geometries and Boundary Conditions

Inaccuracy in defining boundary conditions is one of the biggest uncertainties in acoustic simulation [17], and unreliability from the input data has led to the simulations being rejected by some acousticians historically [8]. Therefore, two geometries from the Benchmark for Room Acoustic Simulation (BRAS) database were utilized to evaluate the acoustic software: a Chamber Music Hall (volume = 2,350 m^3, $f_S \approx 47$ Hz) and an Auditorium (volume = 8,690 m^3, $f_S \approx 29$ Hz). The database offers room models, source and receiver information, absorption and scattering coefficients, and measured IRs. A few corrections were made to apply correct materials, and audience chairs, which is of high importance for the NM simulation, were added to the models.

An additional room was measured by the first author to add an example where potentially more uncertainties are introduced. This space is a Classroom at the Technical University of Denmark (DTU) in building 352 with a volume of 182 m^3 and $f_S \approx 112$ Hz. This room was simulated by the Odeon and Treble software only.

Note that changes in acoustic simulation performance may be due to both boundary conditions and room size.

2.1.2 Source and Receivers

For the BRAS measurements a QSC K8 speaker and the FABIAN dummy head were used and inserted in the simulations. Although, for the Treble simulations an omnidirectional speaker was simulated instead, since the QSC K8 could not be adopted to the software in the timespan of the project.

For the Classroom measurements a Genelec 8020C speaker was used (source directivities can be found in the BRAS database) together with a B&K HATS 4100. The HRTFs were provided by the manufacturer and converted to the file formats necessary for Odeon and Treble.

2.2 Listening Experiment

Three samples were selected for each room to match the suited audio activity of each geometry while having sufficient sound energy to provide perceptual audio information. Samples played in the Chamber Music Hall were two classical and one jazz sample, the Auditorium presented Money for Nothing, a speech sample and a vocal track, and finally the Classroom utilized music only with Get Lucky, Six Blade Knife and Moonlight on Spring River.

In the listening experiments several variations of the simulations were evaluated. Odeon were simulating using the suggested number of late rays from the functions *engineering* and *precision* (the latter having > 10 times more rays), and Treble was simulating using GA only as well as the hybrid using a low and high transition frequency. Raven was simulated only once, resulting in a total of 6 simulated audio systems (5 for the Classroom since Raven was left out). In all geometries the reference was the measured IR convolved with a sample.

All IRs were temporally aligned to each other, convolved with the samples, and peak normalized to prevent clipping before saving as .wav, using MATLAB. Adobe Audition was employed to perform loudness adjustment to -23 LUFS (Loudness Unit relative to Full Scale), following the EBU R 128 standard [18], and adjust to matching signal lengths.

2.2.1 MUSHRA Test

The first experiment is designed to measure *authenticity*, meaning the degree to which a reproduction is perceptually indistinguishable from the reference [19].
The ITU-R BS.1534-3 procedure (MUSHRA), suited for subjective assessment of intermediate audio quality was used, but another attribute was chosen; The sensory measure *Difference* on a scale from very different to very similar, using scale labels similar to [20] and [21].
The test comprised of 1 known reference, 6 audio systems, 1 hidden reference, and 1 hidden anchor. The latter being a low-pass filtered version of the reference used for establishing a lower bound of the scale. Test subjects were asked to identify the hidden reference and rate it at the top of the scale, indicating that the system and reference were identical. The remaining systems should be assessed based on how different they were relative to the reference. An example is shown in Figure 1 below.

Figure 1: MUSHRA test interface in SenseLabOnline listening test software, showing the systems, scales, and description. The dark green dashed box marks the stimulus the assessor identified as the hidden reference, placed in the top, and the light green dashed box marks the anchor. The remaining systems (play buttons) hold a random order of the two Odeon simulations, the three Treble simulations and the Raven simulation (in the BRAS geometries).

The test was evaluated by 21 people, comprising of 5 women and 16 men (age median = 27 years, min = 22 years, max = 61 years), and none reported any hearing impairments.

2.2.2 Attribute Test

The second test will identify singular auditive changes within six selected attributes: Bass Strength, Treble Strength, Width, Localizability, Level of Reverberation, and Clarity. The attributes were selected to give an insightful sound vocabulary from the Spatial Audio Quality Inventory (SAQI)[1] and the Sound Wheel[2] [22] for describing the identified perceptual differences within timbral balance, spatial characteristics, and temporal behaviour. The 6 simulated audio systems and reference were evaluated according to each attribute by 10 ISO 8586-2-categorized expert assessors [23].

2.2.3 Listening Experiment Setup

All tests were conducted in sound-insulated listening booths at FORCE Technology and DTU using Sennheiser HD650 headphones connected to an RME Fireface UC soundcard. The gain of the soundcard was calibrated to give the reference signal a sound pressure level (SPL) of $L_{ref} = 78 \pm 0.25$ dBA, with an individual adjustment allowance of ± 4 dB.

2.3 Statistics

The Welch two-sample two-tailed t-test [24] will determine whether two systems are significantly different from each other or not. For p-values below 0.05 the null hypothesis is rejected, and a significant difference is confirmed.

A Multiple Linear Regression (MLR) is conducted with the results from the MUSHRA test as the dependent variable and with attribute evaluations as independent variables. Weightings assigned to each attribute will determine how they contribute to the overall subjective perception of similarity between simulations and measurements, similar to the studies presented by [25] and [26]. The results will be displayed in the Linear Model (LM) [24], assuming that the error is zero:

$$\bar{y} = b_0 + b_1 x_1 + b_2 x_2 + \cdots + b_n x_n \tag{1}$$

where $\bar{y}$ is the mean of the dependent variable, b_0 is an estimate of the y-intercept, $b_1, b_2, \dots, b_n$ is an estimate of the weight of the coefficients. The mean of the dependent variable is necessary for this analysis where the number of datapoints in the results from the MUSHRA and attribute test are not equal.

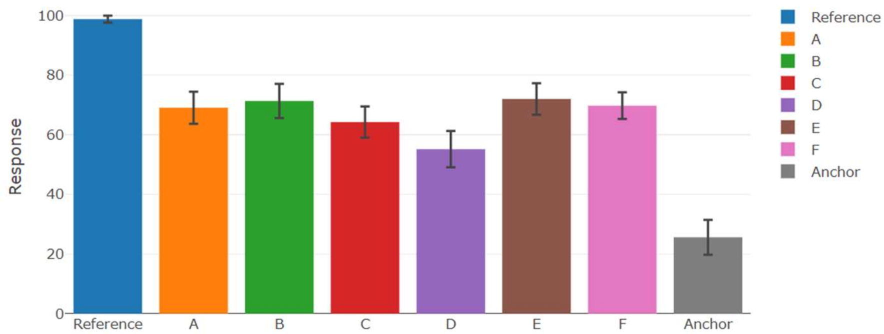
3 Results

In plots comparing the software, the letters A to F are assigned to each simulation to anonymize the software. In the following, *system* refers to the different sound stimuli in the test, including both the reference and the simulations.

3.1 MUSHRA Test Results

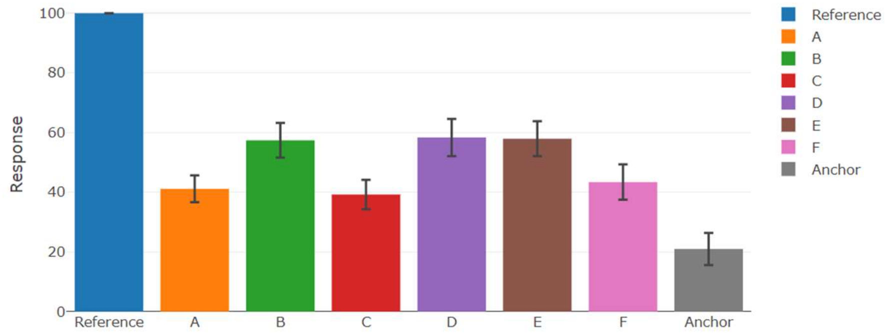
The data from the MUSHRA test is based on 21 participants and are shown in bar plots with means and 95%-confidence intervals. The plots in Figure 2 below reveal a high level of confidence in detecting the reference and anchor across all geometries, and no system is considered insignificantly different from the reference. Furthermore, the results are highly geometry-dependent.

The Chamber Music Hall in Figure 2 (a) reveals that system A, B, E, and F are rated higher than system C, with a rating of system D being “somewhat similar” to the reference. The Auditorium in Figure 2 (b) introduce the largest simulation differences with ratings divided into two groups where A, D, and E are generally rated higher than system A, C, and F. The results from the Classroom in Figure 2 (c) aligns somewhat with the Chamber Music Hall with a higher rating for system B and E closely followed by A and F.

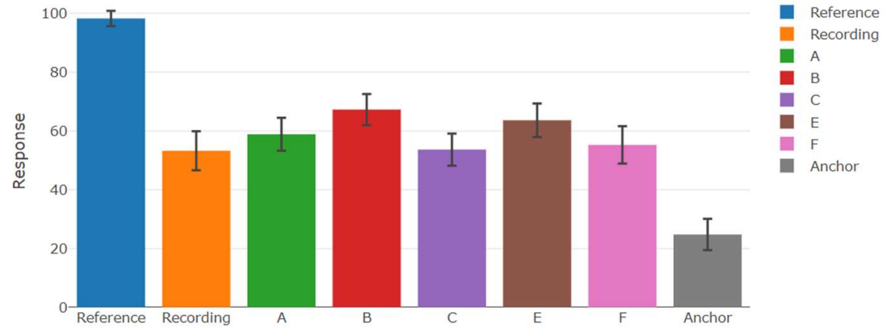


(a) Chamber Music Hall.

[1] https://d-nb.info/1253929041/34
[2] https://forcetechnology.com/en/articles/gated-content-senselab-sound-wheel



(b) Auditorium.

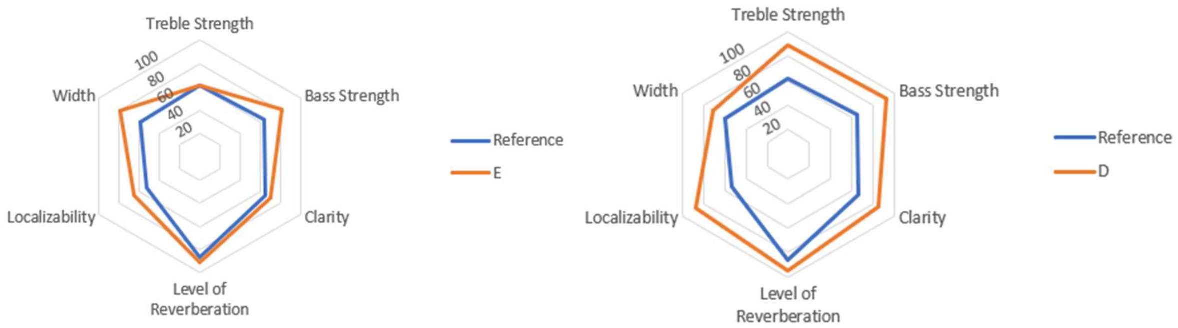


(c) Classroom.

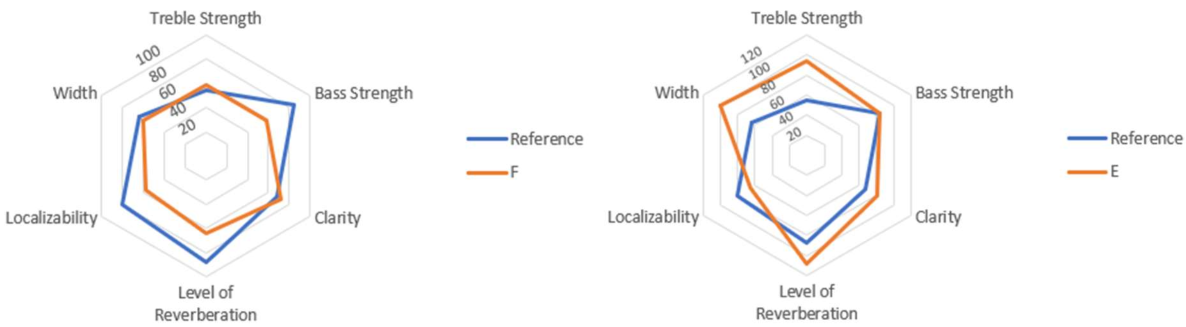
Figure 2: MUSHRA test results with means and 95%-confidence intervals.

3.2 Attribute Test Results

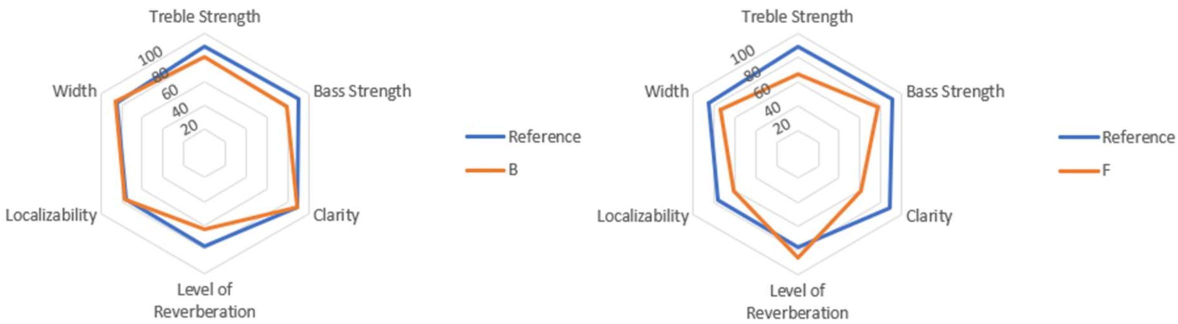
The results from the attribute test are shown with spider plots, with a selection of two plots for each geometry, displaying one simulation most similar to- and most different from the reference.



(a) Chamber Music Hall.



(b) Auditorium.



(c) Classroom.

Generally, the systems differ with geometry and no clear trend suggests that certain attributes (characteristics) are easier for the systems to replicate than others. Conversely, there is no attribute, that all systems fail to replicate. It is noteworthy that a trend of perceptual underestimation of an attribute in one room and overestimation in another is consistently observed across systems.

The results of the Level of Reverberation ratings show a trend where an increase in geometry volume will decrease the accuracy of the perceptual sense of the Level of Reverberation.

A t-test comparison of the two variations in late rays in Odeon introduce no significant differences in the attribute test.

T-test comparisons reveal that Treble using GA only compared to the hybrid simulations highly influence the timbral balance (tone colour) of the sound, which aligns with the expectation that the NM will introdue a perceptual difference in the low frequency, and thereby affect the relative strength of both bass and treble in the audio.

3.3 Multiple Linear regression (MLR)

A reduction of independent variables reveal which attributes are important for the overall perception of similarity, and is performed by creating an MLR with all attributes as independent variables, then a new MLR excluding the attribute with the highest p-value (the least impactful attribute). The reduced MLR is accepted if the reduced and original MLR are not significantly different from each other. The attributes and pertaining weightings are inserted in the regression model:

$$\bar{y} = 89.2 + 0.09\ \text{Loc.} - 0.07\ \text{Treble} - 0.06\ \text{Bass}, \quad R^2 = 0.15. \tag{2}$$

$$\bar{y} = 35.5 + 0.22\ \text{Bass} - 0.21\ \text{Treble} + 0.16\ \text{Reverb.} + 0.13\ \text{Loc.}, \quad R^2 = 0.31. \tag{3}$$

$$\bar{y} = 45.3 + 0.8\ \text{Clarity} + 0.1\ \text{Width} + 0.07\ \text{Treble} + 0.06\ \text{Bass} - 0.04\ \text{Reverb.} - 0.02\ \text{Loc.}, \quad R^2 = 0.12. \tag{4}$$

Despite the low R^2-values, the results from the BRAS rooms present the same attributes. With the Auditorium having the highest R^2-value, the attributes Localizability (Loc.), Level of Reverberation (Reverb.), Treble Strength, and Bass Strength collectively have the highest impact on the overall perceptual similarity.

4 Conclusion

This study demonstrates that neither of the simulations produces authentic binauralizations, thus none of the auralizations demonstrated indistinguishability from the reference. Even though some systems achieved better results compared to other systems this varies depending on the geometry. The key distinctions between the reference and systems also vary across geometries, nevertheless, a trend suggests that the perceptual accuracy of the Level of Reverberation decreases as the volume of the geometry increases.

Among the six tested attributes, Localizability, Level of Reverberation, Treble Strength, and Bass Strength collectively account for most of the variance in the data. This observation is drawn from data obtained in the Chamber Music Hall and Auditorium.

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Localization of sound sources reproduced by immersive and stereo sound systems

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The quest for lifelike spatial sound in music production has driven the development of immersive sound systems. As these systems become more common, understanding their perceptual qualities is crucial. This study examines how listeners perceive directionality in audio from stereo versus immersive setups. Through extensive listening tests, we evaluated participants' ability to locate sound sources across five positions on a virtual stage. The results overwhelmingly favor immersive sound, highlighting its perceptual benefits over traditional stereo setups.

1 Introduction

Accurate localization of sound sources in the horizontal plane relies on two primary mechanisms: interaural time difference (ITD) and interaural level difference (ILD) [1]. ILD is particularly relevant for high-frequency sounds with small acoustic wavelengths relative to the dimensions of the head. For longer sound waves that bypass the head, ILD contributions are minimal, humans rely on ITD for directional perception. Extensive investigations using tones and complex signals have demonstrated superior auditory localization precision for sources positioned in front of listeners [2]. Furthermore, the minimal perceptible angular difference, known as the minimum audible angle (MAA), varies with signal frequency and presentation angle. Generally, MAA values increase for larger angles and may remain indeterminate for specific angles and frequency bands [3]. The quest for achieving optimal sound field replication in musical recordings has driven the evolution of various surround sound systems. The British Broadcasting Corporation (BBC) played a pioneering role by transmitting the first stereophonic radio program as early as 1925 [4]. Subsequently, in the 1930s, Blumlein conducted groundbreaking experiments involving stereophonic microphone configurations [5]. This progression began with stereo setups and advanced to the quadraphonic systems, ultimately leading to the adoption of multi-channel soundscapes facilitated by the widely embraced 5.1 framework, which has further evolved into configurations like the 7.2.4 system. Truly stereophonic sound reaches only a small group of listeners concentrated in regions where left and right signals arrived with similar levels, somewhere in the middle of the audience. Accurate perception of the directional origins of apparent sound sources is therefore not possible for most of the audience. Immersive audio formats can be classified into channel-based, scene-based, or object-based paradigms [6, 7]. Channel-based formats extend beyond the 5.1 configuration by incorporating overhead speakers to create an immersive ambiance while maintaining a straightforward channel-to-speaker mapping. In contrast, scene-based formats generate complex data streams representing comprehensive three-dimensional acoustic environments. Object-based formats combine discrete audio streams with metadata that guides decoders on the precise positioning of these streams. A significant milestone in immersive audio technology occurred in 2016 with the introduction of the L-ISA Immersive Hyperreal Sound system by L'Acoustics [8]. This technology was quickly followed by similar offerings from other manufacturers. In 2017, d&b unveiled the Soundscape system, which was later joined by offerings from sound system pioneers such as Adamson, Martin Audio, and JBL in subsequent years [9, 10]. The Yamaha AFC Image immersive sound system, presented in 2021, represents the evolution of the AFC (Active Field Control) system, which traces its origins back to 1985 [11].
The increasing integration of immersive sound reinforcement systems sparks significant interest from a psychoacoustic perspective, prompting exploration into their inherent attributes and characteristics.

2 Aim

A study was undertaken to assess the subjective perception of directional localization, examining how accurately participants could locate sound sources placed at various positions on the stage. This involved comparing the localization ability when replicating spatial coordinates using stereo versus immersive sound systems. Participants were instructed to identify the origin of sound emanating from five specific locations on the stage.

3 Materials and Methods

To assess the subjective perception of directional characteristics in signals transmitted via stereo and immersive sound systems, a series of evaluations was conducted involving participants. The test focused on localizing sound sources at five positions on stage. In the immersive sound system setup, audio samples were played through specific speakers within the configuration. For the stereo system, the source positions were represented by apparent sound sources created by the speakers. The spatial orientations of signal presentation in the stereo system aligned with the spatial directions of the immersive sound setup's speakers.

3.1 Equipment

During the conference, immersive sound systems were meticulously configured by their respective distributors to demonstrate their capabilities. These systems were optimized to provide omnidirectional acoustic experiences, however, for this assessment, only the front line of the loudspeakers of each system were utilized. These primary loudspeakers were tasked with projecting sound directly from the stage space to the listeners. In the case of the Yamaha AFC system, Nexo PS12 loudspeakers served as the primary speakers, while in the L-ISA L'Acoustics system, A10 Wide loudspeakers fulfilled this role. The placement of loudspeakers in both systems adhered closely to specified guidelines. Sound samples for the experiment were sourced from a ProTools software session. Signal management and routing were facilitated through consoles and engines dedicated to each sound system's operation. Additionally, for the stereo system, outermost loudspeakers of the immersive setup were repurposed to serve as stereo speakers.

3.2 Sound samples

The experimental protocol utilized a pink noise sample, characterized by a frequency bandwidth ranging from 200Hz to 8000Hz, with a duration of one second. This frequency range was chosen to engage both mechanisms involved in sound direction recognition, namely Interaural Time Differences (ITD) and Interaural Level Differences (ILD). Participants were exposed to high-quality audio files adhering to a 24-bit/48KHz resolution standard, ensuring optimal fidelity of auditory stimuli. To maintain a standardized auditory environment, a uniform sound level of 65 dBA was established within the central region of the audience area. This precise control of sound intensity aimed to minimize confounding variables associated with sound level fluctuations, thereby enhancing the consistency and reliability of the experimental conditions.

3.3 Listeners

The participants in the experiment were the attendees of a conference oriented towards musicians and sound engineers. Each participant of this experiment completed pure tone audiometry assessments. Those identified with hearing impairments were systematically excluded from the study to ensure the integrity of auditory perception evaluations. A total of 80 participants were included, strategically divided into four groups of 20 individuals each. Each immersive sound system underwent examination, with two separate groups of listeners participating in tests for each system.

3.4 Procedure

The experimentation was conducted within the confines of the hotel's conference center, which featured rooms with dimensions of 12 meters by 11.4 meters, resulting in an approximate area of 137 square meters, accompanied by a height of approximately 4.8 meters. To replicate a traditional auditorium setup, four rows designated from A to D were meticulously arranged within each room. Each row was equipped with five chairs labeled 1 through 5, corresponding to virtual audience placements. The arrangement represented a conventional auditorium layout, seating participants every third chair and every second row. This configuration effectively simulated an auditorium with seven rows and thirteen seats per row. Figure 1 illustrates the configuration of both auditorium seating and the spatial arrangement of loudspeakers. In the scenario featuring the Yamaha AFC system, a set of five speakers was evenly distributed across the width of the auditorium, spanning 11 meters. In contrast, the L-ISA L'Acoustics system incorporated additional loudspeakers on each side of the stage, serving as extension speakers intended to broaden the auditory spatial field. These speakers are utilized when an object positioned near the edge of the stage requires consideration of its width parameter. However, as our experiment utilized monophonic sound sources, the extension speakers remained inactive, and participants were positioned within an 8-meter width area during the experiment.

The experimental protocol involved five iterations for each group of participants. In each iteration, participants were directed to occupy different rows and seats within the audience area. Consequently, participants experienced five diverse and randomly selected sequences for the presentation of sound samples throughout the experiment. Within each sequence, the sound samples were played back five times, utilizing both the stereo and immersive systems.

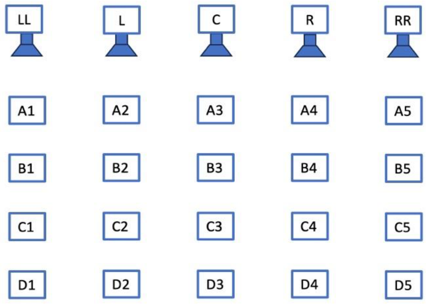
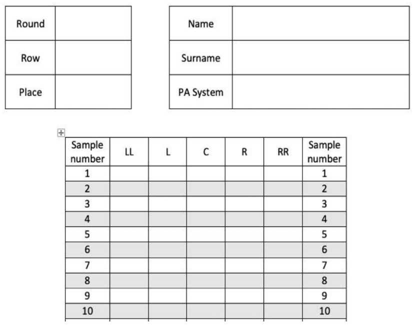


Figure 1: Layout of the audience and speaker system

Participants were provided with a response form. Personal data and ten initial response rows are illustrated in Figure 2. Within these boxes, participants were given the opportunity to indicate one of five directional orientations: LL (far left), L (between far left and center), C (center), R (between far right and center), RR (far right). The directional orientations were defined from the audience's perspective towards the stage, distinguishing between left and right. Audience responses were meticulously recorded to facilitate subsequent result compilation. Each sound system, L-ISA L'Acoustics and Yamaha AFC, underwent the test twice to ensure the robustness and reliability of the experimental findings.



Round		Name	
Row		Surname	
Place		PA System	

Sample number	LL	L	C	R	RR	Sample number
1						1
2						2
3						3
4						4
5						5
6						6
7						7
8						8
9						9
10						10

Figure 2: Questionare used in the experiment.

4 Results

For increased comprehensiveness, the acquired results have been divided into two distinct segments. The first segment provides a comparison between the Yamaha AFC system and the stereo system. Simultaneously, the second segment provides a comparison between the L-ISA L'Acoustics system and the stereo system. This partitioning enables a nuanced investigation of the outcomes, allowing for a focused analysis of the unique performance characteristics of each system when compared with the stereo counterpart.

4.1 Yamaha AFC vs Stereo

Figure 3 illustrates the percentage of accurate responses for each specific angle of signal presentation across every signal direction and individual seats. To improve visual clarity, a graduated color scale has been utilized. Colors transitions from yellow to green, representing an increasing trend in values from lower to higher. Letters from A to D denote rows, while numbers 1 to 5 denote seat numbers corresponding to Figure 1. RR, R, C, L, and LL indicate the positions of the presented sound sources as depicted in Figure 1. "Immersive" and "Stereo" denote the configuration of the audio system utilized to present the sound samples. Figure 4 provides a comprehensive summary, displaying the cumulative percentage of accurate responses across all signal presentations and directions for each individual seat within both the Yamaha AFC and stereo systems.

Stereo RR

	1	2	3	4	5
A	77%	80%	97%	100%	83%
B	57%	70%	93%	93%	100%
C	90%	93%	100%	87%	100%
D	77%	90%	90%	97%	93%

Immersive RR

	1	2	3	4	5
A	73%	70%	97%	100%	87%
B	47%	60%	93%	93%	100%
C	83%	87%	100%	90%	97%
D	70%	93%	93%	93%	93%

Stereo R

	1	2	3	4	5
A	7%	20%	70%	0%	3%
B	20%	13%	63%	3%	0%
C	20%	17%	90%	13%	7%
D	20%	13%	100%	7%	10%

Immersive R

	1	2	3	4	5
A	37%	73%	93%	100%	47%
B	20%	90%	97%	90%	93%
C	70%	83%	100%	70%	83%
D	90%	100%	90%	97%	87%

Stereo C

	1	2	3	4	5
A	0%	3%	87%	0%	10%
B	3%	0%	63%	7%	0%
C	10%	0%	93%	0%	0%
D	0%	0%	73%	13%	3%

Immersive C

	1	2	3	4	5
A	43%	90%	100%	100%	53%
B	43%	87%	90%	93%	63%
C	73%	77%	97%	73%	83%
D	93%	97%	100%	97%	80%

Stereo L

	1	2	3	4	5
A	0%	0%	80%	10%	17%
B	17%	3%	63%	33%	20%
C	0%	0%	87%	13%	20%
D	3%	0%	73%	30%	13%

Immersive L

	1	2	3	4	5
A	83%	87%	100%	97%	43%
B	63%	90%	87%	90%	37%
C	93%	100%	100%	87%	73%
D	93%	100%	93%	97%	87%

Stereo LL

	1	2	3	4	5
A	87%	90%	100%	77%	53%
B	70%	83%	90%	93%	57%
C	90%	97%	100%	100%	63%
D	93%	100%	93%	83%	70%

Immersive LL

	1	2	3	4	5
A	87%	93%	100%	80%	60%
B	67%	97%	97%	90%	67%
C	90%	97%	100%	90%	60%
D	97%	97%	97%	80%	73%

Figure 3: Stereo vs. Yamaha AFC - listeners' responses for all directions of presented signal and for all seats. The labels A through D and the numerals from 1 through 5 correspond to the seats plan (see Figure 1 for details). The colour scale delineates the transition from yellow to green, symbolizing an ascending trend in values from lower to higher. RR, R, C, L and LL denote the positions of the presents sound sources as shown on Fig. 1. Immersive and Stereo denote the configuration of the audio system used to present the sound samples.

Stereo, all samples all directions

	LL	L	C	R	RR
A	33%	37%	87%	39%	34%
B	35%	46%	75%	34%	33%
C	38%	43%	94%	41%	42%
D	38%	46%	86%	41%	39%

Immersive, all samples all directions

	LL	L	C	R	RR
A	58%	95%	98%	83%	65%
B	72%	91%	93%	85%	48%
C	79%	82%	99%	89%	82%
D	84%	93%	95%	97%	89%

Average for all seats and all presented samples

48%

Average for all seats and all presented samples

84%

Figure 4: Stereo vs. Yamaha AFC - values averaged across different directions of presentation. The left-hand section of the figure depicts the outcomes associated with the stereo configuration, while the right-hand section showcases the outcomes tied to the immersive configuration.

Respondents' responses underwent statistical analysis using Bayesian repeated measures ANOVA. The dependent variable was the percentage of correct answers, with the factors being the system and angle. Results indicated that the optimal model accounted for both factors and their interaction (BF10 = 2.5 * 10^270). This model elucidates approximately 45% of the variance in the data (R^2 = 0.45 [0.43, 0.47]). Post hoc analysis further revealed that the Yamaha AFC Image system outperformed the LR (left-right) condition. While differences were not observed for extreme angles, significant disparities were evident for angles in the middle. This observation is clearly depicted in Figure 5, where black dots represent the Stereo configuration and white dots represent the Immersive configuration. Additionally, 95% credible intervals are indicated on the chart with vertical bars.

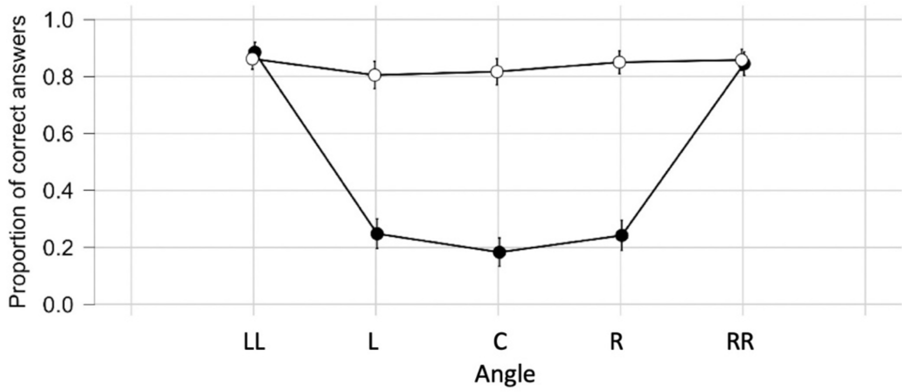


Figure 5: Proportion of correct answers for Stereo and Yamaha AFC Immersive configuration

4.2 L-ISA L’Acoustics vs Stereo

Similarly, mirroring the initial comparison, Figure 6 illustrates the percentage of accurate responses associated with each specific angle of signal presentation, categorized by distinct signal directions, and correlated with individual audience seats. The annotations on Figure 6 mirror those found in Figure 3.

Stereo RR

	1	2	3	4	5
A	83%	93%	100%	90%	73%
B	90%	100%	83%	70%	83%
C	97%	83%	83%	90%	80%
D	87%	93%	83%	80%	63%

Immersive RR

	1	2	3	4	5
A	83%	93%	97%	87%	67%
B	87%	93%	87%	63%	73%
C	97%	87%	73%	87%	80%
D	83%	83%	90%	70%	80%

Stereo R

	1	2	3	4	5
A	13%	17%	90%	0%	37%
B	13%	33%	87%	17%	13%
C	53%	43%	90%	17%	17%
D	50%	57%	67%	30%	27%

Immersive R

	1	2	3	4	5
A	63%	83%	100%	93%	73%
B	73%	90%	93%	80%	73%
C	60%	70%	83%	90%	83%
D	70%	77%	83%	90%	60%

Stereo C

	1	2	3	4	5
A	0%	0%	80%	3%	10%
B	3%	3%	80%	3%	7%
C	17%	0%	87%	10%	17%
D	3%	10%	83%	7%	27%

Immersive 0

	1	2	3	4	5
A	87%	93%	93%	93%	67%
B	90%	97%	87%	70%	70%
C	73%	77%	87%	83%	77%
D	73%	77%	90%	87%	70%

Stereo L

	1	2	3	4	5
A	13%	17%	90%	50%	23%
B	13%	7%	83%	50%	27%
C	10%	20%	93%	60%	33%
D	20%	13%	93%	77%	57%

Immersive L

	1	2	3	4	5
A	77%	87%	93%	83%	57%
B	90%	93%	80%	73%	77%
C	83%	87%	87%	87%	73%
D	70%	83%	83%	87%	67%

	Stereo LL				
	1	2	3	4	5
A	80%	83%	97%	70%	67%
B	83%	93%	67%	73%	77%
C	80%	87%	90%	87%	77%
D	67%	83%	93%	80%	90%

	Immersive LL				
	1	2	3	4	5
A	90%	73%	97%	90%	73%
B	80%	87%	70%	77%	83%
C	80%	83%	83%	93%	80%
D	73%	83%	90%	77%	90%

Figure 6: Stereo vs. L'Acoustics L-ISA - listeners' responses for all directions of presented signal and for all seats. The labels A through D and the numerals from 1 through 5 correspond to the seats plan (see Figure 1 for details). The colour scale delineates the transition from yellow to green, symbolizing an ascending trend in values from lower to higher. RR, R, C, L and LL denote the positions of the presents sound sources as shown on Fig. 1. Immersive and Stereo denote the configuration of the audio system used to present the sound samples.

Once more, the results were subjected to analysis using a Bayesian ANOVA approach. The model incorporating both factors and their interaction emerged as the most suitable (BF10 = 7.77 * 10^166), yielding an R^2 value of 0.38 [0.36, 0.4]. Discrepancies between systems and angles remained consistent, although this time, the percentage of correct answers for middle angles and the LR system showed slight improvement compared to the previous case. These findings are depicted in Fig. 7, where black dots denote the Stereo configuration and white dots denote the Immersive configuration. Vertical bars on the chart indicate the 95% credible interval.

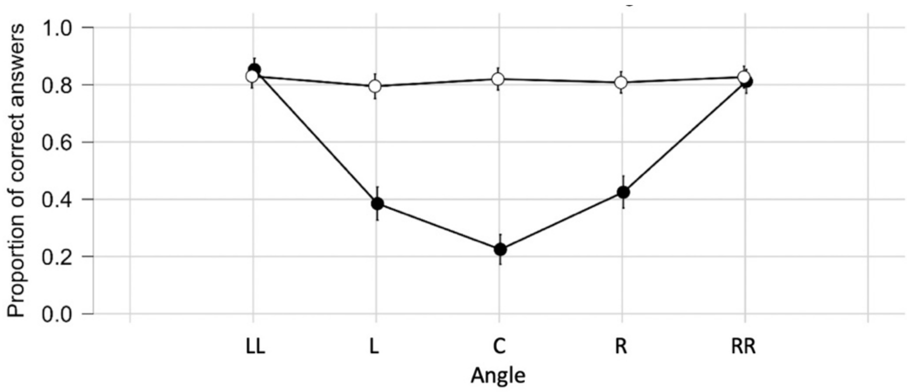


Figure 7: Proportion of correct answers for Stereo and L-ISA L'Acoustics immersive configuration.

Figure 8 displays the percentage of accurate responses for all directions of sample presentation at each site, along with the total percentage of accurate responses for the L'Acoustics L-ISA system. The results for the stereo configuration are depicted on the left, while those for the immersive configuration are shown on the right.

	Stereo, all samples all directions				
	LL	L	C	R	RR
A	42%	43%	91%	42%	38%
B	41%	43%	80%	47%	41%
C	45%	53%	89%	47%	51%
D	53%	55%	84%	51%	45%

	Immersive, all samples all directions				
	LL	L	C	R	RR
A	67%	89%	96%	86%	80%
B	75%	73%	83%	92%	84%
C	79%	88%	83%	81%	79%
D	73%	82%	87%	81%	74%

Average for all seats and all presented samples (Stereo)	Average for all seats and all presented samples (Immersive)
54%	82%

Figure 8: Stereo vs. L'Acoustics L-ISA - values averaged across different directions of presentation. The left-hand section of the figure depicts the outcomes associated with the stereo configuration, while the right-hand section showcases the outcomes tied to the immersive configuration.

5 Discussion

Upon analyzing the results, notable distinctions in correct response rates based on presentation mode were observed within the context of the Yamaha AFC system. Specifically, when sound samples were presented using the immersive AFC system, listeners demonstrated a commendable accuracy of 84% in providing correct responses. In contrast, when subjected to the stereo configuration, their accuracy declined to 48%. Similarly, within the scope of the L'Acoustics L-ISA system, analogous trends were observed, with a correct response rate of 82% recorded for the immersive setup, and a corresponding rate of 54% for the stereo configuration.

Further examination of the Yamaha AFC versus stereo configuration revealed intriguing nuances in the data. There were instances where listeners struggled to provide accurate responses, particularly in stereo mode. Specifically, for 16 distinct combinations of seating position and signal direction, participants failed to yield any correct answers. Conversely, in 13 other place-direction pairings, participants demonstrated a perfect 100% accuracy rate in their responses in the same mode. Interestingly, the Yamaha AFC system in its immersive configuration displayed distinct trends. Unlike its stereo counterpart, no place-direction pairings elicited zero correct responses. In fact, a substantial 19 combinations achieved a remarkable 100% accuracy rate.

Similar trends were observed in the analysis of the L'Acoustics L-ISA versus stereo configuration. Four instances of place-direction combinations resulted in zero correct responses, while in six other combinations, listeners exhibited flawless 100% accuracy. Additionally, the L'Acoustics L-ISA system, when operating in its immersive configuration, showed an absence of place-direction combinations resulting in zero correct responses. However, five combinations yielded a perfect 100% accuracy rate.

This comprehensive analysis highlights system-specific variations in correct response rates, suggesting an interplay between configuration and auditory localization acuity. These findings contribute to our understanding of the nuanced factors influencing sound localization within immersive and stereo contexts, illuminating the complex relationships between audio presentation, spatial awareness, and perceptual accuracy.

6 Conclusions

The results of the analysis notably highlight the discernible advantage of immersive configurations over their stereo counterparts, regardless of the specific system utilized. Both the Yamaha AFC and L'Acoustics L-ISA systems demonstrated an approximate 30% increase in correct response rates for the immersive configuration compared to the stereo configuration. This consistent trend underscores the perceptible advantage offered by immersive soundscapes, emphasizing their potential to enhance the accuracy of auditory localization.

Statistical analyses revealed that both stereo and immersive systems (whether Yamaha or L'Acoustics) yielded similar, favorable results only for the most extreme angles. However, for middle angles (L, C, and R), both immersive systems notably outperformed the typical stereo configuration.

The experimental auditorium layout, while effectively simulating an auditorium setting, represents only a limited subset of a traditional concert hall's expansive space. Given this context, an intriguing avenue for future exploration involves conducting a similar assessment within the capacious auditorium of a large concert hall, accommodating several hundred seats. Such an endeavor holds promise in unveiling insights into the performance of immersive and stereo systems within a more extensive and intricate acoustic environment. The inclusion of multiple seating tiers, including balcony placements, in such a setting could offer a more comprehensive perspective on the potential implications of system configuration on sound localization, contributing to a deeper understanding of spatial auditory perception in larger-scale concert hall scenarios.

7 Acknowledgements

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Comments

The research results presented in this paper have been accepted for publication in the journal Vibration in Physical Systems (ISSN: 0860-6897) Thus, the authors present here the results of their research for presentation at a scientific conference.

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SPATIAL ACOUSTIC MEASUREMENTS IN CONCERT HALLS WITH A REDUCED VIRTUAL ORCHESTRA

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The Virtual Orchestra has previously been used to measure room acoustic conditions of major concert-halls. The idea is to simulate an actual orchestra on stage with an array of loudspeakers and then measure the spatial impulse response in the hall. This approach gives a more complete picture of the acoustic projection and reflection paths between the stage and the audience than traditional measurements with a single omni-directional loudspeaker in few positions.

However, the originally proposed system requires over 24 loudspeakers and is therefore not very convenient for consultants to use from the practical perspective.

In this paper a scaled-down version of the measurement system is presented. This system uses a total of 8 loudspeakers and the spatial impulse response is recorded by a standard A-format microphone. The results are presented with comparative spatial analyses and preliminary results from practical measurements in halls.

1 Introduction

The measurements with the Virtual Orchestra has previously been shown to provide new insights to sound behavior in concert halls and other performance spaces [1][2]. The method enables application of visualization methods [3] for beneficial evaluation and comparisons of the reflection patterns and other spatiotemporal features in different acoustic conditions. However, it is also clear that conducting measurements with 24 or more loudspeakers requires relatively complicated logistics, which is seldom possible in the consulting world. Hence, a more practical approach with a substantially reduced number of loudspeakers, yet mimicking the original spatial distribution of sources, has been experimented and applied recently in acoustic investigations.

2 Measurement setup and analysis

The reduced measurement system consists of 8 loudspeakers: 6 pcs Genelec 8020D and 2 pcs Genelec 8030C. The 8020's are positioned on stands and placed to simulate the main part of instrument sections in typical orchestra seating layout. The larger loudspeakers (8030) are placed on the floor, to simulate the location of double basses and percussion sections.

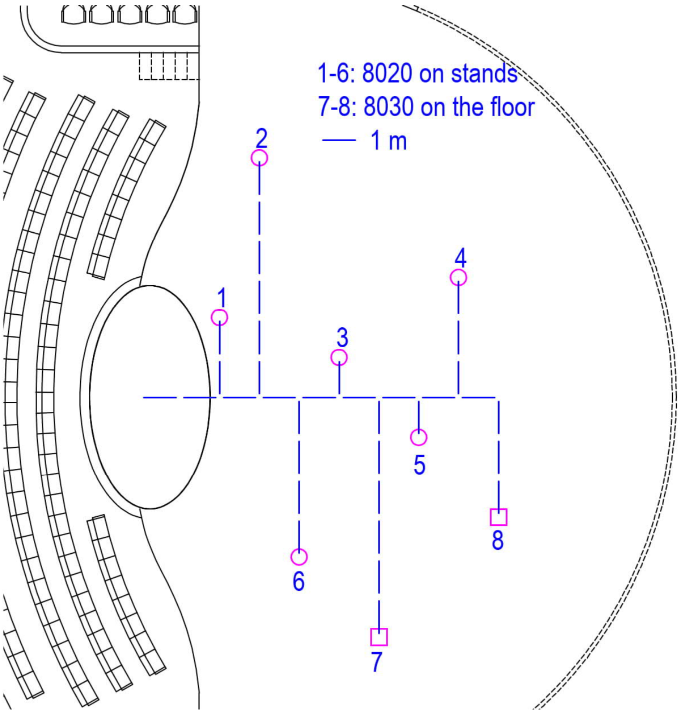

Figure 1. Applied loudspeaker layout

Loudspeaker output gains are calibrated using a pink-noise signal in-situ at 1 m reference distance in front of the final loudspeaker position. Floor loudspeaker levels are adjusted similarly with the loudspeaker pointing temporarily upwards. Although this method includes the influence of the floor reflection, the effect can be assumed nearly equal across different halls with relatively low contribution by the overall acoustic gain of a particular hall stage.

For the receiver, it was decided to use an A-Format microphone instead of the 6-capsule 3D intensity probe which was applied in the originally Virtual orchestra measurements and related studies. The actual measurements are done in practice using the REAPER software for playback and recording and spatial analysis and visualization is conducted using Python and Matlab scripts.

The recorded spatial impulse responses are analyzed with Spatial Decomposition Method (SDM) [4] and visualized using mainly the spatiotemporal and time-frequency visualizations [3].

The spatiotemporal analysis aims to estimate the direction-of-arrival for each audio sample in the measured spatial room impulse response [3]. The estimation is done in short overlapping analysis time-windows. The applied A-format microphone array contains four cardioid capsules in an open tetrahedral configuration at 24 mm distance between capsules. This type of receiver enables the use of so-called B-format signal processing.

Originally [4], the spatiotemporal analysis method is derived for open microphone arrays with omnidirectional capsules. The respective SDM analysis requires processing in short overlapping time-windows. Typical length for analysis windows for large acoustic spaces, such as opera and concert halls, is around 0,75 ms. This value is directly relative with the temporal resolution of the spatial analysis. That is, acoustic events, such as incident reflections that arrive more than 0,75 ms apart can be reliably discriminated from the data. The choice of analysis window is also guided by the dimension of the open microphone array. With larger inter-capsule distances, a longer analysis window is required to include sufficiently overlapping microphone signals for reliable directional estimates [1][4].

In contrast to original implementation with open microphone arrays, corresponding B-format signals analysis can be conducted with signal multiplications without time-windowing, as described in detail in [4],[5] and [6]. In order to obtain comparable results between open microphone array and B-format analyses, the temporal resolution of spatial information in sample-resolution was a low pass filtered with a 0,7 ms smoothing window for more stable application into visualization. Detailed discussion on the temporal resolution of analysis methods can be found in [4] and [5].

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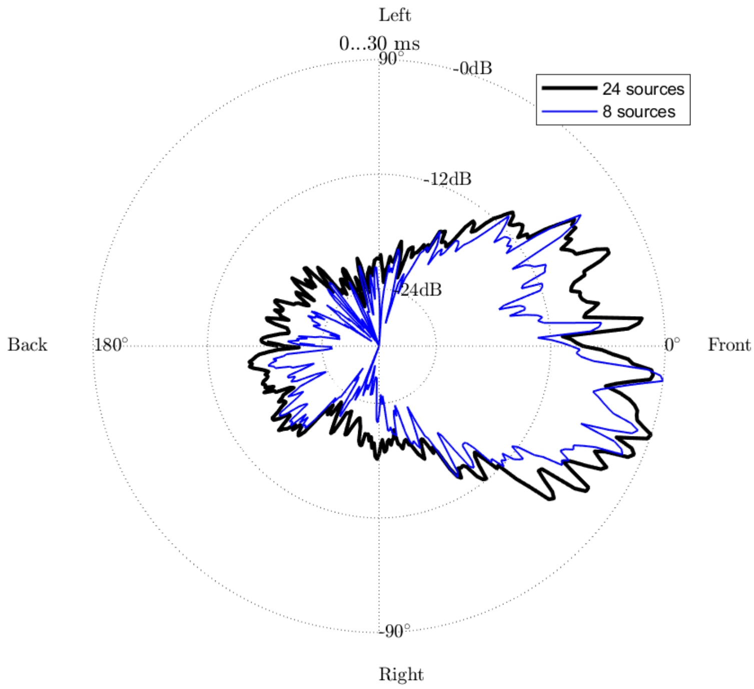
3 Comparison of full virtual orchestra and reduced system

The performance of the applied 8-source loudspeaker array in comparison to full 24-source system was evaluated with spatiotemporal visualizations. An opera hall measurement with 24 sources was analyzed in parallel with the subset of source positions that match with the reduced measurement system. Receiver position distance was approximately 10 m and 2 m off-center to the left side.

A set of comparisons is presented in Figure 2. Energy accumulation over selected forward-integrating time windows is visualized from spatiotemporal analysis with two source configurations. Early lateral energy up to 30 ms from the frontal sector shows slightly reduced energy at the extrema of the stage area. Due to more sparse source positions, certain angles have less direct sound, which is a natural consequence. In the visualization method, the spatial response's from individual sources is combined as the energy average. The curves in Figure 2 are normalized to 0 dB for easier comparison. The median plane example in middle Figure 2 shows the direct sound and early reflections up to 80 ms. The spatial result with reduced a source number correspond closely to 24-source reference.

Directions and level of reflections from high elevations are represented accurately. The longer accumulation of lateral energy within time-window 0…200 ms (Figure 3, bottom) is reduced in 8-source configuration, which results from overall lower sound energy radiated from lesser number of equally powerful sound sources.



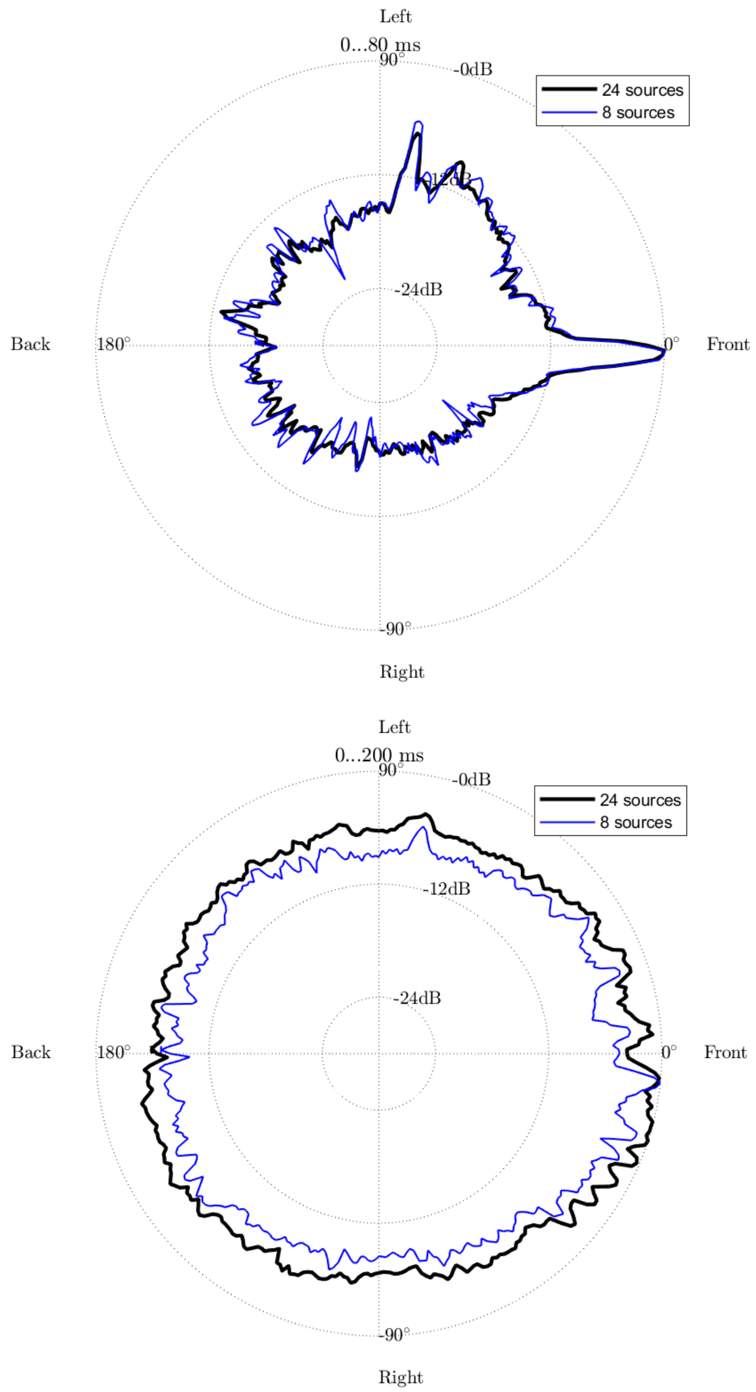


Figure 2. Comparison of energy directions from 3D impulse responses; top: lateral plane, time window 0…30 ms; middle: median plane, time window 0…80 ms; bottom: lateral plane 0…200 ms.

4 Preliminary results

The method has so far been used in measurements at the Bolshoi Opera Hall in Moscow, in the Sibelius Hall in Lahti as well as some halls with electro acoustic enhancement systems, as part of Henrik Möller's PHD research.

Figures 4 and 5 shows example results.

In both cases, the precision of the analysis, was found to be adequate to make an evaluation of the reflection patterns in the rooms. Although the density of the directions from sound sources is naturally sparser in the reduced configuration, the key directions for prominent reflections and their time-of-arrivals remain informative and comparable to the full 24-source reference. In the 8-source configuration, reflection paths by single sources might be emphasized due to smaller base of data averaging. Therefore, the 8-source visualizations may increase the risk of misinterpretation between a highly local reflection phenomenon and a more general feature of the hall. However, this fundamental factor touches to some extent all possible acoustic measurements.

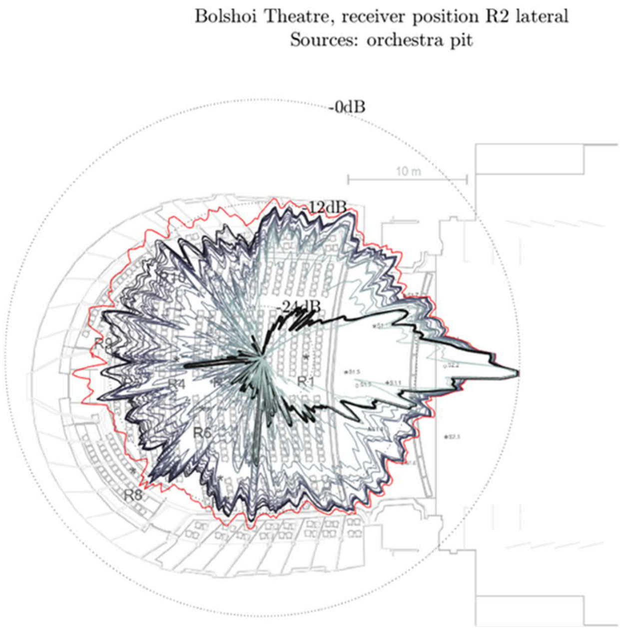


Figure 4. Measured 3D impulse response, lateral plane.

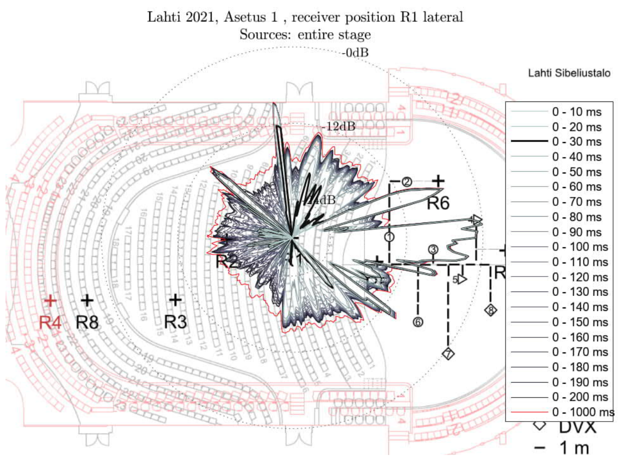


Figure 5. Measured 3D impulse response, lateral plane.

