
# EUROREGIO BNAM2022

Joint Acoustics Conference

9th-11th of May, Aalborg, Denmark

# Book of Abstracts

# Welcome by European Acoustics Association (EAA)

On behalf of the EAA Board, it is a great pleasure to welcome all participants to EUROREGIO 2022, the first face-to-face EAA European Acoustics Conference after the pandemic lockdown in 2020 and 2021 and an excellent opportunity to reconnecting during fruitful presentations and discussions. The congress is merged with the bi-annual Baltic-Nordic Acoustic Meetings being held in the Nordic countries since 1954 and thus BNAM 2022 being the 35th since the start. The EUROREGIO concept is much younger with the conference in 2022 being the 5th since the first being held in 2010 in Ljubljana/Slovenia.

We appreciate very much the great efforts done by the Danish Acoustical Society (DAS) and Aalborg University in organizing the congress, and not least for keeping the "physical" setup as a strong goal and preparing major parts of the conference, while many people still considered the pandemic situation and development too uncertain for a face-to-face conference without virtual attendance in parallel. – We also give our acknowledgments to the Nordic Acoustics Association (NAA) for their institutional support and cooperation.

In society and research, the United Nations Sustainable Development Goals (UN SDGs) have got a rapidly developing attention, and in line with this strong trend and necessity, the conference organizers have encouraged focus on the UN SDGs throughout EUROREGIO 2022.

In the weekend preceding the conference, an EAA Training School is held in the Campus of Aalborg University with distinguished experts in the Acoustics field as lecturers. The School is sponsored by EAA and is part of the Young Forum activities, which also comprise participation in the conference, and invitation to the Young Acousticians Network event on Monday evening (sponsored by DAS).

At last, we want to draw your attention to the option extending your conference paper to a full research article in the EAA journal Acta Acustica, now full open access.

We wish you all, organizers, participants, students, sponsors and exhibitors an excellent and fruitful EUROREGIO 2022.

On behalf of the EAA Board
Birgit Rasmussen
EAA Vice-President

# BNAM and EuroRegio 2022 in Aalborg Denmark

Dear Colleagues in Europe

On behalf of the Nordic Acoustic Association (NAA) I wish you very welcome to BNAM and EuroRegio in Ålborg in May 2022. You will be well greeted by a great city, and I sincerely hope you will enjoy your stay and earn new, valuable knowledge.

I also extend my heartfelt thanks to Lars Bramsløw and Dorte Hammershøi and all their colleagues for their effort in preparing the conference. In agreement with the NAA, the organizing committee has found leading researchers for keynote talks, supporting the UN sustainable development goals.

The NAA was established in the year 1954. The main driving force and the first president of the NAA was the Professor Fritz Ingerslev from Denmark, who saw the value in Nordic collaboration and coordination on acoustics. The biannual Nordic Acoustics Meetings, later Baltic-Nordic Acoustics Meetings, have been held since that same year 1954 on a rotating basis among the member countries. This time, the NAA welcomes joining forces with the European Acoustics Association (EAA).

Coming from the small nation of Icelanders, it was of great value to me to have good and clever neighbours. After my study at the University in Iceland, I studied acoustics first at DTU in Copenhagen and later at the TU in Berlin.

This is the first BNAM conference after the covid pandemic. What we have learned through the crisis is how to convene an e-conference. It is not the same as a conference in the flesh, but still of value. While they can improve the sustainability of conferences and spread the word wider, meeting in person cannot easily be surpassed. So, we are looking forward to once again having a traditional physical conference!

A very clever teacher I had in Berlin once said: “We can divide the world in three parts: Material, Energy, and Information. Albert Einstein taught us the connection between the first two. The Information is however unique: It does not diminish by being spread.”

I wish you a very valuable conference,

Ólafur Hjálmarsson
NAA President

# Welcome to the Joint

# Euroregio / Baltic-Nordic Acoustics Meeting 2022

Our proposal for hosting the Euroregio 2022 as part of our host obligation for BNAM 2022, was motivated by the purpose of the Euroregio conferences; to move around in Europe and embrace the regional conferences. The BNAM conferences have a solid circulation record, and it is always a pleasure to host it in Denmark, where it originally started.

Little did we know of the events that would follow, and the uncertainty that should follow the project for the coming two years. We watched national and international conferences turn digital, and we got used to prepared videos or live presentations with a variety of challenges. The flexibility of viewing presentations offline proved some upsides as well, when everything had to be managed from the same home environment. But even the most professional attempts to make interaction in digital environments would exclude many whatever the options offered.

The idea of conferences is to interact with peers, formally and informally, and to create networks that would support professional life in science, whether academic or industrial. This is the one thing, which the majority of digital conferences have failed to do.

While we wait for new conference formats to emerge, which will eventually combine the best from both worlds, we should celebrate the opportunities that allows us natural interaction with all its pros and cons.

It is therefore with great pleasure that we welcome delegates for the ER BNAM 2022 conference in Aalborg, where we will enjoy the scientific talks, investigate opportunities in the exhibition, and introduce juniors to the real life experience of attending a physical conferences, with all its opportunities for professional interaction.

Dorte Hammershøi
Professor, Aalborg University
Chair

Lars Bramsløw
President
Danish Acoustical Society

# Welcome Exhibitors and Sponsors

On behalf of the EAA, NAA and the Danish Acoustical Society, the organizers are pleased that so many companies have shown interest in the conference and exhibition that takes place in Aalborg 9th-11th May 2022.

We are proud that the sponsorship options have been received well, and that so many different organizations are supporting the Euroregio/BNAM 2022 as either Platinum, Gold or silver sponsors.

The exhibition has since long time been fully booked with a variety of exciting companies, and this demonstrates that physical meetings are important for all of us.

Thanks to all sponsors and exhibitors. We look forward to spending these days with you in the technical sessions and as a part of the social program.

Welcome to Aalborg.

On behalf of the EAA, NAA and the Danish Acoustical Society

Dorte Hammershøi, Chair, Prof. Aalborg University

Lars Bramsløv, President of the Danish Acoustical Society

## The organizing committee

- Dorte Hammershøi, Professor PhD, Aalborg University (chair)
- Lars Bramsløw, Senior scientist, Eriksholm Research Center and president of the Danish Acoustical Society (co-chair)
- Flemming Christensen, Associate Professor PhD, Aalborg University (co-chair)
- Claus Møller Petersen, Senior Consultant, SWECO, Denmark
- Rodrigo Ordoñez, Associate Professor PhD, Aalborg University
- Anders Olsen, Vibratec, Denmark (Sponsors and Exhibition management)
- Dorthe Sparre, Administrative secretary, Aalborg University
- Sreeram Kaithali Narayanan, PhD student, Aalborg University (YAN liason)
- Henriette Bugge Mortensen, Senior consultant, Destination Nord & Partners A/S (registration, hotels and tours)

## Liasons of the Nordic Acoustic Association (NAA)

- Ólafur Hjálmarsson, NAA president
- Finnur Pind, Treble Technologies, Iceland
- Hans Bodén, Professor, KTH Royal Institute of Technology, Sweden
- Henrik Möller, senior consultant, AKUKON, Finland
- Mari Alvik Hagen, Senior consultant, Rambøll, Norway

## Liasons of the European Acoustics Association (EAA)

- Birgit Rasmussen, Senior scientist, Aalborg University (EAA vice-president, conferences)
- Elie Abi Raad, PhD student RWTH Aachen University (YAN chairman)
- Karin Loh, PhD student, RWTH Aachen University (YAN vice-chairman)

## Scientific committee

- Lars Bramsløw, Eriksholm Research Centre, Snekkersten, Denmark
- Dorte Hammershøi, Aalborg University
- Flemming Christensen, Aalborg University, Denmark
- Rodrigo Ordoñez, Aalborg University, Denmark
- Piotr Majdak, Acoustics Research Institute, Austrian Academy of Sciences, Austria
- Douglas Manvell, DMdB, Denmark
- Teresa Carrascal García, Quality in Construction Unit, Eduardo Torroja Institute for Construction Science. IETcc–CSIC, Spain
- Michael Haverkamp, Germany
- Christian Sejer Pedersen, Aalborg University, Denmark
- Claus Møller Petersen, Sweco, Denmark
- Alain Bradette, Sweco, Norway
- Tobias Neher, University of Southern Denmark, Denmark
- Louena Shtrepi, Politecnico di Torino, Italy

# Table of contents

## Monday 9 may 2022 - Active acoustic systems / Structure-borne sound

### Musiksalen

**Chair Person (s): U.P. Svensson**



## Monday 9 may 2022 - Structure-borne sound / Sound quality

### Musiksalen

**Chair Person (s): F. Doleschal**



## Monday 9 may 2022 - Sound quality

### Musiksalen

**Chair Person (s): F. Doleschal**



## Monday 9 may 2022 - Virtual acoustics & auralization

### Gæstesalen

**Chair Person (s): F. Christensen**



## Monday 9 may 2022 - Audio signal processing

### Gæstesalen

**Chair Person (s): J. østergaard**

## Monday 9 may 2022 - Technical audiology
### Gæstesalen
**Chair Person (s): D. Hammershøi**



## Tuesday 10 may 2022 - Noise assessment and road traffic noise 3
### Europahallen
**Chair Person (s): B.M. Larsen**



## Tuesday 10 may 2022 - Keynote 2
### Europahallen
**Chair Person (s): L. Bramsløw**



## Tuesday 10 may 2022 - Noise assessment and road traffic noise 4
### Europahallen
**Chair Person (s): B.M. Larsen**

## Tuesday 10 may 2022 - Objective and perceptual aspects of measured and simulated alternatives to conventional spaces
### Musiksalen
**Chair Person (s): L. Shtrepi**



## Tuesday 10 may 2022 - Room and Concert hall acoustics 1
### Musiksalen
**Chair Person (s): J.H. Rindel, H. Möller**



## Tuesday 10 may 2022 - Current topics in hearing-aid processing and fitting 1
### Gæstesalen
**Chair Person (s): T. Neher**



## Tuesday 10 may 2022 - Current topics in hearing-aid processing and fitting 2
### Gæstesalen
**Chair Person (s): T. Neher**

## Wednesday 11 may 2022 - Room and Concert hall acoustics 2
## Europahallen
**Chair Person (s): J.H. Rindel, H. Möller**



## Wednesday 11 may 2022 - Keynote 3
## Europahallen
**Chair Person (s): C. Møller Petersen**



## Wednesday 11 may 2022 - Room and Concert hall acoustics 3
## Europahallen
**Chair Person (s): J.H. Rindel, H. Möller**



## Wednesday 11 may 2022 - Poster session
## Europahallen
**Chair Person (s): F. Christensen**

## Wednesday 11 may 2022 - Room and Concert hall acoustics 4
## Europahallen
**Chair Person (s): M. Skålevik**



## Wednesday 11 may 2022 - Closing ceremony
## Europahallen
**Chair Person (s):**



## Wednesday 11 may 2022 - Current topics in hearing-aid processing and fitting 3 / Building Acoustics 1
## Musiksalen
**Chair Person (s): P. Rye**



## Wednesday 11 may 2022 - Building acoustics 1
## Musiksalen
**Chair Person (s): B. Rasmussen**

## Wednesday 11 may 2022 - Building acoustics 2

### Musiksalen

**Chair Person (s): B. Rasmussen**



## Wednesday 11 may 2022 - Structure-borne sound & numerical acoustics

### Gæstesalen

**Chair Person (s): A. Olsen**



## Wednesday 11 may 2022 - Numerical acoustics 1

### Gæstesalen

**Chair Person (s): H. Boden**



## Wednesday 11 may 2022 - Numerical acoustics 2

### Gæstesalen

**Chair Person (s): S.H. Nielsen**



## Wednesday 11 may 2022 - Subjective experience of soundscapes

### Gæstesalen

**Chair Person (s): S.H. Nielsen**

# Abstracts

Mo. 8:30 Europahallen Opening ceremony

**Opening ceremony**

***D. Hammershøi***

*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*dh@es.aau.dk*

In May 9-11 2022, Aalborg will host the bi-annual Baltic Nordic-Acoustic Meetings (organized by the Nordic Acoustic Association), which this time is merged with the EUROREGIO (organized by the European Acoustics Association). The conference brings together academics, engineers, consultants and other practitioners in the wide field of acoustics.

While embracing the wide field of acoustics, special attention is given to aspects relating to the United Nations Sustainable Development Goals. Three related keynotes are already scheduled, and we hope that many will take the opportunity to share efforts towards these goals.

The conference is preceded by the PhD training school (Young Forum) on May 7-8 2022. The training school will take place on the campus of Aalborg University (campus zip code 9220 Aalborg ø) utilizing seminar and group rooms for lectures and discussions, and lab facilities for hands-on exercises.

Favorable options also exist for sponsors and exhibitors. The exhibition will be adjacent to the main lecture hall (Europahallen at Aalborg Congress and Culture Centre) in the same airy foyer as lunches and refreshments are served (included in registration).

Denmark is currently experiencing the same rise of COVID-19 infections as many other countries in Europe during winter. Adequate measures are in place to reduce the spread of the virus, incl. effective test and vaccination programs. The current attention makes us confident that we can meet physically in May. We hope that you share our optimism, and welcome the opportunity to re-connect in joint physical spaces in science.

On behalf of the EAA, NAA and the Danish Acoustical Society,

Lars Bramsløw, President of the Danish Acoustical Society Dorte Hammershøi, Chair, Prof. Aalborg University

Mo. 9:45 Europahallen Keynote 1

**Health effects of traffic noise: results from a nationwide study**

***M. Sørensen***
*Danish Cancer Society Research Center, Strandboulevarden 49, 2100 Copenhagen ø, Denmark*
*mettes@cancer.dk*

In 2018 the WHO stated that there was sufficient evidence to conclude that road traffic noise is associated with higher risk of ischemic heart disease. For all other investigated diseases, evidence was found to be insufficient. To gain more evidence on the consequences of traffic noise on human health we performed an epidemiological study based on the entire Danish population We estimated road and railway noise (Lden) at the most and least exposed façades for all residential addresses in Denmark during 1990-2017, and linked these with a national registry with address history for all persons in Denmark. Based on this, we estimated long-term noise exposure for 3.6 million persons aged >35 years. Using national hospital, cancer, prescription, and mortality registries, we identified people with cardiovascular diseases, diabetes, cancer, and dementia during follow-up from 2000-2017. We conducted statistical analyses using Cox proportional hazards models adjusted for individual- and area-level sociodemographic covariates and air pollution. We found that road traffic noise was associated with higher risk of cardiometabolic disease, e.g. a 10 dB higher road traffic noise at the most exposed façade was associated with a 4% higher risk of stroke, 5% higher risk of ischemic heart disease, 4% higher risk of heart failure, 10% higher risk of angina pectoris, and 3% higher risk of diabetes. Additionally, road traffic noise was associated with higher risk of breast cancer and dementia, especially noise exposure at the least exposed façade. Although railway noise was not as consistently associated with disease as road traffic noise, we found that it increased risk of heart failure, breast cancer, dementia, and diabetes. Our results suggest that traffic noise is a risk factor of a number of major diseases besides ischemic heart disease, and that the disease burden associated with traffic noise therefore most likely exceeds previous estimations.

Mo. 11:00 Europahallen Effects of noise and noise policy

**Silent City - Europe's First Living Lab for Traffic Noise**

***S. Enevold***
*Gate 21, Liljens Kvarter 2, 2620 Albertslund, Denmark*
*sif.enevold@albertslund.dk*

Around Copenhagen, in the most noise emission plagued areas in Denmark, 11 municipalities and The Capital Region of Denmark have joined forces to improve citizens' health and quality of life. Traffic noise is one of the largest environmental problems in our society. The partnership encourages exchange of experiences across different local policies and geographical spaces. Since the start in 2015, we have helped to put traffic noise on the agenda, and on that basis implemented several concrete initiatives, including a public consultation at the Danish parliament, preparation of a white paper, targeted press efforts and concrete demonstration projects in a Living Lab. The partnership will help to put the participating municipalities and Denmark on the world map within urban noise control.

**The Effect of the New Median Value of the Revised CNOSSOS Method on Population Exposure**

***E. Hartog Van Banda***
*DGMR Software, Casuaristraat 5, NL-2501 CJ Den Haag, Netherlands*
*ha@dgmr.nl*

In December 2020, the EU commission published amendments on CNOSSOS-EU in report ëC(2020) 9101 final'. These amendments are mainly based on the RIVM-2019-0023 report from August 2019. In report ëC(2020) 9101 final', a new additional method 3, based on using a Median value for the calculation of population exposure is also described. In the original CNOSSOS document, 2 methods are described, Method 1 - Most exposed façade and Method 2 - Length of representative façade. Method 1 is used for buildings that represent a single dwelling or 1 dwelling per floor. Method 2 is used for apartment buildings that have dwellings with a single façade exposed to noise. The new Method 3 can be used for apartment buildings that have more than 1 façade exposed to noise, or for apartment buildings where no information on how many facades are exposed to noise is available. It will be difficult for authorities to create a dataset that distinguishes between a single façade exposed to noise and multiple facades exposed to noise. Therefore, it seems likely that Method 3 will be more frequently used. When comparing Method 2 with Method 3, it is found that Method 3 leads to a higher number of people and dwellings exposed to noise. In this paper the 3 methods are explained and compared by using examples.