5 Conclusions

A smaller, more “travel friendly” version of the Virtual Orchestra has been presented. The first measurements done has shown that, for acoustic analysis of the space, the scaled down version, provides sufficient details for acoustic analysis of the reflection patterns. The reduced number of measurement sources may increase the uncertainty of interpreting single reflection patterns from the resulting visualizations.

It is yet to be investigated how well measurements done with the scaled down version can be used for auralization of rooms.

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Robust 3D Localisation of Anomalies in the Reverbaration Time Signal

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Abstract

Reverberation time measurements form the basis for room acoustic optimizations of existing building structures. During the verification of the achieved room acoustic improvements, anomalies may appear in the reverberation time signal which may be hard to spatially localize, especially in spaces with demanding acoustic requirements such as large, open workspaces or concert halls.

This contribution focuses on the application of the Sound Field Scanning method to the fast spatial localization of room reflections. In this process, an omnidirectional sound source is positioned at an observation point in the room and periodically excited with band-limited pulses. At the same observation point, an acoustic camera system consisting of a rotating linear microphone array is oriented towards the preferred spatial direction. The emitted pulses and associated room reflections are captured on the measurement surface of the rotating microphone array. Acoustic images with high depth resolution are generated in parallel planes to the measurement surface.

In complex situations, the task of spatially localizing anomalies in the reverberation time signal can be reduced to a few measurements from different perspectives, thus, significantly accelerating the problem-solving process with high confidence.

The method is exemplarily described through the room acoustic analysis of a university lecture hall.

Introduction

Sound source localization based on capturing the sound field with a two-dimensional microphone array is an attractive technique because it allows visualizing sound emissions from a distance and does not distort the sound pressure field due to the presence of the measurement device.

These measurement instruments - often referred to as acoustic cameras - are typically implemented as two dimensional microphone arrays with up to 130 microphones distributed over a circular area with a diameter ranging from 35cm up to 2.5m along with an optical camera to overlay the sound pressure map and the optical image of the measurement scene.

The imaging performance of these devices is mostly governed by two contributors, namely the diameter of the microphone array and the number of microphones distributed over that area. The array diameter predominantly sets the spatial resolution. The larger the array diameter, the higher the spatial resolution is, see section "Sensor Concept" for a detailed explanation.

The number of microphones influences the dynamic range of the resulting acoustic image, i.e. the maximum difference in loudness level that can be resolved. Also, a higher count of distributed microphones directly translates into a lower minimum detectable sound pressure level due to an improved signal-to-noise ratio.

While the localization of sound emissions is in many situations straightforward for stationary excitations - including the separation of direct sound events from reflections, the situation may easily get more complicated with localizing short-time sound events in the order of 10ms or less. Considering the fact that sound travels a distance of 1m in air in less than 3ms, a high temporal resolution of sound events is desired in order to locate reflections in space and time.

State-of-the-art acoustic cameras achieve a temporal resolution in the order of 10ms which is insufficient for many room acoustics applications as the resulting acoustic image for a short-time event of less than 10ms represents an overlay of multiple reflections.

These performance criteria have led the authors to propose a new sensor concept targeting at high spatial resolution, high dynamic range and high temporal resolution for the localization of specific portions of the reverberation time signal.

Sensor Concept

The proposed sensor concept is motivated by the underlying physics describing the spatial resolution and dynamic range of a sound imaging system. For simplicity, we consider a sensor with a linear, continuously distributed sensing capability with aperture length *L*. The corresponding normalized, horizontal directivity pattern *D* is given by

$$D(\lambda, \theta) = sinc\left(\frac{L}{\lambda} cos\ \theta\right) \tag{1}$$

where θ is the angle of arrival of an incident sound wave and $\lambda = c/f$ is the wavelength, *c* is the speed of sound in air and *f* is the sound frequency [2] [3] [4]. The shape of the directivity function is depicted in Figure 1. In this configuration, the distributed sensor is most sensitive for sound waves coming in at zero degree and its sensitivity degrades for waves approaching at other angles.

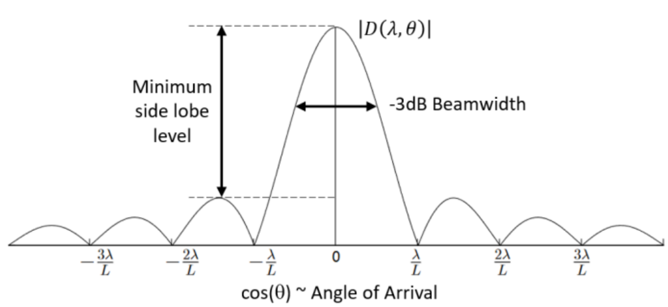

Figure 1. Normalized, horizontal directivity pattern D for a continuous linear array with aperture length L evaluated at wavelength λ.

The spatial resolution of a sound imaging system is typically quantified by the -3dB beamwidth of the main lobe. An improved spatial resolution can therefore be achieved in two ways: (i) increasing the aperture length L or (ii) increasing the frequency of the sound event. Option (i) essentially translates into an increased size of the sensor which, as we will see later on, requires a higher count of distributed microphones and, thus, impacts the hardware complexity. Option (ii) may potentially be an available parameter in applications where the excitation frequency of the ultrasound transmitter can be controlled. Yet, it shall be considered that the higher the excitation frequency is, the more difficult it becomes to implement an ultrasound transmitter with both omnidirectional characteristics and sufficient sound power.

The side lobes in Figure 1 play a special role for real arrays with a finite number of discrete microphones. In fact, the side lobe level quantifies the dynamic range of a sound imaging system. If, for instance, the side lobe level at a certain frequency is 10dB below the main lobe level and assuming that all involved sound sources can be spatially resolved, then the imaging system is still able to localize secondary sources with a pressure level less than 10dB below the most dominant source. The dynamic range can be improved by increasing the number of distributed microphones which, again, impacts the hardware complexity.

In order to get a better impression of the actual numbers that the above formulae suggest, we consider an automotive component with a size of 1m by 1m and key features lying apart in the range of about $d_1 = 10cm$. The distance at which the measurement is conducted is about $d_2 = 75cm$ which ideally requires the -3dB beamwidth of the main lobe to have an opening angle of less than $\theta_{max} = 2\text{atan}\left(\frac{d_1/2}{d_2}\right) = 0.13rad = 7.63°$. When localizing impulse-like sound events in the order of a couple of milliseconds, the lowest localizable frequency shall be around 2.5kHz. Considering that the relation $cos\ \theta = \lambda/L$ shall hold, we can derive the minimum array diameter according as follows: $L_{min} = \lambda/\theta_{max} = 1.06m$ with $\lambda = 343ms^{-1}/2{,}500Hz$, see Figure 1.

Considering the typical landing pattern of a digital MEMS microphone which is in the range of 4mm by 6mm, the hardware implementation of an array with a high count of microphones for optimum dynamic range can easily become a realization problem.

Based on these insights and trade-offs associated with distributing a high count of microphones across a measurement surface with a diameter of at least 1m, the authors propose a sensor concept which enables high spatial resolution and high dynamic range while targeting minimum weight and complexity of the associated sensor hardware.

Hardware implementation

The centerpiece of the sensor concept is a rotating linear array with five distributed microphones which pivots about a non-moving reference microphone. The trajectory of the remaining moving microphones is described by concentric circles. A magnetic rotary encoder which is co-axially aligned with the rotation axis of the array, measures the angular position with respect to the spatially constrained axis of rotation, see Figure 2.

The microphones are based on digital MEMS technology and the corresponding signals are acquired over a common signal path using the time division multiplexing (TDM) method. This method enables the straightforward implementation of a microphone multiplexing scheme for data compression and emulation of arbitrary, even non-implementable two-dimensional array geometries. For instance, the data acquisition can be configured such that the reference microphone along with a second channel which periodically switches between the moving microphones, are recorded, see Figure 3.

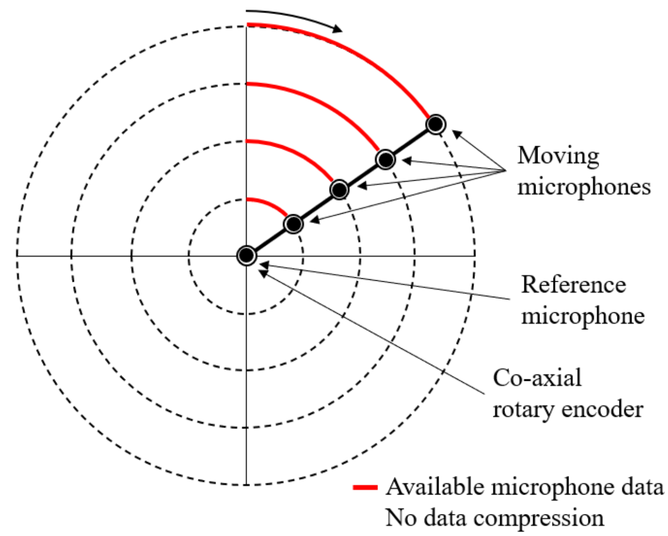

Figure 2. Rotating linear array comprising five microphones pivoting about the reference microphone. The trajectory of the moving microphones is described by concentric circles.

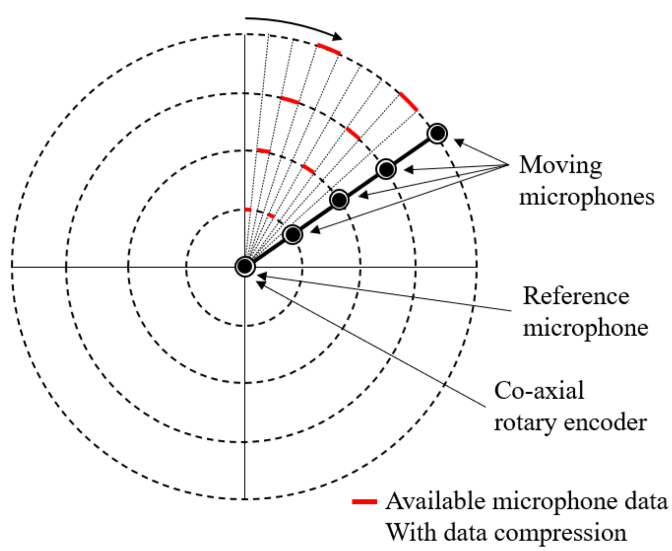

Figure 3. Multiplexing of the moving microphones enables data compression and emulation of arbitrary two-dimensional array geometries.

It is well known that the directivity pattern of the array and the corresponding position of microphones can be optimized to meet certain performance criteria, e.g. the minimum side lobe level at specific frequencies. While two-dimensional arrays with discrete microphone positions require a complete hardware reconfiguration in terms of repositioning the microphones, the rotating linear array merely requires a software reconfiguration to acquire the data at different positions.

Also, the implementation of large arrays with a diameter of more than one meter does not increase the hardware complexity. In fact, the number of microphones distributed along the linear array can stay the same since the fine spatial sampling along the concentric circles guarantees adherence to the spatial sampling theorem [1] [2].

The rotating linear array is a self-powered system and uses wireless technology for data transmission of the audio and rotary encoder data to a processing unit.

Properties of signals acquired by moving microphones

In order to better understand the characteristics of a signal acquired by a moving microphone, we consider a point source with harmonic excitation signal $u(t)$ at the frequency f_0

$$u(t) = Re\{e^{i2\pi f_0 t}\}.$$

Assuming that the corresponding sound wave is independent from the distance to the source and the rotational speed f_{rot} of the moving microphone is constant, the audio signal $y_1(t)$ captured by the moving microphone is given by

$$y_1(t) = Re\left\{e^{i2\pi f_0 t} e^{-i2\pi f_0 \frac{d_1(t)}{c}}\right\}$$

where $d_1(t)$ is the time-varying distance between the sound source and the position of the moving microphone along its circular trajectory with radius R, see Figure 4.

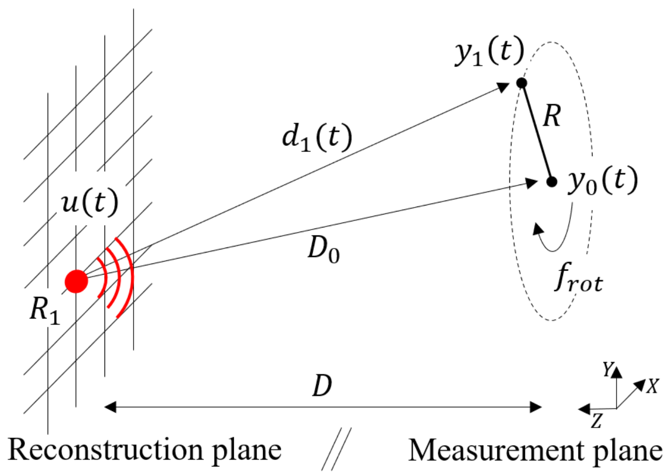

Figure 4. Notations used for describing the setup comprising a point source in the reconstruction plane and a reference microphone and a moving microphone located in the measurement plane.

We further denote the constant distance between the sound source at reconstruction point R_1 and the stationary reference microphone by D_0, the corresponding audio signal by $y_0(t)$

$$y_0(t) = Re\left\{e^{i2\pi f_0 t} e^{-i2\pi f_0 \frac{D_0}{c}}\right\}$$

and the distance between the parallel reconstruction and measurement planes by D. The origin of the Cartesian coordinate system with point representation given by (X, Y, Z) is at the position of the reference microphone and the XY-plane is the measurement plane. Considering the parameters

$$f_0 = 1kHz, f_{rot} = 1Hz, R = 1m, D = 3m, D_0 = (-0.1m, -0.1m, D), \quad (2)$$

we get the following result for the time-varying distance $d_1(t)$

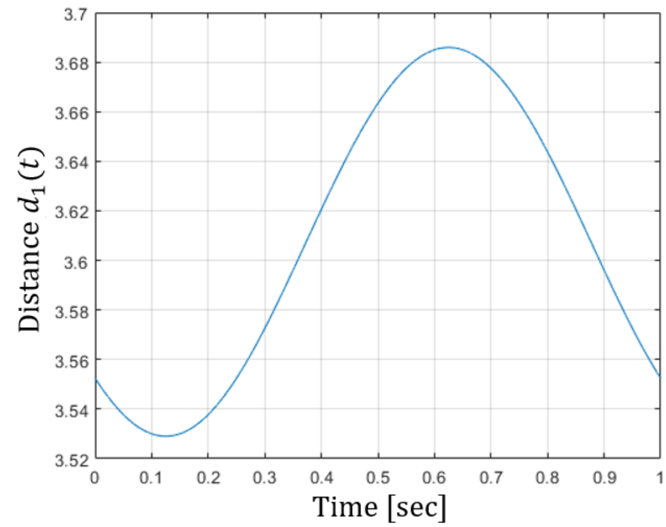

Figure 5. The time-varying distance between the sound source and the position of the moving microphone along its circular trajectory.

and the short-time Fourier transformation of the audio signal $y_1(t)$, see Figure 5 and Figure 6.

As expected, the short-time Fourier transformation of the moving microphone signal $y_1(t)$ is a Doppler-shifted version of the original source signal $u(t)$.

Acoustic Image Computation

The measurement setup depicted in Figure 4 along with the basic observations on the signal properties of the moving microphone and reference microphone signals now enable us to derive an algorithm for the computation of a map describing the distribution of sound sources in the reconstruction plane.

The case of perfect Doppler shift compensation

As a first step, we map the signal $y_1(t)$ of the moving microphone to the spatial position of the reference microphone. This transformation involves backpropagating $y_1(t)$ to the point of the sound source in the reconstruction plane using the time-varying distance $d_1(t)$ and then forward propagating the signal to the reference microphone position using the constant distance D_0.

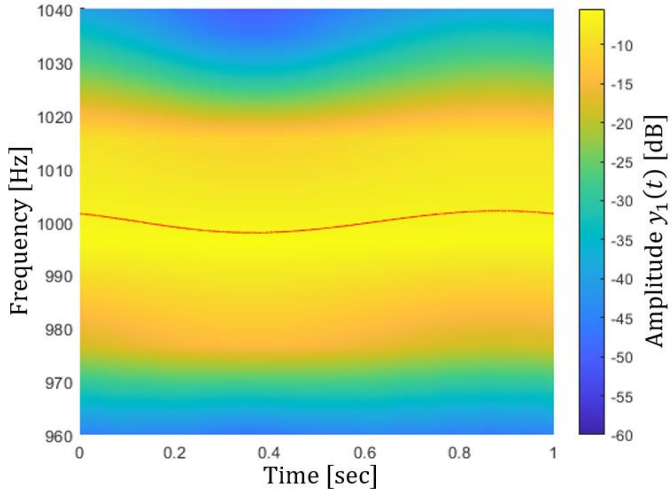

Figure 6. The short-time Fourier transformation of the moving microphone signal for the time-varying distance $d_1(t)$ depicted in Figure 5.

The resulting signal $\underline{y}_1(t)$

$$\underline{y}_1(t) = y_1(t + d_1(t) - D_0) = Re\left\{e^{i2\pi f_0 t} e^{-i2\pi f_0 \frac{(d_1(t) - d_1(t) + D_0)}{c}}\right\} = y_0(t)$$

has the obvious property that the Doppler shift previously induced in $y_1(t)$ is fully compensated and is identical to the signal captured at the position of the reference microphone.

The general case

We now consider the transformation for a point R_2 in the reconstruction plane which is at a constant distance $\underline{D}_0$ from the reference microphone position and away from the point source, see Figure 7.

An additional Doppler shift is induced in the transformed signal $\underline{y}_1(t)$ given by

$$\underline{y}_1(t) = y_1\left(t + \underline{d}_1(t) - \underline{D}_0\right) = Re\left\{e^{i2\pi f_0 t} e^{-i2\pi f_0 \frac{(d_1(t) - \underline{d}_1(t) + \underline{D}_0)}{c}}\right\}.$$

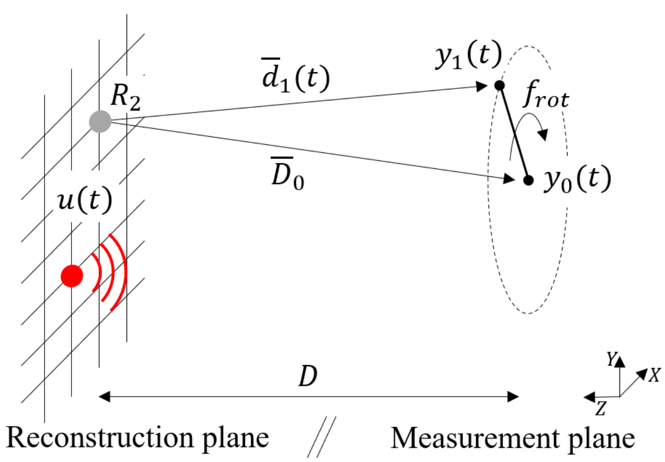

Figure 7. Notations used for describing the setup when mapping the moving microphone signal $y_1(t)$ to the reference microphone position via the point R_2 in the reconstruction plane.

Now, we apply the coherence function $C_{\underline{y}_1 y_0}(f)$ as a frequency-dependent measure of statistical similarity of the transformed signal $\underline{y}_1(t)$ and the signal $y_0(t)$ captured a the reference microphone position,

$$C_{\underline{y}_1 y_0}(f) = \frac{\left|S_{\underline{y}_1 y_0}(f)\right|^2}{S_{\underline{y}_1 \underline{y}_1}(f) S_{y_0 y_0}(f)}$$

where $S_{\underline{y}_1 y_0}(f)$ is the cross-spectral density of the signals $\underline{y}_1(t)$ and $y_0(t)$ and $S_{\underline{y}_1 \underline{y}_1}(f)$ and $S_{y_0 y_0}(f)$ are the power spectral density functions of $\underline{y}_1(t)$ and $y_0(t)$, respectively [4]. The coherence function varies in the interval $0 \leq C_{\underline{y}_1 y_0}(f) \leq 1$. We get a high coherence value at a specific frequency f for points in the reconstruction plane where a sound source is actually located, and a low coherence value for points where there is no or little sound radiation.

We can now use this metric to produce a heatmap representing the distribution of sound sources over the entire reconstruction plane. Considering the parameters from Equation (2) and a point source at the spatial position $(-0.1m, -0.1m, -D)$, we get the color-coded representation of the coherence function $C_{\underline{y}_1 y_0}(f)$ evaluated at $f = 1kHz$ depicted in Figure 8. With this special set of parameters, the resulting heatmap is also referred to as point spread function (PSF) which is used to quantify the performance of an imaging system in terms of spatial resolution and dynamic range [1].

Comparison with fixed arrangement of microphones

In order to appreciate the image quality achieved with one moving microphone only, we can compute the coherence function for a hardware setup with a fixed spatial arrangement of microphones equally spaced along the trajectory of the moving microphone. Figure 9 depicts the result for 12, 24 and 96 microphones.

It is readily visible that the coherence function for the arrangement of 96 microphones approaches the result from the moving microphone. Using 12 microphones only leads to the well-known artifact of grating lobes caused by spatially undersampling the incident sound field [2].

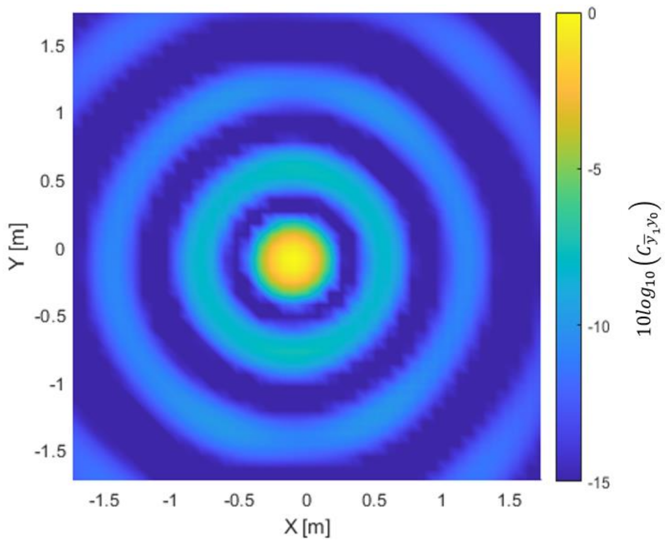

Figure 8. Color-coded logarithmic representation of the coherence function $C_{\underline{y}_1 y_0}(f)$ evaluated at $f = 1kHz$.

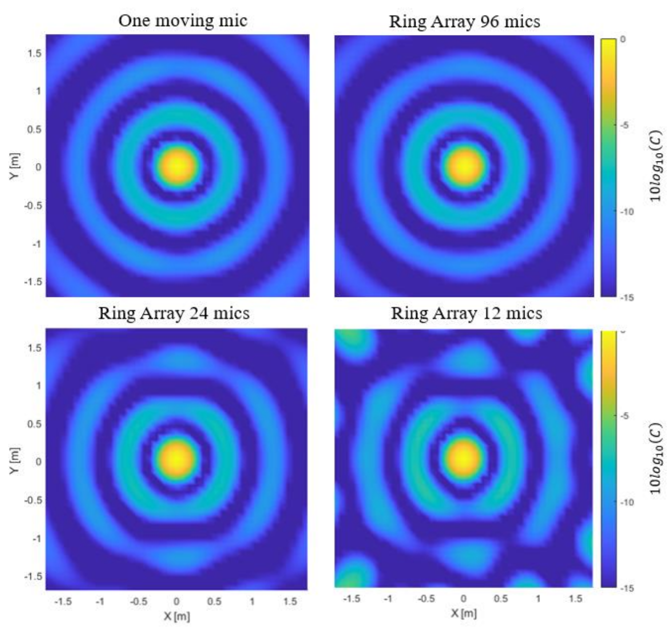

Figure 9. Coherence function $C_{\underline{y}_1 y_0}(f)$ for fixed spatial arrangement of 12m 24 and 96 microphones evaluated versus one moving microphone.

Multiple distributed sound sources

Before we evaluate the capabilities of the proposed sensor concept in real world applications, we finish this section by computing the distribution of multiple sound sources of equal strength emitting a tone at 1kHz and located at positions $P_1 = (-1m, 0, D)$, $P_2 = (0,0, D)$, $P_3 = (1m, 1m, D)$ in the reconstruction plane, see Figure 10.

We shall note that the artifacts surrounding the three sound sources are coming from the mutual, positive interference of the point spread functions at the three spatial positions, thus, degrading the useable dynamic range.

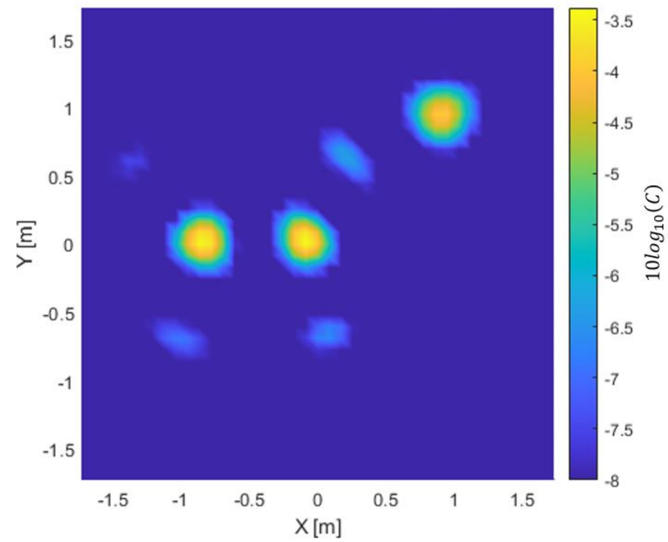

Figure 10. Coherence function $C_{y_1 y_0}(f)$ for multiple sound sources emitting a tone at 1kHz and located at positions $P_1 = (-1m, 0, D)$, $P_2 = (0,0, D)$, $P_3 = (1m, 1m, D)$ in the reconstruction plane.

3D Tracking of reflections

The data acquisition and processing described in the previous section can be adapted to the task of localizing impulse-like sound events with high temporal resolution in a straightforward manner.

As a first step, the sound event under investigation shall be made repeatable. While the linear microphone array is spinning, the repeating sound events are recorded at different rotation angles, see Figure 11.

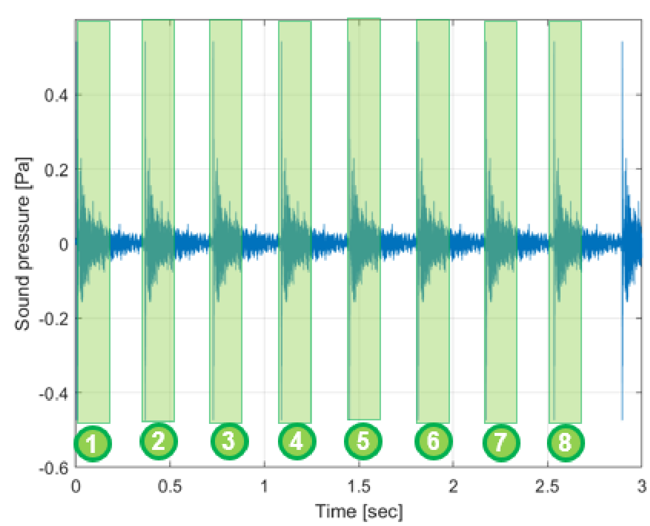

Figure 11. Record of periodic, matching sound events. The user selects a representative impulse and matching impulses are identified based on auto-correlation and frequency domain characteristics.

In a second step, the user selects a representative impulse and all repetitions are searched for in the remainder of the audio signal. The search algorithm uses auto-correlation and frequency domain characteristics in order to robustly identify valid repetitions of the representative impulse.

It is then verified whether the recorded impulses are homogeneously distributed across the measurement surface, see Figure 12.

For doing so, the measurement surface is decomposed into sectors and one sector shall assigned to one impulse at most. Assigning a sector to more than one impulse or not assigning a sector to an impulse at all would be equivalent to introducing an apodization of the function describing the spatial distribution of microphones.

Based on the reduced set of impulses which have been spatially assigned to sectors of the measurement surface, the best homogeneous distribution of these impulses is computed.

The optimization criterion for deriving a best possible distribution of impulses is achieved by regarding each sector as a unit mass with unit distance from the center of the measurement surface.

A compound metric based on the inertia tensor and center of mass of this distribution can be used to derive an optimal subset of spatially recorded impulses.

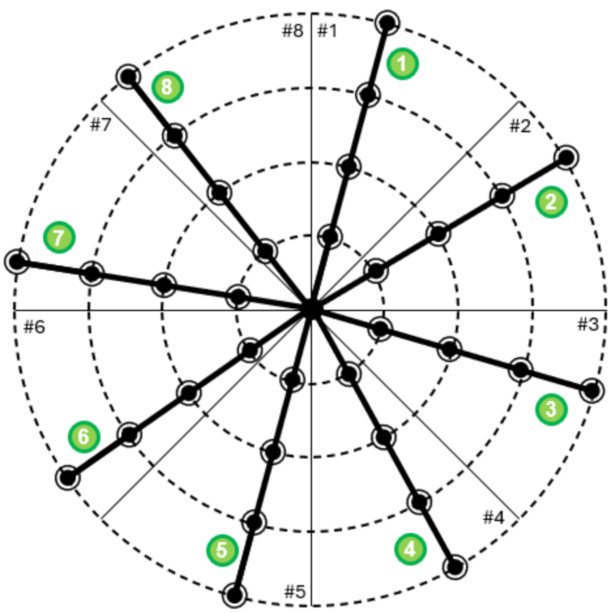

Figure 12. The best possible set of spatially recorded impulses is derived from omitting redundant impulses for specific sectors of the measurement surface (e.g. impulse 4 in sector #3) and considering a compound metric based on the inertia tensor and the center of mass of this spatial distribution.

Now that the relevant impulses have been identified, we partition the selected impulse of the audio signal into overlapping time intervals in the order of milliseconds.

The matching partitions in the other impulses can be found based on their time-offset from the selected impulse, see Figure 13.

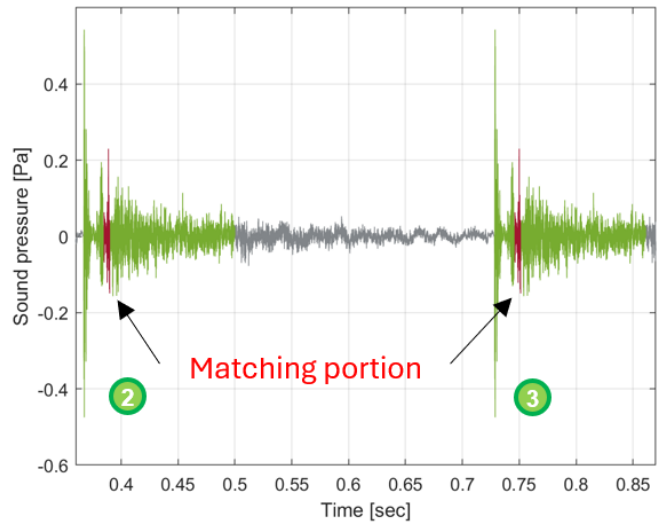

Figure 13. Matching partitions for impulses 2 and 3 which are spatially assigned to sectors 2 and 3, respectively.

An acoustic image for a specific partition of the selected impulse can now be computed based on audio signals only consisting of matching partitions from all relevant impulses. Since the onset of the direct sound from the speaker to the reference microphone of the linear array is known, the distance between the measurement plane and the reconstruction plane for a specific signal portion of the reverberation signal can be computed accordingly. This means that the reconstruction plane moves away from the measurement plane at the speed of sound. Thus, for every signal portion of the reverberation signal a different distance from the measurement plane is chosen.

Applications

The proposed sensor concept is now applied to localizing an anomaly in the reverberation signal which is recorded in the stage area of a concert hall.

Setup

The measurement setup comprises the following devices:

- a rotating sensor with a total of five microphones sampled at 21.3kHz and distributed over a length of 66cm with one reference microphone at the center of rotation and four microphones moving along circular trajectories on a disc with a maximum diameter of 132cm,
- a mobile device (model: Samsung Galaxy Tab S9) for capturing the audio as well as rotary encoder data and sending the data to
- a high performance laptop (model: Dell precision 7680) for computing the acoustic images and
- an omnidirectional speaker (model: Norsonic Nor276 including amplifier Nor 280).

The omnidirectional speaker is positioned in the center of the stage along with the rotating linear array, see Figures 14 and 15.

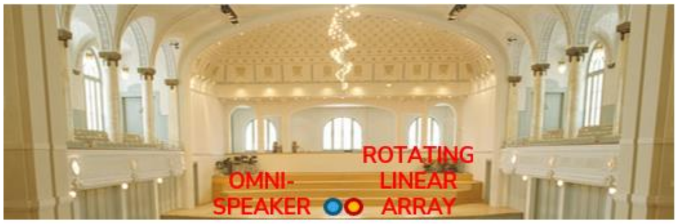

Figure 14. Arrangement of omnidirectional source and rotating linear array in the stage area of a concert hall.

Figure 15. Co-located arrangement of omnidirectional source and rotating linear array.

The sensor rotates at a speed of two revolutions per. The impulse-like sound event is emitted at a repetition frequency of 2Hz. A total of 20 sound events are captured. The optical image is taken with a horizontal field of view of 69,5° and the optical camera – facing away from the auditorium - is pointed in the direction of the left half of the stage area.

Result

Figure 16 depicts the recorded reverberation time signal of the reference microphone where the horizonal axis is expressed as a distance in meters from the measurement plane. The impulse from the direct sound event is visible at the beginning of the chart, i.e. at a distance of 0m away from the measurement plane. At a distance of 17m away from the measurement plane, we can detect an anomaly in the reverberation time signal.

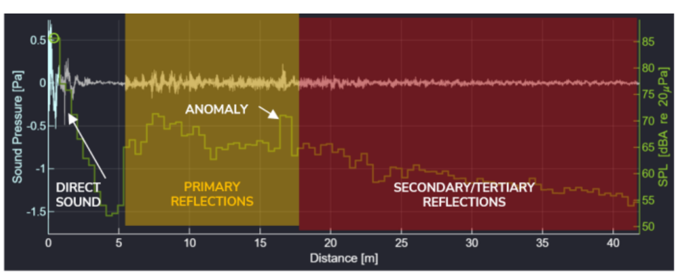

Figure 16. Recorded reverberation time signal with anomaly at a distance of 17m away from the measurement plane.

Figure 17 shows the corresponding acoustic image which localizes the anomaly in the top left ceiling area of the stage. The ceiling is equipped with diffusor panels.

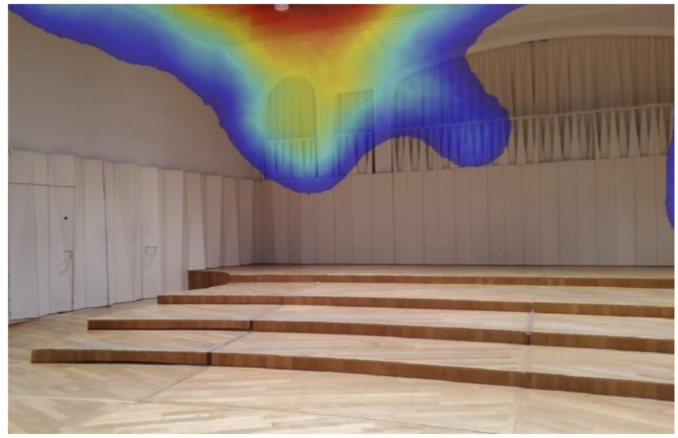

Figure 17. Anomaly in the reverberation time signal at a distance of 17m away from the measurement plane. Selected frequency band is 250Hz – 3.1kHz at a dynamic range of 3dB.

The arrangement of omnidirectional source and rotating linear array can be modified for the analysis of sound propagation from one point in space to an observation point, see Figure 18.

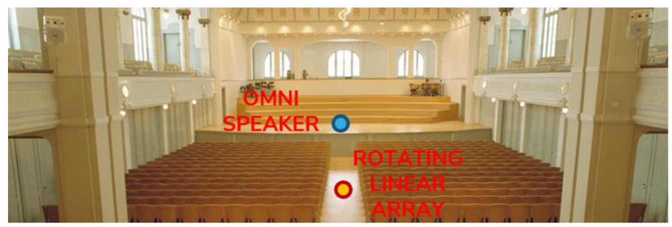

Figure 18. Arrangement of omnidirectional source and rotating linear array for the analysis of sound propagation from the stage area to an observation point in the auditorium.

Figure 16 depicts the recorded reverberation time signal of the reference microphone. The time chart can be decomposed into a time interval of 20-60ms after the direct sound event (primary reflections) and a second time interval of 60-170ms after the direct sound event (critical reflections).

While the primary reflections control the level of the received audio signal, the critical reflections govern the delayed perception of the audio signal.

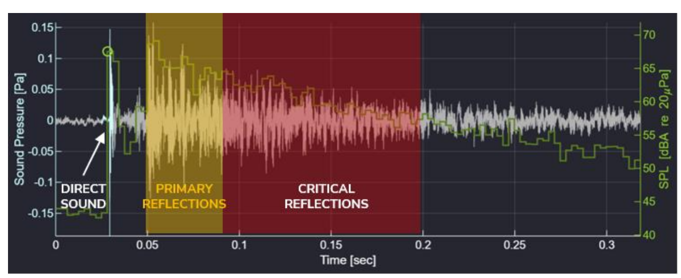

Figure 19. Recorded reverberation time signal with primary reflections (20-60ms after direct sound event) and critical reflections (60-170ms after direct sound event).

The corresponding spatial areas where these reflections are received from at the point of observation are depicted in Figure 20 and 21.

Figure 20. Primary reflections located at the point of observation. Selected frequency band is 250Hz – 3.1kHz at a dynamic range of 2dB.

Figure 21. Critical reflections located at the point of observation. Selected frequency band is 250Hz – 3.1kHz at a dynamic range of 6dB.

In case the delayed perception of audio signals at the point of observation is an issue, the diffusor ceiling in the stage area can be singled out as the primary reflector while the diffusor wall of the stage area can be considered as a secondary effect with sound pressure levels 6dB below.

Summary

This contribution addressed the application of the Sound Field Scanning method to the fast spatial localization of room reflections.

The underlying measurement method is derived and its performance properties are described. A measurement setup comprising the above mentioned sensor, a mobile device, a high performance laptop and omndirectional speaker is used to produce acoustic images for localizing anomalies in the reverberation signal which is recorded both in the stage area and auditorium of a concert hall.

Future work will analyse the effectiveness of the method for optimizing the room acoustics of different environments, e.g. office space and production environment.

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Optimising Railway Track Vibration Isolation by Matching Ground-Borne Noise Level Data to Train Location in a Tunnel

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Tampere Deck and Arena in Tampere city centre is an extensive and diverse hybrid building complex in an extremely challenging location considering ground-borne noise from railway traffic. The building complex consisting of a multipurpose arena for up to 15,000 spectators, a 273-room hotel, a casino, restaurant and commercial spaces, and two office and apartment towers has been built on a bridge deck covering Finland's busy main railway corridor. The numerous InterCity and freight trains traveling along the railway corridor day and night cause significant vibration that required comprehensive mitigation solutions to avoid excessive ground-borne noise levels in sensitive spaces located directly above. The level of attenuation required to fulfil the local noise criteria was not technically achievable by isolation implemented in building foundations and inside buildings, and so certain track sections had to be isolated as well. This article presents the measurement and analysis process used to optimise the extent of isolation in one of Finland's main railway tracks that had been identified as the primary source of ground-borne noise for an apartment tower. A custom lidar-based solution was developed to be able to accurately locate the trains under the deck where the train GPS position resolution was not sufficient or even not available. Simultaneous train locating and measurements of the ground-borne noise caused by the trains inside the building enabled identifying the exact track sections where typical traffic results in excessive noise levels, determining the required length of isolated track, and verifying the requirements for insertion loss. Under-ballast mats were installed under the defined track section, and the measurements were then repeated to confirm the noise criteria were fulfilled.

1 Introduction

Tampere Deck and Arena in Tampere city centre is a recently built extensive and diverse hybrid building complex in an extremely challenging location considering ground-borne noise from railway traffic. A large bridge deck was constructed over a railway corridor to accommodate a building complex consisting of a multipurpose arena for up to 15,000 spectators, a 273-room hotel on floors 6 to 13, a casino, restaurant and commercial spaces, and two office and apartment towers. In the Opaali tower, apartments are located starting from the 3rd floor and in the Topaasi tower the lowest apartment floor is the 7th floor. The bridge deck essentially forms a railway tunnel for the busy main railway corridor of Finland, used by InterCity and freight trains alike, day and night. Under the deck there are 6 tracks with several switches.

The numerous InterCity and freight trains traveling directly below the buildings cause significant vibration that required comprehensive mitigation solutions to avoid excessive ground-borne noise levels in sensitive spaces. Vibration measurements before and during the construction of the bridge deck were conducted at ground level and on the deck to estimate the level of attenuation required to be able to satisfy the local noise level requirements for ground-borne noise caused by the trains.

Attenuating ground-borne noise from train traffic inside the buildings required keeping the buildings structurally separated from the bridge deck and the ground below. The main isolation solution meant placing flexible vibration isolation materials between the deck and the buildings built on it. Multiple unique and completely new details were designed to keep all floating structures disconnected from the bridge deck and the ground. The challenge with new solutions is that there is no previous knowledge on their successful implementation or realistically achievable attenuation.

As the buildings progressed, measurements conducted inside showed ground-borne noise levels in exceedance of the criteria at some locations. The vibration caused by train traffic varied both at ground level and in different parts of the bridge deck much more than had been predicted based on earlier measurements. A thorough investigation on the situation concluded that the level of attenuation required to fulfil the local noise criteria inside the buildings was not technically achievable everywhere by isolation implemented only in building foundations and inside buildings, contrary to what was estimated during the design phase based on earlier measurements. Consequently, the process to design and implement vibration isolation under certain track areas had to be initiated.

First, the most significant track sections causing excess ground-borne noise had to be identified. Comprehensive measurements of ground-borne noise were conducted using a single locomotive as the source, driving it under the bridge deck following predefined paths and speeds. Two tracks and approximate sections of those tracks were identified as the main sources of ground-borne noise for the different buildings on the deck.

To be able to accurately define the lengths of track to be isolated and the level of attenuation required from the isolators, it was necessary to gather measurement results for ground-borne noise inside the buildings while knowing exactly where the train was located in the tunnel. The methods used in the locomotive measurements were not sufficient and so a new measurement and analysis process was developed. A custom lidar-based solution was devised to accurately locate the trains under the deck where the train GPS signal accuracy was not sufficient or even not available, while measuring ground-borne noise inside the buildings. This article presents this new measurement and analysis method, focusing on the apartment tower Opaali and track 1 below it.

2 Using a lidar for tracking train location in a tunnel

2.1 Lidar unit construction

Light detection and ranging i.e. lidar is a way to measure distances using laser pulses continuously produced at a specified rate. A lidar sensor produces laser pulses and receives them back after being reflected from an object in the line of sight of the sensor. Based on the reflected laser pulse the sensor then produces output data that can be used to determine the distance of the object from the sensor. Due to the continuous nature of the measurement, a lidar sensor can be used for automated measurements as well as for fast successive measurements, where manual operation of a laser distance meter would not be practical or fast enough. The sensor can be used to detect an object appearing in the line of sight of the sensor and to determine how long the object was present, even in cases where the object moves at a reasonable speed.

A commercially available lidar sensor was used to construct measurement units to detect bypasses of trains. The lidar units were placed in a perpendicular orientation to the tracks. As the distance measured changes, the bypass of a train in front of each unit can be detected.

Using lidar units to detect train bypasses has benefits compared to recording a video in the same location. Firstly, the distance measurement is read from the lidar sensor 250 times per second. This can provide a more accurate spatial resolution than typical video streams. For a moving train the lidar provides a spatial resolution of 1…10 centimetres, depending on the speed of the train passing the lidar.

Secondly, lidar units are not sensitive to variable or low lighting conditions in the measurement area. Video cameras have limits on the lighting conditions required to be able to record clear images. The railway tunnel in this study has almost no ambient light especially during the night, apart from the train headlights, and thus the lidar units were the only viable option for determining the required information for the train bypasses.

Thirdly, the lidar units convert the data produced by the lidar sensor into a signal that can be connected to a signal recorder. This not only enables synchronization between signals and post-processing but also saves the information in a simpler and more concise format than a video stream, making it significantly easier and faster to use in automated analysis processes.

2.2 Determination of train location

A single lidar unit can detect the presence of a train in front of it and can be used to determine when a train has exited one tunnel section and entered another one. However, the signal from a single lidar unit cannot be used to determine to which direction or how fast the train was going. In order to track the location of the train elsewhere in the tunnel the direction of travel would need to be known. Additionally, knowledge of the length of the individual train assembly would be needed for determining the speed of the train when passing the lidar. This would be based on the assumption that the train is travelling at a constant speed throughout the tunnel.

For most train services, the itinerary with pass-by times based on GPS locations and the length of the train assembly can be found from the open data available in the Digitraffic service that records train traffic information across Finland. The GPS location information tends to have gaps, and for many locomotive bypasses this information is not available at all.

Additionally, the potential change in speed cannot be detected with a single lidar unit, leaving the exact train location uncertain on many occasions.

Two lidar units were used in this study to accurately determine train speed and the direction of travel. The units were installed in the railway tunnel along the same track in predefined locations with a known distance between them. The locations were selected to represent the extremities of the apartment tower directly above the track. At these locations the deck structure also had movement joints. This arrangement made it possible to detect when a train is under the building and which direction it is moving in, and to estimate the impacts of potential speed changes. The length of the train assembly could be calculated from the lidar signals, which aided in recognising the type of train passing through. Video cameras were used as a backup alongside the lidar units to aid in determining the train type in uncertain cases.

In this case, the lidar units were able to observe trains on the two closest tracks (tracks 1 and 2). There were supporting walls and columns blocking the line of sight to the further tracks. Track 1 had previously been identified as the most significant source of ground-borne noise for the Opaali building above and so observing the trains using the other tracks was not necessary. The locations of the lidar units in the tunnel and a photo of the southern unit's line of measurement are presented in Figure 1.

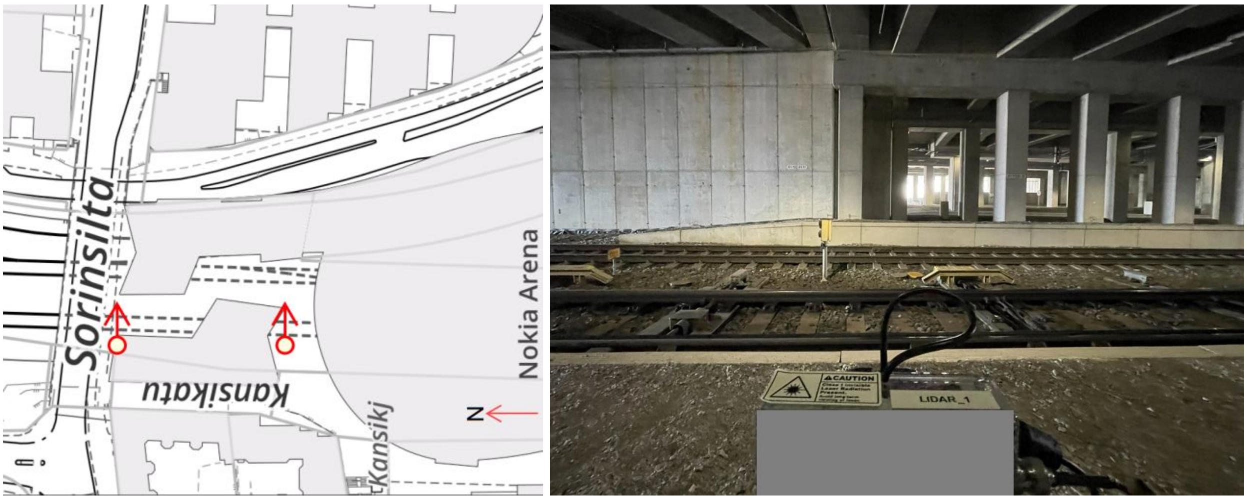

Figure 1: Locations of the lidar units in the tunnel (left) and the southern lidar unit's line of measurement (right).

3 Matching ground-borne noise levels to location

3.1 Simultaneous lidar and noise measurements

Ground-borne noise measurements were conducted inside the building simultaneously with the train bypass measurements in the railway tunnel below. Noise and vibration signals were recorded continuously using microphones and accelerometers connected to signal recorders and sound level meters around the building. This way it was possible to gather information of the ground-borne noise levels caused by each train bypass as the train travels along its track under the building. The noise and vibration measurements and signal processing were conducted following the standard ISO 14837-1:2005 [1] and the preliminary study on traffic-induced ground-borne noise (Research Notes 2468) [2] by the Finnish state-owned research institute VTT. The VTT preliminary study is typically applied similar to local regulation.

The room spaces selected for the noise measurements in the Opaali tower were apartments and apartment hotel rooms on the lowest floors where these types of spaces are located, and where local limits for traffic ground-borne noise level had been observed to be exceeded in previous noise measurements. Apartments and hotel rooms are located starting from the 3rd floor. Ground-borne noise levels were measured in each selected room.

The measurement duration was approximately 3–4 days to accumulate a sufficiently large sample of bypasses for train units in each train class. A large sample is required to be able to establish the long-term statistical distribution of ground-borne noise levels caused by train bypasses as different train units have a large variance in the factors contributing to the ground-borne noise level produced, such as the assembly of the train and the condition of individual wheelsets.

3.2 Combining lidar and noise level data

The train location and ground-borne noise measurements were conducted using numerous separate devices. Ground-borne noise was measured in multiple apartments and hotel rooms across the floor area. Combining the signals from all these devices and correctly matching the ground-borne noise level signals to the train location information required the signal recorders to be carefully time synchronised.

Representative samples of train bypasses with no noise disturbances from other sources were extracted for further analysis from the synchronized signal sets. The lidar and ground-borne noise level signals were then plotted in the same graph, where the time axis was converted into location in the tunnel utilising the known distance between the two lidar units. An example of this type of graph for a locomotive bypass is shown in Figure 2.

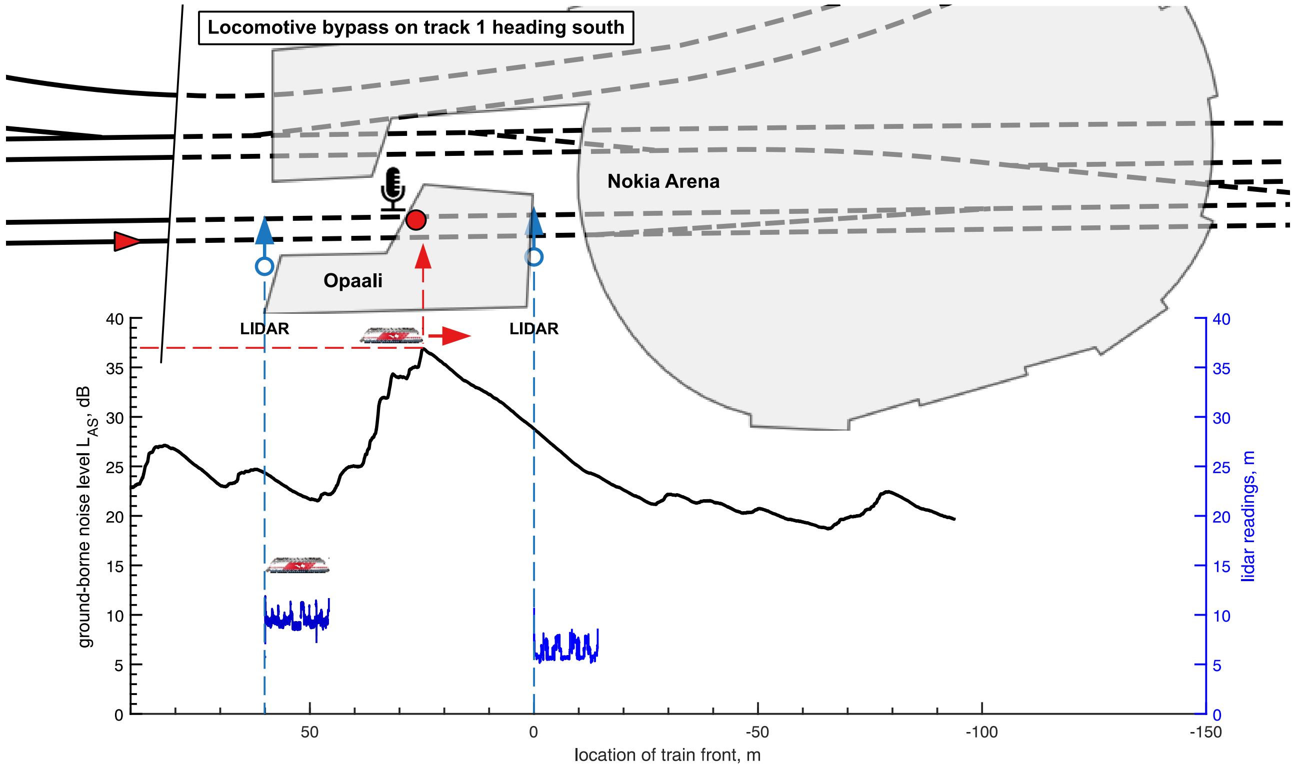

Figure 2: Ground-borne noise levels produced by a locomotive traveling south on track 1 as a function of the corresponding location of the front of the locomotive, which has been matched to the map view. The lidar curves describe the lateral distance measured by each lidar unit as the front of the locomotive moves along the tracks. The ground borne noise levels were measured on the 3rd floor of the building; the measurement position is depicted by a red circle. The blue arrows show the locations of the lidar units in the tunnel. The location where the locomotive caused the maximum noise level has been marked with a red dashed line. The distance axis points north, thus the inverted values. The measurement corresponds to the initial situation prior to isolating the track.

4 Optimising railway track vibration isolation

4.1 The design process

Optimising the vibration isolation of a railway track in simple terms means knowing where isolation is needed and what the minimum attenuation required from the solution is for each section of the track. This information can be determined from the combined train location and ground-borne noise level graphs from which it is easy to see where the train is when the ground-borne noise level exceeds the local criteria. For apartments in Tampere Deck and Arena this criterion is $L_{ASmax} \leq 30$ dB [3], which in practice means that the A and Slow weighted maximum noise level should be less than 30 dB during 95 % of train bypasses. This quantity is evaluated over 1/3-octave bands 16–500 Hz.

Determining the track sections where vibration isolation was needed was done mainly using samples of locomotive bypasses. Locomotives were selected for this purpose to gain an understanding of which specific sections of the tracks were the most likely to result in higher ground-borne noise levels in the different spaces inside the building above. Locomotives give the best location resolution since they are relatively short, resembling the most a moving point source of vibration on the track, compared to longer trains. From Figure 2, it can be seen that the most significant source of ground-borne noise for the particular noise measurement position is the track section approximately in the middle of the building between locations 10 m and 35 m. Similar results were obtained for other noise measurement positions.

As the ground-borne noise levels caused by locomotives are not representative of the levels caused by full trains, locomotive bypasses cannot be directly used to estimate the minimum attenuation required in each track section. Attenuation requirements were assessed by inspecting large samples of InterCity and freight trains, as they represent the vast majority of the traffic in the railway corridor. Due to their length, trains simultaneously excite longer parts of track than locomotives, resulting in a superposition of ground-borne noise produced on multiple shorter track sections all at once. The type of an individual train can be identified from the lidar signal shape and validated against the Digitraffic open data. An example of a combined location and ground-borne noise level graph for an InterCity train travelling south on track 1 is shown in Figure 3, similar to Figure 2. The highest ground-borne noise level is produced when the locomotive pushing the train is near the centre of the building.

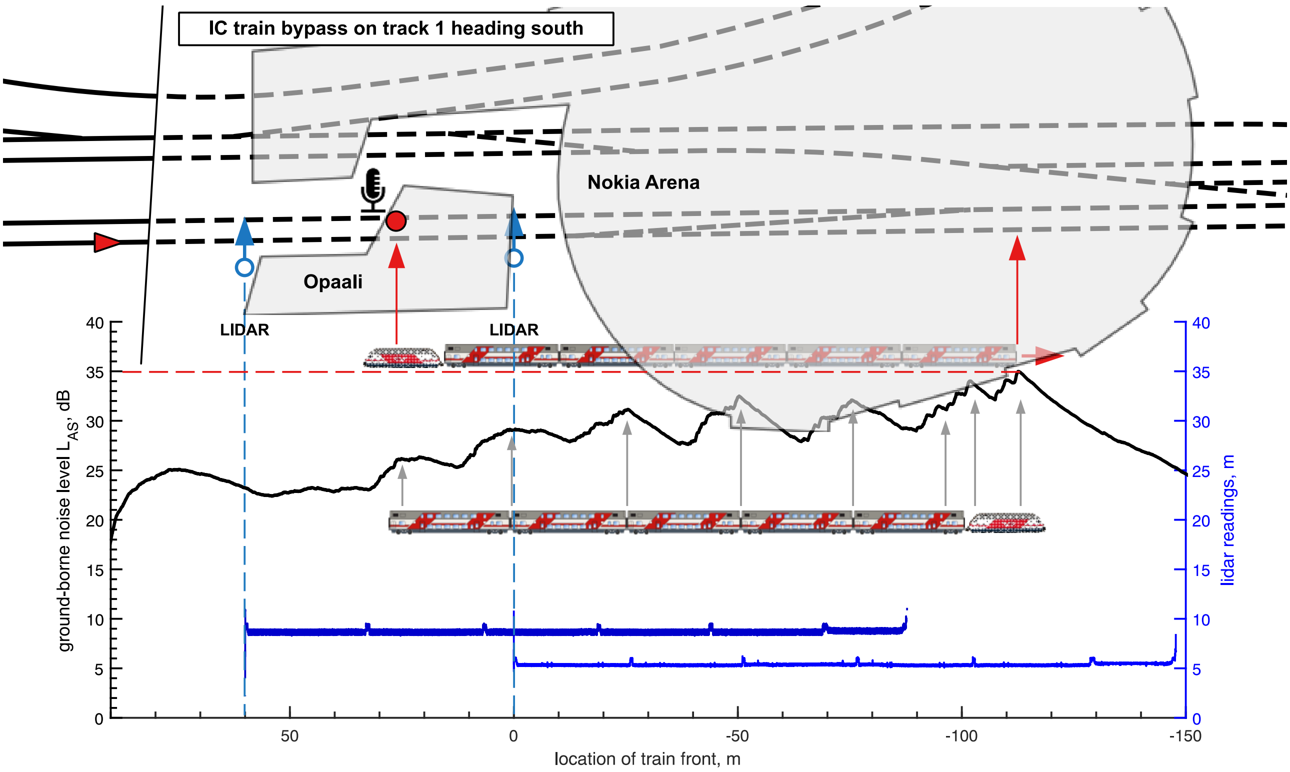

Figure 3: Ground-borne noise levels produced by an InterCity train traveling south on track 1 as a function of the corresponding location of the front of the train, which has been matched to the map view. The lidar curves describe the lateral distance measured by each lidar unit as the front of the train moves along the tracks. The ground borne noise levels were measured on the 3rd floor of the building; the measurement position is depicted by a red circle. The blue arrows show the locations of the lidar units in the tunnel. The location where the train caused the maximum noise level has been marked with a red dashed line; this corresponds to the locomotive pushing the train passing below the building. The bogies of the train produce local maxima in the noise level as they pass under the building (grey arrows). The distance axis points north, thus the inverted values. The measurement corresponds to the initial situation prior to isolating the track.

Graphs similar to Figure 2 and Figure 3 were plotted for a number of locomotives and longer train assemblies running on track 1, and for all the individual noise measurement positions within the building. Based on these graphs, and the measured spectrum information and variation in ground-borne noise produced inside the building by the various trains, it was determined that the area between locations 5 m and 70 m needs vibration isolation, and that the requirement for insertion loss should be 10 dB at frequencies 50–100 Hz. The mitigation criteria included a safety margin taking into consideration the practical limitations of the installation conditions and the implementation of the isolator assembly.

A switch area in the tracks effectively limited the track section where vibration isolation could be installed. It would have been desirable to extend the isolation there, with some locomotive bypasses showing relatively high ground-borne noise levels. Calculations were done to estimate the impact of the vibration isolation of the selected track sections on the resulting ground-borne noise levels to confirm sufficient overall attenuation was achievable even without isolating the switch area.

Due to the relatively large attenuation requirement, under-ballast mats were selected as the isolation method. The area to be isolated included the necessary transition zones with stiffer isolation material between the softer principal isolation material and the normal ballast base. Limits for static deflection of the rail surface under train axle loads were estimated and accounted for in the isolation solution. The design was done in collaboration with the material manufacturer, the track designer and the Finnish Transport Infrastructure Agency, which is responsible for railway tracks in Finland.

4.2 Resulting noise levels and attenuation

After the vibration isolation had been installed under the defined sections of track 1, the ground-borne noise and lidar measurements were repeated to confirm the noise levels now complied with the criteria. The same analysis and assessment process was performed with this new data, including the combined train location and ground-borne noise level graphs. It was concluded that the ground-borne noise levels had been sufficiently attenuated to meet the noise level criteria.

An example of the combined graph after the vibration isolation is shown in Figure 4 for a locomotive bypass on track 1. Comparing these results after the isolation to the results before (Figure 2), the ground-borne noise levels decreased significantly. The maximum ground-borne noise level is now caused at a new location, as the locomotive passes the southern lidar at 0 m and proceeds south to the unisolated switch area.

The insertion loss for the ground-borne noise levels inside the building as a result of the vibration isolation cannot be directly assessed by comparing Figure 4 to Figure 2, as the locomotive bypasses might not have been identical. The ground-borne noise levels inside the building also did not have a large enough signal-to-noise ratio to be able to fully assess the differences in the noise levels caused by the trains, even from a larger sample of different types of bypasses. The insertion loss was, however, estimated to be approximately the required 10 dB.

Instead, the insertion loss of the vibration isolation was estimated for vibration levels measured from a support wall next to track 1 in the tunnel. The vibration levels were measured at the same time during the ground-borne noise measurements, using accelerometers attached to the support wall. The levels and their spectra were calculated for a set of InterCity and Pendolino trains travelling on track 1 both before and after the installation of the under-ballast mats. These values were compared between corresponding train assemblies in each 1/3-octave band to then establish the median differences between the spectra. The median differences approximating the insertion loss describe how much less vibration propagates into the bridge deck structure after the installation of the under-ballast mats. The median insertion loss was estimated to satisfy the requirement of 10 dB at frequencies 50–100 Hz.

5 Conclusions

In this study, a measurement and analysis method combining train location in a railway tunnel to the ground-borne noise level produced by the train in a building directly above the track was developed. The presented approach was required for assessing and optimising the extent of track to be vibration isolated and for estimating the attenuation required. In the tunnel, the GPS-based train location system cannot provide precise information on train location. The poor lighting conditions of the tunnel environment severely limit the frame rate and usable image quality of video recordings.

The train location in the tunnel was determined using two lidar units in known locations along the investigated track. Simultaneous and synchronous recording of the lidar signals and the ground-borne noise signals inside various room spaces inside the building made it possible to evaluate the exact train location and the corresponding ground-borne noise levels in each room at each moment in time. The lidar unit data resulted in location information with a few centimetres' precision and in a format that was easy to record unattended and post process later together with the noise data.

With the help of this method, the exact track section in need of vibration isolation could be determined, thus ensuring the desired end result of attenuating the ground-borne noise levels to comply with the criteria, but without isolating more track than necessary. The minimum required attenuation for the isolation material could also be defined. Additionally, the data gathered in these measurements could be further utilised in modelling the impact of the vibration isolation with the selected products.

Similar location and ground-borne noise measurements were conducted after the vibration isolation had been implemented under the defined track section. It was concluded that the ground-borne noise levels caused by the train traffic in the railway tunnel now comply with the criteria. The insertion loss provided by the vibration isolation was estimated from the vibration levels entering the bridge deck structure and was similar to what was required from the isolation materials.

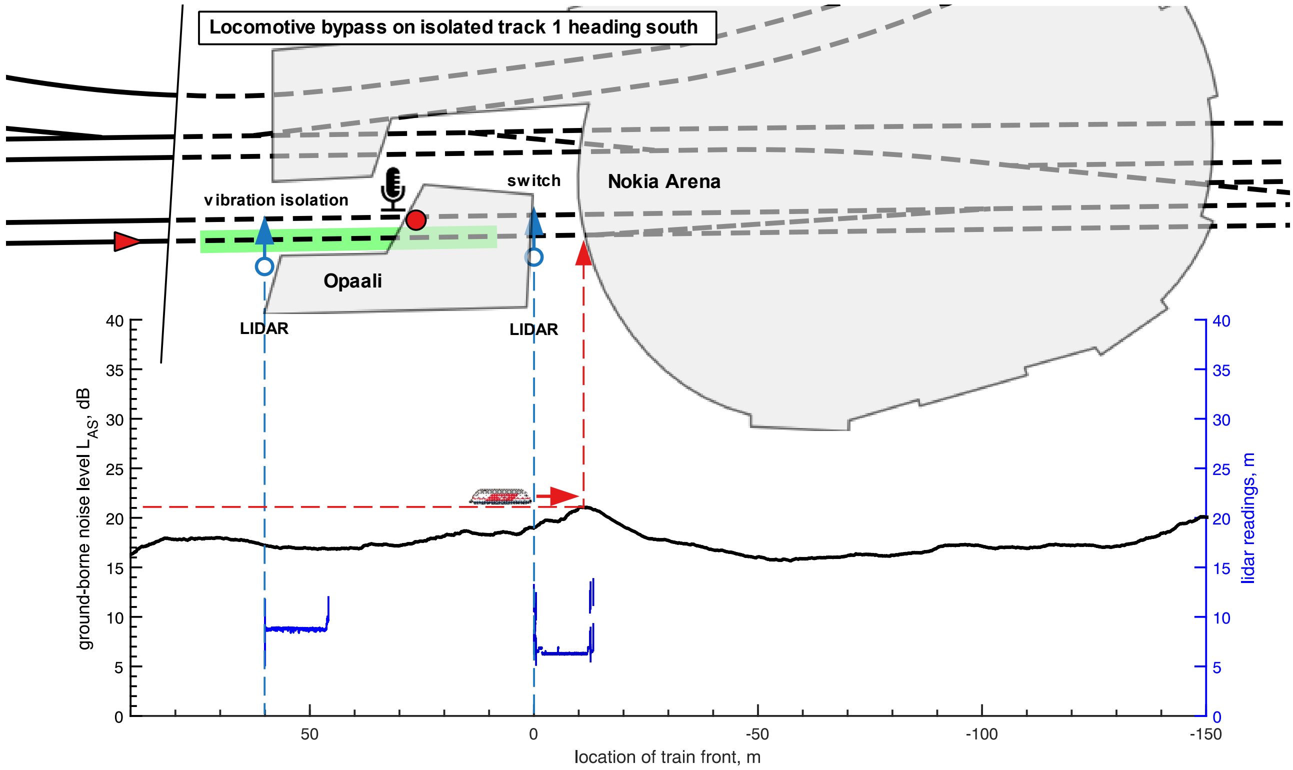

Figure 4: Ground-borne noise level produced by a locomotive traveling south on track 1 and the corresponding location of the front of the train assembly **after implementing the vibration isolation under the track**. The isolated track section has been highlighted in green.

References

[1] *Mechanical vibration – Ground-borne noise and vibration arising from rail systems – Part 1: General guidance* ISO 14837-1:2005, International Organization for Standardization, Switzerland, 2005.

[2] A. Talja and A. Saarinen, *Assessment of traffic-induced ground-borne noise. Preliminary study,* VTT Tiedotteita – Research Notes 2468, VTT, Espoo, 2009.

[3] City of Tampere city plan map 8366, approved by the city council 13.6.2011.

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Pile supported slab mitigating vibration under and next to a railway line

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Pile supported slabs are commonly used as foundations for railway tracks when the soil might be subsiding or the ground under the track is generally weak in stability. It is generally recognized that a pile supported slab under railway track mitigates emitted vibrations. It has also been considered that a pile supported slab next to a railway track could act as a wave barrier. This is the case when new rails founded on pile supported slab are constructed next to an existing track. In this study these effects were evaluated by measurements. The research sites were located in Finland on the Seinäjoki-Oulu railway track, near Ruha, at track kilometer 431+500-432+700. The measurements were made in two sites representing different soils in the summer of 2023. At the site there are two sets of rails next to each other: one founded on soil and the other on a pile supported slab. It was noticed that vibration from the track founded on a pile supported slab is smaller than from the track founded on soil. A commonly used estimate for piled foundations was shown to be good. The measured vibration was 60...90 % lower from a track founded on pile supported slab. The measurement results indicated that track on a pile supported slab could act as a wave barrier and therefore reduce the vibrations emitted to residential buildings next to the railway. The effect is greatest near the track and in the frequency ranges where most of the vibration energy is concentrated.

1 Introduction

Pile supported slabs are commonly used as foundations for railway tracks when the soil might be subsiding or the ground under the track is generally weak in stability. It is generally recognized that a pile supported slab under railway track mitigates emitted vibrations. It has also been considered that a pile supported slab next to a railway track could act as a wave barrier. This is the case when new rails founded on pile supported slab are constructed next to an existing track. In this study these effects were evaluated by measurements.

2 Measurements

2.1 Measurement sites and rolling stock

The research sites were in Finland on the Seinäjoki-Oulu railway track, near Ruha, at track kilometer 431+500-432+700. The measurements were made in two sites representing different soils in the summer of 2023. At the site there are two sets of rails next to each other: one founded on soil and the other on a pile supported slab.

In the first research site (site 1) the soil is silt, clay, clayey silt or muddy silt/clay. The soft layer extends at least 10-15 meters deep from ground surface. In the second research site (site 2) the soil consists of 3-5 meters deep layer of peat. Below that there is a silt or clay layer at least 10 meters deep. In this site the soft layer extends considerably deep.

In both sites soil conditions were assumed to be similar outside the railway area than under a track. Currently, the adjacent areas are in agricultural use. In both sides of the track there is a service road with a gravel surface.

In both sites there was a double-track railway consisting of eastern track (ET) and western track (WT). Both tracks were ballast tracks. The eastern track was founded on pile supported slab. The western track was founded directly on ground. In total 20 trains passed by during measurements in site 1, and 24 during measurements in site 2. The train types were InterCity (IC), Pendolino (PEN), freight train (T) and night train (PYO). The trains are shown in Table 1.

Table 1: List of trains during measurements

Track	Site 1 (June 6-7)				Site 2 (June 7-8)			
	IC	PEN	T	PYO	IC	PEN	T	PYO
Eastern track	4	2	6	-	1	2	9	-
Western track	6	2	-	-	8	2	-	1

2.2 Equipment and analysis

The measurements were conducted on the ground surface using accelerometers. The measurements were carried out in 3 directions (x = parallel to the track, y = perpendicular to the track, z = vertical direction). The data was recorded in wave format so any required analysis could be done afterwards.

The measurement data was filtered using W_m-weighting according to ISO 2631-2 and maximum RMS values (1 second) were used in assessment [1]. The analysis was also carried out in 1/3 octave bands in 1…250 Hz. In frequency band analysis it was decided to use the average of 5 seconds around the maximum value to get more stable and consistent results.

2.3 Measurement points

When assessing the performance of a pile supported slab as a wave barrier, measurements were conducted simultaneously on both sides of the track. All train passages were in the western track. The measurement points are shown in Figure 1 and they were both sides of the track, 23 meters (MP2, MP3) and in 38 meters (MP1, MP4) from the source (WT). Each train passage was studied individually and the difference between results in measurement points in equal distance was assessed. The trains passages that were used in the analysis were 4 IC and 1 PEN in site 1 and 7 IC, 2 PEN and 1 PYO in site 2. Since all freight trains were passing by in eastern track, it wasn't possible to use them in the analysis.

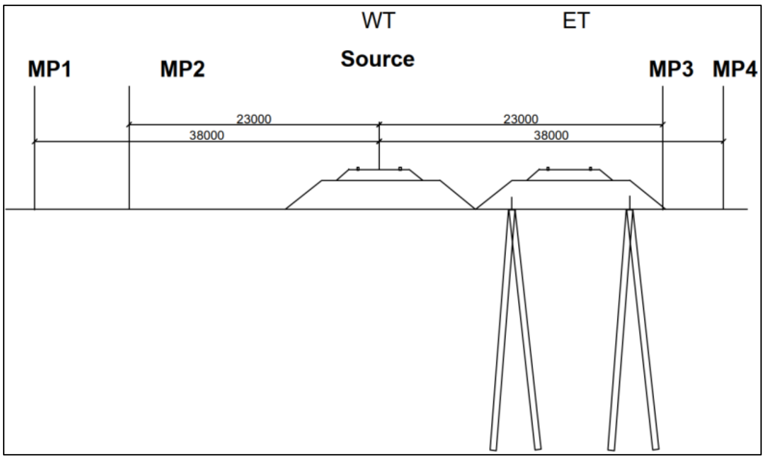

Figure 1: Measurement points when assessing pile supported slab as a wave barrier

When assessing the performance of a pile supported slab as track foundations, measurements were conducted on both sides of the track. Train passages in eastern track were measured in measurement points MP4 and MP5 and respectively passages in western track were measured in measurement points MP1 and MP2. The measurement points are shown in Figure 2 and they were 23 meters (MP2, MP4) and in 38 meters (MP1, MP5) from the source track (WT / ET). The measurement results were averaged by train type and after that results in MP2 were compared to results in MP4 and respectively results in MP1 were compared with results in MP5.

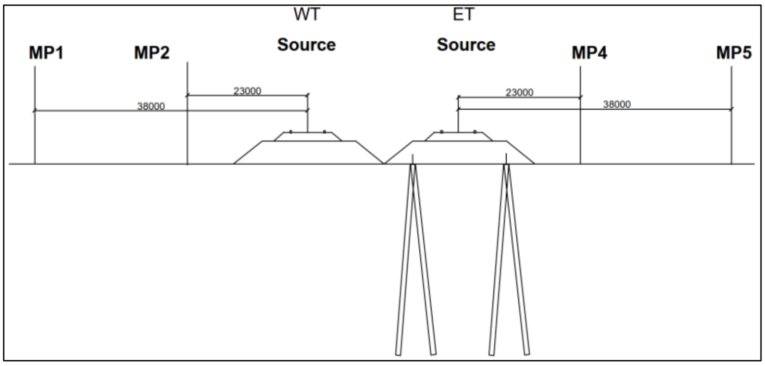

Figure 2: Measurement points when assessing pile supported slab as track foundations

The trains that were used in the comparison were:

- Site 1: 5 IC in western track vs. 4 IC in eastern track
- Site 1: 2 PEN in western track vs. 1 PEN in eastern track
- Site 2: 7 IC in western track vs. 1 IC in eastern track
- Site 2: 2 PEN in western track vs. 2 PEN in eastern track

3 Results

3.1 Pile supported slab as a wave barrier

Measurement results in MP2 were compared to results in MP3 and respectively MP1 to MP4. The differences between the measurement points are considered to represent the effect caused by the pile supported slab. The differences averaged by train type are given as percentages in Table 2 and Table 3. Negative values indicate that the level in east side of the track is smaller, which suggests that the pile supported slab would act as a wave barrier and attenuate vibration.

Table 2: Difference in vibration in two measurement distances, site 1

Train type	23 m distance (MP3-MP2)			38 m distance (MP4-MP1)		
	x	y	z	x	y	z
IC	-6 %	-4 %	-40 %	-11 %	-12 %	-14 %
PEN	28 %	-26 %	-26 %	-31 %	-7 %	26 %
Avg. all trains	1 %	-9 %	-37 %	-15 %	-11 %	-6 %

Table 3: Difference in vibration in two measurement distances, site 2

Train type	23 m distance (MP3-MP2)			38 m distance (MP4-MP1)		
	x	y	z	x	y	z
IC	-81 %	-69 %	-66 %	-56 %	-25 %	-65 %
PEN	-77 %	-75 %	-81 %	-46 %	41 %	-63 %
PYO	-81 %	-78 %	-72 %	-	-68 %	-80 %
Avg. all trains	-80 %	-71 %	-70 %	-54 %	19 %	-67 %

Measurement results in east side of the track are significantly smaller results in many cases. The effect is clearer in vertical direction than in horizontal directions. Also, the effect is greater in small distances and when the ground is softer. In horizontal direction it seems that there might be also some amplification.

In Figure 2 all the measurement results in MP3 and MP4 are given in relation to results in MP1 and MP2. As the Finnish guideline in most cases is 0,3 mm/s, it has been considered in the following.

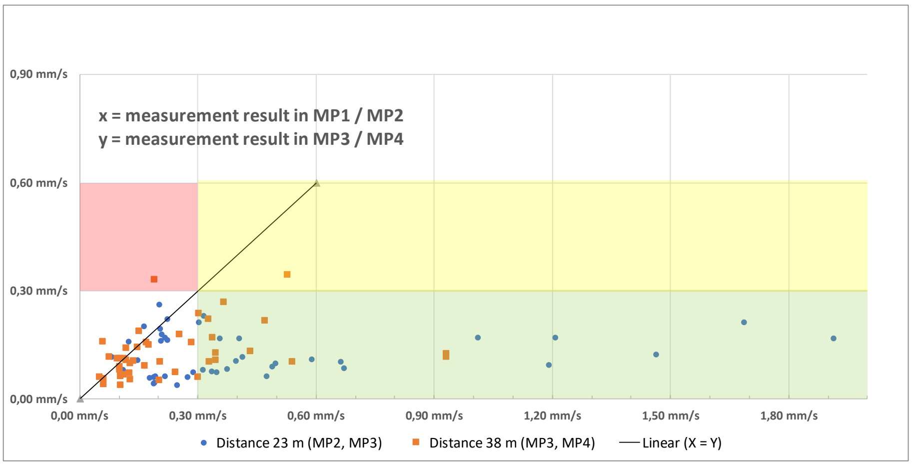

Figure 2: Measured vibration in MP3 and MP4 (east side of the track) in relation to corresponding measurements in MP1 and MP2 (west side of the track). In the white area (bottom left corner) results are less than 0,3 mm/s in both sides of the track. In green area results are more than 0,3 mm/s in MP1/MP2 and less than 0,3 mm/s in MP3/MP4. In yellow area results in MP3/MP4 are smaller than in MP1/MP2, but still over 0,3 mm/s. In red area one result in MP3/MP4 is over 0,3 mm/s while in MP1/MP2 less than 0,3 mm/s.

In Figure 2 many of the results are less than 0,3 mm/s in both sides of the track and therefore they are not very interesting. However major part of the results is under the black line (x = y), which means that most results are smaller in eastern side than western side. This could be interpreted so that the pile supported slab would act as a wave barrier and vibration attenuation would have been achieved. Significant part of the results is over 0,3 mm/s on west side of the track, but less than 0,3 mm/s on east side of the track (green area). Also, one result is more than 0,3 mm/s in both sides of the track, but still smaller in eastern side (yellow area). One of the results is in red area, which means that the result is more than 0,3 mm/s in east side of the track while less than 0,3 mm/s in the western side. This could indicate that in one case the presence of pile supported slab had amplified vibration causing it to exceed the national limits. This particular result has been measured in site 2, 38 m from the track.

In Figure 3 the difference in vibration as percentages in each measurement is shown in relation to the measured vibration on west side of the track (MP1 and MP2). Here it can be seen that the attenuation is dependent on the vibration level and so it is nonlinear. The attenuation is greater when the vibration is greater, and it seems to saturate to approximately 90 %.

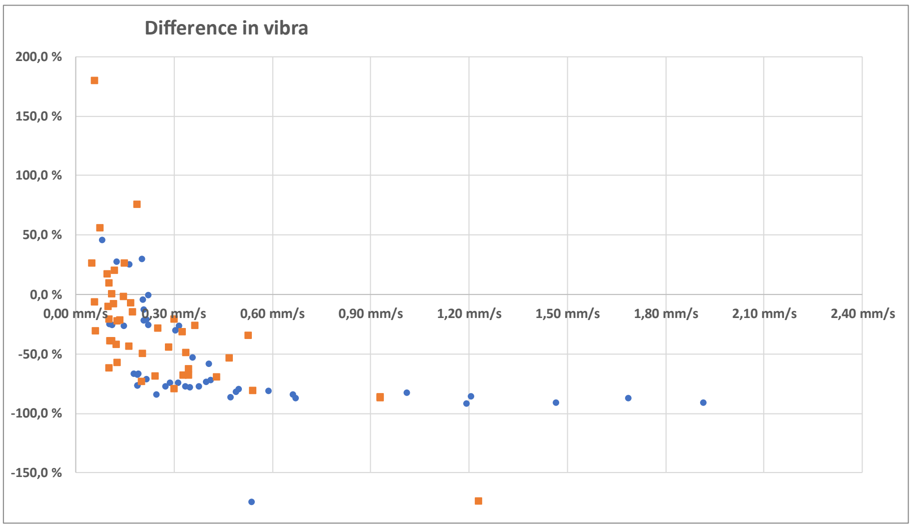

Figure 3: Difference in vibration in relation to the measured vibration on west side of the track (MP1 and MP2).

The measured vibration on different sides of the track and their difference in 1/3 octave bands is shown in Figures 4 and 5. The figures include all assessed trains, and the data is shown as absolute values (mm/s). Most of the previously noticed amplifications seem to be insignificant and the greatest differences are observed in frequency bands where the vibration is greatest.

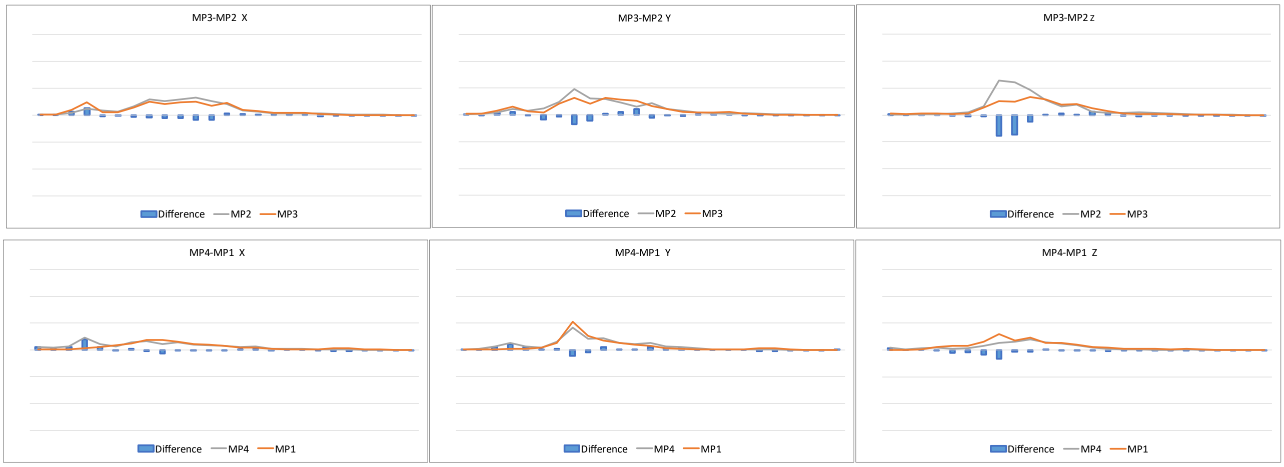

Figure 4: Measured vibration on different sides of the track and their difference in 1/3 octave bands, site 1.

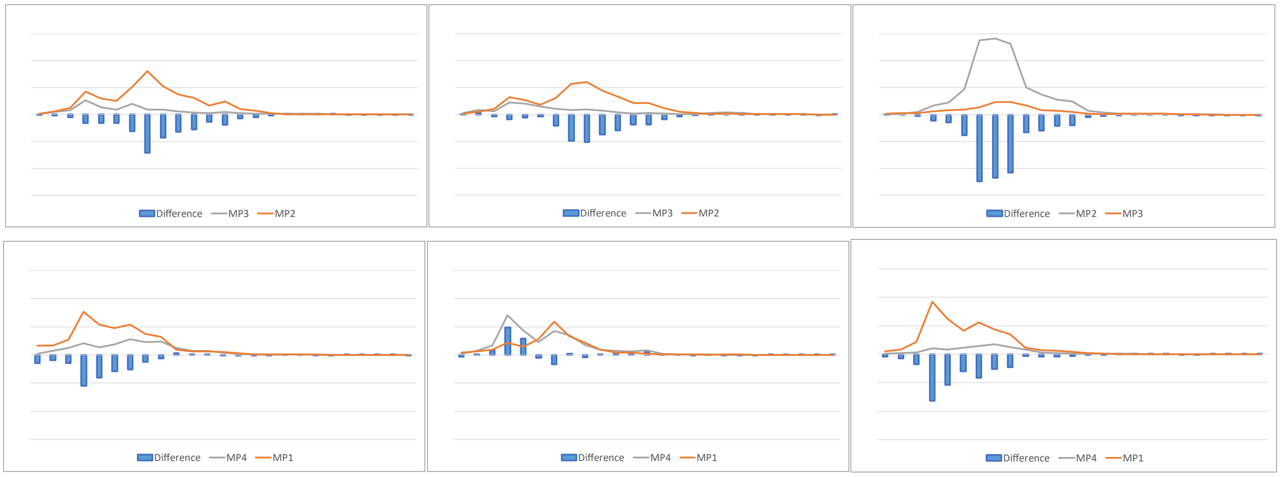

Figure 5: Measured vibration on different sides of the track and their difference in 1/3 octave bands, site 2.

3.2 Pile supported slab as a substructure

Train passages in eastern track measured in measurement points MP4 and MP5 were compared to passages in western track measured in MP1 and MP2. The differences between the measurement points are considered to represent the vibration attenuation achieved by using pile supported slab as a substructure. These attenuation values are given as percentages in Table 4 and Table 5. Negative value indicates that the level in east side of the track is smaller than west side, which suggests that the pile supported slab as a substructure mitigates vibration.

Table 4: Difference in vibration in two measurement distances, site 1

Train type	23 m distance (MP4-MP2)			38 m distance (MP5-MP1)		
	x	y	z	x	y	z
IC	-17 %	-46 %	-67 %	8 %	-51 %	-38 %
PEN	-9 %	-4 %	-37 %	39 %	9 %	-28 %

Table 5: Difference in vibration in two measurement distances, site 2

Train type	23 m distance (MP4-MP2)			38 m distance (MP5-MP1)		
	x	y	z	x	y	z
IC	-87 %	-73 %	-96 %	-91 %	-83 %	-91 %
PEN	-46 %	-12 %	-85 %	-23 %	-34 %	-78 %

Measurement results from traffic in track with pile supported slab give significantly smaller results in many cases. The effect is clearer in vertical direction than in horizontal directions. Also, the effect is greater in small distances and when the ground is softer. It seems that in horizontal direction there might be also some amplification. The results in 1/3 octave bands showed similar behaviour than in the case when pile supported slab was assessed as a wave barrier. The amplification is typically not present in significant frequency bands and the greatest attenuation is observed in frequency bands where the vibration is greatest. Measurement results from site 2 are shown in Figure 6 and the the rest of the data can be found from the publication by Finnish Transport Infrastructure Agency [2].

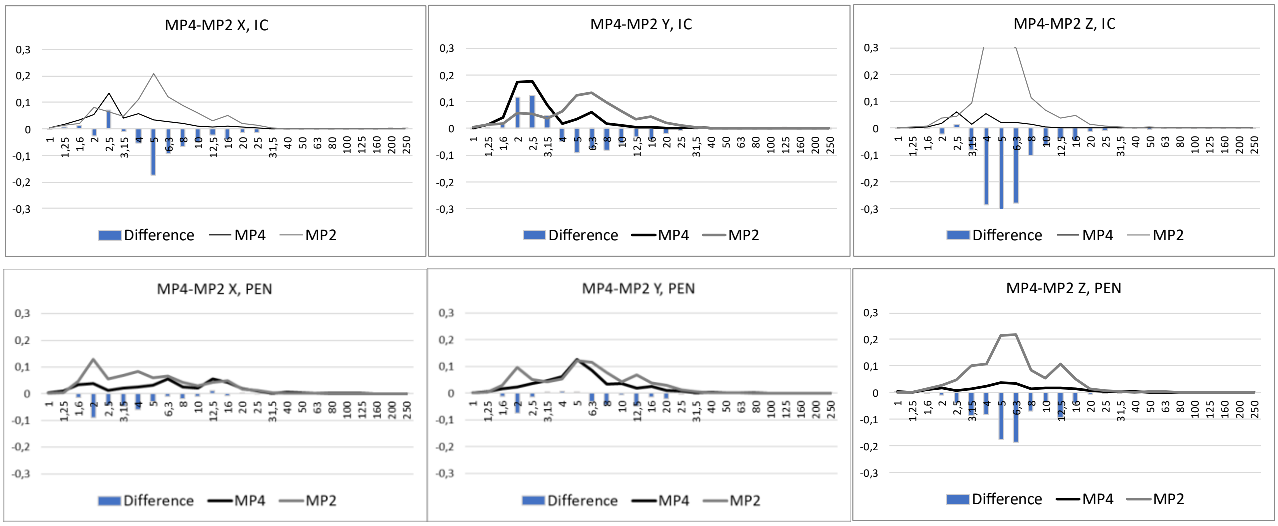

Figure 6: Measured vibration on different sides of the track and their difference in 1/3 octave bands, site 2.

4 Conclusions

This study indicated that a track on a pile supported slab could act as a wave barrier and therefore reduce vibrations emitted from adjacent track to buildings next to the railway. The effect is greatest near the track and in the frequency ranges where most of the vibration energy is concentrated. Attenuation seems to be nonlinear.

It was also noticed that vibration from a track founded on a pile supported slab is smaller than from a track founded directly on soil. A commonly used estimate for piled foundations was shown to be good. The measured vibration was 60...90 % lower from a track founded on pile supported slab. This study was funded by the Finnish transport infrastructure agency and the full report was published in January 2024 [2].

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Sound levels in symphony orchestra musicians

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A symphony orchestra is a complex and powerful sound source, where musicians are both sources and receivers. Instant levels vary a lot from one moment to another, and equivalent levels varies significantly from piece to piece, from one position to another, from one orchestra to another playing the same piece, from one concert hall to another, and so on. Sound level and exposure management has several artistic advantages beyond hearing concerns and should be based on long-term a perspective with a random point sample program for dosimetry over year cycles. As a reference for "normal" exposure, the statistics from more than 1600 dosimeter measurements over 3 years in the Queensland Orchestra is suggested. This paper includes results from etry and other measurements and observations during rehearsals and performances, together with analysis and recommendations for further work in the field. The sound of music in an orchestra is not unwanted and not be handled like noise. Use hearing protection, but not more than needed.

1 Introduction

A symphony orchestra is a complex and powerful sound source, where musicians are both sources and receivers. Instant levels vary a lot from one moment to another, and equivalent levels varies significantly from piece to piece, from one instrument type to another, from one position to another, from one orchestra to another playing the same piece, from one concert hall to another, and so on. Musicians are often concerned about their hearing and the risk of hearing loss, and at least one orchestra has been convicted legally responsible for the hearing loss of a musician. Different kinds of hearing protection are being used, but they come with downsides. Noise is un-wanted sound, while music is not. Still, the sound of music is commonly treated like industrial noise, and its exposure levels compared with limits intended for industry workers. Is this common practice justified? What are the common exposure levels anyway? In general, how do sound levels vary? What are the causes of varying sound levels? Are there other adverse effects from sound levels than health risks? These and other questions are being addressed. Common interpretation of occupational noise regulation limits is at odds with reality in orchestras.

2 Sound levels and level variation from symphonic music

Symphonic music can cause instant A-weighted sound pressure levels (SPL, or L_{pA}) ranging from 20dB to 140dB, depending on time and position in the concert hall or rehearsal room. In various statistical categories, however, the range can be much smaller, and the sound levels can be statistically predictable in terms of probability. This section describes statistical properties in various categories observed from field measurements during symphony orchestra rehearsals and performances.

2.1 Data

2.1.1 O'Brien etry data

A major external source of field measurement data is from the work of O'Brien et.al., reported in 2008, consisting of etry data with equivalent L_{Aeq} and L_{Cpeak} from a total of 1608 sessions, lasting 2.5-3.0 hours, in The Queensland Orchestra, hereafter QSO, over the seasons in the 3-year period 2005-2007 [1]. One half of the sessions are concert performances, mostly in the concert hall of the Queensland Performing Arts Center (QPAC), and the other half partly orchestra pit sessions in the Lyric Theatre and rehearsals in the QPAC Rehearsal Studio. All the raw data has been shared with this author for further and independent analysis.

2.1.2 Own data

Included in the volume of field measured data acquired by this author are etry data in two Norwegian symphony orchestras, in this paper referred to as Orchestra A and Orchestra B, or just A and B, together with sound level data in selected fixed stage and auditorium positions. These measurements include level-time data in 1 second resolution from eters (Casella dBadge) and sound level meters (Norsonic N-140) in metrics $L_{pA,eq}$ and $L_{pC,peak}$ from each session, where some sessions are rehearsals, but most of them are concert performances. Moreover, own data includes 30 hours of binaural data recorded in concert listener positions during concerts with various orchestras in various halls in Europe and USA, i.e. in the Binaural Project[2].

2.1.3 Other

From the Binaural Project, there is more than 1 million sound pressure levels measured in 100ms windows from symphonic music in listening positions. A total of 114 measurements covering 106 h were recorded in musicians in two symphony orchestras, analysed and reported by Schmidt et.al.[3].

Wenmaekers has developed a prediction model for sound levels in musicians in symphony orchestras, published in a series of papers and with data collected in a book [4] (2017), mentioned here for any readers who are interested in comparing predicted and measured results in comparable metrics, in particular the variation in exposure levels over various positions in a symphony orchestra.

2.2 Level distributions and statistical sub-populations

Data can be viewed as a global population of all instant sound pressure levels or as sub-populations based on relevant categories of data. Thus, data from all instants in one instrument or orchestra position could form a sub-population of data. Another population could be the population of sessions in each instrument, based on its $L_{pA,eq}$, $L_{pC,peak}$ or other statistics. Yet another population could be the population of instruments, e.g. their $L_{pA,eq}$ values in one session or any number of sessions. Each population and sub-population can be relevantly described by common statistics like its average, standard deviation, percentiles, skewness of gaussian distribution, and so on, and graphically by e.g. its histogram.

Typically, the global population of all data, with instant levels at all times and all positions, has a gaussian or near-gaussian distribution with the bell-shape slightly skewed to the right in a histogram. In listening positions,

2.2.1 Metrics

Measured sound level metrics $L_{pA.eq}$, $L_{pC.peak}$ from sessions (rehearsal or performance), and instant levels from level-time data in terms of LpA in 1 second resolution, and broadband (0.4-2.5kHz) levels in 1 second, L_{1s} , and 100 milliseconds, L_{100ms}, all in dB, form the database for the work presented in this paper.

2.2.2 Global statistics - listening levels in live symphonic music

Table 1 presents statistics from all instant broadband levels with 100ms resolution measured in listening positions during symphonic music performance, data collected in the Binaural project. N is the count of instants/events, Leq is the equivalent level (level of average energy), L88%, L75%, L50% and L25% are percentile levels, m the average level, s the standard deviation and skew the skewness of the bell shape. All levels are in dB. Figure 1 is the histogram of the global population of broadband listening levels, where the skewness of the bell shaped gaussian distribution can be observed by its somewhat longer tail to the left and shorter to the right. The most frequently occurring level, i.e. the *mode*, is observed to be somewhere in the 63-69dB range. Note that the equivalent level equals the 88-percentile, demonstrating that equivalent level is dimensioned by the louder members of the population.

Table 1 Statistics of listening levels during live symphonic music, see text

N	L_{eq}	L88%	L75%	L50%	L25%	m	s	skew
1011189	80	80	73	64	54	63	15	-0.5

For the purpose of this paper, the skewness of the distribution will be evaluated by the following rules of thumb: 0-0.5 is close to symmetry, 0.5-1.0 is moderately skewed, while >1.0 is highly skewed, and of course the sign of the value determines whether the skewness is positive or negative.

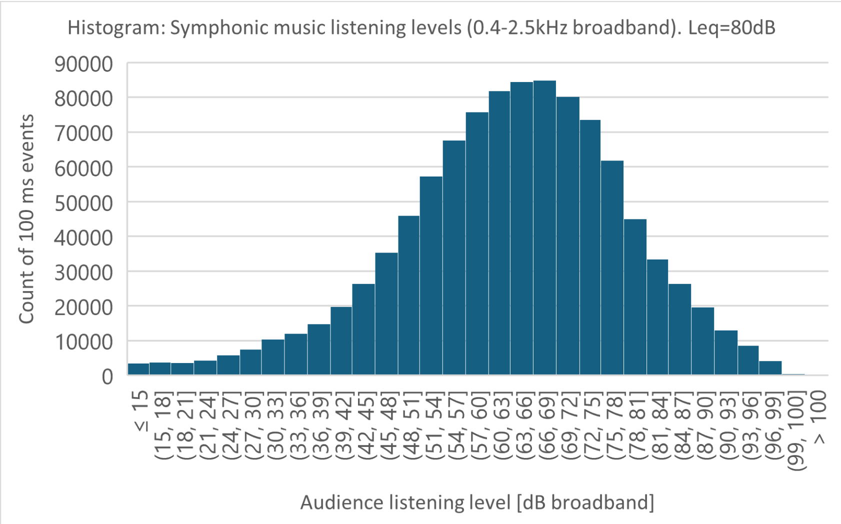

Figure 1 Histogram of 100ms events of broadband sound pressure levels from symphonic music in listening positions

2.2.3 Concerts as sub-populations

If the global data above is grouped into sub-populations of data from the individual concerts, there would an individual set of set of statistics for each concert, e.g. $m_{concert}$ and $s_{concert}$. From the data, $m_{concert}$ would normally be somewhere in the 56-70dB range and $s_{concert}$ in the 10-14dB range, thus a typical concert would have a high likelihood of levels 63±12 dB, i.e. a normal range of 51-75dB. From this we note that the spread of the global data is partly made up by the level variation within in each concert, partly by the variation of average level from one concert to another.

2.2.4 Levels in performers versus levels in listeners

At performers positions, the level distribution profile is observed to be quite equal to that of the listening positions, only with averages 5-7dB higher on stage due to attenuation from stage to the listening positions in the audience area. Similarity of distribution profiles is to be expected since high quality sound transmission is essential in concert halls, and both are determined by the music.

2.2.5 L_{Aeq}, L_{Amax}, and L_{Cpeak} in musicians

In occupational noise & health concerns, the daily or weekly dose of sound exposure is often considered, in addition to the loudest sound during the exposure period.

A daily (permissible) dose is commonly defined as equivalent to the sound exposure from an A-weighted sound pressure level of 85 dB with a duration of 8 hours, denoted $L_{pA,eq,8h} = 85\ dB$. For this reason, $L_{pA,eq}$ is a relevant metric. Moreover, it has been and still is a common, practically hard-wired, metric in measuring equipment, which over the years has led to an accumulating volume of measurement data, reference values and limits based on L_{pA}. The equivalent level is obtained by integrating the squared sound pressure and divide by the integration time.

In order to manage the exposure from very loud events, two metrics have been used, i.e. $L_{pFA.max}$ and $L_{pC,peak}$. The former is the highest A-weighted level measured with the time constant “Fast”, i.e. measured in a 125 ms window, while the latter is the level from the highest occurring squared sound pressure, regardless of duration, in the period being measured. Common limits have been $L_{pFA.max} = 110dB$ and $L_{pC,peak}$ = 130dB or 140dB.

In this paper, the notation L_A will be used for L_{pA}, and $L_{C,peak}$ or just Cpeak for $L_{pC,peak}$.

Based on data from the Queensland project (O’Brien), statistics are given in Table 2. In Timpani and Percussion, equivalent levels may be measurements of sound from others in their position at the back of the orchestra than sound from their own instruments. On the other hand, Cpeak-levels in Timpani and Percussion are most likely determined solely by sound from their own instruments, while Cpeak-levels in other instruments may well be influenced by sound from

Timpani and Percussion. The total number of measured sessions is *N=1619*, each session lasting on average two and a half hours, statistically 154 ± 41 minutes.

Table 2 Statistics from etry in the Queensland Orchestra 2004-2007 (data from O'Brien), averages *(m)* and standard deviations *(s)*, number of sessions measured *N*

	Bass	Bassoon	Cello	Clarinet	Flute	Horn	Oboe	Trombone	Trumpet	Tuba	Viola	Violin1	Violin2	Timpani	Percussion	avr(15)
N	79	135	46	155	151	273	90	157	212	24	79	26	62	63	67	108
LAeq (m)	84,5	87,6	84,6	87,9	87,9	89,2	87,0	88,6	89,4	87,1	85,3	84,2	84,7	87,7	88,8	87,0
LAeq (s)	3,0	2,7	2,7	2,7	2,7	2,7	2,4	3,0	3,1	4,2	3,6	3,0	2,7	3,9	3,1	3,0
LCpeak (m)	123	123	122	123	121	123	121	128	126	127	121	120	120	133	136	124
LCpeak (s)	5,4	5,2	5,1	4,4	4,1	3,7	3,8	4,8	4,6	5,0	5,4	5,7	4,6	4,2	6,2	4,8

In practice, each session would include various compositions with various instrumentation and sound power, and several would consist of a typical all-night concert program. The standard deviation of separate compositions would probably be bigger than the standard deviation of sessions like in this table.

For the average instrument, $L_{A.eq}$ would normally be within 84-90 dB, and on average 87 dB. The loudest positions are those of Horn and Trumpet, normally within 87-92 dB.

As to the loudest instant of a session, $L_{C.peak}$ for the average instrument would normally be within 119-129 dB, on average 124 dB. The positions with the loudest instants are those of Timpani and Percussion, with peaks normally within 129-137 dB and 120-142 dB, respectively. The loudest peaks recorded are a few occasions of 147dB in Percussion.

2.2.6 Dose

Further to section 2.2.5, the dose D [%] can be calculated from sound exposure of any level and duration, with D=100% equal to the common daily permissible dose (DPD), equivalent to 85dB over 8 hours, from the following formula:

$$D\ [\%] = t/480 \cdot 10^{x/10}\ , \qquad (1)$$

where $x=L_{A.eq}(t)-85dB$ measured over work hours a given day, and t is the total duration of the measurement(s), in minutes.

In the Queensland data, statistics of the 1619 measurements reveals that the normal dose is within 17%-137%, with the average dose *D= 77%.*

1 out of 4 sessions exceeded the daily dose.

DPD originates from industry noise regulations, suitable for work environment with little day to day variation in sound exposure, assuming 8 hours a day, 5 days a week. In contrast, occupational sound exposure in a symphony orchestra is very different from that of an industry or industry-like workplace. Sound levels vary from second to second, minute to minute, hour to hour, day to day, and even from one week to another. In order to arrive at empirically based knowledge about permissible doses for orchestra musicians, on would need to know the weekly and yearly doses too. Moreover, literature suggest that hearing loss (PTS, permanent threshold shift) from industry noise accumulates over years [5], implying that long-term exposure time matters, and that even a 10-year dose should be calculated when investigating any adverse effects from sound exposure in musicians.

The weekly dose can be calculated from:

$$D_{week}\ [\%] = t/3360 \cdot 10^{x/10}\ , \qquad (2)$$

where $x=L_{A.eq}(t)-85dB$ measured over work hours a given week, and t is total measurement duration, in minutes.

The principle from (1) and (2) can be extended to any long-term windows as fit. However, measurements and calculations of long-term doses can be impractical, and methods based on point samples and statistics should be consider in order to efficiently acquire a large amount of data with high statistical quality. Given the purpose of these measurement, and because time and resources are limited, the number of measurements would be prioritised over the accuracy of each single measurement.

2.2.7 Queensland statistics as reference for measurements in other orchestras

When seeking a statistical reference for "normal" sound levels in orchestras, it is important to include all relevant types of activity. Since the sound exposure inherent in the activities naturally varies from week to week, but less so from year to year, we would need data from at least one year.

The Queensland data (O'Brien) is acquired over 3 years and can be considered representative for the expected sound exposure in a symphony orchestra. This author has suggested the statistics from these data as an objective reference indication of whether measured levels are normal, or higher or lower than normal, i.e. inside the interval {m-s, m+s} for a given instrument or position. In a number of sound level mapping projects commissioned by symphony orchestras since

2013, this author has found it relevant to compare results with the statistics from the Queensland data, as presented in the next section.

3 etry and sound level mapping commissioned by symphony orchestras

3.1 Random or biased selection?

Some symphony orchestras want, for various reasons, to have knowledge about the sound exposure levels in their musicians, some because of issues or complaints from musicians, others just for the knowledge or because they want to be reassured that they are within normal.

It is important to keep in mind that the reasons for commissioning measurements will inevitably be a filter. If the orchestra for some reason prioritise to measure the presumably loud concert programs, their collected data would obviously not be a random selection, but a collection of sound levels above normal. If they instead seek to manage their seasonal or yearly program with the aim to have levels within normal, a more random selection approach should be chosen.

3.2 Project cases

As a consultant, this author has carried out etry and sound level measurements in several sound mapping projects since 2013, in two orchestras, commissioned by the orchestra administrations, in three concert halls.

- 2013: Orchestra A, Hall 1, Schönberg; Pelleas und Melisande; rehearsals and concert
- 2017: Orchestra A, Hall 1, Williams; Star Wars; rehearsals and 2 concerts
- 2019: Orchestra B, Hall 2 and Hall 3; Sibelius, Shostakovich, Brahms; rehearsals and 2 concerts
- 2020: Orchestra A, Hall 1, Debussy's Images, Ravel's Bolero; rehearsals and 1 concert
- 2022: Orchestra A, Hall 1, Mahler III; rehearsals and 2 concerts
- 2023: Orchestra A, Hall 1, Brahms, Stravinsky, Shostakovich; 2 days, trials with varying acoustics

3.3 Results

In Figure 2, resulting equivalent levels are presented for eters in 14 instruments/positions from the sound level mapping projects described above, together with the results by Schmidt and the 3-year average from Queensland (QSO). The vertical bars represent the "normal" over 3 years, based on the Queensland data. A similar presentation is given for Cpeak-data in Figure 3. Compared to the 4250 total hours of etry in Queensland, the results reported here add up to 554 hours of etry.

3.4 Comments

Equivalent levels in first violins are largely within the normal for their instrument and position in all the cases. The same goes for Timpani, but only two cases are included. All other instruments have levels significantly above normal in one or more cases. The music in Star Wars is very loud for obvious reasons (wars are inherently noisy) due to the extended use of brass in its orchestration. Mahler, too, with a big horn section and strong brass parts which tend to increase levels not only in the brass sections, but also in the woodwinds in front of them, not least because of directivity in trumpets and trombones. The latter is also the reason for the exceptionally high levels in the Double Bass during Star Wars.

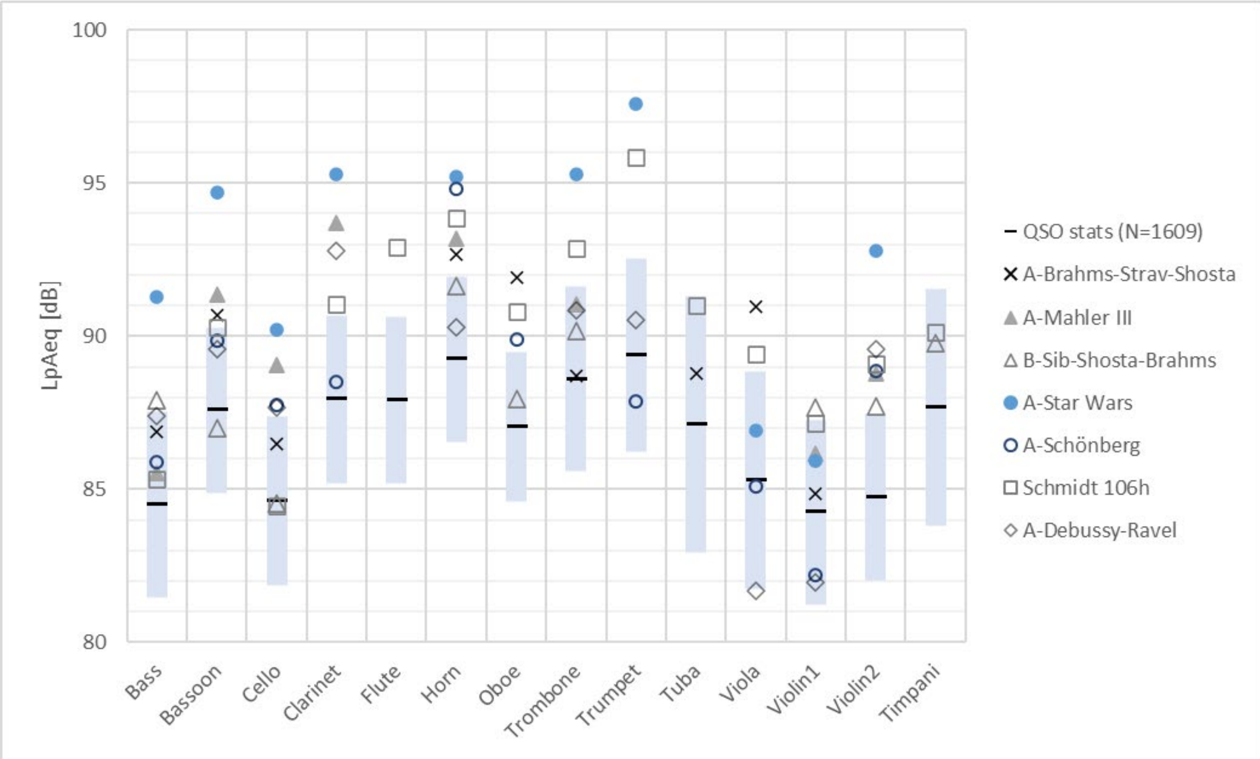

Figure 2 (above) L_{Aeq} from various etry measurements in 14 instrument groups in symphony orchestras; shaded bars indicate normal levels. See text for details and comments.