**Health effects of environmental noise in a wind power area**

***V. Hongisto, J. Radun, H. Maula, P. Saarinen, J. Keränen and R. Alakoivu***

*Turku University of Applied Sciences, Joukahaisenkatu 7, FI-20520 Turku, Finland*

*valtteri.hongisto@turkuamk.fi*

We examined the effects of both wind turbine noise and road traffic noise on symptoms and diseases close to wind turbines and in a control area. Wind turbine sound levels were between 1739 dB LAeq and met the Finnish regulation (40 dB). Daytime road traffic noise levels were 32.5-63.5 dB. Altogether 676 residents responded to the questionnaire. Higher wind turbine sound level was associated with more likely reporting wind turbine noise annoyance and not with reporting of symptoms or chronic diseases. Higher road traffic sound level was associated with higher probability of road traffic noise annoyance, migraine or headache, dizziness, impaired hearing, pressure in ears, tachycardia or heart palpitations, and heart disease. The health effects of wind farms seem to be limited to noise annoyance in areas where all residents are exposed to sound levels under 40 dB. The study has strong practical relevance since new wind power areas always fulfill this 40 dB regulation in Finland. On the other hand, more than 600.000 inhabitants are exposed to road traffic noise levels above 55 dB in Finland. Because high road traffic noise is associated with elevated prevalence of symptoms and a disease in our study, and several previous international studies have found even more serious health effects of road traffic noise, this issue deserves higher concern in the future.

**Developments in International Guidance and Standards for Environmental Noise Assessment**

***D. Manvell***

*DMdB, Teglgårdsvej 52, 2920 Charlottenlund, Denmark*

*douglas.manvell@dmdbsoundadvice.com*

Since its first publication in 1971 of ISO/R 1996:1971 on Acoustics Assessment of Noise with Respect to Community Response, the International Organization for Standardization (ISO) has developed and published a range of standards for environmental noise assessment. The main ISO 1996 series of standards on Description, Measurement and Assessment of Environmental Noise has been developed in phases. And almost 30 years ago, ISO 9613 Attenuation of Sound During Propagation Outdoors was added. More recently, supporting standards such as the ISO PAS 20065 Objective Method for Assessing the Audibility of Tones in Noise and the ISO 17534 series concerning Software for the Calculation of Sound Outdoors have been added, and cooperation initiated with the International Electrotechnical Commission on the IEC 61400-11-2 Assessment of Wind Turbines Noise at receptor positions. This paper provides an overview of ISO standards related to environmental noise assessment. It also describes the historical development of the ISO 1996 series and related standards. And closes with current developments and considerations.

**Uncertainty estimation in environmental road traffic noise measurements using ISO 1996-2:2017**

***S. Domazetovska, M. Anachkova, V. Gavriloski and V. Changoski***

*Faculty of Mechanical Engineering in Skopje, Rugjer Boskovic nn, 1000 Skopje, Macedonia*

*simona.domazetovska@mf.edu.mk*

When measuring the road traffic noise, the environmental noise levels are quantitatively described using the equivalent noise pressure level parameter (Leq). The value of Leq based on the measurements done with sound level meter would probably differ from the true one due to the sources that can cause measurement uncertainty. According to this, the ISO 1996-2:2017 standard proposes calculation of the standard measurement uncertainty, so that the results from the measurements could be more accurate. This paper proposes a guideline on estimating the measurement uncertainty in compliance with the ISO standard for short-term measurements of road traffic noise. To verify the proposed method, twenty short- term measurements are done in same spot in morning and afternoon time period from Monday to Friday for two weeks; and afterwards, the measurement uncertainty is calculated for each measurement. To confirm the accuracy of the results, additional acoustic modeling was made. By comparing the results for the Leq parameter from the measurements and from the acoustic noise maps in accordance with the calculated measurement uncertainty, it could be confirmed the results accuracy. In conclusion, this paper presents a reflection on why estimation of the uncertainty of the measurement is essential in environmental noise analysis.

**Statistical analysis of urban noise measurement data: case study for the city of Skopje**

***M. Anachkova, S. Domazetovska, F. Nikolovski and V. Gavriloski***

*Faculty of Mechanical Engineering, Rugjer Boshkovikj nn., 1000 Skopje, Macedonia*

*maja.anachkova@mf.edu.mk*

Noise from the road transport, particularly from vehicles in urban city areas largely accounts for the general noise level and annoyance of the citizens. The numerous volumes of motor vehiclesflow can be treated statistically, which can establish a deeper insight into the contribution of the road noise to the prevalent noise pollution and its' characteristics. According to ISO 362 and ISO 1996:2, the environmental noise level from traffic is highly dependent on the vehicle category regarding the factor of contribution to the overall urban noise level. The purpose of this study is to analyze the dependence between the number and types of vehicles and measured standardized parameters (Leq, LAF and L95) for noise level assessment by implementing a statistical model analysis of the collected results. The number and the type of the vehicles is obtained from the States' traffic management and control center for a chosen road in the center of the city, whereas noise level measurements have been conducted with a Bruel&Kjaer sound level meter by using a standardized noise level measurement methodology procedure for the selected time period on the given location. This study provides a detailed statistical approach of the collected noise and traffic volume data to in order to obtain conclusions and prediction models for further management of the noise pollution problem in the city.

**Assessment of the proposed changes regarding heavy vehicles in the statistical pass-by draft standard ISO/DIS 11819-1 (2021)**

***M. Geluykens*[a], *A. Barros*[a], *L. Goubert*[b] *and C. Vuye*[a]**

[a]*University of Antwerp, Groenenborgerlaan 171, 2020 Antwerpen, Belgium;* [b]*Belgian Road Research Centre, Boulevard de la Woluwe 42, B-1200 Brussels, Belgium*

*michiel.geluykens@uantwerpen.be*

Exposure to excessive (road) traffic noise can result in sleep deprivation, increased stress, and even heart diseases. Accordingly, accurately determining the acoustic quality of different pavement types is crucial for implementing noise- reducing measures such as low-noise pavements. In this regard, the ISO 11819- 1:1997 standard, which describes the Statistical Pass-By method for assessing the acoustic performance of road pavements, is currently under revision. The proposed changes include combining two vehicles' categories, currently defined as dual- axles (HD) and multiple-axles (HM) heavy vehicles, into a single category: heavy vehicles (H), by adding 2.7 dB to the maximum sound pressure level (LAmax) of HD vehicles. Additionally, a new method is presented to define the noise versus speed relation of this new caterory. In this work, these changes were evaluated based on a large dataset collected by the authors over 8 months, with an abundance of heavy vehicles (2820) while also accounting for passenger cars (2896). The average difference in LAmax between HD and HM vehicles ranged between 2.1 - 2.3 dB, meaning that the draft standard may overestimate this difference. Furthermore, while the draft standard prescribes a coefficient of 25 for the LAmax variation with speed for the road surfaces here investigated, we observed a lower speed dependency, with a coefficient of 20.5 on average. Lastly, we proposed and investigated the introduction of a new vehicle category containing delivery vans. It appears that vans can form a vehicle category with consistent noise behaviour and could thereby aid in describing the pavements' acoustic performance, ultimately contributing to reducing the measurement duration. The noise produced by vans presents a similar spectra shape and coincident noise levels as passager cars at 1000 to 2000 Hz third-octave bands.

**Relationship between tyre-road noise and temperature under noncontrolled traffic flow conditions**

***M. Sánchez Fernández, D. Montes Gonzalez, J.M. Barrigón Morillas, P. Atanasio Moraga, G. Rey Gozalo, R. Vílchez Gómez and A. Bachiller León***

*Universidad de Extremadura, Av. Elvas, s/n, 06071 Badajoz, Spain*

*davidmg@unex.es*

The influence of temperature on tire-road noise level is a topic on which many investigations have been carried out on controlled road traffic flow conditions based on international standards. Its practical application under uncontrolled vehicle flow conditions may be of interest for the assessment and management of road traffic noise through strategic noise maps. This paper presents the results of an investigation carried out by means of in situ measurements of road traffic noise levels and environmental temperature and a discussion is conducted with respect to those findings available in the scientific literature. A negative and highly significant dependence relationship of the road traffic noise level with both pavement and air temperature was found on the two days of measurement, with similar explanations for variability in the range 59-75% for both temperatures but with differences of around 12-14% between the two sets of measurements. When the slope coefficient is compared, increases of approximately 40% for pavement temperature and 36% for air temperature are found from day 2 to day 1. While the values of the coefficient of variation of noise level with temperature range from -0.05 to -0.07 dBA/°C for pavement temperature, they are in the range of -0.14 to -0.19 dBA/°C for air temperature. These values are similar to those published in the scientific literature for pavement temperature, but higher than those reported for air temperature under controlled traffic flow conditions. The followed methodology could be applicable in different conditions.

**Validation Highlights of Nordic and European Road Traffic Noise Standards along a Major Highway in Central Jutland, Denmark**

***J. Khan*[a], *S.S. Jensen*[a], *C. Backalarz*[b], *E. Thysell*[b], *P. Finne*[b] *and O. Hertel*[c]**

[a]*Department of Environmental Science, Aarhus University, Frederiksborgvej 399, 4000 Roskilde, Denmark;* [b]*FORCE Technology, Venlighedsvej 4, 2970 Hørsholm, Denmark;* [c]*Department of Ecoscience, Aarhus University, Frederiksborgvej 399, 4000 Roskilde, Denmark*

*jibran@envs.au.dk*

According to the Danish Environmental Protection Agency (EPA), roadway noise is the main source of noise annoyance. Furthermore, almost one in three Danish homes, and approximately 785000 homes, suffer from noise levels above the recommended limits. Assessment of noise levels is, therefore, indispensable. This study presents validation highlights of Nordic road traffic noise standards, Nord2000 and Road Traffic Noise 1996 (RTN-96), and the European standard, Common Noise Assessment Methods for EU Member States (CNOSSOS-EU), along a major highway (E45) in Helsted, Central Jutland, Denmark. High-quality noise measurements, as per ISO 1996-2 (2017), were performed along E45 at two measuring locations in September - November 2019. In addition, traffic attributes, number of light, medium and heavy vehicles and respective speeds, were also measured. Further, meteorological data (wind speed and direction, temperature, etc.) was also obtained from a nearby meteorological station as well as Denmark's Meteorological Institute (DMI). In short, after data cleaning and imputation, 77 days of measured hourly A-weighted, energy equivalent, sound pressure levels (LAeq,1h) data was available (N = 1834). Measurements results reflect significant variation in LAeq,1h levels at both measuring locations, Receiver 1: range = 56.3 - 78.7 dBA, median = 72.6 dBA; Receiver 2: range = 51.9 - 73.5 dBA, median = 68 dBA. The measured noise levels are compared with the modelled ones (Nord2000, RTN-96, CNOSSOS) using SoundPLAN software, version 8.2. Preliminary results suggest that Nord2000 performs relatively better. The short article and presentation will provide further details of ongoing validation work as well as its strengths, limitations and future prospects.

**Objective measurements and subjective assessment of the speech intelligibility in rooms with an Active Acoustic Enhancement Systems**

***L. Blasinski, A. Pastusiak, J. Kocinski and M. Buszkiewicz***

*Chair of Acoustics, Faculty of Physics, AMU, Uniwersytetu Poznańskiego 2, 61-614 Poznan, Poland*

*lukasz.blasinski@amu.edu.pl*

Speech Transmission Index (STI), apart from the Reverberation Time (RT), remains one of the most important parameters used for the speech intelligibility assessment indoors. Achieving it's sufficient values is crucial not only in classrooms and offices but also at cultural centers like drama theaters, opera theaters and multipurpose cultural halls. As the Active Acoustic Enhancement Systems (AAES) are getting more and more commonly used, the authors decided to verify how such systems affect speech perception. Firstly, in accordance with the 3382-1 and 3382-2 regulations, objective parameters of room acoustics were measured in two rooms equipped with the AAES. The RT values varied from 0.91 to 1.61 [s] for the hall in Mińsk Mazowiecki and from 0.75 to 1.52 [s] for the hall in Puławy. The value of the STI parameter varied from 0.57 to 0.64 in Mińsk Mazowiecki and from 0.64 to 0.69 in Puławy. Despite the significant differences in the RT values, STI coefficient values remain almost unchanged. The auditory test was designed to verify if the subjectively assessed intelligibility would be compatible with the results of the objective measurements. Speech material was obtained by convolving 50-element nonsense word (logatome) lists with the room's impulse responses. Three impulse responses were used: without AAES and with two different AAES setups. Logatome recognition was calculated as a percentage of correct responses. All of the 120 listeners had normal hearing, according to WHO. In summary, subjective measurements' results were compared with the acquired objective STI as well as RT and parameters describing Clarity (C50 and C80) to prove their importance in indoors speech intelligibility evaluation.

**Objective evaluation of multipurpose enclosures equipped with active acoustic enhancement systems.**

***L. Blasinski***

*Chair of Acoustics, Faculty of Physics, AMU, Uniwersytetu Poznańskiego 2, 61-614 Poznan, Poland*

*lukasz.blasinski@amu.edu.pl*

From the acoustical point of view various events requires different acoustic conditions. Research on optimal acoustic parameter values that have been carried out for recent decades and the necessity to provide different acoustic solutions to events of a different type (e.f. symphonic concert or speech), led to construction of rooms with variable acoustic conditions. In such rooms changes in parameters in multipurpose enclosures may be achieved by modifications of volume or a shape of a room as well as through the use of appropriate acoustic treatments. However, one must bear in mind that those modifications are limited and frequently insufficient due to the technical limitations. Therefore, nowadays, active acoustic systems based on signal processing and multisource redistribution of a modified signal in a room become more and more popular. Application of such systems allows adjustments of acoustic parameters required for particular performance. This paper presents results from measurements of objective parameters of multipurpose enclosures equipped with active acoustic enhancement systems. Investigated halls are located in Poland in the following cities: Łańcut, Mińsk Mazowiecki, Puławy, Wieluń and were intentionally built or renovated as multipurpose rooms. All of those were equipped with Yamaha Active Field Control version 3 system. The measurements were performed according to ISO338-1/2 standards. In total 13 active acoustic system settings (presets) were analyzed to investigate their influence namely Reverberation Time (RT), Early Decay Time (EDT), Bass Ratio (BR), Treble Ratio (TR), Initial Time Delay Gap (ITDG), Center Time (Ts), Clarity (C80, C50), Definition (D50), Interaural Cross Correlation (IACC) and Speech Transmission Index (STI). The analysis of the obtained results showed that it is possible to obtain the appropriate acoustic conditions with acoustic parameters values set in the suggested range for a given event using active acoustic systems.

**Solving structure borne noise problems from car wash facility**

***J. Torres and A. Buen***
*Brekke Strand Akustikk, Hovfaret 17, 0275 Oslo, Norway*
*jto@brekkestrand.no*

A car wash facility is a complex system with many noise sources. When the car wash is on the ground floor of a residential building, there are frequently reported noise complaints. The number of complaints usually increases when the car wash service is open 24 hours, and it is constantly in use during the day. This paper shows how to predict propagation of structure borne noise and airborne noise inside a residential building. It is described different measures, not only to solve structure borne noise problems from a car wash facility, but also to reduce airborne noise. It is presented a case study were structure borne noise was the main noise source. In Norway, noise limits for car wash facility installed in the same building should not exceed a sound level of 25 dB inside the residential apartment (Lp,A,T $\leq$ 25 dB). This limit is quite difficult to achieve in some situations.

**Online monitoring using vibrations of fluid density in a pipe in the pulp industry**

***F. Rietdijk***

*Acosense AB, Stampgatan 20A, 41101 Gothenburg, Sweden*

*frederik.rietdijk@acosense.com*

In the pulp industry significant amounts of energy and water are used in the kraft process where wood is converted into pulp. This is done through heating and addition of certain salts. Stable processes are desired to improve efficiency, which requires the monitoring of certain physical properties of various flows. Examples of these are density and dry contents of green and black liquor. Due to the high content of dissolved salts inline instrumentation faces build up on sensors very quickly. Regular cleaning of sensors must be done which can lead to process downtime. Hence a method to measure the properties from the outside is preferred. By exciting the pipe carrying the flow and observing the resonance frequencies it is possible to estimate properties such as density and dry contents from the outside without requiring downtime.

**Subjective and objective evaluation of the tone-to-tone timbre variability of historical flute designs.**

***M. Haverkamp***

*Independent Scientist, Bahnwärterweg 76, 50733 Köln, Germany*

*michael.haverkamp@netcologne.de*

During the eighteenth century, the variability of timbre of music instruments in terms of a ëtone colour' was essential to enhance the emotional involvement of the audience, the naturalness of expression and the language-likeness of instrumental sounds. An expression of multi-sensory content by ëpainting with music' was popular. Timbre is the essential base of the colourfulness of music. Given that flutes of that era show a more simple design than the modern flute, specific fingering patterns are thus needed for the full chromatic scale, which causes a characteristic variation of timbres from tone to tone. This study compares the timbre variability of recorder and transverse flute with the modern Boehm flute.

Possibilities of subjective comparison are discussed. Perceived timbre is seen from a phenomenological point of view. It combines a pitch-dependent portion with effects caused by the distribution of partial tones and formants. For the subjective approach, the pitch dependent portion of timbre perception is removed. This enables subjective evaluation of influence of partial tones on timbre perception.

Objective acoustic analysis of spectra clarifies the associative content with a view to similarities with other instruments. Colour scales are used for a brief overview on the tone-to-tone variability of an instrument. Such colour scales prove to be beneficial to qualitatively document the associative content and its variability in the sense of ëtone colour'.