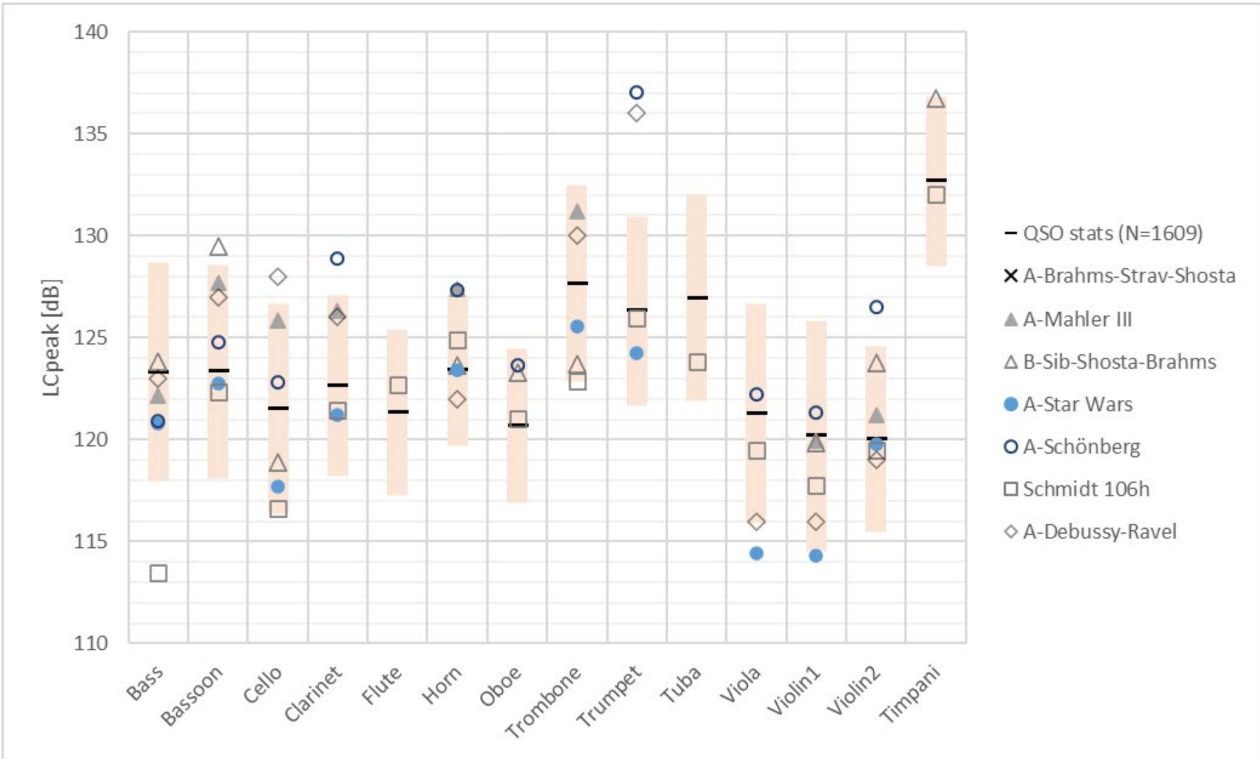

Figure 3 (above) $L_{C.peak}$ from various etry measurements in 14 instrument groups in symphony orchestras; shaded bars indicate normal levels. See text for details and comments.

In contrast to the equivalent levels in Figure 2, Figure 3 reveals that peak levels are largely within normal values, with the exception of Trumpet, which is approximately 1 sigma (1 standard deviation), 5-6 dB, above normal values during Schönberg and Ravel. Basson, Clarinet, Cello and 2nd violin exceed normal values by less than a third of sigma. Interestingly, while the Star Wars concerts exceed the normal range of equivalent levels by ca. 1 sigma in 8 out of 14 instruments, its peaks are within normal, and even slightly below normal in Viola.

One cannot conclude from the results in Figure 2 and Figure 3 whether or not exposure levels in Orchestra 1 and Orchestra 2 are normal in the long run. Since the measurements are commissioned, they are potentially biased. Actually, they are motivated by the orchestra management suspecting the measured projects to be rather loud.

Sound level and exposure management should be based on a long-term random point sample program for dosimetry over year cycles.

3.5 Other observations

3.5.1 Balance between reverberant sound and dry (non-reverberant) sound

With synchronized microphones in suitable positions in the room, it is possible to measure the reverberant sound in the concert hall, with the same resolution (1 second window) as that in the eters. With this information, the dry component of the orchestra, i.e. the sum of non-reverberant sound from own instrument and others', can be calculated at every eter position by subtracting the energy of the reverberant sound from the energy measured at the eter, for every 1 second window. In this manner a total of four time-varying levels with 1 second resolution can be acquired from each individual eter:

- L_{pA} *(t)* *[dB]* *sound pressure level at dosimeter*
- *R (t)* *[dB]* *reverberant energy level on stage*
- *D (t)* *[dB]* *dry, i.e. non-reverberant, energy level at dosimeter*
- *D-R (t)* *[dB]* *dry-reverberant balance*

Here, L_{pA} *(t)* is the time-varying eter signal, *R (t)* the time-varying reverberant sound level on stage, *D (t)* is the time-varying dry component, and *D-R (t)* the time-varying Dry-Reverb balance. The dry component can be calculated with the following formula:

$$D(t) = 10 \cdot log\left\{10^{\frac{L_{pA}(t)}{10}} - 10^{\frac{R(t)}{10}}\right\} \quad [dB] \qquad (3)$$

3.5.2 Possible adaptation to different acoustics the next day

In 2019, Orchestra B rehearsed Wednesday and Thursday, and performed the program in a concert Thursday evening in Hall 2. The next day, Friday, they moved to Hall 3 situated in another city where they had a brief rehearsal and performed the exact same program in the evening as the evening before. Interestingly, the sound levels in dosimeters were 1-2 dB softer, but in the stalls, levels were equal to those the evening before. The difference is not big, but given the repeatability of this orchestra, it is significant.

A possible explanation is that they know both halls very well because they play there frequently. They know that Hall 3 has a higher room gain (G) and know that they can play more relaxed and still achieve the proper levels in the audience.

4 Management of sound level, exposure, and hearing protection

"The more hearing protection they use, the louder they play." Managing director of Orchestra 1.

And - the louder they play, the uglier the sound, one might add.

Sound level management has two quite different advantages. One is obviously about the risk of hearing damage in the musicians, the other on is the artistic one: The sound quality.

4.1 Implications for sound quality

Whenever a musician plays louder or quieter, the sound level changes of course, but more importantly, the sound quality changes. A higher A-weighted sound power output from an acoustic instrument always comes with more brilliance because the power increase is inherently stronger in higher frequencies than in lower frequencies, and vice versa. Louder means more brilliant sound and quieter means more warm sound. Too loud means harsh and too quiet means dull. So – when does the individual musician play too loud or too quiet? The conductor has an opinion of course, an implication from the conductor's interpretation of the actual piece, including the composer's intention.

4.2 Composers' sonic palette

In this section, this author chooses to highlight the composer's intention and its imperative for sound level management.

A composer has an idea about how the piece should sound among the listeners in the audience. In making this idea become real, there are some basic tools, and they are inevitably linked to how the instruments are being played. Various instruments have different sound powers at one and the same dynamic notation, e.g. *mf* when playing mezzo forte, but they also have different frequency profile (spectrum) and therefore different timbre and sonic character. At forte and fortissimo the differences in character will be even bigger. The various power and sound character of various instruments is for a composer like a palette of colours for the painting artist. Instruments with less power, like violin and viola, can be grouped to become loud enough for their voice to be heard. If the composer wants a melody to be voiced with the character of an obo solo, an instrument among the less powerful, it can still be heard with the proper orchestration, e.g. with strings playing softly at piano with a *p* in their scores.

4.3 An orchestra's investment in sound quality

When managing balance between the various voices, it is crucial to be conscious about the intended sonic character. An instrument group can deliver the same loudness in principally two different ways – either a bigger group playing softer or a smaller group playing louder. The former would sound warmer, and the latter more brilliant. For this reason, an orchestra will employ 60 musicians for the full string group because the composition demands it, instead of having just 30 playing 3dB louder. When investing such a big amount of resources to make it sound right, i.e. according to the composer's intention and orchestration, they should naturally be careful not to waist it by having the orchestra playing loud and harsh. However, long-term increased levels because of forced playing easily happens without noticing.

4.4 Factors affecting level and quality

Quite a few factors can lead to excessive powerful playing style, or not so. Some of them are long-term effects, and some are self-reinforcing.

- Conductors' interpretation, personal taste and preference, and methods and tools for evaluating the orchestral sound
- Principle Conductors' philosophy, working methods, consciousness and emphasis on long-term development of the orchestral sound
- Any occurring misbalance is often easier to correct by demanding "more" from some, than "less" from all others
- Early decay time on stage, because proper feedback from the hall, effectively measured by *EDT*, would intuitively reassure the individual musician that music is being conveyed to the listener, and directly stimulate a more relaxed playing style
- Strength of reverberant sound G_r, including the spectral balance (dull-warm-neutral-brilliant-harsh)
    - Similar to *EDT* above, but effective even during sustained notes, chords and other stationary parts
    - Because it offers feedback to the individual musician in terms of "too strong, too soft"
    - Because too high brilliance comes with too dense high-frequency sound which makes hearing own and other's instruments more difficult
- Excessive use of hearing protection (ear plugs), because the intuitive sense of "too much" is suspended

4.5 Noise & Health

This author has repeatedly voiced that the sonic environment in which orchestra musicians work is not noise, and that sound exposure cannot be adequately managed similar to noise in e.g. industry. To the knowledge of this author, there is no scientific substantiation for the equality of sound exposure in the industry and in an orchestra, even if measured daily doses are equal. Even in one of the largest health studies ever performed, HUNT [6], including 250.000 persons since 1987Age, apart from ear infection is the dominant factor in hearing loss, and the only profession with raised risk is hunters because of the gunshots. Musicians, let alone orchestra musicians, are too few to form a statistically valid sub-population.

In a London court, an orchestra was convicted responsible for the hearing damage to one of their musicians, due to close-range exposure from brass instrument(s) in an orchestra pit. This should be a reminder to avoid obvious risk from powerful directive instruments at short distance from an instrument to a colleague's ears, like trumpet and trombone.

An active Noise & Health policy administrated by the orchestra organization is mandatory, but in the end, individually adapted hearing protection and personal, cautious practice is crucial.

- Never point a trumpet or trombone toward a colleague's head at short distance
- Soloist singers should face the auditorium when singing, even in rehearsals
- The use of in-ear hearing protection should only be used to reduce exposure to the recommended limit, 85 dB
    - For most musicians 1- 5 dB would be sufficient in most cases
    - Since the smallest available documented attenuation is 9dB, musicians should learn to roll their own cotton plugs and have the achieved threshold shift tested by audiographer
- A screen can protect against strong exposure from powerful instruments behind in loud parts and improve the overall listening balance in favour of instruments in front, typically the strings. However, increased sound levels from own instruments have been measured and should be considered in the sound exposure perspective
    - Screen type 1: Typically, a plexiglass screen, reflective on both sides, like Falko S300, Figure 4
    - Screen type 2: A U-shaped screen formed in a headrest-like position in which the musician can lean backwards into on demand, w/wo a soft lining, when sound exposure from behind is strong, Figure 4
- If needed, use hearing protection during solitary practice

Figure 4 Screens, Falko S300 (left), Hearwig and Goodear (right)

5 Further work

In future work, this author aims to acquire more data on variation from piece to piece and the population of concerts and projects through a year cycle. This could provide helpful information for the advice of orchestra managements in their pursuit to plan their program in order to achieve a healthy sound exposure load in their musicians.

Moreover, the time-varying Dry-Reverb balance will be investigated further, and so will the possible effect of EDT and G_r on self-reinforcing sound levels.

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Active noise-cancelling headphones: influence on performance, stress, and experience in work context

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Active noise-cancelling (ANC) headphones are increasingly used as an individual noise control device against disturbing speech, but do they really help? Our purpose was to determine the influence of different use settings of on-ear ANC headphones on performance, experience, and physiological stress in conditions resembling an office where task irrelevant speech is present. Laboratory experiment involved 54 participants, who were exposed to five conditions: 1. No headphones, 2. Headphones, 3. Headphones+ANC, 4. Headphones+masking, 5. Headphones+ANC+masking. In all conditions, speech was present in the room at 52 dB L_{Aeq}. Masking played via headphones was 51 dB L_{Aeq} wideband noise. Participants' performance was measured with serial recall and n-back tasks, and physiological stress with heart rate variability. The condition did not influence performance nor physiological stress, but it influenced experience. With masking (conditions 4 and 5), the speech was less annoying than without the headphones (condition 1). With ANC function (condition 3), the sound environment was more pleasant than without the headphones. However, only with both ANC and masking, the sound environment was estimated to impair performance and disturb concentration less than without the headphones. Therefore, using headphones with masking and ANC function can improve experience. Using headphones with ANC function does not properly solve the unnecessary speech problem in work environments as this use setting did not improve performance nor stress level.

1 Introduction

Colleagues' speech, which is irrelevant for the task at hand, is among the most disturbing environmental factors in open-plan offices [1]. Task-irrelevant speech can also influence work performance of cognitively demanding office tasks. The review of Haapakangas provided strong evidence based on psychological experiments, that the performance of short-term memory reduces with increasing Speech Transmission Index, STI, so that the reduction reaches 16% when STI reaches 0.60 [2]. Furthermore, working during speech has been found to increase stress hormone levels [3].

Active noise-cancelling (ANC) headphones are increasingly used in open-plan offices and learning environments as individually controllable noise control devices. However, the benefits of ANC headphones have not been systematically studied. One study examined the effects of ANC headphones on performance [4], but they did not study masking sound, which users often play via (ANC) headphones while trying to concentrate on work.

Commercial ANC headphones can have closed on-ear, open on-ear, and in-ear design. Closed on-ear headphones are often used in offices, since they are easy to put on and put off. They also provide an inherent attenuation of 0−15 dB depending on frequency. The effect of this attenuation on worker's perception has been very little studied.

Our purpose was to examine how closed on-ear ANC headphones influence a working person during task irrelevant speech in five typical use settings explained in Sec. 2.3.

2 MATERIALS AND METHODS

2.1 Study design

This was a laboratory experiment with a repeated measures design. The independent variable was *sound condition* (5 levels). Every participant experienced all five *sound conditions* one after the other. The irrelevant speech was continuously present in the room during the experiment while the *sound conditions* were created with ANC headphones and their different use settings. Dependent variables were perception, performance, and physiological stress.

2.2 Participants

Altogether 57 volunteers participated in the study. The participation criteria were normal hearing, speaking Finnish, age within 18–48 years, and normal health status. Three participants had to be excluded and the final number of participants was 54 (31 females mean age 24 years). The ethics committee of Turku University of Applied Sciences approved the study (statement 1/2020).

2.3 Independent variable

The independent variable was *sound condition.* The five *sound conditions* are defined in Table 1.

Table 1. Measured values of essential acoustic descriptors of *sound conditions* 1–5 and their settings.

Sound condition	STI	Masking in room L_{Aeq} [dB]	Masking via headphones L_{Aeq} [dB]	Masking in overall L_{Aeq} [dB]	Speech L_{Aeq} [dB]	Total L_{Aeq} [dB]
1	0.79	33	0	33	52	52
2	0.59	33	0	23	42	42
3	0.48	33	0	28	36	36
4	0.03	33	51	51	41	51
5	0.00	33	51	51	36	51

Sound condition	**Wearing headphones**	**ANC**	**Masking**
1	NO	Off	Off
2	YES	Off	Off
3	YES	ON	Off
4	YES	Off	ON
5	YES	ON	ON

We did not include silence in this experiment because the number of *sound conditions* was already large and increment of one more condition would drastically drop statistical power. In fact, the difference between silence and speech was studied in a separate experiment, which had basically equivalent experimental procedure and dependent variables [5].

The acoustic descriptors of the *sound conditions* are presented in Table 1. They were measured inside ear channels since *sound conditions* 2–5 involved headphones. A head-and-torso simulator was used at the participants' position. The values in Table 1 consider the diffuse field correction of the torso and, thus, represent the condition prevailing in a diffuse field without the torso.

Irrelevant speech was presented in the room via two loudspeakers during all five *sound conditions* at a constant level of 52 dB L_{Aeq} (A-weighted equivalent sound pressure level). The speech consisted of full sentences taken from an audiobook. However, the order of sentences was mixed and there was no plot to follow.

Weak masking noise was presented in the room via four loudspeakers at a level of 33 dB L_{Aeq} in all *sound conditions* since the room was unrealistically silent without it (17 dB L_{Aeq}). It did not mask the speech but resembled a typical ventilation noise prevailing in most offices.

Sound conditions 2–5 were created with headphones and their different settings. We chose closed on-ear ANC headphones available on the market in 2021 (JBL TUNE 750BTNC). Wearing them provided inherent attenuation by >10 dB above 315 Hz because of sound-insulating muffs. ANC operation reduced the level by 2–14 within 100–630 Hz.

Masking sound was wideband noise played from the headphones at a level of 51 dB L_{Aeq}. The spectrum of speech in the room conformed with standardized speech spectrum of ISO 3382-3. The spectrum shapes of masking in the room (33 dB) and masking in the headphones (51 dB) were equal and conformed with the spectrum used in many commercial sound masking systems.

In *sound conditions* 3 and 5, the ANC operation caused a weak electric noise, 28 dB L_{Aeq}, which can be seen in Table 1.

2.4 Dependent variables

Cognitive performance was assessed with two tests. Visual serial recall (**VSR**) test consisted of 9 digits presented in a random order and the task was to remember the order. The whole test contained 11 repetitions. **3-back** test shows letters on the display one after another with 2.5 second inter-stimulus interval. Task is to tell whether the letter was the same as three letters back or not. The test contained 33 letters. Both tasks were timed, therefore, it was also possible that an answer was not given in time. The variables are explained in Table 2.

Heart-rate variability (HRV) is increasingly used to measure the stress level of human body and it shows increased stress in irrelevant speech condition [5]. HRV was measured through the whole experiment with a chest belt. Stress level was assessed with one variable (HRV LF/HF, see Table 2), which is sensitive to stress caused by irrelevant speech [5].

After VSR and 3-back tasks, questionnaire Q2 inquired task-related subjective experiences about annoyances and workload (Table 2). After each test phase, questionnaire Q3 inquired *sound condition* related subjective perceptions (Table 2).

Table 2. Dependent variables.

Performance variable	**Definition**	
VSR accuracy	The proportion of correct responses	
VSR response time	The time needed to recall all 9 digits	
3-back accuracy	The proportion of correct answers	
3-back reaction time	The average reaction time of right answers	
Psychological variables of Q2	**Item**	**Response scale**
Speech annoyance	"How much speech bothered, irritated, or annoyed you?"	11-point numeric scale
Noise annoyance	"How much other noise than speech noise bothered, irritated, or annoyed you?	from 0 to 10 with labels
Workload	"How loading it was to perform the task?"	0 "Not at all" to 10 "Extremely
Psychological variables of Q3	**Item**	**Response scale**
SE was pleasant	"Sound environment was pleasant."	5-point scale with labeled
SE disturbed concentration	"Sound environment bothered my concentration."	extremes "-2 Completely disagree"
SE impaired performance	"Sound environment decreased my task performance."	to "+2 Completely agree"
Efficient working in SE	"If you should work daily with similar tasks in a similar sound environment, that you just experienced, I could work efficiently for long times."	
Physiological variable	**Definition**	
HRV LF/HF	The ratio of FFT spectrum powers of low (0.04–0.15 Hz) and high (0.15–0.4 Hz) frequency range	

2.5 Procedure

The procedure is described in Table 3. The experiment lasted on average for 2 hours and 18 minutes. The order of five *sound conditions* was randomly assigned for every participant to the five test phases A–E. Every participant performed the experiment alone in a soundproof experimental room.

2.6 Statistical analyses

The *sound conditions* were compared using a repeated measures ANOVA. If the main effect of the *sound condition* was statistically significant (p<0.05), we conducted four individual pairwise comparisons between *sound conditions* 2–5 and *sound condition* 1 was a reference.

Table 3. The procedure of the experiment. During the Test phases A-E, the irrelevant speech was played in the room. The sound conditions 1, 2, 3, 4, and 5 were presented in a random order in Test phases A-E. Orientation task was not analysed but it was used to give participants something to do while orienting to new sound condition. Q1-Q4 refer to different questionnaires. Q2 & Q3 were described above.

Phase, duration	Content
Preparation phase, 22 min	Consent form, heartrate belt wearing, hearing threshold test, Q1
Practice phase, 18 min	Rehearsal versions of orientation, serial recall , and N-back tasks
Test phase A, 20 min	Orientation task, serial recall task, Q2, N-back task, Q2, Q3
Test phase B, 20 min	Orientation task, serial recall task, Q2, N-back task, Q2, Q3
Test phase C, 20 min	Orientation task, serial recall task, Q2, N-back task, Q2, Q3
Test phase D, 20 min	Orientation task, serial recall task, Q2, N-back task, Q2, Q3
Test phase E, 20 min	Orientation task, serial recall task, Q2, N-back task, Q2, Q3
End phase, 6 min	Q4, undressing belt, payment

3 RESULTS AND DISCUSSION

Results are shown in Table 3. *Sound condition* did not affect performance nor physiological stress. Findings were limited to perceptions.

ANC headphones (*sound condition* 3) did not affect speech annoyance nor cognitive performances. The finding regarding performance agrees with the review of Haapakangas et al. [2] stating that reducing STI value from 0.79 to 0.48 should not yet drastically improve cognitive performance. The absence of the effect on annoyance was, however, unexpected and it contradicts the general belief that ANC headphones could protect people from annoying noise during office work. The probable reason for the absence of any effect is that ANC reduces only the sound pressure level of frequencies below 630 Hz while speech intelligibility mostly depends on frequencies above it.

Our findings related to ANC function alone agree with Ref. [4]. Our study provided additional novelty to Ref. [4] since we also inspected the effect of masking played via headphones. When masking was played via headphones without ANC operation (*sound condition* 4) or together with ANC operation (*condition* 5), subjective speech annoyance was significantly lower than in *sound condition* 1.

Headphones with ANC operation (*sound condition* 3) produced one positive effect. Sound environment was estimated to be more pleasant than without headphones (*sound condition* 1).

It was also interesting to see that ANC operation could provide extra benefit if masking was played via headphones (SE disturbed concentration, SE impaired performance).

4 CONCLUSIONS

Active noise-cancelling headphones do not alleviate the adverse performance effects nor reduce the stress levels when human is exposed to 52 dB L_{Aeq} task-irrelevant speech. The study suggests that sound masking should accompany ANC headphones to achieve most positive impacts with ANC headphones. ANC can possibly bring additional benefit, but mostly when masking sound is present.

Our study is limited to specific levels of speech and masking, see Table 1. People in the workplace are exposed to varying levels of surrounding speech and they can choose any masking sound level played via headphones. Thus, our conclusions cannot be generalized to all use settings that people use.

Table 3. The p-values of *sound condition*'s main effect and pairwise comparisons of sound conditions to the *sound condition* 1. The pairwise comparisons were performed only when the main effect was significant (p<0.05).

Variable	Main effect p-value	Pairwise comparisons' p-value 1 vs 2	1vs 3	1 vs 4	1 vs 5
Perception					
Task related					
Speech annoyance	***	ns.	ns.	***	***
Noise annoyance	***	ns.	ns.	***	***
Workload	ns.	–	–	–	–
Sound condition related					
SE was pleasant	*	ns.	**	ns.	ns.
SE disturbed concentration	**	ns.	ns.	ns.	*
SE impaired performance	*	ns.	ns.	ns.	*
Efficient working in SE	ns.	–	–	–	–
Performance					
VSR accuracy	ns.	–	–	–	–
VSR response time	ns.	–	–	–	–
3-back accuracy	ns.	–	–	–	–
3-back reaction time	ns.	–	–	–	–
Physiological stress					
HRV LF/HF	ns.	–	–	–	–

ns. p>0.05 / * p<0.05 / ** p<0.01 / *** p<0.001 / - not tested (no main effect)

5 ACKNOWLEDGEMENTS

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Predictors of noise annoyance and penalty of spectrally different wideband noises

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Numerous objective descriptors are found in literature for assessing noise annoyance of steady-state wideband sounds. However, there is no definitive understanding regarding which descriptors best correlate with subjective annoyance, when spectrum changes. Our aim is to evaluate how various descriptors correspond to differences in noise annoyance resulting from changes in A-weighted sound level and spectrum. This study builds on recent findings from a psychoacoustic experiment by Kuusinen et al. [1], who investigated 23 steady-state wideband noises with diverse spectral shapes within 32–48 dB L_{Aeq}, rated subjectively for annoyance. Their results showed that penalty could be as much as 10 dB due to spectrum shape. Here, we continue this work with a follow-up analysis concentrating on objective noise descriptors. Findings reveal that many descriptors strongly correlated with noise annoyance, but Spectral Centroid and Sharpness emerged as the best candidates to predict spectrum-based penalty. Due to the simplicity of SC calculation, it can be proposed as a practical tool to assess the spectrum-based penalty of environmental noise.

1 Introduction

Sound features like impulsivity and tonality are known to increase noise annoyance beyond what is indicated solely by the measured L_{Aeq} level [2]. This increase in subjective annoyance is considered in many countries by applying a penalty (also known as surplus, sanction, or adjustment), *k*, to the measured L_{Aeq} level, before comparing to the regulated threshold values.

Previous research has shown that also noise spectrum affects annoyance [3], but the application of spectrum-based penalty in noise assessment has received little attention. Kuusinen et al. [1] recently reported a study revealing that spectrum-based penalties for steady-state noise stimuli could be as much as 10 dB. They also observed that the spectrum-based penalties were not affected by the differences in L_{Aeq} level within 32–48 dB L_{Aeq}. They found that spectral centroid was highly correlated with the observed penalty values. However, they did not investigate any other objective noise descriptors.

Our purpose is to continue this work by trying to distinguish the best candidates for the development of the spectrum-based penalty procedure. We examine the correspondences between a variety of noise descriptors encountered in the history of noise assessment in buildings [4].

2 Methods and materials

2.1 Summary of the psychoacoustic experiment

The current work is based on the psychoacoustic experiment of Ref. [1]. Here, we provide an overview of the stimuli and the main results.

The experiment took place in a psychophysics laboratory. Sounds were played back from two loudspeakers. All sounds (experimental and reference) were band limited to 100 and 10000 Hz.

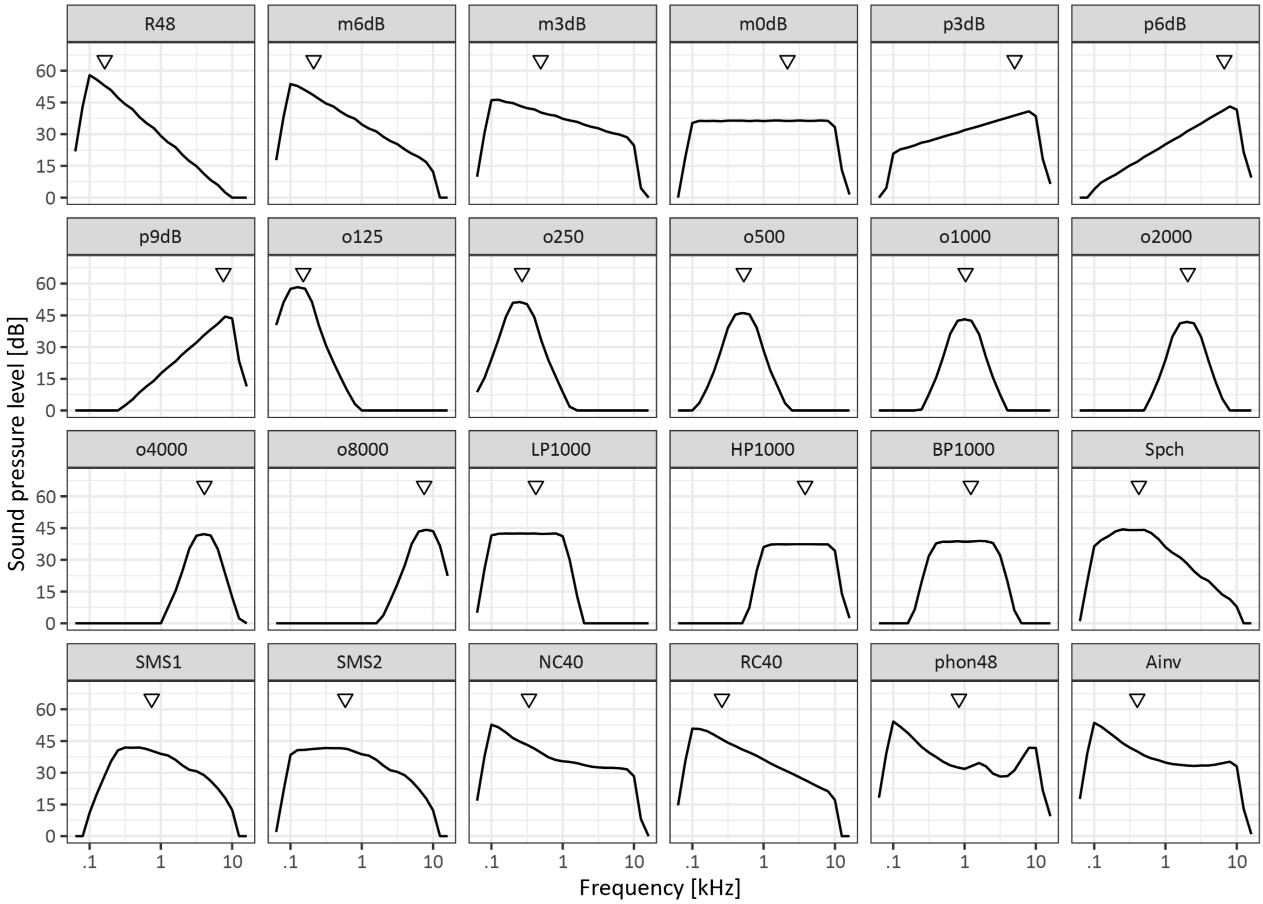

Figure 1. Sound pressure level, SPL, as a function of frequency, f, for the stimuli studied in Ref. [1]. Abbreviations: R48 = Reference sound with 48 dB L_{Aeq}; m6dB, m3dB,…, p9dB = sounds with different spectral slopes; o125,…, o8000 = octave band noises; LP1000, HP1000, BP1000 = wide band noises, RC40 = room criteria curve 40, NC40 = noise criteria curve 40; SIIsp, SMS1, SMS2 = Speech shaped noises, phon48 = equal loudness curve shaped noise, , and Ainv = inverse A-weighting curve shaped noise. All sounds were band limited to 100–10000 Hz. Here, all spectra are normalized to 48 dB L_{Aeq}. Triangles indicate the location of the spectral centroid for each sound.

Figure 1 illustrates the spectral shapes of the stimuli. The sounds included experimental sounds with 23 different spectral shapes, and reference sounds which had a spectrum with a slope of -9 dB per octave (shape of R48). The experimental sounds were presented at 3 sound levels: 32, 40, and 48 dB L_{Aeq}. The reference sounds were presented at 9 levels: 28, 32, 36, 40, 44, 48, 52, 56, and 60 dB L_{Aeq}, so that the lowest- and the highest-level reference sounds were perceived as the least and the most annoying among all the stimuli, respectively. The reference sound spectrum was also selected to be as little annoying as possible, based on the previous findings [3]. All stimuli were presented to listeners in randomized order.

The reference sounds enabled the derivation of the penalty values for the experimental sounds. The penalty is derived as follows: First, a reference line is obtained by fitting a linear function on the annoyance ratings of the reference sounds using L_{Aeq} level as the independent variable. The penalty for an experimental sound was then derived by projecting its annoyance rating onto the reference line and calculating the distance in dB L_{Aeq}. Thus, penalty is an estimate of the level increase needed for the reference sound to be perceived as annoying as the studied sound.

Forty normal hearing participants rated the annoyance of abovementioned 78 sounds using an 11-point scale from 0 (“Not at all”) to 10 (“Extremely”). Results of the experiment are summarized in Figure 2. It displays the correspondence between the mean annoyance ratings and the mean penalty values. It is important to highlight that while the effect of L_{Aeq} level was clear in the annoyance ratings, the penalty values were observed to be independent of the level effect. This can be seen in the marginal distributions in Figure 2.

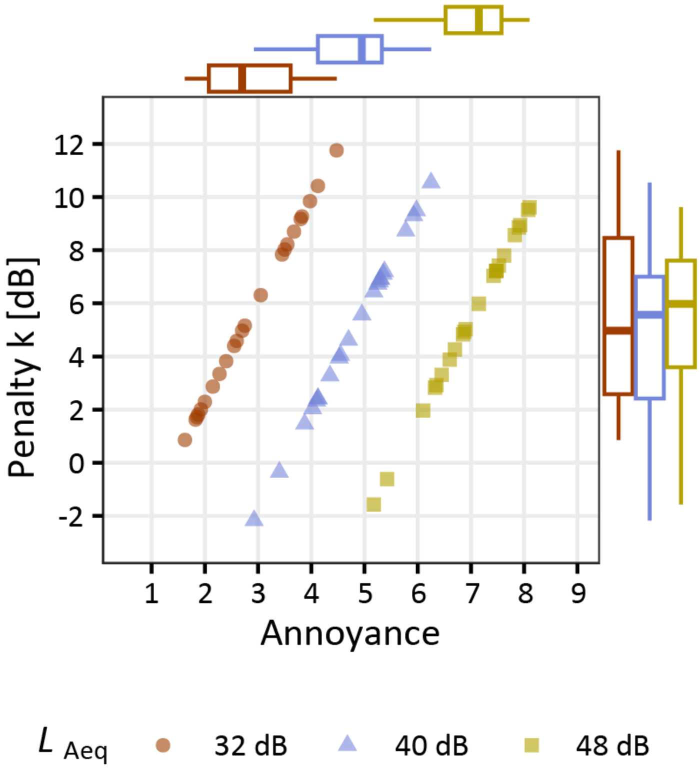

Figure 2. Scatterplot of the mean annoyance rating and the penalty k of the 23 spectrally modified sounds separately for each L_{Aeq} level in the experimental data of [1]. The boxplots on each margin illustrate the marginal distributions of the variables: the thick lines denote the median (i.e., 50th percentile), the boxes denote the 25th and 75th percentile range and the whiskers denote the 0th and 100th percentiles of the data excluding any outliers.

2.2. Noise descriptors

Tables 1 and 2 tabulate the noise descriptors included in the present work. Table 1 includes level dependent noise criteria which are based on SPLs and noise rating curves. Details on these criteria are given in Ref. [4]. In Table 1, the tangent method refers to the procedure where the octave band SPLs are superimposed over the rating curves and the first curve that is completely over the measured values determines the rating curve value. Table 2 includes other descriptors that quantify the shape of sound spectrum.

3 Results and discussion

3.1 Level dependent noise criteria and annoyance

The relationships between the level dependent descriptors and annoyance are illustrated in Figure 3. The marginal distributions (box plots on top) show the effect of sound level on the descriptor values. The level effect is clear for all these descriptors, but for some the distributions between levels are uneven, compared to the distributions of the annoyance ratings (on the right sides). The strongest linear associations were observed with *RNC, PNC NR*, and *NC,* which also exhibit similarly distributed values at each separate L_{Aeq} level. *SIL, NCB*, and *RC* show more scattered and uneven distributions.

3.2 Spectral shape descriptors and penalty

Figure 4 depicts the relationships between the spectral shape descriptors and penalty values. The marginal distributions (box plots on top) show that sound level had little effect on the values of *SC*, *SH*, and $L_{A,lo\text{-}hi}$, but for *QAI,* the level seems have a clear influence on the values. *QAI* also exhibits the largest scatter and weaker association with the penalty than the other three descriptors. *SC* and *SH* exhibit very strong positive association with penalty, while $L_{A,lo\text{-}hi}$ has almost equally strong, but negative association. The scatterplots for *SC* and *SH* are very similar to each other.

Table 1: List of the level dependent noise descriptors.

Abbr.	**Full name / Description.**
SIL	*Speech Interference Level.* Arithmetic average of 500, 1000, 2000, and 4000 Hz octave band SPLs.
NC	*Noise Criterion.* Calculate SIL and select NC-(SIL) curve. If a measured SPL value is over the curve, use the tangent method to determine NC-XX (Hz), with the frequency determining the curve in the parenthesis.
PNC	*Preferred Noise Criterion*. Procedure is the same as for NC, but instead of SIL, PNC rating is based on Preferred SIL, which is the average SPL of 500,1000, and 2000 Hz octave bands. Compared to the NC curves, the PNC curves are also more lenient on the lowest and the highest frequency bands.
NCB	*Balanced Noise Criterion.* Calculate SIL and determine the NCB curve that has the same SIL value. Then indicate the frequency band value that is the closest to the NCB curve.
RC	*Room Criterion*. RC value / curve is determined as the arithmetic average of SPLs over 500, 1000, and 2000 Hz octave bands. Also includes a separate assessment of spectral shape.
NR	*Noise Rating*. Curve value is determined with the tangent method.
RNC	*Room Noise Criterion*. Curve value is determined with the tangent method.

Table 2: List of the spectral shape noise descriptors.

Abbr.	**Full name / Description.**
SH	*Sharpness* [acum]. The center of loudness "mass" on Bark scale. [5]
SC	*Spectral Centroid* calculated by using the measured 1/3 octave band SPL values. SPL values below hearing threshold levels and below 15 dB were set to zero.
$L_{A,lo\text{-}hi}$	Difference between sound energies of low (63-500 Hz) and high frequency (1000–8000 Hz) octave bands [dB]. [6]
QAI	*Quality Assessment Index.* RC -rating is first calculated and spectral deviations from the selected RC -rating curve are calculated for the low (16, 31.5, 63 Hz), mid (125, 250, 500 Hz) and high (1000, 2000, 4000 Hz) frequency region. Spectral deviations are calculated by using averages in sound energy. QAI is defined as the maximum difference in spectral deviations between the low, mid, and high frequency values.

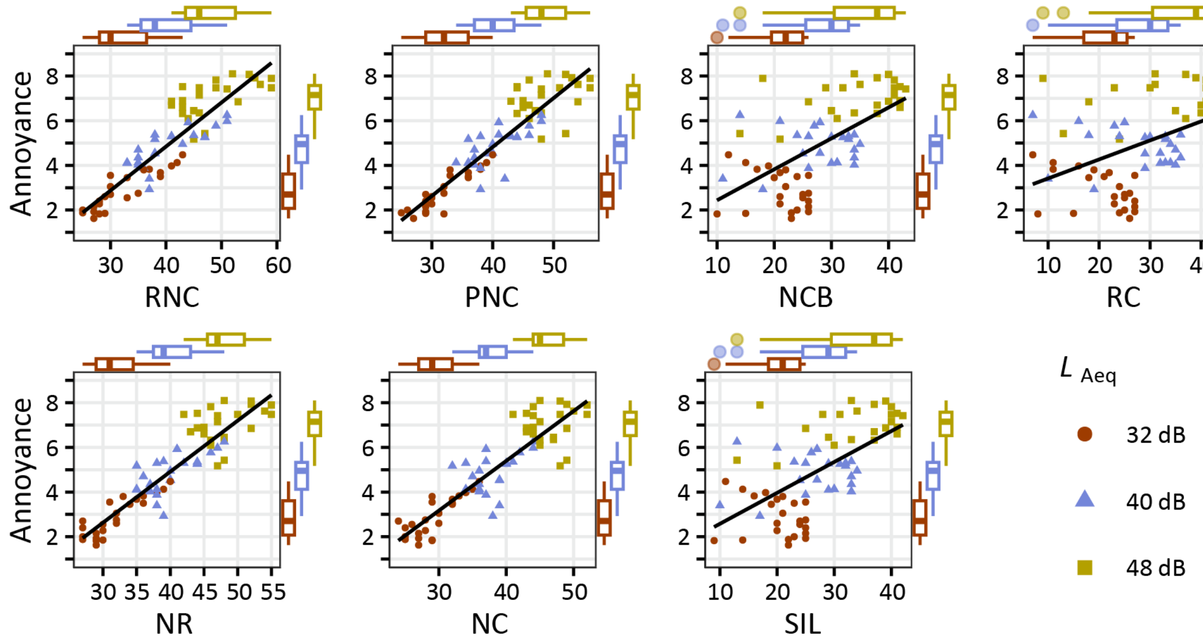

Figure 3: Scatterplots, marginal distributions (as in Fig 2.) and linear correspondences between objective level dependent descriptors and annoyance per each L_{Aeq} level.

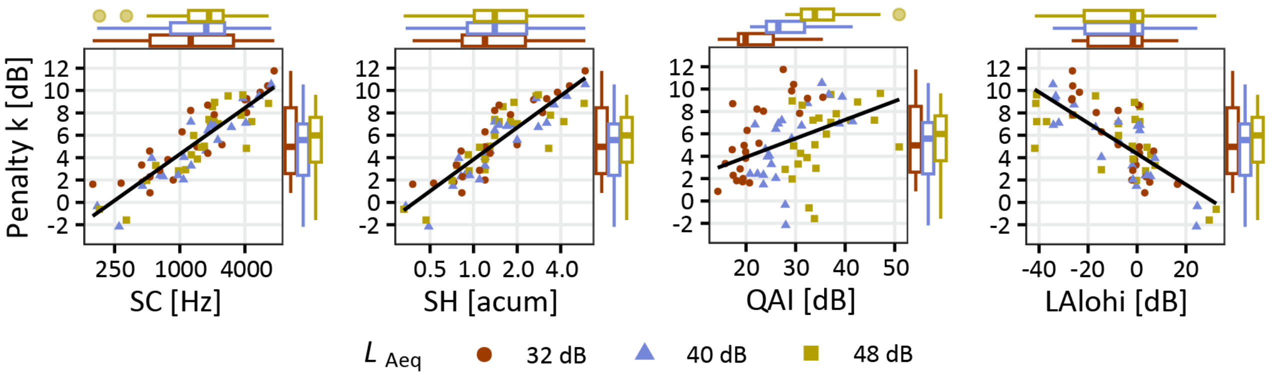

Figure 4: Scatterplots, marginal distributions (as in Fig 2.) and linear correspondences between objective spectral shape indices and the mean penalty k per each L_{Aeq} level.

4 Summary

This work extended the experimental results of Ref. [1] by assessing how different descriptors correspond with the differences in noise annoyance due to changes in A-weighted sound level and spectrum. The work is unique since the spread of studied noise spectra of Ref. [1] was exceptionally large. Their data enabled, for the first time, a critical assessment of different descriptors assessing the spectrum. In our analysis, *RNC*, *PNC, NR,* and *NC* were the most potential descriptors for predicting noise annoyance ratings, while *SC* and *SH* were the best candidates for predicting spectrum-based penalty values. Due to the simplicity of SC calculation, it can be proposed as a practical tool to assess the spectrum-based penalty of environmental noise. We also observed that *QAI* values were clearly level dependent, which was unexpected considering that *QAI* is aimed to quantify the shape of the noise spectrum in relation to the *RC* rating curves. A more detailed investigation with correlation analysis and regression models is on-going to verify these observations.

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An auditory loudness model with hearing loss

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The 'AUDMOD' model presented here models masking patterns, specific loudness, and total loudness for any stimulus and hearing loss, as specified by a standard audiogram. It is a strictly perceptual / behavioural model, based on the Moore & Glasberg roex-filter model, combined with the Zwicker & Fastl loudness model. The auditory-filter bandwidth depends on both signal level and hearing loss, both leading to increased upward spread of masking. The computational requirements are very low, making the model suitable for deep learning applications, including closed-loop hearing loss compensation. Furthermore, the model can be applied in a straightforward manner for analysis and interpretation of hearing aid functionality.

1 Introduction.

Auditory models have gained acceptance and wide use for modelling and understanding hearing and hearing loss. They can be used for quantitative analysis and simulation of sound perception.

Most published auditory models focus on the physiology of hearing [1] at selected output points in the auditory processing chain, e.g. auditory nerve. They can account for important perceptual effects, such as spectral masking, forward masking, hearing threshold etc., but not loudness.

In hearing aids, most fitting rationales apply multichannel compression to compensate for the abnormal loudness perception associated with hearing loss. These rationales are commonly based on simple loudness models [2]–[4]. Hence a signal processing model of loudness can be an important tool in understanding impaired loudness perception and creating appropriate compensation strategies.

An interesting new application of auditory models with hearing loss are 'closed-loop' approaches whereby deep learning is used to directly derive the appropriate hearing loss compensation by comparing a normal-hearing and a hearing-impaired branch of the auditory model. So far, the proposed solutions have not focused on loudness [5]. A simple model of loudness including hearing loss would allow for completely new hearing loss compensation strategies using the same deep learning techniques.

The proposed model was originally developed for a PhD project regarding sound quality metrics for hearing aids [6], [7], implemented in C and later ported to MATLAB.

2 Model design and function.

2.1 Model structure.

The 'AUDMOD' model presented here models masking patterns (excitation), specific loudness, and total loudness for any stimulus and hearing loss, as specified by a standard audiogram. It is a strictly perceptual / behavioural model, based on the roex-filter model [8] combined with a loudness model including recruitment [9]. The auditory filter bandwidth depends on both signal level and hearing loss, both leading to increased upward spread of masking. Due to the simple structure, it is easily differentiable for deep learning applications. An overview of AUDMOD is shown in Figure 1.

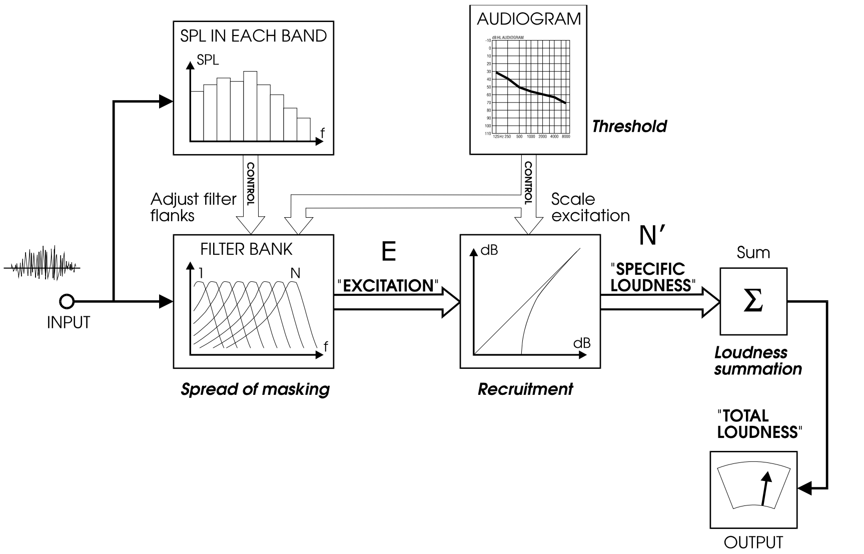

Figure 1: Block diagram showing the structure of AUDMOD.

The processing blocks in the model are as follows:

- FFT spectrum, e.g. 512 lines (not shown), to obtain a power spectrum.
- Corrections for sound-field type (not shown), coupler type and static transmission factors (e.g. middle ear) are applied in the frequency domain.
- The signal power is determined in 1 ERB wide bands, by summing the power spectrum (f) within the limits of each band. These power values are used to adjust the filterbank:
- The FFT power spectrum is then multiplied by a filterbank, consisting of 30 auditory filters whose shape depend on hearing loss and on the signal power. The filter bank concept is based on work from Moore, Glasberg, Patterson and others at the University of Cambridge [8], [10], [11], using 'rounded exponential' (*roex*) filters . The *roex* filterbank output is also called the Excitation pattern.
- The parameters for hearing loss (Threshold) are converted from dB HL to dB SPL and used to influence frequency selectivity in the filterbank and sensitivity in the loudness function.
- The *roex* filterbank output (E) is passed on to the specific loudness function that converts excitation in each channel to specific loudness, (N') according to [9]. The total loudness of an incoming signal can be calculated by summing the specific loudness across bands.

For further details, see [7].

3 Simulations.

The following section shows a few simulations of basic psychoacoustic properties. Comparisons are made to a similar model from Moore & Glasberg made available to us [3], named 'L2003'. AUDMOD used 32 ERB channels and L2003 used higher resolution of 272 channels. All stimuli were sampled at fs = 20000 Hz.

3.1 Frequency masking.

Excitation patterns were calculated for two simple stimuli and three levels: pure tones at 500 and 4000 Hz at 20, 60 and 100 dB SPL. The patterns are for binaural application of the models, with identical stimuli for the two ears (diotic) and a simple doubling of loudness. The results are shown in Figure 2.

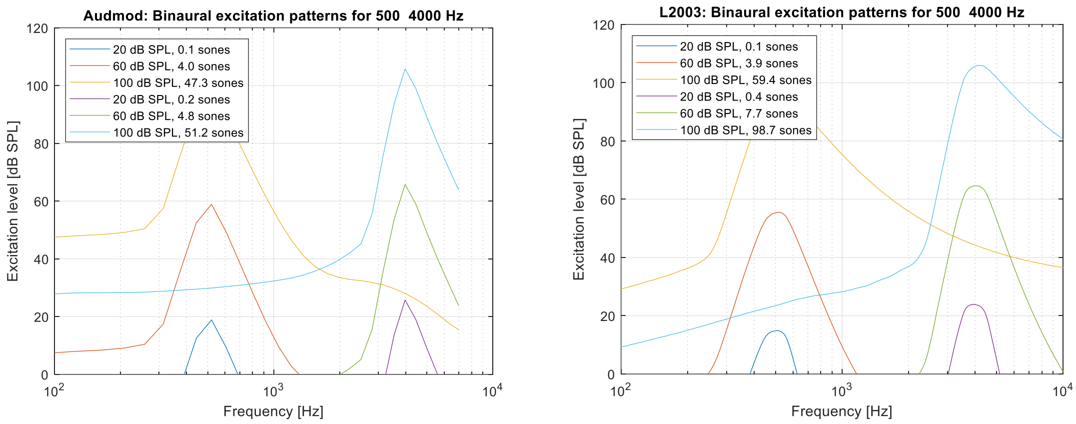

Figure 2: Excitation patterns (binaural mode) for pure tones at 500 and 4000 Hz, for levels 20, 60 and 100 dB SPL. left: AUDMOD, right: L2003.

The two models show similar patterns however L2003 shows more upward spread of masking for 100 dB SPL. Both models also calculate total loudness, which is shown in the figure legends. According to the definition of sones, a 60 dB SPL 1 kHz tone should yield 4.0 sones and 500 Hz slightly less according to the ISO226:2003 standard. Likewise, 4 kHz should yield slightly more than 4.0 sones. While both models agree on 500 Hz, 60 dB SPL, L2003 almost doubles the loudness for 4 kHz, 60 DB SPL (7.7 sones).

3.2 Frequency masking and hearing loss.

Both models can take a standard audiogram with hearing thresholds expressed as dB HL. For simple reference, we have used the standard simplified audiograms from [12], where N2 is a mild, sloping loss, N3 is a moderate, sloping loss and N4 is a severe, sloping loss. N0 was added as a 0 dB loss = normal hearing. The pure tone stimulus was at 1060 Hz, close to 1000 Hz, but matching an AUDMOD centre frequency.

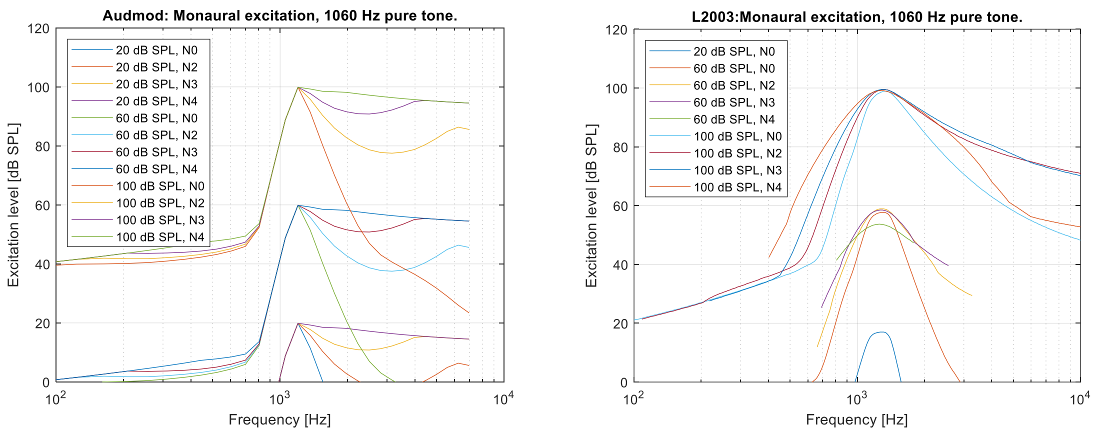

Figure 3. Excitation patterns (monaural mode) for a pure tone, 1060 Hz, at levels 20, 60 and 100 dB SPL. Hearing losses range from N0 (normal hearing) to N4 (severe hearing loss). left: AUDMOD, right: L2003.

The responses to hearing loss are quite different: L2003 has little effect of hearing loss, whereas AUDMOD shows dramatic effects, especially for the severe loss. There is very little frequency selectivity above the stimulus frequency.

3.3 Loudness growth and hearing loss.

An important part of a loudness model is to show plausible loudness growth across levels, according to the classical loudness models. We chose the 1060 Hz tone again and combined it with flat hearing losses (for simplicity) of 0 – 100 dB HL (Figure 4).

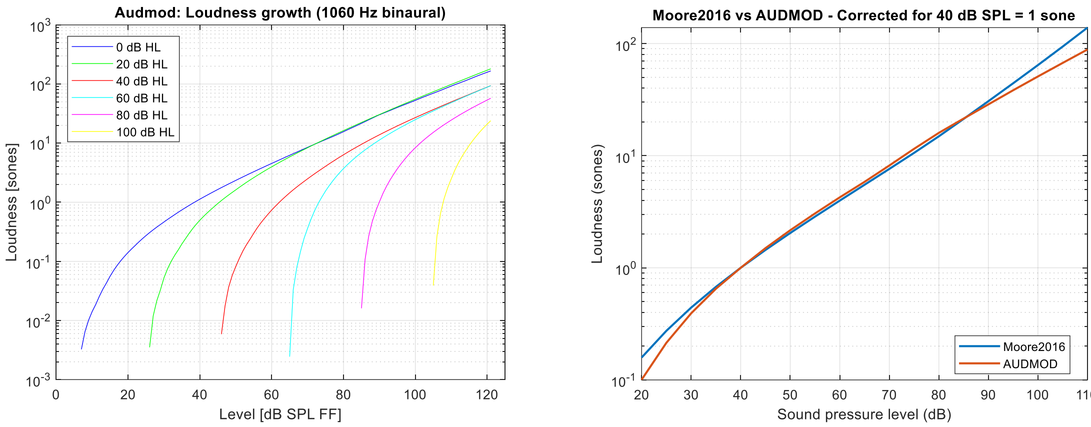

Figure 4. Left: Loudness growth functions for a 1060 Hz tone and different degrees of flat hearing losses. Notice the steep growth of loudness for levels just above threshold. Right: a comparison of AUDMOD to the Moore2016 model.

For normal hearing, the definition is closely matched: 40 dB SPL corresponds to 1 sone. For increasing degrees of flat hearing loss, the threshold is shifted accordingly while loudness for loud sounds remains closer to normal hearing.

For normal hearing, AUDMOD was also compared to the Moore2016 model in [13]. When correcting the Moore2016 model to get 1 sone @ 40 dB SPL, the two models are quite similar across a broad range of input levels. Credits to Pedro Llado, University of Aalto, for this comparison.

The model was validated on broad band stimuli as well, and furthermore against published listening test data on masking and loudness, see [7] for details.

4 Speech sample.

All the simple psychoacoustic simulations shown so far have used simple, stationary stimuli. But the intention of the model was to be used for any real-world signals to provide a time-varying specific loudness pattern for further analysis or as front-end for e.g. other metrics and deep-learning based hearing loss compensation. Hence a simple example is shown below, using a speech sample in quiet (no noise) at 70 dB SPL. A sentence from the new DAST Corpus (Danish Sentence Test, [14]) was chosen: Female speaker F1 speaking: 'Det var enestående at initiativet førte til handling'. The next steps were to include hearing loss and simple hearing loss compensation for a visual inspection of the model output (Figure 5).

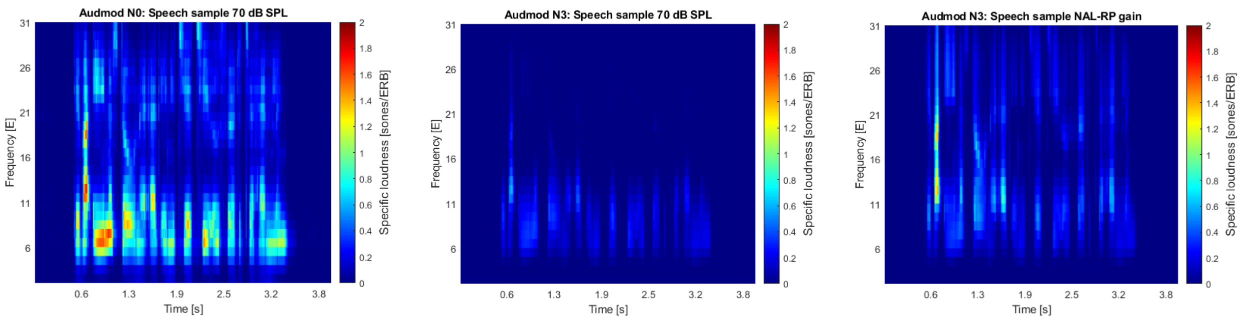

Figure 5: Speech sentence sample at 70 dB SPL. The frequency scale is the ERB-based E-scale according to model channels [10]. Left: unprocessed, centre: with N3 moderate hearing loss, right: N3 loss + NAL-RP linear amplification.

The unprocessed speech sample (left) shows a detailed picture with both intense low-frequency vowels and visible high-frequency consonants. The model then had an 'N3' hearing loss added, with the following audiogram thresholds: [35 35 35 35 40 45 50 55 60 65] dB HL at frequencies [250 375 500 750 1000 1500 2000 3000 4000 6000] Hz, see [12] for details. The resulting model output (center) shows limited audibility: very soft low-frequency sounds and no specific loudness for high frequencies. As a simple hearing loss compensation, linear frequency shaping, NAL-RP amplification for the N3 loss was added, shown in the right pane. Comparing this to the normal-hearing output (right) shows that loudness is partially restored, and some consonants are visible, but also that the reduced frequency selectivity leads to mostly vertical stripes in the pattern, corresponding to pronounced upward spread of masking.

It is likely that modern, multi-channel compression amplification could provide a more 'normal' output from the hearing-impaired model, although not a completely restored output.

5 New applications

AUDMOD is an example of a rather simple, phenomenological model that models the psychoacoustic perception of loudness without and with any hearing loss as defined by audiogram. It is, for example, attractive for two applications:

5.1 Objective metrics

The AUDMOD model can be used as front-end for objective metrics, as was originally done for a sound quality metric by Bramsløw [6]: AUDMOD was combined with a neural network and trained to model ratings of Sharpness, Clearness and Overall sound quality. Model results for both normal-hearing and hearing-impaired listeners were good. Other models use similar combinations of auditory modelling and neural networks, e.g. HASQI [15].

5.2 Hearing loss compensation

The modern deep learning models, e.g. deep neural networks, have opened up ideas of 'inverting' the hearing-impaired auditory model to obtain completely new types of hearing loss compensation. The basic principle is shown in Figure 6: two branches, for the normal-hearing and the hearing-impaired models are compared and a compensation network is added in front of the hearing-impaired model. The compensation network is trained to make the two model outputs as similar as possible, using s suitable cost function. The concept has been successfully implemented by Leer [16] using a more advanced physiological model of hearing.

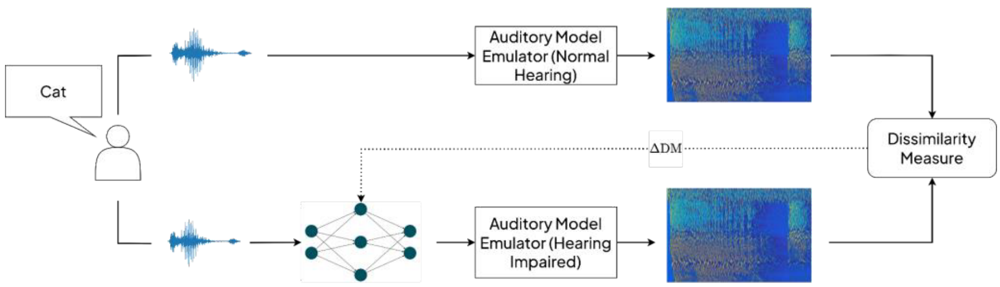

Figure 6. Example of closed loop hearing loss compensation using auditory model emulators and a deep neural network. Auditory model emulators are used for technical reasons but are practically identical. Figure from [16].

The same concept could be implemented using AUDMOD, with the benefit of being computationally more efficient and coding directly for loudness as do most hearing aid amplification rationales, e.g. [2].

6 Summary

AUDMOD is capable of modelling loudness growth for normal audiograms and can be applied to any acoustic stimulus. A simple example with a sentence in quiet, adding the hearing loss and subsequently adding amplification shows that there is still an impaired output from the model.

The computational requirements are very low, making the model suitable for deep learning applications, including closed-loop hearing loss compensation. Furthermore, the model can be applied in a straightforward manner for analysis and interpretation of hearing aid functionality.

It is the intention to publish AUDMOD in the Auditory Modeling Toolbox https://www.amtoolbox.org/, [13], first in MATLAB and later in Python. A corresponding journal publication is also planned.

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Stress effects of impulsive noise – A medical laboratory experiment

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Impulsive sound has been found to annoy people more than steady-state sounds presented at the same L_{Aeq} level. Our study examined the physiological, performance, and subjective effects of impulsive sound on working humans. The conditions were impulsive sound (65 dB L_{Aeq}), steady-state sound (65 dB), and quiet sound (35 dB). We applied between-groups design, where each participant was exposed to one sound condition. Altogether 59 participants were divided into these three groups. Physiological stress was measured with stress hormone concentrations in plasma, heart rate variability (HRV), and blood pressure. Psychological stress was measured with subjective noise annoyance, workload, and fatigue estimations. Performance was measured in tasks requiring constant concentration (visual serial recall, auditory serial recall, N-back). Compared to quiet, impulsive sound caused more annoyance, workload, and lack of energy, raised cortisol concentrations, reduced systolic blood pressure, and decreased accuracy in the 3-back task. Compared with steady-state sound, impulsive sound was estimated more annoying and causing a higher workload as well as lack of energy. Impulsive sound reduced 3-back accuracy while steady-state sound did not. Part of the effects of impulsive sound were due to the increased sound level since physiological stress reaction was also observed for steady-state sound. However, impulsive sound caused an extra effect over steady-state sound reflected in psychological experience and performance. Special care should be paid to the noise control of impulsive sounds in environments where people are performing mental work. Our findings may also have applications in residential environments.

1 Introduction

Impulsive sound has been found to annoy people more than steady-state sound or many other types of sound presented at the same L_{Aeq} level. Therefore, many countries apply a fixed penalty of, e.g., $k = 5$ or $k = 10$ dB (a.k.a., surplus, adjustment, sanction) that should be added to the measured A-weighted equivalent sound pressure level L_{Aeq}, before comparing to the target levels.

Because of such conventions, the presence of impulsivity in environmental noise shall be objectively provable. Therefore, Nordtest (2002) published a method to identify the presence of impulses in sounds. It proposed that the basic descriptors of an impulse are **onset rate R_{on}** (speed of impulse onset in dB/s) and **level difference D_L** (peak height of impulse in dB). The criterion for an impulse is that R_{on}>10 dB/s. However, the significance of an impulse is described by prominence P: P=3·log$_{10}$(R_{on})+2·log$_{10}$(D_L). Nordtest (2002) also contains a penalty model according to which k=1.8·(P-5) if P>5.

Since the penalty model of Nordtest was not based on scientific, peer-reviewed evidence, Rajala & Hongisto (2020) conducted a psychoacoustics experiment to validate the model. They found that penalty due to impulsivity strongly depended on onset rate and level difference. However, Nordtest model overestimated the penalty, and the penalty model should be revised.

Anyhow, impulsivity in sound affects annoyance. Annoyance is usually our first acknowledged reaction to noise. There is very limited research about the effects of impulsive noise on cognitive performance. Moreover, there are no prior studies about the acute physiological effects of impulsive noise. Such research requires a multidisciplinary research team involving experts from psychology, pharmacology, acoustics, and psychophysics.

Our purpose was to examine the physiological, performance, and subjective effects of impulsive sound on working humans. Since any noise can cause at least some of those effects, it is important to investigate non-impulsive noise with the same energy. Furthermore, since spectrum (Kuusinen & Hongisto, 2024), and tonality (Hongisto et al., 2018) also affect annoyance, it is important to keep the spectrum constant between these two sounds. Therefore, our experiment involved three conditions: impulsive sound (65 dB L_{Aeq}), steady-state sound (65 dB L_{Aeq}), and quiet (35 dB L_{Aeq}). All of them had similar spectrum shapes.

This study has been published in detail by Radun et al. (2022).

2 Materials and methods

2.1 Overall design

A medical laboratory experiment was conducted where the independent variable was *sound condition*, and dependent variables were several subjective, psychological, and physiological variables. The experiment had a parallel group design, where each group was exposed to one of the *sound conditions*. Due to the blood tests, approval from the ethics committee was mandatory. The Ethical Committee of the Hospital District of Southwest Finland approved the study (ETMK 20/1801/2018).

2.2 Sound conditions

The experiment involved three *sound conditions*: quiet sound, steady-state sound, and impulsive sound. The exposure time for each *sound condition* was the same. The *sound conditions* are described in Figure 1. The **Impulsive sound** (65 dB L_{Aeq}) was a real pile driving sound recorded at a construction site. The **Steady-state sound** (65 dB L_{Aeq}) was created from pseudorandom noise and the spectrum was aligned with Impulsive sound. **Quiet sound** (35 dB L_{Aeq}) was similar to Steady-state sound but attenuated by 30 dB. It resembled the background noise from typical ventilation noise.

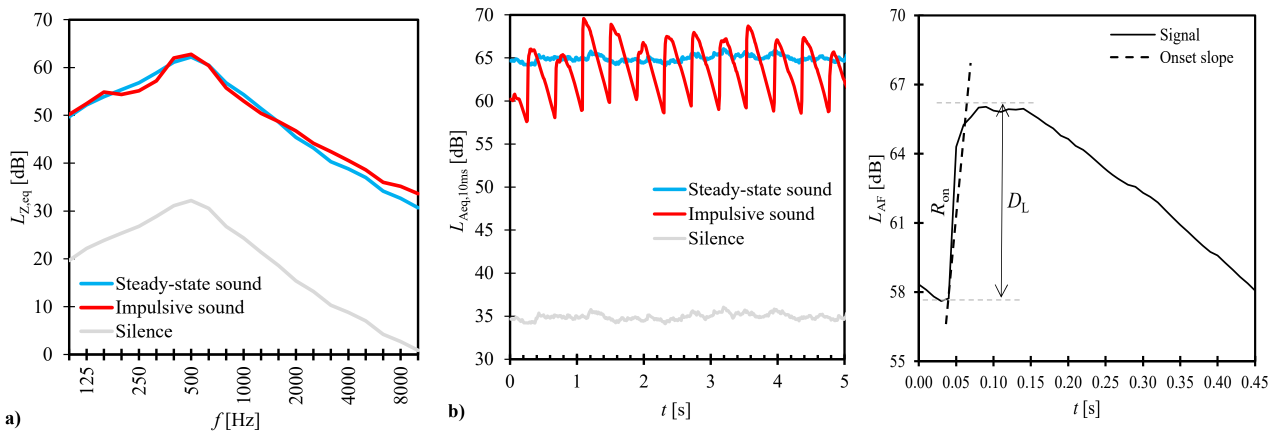

Figure 1. a) Spectra of *sound conditions* showing the unweighted SPL, $L_{p,Z,eq}$, as a function of frequency, *f*. b) Time profiles of *sound conditions* showing the A-weighted SPL with 10 ms time window, $L_{Aeq10ms}$, as a function of time, *t*. c) Microstructure of a single impulse showing the fast-time weighted SPL, L_{AF}, as a function of time. The sampling period was 10 ms. The level difference was D_L = 8.2 dB and the onset rate was R_{on} = 236 dB/s. The impulses were periodic (appr. 2.4 Hz).

2.3 Participants

Altogether 59 participants were involved to this study: 19 in Quiet sound, 19 in Steady-state sound, and 21 in Impulsive sound. The inclusion criteria were: Finnish speaking, not taking regular medications, not smoking, regularly, not needing caffeine before 4 pm., not pregnant, and not breast-feeding. Gender and noise sensitivity were measured before the participants arrived the experiment since they were used to stratify the participants into three balanced groups. Participants were instructed to sleep a normal night before the experiment and wake up at the latest at 8 am. They were instructed not to consume nicotine, alcohol, caffeine, or bananas nor to be exposed to loud noise on the day of the experiment and to consume a light lunch before the experiment at 11 am.

2.4 Laboratory

The experiment was conducted in the psychophysics laboratory of Turku University of Applied Sciences. Two participants were tested at a time. The sounds were produced by two loudspeakers so that the exposure corresponded to the target values of Fig. 1.

2.5 Psychological (subjective) measures

After each task, the participants rated how much background sound irritated, bothered, or annoyed them (*annoyance*) and how demanding or loading performing the tasks was (*workload*). The scale for both questions was from 0 “Not at all” to 10 “Extremely”. The perceived fatigue was measured using Swedish Occupational Fatigue Inventory (SOFI), which gave three scales: *tiredness*, *lack of energy*, and *lack of motivation*.

2.6 Performance measures

N-back is a working memory task, where the participant responses whether the current stimulus is the same as n stimuli back. Three difficulty levels were used n = 0, 1, 2, and 3. Each time, 30+n repetitions of each difficulty level were performed.

Serial recall tasks are also working memory tasks examining how well the participants can keep a list of numbers in their mind. Digits 1–9 were presented in a random order and participants were asked to write the correct order 10 seconds after the last digit was presented. 11 series were used. Two variations of the task were applied: visual serial recall (VSR), where the numbers were presented visually on the display and auditory serial recall (ASR), where the participants heard the numbers from headphones. The level of target speech from on-ear headphones (freely breathing model) was 75 dB L_{Aeq}.

2.7 Physiological (stress) measures

The physiological measures used were stress hormone concentration (cortisol and noradrenaline) determined from plasma, heart rate variability (HRV) measured with a heart rate monitor around participants’ chest, and blood pressure (systolic blood pressure SBP and diastolic blood pressures DBP). Plasma was taken from the peripheral venous access catheter that was placed in participants’ arm in the beginning of the experiment. From HRV, the LF/HF ratio was determined. It describes the activity of parasympathetic and sympathetic nervous system. The larger values mean greater sympathetic nervous system activity, which means more stress. This relation is here called HRV LF/HF, which was calculated for periods of each cognitive task separately (VSR, ASR, and N-back).

Preparation phase →	Practice phase →	Baseline phase →	Break →	Test phase →	Recovery phase →	Ending phase
30 min	25 min	50 min		50 min	20 min	10 min
Informed consent	Q1	VSR, IQ1, N-back, IQ2	10 min	VSR, IQ1, N-back, IQ2	Q2	Sample 6
Catheter on	Practicing tasks	Sample 2		Sample 4		Catheter off.
Heart rate monitor on	Sample 1	ASR, IQ1, N-back, IQ2		ASR, IQ1, N-back, IQ2		Heart rate monitor off
Hearing test		Sample 3		Sample 5		Gift card 70 eur

Figure 2. Experimental procedure. The average duration was 3 h 20 min. Experimental sound was present in Test phase. Sample means that blood sample was taken, and blood pressure was measured.

2.8 Procedure

Procedure is described in Figure 2. Quiet sound (35 dB L_{Aeq}) was present in the room in every phase except in the test phase where the actual *sound condition* (Quiet sound, Impulsive sound, or Steady-state sound) was presented.

The experiment started at 11.45 each day and lasted on average for 3 h 19 min. Afternoon was chosen because diurnal variation in cortisol concentration is the largest in the morning. Cortisol variation also explains the strict requirements and instructions for the participants.

In the preparation phase, the heart monitor and the catheter were put on and hearing was tested. This was followed by the practice phase, where all tasks were explained and rehearsed. Both baseline and test phase involved the same cognitive tasks and subjective estimations, but the experimental sound was presented only in the test phase. The blood samples were taken 6 times during the experiment. Blood pressure was measured each time after taking the blood sample. In the questionnaire Q1, participants reported their current state and other background information. Psychological estimations related to sound were estimated several times during the experiment. Annoyance and workload were estimated after each task (8 times) (IQ1 and IQ2) and SOFI was filled each time after N-back task (4 times) (IQ2). In the recovery phase, participants filled questionnaire Q2 with Quiet sound in the background.

2.9 Statistical analyses

To reduce the influence of individual differences in outcome variables, we used the difference between test phase and baseline phase as the main outcome variable for the psychological and most physiological measures. However, cortisol concentration showed the expected diurnal changes. In addition, there seemed to be large individual differences possibly due to excitement in the baseline phase. Therefore, with cortisol we used the recovery phase as the reference instead of baseline phase. The performance measures showed more variation in performance in the baseline phase than in the test phase possibly due to excitement of the experiment as well as learning the tasks. Therefore, we examined the performance measures using the direct means of the test phase.

The groups were compared with each other using repeated measures analysis of variance, if the test phase had more than one observation from each participant on that variable. In those cases, time was the within-subject variable, *sound condition* was the between-subject variable and noise sensitivity was the covariate. If there was just one observation on that certain variable from the test phase, then univariate analysis of variance was used with *sound condition* as the between-subject variable and noise sensitivity as the covariate. From the performance measures of *N-back* task, only *3-back* is reported here, since it was the only that filled the requirements of repeated measures analysis of variance. Greenhouse-Geisser correction was used if the sphericity assumptions were not filled (ASR and VSR interaction).

Variable	**Pair of *sound conditions***		
	Quiet vs. Steady-state	Quiet vs. Impulsive	Steady-state vs. Impulsive
Psychological measures			
Annoyance	-	-	-
Workload		-	-
Tiredness	n.s.	n.s.	n.s.
Lack of energy		-	-
Lack of motivation	n.s.	n.s.	n.s.
Physiological measures			
Cortisol [nmol/l]	-	-	
Noradrenaline [nmol/l]	n.s.	n.s.	n.s.
SBP [mmHg]		+	
DBP [mmHg]	n.s.	n.s.	n.s.
HRV $_{LF/HF}$	n.s.	n.s.	n.s.
Performance measures			
Auditory serial recall accuracy (ASR	n.s.	n.s.	n.s.
Visual serial recall accuracy (VSR)	n.s.	n.s.	n.s.
N-back RT	n.s.	n.s.	n.s.
3-back Accuracy		-	

No main effect, comparison not performed	n.s.
Indicates lower stress	+
Indicates higher stress	-
Indicates no significant difference, despite of main effect	

Figure 3. The main results of the experiment.

3 Results

The results are summarized in Fig. 3. Some of the main results are shown in Fig. 4.

3.1 Psychological measures

Annoyance, *workload*, and *lack of energy* were affected by *sound condition*. Pairwise comparisons showed that *annoyance* was larger in Steady-state sound than in Quiet sound. However, all three variables were larger in Impulsive sound than in Quiet sound. Moreover, all three variables were higher in Impulsive sound than in Steady-state sound.

3.2 Performance measures

Sound condition only affected the 3-back accuracy. Paired comparison revealed that performance in Impulsive sound was better than in Quiet sound. *Sound condition* did not affect the performance of the other cognitive tasks.

3.3 Physiological measures (stress)

Sound condition affected cortisol concentration in blood plasma. Paired comparison showed that performance was worsened both in Impulsive sound and Steady-state sound compared to Quiet sound.

Sound condition affected Systolic blood pressure. Paired comparison revealed an unexpected result, that the pressure was lower in Impulsive sound than in Quiet sound. The finding could not be explained.

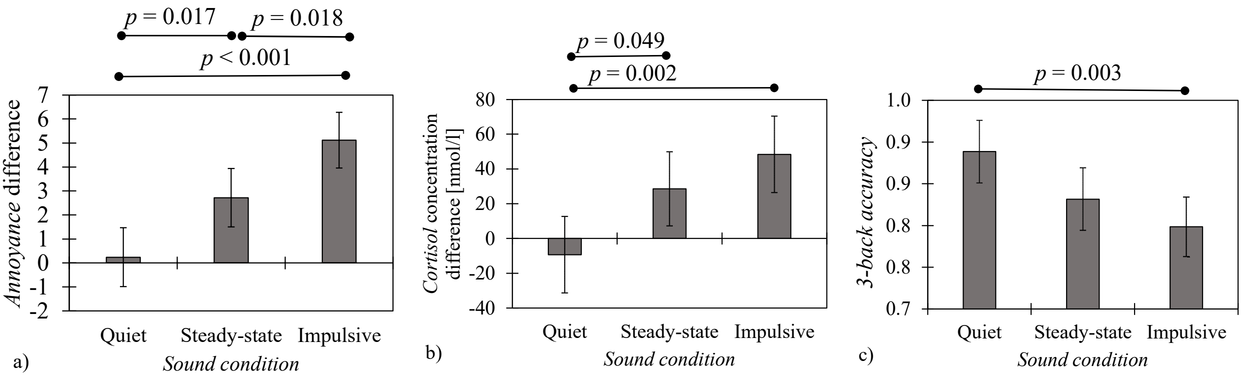

Figure 4. Figures depicting some main results. a) *Annoyance* difference (test phase minus baseline phase) was significantly higher in Impulsive sound than in two other *sound conditions*. b) Cortisol concentration difference (test phase - recovery phase) was significantly higher in Impulsive sound than in Quiet sound. However, difference between Impulsive sound and Steady-state sound did not reach significant level. c) *3-back accuracy* was significantly different between Impulsive sound and Quiet sound.

4 Discussion

Impulsive sound had more adverse effects than Steady-state sound. Four extra effects were found:

1. *Annoyance* was higher in Impulsive sound than in Steady-state sound.
2. *Workload* was higher in Impulsive sound than in Steady-state sound while Steady-state sound did not differ from Quiet sound at all.
3. *Lack of energy* was higher in Impulsive sound than in Steady-state sound while Steady-state sound did not differ from Quiet sound at all.
4. *3-back accuracy* was worse in Impulsive sound than in Quiet sound. Similar effect was not found for Steady-state sound.

The finding 1 agrees with Rajala & Hongisto (2020), according to whom the annoyance penalty of impulses like the pile driving of our experiment, should be more than 5 dB. The fact that significant effect can be seen also in a long-term exposure confirms that penalties should be applied for impulsive sounds.

The finding 2 agrees with Radun et al. (2021), who found that *workload* was not affected by Steady-state sound at 65 dB L_{Aeq} but it was affected by Speech sound at 65 dB L_{Aeq}. Speech is highly impulsive, which may explain the agreement partially. However, speech carries also special information content, which may add workload.

The finding 3 shows Impulsive sound causes different experience than speech. Radun et al. (2021) found that working during speech was more tiring than working during a quiet condition or steady-state sound, but differences in lack of energy were not found, which was the effect caused by impulsive sound.

The finding 4 also agrees with Radun et al. (2020), who found that 3-back accuracy was not affected by Steady-State sound at 65 dB L_{Aeq} but it was affected by Speech sound at 65 dB L_{Aeq}.

It seems that impulsive sounds have more adverse effects on humans than steady-state sounds having the same sound level expressed in L_{Aeq}. It would be interesting to conduct similar research using tonal sounds since similar annoyance penalty procedures are applied for tonal sounds than for impulsive sounds.

5 Conclusions

Previous knowledge has supported that annoyance penalty should be applied for impulsive sounds. Our study supports this way of thinking. However, we found that impulsive sounds have also other extra effects compared to steady-state sounds (workload, lack of energy, reduction of cognitive performance). Therefore, special care should be paid to the noise control of impulsive sounds in environments where people are performing mental work. Our work may also have practical applications in residential environments (shooting ranges, logistic centers, construction sites, ball fields).

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Acoustic comfort assessment in hospital wards: measuring procedures and parameters.

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Noise pollution is one of the most concerning environmental factors, as it can affect people's health. Especially the health of vulnerable people, such as children, the elderly, the sick, or hospitalized. For this reason, special attention must be paid to the design of hospital environments, within which noise from different activities and equipment can affect both the performance of healthcare staff and the quality of sleep of patients. In this context, patients are the most vulnerable users, as their extra-ordinary condition leads to less ability to cope with stress and greater sensitivity to noise. Despite numerous scientific studies showing that the noise levels detected within hospital environments are highly above those indicated by the World Health Organisation, there are few proposals for effective noise reduction in wards. Our research starts from the need to ensure the acoustic comfort of patients during their hospitalization. For this reason, the first purpose was to define a replicable survey method for assessing noise and acoustic quality in hospital wards, using acoustic characterization measurements, sound pressure levels long-time monitoring, and field observations. Specifically, the paper proposes the criteria applied in the selection of the case studies, the acoustic parameters analyzed, and the measurement techniques used. The survey, which can be easily applied to different contexts, has currently been experimented within some Italian University Hospitals.

1 Introduction

Noise pollution is one of the most worrying environmental factors, as it is increasing over time and can affect people's health [1]. Especially the health of vulnerable people, such as children, the elderly, the sick or hospitalized [2]. For this reason, special attention must be paid to the design of hospital environments, which is a complex organism that houses multiple spaces, functions, activities and equipment, but also different user groups - healthcare staff, patients, visiting relatives - with different needs and varying degrees of sensitivity to sound. For patients, excessive noise levels can affect stress levels, quality of sleep, and the recovery process [2, 3]. In addition, excessive noise can adversely affect healthcare personnel's quality of life and performance, causing stress, aggression and distraction, which can lead to medical errors [2, 4, 5]. Most of the noises complained about by patients come from the corridor. Specifically, most of the noises reported by patients during the night are of anthropogenic origin, such as the conversations of patients and healthcare personnel, and the movement of carts and stretchers [6].

The World Health Organization has expressed the values of desirable sound pressure levels for hospital environments, in order to protect people's well-being and health. Specifically, the A-weighted equivalent sound pressure level (L_{Aeq}) should not exceed 35 dBA during the day and 30 dBA during the night, while the A-weighted maximum sound pressure level (L_{AFmax}), for all anomalous events, should never exceed 40 dBA. Furthermore, in the case of areas frequented by patients, the 30 dBA L_{Aeq} should be maintained both at night and during the day [2]. Additionally, studies on the impact of noise on sleep patterns, considering variables such as the variation of the duration and depth of the sleep phases by electroencephalography (EEG), has revealed that an equivalent sound level L_{Aeq} from 45 to 50 dBA can change EEG patterns in approximately 50% of exposed subjects [7].

In spite of this, scientific studies have revealed that sound levels inside most hospital spaces are very high [8]. Moreover, most studies focus only on highlighting the problem, giving little thought to possible solutions. A hospital environment design that takes into account both the acoustic performance, the ward layout and the user behavior could limit patient noise exposure [9].

The normative references on the acoustics of hospital environments in different European countries are various. The most commonly used parameters in national and international standards are airborne sound insulation, impact sound insulation, façade sound insulation, service equipment noise, and reverberation time [10, 11]. Despite this, noise disturbance within a hospital ward is mainly caused by the high sound pressure levels, produced by people's activities and equipment, propagating into the ward and consisting in short term events. In research, the most frequently used parameters to describe the indoor acoustic comfort of hospital environments are L_{Aeq}, L_{Amax}, L_{Amin}, or statistical indices as L_5, L_{10}, L_{50}, L_{90}, and L_{95} [2, 8, 12]. In Italy, the assessment of the acoustic performance of hospital building elements follows the standard UNI 11367 [13], which is mandatory under Ministerial Decree 23/06/2022 [14]. To date, the Italian reference for the interior acoustic quality of public buildings is the UNI EN 11532 series [15], part 4 of which, specific to hospitals, hasn’t yet been published.

In the following we illustrate the method of investigation that has been carried out during a doctoral program, in order to assess the acoustic quality of in-patient wards and propose intervention strategies aimed at the acoustic comfort of all users. In particular, our research focuses on the comfort of the most vulnerable user, the patient, whose extra-ordinary condition leads to less ability to cope with stress and greater sensitivity to noise. The first part of the research focused on defining a replicable survey method for the assessment of the indoor acoustic quality, using acoustic characterization measurements, long-term and short-term monitoring of sound pressure levels and field observations. In addition, interviews were conducted with health personnel.

The paper proposes the criteria applied in the selection of the case studies, the acoustic parameters analyzed, and the measurement techniques used. This method was validated through measurement campaigns carried out in four Italian hospitals.

2 Case studies selection criteria

The first part of the research dealt with defining a process for selecting a typical hospital ward to carry out the survey. The selection can be based on the following two main aspects: first, type of care provided to patients; second, hospital ward configuration; third, patient bedroom layout.

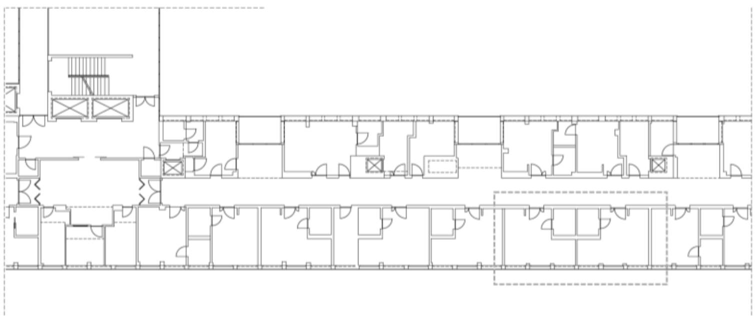

Figure 1: Example of one of the four case studies analysed: Ordinary inpatient care in the Surgical Department. Single corridor ward organization, with two-bed mirrored bedrooms.

Concerning the first topic, for the purpose of this study we propose to select only general low-intensity care wards, since high intensity care wards are characterized by very peculiar noise conditions, and within these specific spaces the acoustic comfort of patients usually has little importance compared with patients’ general health conditions. Then, we focused on the most recurring ward configurations: a double corridor ward layout (two corridors with service in the middle area and bedrooms on the outer sides), and a single corridor ward layout (a single central corridor with rooms on both the left and right sides). Finally, among the wards with these typical configurations, we selected those with one- or two-bed mirrored rooms, with a windowed façade on one side, and private toilets and access door to the other (Figure 1), which represent the most recurrent room type in the modern hospital [16].

For each case study, a representative sample room of the ward acoustics was selected, avoiding marginal locations or special acoustic conditions. Our investigation took place in four wards within the main hospitals of the Tuscan Region (Italy). The in-patient wards examined are of the Maternity, Surgery, Neurorehabilitation and Endocrine Surgery Department.

3 Indoor acoustic comfort evaluation

Since noise disturbance within a hospital ward is mainly caused by the presence of simultaneous sources, and their propagation in the ward, as a result of the performance of the building elements, our survey proposal for the indoor acoustic comfort evaluation is based on achieving the following objectives:

- measurement of Sound Pressure Levels (SPL) in the ward during the day and night;
- measurement of the acoustic performance of building elements;
- use of a qualitative survey to understand the hospital setting.

3.1 SPL measurements

Sound Pressure Levels within a hospital setting are highly variable over time and caused by a variety of sound sources, which is why this investigation is based on:

- Long-time monitoring within a sample room (24 hours minimum) in order to detect temporal and spectral history of noise during the day and night, and assess noise events that might cause disturbance and awakening in patients;
- Short-time monitoring along the ward (20/30 minutes) to detect specific sources of noise or particular activities during the day;
- Measurements at 1 meter from each specific sound source to analyze the specific sources' contribution to the overall noise.

Time and spectral history were sampled with a time constant of 100 ms, in the frequency range from 20 Hz to 20 kHz, by means of a 2-channel real time analyzer 01dB Symphonie, with two ½'' diffuse field microphones. In both long-time and short-time monitoring, the microphones were always positioned at a height of 1.50 m from the floor and at least 1.0 m from any reflective surface. Carrying out monitoring during weekdays (Monday to Friday) is essential, avoiding Saturdays and Sundays, when ward occupancy is reduced.

Three conditions for carrying out the monitoring were evaluated:

- one bedroom in the actual condition of use (occupied room, with the door open);
- one bedroom in an ideal condition (unoccupied room, with the door closed);
- two mirrored bedrooms, one in the actual condition, and one in an ideal condition.

In this way, sound pressure level can be evaluated in terms of L_{Aeq}, L_{AFmax} and statistical levels such as L_{10}, L_{50} and L_{90} during day and night; additionally nighttime noise disturbance can be evaluated in terms of the amount of potentially disturbing events above a fixed threshold.

In addition, short-term monitoring was conducted along the ward and in front of the sample bedroom door, as well as measurements at the specific sources detected during field observations.

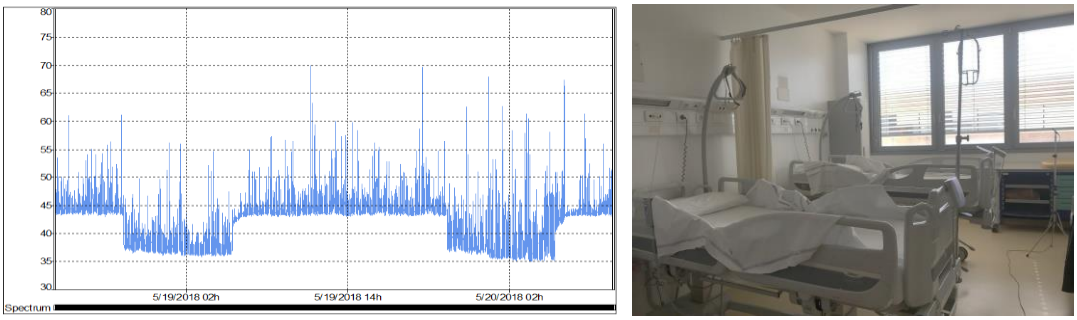

Figure 2: Example of one of the four case studies analysed: Maternity Ward. 48-h long-time monitoring within a typical bedroom in an unoccupied condition and door closed (on the left); photo of the typical bedroom during the long-time monitoring (on the right).

3.2 Measurement of acoustic performance of partitions and rooms

In order to assess the sound propagation within the ward, and patients' exposure to possible sound sources, the acoustic performance of the building elements must be evaluated in terms of reverberation time of the typical room and of the corridor [17], and normalized sound insulation of the partitions between adjacent rooms and between the bedroom and the corridor [18], (Figure 2). In our case studies, measurements were carried out by means of a 2-channel real time analyzer 01dB Symphonie, with two ½'' diffuse field microphones, using the MLS "Maximum Lenght Sequence" technique [19]. This technique was used to reduce excessive noise levels since measurements were carried out during normal activities in the hospital ward. These parameters can be compared with specific national standards. In the Italian context, the comparison was carried out with the 11367:2023 standards [13], referred to by the Italian Ministerial Decree of 23/06/2022 on Minimum Environmental Criteria [14].

Figure 2: Example of one of the four case studies analysed: Endocrine Surgery Ward. Sound insulation measurements of the partition between the bedroom and the corridor (on the left), and between two adjacent bedrooms (on the right).

3.3 A qualitative survey

During surveys, field observations were a valuable way to explore ward recurrences, practices and major sources of disturbance. At this purpose, during the 20/30-minutes short-time monitoring along the ward, noise sources were observed, grouped into seven categories and defined in time and space [20-22], Specifically, the seven groups of sound sources identified are:

- communication;
- anthropogenic noises (e.g., footsteps, coughing, etc...);
- personal electrical devices (e.g., cell phone ringing...);

- ward alarms; medical devices;
- equipment and/or furniture handling (e.g., carts, stretchers, etc...);
- electrical and mechanical systems.

In addition, semi-structured interviews were conducted with healthcare personnel to understand the actual use of ward spaces, functions, but also the needs and occupants' behavior [20, 21, 23].

4 Discussion and Conclusion

Noise pollution is one of the most worrying environmental factors, as it is increasing over time and as it is capable of having direct effects on human health, especially on the health of vulnerable people, including people who are sick or who are in the hospital. Indeed, the acoustic quality of hospital wards is a necessary condition for the comfort of patients and for the recovery process. Despite this, national and international standards usually give limit values for the building elements performances and for SPL values coming from outdoor, but not for the measurement procedure and for the selection of the case studies.

The aim of our research was to define intervention strategies to improve the acoustic quality inside in-patient wards. To do this, a survey protocol was structured which could be applied within general low-intensity care wards. The paper proposed the survey method, validated within four wards of four Italian hospitals, tracing its main phases: case study selection criteria, noise assessment, evaluation of the acoustic performance of the elements, and qualitative survey consisting of field observations and interviews with healthcare personnel.

The proposed survey was very effective in obtaining a large amount of data on the acoustic contexts of the wards, which are currently being processed. In addition, the hospitals involved in the project, as well as staff and patients, showed great interest and cooperation during the investigations, which is a sign of an actual issue that people really feel.

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Structure-borne vibration generated by a pallet jack exiting a service elevator

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Service elevator exhibits higher workloads due to the moving of goods when compared to passenger traffic. The wheel passage of service traffic over the sill on the elevator landing has been found in-situ to induce significant structure-borne noise which might be generally accepted in business spaces but may be found disturbing in apartments. Disruption of sleep may occur when an apartment is situated in the same building and the delivery of goods is carried out in the early morning (e.g. 4 AM). The level of structure-borne vibration generated by service traffic is of interest to prevent excessive noise in apartments. The vibration generation characteristics were studied with preliminary simulations incorporating the finite element method which were compared to in-situ measurements near the elevator landing sill. An impact hammer and a hand operated hydraulic pallet jack with hard nylon wheels exiting the elevator were used for acceleration measurements on the sill and nearby floor. The highest velocity levels were achieved when the fork wheels drop from the elevator onto the sill. The addition of around 90 kg load did not significantly increase the vibration levels in measurements. Loads near maximum capacity may increase the vibration levels significantly according to simulations. Height difference between the elevator and the sill may not be a significant factor when a reasonable range is considered. The preliminary investigation shows promise in sufficiently attenuating the noise by structural isolation and could be used as a starting point for construction and elevator designers to tackle this issue.

1 Introduction

Typically, the first floors of Finnish residential buildings are reserved for businesses such as grocery stores and restaurants. This eases the access to the businesses on ground level and adds privacy to the residential part of the building. The mix of spaces with different intended uses raises sound insulation concern; the airborne and impact sound insulation between residential and business spaces should reflect the noise and vibration characteristics of the business spaces. One such concern is the structure-borne sound caused by service or goods traffic which may occur in unfavourable times, e.g. early in the morning or late at night. This may disturb sleep and annoy the residents. The structure-borne sound transmission through a building caused by the impact of a wheel of transport equipment on the sill of a landing platform is discussed here.

Transient noise is mainly regulated in Finland with a maximum noise level given as an A-weighted and fast time weighted maximum sound pressure level measured in the room ($L_{p,AFmax}$). The maximum allowed value of $L_{p.AFmax}$ is 33 dB for dwellings which is lowered to 30 dB for impulsive noise to account for increased annoyance [1]. Impacts on structures generally result in impulsive sound and the stricter limiting value is used. The regulative performance is usually assessed by noise levels measurements in situ in case the annoyance has already occurred. Sound level measurements may be used to predict the efficiency of noise attenuation procedures, but their applicability is usually limited by the structures of the building in question.

Noise and vibration in the building generated by the operation of an elevator has been studied earlier [2, 3]. The examined sources typically include the elevator assembly that is involved in the movement of the elevator car and sources external to the assembly are less studied. The rolling noise from indoor transport equipment has been researched [4] but, according to the author's knowledge, have not been applied to the special case of transport equipment rolling over an elevator landing sill.

The manufacturer is responsible for the sill of an elevator landing platform since it is part of the elevator assembly. Development of noise attenuating sills and structural isolation designs by physical prototyping and measurements is expensive and time consuming due to product specifications and fire ordinances. Simulations provide an alternative, at least at the concept design stage, where no actual prototypes are needed for analysis, and hard-to-observe phenomena may be characterized. Furthermore, simulation enables us to understand the effects not only in proximity to the sill, but with different building interface combinations. Here, the factors influencing the structure-borne vibration of a wheel are investigated by measurements and simulations.

2 Materials and methods

2.1 Measurements

Acceleration measurements were performed near an in-use service elevator located below ground level in a shopping centre. An overview of the sill construction is shown in Figure 1. The sill profile is overhung into the elevator shaft and the supporting structure consists of three steel brackets (or consoles) that are fixed to the sill with bolts. The brackets are fixed onto the concrete wall of the elevator shaft. This solution is one of the two general design solutions of the manufacturer.

Figure 1: The measurement site: the investigated sill with an attached accelerometer (left) and the sill supporting structure (right).

Measurement points were located on the sill and on the concrete floor connected to the sill. Excitations were performed with two different sources: an in-use hand operated hydraulic pallet jack with uncovered hard nylon wheels (worn) and an impact hammer. The pallet jack was run over the sill 10 times for each configuration. The measurements were used for comparison with the simulation model. The maximum fast time weighted vibration velocity level $L_{v,Fmax}$ in 1/3-octave frequency bands between 20 to 500 Hz and the wideband maximum A-weighted fast time weighted vibration velocity level $L_{v,AFmax}$ were chosen as inspected quantities to match the sound pressure equivalents.

The pallet jack model was Still HPS25 with a wheel configuration 2 / 4 and a maximum load capacity of 2500 kg. The pallet jack wheels are shown in Figure 2. Steering wheel diameter was 200 mm and fork wheel diameter 80 mm. The impact hammer was Dytran 5803A. A single pallet jack measurement consisted of pulling the whole pallet jack out of the elevator until the forks are around 0,5 m away from the sill. The velocity of the pallet jack was determined with video feed and accelerometer signals. The measurements with the impact hammer included hitting the first edge of the sill in approximately vertical direction.

Figure 2: The pallet jack's steering wheel (left) and fork wheels (right).

Accelerations on the sill and on the concrete floor at distances of 1 m and 3 m from the sill outer edge and the force of the impact hammer were registered. The accelerometer on the sill was Wilcoxon WR786A and MMF KS48C on concrete. Pallet jack was either empty or loaded with four Euro pallets (á ~23 kg). The weight on steering wheel was measured to be 47 kg unloaded and 72 kg loaded. The fork wheels had a combined load of 18 kg unloaded and 88 kg loaded. Three initial sill height differences were investigated: flush with elevator, 10 mm lower than the elevator (drop) and 6 mm higher than the elevator (rise). The horizontal gap between the elevator and the sill was measured at around 27 mm. The sill height difference is affected by the load and hence changed during the run when the pallet jack is only partly in the elevator.

2.2 Simulations

The applicability of a sufficiently simple simulation model for prediction of structure-borne vibration of wheel contact and the use of the model for vibration attenuation studies are of interest. A 2D model of the investigated sill construction was employed. Simulations were performed in time implicit analysis with COMSOL 6.2 Multiphysics software. The geometrical model is shown in Figure 3. The model consisted of an elevator platform, landing sill, landing sill support structure (brackets or consoles), elevator shaft wall and floor under the sill, elevator wall over the sill, landing floor and insulation layer against ground under load-bearing structures. Due to the nature of the 2D simulation, the brackets were modelled as continuous in depth.

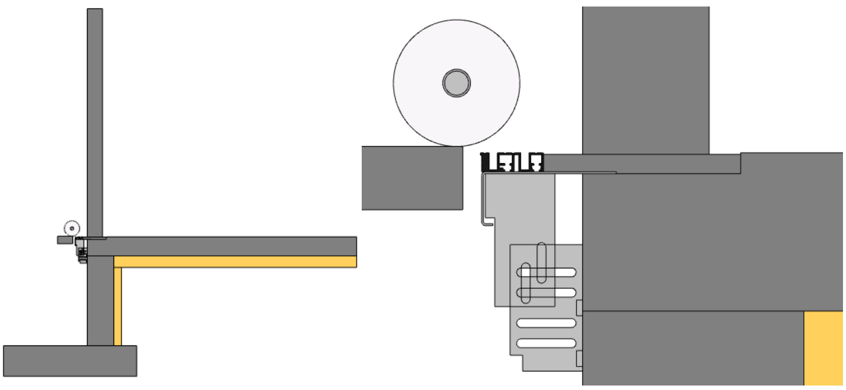

Figure 3: The model for the simulation (left: whole model, right: near the sill).

The structural drawings of the measurement site were not available and the actual thicknesses, structures and layers behind visible surfaces are unknown. General design guidelines and national solutions were followed. The structures were extended to around 3 m from the sill and their ends were attenuated with Low-Reflecting Boundaries. Maximum mesh size was 75 mm and was limited by the approximated minimum bending wave speed in concrete of around 224 m/s, maximum frequency band of interest (500 Hz) and N=5 elements per wavelength. CFL number of 0.15 was used for solver timesteps which corresponds to a timestep of 0.05 ms here. Simulation time was set to 0.3 s. A finer mesh of 0.5 mm was used for contact regions of wheel and sill.

Material properties are given in Table 1. The material parameters were based on literature and were not measured. Model consisted of linearly elastic materials and spring foundations, latter which were used to model the ground support. Minor Rayleigh damping was added to linearly elastic materials (ζ_1 = 0.01, f_1 = 40 Hz, ζ_2 = 0.01, f_2 = 1200 Hz), which corresponds to a loss factor of around 0.01 between 100...500 Hz. Solids were generally meshed together (i.e., bonded to each other) excluding the wheel and partly the bracket to concrete connection. A contact was added to the part where the bracket was not connected to the concrete with bolts. The bracket itself consisted of two bolt connected parts which were modelled as bonded connections here. The load bearing structures rested either on insulation boards or were supported by spring foundations (elevator shaft). The insulation boards were supported by spring foundations.

The wheel consisted of a rotating outer part and a horizontally moving non-rotating hub. Rotation of the wheel was achieved by contact friction and initial rotational velocities. The wheel hub moved at a constant velocity. The contacts between the wheel and the sill were modelled with Nitsche's incomplete formulation with exponential dynamic Coulomb friction (static friction coefficient μ_{stat} = 0.5 and dynamic friction coefficient μ_{dyn} = 0.3). The contacts between the wheel and the initial platform were modelled with penalty method with the same friction model and parameters.

A single wheel simulation took around 8 hours to solve with 8 cores total. No benefit in simulation time was achieved in this model when the core count is increased to 16. The simulation had around 83000 degrees-of-freedom and had a modest memory usage (less than 8 GB).

Table 1: Material parameters used in simulations.

Component	Material	Modulus of elasticity E	Density ρ	Poisson's ratio v
Sill	Aluminium	70 GPa	2700 kg/m^3	0.35
Sill support structure	Steel	210 GPa	7850 kg/m^3	0.3
Leveling	-	13 GPa	2000 kg/m^3	0.3
Load bearing structures	Concrete	30 GPa	2500 kg/m^3	0.3
Insulation layer against ground	EPS	12 MPa	20 kg/m^3	0.12
Ground (500 mm thickness)	-	1 GPa	-	0.3
Wheel, general	Nylon	4 GPa	1150 kg/m^3	0.4
Wheel, hub	Steel	210 GPa	7850 kg/m^3	0.3
Elastomer	-	0.2 MPa	150 kg/m^3	0.45

2.3 Model comparison methodology

The simulation model was compared to measurements in two steps with measured impulse hammer force excitations: first the maximum vibration velocities on measurement points on concrete were compared, and secondly, the maximum vibration velocities on measurement points on the sill and concrete were compared. Lastly, the steering wheel overpass was simulated with the setup corresponding to the measurements. The steering wheel was chosen due to its simpler modelling principles.

3 Results and discussion

3.1 Measured maximum vibration velocity levels generated by a pallet jack

The measurements were divided into six categories depending on the relative sill height (flush, drop, rise) and whether load was added on the pallet jack. The average velocity of the pallet jack before the sill was determined to be 0.6 m/s which was used in simulations. The velocity varied between 0.4 m/s and 0.9 m/s. The measured maximum A-weighted fast time weighted vibration velocity levels $L_{v,AFmax}$ were bandpass filtered to include only 20…500 Hz third octave bands to accommodate the chosen simulation limits. Bandpass filtered maximum vibration velocity level $L_{v,AFmax,20\text{-}500Hz}$ on the concrete floor at 3 m from the sill for each measurement is shown in Figure 4 for different wheelsets. The fork wheels are found to induce around 10…15 dB higher vibration levels than steering wheels. The highest vibration velocity levels are achieved when the pallet jack drops from the elevator onto the sill (measurements 21…40). The addition of load does not seem to have a significant effect on velocity levels, but the added load was small compared to the capacity of the pallet jack.

Based on the measurement results (Figure 4) and the maximum allowed maximum sound pressure level $L_{p,AFmax}$ = 30…33 dB, one may approximate the needed attenuation in vibration velocity level $\Delta L_{v,AFmax,20\text{-}500Hz}$. A rough estimate for the formed sound pressure levels $L_{p,AFmax}$ in an apartment nearby the elevator shaft were made using typical values for vibration attenuation and sound radiation [5, 6]. The attenuation needed from the loaded floor to the apartment is found to be as high as 35 dB.

3.2 Comparison between measurements and simulations

The maximum vibration velocity levels on sill and on concrete are shown in Figure 5 when the impact hammer was either hit on the sill or on the concrete 30 cm from the sill. The measured velocity levels on the sill are found to be around 20…30 dB higher than simulated values. On the other hand, the measured vibration velocity levels on concrete 3 m away from the sill are lower than simulated values. A more adequate agreement is found when the concrete is excited. Measured structural loss factors were found to be higher on the sill than simulations but adequate accuracy was found on concrete. Removing Rayleigh damping on the sill components did not affect results significantly. These findings may indicate that the sill and its connections are not accurately modelled due to the depth-wise continuous brackets in simulation.

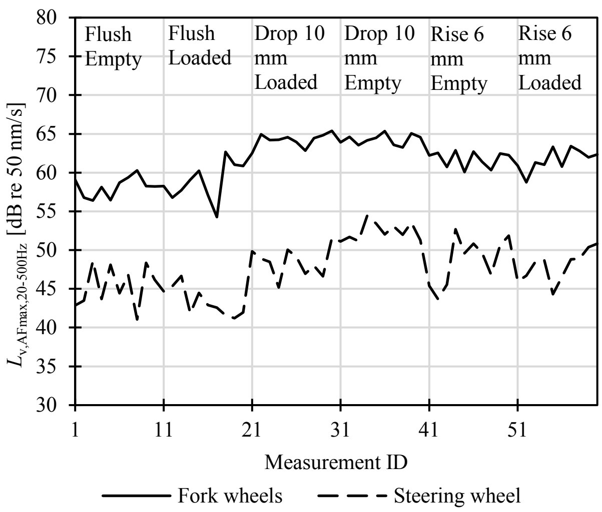

Figure 4: Measured bandpass (20-500 Hz) filtered A-weighted maximum vibration velocity levels $L_{v,AFmax,20\text{-}500Hz}$ of pallet jack overpasses for steering and fork wheels separately.

The measured and simulated maximum vibration velocity levels of the empty pallet jack's steering wheel on concrete floor 3 m away from the sill are shown in Figure 6 for different sill height difference cases. High deviation from measurements is found at frequency bands 160…500 Hz for all cases. The deviation is remarkably lower in frequency bands 50…125 Hz. The loaded pallet jack results are omitted due to the clarity of the paper. The loaded measurements tend to have the same or a little lower velocity level than the unloaded measurements and in contrast the simulations have a slight increase in levels due to load. The errors of impact hammer and pallet jack induced velocity levels are different in shape which implies inaccuracies in the wheel model, its load and contact modelling.