The results indicate that the total sound of each instrument is based on a variety of spectral properties/timbres rather than unique properties for all tones. Surprisingly, this also applies to the modern flute that has been developed for similarity of perceived tone quality throughout the scale. Each flute type shows variable references to sounds of other instruments. Timbre variation is not an error state, but it essentially contributes to the music aesthetics of the Renaissance and Baroque era.

**Interdependencies of Humming, Rumbling and Booming**

***F. Doleschal, G.-T. Badel and J. Verhey***

*Otto von Guericke University Magdeburg Dep. of Exp. Audiology, Leipziger Str. 44, 39120 Magdeburg, Germany*
*florian.doleschal@med.ovgu.de*

Humming, rumbling and booming are sensations that require the presence of low-frequency tonal components in the noise. If it only contains a low-frequency tone the sound is perceived as humming, if this component is modulated it is perceived as rumbling and if more tonal components with higher frequencies are present that are related to the low-frequency tonal component (e.g. are harmonics), the sound is perceived as booming. Sounds eliciting these sensations are often observed in vehicle acoustics when downspeeding the engine, which is a strategy to reduce fuel consumptions in vehicles with combustion engines. The present study investigates how these sensations interact using original sounds from the vehicle interior and sounds where parts of the spectrum are altered to change the magnitude of the sensations. The results show that the three sensations correlate in some aspects. Furthermore, dominance effects have been observed, so that measures to manipulate a certain sensation do not take effect if another sensation is much more pronounced.

Mo. 15:15 Musiksalen Sound quality

**The sound of coffee. A predictive model for sound quality of espresso machines**

***L. Rossi*[a], *C. Acanfora*[a], *M.C. Vola*[a], *A. Palmese*[a] *and C. Carbonini*[b]**

[a]*Capgemini Engineering, Strada del Drosso 33, 10100 Torino, Italy;* [b]*Lavazza Spa, Strada Settimo 410, 10100 Torino, Italy*
*laura.rossi@capgemini.com*

The optimization of sound quality of home appliance is a continuous challenge for designers and engineers as it reflects the direct requirement of final users. Customers are looking for acoustic comfort in their homes and this comes from the choice of equipment characterized by the emissions of not disturbing and pleasant sounds. During the development of a new product is therefore becoming more and more important to understand the user expectation in terms of perceived quality and define a feasible target. Sound quality tools and psychoacoustics metrics can help in objectifying the subjective perception. In our work, focused specifically on the noise emitted by domestic espresso coffee machines, we develop a mathematical model that allow the prediction of sound quality of coffee brewing. The SQI (Sound Quality Index) value corresponds to the evaluation that a human jury would express if called to assess the sound submitted to the model. The model has been developed using Machine Learning supervised techniques aimed to predict the perceived sound quality, based on acoustic measurements of a huge set of machines. The validation phase highlighted the goodness of the model, comparable with the precision of a human evaluator. The SQI can be used with data coming from acoustic measurements on prototypes as well as fully developed products. The possibility to rapidly evaluate sound quality, also in early stages of product development, can be a great advantage in the user centred design process.

**Experimental evaluation of earplug behavior in front of high- level impulse noises.**

***C. Blondé-Weinmann*[a], *T. Joubaud*[a], *P. Hamery*[a], *S. Demezzo*[a], *V. Zimpfer*[a] *and S. Roth*[b]**

[a]*French-German Research Institute of Saint-Louis, 5 Rue du Général Cassagnou, 68300 Saint-Louis, France;* [b]*Laboratoire Interdisciplinaire Carnot de Bourgogne, Site UTBM, rue de Leupe, 90400 Sevenans, France*

*cyril.blonde-weinmann@isl.eu*

High-level impulse noises such as weapon noises or mine charges can cause irreversible damage to the hearing system. Hearing protectors are used to reduce airborne sound propagation. However, the resulting attenuation of protectors worn singly or in combination is limited and may be insufficient in some extreme situations. One reason for these limitations is the behavior of protectors under high solicitations. For example, high-level impulse noises could induce a slight earplug movement in the ear canal. In order to quantify this effect, a new experimental set-up with an artificial simplified ear canal is developed. Thus the earplug movements are measured with a laser vibrometer and a high-speed camera under various configurations (high impulse stimulation levels, ear canal lengths, earplug adjustments, and positions). These investigations highlight relative displacements that can exceed one millimeter and modify the final earplug position in the ear canal. The observed effects could be responsible for an alteration of the protection efficiency. New hypotheses are formulated to limit these phenomena and improve hearing protectors.

**Perception of Virtual Reality Based Audiovisual Paradigm for People with Hearing Impairment**

***K. Sun, N.H. Pontoppidan, D. Wendt and L. Bramsløw***

*Eriksholm Research Centre, Oticon A/S, Rørtangvej 20, 3070 Snekkersten, Denmark*

*knsu@eriksholm.com*

Integrating new and emerging technologies, such as virtual reality (VR), with established test methods to improve the ecological validity is gradually used by practitioners in some fields (e.g., soundscape research) but remains limited use in hearing science. In this paper, an audiovisual setup was introduced to administrate an augmented speech-in-noise test for people with hearing impairment (PHI). The environment stimuli containing four competing talkers was recorded with 360-degree video and ambisonics audio. In the scene recreation, the video was displayed in the VR headset, while the audio was presented to the participants via loudspeaker array. Furthermore, a (silent and fixed) physical avatar was included in the video recording as a placeholder for the audio stream of the target speech which was added into sound presented over the loudspeaker array. This setup was used to test the effect of different hearing aid (HA) settings under the same condition for each participant. In total, 27 PHI participants were tested.

The simulator sickness questionnaire (SSQ, Kennedy et al., 1993) was used to quantify the sickness measurement. The data showed a low degree of nausea and disorientation, but scattered oculomotor responses. Moreover, a general questionnaire assessing scene recreation, test method and outcome expectation was administrated. In general, the audiovisual system received high appraisal in realism; the augmented speech-in-noise test method was well accepted; participants highly agreed that the difference between programs could be distinguished. However, the sense of physical immersion decreased due to the weak binding between the avatar and the target speech. Furthermore, when comparing the three components (nausea, disorientation, oculomotor) in SSQ and items in the general questionnaire, the oculomotor was found significantly correlated to the perceived binding of target speech and the avatar. Specifically, participants who were less convinced that the speech came from the avatar, also rated the oculomotor higher.

**Perceptual evaluation of auralizations from a wave-based method in a virtual environment**

***T. Pathre*[a], *M. Hornikx*[a] *and A. Kohlrausch*[b]**

[a]*Eindhoven University of Technology, Department of the Built Environment, P.O. Box 513, 5600 MB Eindhoven, Netherlands;* [b]*Eindhoven University of Technology, Human-Technology Interaction, P.O. Box 513, 5600 MB Eindhoven, Netherlands*
*t.u.pathre@tue.nl*

In recent years, Virtual Reality (VR) has become a powerful tool in studies involving perceptual evaluation of auralization methods. The aim of this study was to investigate the extent to which binaural auralizations from a wave-based method are perceptually close to those from measurements for two different room acoustical scenarios. The signals used for auralization are simulated using the Discontinuous Galerkin method. The simulations encompass geometric details of the room, material properties of the objects in the room and receiver directivity including head orientation. The measured Binaural Room Impulse Responses (BRIRs) were limited to an upper frequency of 2.5 kHz to match the bandwidth of the simulations. The objective results show a fairly good agreement between the measured and simulated data, thereby indicating that the wave-based method can be held as a reference for achieving perceptually realistic auralizations. To evaluate this, a listening experiment was carried out in an interactive VR environment employing a 3D model of the room integrating a dynamic convolution framework with headphone reproduction. The BRIRs were convolved in real-time with two types of source signal - percussion and female speech. Twelve normal hearing participants performed a rating task for judging differences between measured and simulated BRIRs for four perceptual attributes - reverberance, clarity, bassiness and externalization. The results show that the mean difference between measurement and simulation is statistically significant for reverberance, clarity and bassiness but not for externalization. Additionally, this study highlights a state-of-the-art experimental framework which can be extended for perceptual evaluation studies independent of the computational method used to derive the auralizations.

Mo. 13:15 Gæstesalen Audio signal processing

**Effect of Wireless Transmission Errors on Sound Zone Performance at Low Frequencies**

***C.S. Pedersen*[a], *M.B. Møller*[b] *and J. østergaard*[a]**

[a]*Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark;* [b]*Bang & Olufsen A/S, Bang & Olufsen Allé 1, 7600 Struer, Denmark*

*cp@es.aau.dk*

Sound zones can provide different audio content to several persons in the same room by creating separate zones of individual audio content with minimal interference between the zones. Optimal strategies for creating sound zones depend on the wavelength of the sound. For low frequencies it is advantageous to distribute a number of woofers in the room and use sound field control based on measured transfer functions. For such a distribution of woofers it is desirable to stream the audio signals wirelessly to the woofers, but wireless streaming to multiple receivers can be challenging since transmission artefacts may occur. As the sound field control depends on the superposition of the sound from all the woofers, any errors in the reproduced sound will not only affect the audio quality, but also have a negative effect on the sound zone generation. This study investigates relevant types of degradation that transmission errors can have on the sound zone performance for a specific sound zone setup with eight woofers. Errors such as transmission delays and data dropouts are simulated. It is demonstrated that all these transmission artefacts lead to decreased contrast between the sound zones which is mainly caused by increased leakage to the dark zone. The decrease in contrast depend on which woofer is affected and the type of transmission artefact, and for continuous data dropout it ranges from 3.3 to 17.9 dB reduction in the mean contrast between the sound zones. A mitigation strategy where the affected woofer is excluded from the sound zone control shows promising results with slightly lower contrast than the error-free system (mean contrast reduction of 0.5 to 2.9 dB depending on woofer), but with much better performance than the system affected by transmission errors.

Mo. 13:45 Gæstesalen Audio signal processing

**The Effect of Fixed-Point Arithmetic on Low Frequency Sound Zone Control**

***P. Koch***[a] ***and J. østergaard***[b]

[a]*Aalborg University, Dept. Electronic Systems, Fredrik Bajers Vej 7B, 9220 Aalborg ø, Denmark;* [b]*Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*pk@es.aau.dk*

To generate sound zones, a number of loudspeakers collaborate with each other and the enclosed space to create constructive and destructive acoustic interference leading to spatial zones with high (bright zone) and low (dark zone) sound pressure levels, respectively. The signals to be played out by the loudspeakers are commonly created by filtering the audio signals using Finite Impulse Response (FIR) filters. In this work, we are interested in using finite precision multiply- accumulate arithmetic in the FIR filter operations in order to reduce the overall computational complexity, thus potentially minimizing the execution time as well as the power consumption, and the physical size of a real-time hardware implementation. Our target hardware platform is FPGA-based, enabling the possibility to adjust accordingly the word lengths of the multiplier and the adder. In a simulation study based on a setup with 8 subwoofers and for the frequency range 50-600 Hz, we demonstrate that it is sufficient to use 8 and 12 bits signed fixed-point arithmetic for the multiplier and adder, respectively, for generating two sound zones. Specifically, we observe a reduction in the ratio of sound pressure levels between the bright and the dark zones of approximately 0.1 dB compared to the performance obtained using a 32-bit reference fixed-point arithmetic. Moreover, we observe that reducing the word length of the adder below 12 bits, significantly increases the power spectral density of the dark zone. A similar trend is not observed for the multiplier. Thus, to create low frequency destructive interference for sound zone applications, the 2's complement fixed-point word length of the adder needs to be no longer than 12 bits. As compared to most commercially available fixed-point DSP-processors based on a 16-bit architecture, using a reconfigurable hardware platform thus paves the way for a complexity reduction of the arithmetic units.

**A comparison of audio-to-tactile conversion algorithms for melody recognition**

***R. Paisa***[a]***, J. Andersen***[b]***, N.C. Nilsson***[a] ***and S. Serafin***[c]

[a]*Aalborg University, Lyongade 7, 1TH, 2300 Copenhagen S, Denmark;* [b]*The Royal Danish Academy of Music, Rosenørns Alle 22, 1970 Frederiksberg, Denmark;* [c]*Aalborg University, AC Meyers Vaenge 15, 2450 Copenhagen, Denmark*

*rpa@create.aau.dk*

Besides language, music is one of the two major acoustic channels for expression of human nature and it is ubiquitous to all cultures. Due to a tight correlation between auditory and haptic stimuli, more and more attention is focused on the importance of the latter sensation in a musical context. For the hearing impaired especially, tactile feedback has been investigated extensively for it's musical applications and hearing assistive devices, as early as 1983. This study compares three common audio-to-haptic signal processing algorithms designed for full range vibrotactile transducers used for tactile augmentation of music. The focus is on melody discrimination over three instruments: double bass, digital subtractive synthesizer with a sawtooth oscillator and trumpet. The transducer used is a high fidelity Tactuator BM3C, enclosed in a custom anatomical handheld case, inspired by an orthopedic resting hand splint. An evaluation was conducted on 34 participants and used a within-group design with three alternative forced choice task assessing the participants ability to match melodies to tactile stimuli. The results indicate that no algorithm performs better than others, which is in line with the literature regarding the overall poor frequency discrimination of the skin. Nevertheless, post experiment interviews suggest that some participants perceived multiple frequencies simultaneously, on different areas of their hand, similar to auditory polyphony.

Mo. 15:15 Gæstesalen Technical audiology

**A data-driven distance metric for evaluating the effects of dynamic range compression in adverse conditions**

***N. Overby, T. Dau and T. May***

*Technical University of Denmark, Department of Health Technology, 2800 Kgs. Lyngby, Denmark*

*niov@dtu.dk*

Dynamic range compression aims to restore audibility for hearing-impaired listeners and is one of the most essential building blocks in modern hearing aids. However, the choice of suitable compression parameters, such as the time constants associated with the level estimation stage, depends on the acoustic conditions, and the perceptual benefit of different parameter configurations is still controversial. Listening tests can provide an accurate assessment of the perceptual effects of compression in a limited set of acoustic conditions, but they are time-consuming and can therefore not be used to optimize the various compression parameters across experimental conditions. While several studies have attempted to link the perceptual outcomes of dynamic range compression to a set of objective metrics, there is no agreement on how to objectively quantify the effects of compression. In the current study, a data-driven distance metric was developed based on objective metrics to analyze different compression systems. This analysis included slow- acting, fast-acting, and ëscene-aware' compression that adaptively switched between fast- and slow-acting compression depending on the target source activity. In addition, a reference system termed ësource- independent compression' was considered that had access to the individual speech and noise signals. A comprehensive list of objective metrics was considered to evaluate the effect of the different compression systems in a wide variety of acoustic conditions, including both interfering noise and room reverberation. Factor analysis was then applied to derive a compact set of interpretable features that explained the effects of compression as linear combinations of sparsely selected objective metrics. The Euclidean distance, within the reduced dimensionality representation, was used to compare the similarity between the compression systems. This newly developed distance metric allows a systematic analysis and optimization of the parameters of dynamic range compression systems by minimizing the Euclidean distance with respect to the source-independent compression system.

**Conception of the listening test procedure for quantifying speech intelligibility in Slovak language - a preliminary study**

***D. Húdoková*[a], *V. Chmelík*[a], *L. Zelem*[a], *D. Urbán*[a] *and M. Rychtarikova*[b]**

[a]*Slovak University of Technology in Bratislava, Radlinského 2766/11, 810 05 Bratislava, Slovakia;* [b]*KU Leuven, Celestijnenlaan 200 D, 3000 Leuven, Belgium*

*dominika.hudokova@stuba.sk*

This paper describes the methodology used for the conception of the Slovak sentence listening test, suitable for speech reception thresholds (SRT) experiments in quiet and noise. The procedure on choose of suitable sentences in Slovak language is addressed and method on sorting them according to their intelligibility is described. Results shown in this paper are based on a preliminary headphone-based listening test experiments performed on 15 normal hearing female and male human subjects. The task of each respondent was to repeat presented sentence in noise, with individualized signal to noise ratio, tuned for each test person in a pre-test by means of the method of adjustment. The application of the newly developed well- balanced sets of sentences in Slovak language is twofold. Its direct usage can be seen in audiological experiments with severely hearing-impaired individuals and cochlear implant users with Slovak as native language. Secondly, it can serve as a tool for validation of speech related room acoustic quantities, such as STI and D50 or U50 for Slovak language.

**Situations where users always wear or take off their hearing aids**

***S.K. Narayanan*[a], *S.S. Houmøller*[b], *A. Wolff*[c], *L.-T. Tsai*[d], *K. Lund*[a], *D.D. Hougaard*[c], *M. Gaihede*[c], *J.H. Schmidt*[b] *and D. Hammershøi*[a]**

[a]*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark;* [b]*Research Unit for ORL - Head & Neck Surgery and Audiology, University Hospital &University of Southern Denmark, 5000 Odense, Denmark;* [c]*Department of Otolaryngology, Head & Neck Surgery and Audiology, Aalborg University Hospital, 9000 Aalborg, Denmark;* [d]*Department of Aging Medicine and Aging Research, University Hospital Zurich and University of Zurich, 8091 Zurich, Switzerland*

*skn@es.aau.dk*

It is well known that hearing aids (HAs) are not worn by HA users all the time. Understanding different situations in which the users always wore or took off their HA can provide insight into their exposed soundscapes and preferences. This can also provide knowledge of the shortcomings in the functional and user experience aspects of the HAs. The data collected from 1487 HA users in the Better hEAring Rehabilitation (BEAR project) is used in this study. Users were asked two separate questions as part of the one-year online follow-up; 1) if in any situation they always took off their HAs, and 2) if in any specific situation they always wore their HAs. If they answered ”yes” to these questions, ten choices of situations were presented: watching television, shopping, walking, cycling, in church, when using the telephone, during driving, at home, at work, and others. If the user chose ”other,” they were asked to elaborate. The result showed that more than 57% of the users took off their HAs in one or more of these situations. From the responses to the choice of situations, it was found that most of the HA users preferred to take off their HAs during physical activities. HA users preferred always to wear their HAs at work and while watching TV. The significance of the acoustics of the environment versus user-related factors will be further explored.