Uncertainty is included in the structural model depicting the measurement site since no precise structural drawings were available. The sill connection at the floor level was unknown and believed to be somewhat loose due to errors or wear since the leveling exhibited a hollow sound when hit with a hammer. Actual material parameters were also unknown, but it is believed that the realistic range of these parameters for the structural parts of the model do not explain the large discrepancies of order 20 dB since the velocity levels between points on the concrete wall reached adequate accuracy. This might not be the case for the material properties of the Nylon wheel where plastic deformation may occur on impact and elastic properties may have a larger range of acceptable values. The wheel surfaces also had wear and impurities. The natural frequency of the hydraulic cylinder of the pallet jack may also be around 80…200 Hz depending on the actual cylinder dimensions and oil bulk modulus [7]. The external load would then be sprung, and vibration would be attenuated at frequencies higher than the natural frequency more effectively. The chosen modelling method of the load is in principle unsprung.

Solver inaccuracies may occur when the mesh or solver timestep is too coarse. It was found that a CFL number 0.1 did not improve accuracy but would increase the solution time significantly. Decreasing the maximum mesh size provides more accurate resonance frequencies but did not affect the overall velocity levels in 1/3-octave bands significantly. The improved accuracy is believed to be linked to the insulating layer and ground spring supports. Hence, errors of this magnitude must come elsewhere.

According to the analyses, one of the main sources of inaccuracy was the chosen modelling approach for the studied sill type. The implementation of a 2D model offers simplicity and speed of calculation but ignores the discontinuous nature of the sill supporting structure. The bending of the sill between the brackets is not accounted for. Hence, the bracket connections are deemed too stiff and vibration-conducting in simulations. In analyses not shown here, it was found that modifying the nature of connection between sill and brackets into ideal springs or nonlinear displacements would affect the velocity levels in the concrete floor significantly. This, however, would need a separate analysis to correctly model the equivalent connection.

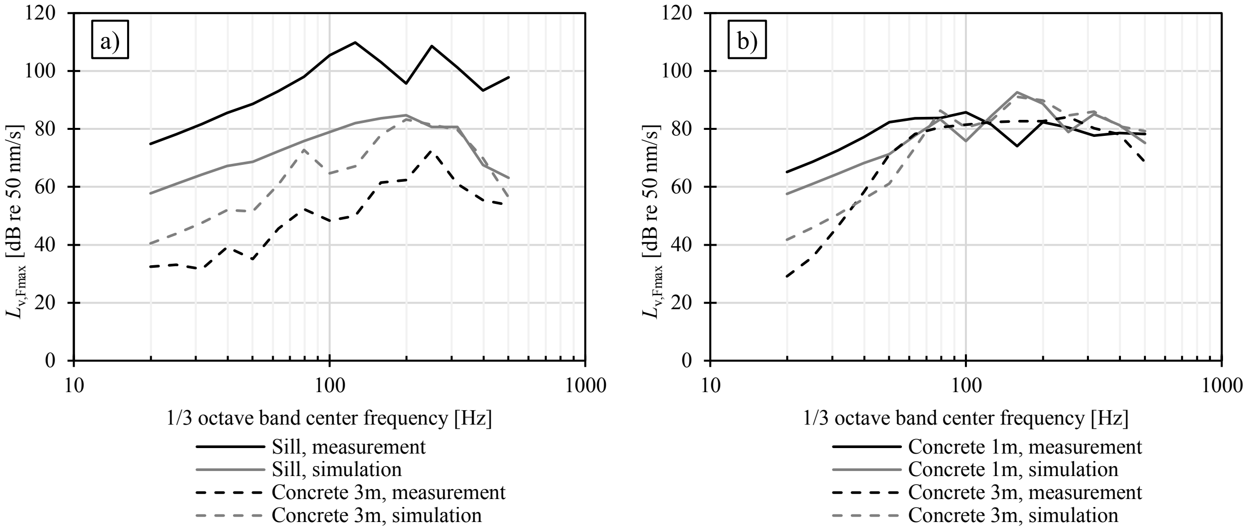

Figure 5: Maximum vibration velocity levels $L_{v,Fmax}$ in 1/3 octave bands of positions on sill and concrete for measurement and simulation when impact hammer excitation is on either a) sill or b) concrete 30 cm from sill.

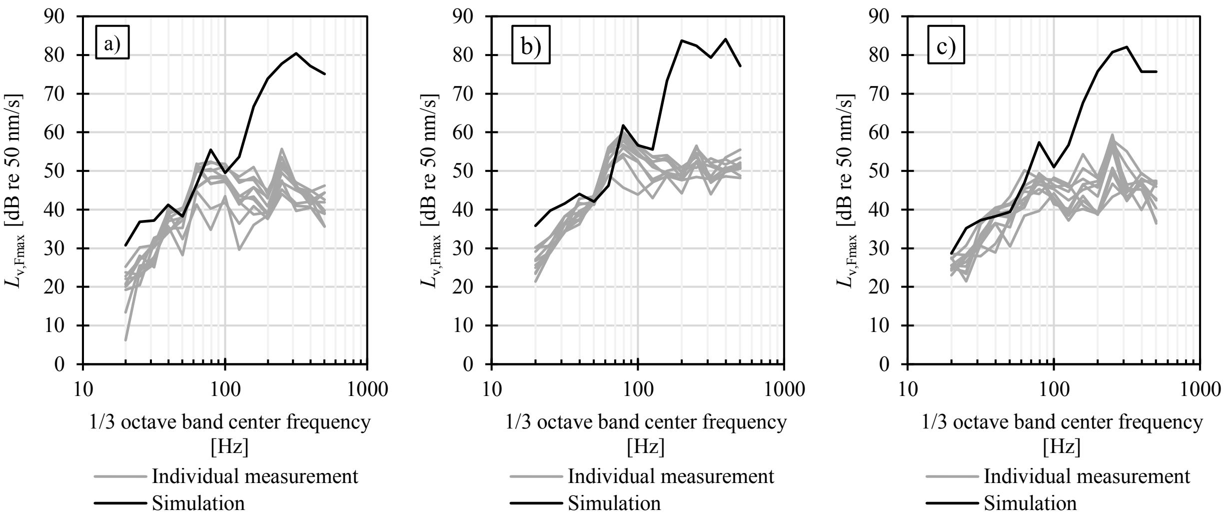

Figure 6: Measured and simulated maximum vibration velocity levels $L_{v,Fmax}$ in 1/3 octave bands generated by an *unloaded* steering wheel on concrete at 3m from the sill when the height difference from the elevator to sill is a) flush, b) 10 mm drop and c) 6 mm rise.

3.3 Parameter study

Even though the agreement between the measurements and simulation results presented in section 3.2 was not entirely adequate, it is interesting to study how the model behaves under different parameters. Sill height differences of -3 mm, 0 mm and 3 mm were studied, which correspond to a more realistic normal operation compared to the ones specified in the measurements in section 3.1. The wheel diameter was set to 80 mm since the fork wheels were found to have higher maximum velocity levels. The applied unsprung load onto the wheel hub was set to either 9 kg (empty) or 1000 kg (loaded). The open gap between the elevator and sill was set to 30 mm which corresponds to the maximum design value. The results are shown in Figure 7. The sill height difference does not affect the results significantly. High load increases the vibration velocity levels significantly (10...30 dB) especially in the low frequencies when the load is unsprung. This would further increase the vibration attenuation need approximated in section 3.1. Interestingly, the flush case is found to have the largest vibration velocity levels when the wheel load is 9 kg (empty). This may be due to the wheel dropping in the gap with more momentum and may be a dimension-specific finding.

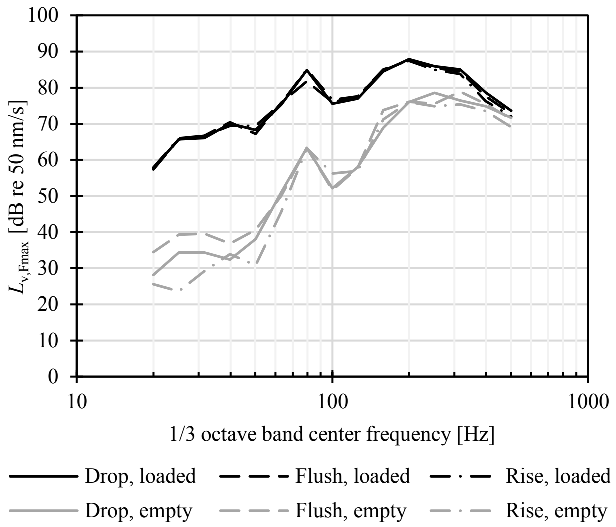

Figure 7: Simulated maximum vibration velocity levels $L_{v,Fmax}$ in 1/3 octave bands generated by a fork wheel on concrete at 3m from the sill with varying sill height difference and load.

3.4 Reduction of vibration velocity levels

The effect of reducing vibration velocity levels by separating the concrete floor from the elevator shaft wall was investigated when a small wheel drops 3 mm onto the sill (section 3.3). The separation was modelled as a 12.5 mm thick vertical strip of linear elastic material (elastomer) between the concrete slabs and between the supporting insulating layers. Elastomer's material properties are given in section 2.2. The insertion loss (*IL*) in maximum vibration velocity levels $L_{v,Fmax}$ is presented in Figure 8. Differences between simulation cases are found under 160 Hz, which indicates the result is excitation dependent. When high load is added, the differences diminish, and the excitation will become less varied.

The achieved level reduction in overall A-weighted vibration velocity level $L_{v,AFmax}$ (in 1/3 octave frequency bands 20…500 Hz) was approximated by subtracting the simulated insertion loss $IL(L_{v,Fmax})$ from a measured vibration velocity level $L_{v,Fmax}$. Measurements were used to not overestimate the role of frequencies over 125 Hz since they are overestimated in simulations. The average attenuation of $L_{v,AFmax}$ is found to be in the range of 20…40 dB. The structural isolation of the elevator shaft is found to have superior attenuation potential to the examined operational differences in section 3.3. Realistic structural isolation designs and attenuation goals should be explored per the actual building frame in question.

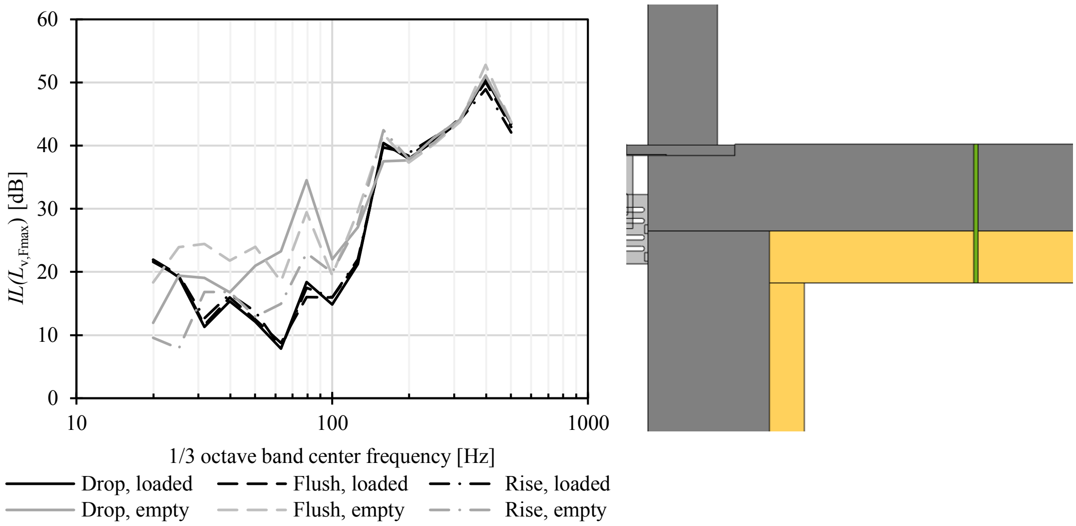

Figure 8: Left: simulated insertion losses IL($L_{v,Fmax}$) in 1/3 octave bands of a vertical strip with varying sill height difference and load when the position is on concrete at 3m from the sill. Right: the position and extent of the vertical strip (green).

4 Conclusions

Maximum vibration velocities of a pallet jack exiting a service elevator were investigated with measurements and simulations. A large difference between vibration velocities of steering and fork wheels were found in measurements. Large errors between simulation and measurements were found for both excitation types. The errors are believed to be associated with the structural modelling of the sill and its connections since the errors are found when the sill is excited with an impact hammer. The 2D presentation of the measured sill type is probably too stiff. In further research, the uncertainty caused by the unknown structural depictions should be eliminated to have a more accurate comparison with simulations before the wheel excitation is altered further. Errors in the wheel response could be associated with the plastic deformation of the wheel, bending of the sill or the sprung mass principle over the hydraulic pump.

In simulations, realistic sill height differences of ±3 mm had minor effect on maximum vibration velocity levels and very precise height tolerance is not necessarily needed for vibration attenuation purposes. However, a very high load increases the velocity levels significantly when the load is unsprung. The largest vibration velocity levels in simulations are generally found in frequency bands over 125 Hz where the largest errors are also found. A preliminary vibration attenuation procedure was examined with simulations where the floor is separated by a vertical elastomer strip at 1 m from the sill. The attenuation results vary between simulation cases and approximately the average attenuation in the maximum A-weighted vibration velocity level $\Delta L_{\mathrm{v,AFmax,20\text{-}500Hz}}$ is 20…40 dB when subtracted from measured values. The preliminary investigation shows promise in sufficiently attenuating the noise generated by a wheel overpass over a landing sill by structural isolation but should be extended to actual design possibilities. The results could be used as a starting point for construction and elevator designers to tackle this issue.

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Survey measurement of impact sound insulation of concrete walls

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Structure borne noise from appliances attached to walls, e.g., heat pumps, air conditioners, elevators, ventilation, cooling appliances, and doors, can cause noise in residential buildings. Impact sound insulation of floors is usually measured using a tapping machine. However, similar instrument is not available for impact sound insulation measurements of walls. This study investigates the possibility of impact sound insulation measurement using an impact hammer, force transducer, and sound level meter. One 160 mm concrete floor and four similar 160 mm concrete walls were tested. The force, and sound pressure levels of series of impacts were measured simultaneously. Impact force levels could be produced manually with tolerable repeatability with the impact hammer. Sound pressure levels in the receiving rooms had similar variation as in any sound insulation measurements. Reverberation times were measured to enable determination of normalized impact sound pressure levels. The weighted normalized impact sound pressure level $L'_{n,w}$ of the four walls was 89–93 dB. We also compared the hammer method against tapping machine on a 160 mm steel-reinforced concrete floor and the $L'_{n,w}$ results differed only 3 dB. The results are promising but more research is needed for different heavy-weight materials such as light concrete, heavy concrete, and brick.

1 Introduction

Structure borne noise from appliances attached to walls, e.g., heat pumps, air conditioners, elevators, ventilation, cooling appliances, and doors, can cause noise in residential buildings. Impact sound insulation of floors is usually measured using a tapping machine. However, similar instrument is not available for impact sound insulation measurements of walls.

This study focuses on the measurement of impact sound insulation of walls which is not possible using standard tapping machine. We investigated the measurement method presented previously by Bailhache et al. [1]. The method uses impact hammer, force sensor, and sound level meter to determine impact sound insulation in a comparable way to method using standard tapping machine.

2 Materials

2.1 Measurement location

The measurements were conducted for five constructions of an acoustics laboratory (Figure 1). Four 160-mm-thick steel-reinforced concrete walls, and one 160-mm-thick steel-reinforced concrete floor were investigated. The sound pressure levels (SPL) were measured in reverberation rooms 1–3. The volume of rooms 1–3 was 76 m^3, 69 m^3, and 201 m^3, respectively.

Walls **1** and **2** are part of room 3 which is entirely isolated from the building using vibration isolators. Walls **3** and **4** are part of room 1 which is also entirely isolated from the building using vibration isolators. The floor **5** is mounted on a steel frame that is rigidly connected to the building. However, room 2 underneath is entirely isolated from the building using vibration isolators. The numbers **1**–**5** are used as reference in the following sections.

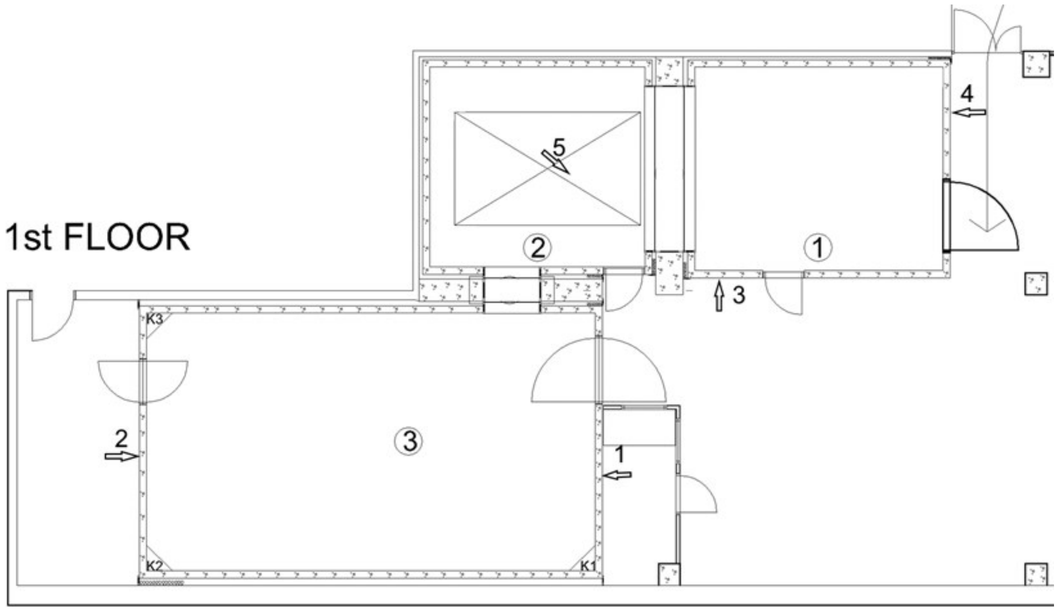

Figure 1: Layout of the 1st floor of the acoustics laboratory (rooms 1–3). The investigated walls and the floor are indicated with arrows. The floor was on top of room 2 and tapping machine was located on 2nd floor.

3 Measurement methods

The method using an impact hammer with force sensor and sound level meter has been introduced in ISO 12354-2 Annex F [2]. The force level of the impact, $L_{F,hammer}$ [dB re μN], and SPL in the receiving room, $L_{p,hammer}$ [dB re 20 μPa], are measured simultaneously. The SPL of standard tapping machine, $L_{p,TM}$ [dB], is determined indirectly using force level correction that depends on the force levels produced by the hammer, $L_{F,hammer}$, and the standard tapping machine, $L_{F,TM}$ [dB]. The method is expected to be applicable for heavy monolithic building elements when low point mobility can be assumed, and frequency is above the critical frequency.

3.1 Force level and SPL

The force level of impacts, $L_{F,hammer}$, was measured using impact hammer (Kistler 9726A5000) which has force sensor and steel tip. The signal was filtered using 10-Hz-high-pass-filter, recorded with 12.8 kHz sampling rate, and analysed using a real time analyser (Soundbook MK2).

The SPL of hammer impacts in the receiving room, $L_{p,hammer}$, was measured using microphone (GRAS 40AF), preamplifier (GRAS 26AK), and the real time analyser. Four impact positions were used for constructions **1**–**5**. The measurements were conducted in five microphone positions for each impact position. The measurement time was 5 seconds (10 impacts).

Standard tapping machine produces 10 impacts per second. In our study, two hammer impacts were produced per second (120 BPM). This meant that equivalent SPL of the hammer impacts was lower than that produced by tapping machine even when the force level of a single impact was equivalent to the single impact of the tapping machine. Therefore, the measured SPL of the hammer impacts needed to be corrected with a constant K [dB] taking into account the different impact frequencies:

$$K = 10\log_{10}(10/2) \approx 7\ dB \tag{1}$$

The SPL of standard tapping machine was calculated by

$$L_{\text{p,TM}} = L_{\text{p,hammer}} - L_{\text{F,hammer}} + L_{\text{F,TM}} + K \tag{2}$$

where $L_{F,TM}$ [dB] was the theoretical force level of standard tapping machine according to EN 12354-5 Annex F [3]. $L_{F,TM}$ is presented in Table 1.

3.2 Reverberation time

The total sound absorption in the room affects the normalized impact SPL, L'_{n} [dB], according to ISO 16283-2 [4]. The sound absorption area, A [m^2], was determined using the mean reverberation time, T [s] by

$$A = 0.16V/T \tag{3}$$

where V [m^3] was the volume of the room. The reverberation time was measured according to ISO 3382-2 [5] using two loudspeaker positions and three measurement positions. The test signal was interrupted pink noise. The decay range in determination of T was –5 to –25 dB. The measurements were conducted in 1/3-octave bands 100–3150 Hz. The results of six combinations were averaged to obtain the mean reverberation time.

3.3 Normalized impact SPL

The normalized impact SPL, L'_{n}, was determined by

$$L'_{n} = L_{p,TM} + 10\log_{10}(A/A_0) \tag{4}$$

where A_0=10 m^2. The normalized impact SPL, L'_{n}, of the floor **5** was also measured using standard tapping machine (Nor211A) according to ISO 16283-2 [4]. The weighted normalized impact SPL, $L'_{n,w}$, was determined according to ISO 717-2 [6] (Figure 2b).

4 Results

The normalized impact SPL of the walls **1**–**4** and floor **5** is presented in Figure 2a. They were measured using the impact hammer method. The normalized impact SPL of the floor **5** is presented in Figure 2b. The results were measured using tapping machine and impact hammer method. The averages of $L_{F,Hammer}$ are presented in Table 1. The averages of measured $L_{p,Hammer}$ are presented in Table 2. The reverberation times are presented in Table 3. The weighted normalized impact SPLs of **1**–**5** are presented in Table 4.

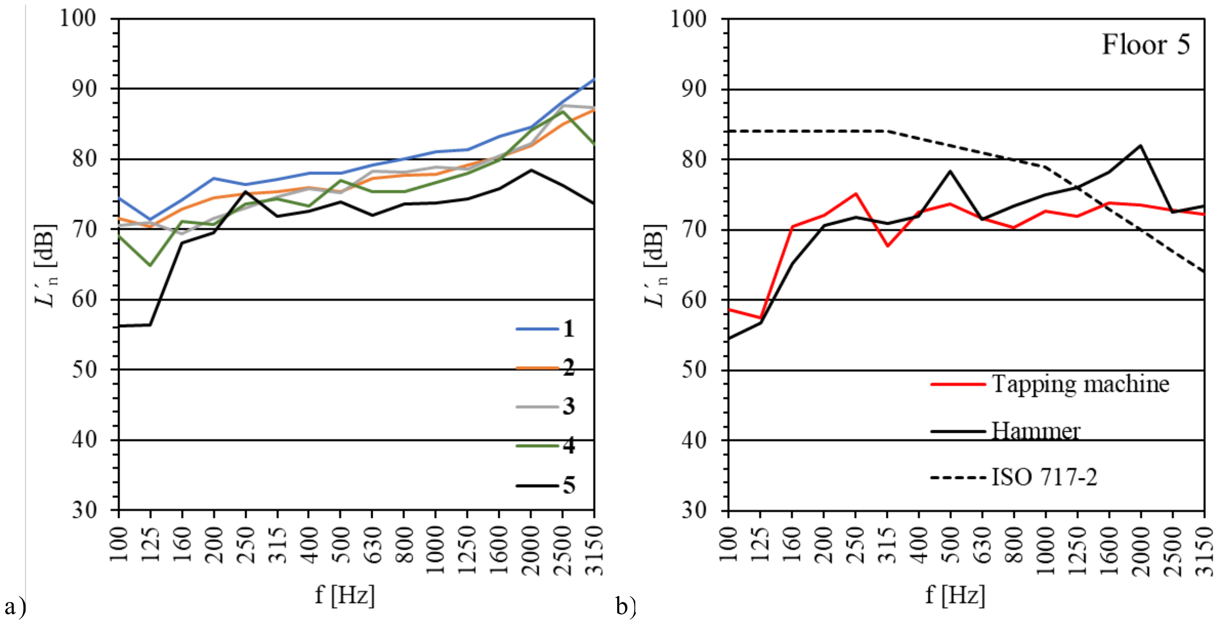

Figure 2: a) The normalized impact SPL, L'_{n}, of the walls **1**–**4** measured using impact hammer. b) L'_{n} of the floor **5** measured using both impact hammer and tapping machine. The reference curve is at $L'_{n,w}$ = 82 dB. All walls and floor were made of 160 mm steel-reinforced concrete.

Table 1: Theoretical force level of tapping machine $L_{F,TM}$ [3] and the averages of measured force levels $L_{F,Hammer}$ in constructions **1**–**5**.

f	[Hz]	100	125	160	200	250	315	400	500	630	800	1000	1250	1600	2000	2500	3150
$L_{F,TM}$	[dB]	139.0	140.0	141.1	142.0	143.0	144.0	145.1	146.0	147.0	148.1	149.0	150.0	151.1	152.0	153.0	154.0
1 $L_{F,Hammer}$	[dB]	131.1	132.0	133.1	134.1	135.0	136.0	136.9	137.7	138.4	139.0	139.4	139.4	138.6	136.4	131.1	124.0
2 $L_{F,Hammer}$	[dB]	131.3	132.3	133.3	134.3	135.2	136.2	137.1	137.9	138.6	139.2	139.4	139.3	138.4	136.0	130.2	124.0
3 $L_{F,Hammer}$	[dB]	131.4	132.4	133.4	134.4	135.4	136.3	137.2	138.0	138.8	139.3	139.6	139.5	138.4	135.9	129.5	125.3
4 $L_{F,Hammer}$	[dB]	132.8	133.8	134.8	135.8	136.7	137.6	138.4	139.1	139.6	139.9	139.7	138.9	136.8	132.7	126.7	127.4
5 $L_{F,Hammer}$	[dB]	133.6	134.6	135.6	136.5	137.4	138.3	139.0	139.5	139.8	139.8	139.2	137.7	134.5	127.7	121.7	124.7

Table 2: The averages of measured SPLs $L_{p,Hammer}$ in the receiving room while measuring constructions **1**–**5**.

f	[Hz]	100	125	160	200	250	315	400	500	630	800	1000	1250	1600	2000	2500	3150
1 $L_{p,Hammer}$	[dB]	60.7	58.4	60.9	62.2	61.7	62.6	62.9	62.9	63.7	63.8	64.1	63.1	62.5	60.3	57.0	51.3
2 $L_{p,Hammer}$	[dB]	59.7	58.9	62.1	62.0	62.5	62.7	63.3	62.1	63.9	63.6	62.8	62.5	61.2	58.9	54.5	48.4
3 $L_{p,Hammer}$	[dB]	60.9	61.9	60.7	62.6	64.3	65.1	66.2	66.1	69.0	68.3	67.9	65.5	64.9	62.5	59.8	53.2
4 $L_{p,Hammer}$	[dB]	63.9	59.5	65.3	64.6	66.5	67.3	66.2	69.7	67.8	66.9	66.6	65.4	63.5	62.0	57.0	51.0
5 $L_{p,Hammer}$	[dB]	46.2	45.1	56.0	57.7	63.8	60.4	61.2	61.4	59.3	59.9	58.4	56.3	53.2	47.4	38.1	37.0

Table 3: The reverberation time T in the receiving room used to test constructions **1**–**5**.

	f	[Hz]	100	125	160	200	250	315	400	500	630	800	1000	1250	1600	2000	2500	3150
1	T	[s]	4.3	5.1	4.6	3.2	3.5	3.5	3.3	3.3	3.3	3.1	3.0	2.7	2.4	2.2	1.9	1.5
2	T	[s]	6.1	6.6	8.0	5.4	5.3	5.3	5.4	4.9	5.1	4.9	4.5	4.1	3.6	3.2	2.7	2.2
3	T	[s]	3.9	4.3	4.7	4.5	4.6	4.1	4.1	4.7	4.8	4.6	4.3	3.4	3.0	2.7	2.2	1.7
4	T	[s]	7.6	7.3	6.8	6.5	5.1	5.3	5.5	5.6	5.7	5.7	5.1	4.3	3.8	3.3	2.8	2.2
5	T	[s]	1.9	1.4	1.2	1.3	1.4	1.5	1.6	1.4	1.5	1.6	1.5	1.5	1.4	1.2	1.1	1.0

Table 4: The weighted normalized impact SPL, $L'_{n,w}$, in **1**–**5**. Results of impact hammer method are marked with (H) and results of tapping machine method with (TM).

	1 (H)	**2 (H)**	**3 (H)**	**4 (H)**	**5 (H)**	**5 (TM)**
$L'_{n,w}$ [dB]	93	90	91	89	82	79

5 Discussion

The impact hammer method of Ref. [1] produced promising results. The weighted normalized impact SPL of the walls **1**–**4** was 89–93 dB $L'_{n,w}$ which is tolerable variation for measurements of similar structures in different locations. The result of floor **5** was $L'_{n,w}$ = 82 dB which was 3 dB higher than the result $L'_{n,w}$ = 79 dB measured using the standard tapping machine. Such difference is acceptable for survey measurements.

The normalized impact SPL L'_n of the floor **5** was in good agreement with measurement result using tapping machine except in 2000 Hz. L'_n of the floor **5** with hammer was 7–11 dB lower than the L'_n of the walls **1**–**4**. A probable reason for this is that the walls were part of resiliently mounted reverberation rooms (lower coupling loss factor) while the floor was mounted on a rigid building frame (higher coupling loss factor). Similar effect of resilient joints on airborne sound insulation has been investigated in laboratory by Keränen and Hongisto [7].

In walls **1**–**4**, floor to wall joint had significantly lower velocity level difference than the joint between the floor **5** and the surrounding building. This caused higher impact SPLs in measurements **1**–**4** because the connected floor and wall surfaces (110–220 m^2) also radiated sound into the receiving room. In **5**, only the floor (10 m^2) radiated sound into the receiving room.

Standard deviation of $L_{F,Hammer}$ was under 3 dB within 1/3-octave bands 100–3150 Hz in constructions **1**–**5**. This standard deviation is very low and verifies that it is possible to produce adequate accuracy using manually operated hand-held impact hammer. Standard deviation of $L_{p,Hammer}$ was 1–5 dB within 1/3-octave bands 100–3150 Hz in constructions **1**–**5**. The standard deviation is, typically, within the same range in impact sound insulation measurements in field conditions [8].

All studied constructions were made of 160 mm thick reinforced concrete. Walls were prefabricated elements and floor was cast on site. Wall **4** had plastered and painted surface that deteriorated during the measurements due to hammer impacts. This caused more deviation in force levels than in the measurements of walls **1**–**3**. This affected also the deviation of SPLs $L_{p,Hammer}$. The softer surface is possible reason for the lowest value of the walls, $L'_{n,w}$ = 89 dB.

6 Conclusion

We evaluated the hammer method of Bailhache et al. to determine the impact sound pressure level of vertical surfaces, where tapping machine cannot be applied. We tested the method for four different 160 mm steel-reinforced concrete walls and the results ranged from 89 to 93 dB $L'_{n,w}$. We also compared the hammer method against tapping machine on a 160 mm steel-reinforced concrete floor and the $L'_{n,w}$ results differed only 3 dB.

The results are promising. The method can be applied, e.g., for testing the impact SPL of vertical constructions. However, more research is needed for different heavy-weight materials such as light concrete, heavy concrete, and brick.

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Gymnasium activity noise in residential and educational buildings

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Sports and exercise activities in gymnasiums are a major source of noise to adjacent spaces. For example, the activities may include running, ball games, jumping and other impulse sound events. Ideally, architectural layout planning would be guided so that the gymnasium will be away from noise sensitive spaces. In many renovation projects and some new builds, it is not possible to avoid challenging situations. Arguably cases where noise sensitive spaces are below a gymnasium, are the most technically demanding ones for acoustics. In Finland, acoustic design of such cases is based on a statutory requirement to consider the intended use of the spaces and not to cause harm to the inhabitants. Acoustic guidelines in terms of airborne sound insulation $D_{\mathrm{nT,w}}$ and impact sound insulation $L'_{\mathrm{nTw}} + C_{\mathrm{I,50\text{-}2500}}$ are typically insufficient for the assessment of user comfort and possibility of harm caused. Noise level guidelines for assessing sleep disturbance or other health effects in terms of $L_{\mathrm{Aeq,07\text{-}22h}}$, $L_{\mathrm{Aeq,22\text{-}07h}}$ and $L_{\mathrm{Aeq,1h}}$ generally require real world measurement data. This paper explores acoustic design considerations of cases where gymnasiums are located above residential and educational spaces in both renovation and new build projects.

1 Introduction

Typically, sports facilities are located away from noise sensitive spaces, but in some cases challenging adjacencies cannot be avoided. Especially in new buildings it is possible to guide space planning towards solutions, where noise sensitive spaces would not be placed next to gymnasia. A more common scenario would be an existing school building where the gymnasium is not on the ground floor.

Gymnasium activities may include sports as well as events. The noise levels caused by different activities vary significantly so it is important to know what to design for as early as possible. In the cases presented here, the main design concern was impact noise from sports and exercise activities.

Designing for impact sound insulation of floor structures typically includes studying the base floor and flanking sound paths and predicting the impact of sound insulation improvement layers. Studies for the base floor and flanking sound paths may include testing, simulation, and literature research. For the sound insulation improvement layers, such as flooring, ceilings and wall linings, simulation and literature research are typically done before testing can be conducted.

In the case studies presented here, the acoustic design focus has been on achieving sufficient impact sound insulation performance of the floor structures. Some of the design tools utilized have included commercial sound insulation prediction software, sound insulation calculation models presented in literature, sound insulation data presented in literature, and sound insulation data from in-situ testing.

Impact sound insulation performance of floor structures is typically characterized by standardized quantities. In the case studies presented here, some alternative drop impact sound sources are used. These sound sources have been selected to better represent other impact sounds than walking.

2 Statutory requirements and guidelines in Finland

2.1 Sound insulation

The statutory requirement for airborne sound insulation is $D_{nT,w} \geq 55$ dB and for impact sound insulation it is $L'_{nT,w} + C_{I,50\text{-}2500} \leq 53$ dB between dwellings [1]. Sports and exercise facilities specific guideline values are $D_{nT,w} \geq 57$ dB and $L'_{nT,w} + C_{I,50\text{-}2500} \leq 46$ dB to adjacent spaces [3][4]. However, the statutory requirements also state that "*…if the dwelling* […] *is structurally connected with spaces where intense, particularly annoying or low frequency noise is generated, in the implementation process special consideration shall be given to the implementation of sufficient sound insulation.*"

For educational buildings, the sports and exercise facilities specific guidelines are typically followed.

2.2 Noise levels

The noise level limits related to gymnasium activity are meant to evaluate adverse health effects. The statutory requirements for noise in living spaces within dwellings are $L_{Aeq,07\text{-}22h} \leq 35$ dB in the daytime and $L_{Aeq,22\text{-}07h} \leq 30$ dB in the night-time [2]. Additionally, in residential rooms intended for sleeping low-frequency noise or other noise that may cause sleep disturbances and is clearly distinguished from the background noise shall not exceed $L_{Aeq,1h} \leq 25$ dB [2]. Furthermore, limits on one-third-octave bands are given for low frequency noise in residential rooms intended for sleeping as presented in Table 1 below [2].

Table 1: One-third-octave band limits for low frequency noise in residential rooms intended for sleeping.

One-third-octave band [Hz]	20	25	31,5	40	50	63	80	100	125	160	200
Nighttime (hrs. 22-07) $L_{eq,1h}$ [dB]	74	64	56	49	44	42	40	38	36	34	32

For teaching spaces, or other spaces where speech needs to be intelligible, without a sound amplification system the statutory requirement is $L_{Aeq,07\text{-}22h} \leq 35$ dB in the daytime and with an amplification system the requirement is $L_{Aeq,07\text{-}22h} \leq 40$ dB [2]. In working spaces, the requirement is $L_{Aeq,07\text{-}22h} \leq 45$ dB.

Penalties for impulsive and tonal noise are described in the statutory requirements [2]. For impulsive noise the penalty is 5 dB or 10 dB based on the audibility and sound level of the impulsive noise event. For tonal noise the penalty is 3 dB or 6 dB based on the audibility of the tonal component of the noise. In case of simultaneous impulsive and tonal noise at the same time, only the higher penalty value is applied for the duration of the impulsive and tonal noise event. [2][5]

However, equivalent sound levels are not sufficient for evaluating the disturbance of short-term impulse noise events. Therefore, assessing the maximum sound levels is advised. Noise level limits for the maximum sound levels should also be implemented in the future.

3 Acoustic design process

3.1 Building design information

The first step in the acoustic design process is to identify noisy spaces such as a gymnasium space and any adjacent noise sensitive spaces. At this point it is necessary to inform the client and the design team of such adjacencies and of the potential implications. If the space arrangement cannot be altered towards a more favourable one, then technical solutions need to be explored.

Main acoustic concerns often occur at the immediate separating structures. By studying the proposed or existing wall and floor constructions an overview of potential acoustic issues is formed. Structure build-up airborne and impact sound insulation is evaluated by calculations or comparing them with on-site test results of similar structures. Potential risks may include low overall mass, or lightweight outer leaves of the structures.

The load-bearing capacity of structures can restrict the acoustic solution that can be implemented, even in new buildings. The load capacity restriction may come from the building foundations or from the superstructure.

In renovation projects there are cases where the existing floor has infill materials that need to be removed. In buildings from the late 1800s to the mid-1900s in Finland the infill consists of wood shavings, brick pieces and other construction waste material. With current building regulations the replacement infill can rarely match the mass of the removed material, which means a lower overall weight of the floor structure. Lower weight generally means lower sound insulation and a more challenging acoustic design case. Replacing lightweight insulation infills with new similar material typically does not affect the acoustical performance of the structure.

Further studies into doors, windows, ventilation duct systems and other potential flanking sound paths are needed for a more detailed assessment.

3.2 Baseline testing

In case of existing structures, it is important to conduct in-situ testing. Testing may include measurements, for example, of airborne sound insulation, impact sound level, activity sound level, controlled event sound level, or vibration level.

Standardised airborne sound insulation and impact sound level test methods provide consistent and easily comparable data. However, in case of sports and exercise activities the standardized test results may not always be sufficient for evaluating the noise conditions. For example, when the activity impact forces exceed those of the ISO standard tapping machine.

In addition to standardized impact sound test results the maximum sound levels such as L_{AFmax} of real-life gymnasium activities are particularly interesting.

Activity sound level monitoring can be useful for finding the most significant noise sources. However, it requires that the noise events are identified either in person, recognised automatically or by browsing through video and audio recordings after the monitoring period. Controlled test settings may be more efficient if they can be selected so that the results are comparable with typical use cases.

In the cases presented here, controlled event sound level testing has been done with various simulated activities. Here are some examples of the methods used and photos of some of them in Figure 2:

- Impact sound source (ISO standard tapping machine).
- 2 kg medicine ball drop from 1 m height.
- 12 kg kettlebell drop onto 25 mm elastic pad from 0,7 m height.
- 35 kg kettlebell drop onto 25 mm elastic pad from 0,5 m height.
- Walking with and without shoes.
- Jogging with and without shoes.
- Jumping.
- Basketball dribbling.

Figure 1: Examples of controlled sound sources.

Comparison between some of the controlled noise sources from a case study is presented below in Figure 2. While the standard tapping machine signal to noise ratio in the low frequency range was adequate, the medicine ball and kettlebell drops provided signal levels which were closer to those of the simulated activities. This suggests that the tapping machine as a noise source could be insufficient for structure-borne noise evaluation in cases where the structure-borne noise isolation is high.

The medicine ball drop is comparable with the standard rubber ball drop. The resulting spectrum corresponds better with that of simulated activities than the ISO standard tapping machine result, which has been shown for the standard rubber ball drop as well [6]. It can also be seen in Figure 2 that the 12 kg kettlebell drop seems to better correspond with higher impact activities, than the medicine ball drop.

Based on the various test methods and case experiences, three impact sound test types would be recommended for gymnasium floors:

- Standard tapping machine impact sound level.
- Standard rubber ball or equivalent ball drop impact sound level.
- 12 kg kettlebell or equivalent device drop impact sound level.

The standard tapping machine impact sound level test is recommended for comparison with standard guidelines. However, these results alone should not be used for noise mitigation design.

The standard rubber ball or equivalent ball drop impact sound level test is recommended for objective evaluation of low impact activity sound level. Activities such as walking or jogging could be seen as low impact activities.

The 12 kg kettlebell or equivalent device drop impact sound level test is recommended for objective evaluation of intermediate impact activity sound level. Activities, such as ball sports could be seen as intermediate impact activities.

Higher impact activities, such as rhythmic group exercises, exercises including weight drops or weightlifting will likely require heavier weight drop tests.

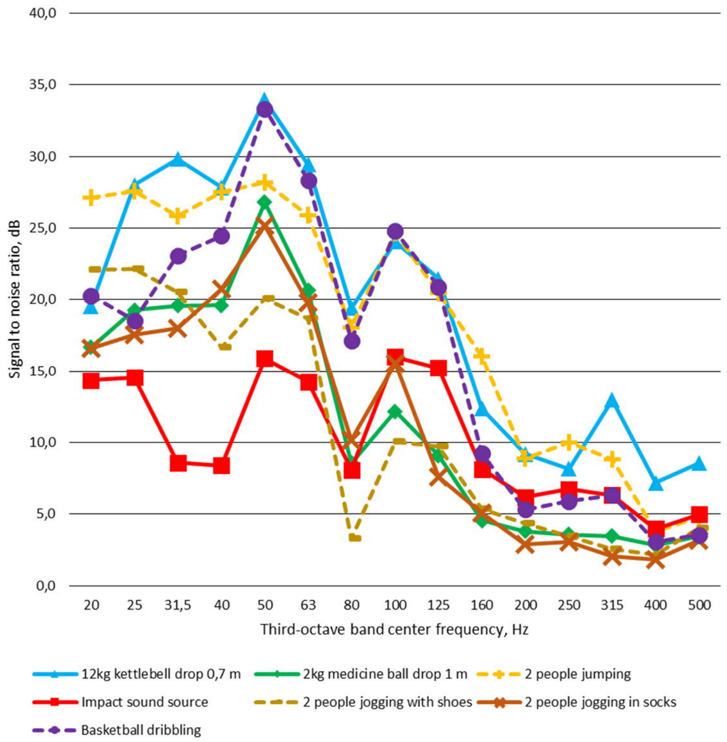

Figure 2: Signal to noise ratio for various controlled sources in Case C.

3.3 Proposed acoustic solutions

Floor build-up sound insulation is analysed based on calculations and simulations, as well as comparing them with baseline test results. Analysis findings are compared with the acoustic design requirements and needs for improvements are identified. Improvements to gymnasium floor structure impact sound insulation generally involve floating floor and suspended ceiling solutions.

Floating floor design needs to consider the bearing structure properties and the expected sound and vibration excitation sources. The floating floor natural frequency is designed to be well below the excitation frequencies in order to gain sufficient attenuation. Similarly, the load-bearing structure would ideally be as stiff as possible, so that the achievable insertion loss is as high as possible.

Suspended ceiling solution design is similar to the design of floating floors when elastic hangers are used. Some benefits of suspended ceilings when compared to floating floors are generally larger air gaps and even completely detached mounting. Some disadvantages are relatively low possible mass and more complex coordination with building services installations.

The gymnasium activity induced vibration should also be taken into consideration to avoid secondary noise sources, such as rattling of ceilings.

Other impact sound improvement options may include resilient floor coverings, load-bearing floor mass addition and wall linings. Resilient floor coverings provide impact cushioning, which is beneficial for reducing impact sounds. Adding mass to the load-bearing floor is not easy, since stiffness comes easily along with mass and increasing both would diminish the overall effect. Wall linings are typically needed if flanking sound paths along vertical structures require attenuation. When considering wall linings it needs to be noted, that for low frequency attenuation large air gaps may be required.

Combinations of floating floors, suspended ceilings and load-bearing floor improvements increase construction complexity, so an ideal solution would involve as few of the options as possible. If there are limitations regarding overall weight and build-up height, then the solution needs to be optimized within the limitations. For example, concentrating mass on the radiating receiver side plate may be more efficient than on the source side floating floor plate.

In general, a floating floor solution would be recommended rather than suspended ceiling solutions. The floating floor addresses the sound source directly while suspended ceilings only affect one room surface radiating sound.

3.4 Commissioning testing

After the designed solutions are constructed on site, their performance needs to be verified with acoustic testing. The same test methods which were used for baseline testing should be used for commissioning testing as well.

In case of novel solutions, it is useful to conduct commissioning testing in stages to assess agreement with predicted performance. Especially testing the bare load-bearing structure performance against a prediction model provides valuable information about the expected outcome. A proposed outline for a commissioning testing plan would be as follows:

- Bare load-bearing structure performance.
- Load-bearing structure together with the floating floor solution before installation of flooring.
- Load-bearing structure together with the floating floor and ceiling solutions before installation of flooring.
- Load-bearing structure together with the floating floor, ceiling and wall solutions before installation of flooring.
- Final solution with flooring installed.

It is not always possible to test the performance at each stage.

4 Case studies

4.1 A gymnasium above residential use in a new building – Case A

Located in Helsinki, this building completed in 2019 has residential use on the lower floors and a gymnasium on the top floor. This unique arrangement meant that typical sports flooring solutions would not have been sufficient. Stringent acoustic requirements meant that a room-in-room solution was needed. Since the gymnasium could not be supported on separate foundations, a heavy-duty floating floor was designed.

The design for the floor structure build-up was as follows:

- 55 mm timber sports flooring
- 105 mm reinforced concrete
- 265 mm hollow core concrete slab
- 150 mm cavity with elastic bearings:
    - 15 mm steel plate
    - 100 mm elastomer bearings
    - 35 mm cement screed plinth
- 320 mm hollow core concrete slab

Commissioning testing was carried out on site at the end of the construction stage before the installation of the sports flooring. The testing was performed between the gymnasium and two of the apartments below. The measurement results were limited by background noise conditions, and the 35 kg kettlebell drop was only barely audible. Commissioning test results are presented in Table 2 below.

Table 2: Commissioning test results for Case A.

Test type	Test result
Airborne sound insulation	R'_{w} = 67…69 dB[1)]
Impact sound level	$L'_{n,w} + C_{I,50\text{-}2500}$ = 33…37 dB[1)]
Ball drop: 2 kg medicine ball drop from 1 m height	Not registered on test equipment
Kettlebell drop: 35 kg kettlebell drop from 0,5 m height	L_{AFmax} = 34 dB[1)], L_{AE} = 42 dB[1)]
1) Result limited by background noise level.	

The spaces are in use and no adverse feedback has been received.

4.2 A gymnasium above educational use in a new building – Case B

Located in Helsinki, this educational building was completed in 2023. The gymnasium is located on the second floor and there are educational spaces on the first floor below. The acoustic design focused on achieving a low impact sound level with a simple floor structure.

The design for the floor structure build-up was as follows:

- ca. 11 mm resilient sports flooring
- 22 mm chipboard
- 15 mm flooring gypsum plasterboard
- 22 mm chipboard
- 150 mm flooring joists c/c 600 mm, with 25 mm elastomer bearings c/c 600
- 320 mm hollow core concrete slab
- ca. 900 mm air gap and ceiling suspension system
- 50 mm demountable mineral wool tile ceiling

Commissioning testing was conducted between the gymnasium and one of the teaching spaces below. The commissioning testing results are presented in Table 3 below.

Table 3: Commissioning test results for Case B.

Test type	Test result
Impact sound level	$L'_{nT,w} + C_{I,50\text{-}2500}$ = 27 dB[1)]
1) Result limited by background noise level.	

The spaces are in use and no adverse feedback has been received.

4.3 A gymnasium above educational use in an existing building – Case C

Located in Helsinki, this educational building, originally completed in 1954, underwent renovations during 2020 to 2021. The gymnasium is located on the first floor and there is educational and office use on the floor below.

The existing floor structure build-up, most recently renovated in 2006, was as follows:

- 77 mm timber sports flooring
- 3 x 15 mm flooring gypsum plasterboard
- ca. 360 mm concrete beams ca. c/c 1100 mm, with floor build-up between beams:
    - 22 mm timber battens 22x100 c/c 200 mm
    - 120 mm timber joists 120x45 c/c 400 mm, with mineral wool infill in cavity
    - 120 mm timber joist support attached to concrete beams, with mineral wool infill in cavity
    - ca. 50 mm lightweight aggregate infill
- ca. 50 mm reinforced concrete slab

- 70 mm timber battens 70x45 c/c 600 mm, with mineral wool infill in cavity
- 25 mm resilient acoustic channels c/c 400 mm
- 2 x 12,5 mm gypsum plasterboard

Baseline testing was performed between the gymnasium and one of the rooms below. The baseline testing results are presented in Table 4 below. It is notable, that the rattling noise from the ceiling was prominent.

Table 4: Baseline test results for Case C.

Test type	**Test result**
Impact sound level	$L'_{nT,w} + C_{I,50\text{-}2500} = 42$ dB[1)]
Ball drop: 2 kg medicine ball drop from 1 m height	$L_{AFmax} = 63$ dB
Basketball dribbling.	$L_{AFmax} = 50$ dB, $L_{Aeq} = 42$ dB
1) Result limited by background noise level.	

The gymnasium floor structure was not altered in the renovation, but the ceiling structure in the room below was disconnected from the underside of the concrete slab. Since the existing floor structure was not altered and there were room height limitations in the spaces below, the possible solutions were limited.

The constructed floor structure build-up at the time of commissioning testing was as follows:

- 77 mm timber sports flooring
- 3 x 15 mm flooring gypsum plasterboard
- ca. 360 mm concrete beams ca. c/c 1100 mm, with floor build-up between beams:
    - 22 mm timber battens 22x100 c/c 200 mm
    - 120 mm timber joists 120x45 c/c 400 mm, with mineral wool infill in cavity
    - 120 mm timber joist support attached to concrete beams, with mineral wool infill in cavity
    - ca. 50 mm lightweight aggregate infill
- ca. 50 mm reinforced concrete slab
- air gap
- 100 mm Z-profile steel joist, with 50 mm mineral wool infill in the cavity
- 2 x 13 mm gypsym plasterboard

Commissioning testing was conducted between the gymnasium and two of the rooms below. The commissioning testing results are presented in Table 5 below. It can be seen from the results that an improvement was achieved. The most significant improvement was tested for the 2 kg medicine ball drop, and it was the result of a significant reduction in secondary sound from rattling installations on the ceiling and on the walls.

Table 5: Commissioning test results for Case C.

Test type	**Test result**
Impact sound level	$L'_{nT,w} + C_{I,50\text{-}2500} = 40$ dB
Ball drop: 2 kg medicine ball drop from 1 m height	$L_{AFmax} = 43$
Kettlebell drop: 12 kg kettlebell drop from 0,7 m height	$L_{AFmax} = 51$ dB
Basketball dribbling.	$L_{AFmax} = 45$ dB, $L_{Aeq} = 40$ dB
Jogging with shoes, 2 people.	$L_{AFmax} = 38$ dB, $L_{Aeq} = 35$ dB
Jogging without shoes, 2 people	$L_{AFmax} = 44$ dB, $L_{Aeq} = 36$ dB
Jumping, 2 people.	$L_{AFmax} = 48$ dB, $L_{Aeq} = 39$ dB
1) Result limited by background noise level.	

The spaces are in use and some feedback about the gymnasium activity noise was received after commissioning. This shows that even though the statutory requirements are met, it does not always guarantee user satisfaction.

4.4 A gymnasium above workspaces in an existing building – Case D

Located in Helsinki, this building originally completed in 1929 is undergoing renovations in 2023-2024. The gymnasium is located on the top floor and there are workspaces on the floors below.

The existing floor structure build-up was as follows:

- ca. 58 mm timber floor planks 70x58
- Cardboard layer
- ca. 90 mm timber battens 100x70 and 100x20 attached to concrete beams, with old construction material infill in cavity.
- ca. 330 mm T-profile concrete beams 580/160x330 ca. c/c 1150 mm, with old construction material infill in cavity.
- ca. 40 mm reinforced concrete slab
- ca. 20 mm air gap and ceiling suspension system
- ca. 20 mm demountable mineral wool tile ceiling system

Baseline testing was performed between the gymnasium and the workspace below. The baseline testing results are presented in Table 6 below.

Table 6: Baseline test results for Case D.

Test type	**Test result**
Airborne sound insulation	$D_{nT,w}$ = 62 dB
Ball drop: 2 kg medicine ball drop from 1 m height	L_{AFmax} = 43 dB
Kettlebell impact: 12 kg kettlebell lay down by hand.	L_{AFmax} = 56 dB
Walking without shoes, 2 people.	L_{AFmax} = 34 dB
Jogging without shoes, 2 people	L_{AFmax} = 33 dB
Jumping, 2 people.	L_{AFmax} = 56 dB
1) Result limited by background noise level.	

In addition to controlled baseline testing, activity sound level logging was performed in the workspace below the gymnasium. Some results for the activity sound levels are presented in Table 7 below.

Table 7: Activity test results for Case D.

Activity type	**Test result**
Uneven bars gymnastics	L_{AFmax} = 30…36 dB
Jumps and landings on the floor	L_{AFmax} = 48…60 dB
Shifting or adjusting of gear on the floor	L_{AFmax} = 47…63 dB
Running on the floor	L_{Aeq} = 41…48 dB
Gymnastics session	$L_{AFmax,99}$ = 50…54 dB
1) Result limited by background noise level.	

The acoustic design for improving the floor structure impact sound insulation performance was based on preserving the existing floor structure and building a new floating floor on top of it. The existing load-bearing structure limited the amount of weight that could be added, and the height of the floating floor needed to be as low as possible.

Small scale mock-up testing was conducted for various proposed floating floor build-ups, which included elastomer bearings, recycled PU foam granulate sheets, steel springs and various board and rubber sheet layers. Based on the mock-up tests a developed design for a floating floor build-up using elastomer bearings was selected:

- ca. 23 mm timber flooring
- 18 mm plywood
- 12 mm granulated rubber mat
- 2 x 18 mm plywood
- 50 mm elastomer bearings 50x50 c/c 600 mm, with polyester fibre wool infill in cavity
- Existing floor structure

A small test construction of the designed floating floor structure was built, and controlled testing was conducted. The test construction test results are presented in Table 8 below. An improvement was achieved compared to the baseline. The most significant improvement was for the sharp knocking type impact of laying down a kettlebell by hand, which is expected from a floating floor type solution.

Table 8: Small test construction test results for Case D.

Test type	Test result
Ball drop: 2 kg medicine ball drop from 1 m height	L_{AFmax} = 36 dB
Ball drop: 4 kg medicine ball drop from 1 m height	L_{AFmax} = 42 dB
Kettlebell impact: 12 kg kettlebell lay down by hand.	L_{AFmax} = 31 dB
1) Result limited by background noise level.	

5 Conclusion

The case studies presented here briefly highlight some of the technical challenges of designing sufficient impact sound insulation for noise-sensitive spaces located below gymnasia. Renovation projects presented more difficult acoustic design scenarios than new buildings.