**Innovation Loam Noise Barrier**

***M. Chudalla and F. Strigari***

*Federal Highway Research Institute, Brüderstraße 53, D-51427 Bergisch Gladbach, Germany*

*chudalla@bast.de*

Cob (in German ”Wellerlehm”) and stamped loam are traditional earthen construction methods that have been used in Europe for hundreds of years. With a pilot project near Nebelin in Brandenburg, this forgotten construction technology, which can now be considered innovative, is to be applied to a noise barrier at a planned service area on the A14 motorway. The project is called the ”Alhambra of Brandenburg”. The aim is to determine the necessary boundary conditions for the construction of noise protection walls in earthen construction, which meet both the technical requirements of the German ”ZTV-Lsw” (Additional Technical Contractual Conditions for noise barriers) as well as ecologically sensible criteria.

Due to the wall thickness of a loam noise barrier of up to one metre and the density of the building material alone, achieving a sufficient airborne sound insulation does not represent a big problem. However, to improve the absorbing properties, the dense and sound-reflecting surface of the cob must be supplemented by a kind of facing shell. Promising results in this regard are shown by initial measurements in the impedance tube with Heraklit and hemplime, a composite material made of hemp, light wood and lime binder. In further test series, the material compositions of the sample bodies are varied in order to optimise the absorption spectrum to the road traffic spectrum.

It is hoped that the natural and locally occurring materials will contribute to a reduction in energy consumption and, due to the high content of renewable raw materials, a favourable CO2 balance and a reduction in greenhouse gas emissions. Furthermore, the purely naturally occurring raw materials make the building completely recyclable.

**Measurement of the acoustic effectiveness of diffractors**

***F. Strigari, R. Becker and W. Bartolomaeus***

*Federal Highway Research Institute, Brüderstraße 53, D-51427 Bergisch Gladbach, Germany*

*strigari@bast.de*

One possible approach for innovative noise protection from road traffic noise is to use the diffraction principle to cause an upward diffraction of the sound. Technically, this involves periodic grid structures with resonance chambers of different depths. These produce a reduction effect that extends over a broadband frequency range. Diffractors can be embedded in the side space next to the road or mounted on a noise barrier.

In the present study, the acoustic effectiveness of two diffractor types was determined by means of controlled and statistical pass-by measurements. In order to ensure the best possible quantification of the noise reduction potential, the measurements were carried out on two times four microphones, simultaneously at the diffractor and at a reference. An equivalent noise barrier of the same height or the propagation over free field served as reference. The simultaneous measurements at the diffractor and the reference allow a direct comparison of the same emission source with almost identical emission strength and characteristic. The evaluation of the level differences shows a significant and height-dependent additional reduction effect by the diffractors. The analysis of the one-third octave bands illustrates the working principle of the diffractors and their optimisation for the road traffic noise spectrum.

In addition to classifying the acoustic effectiveness of the investigated diffractors, the general suitability of the measurement principle is also discussed. The results moreover provide an important input for possible approaches to consider diffractors in the calculation of sound propagation along roads, as presented in the contribution ”Simulation of the sound field behind diffractors”.

**Simulation of the sound field behind diffractors**

***W. Bartolomaeus, R. Becker and F. Strigari***

*Federal Highway Research Institute, Brüderstraße 53, D-51427 Bergisch Gladbach, Germany*

*bartolomaeus@bast.de*

In analogy to optical reflection gratings, acoustic diffractors cause an upward diffraction of the sound due to their periodic structure. They can be recessed in the roadside space or mounted on a noise barrier. The prerequisite for their effectiveness, the coherence of the incident sound, is fulfilled for passing vehicles at small angular differences and not too high frequencies. By means of chambers of different depths, a spectral broadband effect of the diffractors is achieved. Based on the results presented in the article ”Measurements of the acoustic effectiveness of diffractors”, the question arises how the acoustic effect of diffractors could be taken into ac-count in the calculation of sound propagation from roads. For this purpose, the propagation of traffic noise over flat terrain described by Rasmussen already in 1982 was extended with an impedance jump road - grassland and applied to the situation with diffractors. By comparing the sound signals of pass-by measurements recorded at different positions, it should be possi-ble to model the acoustic effect of an additional strip of diffractors placed next to the road by a corresponding strip with a frequency-dependent acoustic impedance. The transformation of the results obtained here into the purely energetic approach of ISO 9613-2 for sound propagation outdoors is certainly not easy. An indirect possibility would be the introduction of a frequency-dependent vertical directivity of the sound sources, which takes the effect of the diffuser into account.

Tu. 10:30 Europahallen Keynote 2

**Deep machine learning in audiology: Models of speech perception and self-conducted speech audiometry**

***B.T. Meyer***
*Carl von Ossietzky University, Carl-von-Ossietzy-Str. 9-11, 26129 Oldenburg, Germany*
*bernd.meyer@uni-oldenburg.de*

Deep machine learning resulted in a major performance boost in many research fields such as computer vision, natural language processing, and automatic speech recognition (ASR). In my talk, I will provide examples how this technology - specifically ASR based on deep learning - can be exploited in the context of audiology and speech enhancement. First, models of speech perception will be introduced that aim at predicting speech intelligibility, as well as the perceived speech quality, and the subjective listening effort for normal-hearing and hearing-impaired listeners. At their core, phoneme probabilities from a deep neural network are calculated; the degradation of these probabilities in the presence of noise, reverberation or other distortions is quantified, which results in the model output. In some cases, these algorithms outperform baseline models even though they operate on a mixture of noise and speech - in contrast to other approaches that often require separate noise and speech inputs. This implies a reduced amount of a priori knowledge for these algorithms, which could be interesting for applying them in the context of hearing research. The models were trained with hundreds or thousands of hours of speech and are harder to analyze in comparison to established models; yet they are not black boxes since we have various methods to study their properties. Another exciting application of deep learning is the enhancement of speech signals by noise suppression or speaker separation, which will be briefly covered. Finally, noise-robust ASR could be used for conducting listening tests where the subject listens to a noisy sentence and responds with the words he or she recognized. I will present some of our results for speech audiometry using an ASR system in the clinic and briefly introduce an Alexa skill for performing a screening procedure in the living room.

**Shore power connection for offshore vessels - Measured noise reduction in port and dock**

***B.M. Larsen***

*Multiconsult Norway, Fjellgata 6, 4612 Kristiansand S, Norway*

*bernt.mikal.larsen@multiconsult.no*

The presentation will summarize measured noise reduction for drilling, plumbing and offshore support vessels due to shore power connection. The noise level at office or storage building in distance of 100-250 m from the vessels without significant influence from other background noise has been registered continously through days and nights. Noise levels at nights without and with shore power connection has been compared. A drilling vessel had Lw 110 dB and a noise reduction due to shore power connection of 18 dB. A plumbing vessel had Lw 105 dB without shore power, and a noise reduction of 15 dB due to the effect of shore power. Both drilling and plumbing vessel seem to have Lw of 90-92 dB with shore power connection. Two different offshore support vessels have also been measured with sound power level Lw of 107 dB and 100 dB. The noise reduction with shore power connection is 13 dB for the first and 6-7 dB for the latter. With shore power connection, both offshore suppert vessels have Lw of 94 dB. The two offshore support vessels show that shore power can give different noise reduction for ships with different design, and that the sound power level with shore power connection seems to be the same within a category of vessels, due to the fact that ship noise level from ventilation and fans seems to be the same. All of the vessels measured have sound power level between 90 and 95 dB with shore power connection.

**Applicability of ISO standard 3744 to UA**

***J. Treichel*[a], *J. Foerster*[b], *A. Volkert*[c] *and T.J. Lieb*[c]**

[a]*Umweltbundesamt, Wörlitzer Platz 1, 06844 Dessau-Roßlau, Germany;* [b]*Technische Universität Berlin, Marchstr. 12-14, 10587 Berlin, Germany;* [c]*Deutsches Zentrum für Luft- und Raumfahrt, Lilienthalplatz 7, 38108 Braunschweig, Germany*
*julia.treichel@uba.de*

Unmanned Aircraft Systems (UAS) are used for a variety of purposes. Especially the industrial or professional use of UAS will lead to an increasing number of possible applications. The steadily rising number of drones raises the question of noise impact from these unmanned aircrafts. For the first time, an EU regulation provides a uniform noise rating for drones. This involves the introduction of a label for the guaranteed sound power level. This level is to be determined via EN ISO 3744:2010 by means of an enveloping surface method. Manufacturers are required to document the guaranteed sound power level as part of their CE marking. In addition, the EU regulation specifies a maximum permissible sound power level. The permitted level depends on the weight of the drone. The German Environment Agency has therefore started with acoustic investigations of drones. Various small UAS models (multicopter) were used in the measurements in accordance with the EU regulation. This paper presents the results of the measurements and shows whether the requirements of the EU regulation are to be complied with. The challenges for users of the applicable measurement standard are also highlighted.

**Electric Cars - Noise Simulation of AVAS Effects**

***S. Eggers***[a] ***and H. Steven***[b]

[a]*LÄRMKONTOR GmbH, Altonaer Poststraße 13b, 22767 Hamburg, Germany;* [b]*HS Data Analysis and Consultancy, Dorath 1, 52525 Heinsberg, Germany*

*s.eggers@laermkontor.de*

Besides their environmental impact on e.g. air pollution, electric cars are often expected to bring a noticable reduction in noise emissions. Even for speeds on urban roads, the noise contribution of the tyres (i.e. rolling noise) is dominant. A reduction in urban areas could still be expected as the engine noise is significantly lower. Since mid 2021, the "Acoustic Vehicle Alert System" (AVAS) is mandatory for new electric vehicles to improve safety for pedestrians and cyclists, creating artificial noises for speeds up to 20 km/h. Overall, the impact of AVAS on urban road noise remains unclear as the emission varies between different cars and can also be in magnitude of combustion engine noise emissions. As part of recent research, a noise simulation was carried out using detailed noise emissions for single cars (TraNECaM). Rolling noise and engine noise are calculated for various traffic situations in steps of single seconds, depending on the acceleration and speed. The model allows the detailed simulation of AVAS as the emission is only active for low vehicle speeds. Simulations were carried out for different crossing situations with percentages of cars passing at higher speed and stopping. Results will be presented as well for the average noise level as for an assessment of the changing noise level and single noise events.

Tu. 8:30 Musiksalen Objective and perceptual aspects of measured and simulated alternatives to conventional spaces

**Acoustic labels as a simple, easy-to-understand visual representation of the acoustic suitability of unconventional spaces**

***K. Jambrosic and M. Horvat***

*Faculty of Electrical Engineering and Computing, Uni Zagreb, Unska 3, 10000 Zagreb, Croatia*

*kristian.jambrosic@fer.hr*

Instead of building many dedicated spaces that are acoustically optimised for specific types of events, many municipalities are forced by financial constraints to use multi-purpose halls and/or unconventional spaces as venues for a variety of different events. Depending on immediate needs, many public spaces are used as conference or lecture rooms, as well as venues for other speech-based events, but also as venues for unamplified or amplified musical performances, exhibition spaces, cinemas, theatres, etc. Many historic buildings are also frequently used in this way due to their attractiveness to visitors and their historical and cultural significance. In all these cases, the acoustic suitability of a space for a particular type of event must be investigated and determined. Many objective room acoustic parameters have been developed and are commonly used to evaluate the acoustic quality of both performance spaces and ordinary rooms. These parameters are not necessarily intended for the evaluation of unconventional spaces or spaces that are used temporarily or occasionally as performance spaces, e.g. an industrial hall used as a theatre, or a stone atrium used as a concert venue. Nevertheless, a rough rule-of-thumb evaluation can be made using these parameters to assess the acoustic suitability of such unconventional spaces for certain types of events. To help non-experts understand the results of such an evaluation, simple methods of visually representing the results are proposed, leading to a kind of acoustic label describing the acoustics of a room, similar to a product label describing a product by listing its main characteristics, or an energy label as the indicator of the energy efficiency of a product. A crude version of this approach has already been used to assess the suitability of several historic and contemporary spaces for different types of events, and is now being refined and extended.

Tu. 9:00 Musiksalen Objective and perceptual aspects of measured and simulated alternatives to conventional spaces

**Alternative Spaces for Learning: Reversible Acoustic Treatments for Transformation of Spaces into Classrooms During COVID Era**

***D.I. Schiavon, L. Shtrepi and A. Astolfi***

*Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy*

*louena.shtrepi@polito.it*

The design of acoustic comfort in classrooms is a challenging issue. It became a crucial aspect during the Covid-19 pandemic regarding class life organization, as for the latter large spaces for teaching were required in order to guarantee the minimum distance between the occupants to limit the spread of COVID-19, and it was important to ensure their acoustics functionality for the performance of lessons. To this aim, this investigation is to provide a guide (for school principals, administrators, safety managers, architects and engineers, acousticians) where easily implementable solutions for different types of school buildings are detailed in various case studies and they are an example in case of environments similar in shape and volume. In particular the focus is put on the transformation of school spaces created for different purposes than normal classrooms (e.g. corridors, atria, sports halls) into classrooms for teaching. Different layouts of the furniture and sound-absorbing materials have been applied and analyzed by simulations (Odeon 16) comparing the results with the standardized optimal values and with those of the actual environments obtained through an extensive measurement campaign. Eight schools were taken into account as case studies, representing the Italian school heritage and including elementary schools, middle schools and high schools. A total of 26 different spaces with a volume varying in the range 135-2800 m3 were analyzed. For each of them, both general analyzes at the overall environment level and more specific ones for single receivers located over the area occupied by the students were carried out. As a result, it should be highlighted that the proposed solutions could not be fully acoustically optimized for all of the spaces, as the analyze also took into consideration an affordable cost, the speed of realization and the reversibility of the intervention.

Tu. 9:30 Musiksalen Objective and perceptual aspects of measured and simulated alternatives to conventional spaces

**The development of modern, interactive acoustic courseware material within the Acoustics Knowledge Alliance project**

***M. Horvat*[a], *K. Jaruszewska*[b], *S. Raetz*[c], *E. Carayol*[d], *Y. Sluyts*[e], *A. Herweg*[f], *L. Aspöck*[g] *and S. Zeman*[h]**

[a]*Faculty of Electrical Engineering and Computing, Uni Zagreb, Unska 3, 10000 Zagreb, Croatia;* [b]*KFB Acoustics Sp. z o.o., Mydlana 7, 51-502 Wrocław, Poland;* [c]*LAUM, Le Mans Université, Avenue Olivier Messiaen, 72085 Le Mans, France;* [d]*Kahle Acoustics, Rue Émile Banning 24, B-1050 Brussels, Belgium;* [e]*KU Leuven, Celestijnenlaan 200 D, 3000 Leuven, Belgium;* [f]*HEAD acoustics GmbH, Ebertstraße 30a, 52134 Herzogenrath, Germany;* [g]*Institute for Hearing Technology and Acoustics, RWTH Aachen, Kopernikusstraße 5, 52074 Aachen, Germany;* [h]*Jazzy Innovations sp. z o.o., ul. Zygmunta Starego 22/42, 44-100 Gliwice, Poland*

*marko.horvat@fer.hr*

The impact of sound and acoustic phenomena on daily life and its overall quality is not sufficiently perceived by the general public. In addition, in practically all fields of acoustics, there is a lack of professionals with the necessary knowledge and training to solve the problems that occur in everyday life. In order to raise general awareness of the importance of acoustics and to disseminate knowledge to both professionals and lay people, several universities and companies have joined forces in a consortium with the goal to facilitate the development and dissemination of contemporary teaching materials that go beyond the usual forms and approaches to knowledge transfer. As a result, the freely accessible platform Acoustic Courseware (ACOUCOU) was launched. Over the years, the consortium has developed different courseware on acoustics for specific user groups within the scope of several projects. The latest, ongoing project is the Acoustics Knowledge Alliance (ASKnow) project, funded by the ERASMUS+ programme. The aim is to develop modern, interactive content that will form a well-rounded package of teaching materials. These materials will be used in five specific areas of acoustics, i.e. in five corresponding courses: Acoustic Fundamentals, Psychoacoustics, Acoustic Simulations and Auralisation, Electroacoustics, and Room and Building Acoustics. This paper provides an overview of the ASKnow project, its tasks and objectives. It describes the nature and structure of the developed materials and discusses the development procedures used in their creation. The challenges encountered along the way are briefly outlined. The project has entered the phase of transforming the developed teaching materials into their final form, i.e. the interactive online content. This provides an opportunity to present selected extracts of the raw materials developed during the first half of the project, as well as examples of the finished prototypes of interactive online lessons.

**Acoustical aspects not covered by the common, standardised room acoustic parameters**

***T. Halmrast***

*Cowi and Univ. of Oslo/Musicology, Spångbergveien 28a, 0853 Oslo, Norway*

*torhalm@online.no*

The standardised room acoustic parameters are of great help in the design and evaluation of concert halls. However, there might be excellent halls that do not fulfil the criteria, and, on the other side; halls that fulfil most of these criteria that still are not of high quality. This paper will present some acoustical aspects not covered by the common, standardised parameters. As an introduction, we might remember the words by Albert Einstein: ”Not everything that counts can be counted, and not everything that can be counted counts”. The paper will give examples of some aspects not covered by the standardised parameters, such as: Directivity of the source, Level of the Reverberation, Investigations of the very early part of the Impulse Responses, Room Acoustic Attack, Time limits for room acoustic parameters, Direction of perceived reverberation, What is an echo?, Frequency Response/Resonances, What is Bass Response?, Critical Distance in detail, Focusing effects from corners and curved surfaces, Early reflections/Comb filter coloration, Direction of early reflections to the audience, Envelopment from the rear of the hall, Empty/Occupied stage, Response back to the musician, Flutter Echoes, Seating arrangements for the orchestra, etc. Many of these aspects can be counted and measured, but by non-standardised measurements. We need to learn which of these aspects that counts the most! Acoustic design is like cooking a good meal or composing music: We know how each ingrediences tastes and how each note sounds, but the problem is to choose the right amount of each of them. Ref.: Halmrast, T: Acoustics in Between: Perception of Sound in Rooms Beyond Standard Criteria Psychomusicology: Music, Mind, and Brain. Special ed. in honour of Leo Beranek. American Psychological Association 2015, Vol. 25, No. 3

**A Two Stage Embedded Genetic Algorithm to Optimise Ceiling Reflections**

***J. O'Keefe***

*O'Keefe Acoustics, 10 Ridley Gardens, Toronto, Canada M6R 2T8*

*john@okeefeacoustics.com*

Recent studies have explored Machine Learning techniques to optimise individual NURB-based acoustical reflectors. Reconfiguring the shape of an individual reflector with the freedom that only NURB geometry allows. This study, conversely, limits itself to primitive geometries such as triangles, polygons, circles and extruded versions of each, i.e. partial spheres, etc. These primitive shapes are collected in an array as they might be in a theatre or concert hall ceiling. An embedded, two stage Genetic Algorithm (GA) is then employed to optimise their size and orientation to encourage uniform laterally reflected sound. Each of the reflectors in the array is set at a given anchor point, starting out parallel to the receiving surface. To initiate the first GA, the given reflector is oriented to direct sound towards an Aiming Point on the receiving surface. The first GA then optimises that (individual) reflector to cast as many reflections as possible across the receiving surface, as laterally as possible. This is repeated for each reflector in the ceiling array. Several similar arrays are then created and these form the population for the second GA. An early experiment with this 2-stage GA suggests that it might provide higher and more uniform Lateral Fractions than a human design.

**Acoustic Simulation of Small Rooms**

***B. Mondet***
*COMSOL, Diplomvej 373, 2800 Kgs. Lyngby, Denmark*
*boris@comsol.dk*

Ray tracing is currently the most common method to perform room acoustic simulations. Despite all its advantages, the approximations associated with it greatly limit its capacities in the case of small volumes. This study explores potential solutions for the simulation of rooms where modal behavior plays an important role in the frequency range of interest. A seminar room is modeled according to available published data comprising the geometrical model of the room, random incidence absorption and scattering coefficients for the different surfaces present, as well as in situ measurement results. Simulations are performed in COMSOL Multiphysics using several variants of ray tracing and finite element method (FEM). The results from the different methods are analyzed both in terms of accuracy relative to the measurements and computation time.

Tu. 8:30 Gæstesalen Current topics in hearing-aid processing and fitting 1

**Deep neural networks for speaker voice separation and noise reduction for hearing impaired listeners.**

***L. Bramsløw*[a], *G. Naithani*[b] *and T. Virtanen*[b]**

[a]*Eriksholm Research Centre, Oticon A/S, Rørtangvej 20, 3070 Snekkersten, Denmark;* [b]*Tampere University, Korkeakoulunkatu 7, 33720 Tampere, Finland*

*labw@eriksholm.com*

Deep neural networks (DNN) have demonstrated substantial user benefits for speech-in-noise enhancement for hearing-impaired listeners and voice-on-voice enhancement. Recently, deep neural network algorithms have shown great potential in tasks like blind source separation of a single-channel (monaural) mixture of multiple voices. The idea is to train the algorithm on relatively short samples of clean speech, thus learning the characteristics of each voice. Once trained for those specific voices, the network can then be applied to mixtures of new speech samples from the same voices. The present work has applied several low- latency DNN architectures for 1) separating two known voices and 2) enhancing speech from more common noise types: a party noise and a shopping centre ambient noise. Speech from a 12-voice Danish HINT corpus was used for training, from which two male and two female voices were used in a HINT listening test, recording the word scores. In experiment 1, two known voices were separated, using three different DNN algorithms. A 37%-point word score recognition in 15 hearing-impaired listeners was shown when selecting one voice and 13%-point in the same listeners when presenting the two separated voices dichotically to the two ears. These statistically significant benefits indicated a large potential for separating two voices in quiet. In experiment 2, voice in noise was scored by 21 hearing-impaired listeners. Five different DNN architectures, employing both voice-dependent and voice- independent training. In party noise, a modest, but statistically significant, word score improvement of 16%-point was found for a voice-dependent and a voice- independent DNN type, while two other DNN types showed an improvement of 12%-point. In the more stationary shopping centre noise, no improvements were found. It was assumed that the more modulated party noise provided more glimpsing opportunities for the DNN algorithm, compared to the shopping centre noise.

Tu. 9:00 Gæstesalen Current topics in hearing-aid processing and fitting 1

**Benefit of impulse noise reduction in commercially available hearing aids**

***H. Husstedt*[a], *W. Hilgerdenaar*[a], *M. Frenz*[a], *F. Denk*[a] *and J. Tchorz*[b]**

[a]*German Institute of Hearing Aids, Anschützstr. 1, 23562 Lübeck, Germany;* [b]*Technische Hochschule Lübeck, Mönkhofer Weg 239, 23562 Lübeck, Germany*

*h.husstedt@dhi-online.de*

In this contribution, we discuss the benefit of impulse (or transient) noise reduction (INR) in hearing aids based on a study with six commercially available state-of-the-art behind-the-ear (BTE) hearing aids. Anechoic recordings of nine impulse signals were presented via a group delay equalized loudspeaker in an audiological test room. First, the hearing aids were mounted on a head and torso simulator, and the C-weighted peak sound pressure levels at the ear simulators were measured. Second, 24 hearing-impaired and 20 normal-hearing test subjects rated how uncomfortable they experienced the same impulse signals on a nine-point scale from ”not uncomfortable” to ”extremely uncomfortable”. The experienced discomfort was rated by all subjects while unaided, and by the hearing-impaired subjects additionally while wearing the six hearing aids with and without activated INR. During these measurements, a fitting according to NAL-NL2 was used in all hearing aids, and the impulse signals were presented in quiet. Further technical measurements included additional hearing aid settings to analyze the effect of other common hearing aid features on impulse noise and speech such as compression, noise reduction for speech enhancement and the output limiter. Based on these results, we discuss the following research questions: (I) What is the unique benefit of INR compared to other common features in hearing aids? (ii) Do hearing aid user with and without INR experience more discomfort from impulse noise compared to normal-hearing listeners without hearing aids? (iii) What should be the audiological goal of INR in hearing aids?

**Multi-modal hearing devices and their evaluation**

***V. Hohmann, H. Kayser and G. Grimm***

*Carl von Ossietzky Universität Oldenburg, Auditory Signal Processing and Cluster of Excellence Hearing4all, Ammerländer Heerstraße 114-118, 26129 Oldenburg, Germany*

*volker.hohmann@uni-oldenburg.de*

Spatial filtering and decomposition of sounds into acoustic source objects is increasingly investigated for speech enhancement in hearing aids. However, with increasing performance and availability of these space-aware hearing aid algorithms, knowledge of the user's personal listening preferences and of the attended source becomes crucial. For instance, such an algorithm combines information of the subject's eye- and head-movement behavior with an acoustic analysis of the sound source positions to identify the attended source(s) from a mixture of sources. Gaze direction is recorded by electrooculography (EOG) combined with a head tracking system, which could in principle be integrated in hearing aids. The spatio-temporal distribution of source positions is estimated from input signals to a binaural hearing device. Beamformer techniques are then used to enhance the source (or the sources) that are estimated as being attended. By using (electro-)physiologic sensor signals to estimate attention and behavior, the hearing device becomes part of the active communication loop between environment and behaving subject. Such closed-loop hearing devices require subject-in-the-loop evaluation methods, which simulate realistic interactive environments to reproduce natural behavior. This means that audio-visual signals are required that also include behavioral simulation of the sources, e.g., simulated lip movement, or simulated conversational gaze behavior. The current status of our developments towards such an audio-visual interactive lab are presented in this talk in addition to experimental data gathered in this lab on closed-loop hearing device algorithms.

Project funded by DFG SFB 1330 project B1

**Hearing-aid delay in open-fit devices: Preferred sound quality in normal-hearing and hearing-impaired listeners**

***G. Stiefenhofer*[a], *E. Lundorff*[a], *R. Skipper*[b] *and T. Neher*[c]**

[a]*WS Audiology, Nymøllevej 6, 3540 Lynge, Denmark;* [b]*Odense University Hospital, Dept. of Audiology, J. B. Winsløws Vej 4, 5000 Odense, Denmark;* [c]*University of Southern Denmark, Campusvej 55, 5230 Odense, Denmark*

*georg.stiefenhofer@wsa.com*

Sound quality is an important contributor to hearing-aid satisfaction and uptake. A key factor influencing sound quality in open-fit digital hearing aids is the inherent processing delay. Delay-based distortions occur when the processed and delayed sound mixes with the direct sound that enters the ear canal through the vent. Depending on the magnitude of the delay, the delay-based distortions are often perceived as coloration of the sound, echoes and/or altered spatial qualities. The upper limit of acceptable hearing aid delays has been established in earlier studies and is generally considered to be ∼10 ms. However, the lower limit needed for ensuring optimal sound quality has not been established for realistic, open-fit hearing aids so far. The current study investigated the relationship between preferred sound quality and five hearing-aid delays ranging from 0.5 to 10 ms using pairwise comparisons. Three different everyday sounds were processed using a realistic hearing-aid simulator that preserved the specific open-ear acoustics. Thirteen normal-hearing listeners and 20 listeners with mild-to-moderate hearing losses took part. The normal-hearing group showed a strong preference for the shortest (0.5 ms) delay for all three sounds. For the hearing-impaired group, delay preferences were more variable and depended on the type of sound. A moderate correlation between the participants' pure-tone average hearing losses and their preference scores was found. The participants who preferred the shortest delay had near-normal hearing thresholds below 2 kHz. Overall, these results indicate that very short hearing-aid delays are beneficial for persons with mild hearing impairments.

Tu. 11:45 Gæstesalen Current topics in hearing-aid processing and fitting 2

**Towards auditory profile-based hearing-aid fittings: BEAR rationale and clinical implementation**

***R. Sanchez-Lopez***[a]***, M. Fereczkowski***[b]***, M. Wu***[b]***, S. Santurette***[c]***, M. Baumann***[c]***, B. Kowalewski***[d]***, T. Piechowiak***[e]***, G. Ravn***[f]***, S.K. Narayanan***[g]***, T. Dau***[a] ***and T. Neher***[b]

[a]*Technical University of Denmark, Department of Health Technology, 2800 Kgs. Lyngby, Denmark;* [b]*University of Southern Denmark, Campusvej 55, 5230 Odense, Denmark;* [c]*Oticon A/S, Kongebakken 9,, 2765 Smørum, Denmark;* [d]*WS Audiology Denmark A/S, Nymøllevej 6, 3540 Lynge, Denmark;* [e]*GN Store Nord A/S, Lautrupbjerg 7, 2750 Ballerup, Denmark;* [f]*Force Technology A/S, Agro Food Park 13, 8200 Aarhus, Denmark;* [g]*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*rsalo@dtu.dk*

Although modern hearing aids (HAs) come with many advanced signal processing algorithms, hearing loss compensation is still largely based on the prescription of frequency-specific gain. Arguably, advanced HA features could be adjusted more systematically to the hearing deficits of individual patients (e.g., based on speech-in-noise tests). Importantly, however, such personalization attempts would need to be based on suitable, time-efficient diagnostic measures.

As part of the Danish ëBetter hEAring Rehabilitation' (BEAR) project, we recently proposed a data-driven method for auditory profiling to enable the stratification of patients based on measures beyond the pure-tone audiogram. We then tested a heterogeneous group of listeners using a test battery designed to tap into different aspects of supra-threshold auditory processing. When analyzing the collected data, we found evidence for four distinct auditory profiles that varied primarily along two dimensions of hearing deficits: speech intelligibility-related deficits and loudness-related deficits. Here, we describe the theoretical basis for a HA fitting strategy - the BEAR rationale - that we developed to individualize HA settings to account for the deficts observed for these auditory profiles. Furthermore, we summarize results from a pilot study based on a HA simulator that we conducted as a first step towards evaluating the BEAR rationale. Finally, we present how we translated the new rationale to clinically usable fitting solutions and devices based on real- ear gain adjustments and SNR improvement measurements.

Collaboration and support by Innovation Fund Denmark (Grand Solutions 5164- 00011B), Oticon, GN Resound, Widex and other partners (University of Southern Denmark, Aalborg University, the Technical University of Denmark, Force, and Aalborg, Odense and Copenhagen University Hospitals) is sincerely acknowledged.

**Hearing-aid amplitude compression for listeners with rollover at above-conversational speech levels**

***M. Fereczkowski*[a], *S. Christiansen*[b], *R. Sanchez-Lopez*[c] *and T. Neher*[d]**

[a]*University of Southern Denmark, Fioniavej, 5230 Odense, Denmark;* [b]*University of Southern Denmark, Fioniavej 36, 5230 Odense, Denmark;* [c]*Technical University of Denmark, ørsteds Plads 352, 2800 Lyngby, East Timor;* [d]*University of Southern Denmark, Campusvej 55, 5230 Odense, Denmark*

*mfer@health.sdu.dk*

Hearing loss typically leads to a reduced auditory dynamic range (DR). To compensate for that, hearing aids (HAs) apply DR compression. In clinical HA fittings, attack times (ATs) are usually short (e.g., 10 ms), release times (RTs) long (e.g., >500 ms), and compression ratios (CRs) relatively low (e.g., 2:1). While this may be suitable for the ëaverage' listener, it may be suboptimal for listeners showing reduced speech recognition at above- conversational levels - so-called rollover - since this effectively limits the available DR. To minimize the effects of rollover, higher CRs could be beneficial. Since high CRs combined with short ATs lead to more signal distortion, long ATs could be advantageous. Here, the aim was to evaluate two compressors with relatively high CRs: FAST (short AT) and SLOW (long AT). These two compressors were compared to a reference compressor (REF) with moderate CRs, comparable to those used in clinical HA fittings. Twenty hearing-impaired listeners exhibiting rollover at above-conversational speech levels were tested. For REF, NAL-NL1-based insertion gains were prescribed, and the AT and RT were set to 10 and 800 ms, respectively. For FAST, the same time constants were used, but the CRs were set to 5:1. For SLOW, the AT was set to 800 ms. The three compressors were evaluated using sentence- recognition and preference-judgment tasks. It is expected that SLOW and REF will lead to comparable results at moderate input levels, but that SLOW will be beneficial at low and high levels. Due to more signal distortions, the results obtained with FAST are expected to be poorer than those obtained with SLOW.

Collaboration and support by Innovation Fund Denmark (Grand Solutions 5164- 00011B), Oticon, GN-Resound, Widex and other partners (University of Southern Denmark, Aalborg University, Technical University of Denmark, Force, and Aalborg, Odense and Copenhagen University Hospitals) is sincerely acknowledged.

**Orchestra Conductor Acoustics**

***M. Skålevik***
*AKUTEK and Brekke & Strand, Bølstadtunet 7, 3430 Spikkestad, Norway*
*msk@brekkestrand.no*

While the playing and listening conditions for musicians have been studied and taken into consideration in research and design practice since Gade's early work in Podium Acoustics in the 1980s, the impact of room acoustics on the conductors working conditions are only rarely addressed in scientific papers. In preparation of a concert performance, conductors typically need to hear in considerable detail what is going on in the orchestra, perhaps even more during rehearsal than during the actual concert. Importantly, the conductor needs to prepare the orchestra sound and balance to be presented to the audience. In this paper, sound from a symphony orchestra in various rehearsal rooms and a typical concert hall is simulated. Features of orchestra sound at the ears of the conductor's ears and at the ears of the audience is studied. The impact of varying acoustics and the difference between sound at conductor's ears and sound audience ears are commented. In particular, the significance of compliance with ISO-23591 is studied. A questionnaire suggested by this author for a survey among conductors is presented and discussed, the purpose being to provide a basis for finding acoustic measures that can describe critical aspects from the conductors' perspective.