It is noted that for intermittent or impulsive noise sources, which is common for gymnasia, the standard impact sound level measurements may not be sufficient for evaluation. Maximum sound level measurements such as $L_{AFmax,T}$ would be advisable to be tested with the appropriate drop test methods or activities.

Further research into impact sound sources and their correspondence with sports and exercise activity impact sounds would be beneficial for easier comparison between cases. For example, transfer functions for different gymnasium activities to be used with the ISO standard tapping machine would be interesting. The standard rubber ball drop method and the medicine ball drop method used in some of the cases show agreement with low impact exercise. Intermediate impact exercise and ball games may require heavier impact sources, as is suggested by the comparison between the 12 kg kettlebell drop and basketball dribbling sound levels.

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ACOUSTIC DESIGN OF HOTELS: COMPARISON BETWEEN DIFFERENT HOTEL BRAND ACOUSTIC REQUIREMENTS

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All major hotel brands have their own design guidelines, including requirements of acoustics. Typically these requirements will include both sound isolation requirements, background noise requirements as well as requirements for facade sound isolation.

These requirements are typically based on American or European standards, which can present some challenges when designing hotels in locations which does not have acoustic regulations.

In the paper we will describe hotels designed to different brand standards and make a comparison between some of the standards and local regulations.

1 Introduction

During Soviet times, Georgia was one of the main holiday destinations. After the collapse of the USSR, the amount of tourist shapely declined. Since around 2010, tourism is again becoming an increasingly important component of the country's economy. Since 2020, the tourism industry accounts for more than 7% of the country’s GDP and in 2025, more than 10 million visitors are expected [1].

So the hotel industry in Georgia is rapidly developing, with brands such as Radisson, Hilton, Accor entering the Georgian market in force.

Most of these hotel brands have their own construction manual, usually including some degree of acoustic requirements. The acoustic requirements in the hotel brand manuals, will typically depend on the geographical origins of the hotel chain. In other words, the requirements are not necessarily in line with local requirements, and often requirements are presented in units not normally used in the area.

In this paper, we will explore acoustic requirement differences between the acoustic requirements of the different brands.

2 Sound isolation requirements.

The sound isolation requirements for the chains presented here, can be seen in Table 1. As can be seen, the requirements differ quite significantly. Also, even the basic 55 dB requirement are actually higher than legislative requirements between dwellings in both Georgia and other former CIS countries.

Sound insulation requirements of partition wall between guestroom and corridor depends on sound insulation requirement of the guestroom entrance door and should be 10dB higher than that of the door. This requirement varies between different hotel brands.

Table 1: Sound isolation requirements

Hotel Brands	Hilton	Accor	Radisson	Marriot (Le Meridien)	IHG (Indigo)
Between rooms	$R_w \geq 55$dB	$D_{nTw} + C \geq 51$dB	$R_w \geq 55$dB	STC ≥ 52	$D_{ntw}+C_{tr} \geq 43$dB
Door to corridor	R_w 38dB	R_w 42dB	R_w 38dB	-	R_w 29dB
Wall to corridor	R_w 48dB	R_w 52dB	R_w 48dB	-	R_w 39dB

The impact noise requirements are similar for all brands, L_{nw} 60 – 63 dB, however normally some floating construction is used in the hotel rooms (typically at least floating parquet) and wall to wall carpets are used in the corridors, meaning that the impact noise levels will typically be between 50 dB and 58 dB.

Not all brands have sound isolation requirements to other space, so typically we will suggest requirements depending on the layout of the spaces. The typical problems are the "Skybar" or the Gym on the top floor of the building. Also, many of the hotels have conference facilities with movable walls etc. In these cases, we will advise the client to realistic sound isolation requirements, and in many cases, help the client to find appropriate products (not all products are necessarily available in the market).

3 Room acoustic requirements

Typically, there are no requirements for room acoustic conditions in the guest room. Also, most brands do not have requirements for lobby spaces and similar, only for conference rooms.

However, we will typically set requirements also for more general spaces, so that for instance the speech communication around the check in counters are ensured and that it is possible to have a lobby bar without too much noise problems.

In order to achieve this, it is essential to have a good cooperation with both the building owner and the architectural designers, who in many cases never before has considered acoustics as a design parameter.

4 Background noise requirement

The typical problems concerning the technical equipment, will be the fan-coils. Efficient cooling of the rooms are necessary, however finding fan coil units with sufficient low noise levels and sufficient cooling capacity (and low price) can be a challenge. In particular, as some operators require that the noise levels should be fulfilled for the fan-coils running at full speed, which is a challenge.

For background noise in guestrooms coming from outside, different hotel brands have different requirements.

As well as calculation method for façade sound insulation. Background noise requirements and its calculation method is presented below. as example was taken budget hotels

The Day time period are from 07:00 to 23:00 and the Nighttime period from 23:00 to 07:00

Table 2: Background noise from outside requirements

Hotel Brand	Accor greet		Hilton garden inn		Radisson red	
	Daytime*	Nighttime*	Daytime*	Nighttime*	Daytime*	Nighttime*
Background noise requirements	$L_{A10} \leq 35$dB $L_{A1} \leq 40$dB	$L_{A10} \leq 30$dB $L_{A1} \leq 45$dB	$L_{Aeq} \leq 40$dB	$L_{Aeq} \leq 35$dB	$L_{A10} \leq 35$dB	$L_{A10} \leq 30$dB
Calculation method	L_{A10} and L_{A1} values should be calculated from noisiest 10-		L_{aeq} should be calculated from noisiest 2-hour period of day and night time		L_{A10} should be calculated from daytime and nighttime period	

	minute period of day and nighttime		

Both Accor and Radisson use the L_x method of noise evaluation where L_{A10} is the A-weighted, sound level, just exceeded for 10% of the measurement period and the L_{A1} the A-weighted, sound level, just exceeded for 1% of the measurement period, calculated by statistical analysis.

In the following, some examples for hotels are presented, based on actual measurements for hotel projects.

4.1 Accor traffic noise analysis

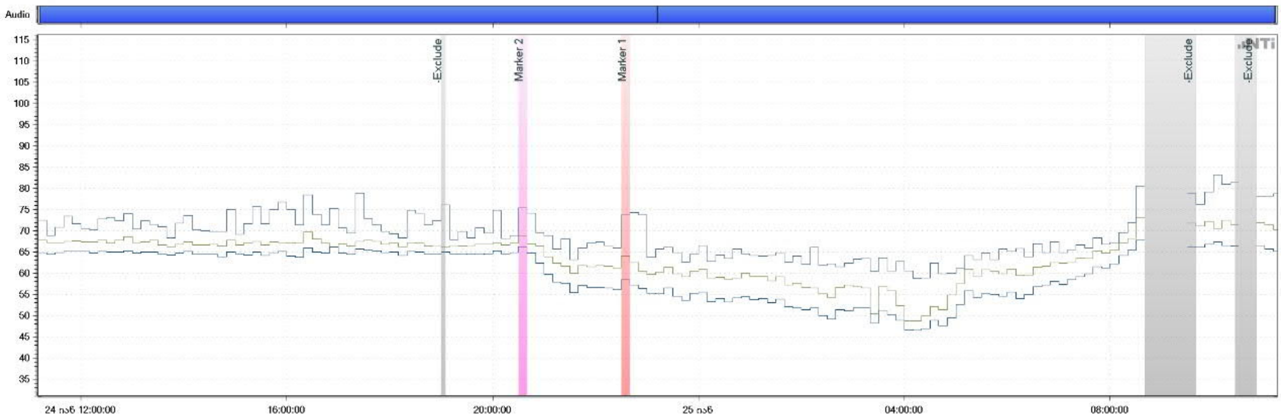

Figure 1: 24 Hour noise measurement on site for an Accor brand hotel

L_{A10} and L_{A1} from noisiest 10 min. period of daytime	L_{A10} and L_{A1} from noisiest 10 min. period night time
L_{A10} = 74dB, L_{A1}=75dB	L_{A10} = 64dB, L_{A1}=69dB

4.2 Hilton Traffic noise analysis

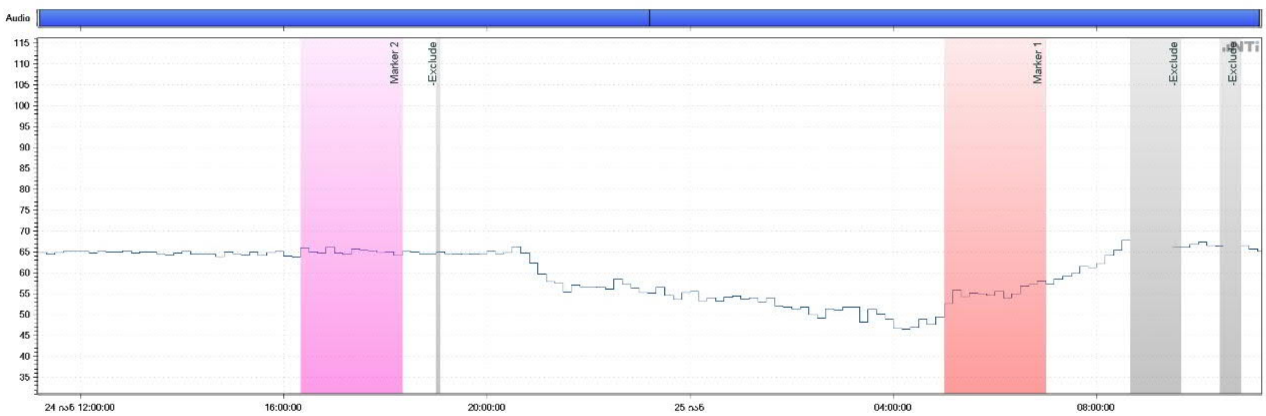

Figure 2: 24 Hour noise measurement on site for a Hilton brand hotel

L_{Aeq} calculated from noisiest 2 hour period of daytime	L_{Aeq} calculated from noisiest 2 hour period of nighttime
L_{Aeq}=65dB	L_{Aeq}=55dB

4.3 Radisson Traffic noise analysis

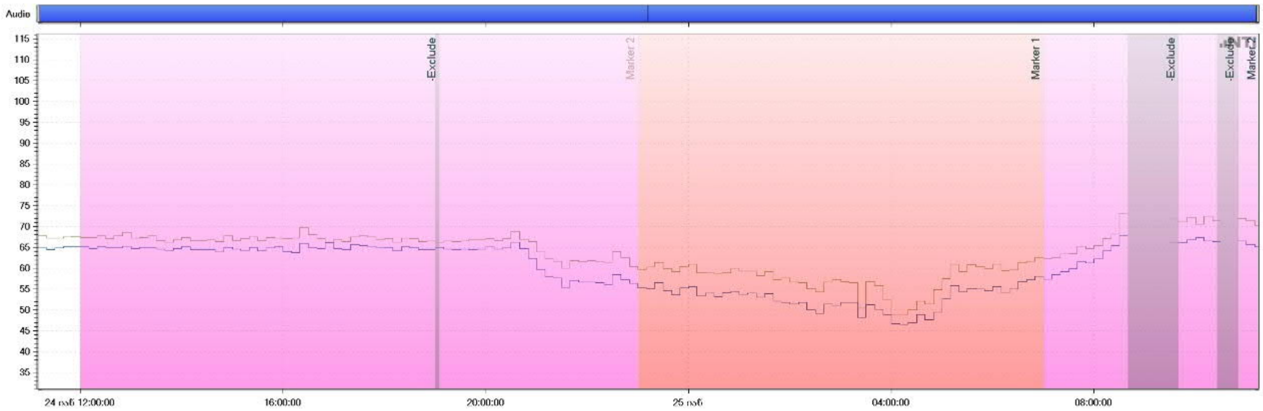

Figure 3: 24 Hour noise measurement on site for a Radisson brand hotel

L_{A10} calculated from period of daytime	L_{A10} calculated from period of nighttime
L_{A10} = 67dB,	L_{A10} = 59

For Façade sound insulation calculations, we are using Finish method, below is presented ΔL values for calculation and calculated requirement for façade.

For this example, the room parameters are: total room area 14m^2, façade wall 9.2m^2, window area 3.1m^2

Hotel Brand	Accor	Hilton	Radisson
ΔL values for calculation	34	30	32
Façade sound insulation requirement R_W+C_{tr}	38	34	36

5 Summary

The acoustic requirements for different hotel brands are mainly based on the regulations in their “Home countries”. This sometimes creates challenges when building in new (developing) regions with different or no acoustic requirements in te building code and with different building traditions.

However, we have found that by having the acoustic consultant involved from the beginning of the project, it is possible to both achieve the requirements of the brand standards and still doing the construction in a way that can be done by local construction companies.

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The end justifies the means – Sprayed sound-absorbing coating on non-acoustic materials

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ABSTRACT

There is a growing demand in the market for elegant but at the same time more sustainable acoustic solutions. To achieve good acoustic environments there are several ways to tackle the problem. Do we go to high-end luxury road, or could there be more affordable solutions with the same end results? And how to pull off the trick? In this study we present results from acoustical measurements taken in four different spaces of varying functionality where 10 mm thick acoustic coating has been installed directly on non-acoustical hard surfaces. The acoustic coating absorbs sound without the need of an acoustic base material underneath. When installed on existing surfaces, the carbon footprint of the building is reduced. The coating can be tinted to any color, and its texture can be customized from rough to smooth, thus architectural visual changes can be minimized. We demonstrate that the presented acoustic coating is a cost-efficient seamless acoustic solution that can be used to achieve excellent acoustic comfort, even though the thin layer sounds like insufficient. We demonstrate that coating can easily have larger surface area than traditional acoustic tiles, thus the required total absorption area is achieved with only 10 mm thick coating. The use of the acoustic coating, however, requires acoustic engineers, designers, and architects to rethink their acoustic plans with a more creative and sustainable mind.

1. INTRODUCTION

Traditional room acoustic solutions often involve suspended ceiling systems or glued sound-absorbing panels. In terms of aesthetics, these solutions might not be the preferred option. They may even be impractical or impossible to use in spaces where there are strict requirement to preserve the architectural visual design, such as in historical or architecturally protected buildings. Acoustic coatings offer an alternative solution to improve room acoustics while maintaining the visual appeal of spaces. These coatings can even seamlessly blend into the surfaces of spaces, sometimes becoming entirely unnoticeable.

Most of the acoustic coatings available on the market are non-sound absorbing materials. Instead, they are applied on top of acoustical panels. In this system, the function of the acoustic

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coating is hiding the seams between the underlying acoustic panels, while the acoustic panels provide the sound absorption properties. A well-known challenge in the use of non-sound absorbing coatings is that the thickness of the coating is critical to reach the full functionality of the acoustic solution. Too thick layer of coating will degrade the acoustical properties of the underlying acoustic panels and, consequently, the acoustical properties of the system. Too thin layer of coating will fail too entirely hide the seams between acoustic panels, thereby deteriorating the aesthetics of the solution. Generally, the critical thickness of non-sound absorbing coatings is 3-4 mm.

On the other hand, sound absorbing coatings offer several advantages compared to non-absorbing coatings. In first place, the thickness of the coating is not critical to achieve the full functionality of the acoustic solution. It must be thick enough to fully hide the seams between the underlying panels. In most cases, the minimum thickness required for this purpose is 5 mm. Too thick layer of coating will not degrade the acoustic properties of the underlying panels as sound waves will always go through the porous structure of the sound absorbing coating. Secondly, sound absorbing coatings can be utilized without the needs of employing underlying acoustical panels. They can be installed directly on non-acoustical hard surfaces and improve room acoustics. Their main constraint, however, is the limitation of their acoustic properties mainly due to their relatively small thickness. Nevertheless, the coating can easily have larger surface area than traditional acoustic tiles, thus the required total absorption area can be easily achieved by increasing the total surface area covered by the coating. Moreover, it can substitute other building materials like fillers and paints, providing sound absorption properties to surfaces primarily intended only for visual purposes.

In this study, we present four cases where biobased sound-absorbing acoustic coatings, installed directly on non-acoustic surfaces, have been used to improve the acoustic environment. The acoustically treated premises are an open-plan office space, a five-floor staircase, a restaurant, and a spa. High visual requirements were set in all of the cases for the sound absorbing structures. In practice, the acoustic treatment had to be unnoticeable. The acoustic of all the premises were evaluated via acoustical measurement of reverberation time, T_{30}, and speech clarity, C_{50}, taken before and after installation of the acoustic coating. The biobased sound absorbing coating used in all the premises is carbon negative. Thus, since the coating has been installed directly on existing surfaces, the carbon footprint of all the buildings has been reduced after installation of the coating.

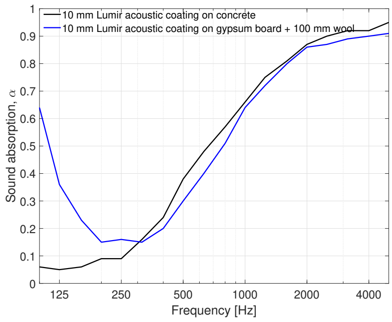
2. MATERIALS AND METHODS

2.1. Materials

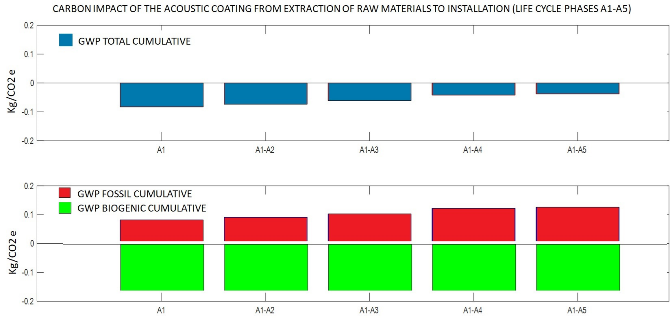
A sprayable biofibre-based sound absorbing coating manufactured by Lumir [1] was used as the acoustic solution for the acoustic design of all the premises presented in this study. In all the spaces, the coating was directly sprayed on existing non-acoustical hard surfaces. The thickness of the coating is 10 mm, the color and surface of the coating was tailored to meet client´s specifications, which generally requires minimizing visual changes to the architectural design of the premises. The coating can be sprayed almost on any surface. In most cases, installation of the coating involves first applying primer to the underlying surfaces to enhance adhesion of the coating. When sprayed on acoustical surfaces, such as mineral wool or perforated gypsum, the sound absorption coefficients of the acoustic solution improve at low and mid frequencies, leading to A-C sound absorption class depending on the underlying acoustical structure. The sound absorption coefficients of the 10 mm coating sprayed on two different non-acoustical hard surfaces are presented in Figure 1a:

- 10 mm biofibre-based acoustic coating sprayed on concrete. Sound absorption class D, α_w = 0.35 (MH).
- 10 mm biofibre-based acoustic coating sprayed on plain gypsum board with 100 mm mineral wool behind. Sound absorption class D, α_w = 0.35 (MH).

In addition to the acoustic properties, the biofibre-based sound absorbing coating acts as a carbon sink



(a) Sound absorption coefficients of biobased acoustic coating.



(b) Carbon impact of acoustic coating.

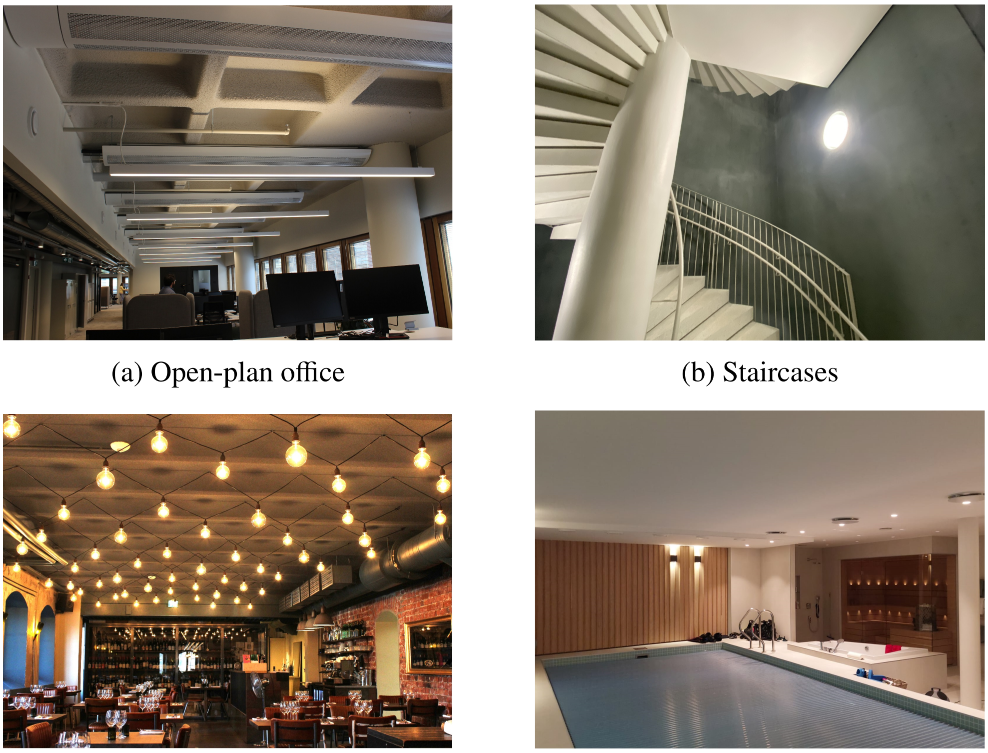
Figure 1: (a) Sound absorption coefficients of biofibre-based acoustic coating sprayed on different non-acoustic surface measured according to standard ISO 354 by an accredited laboratory. (b) Carbon impact of the biofibre-based acoustic coating according to results from environmental product declaration (EPD) report, including life cycle phases A1-A5, from extraction of raw materials to coating installation. The EPD has been conducted in accordance with EN 15804+A2 and ISO 14025.

during its operational life as it comprises approximately 80 weight percent (wt%) cellulosic fibres as its primary raw material. Under normal conditions, the operational life of the coating extends to several decades. Cellulose, the main structural component of natural fibers, consist about 49 wt% of carbon. Plants acquire this percent of carbon mainly as carbon dioxide (CO_2) from the atmosphere during their growth phase. The carbon dioxide is then processed into cellulose and other components via biosynthesis, with oxygen resulting as a side product. As a rule of thumb, carbon bound in 1 kg of cellulose, often referred as biogenic carbon, corresponds to roughly 1.5 kg of atmospheric carbon dioxide [2]. It can be estimated that the biofibres incorporated into the coating (80 wt%) capture approximately 1.2 kg of atmospheric CO_2/m^2.

According to Life Cycle Analysis (LCA), the coating is carbon-negative from life-cycle phase A1 to A5, including acquisition of raw materials, coating production and installation, see Figure 1b. Thus, the coating stores into its structure more carbon dioxide over its operational life than is released during its manufacturing and installation processes. The carbon footprint of building materials accounts for the carbon sequestration (biogenic carbon) and emissions associated to the building materials over their lifecycle. The carbon footprint of the biobased sound-absorbing coating is -0.038 kg CO_{2e} per kg of product, which, based on the density and thickness of the coating, implies that installation of the acoustic coating on existing surfaces results in a reduction of the carbon footprint of the building of around 46 g CO_{2e}/m^2.

2.2. Methodology

Four premises –an office, a staircase space, a restaurant, and a spa– have been acoustically treated with the use of 10 mm acoustic coating applied on non-acoustical hard surfaces. Assessment of acoustic parameters, including reverberation time (T_{30}) and speech clarity (C_{50}), was conducted in accordance with ISO 3382-1 (2009) both before and after implementing acoustic treatment (except for the open-plan office, where measurements where taken only after installation of acoustic coating). All the measurements were taken under unoccupied conditions. The software ARTA [3] was utilized to capture impulse responses via the inverse swept-sine technique [4]. The sound was emitted from an omnidirectional sound source, model LS02, and an omnidirectional 1/4-inch measurement microphone (Superlux ECM-999) was employed for recording. Moreover, the presented results represent averages from multiple measurements, with a minimum of two sound source and two receiver positions. Results on the effect of the acoustic coating on the carbon footprint of the acoustically treated spaces is also provided based on data from LCA.



(a) Open-plan office
(b) Staircases
(c) Restaurant
(d) Spa space

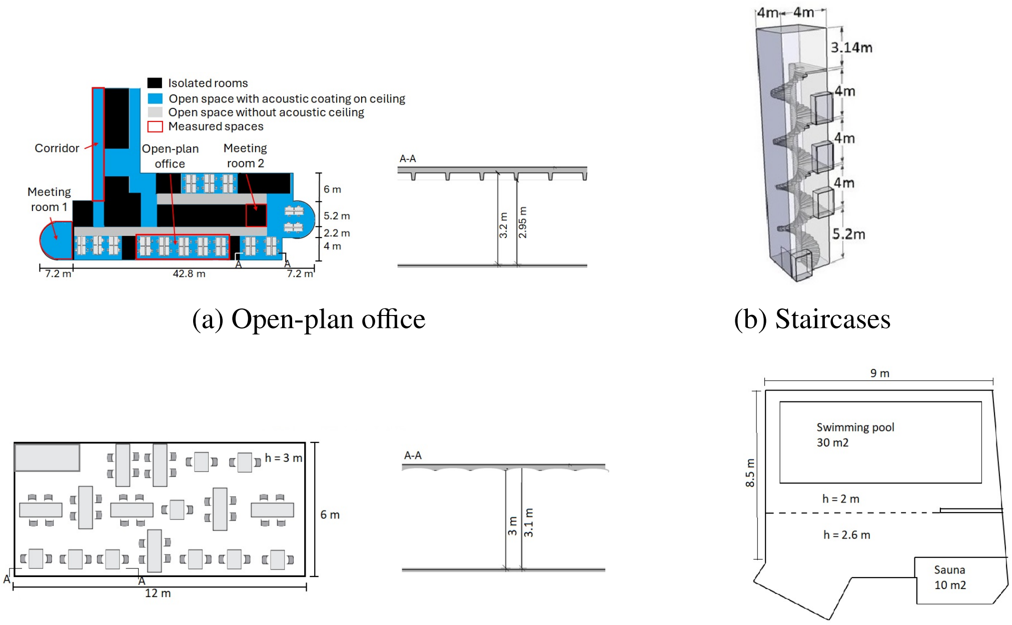
Figure 2: Images of the four spaces acoustically treated with 10 mm biobased acoustic coating sprayed on non-acoustical surfaces.

Figure 2 illustrates the four premises treated with the 10 mm acoustic coating. Layout and sections are shown in Figure 3. There were strict requirements for the acoustic treatment to avoid changes in the appearance of all the premises, taking into account the color and the structure of the surfaces. All the premises are briefly described below.

Office: the office was located in Porkkalankatu 3, Helsinki. The ceiling of the office was concrete vaulted slab, the structure can be seen in Figure 2a. The whole surface of the concrete vaulted slab on the ceiling was primed and sprayed with the 10 mm acoustic coating, including vertical and horizontal surfaces. Each of the concrete vaulted slab units added a total of 1.2 m^2 of acoustic coating on their vertical surfaces. Acoustical measurements were taken in a corridor, two meeting rooms, and an open-plan office. All the measured spaces had a thin sound absorbing carpet, upholstered chairs, and some of the workstations were equipped with 1.4 m high sound absorbing screens.

Staircases: the five-floor spiral staircase is located in the School of Business building at Aalto University. The acoustic design of the staircases is part of the artwork, Mare Tranquillitatis, by the artist group IC-98. The artwork aimed to create a zone of complete silence that serves as a place of tranquillity and confrontational encounter, as described by the authors. All the walls, the ground floor, and the ceiling are painted concrete. The stairs and landings were mosaic concrete, and the underneath of the stair-landing was plywood with an air cavity behind it. A concrete pile of 0.5 m of diameter was stranded in the middle of the spiral staircase from the ground floor up to a height of 18.5 m. All the walls and ceiling were treated with the 10 mm biobased acoustic coating directly on the primed concrete surface. The plywood in the underneath of stair-landings was exchanged with perforated gypsum boards sprayed with the 10 mm acoustic coating.

Restaurant: the space is located in Tehtaankatu 27-29, Helsinki. All the walls were brick surfaces, there was a thin carpet on the floor, and the ceiling was a vaulted brick structure as shown in Figure 2c. The vaulted brick ceiling was primed and acoustically treated with the 10 mm biobased acoustic coating.



(a) Open-plan office

(b) Staircases

(c) Restaurant

(d) Spa

Figure 3: Layout and sections of the four spaces acoustically treated.

Spa: the spa space is part of a private house located in Helsinki. The spa has a 30 m^2 swimming pool, sauna and a space for showers. All the walls were hard surfaces. Wooden slat surface was found on one of the walls. The floor had tiles and the ceiling was suspended plain gypsum. The 10 mm acoustic coating was sprayed directly on the plain gypsum ceiling after spraying a layer of primer.

3. RESULTS AND DISCUSSION

Table 1 presents a summary of room volumes, floor surface area, total surface areas on ceilings and walls covered with acoustic coating, average reverberation time before and after installation of acoustic coating, as well as the impact of the acoustic coating on the carbon footprint of the building after installation.

3.1. Room acoustics

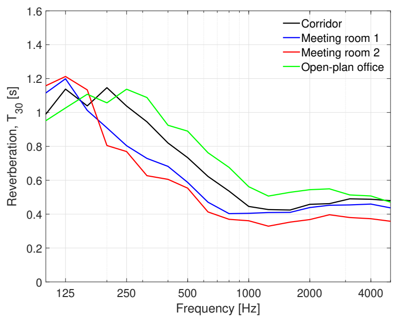
Figure 4 illustrates results from the acoustical measurements taken in the different premises presented in this study. It can be seen that the installed acoustic coating has led to significant improvement in reverberation time and speech clarity. Reverberation time has been decreased in all the spaces according to the sound absorption properties of the acoustic coating. The most significant reduction of reverberation time happens at frequencies above 500 Hz. However, the coating has also improved the room acoustics of all spaces at frequencies below 500 Hz. The influence of the acoustic coating on the mid and low frequencies depends on the acoustics of the space before installation as well as on the total surface area covered by the coating. Interestingly, reverberation time measurements in the spa, after installation of the acoustic coating, presents a strong flutter echo at around 3 kHz. In this space, furniture was minimal and sound absorbing surfaces were installed only on the ceiling, thus all the vertical surfaces were left to reflect sound. The flutter echo may arise between two walls, most probably between the walls in the shower area.

One of the main advantages of the acoustic coating compared to acoustic panels is that it can be installed on any surface without affecting the architectural design of the spaces. In addition, the visual outlook of the space is untouched, as the coating can be colored and the roughness of the

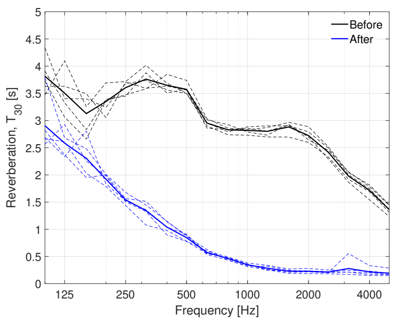
Table 1: summary of room volumes, floor surface area, total surface areas on ceilings and walls covered with acoustic coating, average reverberation time before and after installation of acoustic coating, as well as the impact of the acoustic coating on the carbon footprint of the building after installation.

	Volume [m^3]	**Floor area[a] [m^2]**	**Acoustic coating on ceiling [m^2]**	**Acoustic coating on walls [m^2]**	**T_{30} before[b] [s]**	**T_{30} after[b] [s]**	**Bound atmospheric carbon on coating[c] [kg CO_{2e}]**	**Carbon footprint of installed coating[d] [kg CO_{2e}]**
Open-plan office	377	74	120	0		0.7	210	-8.1
Corridor	95	30.1	48	0		0.6	58	-2.2
Meeting room 1	139	45	66	0		0.5	79	-3
Meeting room 2	81	26	44	0		0.5	53	-2
Staircases	331	-	63	188	2.8	0.7	301	-11.5
Restaurant	216	72	90	0	0.7	0.5	108	-4.1
Spa	190	80	80	0	2	1.1	96	-3.7

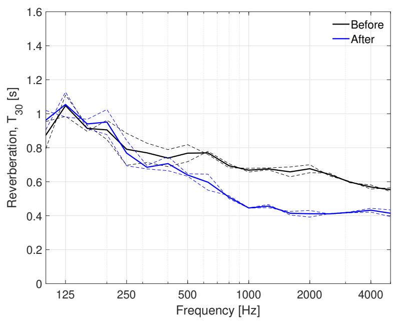
[a] Only the floor area below the ceiling areas covered with acoustic coating has been considered.
[b] Arithmetic average reverberation time across the third-octave-bands from 250 Hz to 4000 Hz.
[c] Biogenic carbon, the carbon content stored in the structure of the coating mainly due to the organic raw materials used for its production.
[d] Includes biogenic carbon and carbon emissions produced during the extraction of raw materials, coating manufacture, transportation and installation.



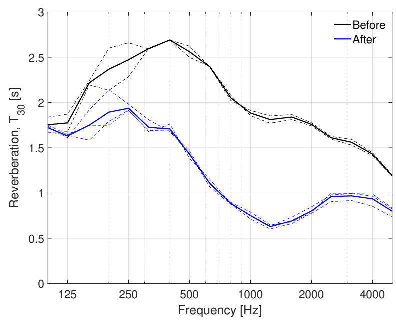
(a) Office, T_{30}



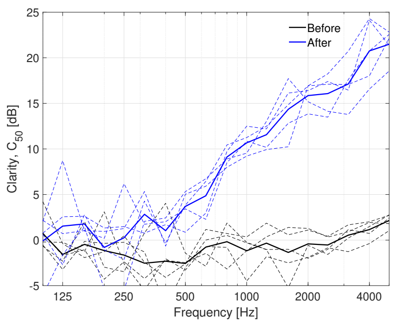
(b) Staircases, T_{30}



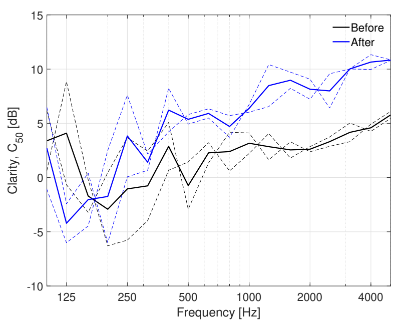
(c) Restaurant, T_{30}



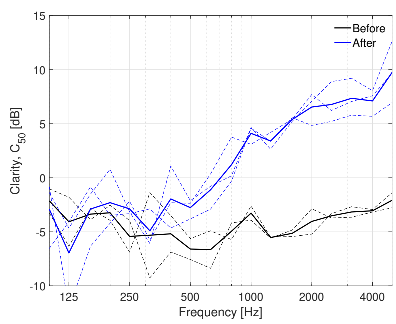
(d) Spa, T_{30}



(e) Staircases, C_{50}



(f) Restaurant, C_{50}



(g) Spa, C_{50}

Figure 4: Reverberation time, T_{30}, and speech clarity, C_{50} measured in the acoustically treated premises after installation of biobased acoustic coating on non-acoustical hard surfaces.

finished surface adjusted. Therefore, the coating can easily cover larger surface areas than traditional acoustic tiles. Despite the coating having poorer absorption at frequencies lower than 500 Hz, by increasing the total surface area of the absorptive material, it can achieve the same absorption area as more efficient absorbers.

For example, in the open-plan office, the ceiling area covered with the acoustic coating is 120 m^2. The maximum ceiling area that could be covered using acoustic tiles would be 74 m^2, as acoustic tiles could be installed only on horizontal surfaces of the concrete vaulted slabs to preserve the aesthetics of the ceiling. Figure 5 illustrates the total absorption area achieved with the 10 mm acoustic coating versus the total absorption area that could be achieved using A-, B-, or C-class acoustic tiles. It can be seen that at 500 Hz, the total absorption area achieved by the acoustic coating is equal to that achieved by the acoustic solution based on C-class acoustic tiles. Above 700 Hz, the total absorption area is considerable greater for the acoustic coating compared to the acoustic tiles-based solutions.

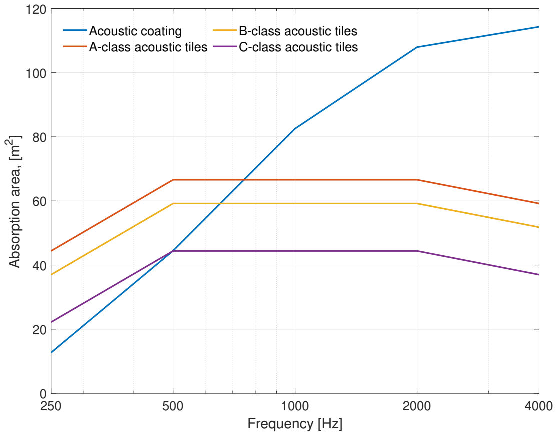


Figure 5: Total absorption area calculated for 10 mm biobased sound absorbing coating installed on 120 m^2 versus absorption area for acoustic tiles installed on 74 m^2.

Speech has been reported as the most disturbing sound source in open-plan offices [5]. In such spaces, it is beneficial to maximise the area of sound absorbing surfaces to avoid spreading of speech noise between workstations. Thus, maximising sound absorbing area for frequencies above 500 Hz would reduce the radius of distraction, and more importantly, it would decrease speech intelligibility between workstations as speech intelligibility would be high only at very short distances. This is not usually a concern as employees in an open-plan office are generally distributed in teams working in common subjects. If additional absorption area is required at mid and low frequencies, it could be easily increased by spraying the acoustic coating on other surfaces such as on the walls. Due to the toughness of the coating, it is suitable to be installed on walls. This addition of acoustic coating on walls would not alter the visual aesthetic of the space while it significantly aid in preventing the propagation of noise.

Expanding the application area of the acoustic coating may raise concerns about the costs associated with the acoustic treatment. However, among seamless acoustic solutions utilizing acoustic coatings, spraying a 10 mm acoustic coating on existing surfaces emerges as the most cost-effective option as it eliminates material costs associated with underlying acoustic materials. In comparison, the material costs for 1 m^2 of a seamless acoustic solution involving acoustic coating sprayed on top of acoustic underlying material equal those of 1.5 m^2 of acoustic coating directly sprayed on existing surfaces. Additionally, installing the coating without using underlying acoustic materials significantly reduces installation time, resulting in lower overall installation costs. Moreover, the use of tinted acoustic coating could substitute the use of other building materials such as fillers and paints, thus leading to further savings.

3.2. Environmental viewpoint

The carbon impact of the acoustic coating on all the spaces is reported in Table 1. The biobased acoustic coating installed in all the spaces bound approximately 1.2 kg CO_{2e}/m^2. The carbon footprint of each of the premises, including biogenic and carbon emissions, was reduced by 48 g of CO_{2e}/m^2. In individual spaces, the decrease of the carbon footprint by a biofibre-based acoustic coating might seem insignificant, especially in the presented case studies, as the rooms are quite small. However, the carbon impact of the coating is much more significant when one considers bigger buildings, as well as all the building materials that can be omitted due to the installation of the acoustic coating, such as paints and fillers. Furthermore, compared to other traditional acoustic solutions, such as glass wool or perforated gypsum, the use of glass wool would increase the carbon footprint of the building by 3.1 kg of CO_{2e}/kg, whereas the use of perforated gypsum would increase the carbon footprint of the building by 1.9 kg of CO_{2e}/kg [2]. For example, taken the open-plan office case, installing 74 m^2 of 20 mm glass wool of density 90 kg/m^3 would increase the carbon footprint of the building by 133.2 kg of CO_{2e}. This value, compared to the reduction in the carbon footprint of the building achieved with the installation of the biobased sound absorbing acoustic coating on 120 m^2, -8.1 CO_{2e}, demonstrates the capability of the acoustic coating as one of the most sustainable seamless acoustic solution.

4. CONCLUSIONS

This study presents the use of a biobased sound absorbing coating, directly sprayed to existing non-acoustic surfaces, as the primary measure in the acoustic treatment of any space where the main objective of the acoustic design is to mitigate propagation of speech noise, such as shopping malls, large reception areas, corridors, restaurants, and open-plan offices. The 10 mm acoustic coating has its own limitations, especially at frequencies below 500 Hz. At such frequencies, if greater absorption is needed, acoustic designers count with several other tools to absorb sound, such as acoustic screens, carpets, or some sound-diffusing elements such as shelves. On the other hand, the acoustic coating permits the treatment of larger surface areas, thereby augmenting the total absorption area below 500 Hz. This, in turn, leads to an enhanced absorption at a wider frequency range. Moreover, the acoustic coating could substitute other building materials like fillers and paints, providing sound absorption properties to surfaces primarily intended only for visual purposes.

The presented case studies proved that acoustical treatment can be done while respecting the architectural visual aesthetics and even reducing the carbon footprint of the buildings. The increase use of such carbon-negative building materials is indispensable towards an economy with net-zero greenhouse gas emissions, where buildings will reverse their role in the fight against climate change.

ACKNOWLEDGEMENTS

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Structure-borne noise emitted by building service equipment: laboratory measurements and modelling

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Effective control of structure-borne noise is increasingly important as modern buildings often contain numerous building service equipment that can generate significant structure-borne noise. Suitable and sufficiently accurate computational methods are often limitedly available. Novel modelling methods for structure-borne noise have been developed, based on initial data obtained from laboratory measurements of equipment structure-borne noise emissions. Using initial values, such as the equipment's blocked force levels, simplifies the modelling process and results in more accurate predictions of the equipment's sound in situ. Similarly, the newly published standard EN 12354-5:2023 is based on a computational technique similar to that used in this research. Furthermore, the parameters defined in the new version of the waste water installations measurement standard, EN 14366-1:2023, are better suited to computational modelling needs. Noise levels measured *in situ* have been found to correlate well with values modelled according to these new methods, with a possible calculation accuracy of ± 3–4 dB.

1 Introduction

Modern buildings can have multiple sources of structure-borne noise. In residential buildings, for instance, structure-borne noise may arise not only from rotating devices such as fans, pumps, or compressors but also from wate water installations or water appliances. Currently, equipment manufacturers rarely provide sufficient initial values necessary for calculating structure-borne noise. While data on noise from waste water installations and water fixtures are more readily available, there is a significant lack of such information for other devices such as heat pumps or fans. On the other hand, computational methods for determining these initial values are also extremely limited. Values based on measurements, however, can be very effectively utilized.

Zenner Engineers has conducted extensive research on utilizing laboratory measurements of various structure-borne noise excitations in modelling. Specifically, the focus has been on the structure-borne noise emitted by waste water systems and water appliances. The goal is to gather broader information and create a database on other noise sources as well, such as pumps.

It is crucial to encourage industry participants to report structure-borne noise excitation levels in addition to the current airborne sound power levels. The latest versions of standards such as EN 12354-5:2023, EN 15657:2017 and EN 14366-1:2023 [1-3] align well with this concept, as the calculations are based on initial values obtained through measurements.

The utilization of a standardized and well-known impact sound source device facilitates modelling in many cases, enhancing the accuracy and applicability of the results.

In the case of heavy structures with low mobility, good computational results are typically achieved. However, modelling as well as the initial measurements is more challenging with lightweight structures. Typically, heavier pipe systems and vibrating machines or devices are mounted to massive structures to mitigate this issue.

2 Modelling principles and laboratory measurements

2.1 Modelling principles

Standard EN 12354-5 [1] outlines general methods for calculating the sound pressure levels produces by service equipment installed *in situ* within buildings. The primary input for structure-borne noise calculations is the installed power level of the source, $L_{\mathrm{Ws},i}$, which can be derived from either the equipment single equivalent free velocity level, $L_{v\mathrm{f,eq}}$, or the equipment single equivalent blocked force level $L_{F\mathrm{b,eq}}$. This research primarily focuses on the latter.

In typical cases where the mobility of the receiver is significantly lower than that of the source, such as when service equipment is attached to a heavy structure, Standard EN 12354-5 provides a simple calculation model for computing structure-borne noise. This model is based solely on a single source characteristic, $L_{F\mathrm{b,eq}}$, and the well-known characteristics of the ISO standard tapping machine. The calculation formula for the apparent structure-borne sound pressure levels $L'_{\mathrm{ne,s}}$ generated *in situ* by any equipment connected to a heavy wall or floor is given by [1]:

$$L'_{\mathrm{ne,s},i} = L'_{\mathrm{n},i} + L_{F\mathrm{b,eq}} - L_{F\mathrm{b,eq,stm}} \tag{1}$$

In Equation (1), $L'_{\mathrm{n},i}$ represents the apparent impact sound pressure level of element *i,* which can either be measured on-site or calculated using methods given in Standard EN ISO 12354-2 [4]. The only source-dependent characteristic, $L_{F\mathrm{b,eq}}$, required for the calculation model, can be determined through laboratory measurements as specified in standards such as EN 15657 or, for waste water systems, in EN 14366-1. The final parameter in the equation, the single equivalent blocked force level of the ISO standard tapping machine, $L_{F\mathrm{b,eq,stm}}$, can easily be calculated from literature [5].

If blocked force levels were provided by manufactures or measured in the laboratory, the structure-borne noise levels could be very easily determined for any location. The objective is to promote the widespread adoption of reporting this characteristic and to establish it as common practice for manufacturers to include this value in the technical specifications of their devices. It is also particularly important that the determination of input data for the calculation defines the excitation as a whole, including pipe clamps, device support frames and similar components.

It should also be noted that Standard EN 15657 also provides calculation formulas to calculate the installed structure-borne power of any source under various source-receiver mobility conditions, accommodating both lightweight and massive building elements. However, modelling, as well as the initial measurements, are more challenging with lightweight structures and are not the focus of this research.

2.2 Laboratory measurements

Zenner Engineers has conducted extensive research into the mechanisms of waste water noise generation. As part of these studies, both airborne and structure-borne noise levels generated by various waste water products and different installation methods were measured. The measurement system, implemented with some modifications, complies with Standard 14366-1. In this system, a water flow rate of 0–8 dm^3/s is generated within the pipeline traversing the test rooms using a compressed air system. The total length of the sewer pipeline is 10 meters, approximately 6.5 meters of which is vertical, and is supported by a massive test wall ($m' = 400\ \mathrm{kg/m}^2$) that separates the source and receiving rooms. Airborne noise emitted by the sewer pipe is measured in the source room, while the structure-borne noise is simultaneously measured in the receiving room.

The characteristics of the test wall have been precisely defined through measurements. Given that the mobility and structural sensitivity of the test wall are accurately known, the measurement system is capable of providing highly reliable initial values essential for the calculation of structure-borne noise, as discussed in the previous chapter.

The system facilitates highly flexible investigations of different pipe materials, components, support methods, and enclosure or insulation solutions. These measurements have fled to the creation of extensive databases for various products, significantly benefiting acoustic design. The conducted studies have also resulted in the production of one diploma thesis [6].

2.3 Examples of laboratory measurements

Figure 1 presents two examples blocked force levels, $L_{F\mathrm{b,eq}}$, measured in the laboratory for a typical heavy plastic pipe attached to a wall with two types of clamps: rigid and vibration-isolated. For comparison, the figure also includes the standard blocked force levels, $L_{F\mathrm{b,eq,stm}}$, for an ISO tapping machine (in dB re 1 µN).

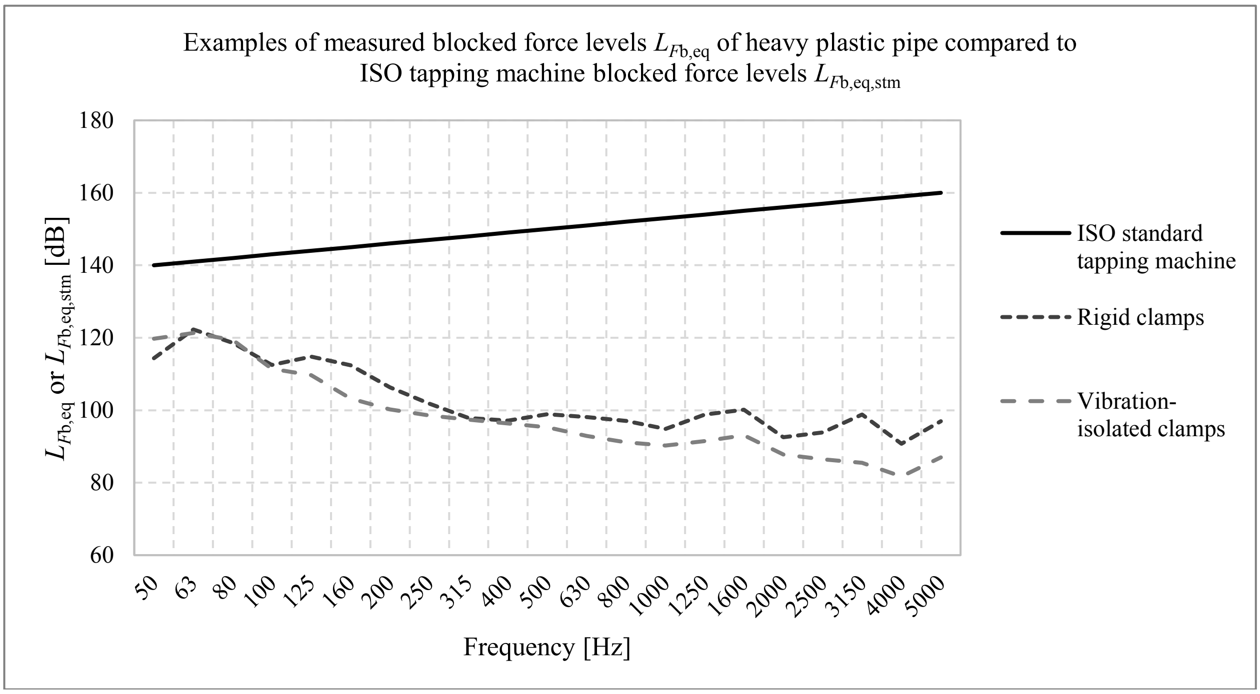

Figure 1: Examples of measured blocked force levels $L_{F\mathrm{b,eq}}$ of heavy plastic pipe in comparison with ISO standard tapping machine blocked force levels $L_{F\mathrm{b,eq,stm}}$.

The results show that the blocked force levels measured with vibration-isolated clamps are approximately 5–10 dB lower than those measured with rigid clamps. This directly leads to a lower emitted structure-borne noise when using isolated clamps. It is important to note that clamp products of different types and from different manufacturers can vary significantly. Therefore, the vibration isolation properties must be measured separately for each product type.

3 Comparison to field measurements

The research included also actual noise measurements at the installed site to validate the accuracy of the calculation model. Figure 2 presents an example of calculated and on-site-measured structure-borne waste water noise when the sewer system is attached to two different types of walls. In the example, the sewer pipes, made of heavy plastic material, were supported by typical vibration-isolated clamps. During the measurements, water was supplied into the pipe system at a constant flow rate of 4 dm³/s.

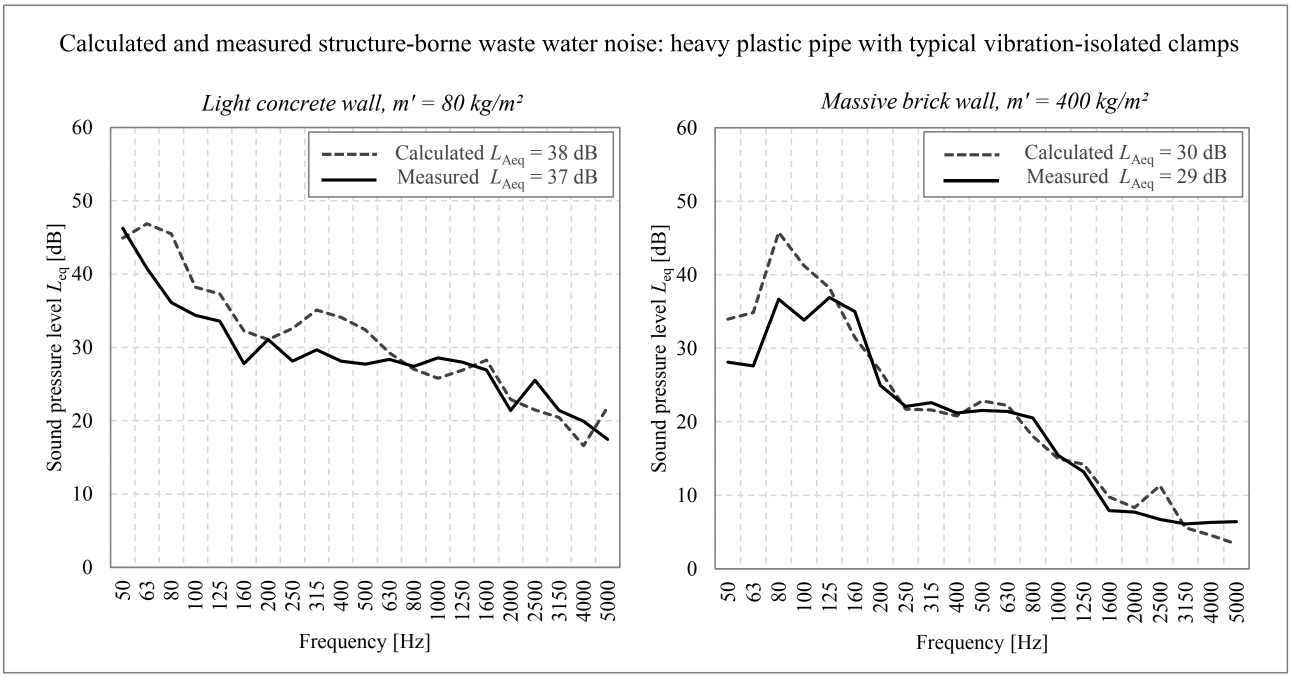

Figure 2: Examples of calculated and field-measured structure-borne waste water noise of heavy plastic pipe attached to two different walls with typical vibration-isolated clamps. Flow rate 4 dm³/s.

The results demonstrate a relatively good correlation between the predicted and measured structure-borne noise levels. For instance, the model predicted a structure-borne noise levels of 38 dB and 30 dB, while the measured levels were 37 dB and 29 dB, respectively, confirming that the model's precision can be within a 3–4 dB margin of error. This example illustrates the practical application of acoustical modelling in assessing and managing structure-borne waste water noise in building environments.

4 Summary and conclusions

Accurate modelling of the structure-borne noise caused by building service equipment necessitates the reliable determination of initial values through laboratory measurements. With these initial values, it becomes feasible to accurately calculate the structure-borne noise emissions of devices in various situations and locations.

In this study, the accuracy of the calculation model proved to be robust, particularly in scenarios involving short transmission paths and relatively massive structures. The reliability of the calculations was validated through field measurements.

The long-term goal is to promote the widespread adoption of measured initial data values for equipment structure-borne noise emissions and to establish it as common practice for manufacturers to include these values in the technical specifications of their devices.