**Feasibility of inhomogeneous MPPs in multipurpose halls: The case study of Bilkent Concert Hall in Ankara**

***E. Fasllija***[a] ***and S. Yilmazer***[b]

[a]*Bilkent University, Department of Interior Architecture and Environmental Design, 06800 Ankara, Turkey;* [b]*Purdue University, Ray W. Herrick Laboratories, West Lafayette, In, 47907, USA*

*e.fasllija@bilkent.edu.tr*

For multipurpose halls, there are additional constraints in planning for good visual conditions and sightlines, meeting specific social demands concerning layout, and providing good acoustic conditions. Regarding the latter, they require a compromise between the auditory needs of speech and music. While conventional materials fail to absorb sounds in the low-frequency range, Helmholtz resonators show a very narrow absorption bandwidth. The current study investigates the potential created by inhomogeneous Micro-Perforated Panels (MPPs), which literature demonstrated to have a wider absorption bandwidth in the low-mid frequency spectrum. This work uses Bilkent concert Hall, a 700-seat capacity multipurpose trapezoid-shaped hall located in Ankara, Turkey, as a case study to examine its current acoustic condition and provide material-based solutions for the existing problems. It analyzes the hall by assessing the fundamental acoustic parameters such as reverberation time, sound pressure levels, and speech transmission index. Comparative simulations between the existing and the proposed material-based solution of the Bilkent Concert hall showed an improvement in the reverberation time and other objective acoustic criteria when inhomogeneous MPPs were used as wall panels instead of conventional materials. This comparison enhances the existing problems in the multipurpose hall and emphasizes the viability of those ”next generation” absorbing materials, which are at the same time are in demand from architects due to their design feasibility.

**Room Acoustic design for electronic enhancement systems**

***H. Möller***
*Akukon Ltc, Hiomotie 19, 00380 Helsinki, Finland*
*henrik.moller@akukon.com*

Within the last 10-15 years, electroacoustic enhancement systems have become a real tool to improve the acoustic conditions in multi-functional halls, both as full systems, where the total acoustic field is created electronically, and as an extension of existing acoustic conditions in the hall. However, one common misconception is that it is possible to ”repair the acoustics” or fix problems with an electroacoustic enhancement system. The fact is that good acoustic conditions are necessary for a the successful implementation of an electroacoustic enhancement system. This paper will give a brief overview of the current technology and present the typical acoustic requirement and examples of solutions.

**Sound insulation, residents' satisfaction, and design of wooden residential buildings**

***F. Ljunggren***

*Luleå University of Technology, Luleå University of Technology, 97187 Luleå, Sweden*

*fredrik.ljunggren@ltu.se*

Wood-based multi-family houses continues to gain popularity. Related to acoustics, low-frequency sound insulation as well as appropriate single number quantities for the evaluation of sound insulation have been in focus for a long time. In a series of Swedish research projects running for 12 years, the correlation between rated annoyance from residents and measured airborne and impact sound insulation, with alternative frequency ranges and weightings, have been studied. In total, 38 building cases of various constructions were involved and more than 1200 questionnaire responses were collected. While the building code's present evaluation parameter for airborne sound insulation, D'nT,w + C50-3150, seems to be working well, the situation is different with respect to impact sound insulation. L'nT,w as well as L'nT,w + CI,50-2500 show weak correlation with the rated annoyance from the residents. The reason is that frequencies below 50 Hz are overlooked, although they dominate the response from walking in many common, particularly lightweight, floor constructions. The strongest correlation with the rated annoyance from impact sound, including both lightweight and heavyweight constructions, was found when the measured frequency range was extended down to 25 Hz, using L'nT,w + CI,25-2500. Because footstep noise rendered the highest degree of annoyance in the survey, a somewhat more restricted requirement than what it used today is suggested to offer a higher degree of protection against unwanted impact sounds. It is a delicate challenge to design wood-based floor constructions with great sound insulation at low frequencies to meet the higher requirement. A tested innovative floor design based upon two high-density cross-laminated timber plates with an intermediate damping layer may serve as the basis for future constructions.

**A light-weight wireless omni-directional sound source for room acoustic measurements**

***C.L. Christensen, G. Koutsouris, A. Richard, J.H. Rindel and A.K. Nørgaard***
*Odeon A/S, DTU Science Park, Diplomvej Bldg. 381, 2800 Kgs. Lyngby, Denmark*
*clc@odeon.dk*

One of the biggest challenges in room acoustic measurements, according to the ISO 3382 standard (parts 1-3), is to use sources that maintain omnidirectional directivity (with small deviations) across the frequency range, from the 125 Hz to 4 kHz octave bands. Conventional omnidirectional sources can achieve adequately omnidirectional directivity with the use of more than one loudspeaker unit (e.g. twelve units in a dodecahedron). However, these sources tend to have poor performance at low frequencies, because of the restricted volume shared by the units, as well as non-uniform directivity at high frequencies. In addition, their size makes them far from ideal point sources and discourages acousticians from performing measurements more often, due to their reduced portability. In this paper, a novel sound source is presented, which solves the aforementioned problems. The source consists of a single loudspeaker unit, which keeps the size small and close to an ideal point source. The loudspeaker unit is backed by a cavity with sufficient volume for efficient low-frequency performance. In front of the loudspeaker unit is an acoustic lens with a doughnut-like shape, designed in such a way that the high- frequency radiation is close to omni-directional. This acoustic lens contains all electronics required to make the sound source fully portable and wireless. Free- field measurements of the sound source show that the frequency response is fairly flat over a wide frequency range and that the directivity pattern is well within the limits of ISO 3382, also for frequencies outside the default range (63 Hz and 8 kHz octave bands). An application example is presented, showing that the source is well-suited for room acoustic measurements in small to medium-sized rooms. In addition, the source can be used for measuring the room frequency response, e.g. from 20 Hz to 200 Hz, when positioned in a corner.

**Effective sound absorption of architectural ETFE membranes in the lab**

***Y. Sluyts, C. Glorieux and M. Rychtarikova***
*KU Leuven, Celestijnenlaan 200 D, 3000 Leuven, Belgium*
*yannick.sluyts@kuleuven.be*

In this paper we report the determination of the sound absorption characteristics of architectural ETFE membranes by a modified ISO354 measurement method. Despite the large-scale deployment of these systems in the building industry, little work has been published on this topic. Architectural ETFE membranes and cushions, often based on multilayer foil structure, are gaining popularity amongst architects and designers in the building industry. As the sustainability topic evolved from an elusive goal to a stringent requirement over the past decades, designers have been seeking new materials, such as ETFE membranes, that fit this paradigm. The lightweight membranes allow for a more efficient structural design that leads to the reduction of material use for structural purposes. ETFE foils themselves require less energy for production and transportation than glass panes. The lightweight nature of the membranes results in acoustic transparency at low frequencies. In a typical setting where the membranes are used in a roof or wall structure, this behaviour leads to low frequency acoustic waves not being reflected, which, by its reducing effect on the sound pressure level and reverberation time, is highly beneficial for the indoor acoustic comfort. In terms of determination of this ”effective absorption” in a reverberant chamber, this acoustic transparency poses a challenge. By using a glass wool filled box below the membrane under the test sample, where the high absorption of the large space in the box was acting as an acoustic sink, an outdoor (100% absorbing) environment was mimicked. Reliable values were found for the frequency dependence of the ”effective” absorption despite the small size of the sample, and implemented in the framework of simulations, auralizations and measurement interpretation of real spaces based on ETFE membranes and cushions.

We. 13:00 Poster session

**Simple dynamic measurement system for testing IMU sensor precision in spatial audio**

***P. Franček, K. Jambrošić, M. Horvat and V. Planinec***

*Faculty of Electrical Engineering and Computing, Uni Zagreb, Unska 3, 10000 Zagreb, Croatia*

*petar.francek@fer.hr*

Spatial sound is used in a variety of systems these days. The application of high-quality audio systems with real-time processing ranges from hi-fi systems to systems for live sound and virtual reality. The data needed to (re)create realistic surround sound audio is often collected using Inertial Measurement Unit (IMU) sensors. With technological progress, the number of such sensors in many devices is constantly increasing. By combining the data from all sensor devices, a dynamic virtual audio field can be created that provides an immersive experience for the user. When processing the collected raw data, attention must be paid to the precision and tolerance of the sensors. The IMU sensors used in this type of measurement usually consist of an accelerometer, a gyroscope and sometimes a magnetometer, all of which are triaxial. When processing and editing the measured data, the effects of low precision and accuracy, parameter drift and latency can greatly affect the measurement uncertainty and the overall user experience. Sensor devices used for spatial audio fall into three groups: VR headsets, smartphones and DIY sensor systems. However, the cost of a device does not necessarily guarantee the accuracy of the measured data. This paper presents the results of dynamic measurements on sensors used in embedded systems. Several different types of sensors with different shapes and sizes are used. Some are built on PCB boards with processing units, while others are implemented as modules on dedicated PCBs. A microcontroller is used for data processing. The most widely used open-source platform Arduino is used for the measurements. To avoid the magnetic influence of the moving parts in the setup, a simple aluminium/plastic pendulum is used. To minimise the measurement error, all measurements are taken simultaneously. The measurement results are compared and the quality of the measured sensors is evaluated.

We. 13:00 Poster session

**A parameter-conditional neural network framework for modelling parameterized auditory models**

***P. Leer*[a], *J. Jensen*[b], *Z.-H. Tan*[c], *J. østergaard*[c] *and L. Bramsløw*[a]**

[a]*Eriksholm Research Centre, Oticon A/S, Rørtangvej 20, 3070 Snekkersten, Denmark;* [b]*Oticon, Kongebakken 9, 2765 Smørum, Denmark;* [c]*Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*
*pelb@eriksholm.com*

The use of auditory models is important for designing speech and audio processing algorithms for hearing assistive devices. These auditory models are often parameterized by a set of parameters relating to auditory function, e.g., hair cell loss or synaptopathy. In practice, the computational load of these auditory models can be very high thus limiting the feasibility of using the models as bio- inspired loss functions for deep learning based hearing loss compensation strategies or denoising strategies. Previous efforts have addressed this problem by training a neural network for each parameter configuration of the auditory model which greatly reduces the computation time of the auditory model but requires a new network to be trained whenever the parameterization changes. In this paper we propose an approach where a single neural network is trained, once and for all, to accurately simulate auditory models across their parameter spaces by conditioning the weights of the network on the parameters of the respective auditory models. This approach enables greater flexibility than training a single model for each parameter configuration, as any parameterization can be acquired on the fly. The accuracy of the neural network is shown to be robust across both unseen inputs and standard audiograms.

We. 13:00 Poster session

**Acoustic/structure interaction in timber buildings: resilient profiles friction influence**

***P. Brugnara, C. Luzzani and A. Speranza***
*Rotho Blaas srl, Via dell'Adige, 39040 Cortaccia, Italy*
*paola.brugnara@rothoblaas.com*

The wooden structures are statically assembled with different types of connections. Mainly we talk about screws, angular and hold-down. These connection systems have the task of precisely coupling the different structural elements to form a single large joint structure. While in parallel to overcome the problem of transmission of vibrations, in recent years resilient profiles in EPDM or polyurethane are very often being used which have the task of decoupling and reducing the transmission of vibrations between the different elements. In the past, Rothoblaas investigated the resilient wood-profile interface from an acoustic point of view to understand the influence of fastening systems in the reduction of flanking transmission. The new investigation is focused on resilient wood-profile interface from a structural point of view in order to better understand the sliding mechanisms and therefore also the interaction with the connection system. For this reason we have founded an experimental campaign with the University of Graz and Innsbruck to try to derive some experimental value on wood-resilient profile friction and how much this can change the mechanical strength in the presence of structural screws. To date, friction is not considered in the static design of wooden structures, since Eurocode 5 does not foresee it, but the intent of this research was to know the global behavior of the proposed solutions and to be able to deliver experimental data also to those designers who want to explore this field still new and in great evolution.

**Free-field correction values for RadioEar DD65 v2 Circumaural Audiometric Earphones**

***C.S. Pedersen***[a]***, R. Ordonez***[a] ***and D. Hammershøi***[b]

[a]*Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark;* [b]*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*cp@es.aau.dk*

Speech audiometry with earphones requires frequency dependent calibration based on free-field correction values for the particular earphone used. The free-field correction values are derived from a loudness matching experiment between free-field frontal incidence third-octave-band noise signal from a loudspeaker and the required earphone level as measured on an IEC 60318-1:2009 acoustic coupler. This paper describes the determination and results of free-field correction values for RadioEar DD65 v2 circumaural audiometric earphones for all third-octave band frequencies from 125 Hz to 8 kHz, following the procedure described in the IEC 60268-7:2010 standard using 23 test subjects who also participated in the determination on Equivalent Threshold Sound Pressure Levels for the same earphones. The standard contains details on setting up the loudness comparison to have a controlled sound field and great care was taken to conform with this in the experimental setup. The standard also contains practical details on approximate stimulus and pause durations, and on presentation order; first loudspeaker then earphones. However, when it comes to the choice of psychometric method for obtaining loudness match i.e. how the level of the earphones is varied between presentations and how the final loudness match level is obtained it is just generally stating that the level of the earphones is varied until equal loudness is obtained. As this choice is up to the experimenter, care was taken to eliminate potential biases and a two-alternative forced choice paradigm was chosen. The results are compared to data from another laboratory and a similar type of earphones and possible explanations for minor differences are discussed.

We. 13:00 Poster session

**Equivalent Threshold Sound Pressure Levels (ETSPL) and Insertion Loss for the RadioEar DD65 v2 Circum-aural Audiometric Earphones**

***R. Ordonez*[a], *C.S. Pedersen*[a] *and D. Hammershøi*[b]**

[a]*Department of Electronic Systems, Aalborg University, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark;* [b]*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*rop@es.aau.dk*

Assessment of hearing thresholds requires proper calibration of the audiometric equipment that considers the sensitivity of the earphones used. For this purpose, the ISO 389 series defines the Reference Equivalent Threshold Sound Pressure Level (RETSPL), as the sound pressure level produced by the earphone actuated by the voltage corresponding the median auditory threshold of a young, healthy population, measured in a standardized acoustic coupler. As such RETSPL depend on the type and sensitivity of a given transducer. This paper describes the assessment of the Equivalent Threshold Sound Pressure Levels (ETSPL) and the measurement of insertion loss of the RadioEar DD65 v2 audiometric earphones. Hearing thresholds were determined for twenty-nine test subjects at eleven audiometric frequencies from 125 to 8000 Hz according to the specifications given in ISO 389-9 (2009) and ISO 8253-1 (2010). Insertion loss of two sets of earphones was measured in third-octave bands between 50 and 16000 Hz using an acoustic test-fixture in a reverberant sound field according to ISO 4869-3 (2006). The results are reported as ETSPL measured in an acoustic coupler conforming with the specifications given in IEC 60318-1 (2009) and compared to reference data used as a basis for the RETSPL for circum-aural audiometric earphones, ISO 389-8 (2004). Results show that the auditory thresholds obtained with the RadioEar DD65 v2 circum-aural audiometric earphones are in close agreement with reported values for the Sennheiser HDA200 circum-aural audiometric earphones and the reference values given in ISO 389-8 (2004). The insertion loss shows minimum values at the low frequency range (below 160 Hz) of 5 dB, then it gradually increases up to 45 dB at 2000Hz and above.

We. 13:00 Poster session

**Data-driven Optimization of Parametric Filters for Simulating Head-Related Transfer Functions in real-time Rendering Systems**

***F. Schwark***[a]***, M.R. Schädler***[a] ***and G. Grimm***[b]

[a]*Carl von Ossietzky Universität Oldenburg, Ammerländer Heerstraße 114-118, 26129 Oldenburg, Germany;* [b]*Carl von Ossietzky Universität Oldenburg, Auditory Signal Processing and Cluster of Excellence Hearing4all, Ammerländer Heerstraße 114-118, 26129 Oldenburg, Germany*

*fenja.schwark@uni-oldenburg.de*

In real-time virtual acoustic rendering, the head related directional properties of the receiver, i.e., the listener, are often modeled by convolution of the signal with measured head related impulse responses (HRIRs). However, the computational cost of HRIR-convolution is rather high, especially when targeting low-delay of the acoustic signal, and interpolation needs to be applied to simulate all source directions, depending on the spatial resolution of the HRIR catalogues used. In order to reduce the computational cost in low-delay real-time virtual acoustic rendering, this study uses a parameterized digital filter model with delay lines to approximate the direction-dependent features of the head. A data-driven optimization method for the filter parameters is introduced that aims at matching the direction-dependent features of modeled and measured HRIRs using a spectral distance metric. Using an objective binaural speech intelligibility model, it was shown that the estimate of the speech intelligibility for the optimized model approaches the estimate for the measured HRIRs. This suggests that the parameterized HRIR model may be sufficient to enable plausible spatial perception in virtual acoustic scenes. With the parameterized HRIR model a reduction of computational cost in the order of two magnitudes is possible for virtual acoustic scenes with a small number of objects. Further work will include subjective testing of the model, compared against measured HRIRs.

**Efficacy of Gamification in Out-of-Clinic Hearing Diagnostics**

***P. Rye, R.O. Andersen and D. Hammershøi***

*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*

*par@es.aau.dk*

We are facing an expected increase in the number of people affected by hearing disorders and an ongoing ambition for improved efficiency, e.g. through expeditious or fewer clinical visits. Recent research suggest that classifying patients according to archetypal profile based on additional supra-threshold diagnostic measures allow a consistent treatment approach targeting the needs and preferences of the specific archetypal profile. The potential benefits of such detailed individual patient profiles are likely to necessitate more diagnostic test time for proper classification. Out-of-clinic diagnostic hearing testing proposes a tempting remedy for the above challenges. However, the lack of in-person guidance during testing may lead to unnecessary confusion or frustration and result in poor reliability proving detrimental to the goal. The current study investigates whether a gamification approach to out-of-clinic hearing diagnostics could help counteract such insufficiencies. The proposed gamification approach focusses on simple, effective instructions and seek to provide intuitive interaction with the diagnostic tests. As an example of this approach, tablet-based implementations of 3I-3AFC tests with immediate trial feedback were developed and tested on 18 test subjects. For Spectro-Temporal Modulation and Binaural Pitch the gamified paradigm was compared with a more traditional button-based approach. Acceptance scores for a sample of the intended patient group are reported using the System Usability Scale. Through exit interviews and data analysis particular strengths and weaknesses of the gamified approach was revealed. For several test subjects the additional gamified element detracted from the main task during trials and also resulted in excessive test time.

**Linguo-Acoustic Aspects of Speech**

***M. Hudcovičová*[a], *B. Petrášová*[a] *and M. Rychtarikova*[b]**

[a]*University of Ss. Cyril and Methodius, Nám. J. Herdu 2, 91701 Trnava, Slovakia;* [b]*KU Leuven, Celestijnenlaan 200 D, 3000 Leuven, Belgium*

*monika.rychtarikova@kuleuven.be*

Speech is the ability making human beings unique. Linguistic diversity reflecting wide variety of speech segments does not only bring about different speech sound production mechanisms, but because of lack of appropriate acoustic research, reminds us of the need to document acoustic properties of individual sounds, as well as their possible combinations in speech in various languages. The paper presents the issues connected with the process of preparing a digital database of Slovak sentences that will be used for testing speech intelligibility in various acoustic conditions. The sentences produced as spoken utterances will be selected according to the most significant linguistic criteria to meet all the requirements for creating the relevant database. The standardised sentences will be recorded and subsequently tested in simulated laboratory conditions, while paying attention to variables in different acoustic environments. The phonetic features of the material analysed will be evaluated by acoustic analyses using the computer program PRAAT (Praat software Version 6.1.47; Boersma & Weenink) which is also used to create auditory spectrograms. The program helps to test speech intelligibility, as it measures pitch, formants, intensity and duration of sounds. These aspects are fundamental for perceiving speech sounds and decoding them properly. Based on the rules of phonotactics, combinations of vowel and consonant phonemes (in open/ closed syllables, mono-/ polysyllabic words) reflecting higher or restricted level of speech intelligibility in different acoustic environments will be studied. The linguistic data will be collected via the Slovak National Corpus, version prim 9.0. In order to construct the sentences from the most frequent words, the electronic statistical tools will be applied: ARF (average reduced frequency) and potential occurrence in one million words. For detecting the co-occurrence of the words, the statistical tools of the electronic corpora will be applied: MI Score and T Score.

**Transformation of office space for laboratory listening room**

***L. Zelem***[a]***, V. Chmelík***[a]***, D. Urbán***[a] ***and M. Rychtarikova***[b]

[a]*Slovak University of Technology in Bratislava, Radlinského 2766/11, 810 05 Bratislava, Slovakia;* [b]*KU Leuven, Celestijnenlaan 200 D, 3000 Leuven, Belgium*

*lukas.zelem@stuba.sk*

The issue of sustainability in the society leads to efforts associated with changing the purpose of conventional spaces while maintaining the essence of the original building. Therefore, the change of use of the spaces is a common process during the buildings' life cycles. However, if the purpose of the room with lower requirements is transformed to room with higher requirements, in terms of building physics, it is necessary to look at the given issue from multiple point of view. One of the basic points of view is to critically evaluate, whether there is a space in the building that has the potential to fulfil the requirements for a new purpose. This article is aimed at the transformation of the so-called office spaces to acoustic laboratory designed for subjective laboratory listening tests. Within a given transformation, it was necessary to select a room, which eliminated possible sources of interfering sounds by its location. Subsequently, the construction adjustments were necessary. These treatments were focused on building and room acoustics. The aim was to create a room with as low background noise level as possible and high sound absorption at the same time. During the conversion process, we had to face several issues resulting from the original design of the space. Airborne sound insulation is improved using gypsum board lining system. The sound attenuation of the newly designed of listening room is ensured by materials with high sound absorption. Thanks to these adjustments, it was possible to significantly increase airborne sound insulation and reduce the average reverberation time. The laboratory is recently used for research as well as teaching process.

**Sound producing mechanism in the temperature inversion layer and its sensitivity to geomagnetic activity**

***U.K. Laine***

*Aalto University, Otakaari 3, PL 12200, 02150 Espoo, Finland*

*unto.k.laine@aalto.fi*

Sounds associated with aurora borealis, so called Auroral Sounds (AS), are explained by electrical discharges occurring in the Temperature Inversion Layer (TIL) approximately 75 meters above the ground [1]. It is assumed that under favourable weather conditions the number of AS events depends on the activity of the geomagnetic (GM) storm. Historical testimonies of AS indicate that the GM storm should be strong enough to rise bright and lively auroras to the zenith before these sounds can be observed. The goal of this study is to test the relevancy of this claim by studying the sensitivity of the AS producing mechanism under moderate GM activity. A four-hour sound measurement was performed around the local magnetic midnight on Jan 25- 26, 2022 at Fiskars village in southern Finland. Sixty AS event candidates were manually selected and their temporal distributions and period histograms constructed for statistical comparisons with the GM data measured simultaneously by the Finnish Meteorological Institute (FMI) at Nurmijärvi Geophysical Observatory (NGO). The data was also used for regression models to study possible causal relationship between the GM activity and the AS events. A strong causality was found. The temporal distribution of the sounds was predicted by the GM activity with 90% accuracy after a delay of 21 min. The results show that in good weather conditions the AS events are much more common than previously thought.

**ISO 235891 Acoustic quality criteria for music rehearsal rooms - why, and what does it say**

***J.G. Olsen***

*Council for Music Organizations in Akershus, Norway, Trondheimsveien 50E, 2007 Kjeller, Norway*

*jon.olsen@musikk.no*

The presentation will explain the main framework and principles in the new ISO standard 23591, published in September 2021. Each week, more than 2,5 million music rehearsal rooms are used throughout Europe. With an average ensemble size og 30, this means 75 million singers and musicians use rooms for music ensemble rehearsals. The acoustic properties of a room are crucial for the interaction between the room and the music instrument and voice. The rooms are used by professionals, students and amateurs. We know that the majority of the rehearsal rooms is not suited to the type of music which in is rehearsed in them. The scope was to ”specify global, differentiated criteria for acoustic conditions and characteristics at music rehearsal rooms and spaces.” The criteria in the standard are differentiated between three music types, quiet acoustic music, loud acoustic music and amplified music. The standard specifies criteria for four different room sizes, corresponding to different ensemble sizes: Individual practice rooms, small ensemble rooms, medium ensemble rooms and large ensemble rooms. The important criteria are net volume and net area, room geometry (form and dimensions), reverberation, acoustic treatment of the surfaces and background noise. The determination and importance of sound strength (G) in a room and how this affect the music rehearsal conditions is described and evaluated in an important annex. The presentation will also present and explain how the standard is intended to be used when planning new buildings and rooms or refurbishment of existing ones, and how the standard can be used to assess the suitability of existing rooms for different music purposes.

**ISO:23591 and amplified music**

***N. Adelman-Larsen***
*Flex Acoustics, Ny Carlsbergvej 27, 1760 Copenhagen, Denmark*
*nwl@flexac.com*

It is common for musicians to use ear protection during solo- or group-rehearsals within the genres of amplified music (pop, rock etc.). This is of course due to the often loud nature of some of the instruments involved, for instance the drum- set. In fact, the ”strength” parameter, and thus the loudness aspect of the Reverberation Time parameter, is of little interest when addressing acoustics for amplified music. The room is not as an outset supposed to help bringing out the amplified sound. However, the notion of the critical distance (Hallradius) parameter has incorrectly led some acousticians to believe that as short an RT as possible would be the answer. Further, especially in larger rooms RT at bass frequencies can become dominant thereby destroying the clarity of the music which is bass heavy. Also, especially in larger rooms, if RT at higher frequencies is too low, then the dynamics of the music and vivacity of the performance suffers. In fact, there is a narrow field of tolerances of RT at various frequencies that should be fulfilled to obtain optimal musical conditions for amplified music in rehearsal rooms and spaces. ISO:23591 is a non-mandatory recommendation and therefor constitutes best practice.

**Development of a Technical Memorandum Describing Optimal Room Acoustic Parameter Ranges for Musical Performance and Rehearsal Spaces**

***E. Green*[a], *M. Barron*[b], *E. Kahle*[a], *T. Halmrast*[c], *W. Lachenmayr*[d], *H. Möller*[e], *U. Stephenson*[f], *L. Shtrepi*[g] *and B. Støfringsdal*[h]**

[a]*Kahle Acoustics, Avenue Moliere 188, 1050 Brussels, Belgium;* [b]*Retired (ex University of Bath, UK), Combe Royal Cottage, Bathwick Hill, BA2 6EQ Bath, UK;* [c]*Cowi and Univ. of Oslo/Musicology, Spångbergveien 28a, 0853 Oslo, Norway;* [d]*Müller-BBM, Helmut-A.-Müller-Str. 1-5, 81252 Planegg, Germany;* [e]*Akukon Ltc, Hiomotie 19, 00380 Helsinki, Finland;* [f]*HafenCity Universität Hamburg, Raumakustik, Klaus Groth Str. 16A, 23611 Bad Schwartau, Germany;* [g]*Politecnico di Torino, Corso Duca degli Abruzzi 24, 10129 Torino, Italy;* [h]*COWI, Langebruvegen 15, NO-6800 Førde, Norway*

*ekahle@kahle.be*

A working group has been convened by the European Acoustics Association to develop optimum Room Acoustic Parameter (RAP) ranges for musical performance and rehearsal spaces. A Technical Memorandum, outlining the proposals of the working group is in production: the main contents are summarized in this article. The main focus has been to develop optimum parameter ranges for concert halls, as well as for other spaces for music rehearsal and performance, such as chamber music halls and orchestra rehearsal halls.

It is widely accepted that room acoustic quality is multi-dimensional: multiple RAPs are therefore required to quantify acoustic quality. The RAPs as currently implemented do not however provide a robust differentiation of acoustic quality - both good and bad sounding halls can fall within the RAP ranges currently proposed in the literature. The Technical Memorandum describes a selection process for RAP ranges, with the aim of setting projects on a good path towards acoustical excellence, and to improve the quantification of the outcome.

While the possibility to develop novel parameters was at first discussed, is has been agreed that the RAPs already defined in ISO 3382 Part 1 will be used. The RAPs are physically linked; therefore, the RAP optimum ranges must form an internally consistent set, taking into consideration their variation throughout the space. Other important aspects that must be considered in the definition of optimal RAPs that are often overlooked are the orchestra/ensemble size, music type or genre, room volume available, ideal loudness during loud (forte) passages, variation of parameters with acoustic volume and with distance from the source.

It has also been established that measurement protocols should be refined to improve the differentiation between spaces of differing acoustical quality.

We. 15:15 Europahallen Closing ceremony

**Closing ceremony**

***D. Hammershøi***
*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark*
*dh@es.aau.dk*

The conference has been organized by the Danish Acoustical Society, Aalborg University and VisitAalborg on behalf of the Nordic Acoustic Association and the European Acoustics Association.

The conference merges the bi-annual Baltic Nordic-Acoustic Meetings, and the EuroRegio conference, which is held in years in between Forum Acusticum and EuroNoise. The conference format comply with the EAA format, where written papers are optional (normally obligatory for BNAM).

The ERBNAM 2022 organization committee would like to thanks all the participants, presenters, exhibitors, keynote speakers and company sponsors for their valuable contribution to this conference.

We hope that you have enjoyed your stay in Aalborg and we look forward to meeting you again in the future.

**Assessing the generalization gap of a deep neural network-based binaural speech enhancement system in noisy and reverberant conditions**

***P. Gonzalez*[a], *T.S. Alstrøm*[b] *and T. May*[a]**

[a]*Technical University of Denmark, Department of Health Technology, 2800 Kgs. Lyngby, Denmark;* [b]*Technical University of Denmark, Department of Applied Mathematics and Computer Science, 2800 Kgs. Lyngby, Denmark*

*phigon@dtu.dk*

Noisy and reverberant speech signals are influenced by many factors, such as the spectro-temporal characteristics of the target speaker and the interfering noise, the room acoustics, the signal-to-noise ratio (SNR) and the position of the different sources in the scene. This large variability of acoustic conditions poses a major challenge for deep neural network (DNN)-based speech enhancement systems, since any mismatch between training and testing can substantially reduce their performance. In addition, the generalization capability of DNN-based systems is typically assessed by testing the system with an arbitrarily chosen speech, noise or binaural room impulse response (BRIR) database that was not seen during training. This poses a problem, as the difficulty of the speech enhancement task can substantially vary across databases, which strongly influences the results and complicates a comparison across studies. The present study systematically investigates the influence of six acoustic scene dimensions on the generalization capability of a binaural DNN-based speech enhancement system, namely the target speaker, the noise type, the room, the SNR, the target direction and the mixture level. We propose a new measure of generalization potential, which is referred to as the generalization gap. It is expressed in percentage and is defined as the performance distance to a reference model that was trained on the test condition. The generalization gap is estimated by cross-validating over multiple speech, noise and BRIR databases, such that the comparison across studies is possible. We find that a mismatch along the speech dimension induces the largest generalization gap (49% in terms of mean squared error (MSE)), followed by the noise type (36%) and the room (30%). The SNR, direction and level dimensions can also induce significant generalization gaps, but these are easily reduced by training on diverse datasets that present a wide range of SNRs, directions and levels.

**Comparison of Speech Reception Thresholds for diotic, dichotic and antiphasic headphone presentations of digits-in-noise triplets using Dantale I material**

***P. Rye*[a], *R.O. Andersen*[a], *G. Ravn*[b] *and D. Hammershøi*[a]**

[a]*Aalborg University, Section for AI and sound, Department of Electronic Systems, Fredrik Bajers Vej 7, B5, 9220 Aalborg ø, Denmark;* [b]*Force Technology A/S, Agro Food Park 13, 8200 Aarhus, Denmark*

*par@es.aau.dk*

A complete and accurate diagnosis is the prerequisite of efficient treatment of any disorder. For hearing disorders, the primary diagnostic tools for characterizing the type and severity typically include a pure-tone audiogram describing the sensitivity loss across the frequency range typical for speech and one or more speech recognition tests. While there is typically a high level of correlation between the audiogram and the speech recognition tests, the speech test may reveal different characteristics of the hearing disorders. The research interest in out-of-clinic versions of speech in noise has increased in recent years. Different implementations exist in various languages, e.g. the digits-in- noise triplet test made available by the World Health Organization. The present study examines a similar triplet test featuring digits in Danish originating from the DANTALE I material. The original masking noise of the DANTALE I material did not contain sufficient high-frequency content to mask the digits effectively and equally well, so a new masking noise was derived from the digit material. Three different spatial configurations of the presented speech in noise were tested: diotic speech in dichotic noise, and antiphasic speech in both diotic and dichotic noise. Initial test on 18 subjects (5 subjects with better hearing ear pure-tone average (PTABE) $>$ 30dB) in the age range 53-81 years (mean 64.4 years) shown reasonable correlation ($r > 0.86$ for all three configurations) between PTABE and the estimated speech reception thresholds.

**Laboratory tests of impact sound insulation of timber floors**

***V. Hongisto, R. Alakoivu, J. Laukka, J. Keränen, J. Virtanen and J. Hakala***
*Turku University of Applied Sciences, Joukahaisenkatu 7, FI-20520 Turku, Finland*
*valtteri.hongisto@turkuamk.fi*

The subtask of a Finnish-Swedish consortium project attempts to develop single- number quantities that could better explain the annoyance of impact sounds transmitted through timber floors. The subtask consists of two parts: I. Measurements of floors, and II. Psychoacoustic experiment. The purpose of this paper is to report the results of I. More than 25 timber floors were tested in the new impact sound insulation laboratory of Turku University of Applied Sciences in Turku, Finland. The test opening size was 10 m2. The load bearing slab options were CLT 260 mm and box slab 370 mm. Various floor coverings and suspended ceilings were tested with them. The measurements were conducted within 205000 Hz using rubber ball, tapping machine, and loudspeaker. The range of values was 2480 dB (Ln,w), 40 dB (L'n,w+CI50-2500), 4174 dB (LiA,Fmax), and 4275 dB (Rw). The database is useful as such for education, industry, and design sector. Since European regulated values of range within 4868 dB (L'n,Tw), we will choose such floors to the part II of the subtask.

**Whole glass facade in office building - Measured noise level and requirement for facade**

***B.M. Larsen***
*Multiconsult Norway, Fjellgata 6, 4612 Kristiansand S, Norway*
*bernt.mikal.larsen@multiconsult.no*

The presentation will summarize calculated and measured noise level from road traffic in office building with whole glass facade. Measurements from project show that the difference between laboratory value and field value is as much as 10 dB for a whole glass facade. The difference is explained by the fact that a whole glass facade needs a total correction for both the effect of weakening due to profile system and area correction due to the size of the glass. Due to both of this effects, the required sound isolation from laboratory should normally be at least 8-10 dB higher than the value achieved for the facade in field. The experience is based on a new office building called Baneheia Park in Kristiansand in Norway. With a whole glassfacade with Rw+Ctr 46 dB in a noisy situation, both calculated an measured noise level from road traftic was Ld 39-40 dB. In Norway the required noise level in offices is Ld 35 dB from road traffic. If effect of reduced sound isolation due to profile system filled with mineral wool (4-6 dB) and reduced effect of sound isolation due to area correction (3-4 dB) were included, the requirement for the fasace in the given situation should have been Rw+Ctr 51 dB. With such facade, the indoor noise level of Ld 35 dB would have been achieved.