The accuracy of modelling deteriorates in the case of lightweight structures and in situations where the number of transmission paths or joints increases. Achieving reliable modelling in these situations remains an area requiring further research.

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Harmonizing sustainability & acoustics: challenges in mass timber construction

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Sustainable buildings have been in rising demand for a few years, leading to increased use of wood construction due to their sustainable character. Following this trend, mass timber solutions, including cross-laminated timber (CLT), can be an excellent substitute for more traditional, stiffer and heavier building materials, such as concrete and steel, when some of their inherent properties aren't required. However, while CLT has many advantages as a sustainable building material, it can also pose some rather unique challenges regarding acoustics. Due to its orthotropic character and low mass density, the material is less effective acoustically, resulting in increased direct sound transmission and flanking transmission paths. CDM Stravitec has developed and tested floor-ceiling solutions to reduce direct sound transmission. To reduce flanking sound transmission, elastic decoupling materials can be installed, and complementary anchors with acoustic decoupling features need to be installed to maintain structural stability. Acoustical and structural integrity tests were performed on the developed flanking sound isolation solutions. CLT panels are also an ideal building material for 3D modular constructions. However, transferring vertical and horizontal forces between stacked modules can compromise acoustic requirements. CDM Stravitec has developed solutions using pre-compressed elastomer decoupling techniques to transfer forces without compromising sound insulation. Structural integrity tests on the couplers and in-situ measurements of airborne sound isolation have shown that good coupling design can achieve the correct load transfer and high sound insulation between the modular structures. This paper presents the results of the above-mentioned test campaigns and the main findings.

1 Introduction

Mass timber solutions, including cross-laminated timber (CLT), have many advantages as sustainable building materials, but they can also pose some rather unique challenges regarding acoustics.

CDM Stravitec has developed solutions to increase airborne and impact sound isolation for CLT structural floors, elastically decouple building parts, and stack modular construction without compromising the acoustic decoupling of the modules.

Section 2 will describe the test campaign executed in the Buildwise, Belgium laboratory to define the airborne and impact sound isolation of various floating floors installed on CLT structural slabs. Section 3 describes an in-situ experiment to define the efficiency of a resilient polyurethane strip in combination with an angle bracket with acoustic decoupling features. Section 4 discusses the influence of angle brackets with acoustic decoupling features on the sound transmission between 2 modular units.

2 Airborne and impact sound isolation of CLT structural floors

2.1 Test setup

Tests A to P were carried out on a bare 5-layer cross-laminated timber (CLT) slab 180 mm thick (BS1). Tests Q, R, and S were carried out on a bare 5-layer cross-laminated timber (CLT) slab 140 mm thick (BS2). Both were over 260 cm x 442 cm. Each test element was mounted according to the NBN EN ISO 10140-3 standard in a similar manner to the actual construction, and tests were carried out on each system described in this paper.

2.2 Test results

Table 1 describes the different tested setups and an overview of the global ratings for all setups. The presence of a dropped ceiling is shown together with the elastic supports and board materials of the tested assembly. All tests are performed on dry, panelized, floating floor systems except setup G, in which a gypsum topping of 50 mm thickness is installed on top of the structural CLT slab.

Table 1: Section of tested setups and results overview (global ratings)

Setup	Dropped Ceiling	Elastic Support	Floating floor	Build-up Height(*)	Dry Screed Load [kg/m²]	$L_{n,w}$ (C_i) [dB]	ΔL_w ($C_{i,\Delta}$) [dB]	R_w (C;C_{tr}) [dB]
BS1	n.a.	n.a.	n.a.	n.a.	n.a.	87 (-5)	n.a.	39 (-1;-4)
BS2	n.a.	n.a.	n.a.	n.a.	n.a.	88 (-5)	n.a.	38 (-1;-3)
A	n.a.	Stravifloor Mat-W8$_a$	HydroFlam® 18 mm + Damping Layer 5 mm + OSB/3 18 mm	49 mm	26	67 (0)	23 (0)	50 (-1;-6)
B	Yes(**)	Stravifloor Mat-W8$_a$	HydroFlam® 18 mm + Damping Layer 5 mm + OSB/3 18 mm	49 mm	26	53 (0)	35 (3)	64 (-2;-8)
C	Yes(**)	Stravifloor Mat-W8$_a$	HydroFlam® 18 mm + OSB/3 18 mm	44 mm	22	53 (1)	34 (2)	63 (-2;-8)
D	n.a.	Stravifloor Mat-W25	Plywood 19 mm + Fermacell® Powerboard H20 12,5 mm + Plywood 19 mm	75.5 mm	46	61 (0)	27 (0)	53 (-1;-6)
E	n.a.	Stravifloor Mat-W25 strips [o.c. 610 mm]	Plywood 19 mm + Fermacell® Powerboard H20 12,5 mm + Plywood 19 mm	75.5 mm	36	56 (0)	32 (5)	59 (-3;-9)
F	n.a.	Stravifloor Mat-W25 strips [o.c. 610 mm]	Plywood 19 mm + Plywood 19 mm	63 mm	23	60 (0)	28 (5)	55 (-2;-9)
G	n.a.	Stravifloor Mat-W25	Gypsum topping 50 mm	75 mm	92	65 (0)	21 (0)	56 (-1;-7)
H	n.a.	Isolated Channel-M30 [Pad-M30 (30 mm)] (o.c. 610 mm)	HydroFlam® 18 mm + Fermacell® Powerboard H20 12,5 mm + OSB/3 18 mm	78.5 mm	35	54 (0)	34 (4)	62 (-2;-8)
I	n.a.	Isolated Channel-M30 [Pad-M30 (30 mm)] (o.c. 610 mm)	HydroFlam® 18 mm + OSB/3 18 mm	66 mm	26	57 (0)	31 (4)	60 (-3;-9)
J	n.a.	Isolated Channel-M30 [Pad-M30 (30 mm)] (o.c. 610 mm)	Plywood 19 mm + Plywood 19 mm	68 mm	23	57 (1)	30 (4)	59 (-3;-9)
K	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm])] (o.c. 610 mm)	Plywood 19 mm + Damping Layer 5 mm + Plywood 19 mm	93 mm	28	55 (-1)	34 (2)	64 (-2;-8)
L	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm)] (o.c. 406 mm)	Plywood 19 mm + Damping Layer 5 mm + Plywood 19 mm	93 mm	28	55 (-1)	35 (3)	63 (-2;-8)
M	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm)] (o.c. 406 mm)	Plywood 19 mm + Plywood 19 mm	88 mm	23	55 (0)	34 (4)	62 (-3;-9)
N	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm)] (o.c. 406 mm)	Plywood 15 mm + Fermacell® Powerboard H20 12,5mm + Plywood 15 mm	100.5 mm	32	54 (0)	35 (2)	63 (-2;-8)
O	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm)] w/ 30	3x Fermacell® Powerboard H20 12,5 mm + Plywood 19 mm	136.5 mm	52	47 (0)	42 (1)	67 (-2;-7)

		mm overheight (o.c. 406 mm)						
P	n.a.	Isolated Channel-M50 [Pad-M50 (50 mm)] w/ 30 mm overheight (o.c. 406 mm)	Plywood 19 mm + Plywood 19 mm	118 mm	23	53 (0)	36 (2)	65 (-2;-7)
Q	Yes(***)	n.a.	n.a.	-	n.a.	56 (-2)	31 (-5)	67 (-52;-7)
R	Yes(***)	Stravifloor Mat-W8$_a$	OSB/3 15 mm + Fermacell® Powerboard H20 12.5mm + OSB/3 15 mm	42.5 mm	32	43 (2)	44 (3)	74 (-4;-11)
S	Yes(***)	Stravifloor Mat-W8$_a$	OSB/3 15 mm + OSB/3 15 mm	30 mm	22	46 (1)	42 (3)	71 (-3;-10)

(*) Not including bare slab or dropped ceiling if applicable.

(**) 2 layers of 12.5 mm gypsum hung on metal grillage 150 mm below CLT slab.

(***) 2 layers of 18 mm gypsum hung on the metal grid with Stravilink CC-150 clips (on a grid of 600 mmx800 mm) 100 mm below CLT slab. Cavity filled with 50 mm insulation material.

2.3 Conclusions

When looking at floor-ceiling setups tested on 180 mm CLT slab, we can say:
There is an improvement in airborne and impact sound insulation of around 14 dB due to installing a suspending ceiling. The improvement is across all frequencies above 80 Hz. We see a negative effect of the mentioned dropped ceiling for low frequencies. However, it is important to mention that the installed dropped ceiling doesn't use resilient hangers or insulation material in the void and is not an acoustical dropped ceiling. The little negative effect of the dropped ceiling at low frequencies can be easily solved by adding insulation material in the void to avoid standing waves and using resilient hangers rather than stiff ones.
There is a significant improvement in airborne and impact sound insulation (around 3 dB) when using strips of 100 mm Stravifloor Mat-W25 spaced 610 mm versus full surface support with the same resilient material.
The full-surface wet systems tested can perform up to 3 dB better in airborne noise insulation but have lower performance (up to 4 dB) in impact noise insulation, with the most significant differences at frequencies above 160 Hz.
When comparing setups using discrete bearings with setups using mats as resilient support, there are improvements in airborne sound insulation up to 10 dB and 5-7 dB in impact sound insulation; those improvements are visible across the complete frequency spectrum.
The implementation of Fermacell® Powerboard H20, 12.5 mm thick and with a surface density of 13.5 kg/m², increases airborne and impact sound insulation by approximately 3 dB.
The current study used three types of wooden boards, HydroFlam®, OSB/3, and plywood, for testing. We observed no significant differences in acoustic performance among the test setups, which differed only in the type of wooden board used. This finding can be attributed to the similarity in thickness and density of the boards.
This study investigated the acoustic performance of test setups with channel spacing of 406 mm and 610 mm between bearings while maintaining a constant distance of 500 mm between bearings in the other direction. Results showed no significant difference in acoustic performance for frequencies starting from 50 Hz.
In lightweight acoustic floor systems, high-damping viscoelastic acoustic membranes (damping layers) are added between wood-based panels to address the dips in transmission loss in the resonance and coincidence-controlled regions, constrained layer damping (CLD) technique. This study found no significant difference between the results of test setups with and without a damping layer, except for slightly better results at the lowest end of available data (< 50 Hz) and higher transmittance above 800 Hz. This is because the impact of the standardized tapping machine used in the tests was insufficient to generate high shear loads in the damping layer. Therefore, no significant energy was lost in this layer during the tests. However, the panels and damping materials are expected to be more compressed for higher loads, resulting in higher deformation and shear deformation and a more pronounced benefit of constrained layer damping.
Comparing the setups using 30 mm bearings with those using 50 mm bearings, it is observed that there is a 2-3 dB improvement in airborne sound insulation and impact sound insulation. Notably, the improvements are predominantly observed in the low-frequency range due to the overall stiffness of the system and the increase in the void, which results in the reduced impact of stiffness of the entrapped air.
Increasing the air void between the floating floor system and the supporting structural floor from 50 mm to 80 mm results in a noticeable enhancement in airborne and impact noise insulation. The shift of the R_w curve towards the left at lower frequencies confirms this observation. As the air void becomes larger, this can be attributed to reduced air spring stiffness.

A system can be designed with a total build-up height (excluding structural slab) of 136.5 mm by combining an acceptable number of boards to achieve a high surface load of 52 kg/m² with an overheight of 30 mm and pads of 50 mm. This system can achieve global $L_{n,w}$ = 47 dB and R_w = 67 dB.
Comparing setups using 2 plywood boards and dry screed (setup P) with setups using a plywood board combined with 3 layers of Fermacell (O), an influence can be observed on sound insulation at low frequencies due to the added surface mass.
When looking at floor-ceiling setups tested on 140 mm CLT slab, we can say:
Due to the installation of an acoustical suspended ceiling, there is a significant improvement of up to 42 dB in airborne and impact sound insulation. The improvements are specifically better on medium and higher frequencies.
Combining a floating floor with a suspended ceiling improves airborne and impact sound insulation. Up to 7dB is achieved for the airborne sound insulation and up to 13 dB for the impact sound insulation on all frequencies, specifically below 80 Hz and above 125 Hz.
When looking at the results achieved on the first slab using a non-acoustical drop ceiling and on the second slab using an acoustical ceiling, a performance increase of 8 dB for both $L_{n,w}$ and R_w is observed. That shows the benefits of designing a solution that correctly treats both sides of the CLT slab. However, it is better to mention another difference between setups from both test campaigns, the thickness of the OSB boards used (15 mm vs 18 mm).
Higher differences between both test campaigns are mainly from 100Hz to 2000Hz. Also, the performance decrease below 80 Hz compared with the bare slab in the first campaign almost no longer occurs when using an acoustical suspended ceiling.

3 Improving flanking sound transmission in CLT constructions

An in-situ test campaign was carried out on a T-junction to learn more about the influence of elastic interlayers and acoustic brackets on the flanking sound transmission. The campaign was executed on a construction site situated in Lier, Belgium. The construction is a residential multi-storey building with structural elements at all levels above ground made of CLT panels. At the time of measurements, only the structure was present without finishing. Almost all walls are supported by an elastic layer. The test campaign aims to determine K_{ij} for the different transmission paths.

Figure 1 shows the geometry of the measured T-junction. The geometry and material properties of each component in the T-junction are known.

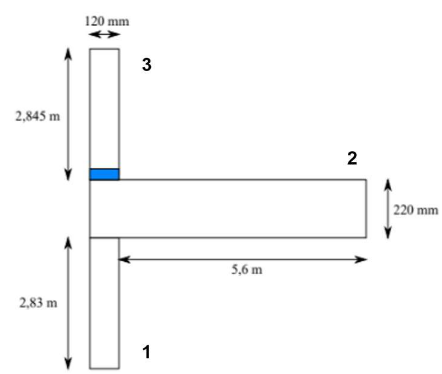

Figure 1: Geometry of measured T-junction. Taken from "Voorspelling van flankerende geluidtransmissie in lichtgewichtconstructies"[1]

Experiments to define K_{ij} for each of the transmission paths is done for 3 different connection types:

1. Straviwood WallBreak-S (Elastic strip, 12.5 mm)
2. Straviwood WallBreak-S (Elastic strip, 12.5 mm) + metal L-bracket (rigid connection)
3. Straviwood WallBreak-S (Elastic strip, 12.5 mm) + Straviwood WallBracket (acoustical wall angle bracket)

Measurements are done following international norm ISO 10848-1[2]. The positioning scheme of accelerometers and impact generation and the method of processing measurement data can be found in "Voorspelling van flankerende geluidtransmissie in lichtgewichtconstructies"[1].

The graphs below show measurements of cap K_{ij} and results from the empirical formula in Annex F of ISO 123543 for the different transmission paths in the T-junction.

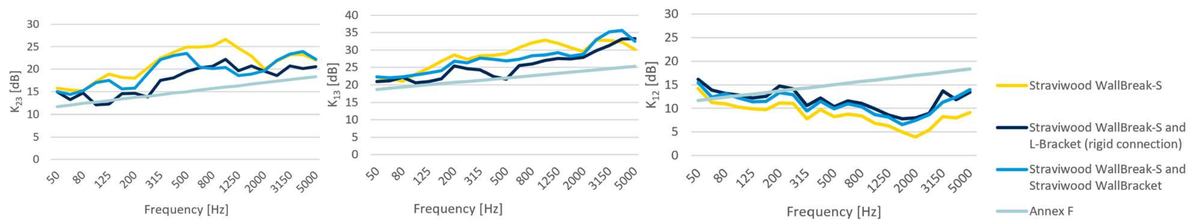

Figure 2: Vibration reduction index

When looking at the vibration reduction index of path 2 to 3, the path from the upper wall to the floor through the elastic layer, all measurements result in a higher vibration reduction index than the empirical formula of ISO 12354, showing that the elastic layer has a positive effect.

The measurement without any fixation shows the highest values. The measurements with angle brackets show lower values of K_{ij} compared to the setup with just the elastic layer. Acoustically decoupled angle brackets (Straviwood WallBracket) can mitigate the additional transmission over the junction, but still result in lower values of K_{ij} compared to connections without brackets.

When analysing K_{ij} for paths 1 to 3, the path from the upper wall to the lower wall, one can clearly see the same happening with the highest values for the setup without angle brackets and the lowest measurement values for the setup with fixed angle brackets. The empirical formulas of ISO 12354 result in too low values compared to the measurements.

When looking at the vibration reduction index for the floor – lower wall path, one sees that the vibration reduction index measurements show lower values than the empirical formula. This might be a result of the presence of an elastic layer between the floor panel and the upper wall. Due to the presence of the layer, the vibrations are directed to the non-isolated building element, resulting in a lower vibration reduction index from element 1 to element 2 and vice versa.

4 Improving flanking sound transmission in CLT modular construction

As modular timber structures in cross-laminated timber grow larger and higher, the modules need, from a structural point of view, to be more solidly connected to each other in the horizontal direction. This, unfortunately, results in acoustic contact bridges. In the Netherlands, a test arrangement was made with two modules in cross-laminated timber and various interconnection methods were used to test them. The connections ranged from full dilatation (without connection) and rigid metal connections to 3 variants of Straviwood ModuLink (an acoustical bracket for structural joints) with different stiffnesses. Apart from these, no other connections were made between the 2 modules.

The study was carried out on two used Finch Buildings modules, each measuring approximately 4 x 8.5 m. To minimize noise transmission through the supporting structure, each module was installed on 6 isolation pads in CDM-104 material with 300 x 140 x 12.5 mm dimensions, supplied by CDM Stravitec. The space between the modules' floors and the supporting structure was filled with rockwool. The modules were horizontally connected at roof level with 4 anchors with a horizontal spacing of 2.25 m.

Different connection methods were made and tested. The relevant specifications that influence the setup's acoustic behaviour are listed below.

Separating wall: 140 mm CLT5S – cavity of 100 mm filled with 50 mm glass wool – 140 mm CLT5S.

Acoustic anchors – type Straviwood ModuLink, with resonance frequency of approximately 15 Hz:

Type C1: type 0.8 kN SLS – 2 x CDM-102 isolation pads with 140 x 45 x 25 mm + 2 elastomeric sleeves M12.

Type C2: type 6.7 kN SLS – 2 x CDM-105 isolation pads with 140 x 45 x 25 mm + 2 elastomeric sleeves M12.

Type C3: type 17 kN SLS – 2 x CDM-106 isolation pads with 140 x 45 x 25 mm + 2 elastomeric sleeves M12.

Tests were performed following norm NEN-EN-ISO 717-1:20134 and normative references herein. In total, 5 variants with different connection methods were tested, ranging from fully dilatated (C0 – no connection at roof level) to over 3 acoustic anchors – type Straviwood ModuLink (C1, C2, and C3) and hard connections with stiff metal anchors (C4).

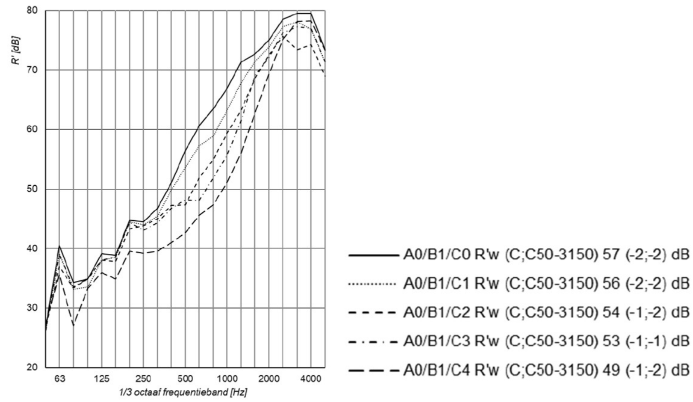

Figure: Airborne sound isolation results of different test setups

The measurement results show roof-level connections influence the overall sound insulation between modules. In the case of rigid steel couplings, this effect occurs over almost the entire frequency range, leading to a decrease in airborne sound insulation R'_w of 8 dB. When Straviwood ModuLink anchors are used, the effect is limited to 1 to 4 dB, depending on the stiffness/load capacity of the anchor.

Based on these results, one can analyse the change in vibration reduction index ΔK compared to a fixed connection (Setup C4 in the discussed test campaign). This calculation can be done based on formulas 18 and 20 in ISO 12354-13, from which one can deduce per situation the direct part $(R_{D,w})$ and the flanking part $(R_{F,w})$.

With ΔK the improvement of the vibration reduction index compared to the fixed connection results.

Table 3: Measurement results for different connection types

Connection type	R'_w (dB)	$R_{D,w}$ (dB)	$R_{F,w}$ (dB)	$\Delta K^{(*)}$ (dB)
C0	57	57	>67	>17
C1	56	57	63	13
C2	54	57	57	7
C3	53	57	55	5
C4	49	57	50	-

(*) improvement of the vibration reduction index compared to the fixed connection results.

The vibration reduction index ranges from 5 dB to 13 dB for the different brackets, with the one using the isolators with the lowest stiffness having the highest vibration reduction index.

The performance obtained in practical situations may deviate slightly from the results in the test setup. This is caused partly by the different geometry of rooms and the spacing between the structural Straviwood ModuLink anchors. From this study, high-quality acoustic connectors make an important contribution to achieving a project's limit and target values for sound insulation.

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Improving sound transmission in timber buildings: the role of flexible interlayers

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In buildings, sound transmission is influenced by both direct transmission and flanking transmission through the structure. In timber structures, the impact of sound propagation through wall/ceiling junctions should be considered as it can significantly affect the acoustic performance of these elements. Elastic separating layers can reduce flanking transmission and prevent the propagation of vibration through a timber structure.

The objective of this study was to characterize flexible interlayers made of homogeneous viscoelastic materials and develop test methods to evaluate their mechanical and acoustic properties. The focus was on measuring Kij on large-scale CLT mock-up to determine the acoustic benefits of flexible interlayers. The study found that the use of elastic separating layers can reduce the amount of flanking transmission between rooms and therefore improve the acoustic performance of the CLT structure. The mock-up testing included various junction types, "T-junctions" simulating perimeter walls and "X-junctions" simulating internal walls. The setup included the possibility of applying a load on the joint to simulate the behaviour of resilient interlayers in tall buildings and allowed characterize different types of interlayers (not limited to softer interlayers as previously done). The measured Kij and Δl values can be incorporated into the formulas of ISO 12354 to estimate the flanking sound reduction index.

1 Standard reference

1.1 Calculation model

ISO 12354 specifies calculation models designed to estimate the airborne sound insulation between adjacent rooms in buildings, primarily using measured data which characterize direct or indirect flanking transmission.

The simplified version of the calculation model in ISO 12354 predicts the weighted apparent sound reduction index on the bases of the weighted sound reduction indices of the elements involved, including the direct transmission and the flanking for each path. For each transmission path the weighted sound reduction index is predicted from the input data on the elements and junctions.

The weighted flanking sound reduction indices are determined from the input values according to Formula (20) of ISO 12354-1 and collapsed in Equation (1)

$$R_{ij,w} = \frac{R_{i,w}+R_{j,w}}{2} + \Delta R_{ij,w} + K_{ij} + \left(10lg\frac{S_s}{l_0 l_f}\right)\ [dB] \quad (1)$$

The quantity related to the vibration power transmission over a junction between structural elements, is characterize by the vibration reduction index K_{ij}.

Kij can be deducted from empirical data or from simulation with additional input data.

The presence of the flexible interlayers in junction can modify the junction performance.

If the transmission path crosses one joint:

$$K_{ij} = K_{ij,rigid} + \Delta_l\ [dB] \quad (2)$$

If the transmission path crosses two joints:

$$K_{ij} = K_{ij,rigid} + 2\Delta_l\ [dB] \quad (3)$$

Scope of the work was to provide reliable and measured data which characterize the vibration reduction index K_{ij} with flexible interlayers and the correction of Kij in presence of flexible interlayers in junction. Basic formatting instructions

1.2 Determination of K_{ij} and Δ_{ij}

The vibration reduction index Kij is a quantity related to the transmission of vibrations through the structural elements of a junction. The vibration reduction index Kij shall be measured using structure-borne excitation and calculated according to the equation (4)

$$K_{ij} = \frac{D_{v,ij}+D_{v,ji}}{2} + 10log\frac{l_{ij}}{\sqrt{(a_i a_j)}}\ [dB] \quad (4)$$

Vibration measurements shall be carried out using accelerometers mounted directly onto the surface of the test element. It shall have a sufficient sensitivity and low noise in order to obtain a signal-to- noise ratio of the measurement chain that is adequate to cover the dynamic range of the response of the structure.

To generate a vibrational field, the excitation shall be steady-state (shaker) or transient (hammer or impact machine). In each frequency band, the measured velocity level on the receiving element shall be at least 10 dB higher than the background noise level in any frequency band. If this is not fulfilled, corrections shall be applied as described in ISO 10140-4. The correction value shall not exceed 1,3 dB.

The measurements shall be performed using one-third octave band filters having at least the following centre frequencies, in hertz: 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150.

The $\overline{K_{ij}}$ value, is the arithmetic average of K_{ij} within the frequency range 200 Hz to 1 250 Hz (one-third octave bands)

$\Delta_{l,ij}$ for a specific path is calculated as difference between $\overline{K_{ij}}$ of the same path with and without resilient interlayer. $\overline{K_{ij}}$ is the arithmetic average of K_{ij} within the frequency range 200 Hz to 1 250 Hz (one-third octave bands).

$$\Delta_{l,ij} = \overline{K_{ij,with}} - \overline{K_{ij,without}}\ [dB] \quad (6)$$

In order to define a standardized way to characterize the performance of the flexible interlayers, an European Assessment Document (EAD) was written. The definition of an EAD permit to issue an European Technical Assessment (ETA), that allows manufacturer to introduce innovative products and new features to the entire European market.

2 Measurements and results

One of the most important objectives of the project was to assess Kij on a big-scale CLT prototype to evaluate the improvement that can be obtained using flexible interlayers XYLOFON: monolithic polyurethane mixture 6mm thick.

For this project two types of junctions were tested:

- "X-junctions" simulating the junction of internal walls.
  Top wall: 5-ply CLT, 100 mm, (2,4 m x 3 m)
  Floor: 5-ply CLT 100 mm (2,4 m x 7,1 m)
  Bottom wall: 5-ply CLT, 100mm, (2,4 m x 3 m)

Figure 1: big-scale CLT prototype with X-junction

- T-junctions" simulating the junction of perimeter walls
  Top wall: 5-ply CLT, 100 mm, (2,4 m x 3 m)
  Floor: 5-ply CLT 100 mm (2,4 m x 3,5 m)
  Bottom wall: 5-ply CLT, 100mm, (2,4 m x 3 m)

Figure 2: big-scale CLT prototype with T-junction

In the notch of the top wall it is possible to install a hydraulic jack that allows applying a load to the wall and imposing the correct level of load on the interlayer.

2.1 Influence of the interlayer (with and without flexible interlayer):

The scope of the measurements is to understand the acoustic improvement of the monolithic resilient interlayer in operational condition, considering fastening system and the response on CLT structure.

In some cases, the flexible interlayers are used only between the upper wall and the floor even if the best practise requires to install them both: between the upper wall and the floor and between the lower wall and the floor.

If the material is properly loaded, the improvement is significant, even if the material is quite thin (6mm) because of the viscous properties of the monolithic structure.

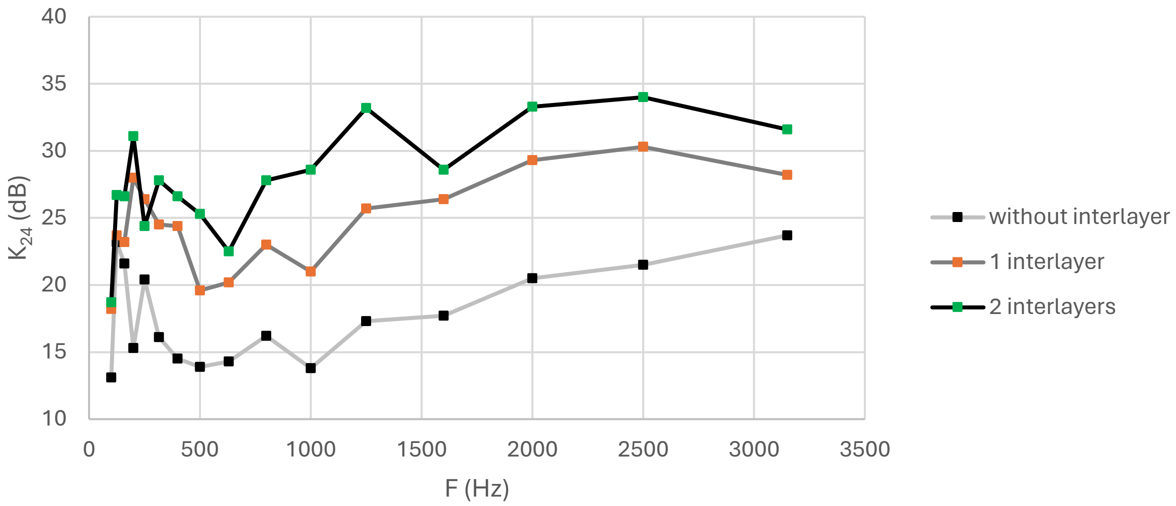

Figure 3: comparison between not using interlayer, using 1 interlayer and using 2 interlayers

2.2 Influence of the load:

Flexible interlayers must be properly loaded in order to optimize their performance and reduce the propagation of the vibration through the structure.

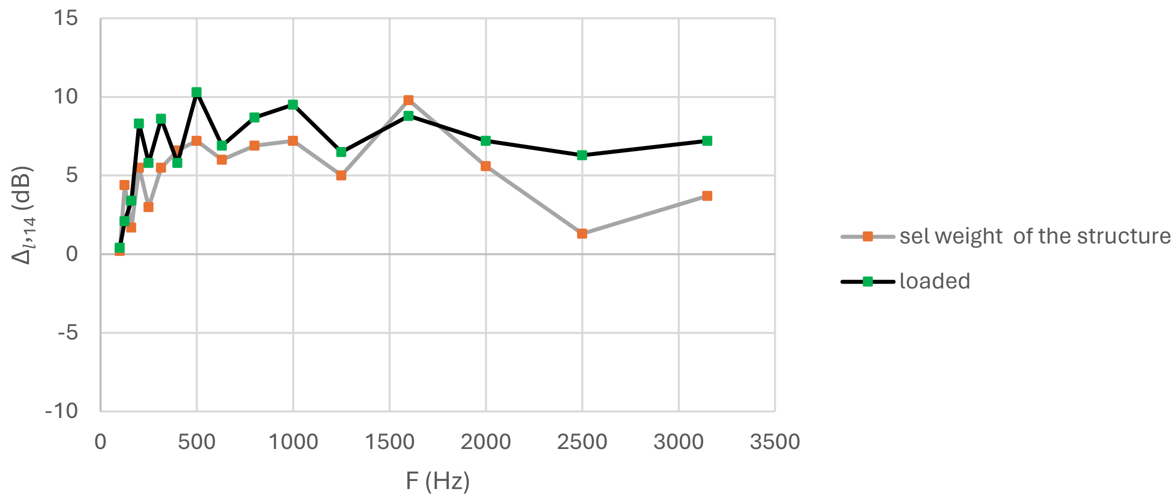

Figure 4: comparison between interlayer under self-weight structure and interlayer properly loaded

The proper load of the structure is defined according to the compressive modulus of the product.

Different hardnesses were tested, in order to demonstrate that alle the assumption can be considered valid also for harder version.

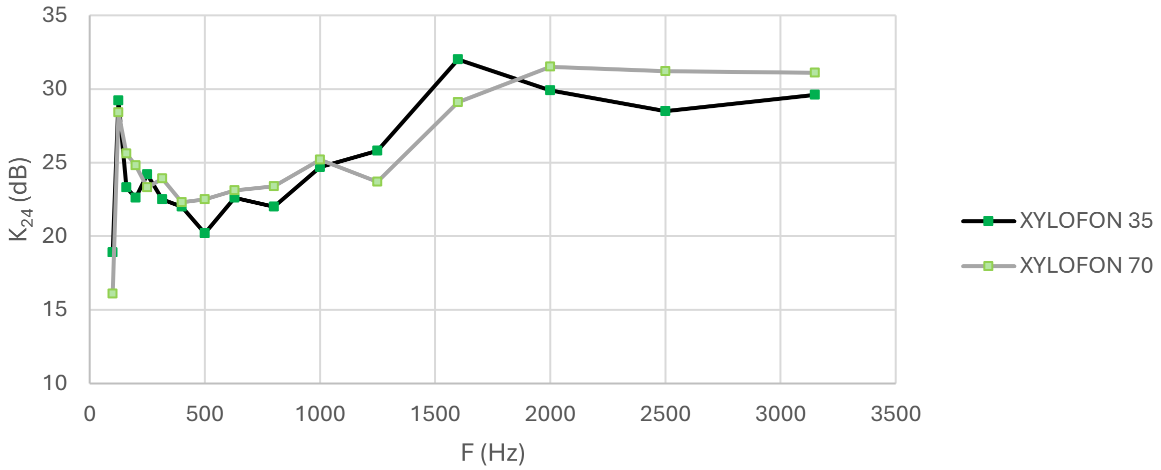

Figure 5: comparison between hardnesses, the difference is negligible

2.3 Influence of the type of junction (X vs T):

If we compare two setups without interlayer where the only variable that changes is the type of joint (X or T), we notice that the X joint tends to dissipate more energy than the T joint.

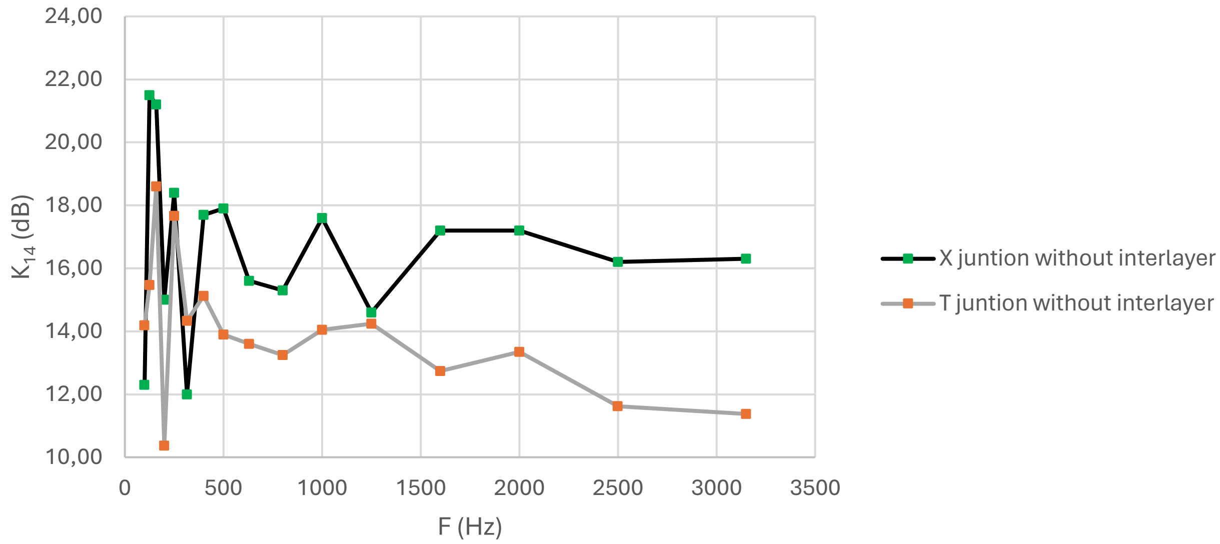

Figure 3: comparison without interlayer: X-juntion vs T-juntion

The X-junction without interlayers dissipates more energy compared to the T-junction. This difference tends to diminish in the configuration with the flexible interlayer because it dissipates the energy.

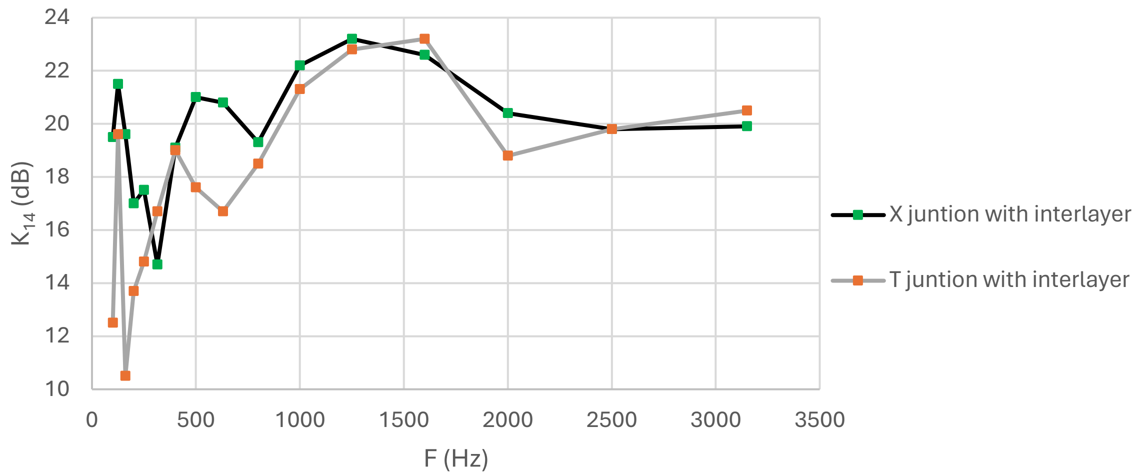

Figure 4: comparison with interlayer: X-juntion vs T-juntion

3 Conclusion

To date, there are no common standards that describe in detail how to characterize the interlayers and which performance shall be determined to guarantee a proper static and acoustic performance.

The study shows the variables that need to be considered in the analysis of the acoustic performance of flexible interlayers.

Thin monolithic interlayers guarantee low deformation, are statically safe and thank to the viscosity they behave very good in acoustic too.

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FEM-based simulation procedure to predict impact sound insulation of a timber floor

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During the last decades, researchers have actively developed simulation procedures using the finite element method (FEM) to predict impact sound insulation of timber slabs and floors. In the case of full timber floors, the simulations presented in the research literature have been limited to low-frequency analyses, despite the obvious ability of the method to work in a broader frequency range as well. The validation of simulation models has again meant model calibrations and experimental modal analyses, which can rarely be carried out in connection with typical product development tasks. The purpose of this study was to determine the applicability of a modern FEM-based simulation procedure for predicting the normalised impact sound pressure level L_n of a timber floor, when information provided by material manufacturers is used as input data. Force excitation caused by an ISO standard tapping machine was determined by means of a validated simulation analysis utilizing explicit time integration and FEM. The modelling of sound radiation itself was performed in a frequency domain FEM analysis. The model of the timber floor was validated blindly, i.e. the simulation model was prepared before the laboratory measurements of impact sound insulation. Based on the results, it was possible to predict the laboratory measurement results of L_n, and the single-number quantities with a reasonable accuracy. The differences between the measurement and simulation results could be explained by uncertain material properties.

1 Introduction

Impact sound insulation (ISI) is one of the principal technical parameters dimensioning the structural layers of timber floors in apartment buildings. Thus, evaluating the ISI of the floor is probably the most important task of acousticians working in timber construction. Instead of choosing acoustical solutions for the floors based on experience gained in previous projects or from laboratory measurements, an appealing approach would be to use prediction tools to evaluate the ISI of the floor. Here, the focus is on the ISI prediction of timber floors with the finite element method (FEM) with an emphasis on simulating the normalised impact sound pressure levels L_n generated by the ISO standard tapping machine (STM) [1,2].

FEM has previously been used to simulate the ISI of timber floors excited with the STM. For example, Rabold [3], Rabold et al. [4–6], Kohrmann [7], Kohrmann et al. [8], and Coguenanff [9] have simulated the low-frequency ISI of different timber floors below 200 or 250 Hz applying FEM. Additionally, FEM has been applied for predictions in a broader frequency range. Previously, this has successfully been carried out for concrete slabs [10,11], but Wang et al. [12] have presented a FEM model simulating the radiated impact sound pressure level of a timber rib slab. However, according to the authors' knowledge, validated FEM models predicting the ISI of full timber floors in a broader frequency range have not been published so far. Thus, a full understanding of the suitability of the FEM for research and development (R&D) purposes of timber floors has not yet been created.

The purpose of the study reported in this paper was to apply a FEM procedure to imitate the ISI laboratory measurements of a full-scale timber floor in a simulated R&D task. The proposed FEM model predicting the normalised impact sound pressure level L_n of a STM driven bare timber floor was created based on material data presented by the product manufacturers and validated against the laboratory measurements. To demonstrate the ISI prediction process of the full floor with the floor covering, the simulation results of the bare floor were supplemented with ΔL measurement results. The computations were performed up to 1000 Hz 1/3-octave band. This paper is based on a wider study reported in the reference [13].

2 Materials and Methods

2.1 Floor structure

The structure under study was a full timber floor with a rib slab as its bearing structure and a floating multilayer parquet as its floor covering. In addition, the full floor consisted of plasterboards attached onto the slab, a suspended plasterboard ceiling, and a glass wool layer installed between the ribs (Figure 1). The width of the floor was 3020 mm, and the length 3870 mm. In the laboratory measurements and in the simulations, the floor was supported to the laboratory opening at their short ends by elastomer strips installed between the LVL beams of the rib slab and the supporting steel frame of the opening.

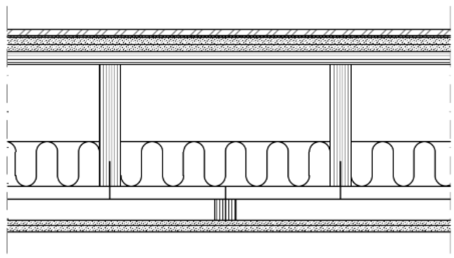

Figure 1: Floor structure. The layers of the floor from top to bottom were: a multilayer parquet (thickness h = 14 mm) on an underlayment (h = 3 mm), two 15.5 mm plasterboard layers, rib slab with a 27 mm LVL deck and 260 mm LVL beams (width 45 mm, c/c = 490 mm), a 95 mm glass wool layer between the ribs, overhead boards from LVL deck (h = 27 mm, b = 100 mm, c/c = 550 mm) screwed below the ribs, LVL battens from LVL beams (h = 45 mm, b = 45 mm, c/c = 490 mm) screwed to the overhead boards between the ribs, and two 12.5 mm plasterboard layers.

2.2 Simulations

Impact sound pressure level L_n generated by a STM on the timber floor was computed applying a three-stage method as illustrated in Figure 2. At the first stage, impact force excitation generated by the STM was determined by using explicit dynamics analysis with Ansys LS-DYNA (smp s R10.1.0 Revision: 123264) following the procedures presented in reference [14]. With a post-processing method [14–16], the impact force pulses were converted into frequency domain to present continuous operation of the STM. The second stage involved FEM simulations of the impact sound radiation of the floor excited by the previously determined excitation. At this stage, the computations were performed in frequency domain by using COMSOL Multiphysics 6.1 with a 2 Hz frequency resolution which corresponds to the excitation line spectra of a STM. The radiated sound power of the floor was solved directly with the FEM model by applying a half-infinite acoustical fluid domain below the floor. At the third stage, the simulated results for the radiated sound power were post-processed to present the normalised impact sound pressure level L_n in a receiving room. At this stage, the effect of the floor covering on the ISI was also introduced.

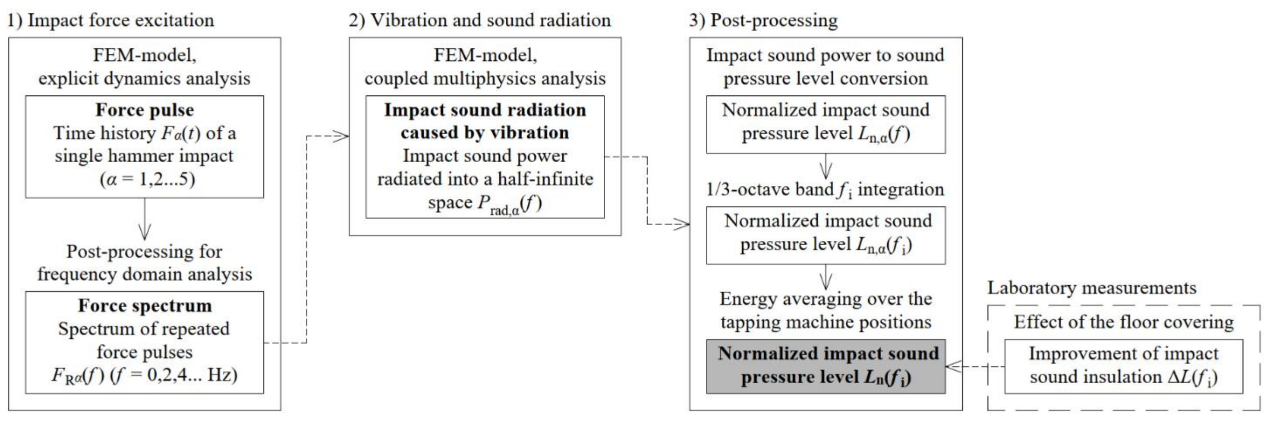

Figure 2: A three-stage FEM procedure to determine the normalised impact sound pressure level L_n of a timber floor excited by a STM.

The procedure was applied to predict the ISI of the timber floor in the frequency range enveloping the 1/3-octave bands 50–1000 Hz. The actual FEM simulations were carried out for the floor without the floor covering. The prediction result of L_n for the full floor was achieved simply with a difference between the results for the bare floor and for the measured improvement of impact sound insulation (ΔL) of the floor covering. This was performed to demonstrate the application of the proposed FEM procedure for the full timber floor. Because it has been noted that the floor coverings should be measured on a floor representing the behaviour of the timber floor under study [17], the ΔL of the floor covering was calculated from the ISI laboratory measurements of the timber floor of the present study.

Main geometry of the simulation model has been illustrated in Figure 3 together with the computational mesh at 1000 Hz. The frequency-dependent element mesh consisted of quadratic hexahedral and tetrahedral Lagrange elements. The minimum criteria for the mesh density to be met was to achieve at least five elements per wavelength in all the domains to represent the waves on the mesh. In case of the structural domains, bending waves were taken into account in the requirements considering that the structures behave as ribbed plates. The materials of the floor were described as linear elastic. The material properties were provided by the product manufacturers (public datasheets) and supplemented with information based on the values reported in the literature. [13]

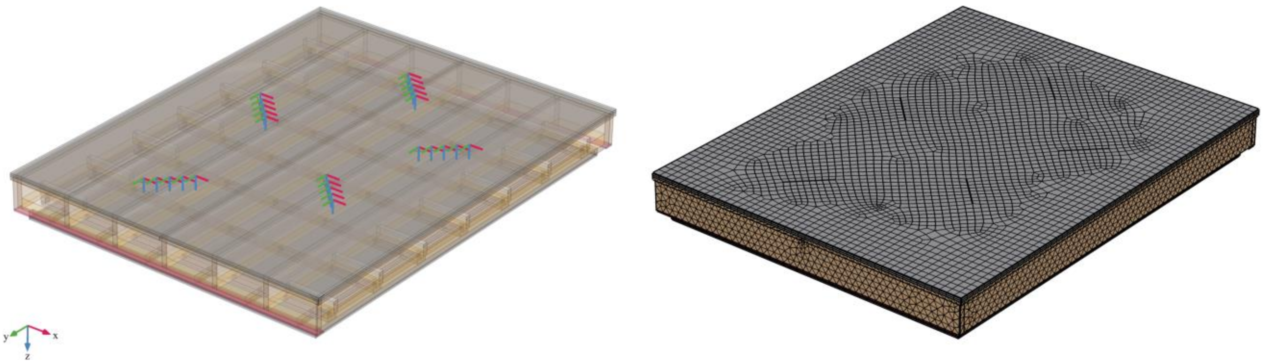

Figure 3: Geometry and mesh (at 1000 Hz) of the FEM model. The half-infinite airspace below the floor has been hidden to highlight the structural features of the floor.

2.3 Impact sound insulation measurements

Impact sound insulation of the floor with and without the floor covering was measured in accordance with the standard ISO 10140-3 [18] in an accredited building acoustics laboratory (Eurofins Expert Services Oy, Espoo, Finland). The experiments were performed to achieve 1/3-octave band results for the normalised impact sound pressure level L_n in the frequency range 50–5000 Hz. STM was used as an impact sound source in five predetermined source positions. The improvement of impact sound insulation ΔL of the floor covering on the floor under study was determined based on the measurements.

2.4 Model validation

The impact force excitation model was validated by comparing the simulated impact force to the experimental results presented in a previous study [19] at top of the centre beam and between the beams on the floor F9.0 resembling the floor under study without the floor covering. The model for simulating the impact sound radiation of the bare floor was validated by a comparison with the ISI measurement results. The validation was performed blindfolded, i.e., the simulation model of the bare floor was fully constructed before the ISI measurements were performed and the results were given to the authors. Moreover, the simulations and the measurements were performed with the same source positions. The object of the validation was to find out how satisfactory the simulation procedure performs in predicting the normalised impact sound pressure level L_n of a timber floor if only information of the construction and the used materials were known. As noted above, this represents a R&D task, where the prediction is performed based on available material data.

3 Results and discussion

3.1 Impact force model validation

Figure 4 compares the simulated impact force results with the measurement results [19]. The validation model predicted the measured impact force pulse on the previously studied floor F9.0 at the top of the centre beam and between the beams with a reasonable accuracy. Thus, the impact force model was regarded as valid to be applied in predicting the impact force excitations of the floor under study. The validated impact force models were modified to correspond with the timber slabs of the bare floor and applied to compute its impact force excitations following the same procedure.

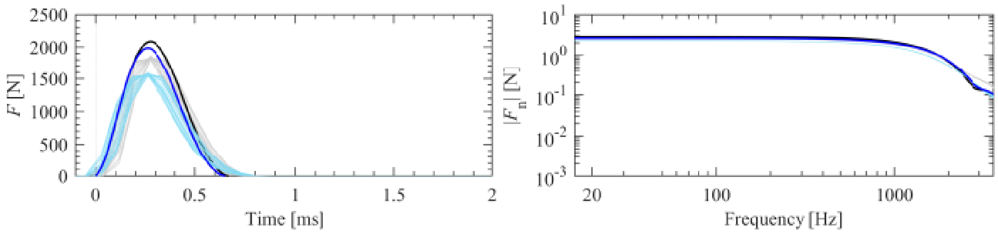

Figure 4: Impact force model validation: Simulated impact force pulses $F(t)$ and magnitudes of their single-sided amplitude spectra F_n (black (at top of the beam) and blue (at between the beams) lines) and the corresponding measurement results from the experiments [19] (thin grey (at top of the beam) and light blue (at between the beams) lines).

3.2 Simulation and measurement results for normalised impact sound pressure levels

Figure 5 compares the measurement and prediction results for the L_n. Additionally, the measurement and simulation results have been given as single-number quantities $L_{n,w}$, C_I, and $C_{I,50\text{-}2500}$ calculated according to the standard ISO 717-2 in Table 1. These values were determined also for the prediction results although the simulated frequency range did not cover all the frequencies needed in the standardised single-number rating. Based on the results, the equivalency between the measurement and prediction result was reasonable although the exact material properties of the floor were not known. The greatest deviations between the results occurred at 80, 315 and 500 Hz frequency bands. The differences in single-number quantities varied between 0 and 4 dB.

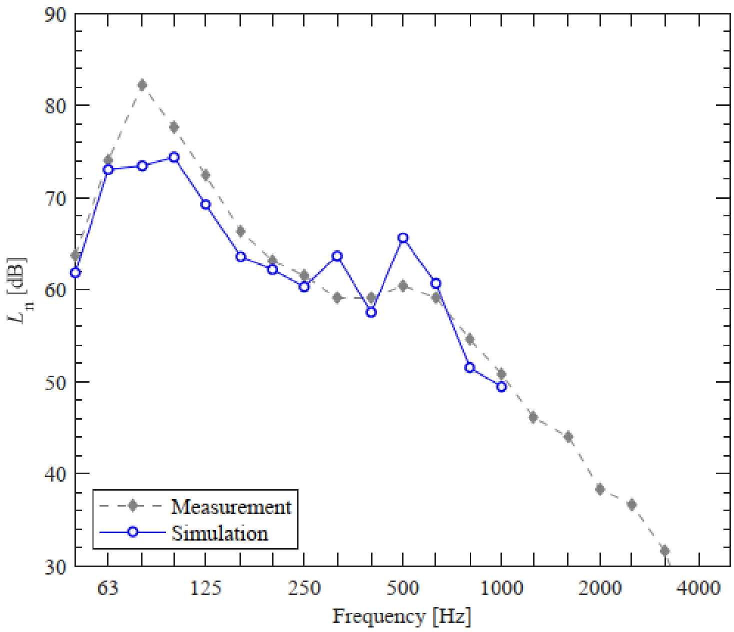

Figure 5: Simulated and measured 1/3-octave band integrated normalised impact sound pressure levels L_n.

Table 1: Single-number quantities determined based on the simulation and measurement results for L_n.

Single-number quantity	Measurement	Simulation	Difference (meas.–sim.)
$L_{n,w}$	60 dB	60 dB	0 dB
C_I	4 dB	2 dB	2 dB
$C_{I,50\text{-}2500}$	9 dB	5 dB	4 dB

The performed simulations mainly utilised data provided by the product manufacturers instead of measured material properties of the actually studied floors. This starting point corresponds with a R&D task where similar source information represents the best available data for the analysis. However, because of the chosen method, it remains unclear whether the actual material properties would improve the equivalence between the simulation and measurement results for the ISI. Thus, it is highly recommended that a study of this type is repeated with known material data of measured floors. This would also bring insight into the possible modelling inaccuracies.

The modelling was performed according to the best current knowledge of the authors but due to the complexity of the studied structures inaccuracies can occur. These inaccuracies can be related to the mesh density, boundary conditions and contacts/constrains between parts, to give examples. Moreover, according to the wider study, it was suggested that the material parameters applied in the simulations did not fully correspond to those of the measured floor [13]. Due to the deterministic nature of the FEM simulations, changing the parameters will have an effect on the results. Another point of view is that it is not known how sensitive the simulation model (or even the measurement result) is for different kind of changes in the timber floors. These changes include variations in materials, dimensions, and joints, to give examples. For this reason, it is recommended that a sensitivity analysis for the simulations is performed.

4 Conclusions

The study reported in this paper investigated the ability of FEM simulations to predict impact sound insulation (ISI) of a full timber floor from a R&D perspective. The simulation procedure was applied to find out whether the currently known FEM-based simulation procedures can be utilised to imitate ISI laboratory measurements of timber floors driven by an ISO standard tapping machine (STM) in R&D tasks. Based on the results, the FEM simulation procedure was able to predict the laboratory measurement result of the L_n of a full timber floor with reasonable accuracy. Therefore, the method can be considered suitable for the studied purpose, although further research on the subject is still needed.

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