**The sound insulation of façades at infrasound frequencies**

***V. Hongisto, J. Keränen and J. Hakala***

*Turku University of Applied Sciences, Joukahaisenkatu 7, FI-20520 Turku, Finland*

*valtteri.hongisto@turkuamk.fi*

The sound pressure level (SPL) caused by environmental noise inside a dwelling is usually calculated by subtracting the outdoor-indoor level difference (DL) of the façade from the outdoor SPL. However, there is very little DL data available below 200 Hz. The purpose of this study was to determine the sound reduction index (SRI) within 50-5000 Hz according to ISO 16283-3 and the DL within 5200 Hz for a broad range of different vertical façade constructions. The statistical distributions of SRI and DL were determined. Thirteen residential buildings involving twenty-six façade constructions were measured. Sound stimuli was produced by three separate loudspeakers. The indoor SPL was measured in four corner and five central positions in the receiving room. The inter-façade differences of SRI within 505000 Hz were 1736 dB. The differences for DL were smaller within 5200 Hz: 1230 dB in corner positions and 1430 dB in central zone positions. The DL values measured in corner positions were significantly lower than the DL values measured in central zone positions within 40200 Hz. The DL values in corners agreed with three previous Danish studies only at few one-third octave bands within 8 to 50 Hz. The DL values were, in general, larger below 50 Hz. This study is unique because we were able to measure DL values down to 5 Hz using loudspeaker excitation. We recommend that the mean value minus two standard deviations can be used to assess the SPL of environmental noise indoors when the SPL outdoors is known but the SRI of the façade cannot be measured. This value is exceeded in 84% of cases. Therefore, it represents a feasible underestimate of the DL of a typical façade.

**Numerical investigation on sound transmission behavior of multilayered panels with periodic arrays of spring-mass resonators**

***M. Jovanoska and T. Samardzioska***
*Faculty of Civil Engineering, UKIM, blvd. Partizanski odredi 24, 1000 Skopje, Macedonia*
*m.jovanoska@gf.ukim.edu.mk*

Recent developments in engineering and sustainability initiatives have resulted in building trend of lightweight partitions, utilizing new materials and technologies. At the same time, noise pollution is becoming a growing problem across the globe, as a result of present-day life dynamics. The impact of the increased noise levels on human health and wellbeing is perceived. All this implies the need for sound insulation enhancement of the lightweight partitions. Theoretical and numerical investigation of multilayered panels under acoustic excitation are conducted using different methods. The sound transmission loss (STL) of unbounded multilayered panels is calculated using the theory of three- dimensional elasticity and well-known transfer matrix method (TMM). Additionally, the finite element method (FEM) is also used for sound transmission loss calculation and for obtaining the dispersion diagrams. Based on the dispersion curves, detection of the band gaps is made possible. In order to improve the sound transmission loss in a specific frequency region, periodic resonant units (spring-mass resonators) are tuned and introduced to the considered panels. It can be shown that the resonance mode of the spring-mass resonators couples with the plate vibration in the way of breaking the mass law and overcoming some phenomena like coincidence effect and mass-air-mass resonance. These units can be attached or embedded in the host panel. Such periodic structures recently have been recognized as acoustic metamaterials. Acoustic resonant metamaterials are artificial periodic structures with unique acoustic wave manipulation properties, owing to the dynamic influence of their local resonant units. The drawback of this concept is that this ”unordinary” power for wave manipulation works only in specific narrow frequency band associated with the resonant frequency of the resonant units. Nevertheless, this phenomenon has potential to be useful ”tool” for correcting/complementing lightweight partition systems.

**Laboratory test method to determine noise from wastewater installations**

***S. Conta and A. Homb***

*SINTEF Community, Høgskoleringen 7B, 7034 Trondheim, Norway*

*simone.conta@sintef.no*

We are working on a project which aims at improving sewage pipe material and piping system to reduce the radiated noise. One challenge we faced initially was to identify a suitable experimental method that fulfilled following requirements: allows for testing of the complete system and single components, fits our standard laboratory, suitable for measuring both airborne and structure-borne noise, reliable and cost effective. EN 14366 describes a laboratory measurement procedure to determine noise from wastewater installations. This method requires a setup comprising of two rooms: a source room to determine the airborne sound and a receiving room to indirectly determine the structure-borne sound by measuring the sound pressure level. As the test method was designed to compare pipe systems, the system layout is strictly defined, and the setup is not well suited to investigate system components or selected system sections for R&D purposes. The requirements on the laboratory facility are extensive and test of single components is not foreseen in the standard. In this paper we briefly present findings from a literature survey on the measurements of noise from wastewater installations. Moreover, we present an alternative setup and test method that we designed for our purposes. This setup allows the investigation of vertical and horizonal pipes or even just single components (e.g., bends). We will present sample results and are looking forward to an active discussion of pros and cons.

**On studies of wave propagation by means of conventional finite element modal analysis**

***S. Sorokin*[a], *P. Broberg*[a], *M. Steffensen*[b] *and L. Ledet*[c]**

[a]*Aalborg University, Fibigerstraede 16, 9220 Aalborg, Denmark;* [b]*MEK DTU, Nils Koppels Allé Bygning 404, 2800 Kgs. Lyngby, Denmark;* [c]*Grundfos Holding A/S, Poul Due Jensens Vej 7, 8850 Bjerringbro, Denmark*

*svs@mp.aau.dk*

In this paper, we demonstrate how conventional modal analysis of a slice of an infinite waveguide provides essential information on its properties. As a pre- requisite to accomplish this task, a bi-orthogonality relation for an elastic 3D waveguide of a uniform cross-section is formulated and converted to the finite element format. The sets of ëclass consistent' boundary conditions, which emerge from its formulation, are applied for analysis of free vibrations of a slice of the homogeneous waveguide. Obtained eigenfrequencies and mode shapes are used to reconstruct dispersion diagram for propagating waves and, in the presence of material losses, to assess their decay rates. For a periodic waveguide, the modal analysis of a symmetric unit periodicity cell with the same ëclass consistent' boundary conditions is used to identify partial and full stop-bands. The computational efficiency of the proposed method is assessed.

**Modeling of multiple reflections between noise barriers and trains using the boundary element method**

***C.H. Kasess*[a], *T. Maly*[b] *and W. Kreuzer*[a]**

[a]*Acoustics Research Institute, Austrian Academy of Sciences, Wohllebengasse 12-14, 1040 Vienna, Austria;* [b]*Institute of Transportation, TU Wien, Karlsplatz 13/230-2, 1040 Vienna, Austria*

*christian.kasess@oeaw.ac.at*

In addition to the desired acoustical effect, noise barriers also pose an optical barrier obstructing the view. Transparent elements, which are typically highly reflecting, could reduce this visual barrier effect. Unfortunately, in particular for railway traffic highly reflecting surfaces oriented towards the noise source may produce considerable multiple reflections between train and barrier, potentially reducing the insertion loss of the barrier. Current noise mapping methods such as the EU directive 2002/49/EC (Annex II) provide means to approximately consider such reflections assuming parallel planes and a constant absorption coefficient of the barrier. There is, however, the need for detailed investigations how the placement and dimensions of sound hard elements, the distance of the train to the barrier, the train cross- section, the distance to the observer, and other variables influence the insertion loss of the barrier. The work presented here aims at investigating the different factors systematically using 2.5D-boundary element (BE) calculations as well as an extensive measurement campaign where acoustic passby measurements of 4 different barrier variants combining highly absorbing and reflecting materials were performed. These measurements provide the basis for the validation of the BE calculations. Based on this calibration, simulations of many different configurations are performed and a simplified, practical calculation model will be derived which can be used in noise mapping applications. Measurement results show reductions in the shielding effect of the barrier, depending on the position and the train type. Concerning the simulations, the BE method proved suitable for modeling the effects of reflecting panels, however, in particular the source positioning can produce considerable variance in the BE calculation results and thus the derivation of an adequate source model is the current challenge.

**Nonlinear Behavior of Perforates**

***M.H. Jensen***
*COMSOL, Diplomvej 373, 2800 Kgs. Lyngby, Denmark*
*mads@comsol.dk*

Perforates or microperforated plates (MPPs) are found in a large array of acoustic applications where controlled damping is a requisite. Applications range from room acoustic treatments, grills used in loudspeakers, to diaphragms and backplates in microphone applications. Damping is a result of thermal and viscous dissipation in the acoustic boundary layer. As the perforates gets smaller or sound pressure levels higher, the local particle velocity will increase and lead to additional resistive losses due to local nonlinear effects like vortex shedding. This results in the transfer impedance becoming level dependent. Here, linear and nonlinear physical phenomena are modeled in detail using a full thermoviscous description of the governing equations and solved using the finite element method (FEM) in COMSOL Multiphysics. Results are obtained for a series of configurations and also compared with analytical and semianalytical results.

**Measured and Computed Reflection from a Finite Plate**

***S.U. Vikøren and U.P. Svensson***

*Norwegian University of Science and Technology, O.S. Bragstads plass 2, 7034 Trondheim, Norway*

*peter.svensson@ntnu.no*

Inaccuracies in computation of sound propagation through complex geometries are related in part to inadequate modelling of edge diffraction. Prediction methods need to be evaluated against reference results which could be benchmark measurements, of adequate accuracy. This paper presents impulse response measurements for a rigid 1x1m plate in an anechoic room. The plate was made of 20 mm plywood with a smooth and hard surface. The measurements used a 25 mm diameter tweeter as a source and a 1/2-inch microphone. Using fixed source and receiver positions, the plate was suspended from the ceiling, and rotated in steps of 15 degrees around its vertical symmetry axis. This gave cases with and without specular reflections, with and without direct sound. A signal-to-noise ratio larger than 30 dB was found for the frequency range of 125 Hz to 20 kHz. Calculations were made with the Matlab Edge diffraction toolbox, modeling the loudspeaker as a point source, and the plate as thin or thick. Measured and predicted 1/6-th octave band levels were compared and for the 1/6-th octave band from 125 Hz to 16 kHz, and across 12 plate rotation angles, the 2.5\% and 97.5\% percentiles and median values were computed. For one of the two setups, using a thick plate model and diffraction order 3, the percentile values were -1.1 dB and +1.8 dB, respectively, with a median value of 0.36 dB. For diffraction order 2, and a thick plate model, these results were only marginally worse, with the percentile values -1.4 dB and +1.8 dB.

**Mesh2HRTF / NumCalc: An Open-Source Project to Calculate HRTFs and wave scattering in 3D**

***W. Kreuzer*[a], *K. Pollack*[a], *P. Majdak*[a] *and F. Brinkmann*[b]**

[a]*Acoustics Research Institute, Austrian Academy of Sciences, Wohllebengasse 12-14, 1040 Vienna, Austria;* [b]*Technische Universität Berlin, Einsteinufer 17c, 10587 Berlin, Germany*

*wolfgang.kreuzer@oeaw.ac.at*

Mesh2HRTF (mesh2hrtf.sourceforge.io) is an open-source project originally developed to provide users with an easy-to-use platform for the calculation of the head-related transfer functions (HRTFs) based on the boundary element method (BEM). However, its BEM module NumCalc with its easy-to-use input file is flexible, and thus it can be used as a stand-alone application for solving the Helmholtz equation in various other contexts such as calculating the sound field inside of a car or a duct, or the calculation of the scattering of sound waves in an open field around an object in 3D. NumCalc is a combination of the collocation approach based on the Burton-Miller formulation of the boundary integral equation and the fast multipole method (FMM). In this talk we will discuss novel features of NumCalc such as the definition of frequency-dependent admittance boundary conditions, and point to its current limitations with respect to the numerical accuracy and memory consumption.

**A Digital Twin for full-field monitoring in multi-channel control with applications to direct field acoustic testing**

***A. Garcia De Miguel*[a], *O. Atak*[b] *and M. Alvarez Blanco*[a]**

[a]*Siemens Digital Industries Software, Research Park 1237, Interleuvenlaan 68, 3001 Leuven, Belgium;* [b]*Siemens Digital Industries Software, Francis House, 112 Hills Rd, CB2 1PH Cambridge, UK*

*alberto.garcia_de_miguel@siemens.com*

This work proposes the use of a Digital Twin of the test plant of direct field acoustic excitation (DFAX) tests for on-site model update and full-field monitoring. DFAX is an increasingly adopted method to conduct environmental acoustic tests of space hardware which avoids the need of large reverberant rooms by exciting the structure via direct acoustic radiations generated by a set of high-powered loudspeakers. Based on previous research, a comprehensive Digital Twin of the electro-acoustic plant is utilized to simulate the frequency responses of the main sub-systems involved in the control system, including digital audio processing, loudspeaker performance, sound propagation and structural response. In this study, the main sources of modeling uncertainty in DFAX tests are first studied via sensitivity analyses. Then, a model update approach is proposed to tune the electro-acoustic contributions in the simulation based on experimental data obtained during the test. The outcome of this step is the availability of an improved Digital Twin that represents with higher fidelity the responses of the physical system. An evaluation of the gains in model accuracy is presented via comparison with measured data, demonstrating the validity of the proposed approach for DFAX. Such model can be subsequently linked to the multiple-input multiple- output (MIMO) control loop to conduct a hybrid virtual/physical test in which the field responses are measured not only in a few physical sensors, but over the entire test volume with full-field capabilities. Consequently, the resultant Digital Twin enables test engineers to monitor the system behavior at all locations, thus identifying with higher confidence the appearance of pressure fluctuations due to acoustic reflections and standing waves, as well as areas where the structural integrity might be at risk during the test.

**Spatial Transformations Effect to Soundscape: The Case of Istanbul Land Walls**

***Z.S. Akdemir***[a] ***and Ö.E. Aktuğlu Aktan***[b]

[a]*Yildiz Technical University, Barbaros Bulvarı Yıldız kampüsü, 34349 Istanbul, Turkey;* [b]*Yıldız Technical University, Barbaros Bulvarı Yıldız kampüsü, 34349 Istanbul, Turkey*

*zeynepsenaakdemir@gmail.com*

Sound atmosphere gives cues about environmental perceptions and memories. Land walls of Istanbul plays an important role in Istanbul's historical, cultural, sensory, and spatial memory. The establishment of the Ottoman Empire, land walls lost their defensive function and became a part of civil life. The land wall transformation has mostly been detached from the positive perception and cultural context of open green spaces. This perception replaced them with unplanned and uncontrolled spatial transformation (Kıran, 2019). As a result of highways and uncontrolled constructions, Istanbul land wall environment sound atmosphere was adversely affected. Especially in the region between Yedikule and Topkapı, it intensified the population and transformed the demographic structure. In this article, today's sonic and perceptual environment around the Istanbul land walls will be studied. This research aimed to contribute sonic perception and experience of the people in the field. The effect of the determined spatial transformation on the sonic landscape comfort in the adult user group will be investigated. While defining sonic landscape comfort and perception, the standard methodologies described in ISO/TS 12913-2 (2018) will be used. Surveys, sound walks, on-site sound recordings and listening to volunteers under special conditions methods will be used. Many types of data gathering methods will be use. The obtained data will be separated with the help of analysis programs. The soundwalking and spatial acoustic comfort obtained; hence the outcome will be evaluated by classifying the survey data. The result will be evaluated in terms of identified sound source, affective quality, and overall sound environment.

**Heart rate-based dynamic sound intervention - a pilot study**

***S. Raahede, H. Holm and S.H. Nielsen***

*SoundFocus ApS and Aarhus University, Sindalsvej 36b, 8240 Risskov, Denmark*

*sr@soundfocus.dk*

In this paper we present a pilot study on the prospects of working with dynamic sound intervention that adapts to physiological data. The study concerns an investigation of whether a dynamic synth-based soundscape, that adapts in tempo to the heartrate of a user, can lower and stabilize the heart rate faster than a 60 beats per minute (BPM) static synth-based soundscape. The assumption of the dynamic soundscape is that if the tempo constantly decreases with the adaptation of 2% below the actual heart rate, this will aid in bringing the user from a stressed to a more relaxed state (after 1 minute of physical exercise). To test the assumption, we included 8 test subjects in a repeated measures test with three different conditions of silence, static, or dynamic sound stimuli. The study furthermore includes short interviews on how the test subjects experienced the different conditions. Using the velocity of decreasing the heart rate as a measure, the influence of the three different conditions was estimated. A small tendency was found in the velocity being slightly faster in the condition of static and dynamic stimuli than in silence, but none of the results present any levels of statistically significance (p-value 0.16 at best). The difference detected between the static and dynamic stimuli is minimal, with the dynamic condition having a slightly faster heart rate decrement velocity. This is consistent with most of the subjects perceiving the two sound conditions to appear the same. The paper discusses these findings and presents critical thoughts on what could be considered in a further development of a concept like this. Even though the study cannot reject the null hypothesis, the paper brings a new perspective to the area of working with sound intervention in health.

# AUTHOR INDEX