ČESKÉ VYSOKÉ UČENÍ TECHNICKÉ V PRAZE

ČESKÁ AKUSTICKÁ SPOLEČNOST

5. STUDENTSKÝ AKUSTICKÝ SEMINÁŘ

25. ledna 2024

Praha

Redakce sborníku: Petr Honzík, Václav Vencovský, František Rund

Sborník 5. studentského akustického semináře

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URL: https://mmtg.fel.cvut.cz/student-acoustic-seminar/

Příspěvky otištěné ve sborníku nebyly redakčně upravovány.

ISBN 978-80-01-07281-3

Vydalo: České vysoké učení technické v Praze, 2024
Zpracovala: Fakulta elektrotechnická
Kontaktní adresa: Petr Honzík, Technická 2, 166 27 Praha 6
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24 stran
1. elektronické vydání

Obsah

Biografie prezentujících autorů

Ruoting Wang

I am ruoting Wang, and I am a recent undergraduate student majoring in electrical engineering and computer science at the Czech Technical University. My thesis is mainly dedicated to studying the effect of DPOAE in the field of acoustics.

Aneta Furmanová

Graduated from the Faculty of Electrical Engineering at the Czech Technical University in Prague (CTU, FEE) in 2023 with a Bachelor's Degree in Medical Electronics and Bioinformatics. In her bachelor's thesis, she focused on optimization of sound absorption in rectangular acoustic black holes. Currently a master's student of the same programme at CTU, FEE, with specialization on Bioinformatics. Now focusing on numerical optimization and machine learning in sound transmission in locally periodical structures.

Michal Reiser

Narozen ve Strakonicích v roce 2000. Po zakončení studia na Střední průmyslové školy v Písku nastoupil na bakalářský obor Kybernetika a Robotika na ČVUT FEL, který úspěšně absolvoval s bakalářskou prací zabývající se možným využitím neuronových sítí pro návrh rezonátorů. Nyní studuje magisterskou etapu tohoto oboru. Kromě účasti na projektech katedry fyziky také působí ve skupině Multirobotických systémů.

Jan Plaček

Jan Plaček was born in 2000 in Mladá Boleslav. He graduated from The Dr. Josef Pekař Gymnasium in Mladá Boleslav in 2019. The same year, he enrolled in the Czech Technical University, at the Faculty of Electrical Engineering. In 2022, he obtained a Bachelor's degree in Electronics and Communication. He has since continued his studies in the subsequent Master's program in Audiovisual Technology and Signal Processing. Currently, he is working on his master's thesis on the topic of Theoretical model of an electrostatic microphone with nonlinearity.

Jan Kubis

Absolvent magisterského studijního programu Řídicí a informační systémy na Vysoké škole báňské – Technické univerzitě v Ostravě. Nyní studuje v 1. ročníku doktorského studijního programu Akustika na FEL ČVUT. Téma disertační práce „Modelování kochleární mechaniky“.

Zuzana Štanclová

Born in 1999 in Zábřeh, she loved singing from a very young age but because of the encounter with certain voice problems, it was necessary to make an effort to the right usage of her voice. After graduating from Gymnázium in Zábřeh, she continued her studies at Palacky University in Olomouc where she obtained Bachelor's and Master's degrees in biophysics (both with honours). Now she is a Ph.D. student there under the supervision of Prof. RNDr. Jan Švec Ph.D. et Ph.D. working on various aspects of voice production in singers.

Abhishek Singh

Abhishek Singh is an acoustician with experience and research interest in sound measurements, metamaterials, noise cancellation, and piezoelectricity. He is a graduate of the Indian Institute of Technology, Mandi, India and currently pursuing his doctorate at the Department of Physics, FEE, CTU in Prague.

Miroslava Kozlová

Born on February 2, 1992, in Jindřichův Hradec, she obtained her master's degree in Physical Measurement and Modelling in 2018 from the University of South Bohemia in České Budějovice. Subsequently, she joined the Robert Bosch company as a test engineer and is currently working at the Competence Center for Noise, Vibration, and Harshness. Additionally, she is pursuing a remote Acoustics PhD program at the Czech Technical University in Prague.

Martin Novotný

Martin Novotný was born in 1997 in Prague, Czech republic. He received the M.Sc. degree in Electronics and Communication from the Faculty of Electrical Engineering, Czech Technical University in Prague (FEE CTU) in 2022. He recently works as a part of the Department of Radioelectronics (FEE CTU) Multimedia Technology Group. His main research interests are Virtual reality systems with regard to their usage for virtual acoustics, signal processing and compression algorithms of audiovisual signals.

ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Effect of swept-sine speed on distortion-product otoacoustic emissions

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1 INTRODUCTION

This work utilizing the novel Synchronized Swept-Sine methodology [3], conducts controlled experiments to understand the nuanced impact of varying the speed of swept sines on DPOAE measurements. The findings indicate that the DP-gram, remains roughly consistent across four different frequency sweep rates.

2 METHODS

The theory behind SSS signals is described in [1]. Its application for DPOAE measurements is then described in [2]. The input signal x(t) is a sum of two SSSs：

$$x(t) = \sin\left(\varphi_1(t)\right) + \sin\left(\varphi_2(t)\right)$$

The SSS technique allows for detection of these higher harmonics because those appear in the “virtual” impulse response yielded by [4]:

$$h(t) = F^{-1}\left\{\frac{Y(f)}{S(f)}\right\}$$

By converting DPOAE component into the frequency domain, we can construct a DP-gram.

For a discrete signal x[n] sampled over N points, the RMS value (P) is calculated as follows:

$$P = \sqrt{\frac{1}{N}\sum_{n=1}^{N}|x[n]|^2}$$

The formula to convert a pressure to sound pressure level in dB:

$$P_{dB} = 20log_{10}\left(\frac{P}{P_0}\right)$$

3 RESULTS

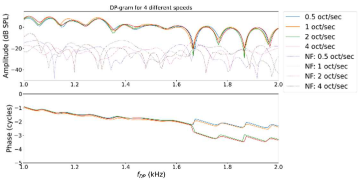

Figure 1. *Amplitude and phase plots of DP-gram from subject s039 for different speeds of sweeping.*

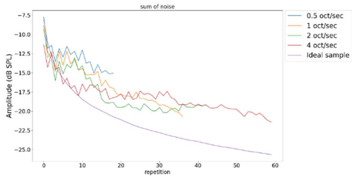

Figure 1. *Sum of noise level for subject s039 following Figure 1.*

From Fig.1 we can observed there is no systematic shift in the fine structure of DP-gram amplitude with sweep rate. It can be preliminarily concluded that the DP-gram is roughly the same at four different frequency sweep speeds.

In Fig.2, as the number of repetitions increases, there is a consistent decrease in noise levels. This observation holds true across different sweep rates. This similarity suggests that increasing the sweep rate up to 4 oct/sec may not necessitate a proportional increase in the number of measurements.

4 CONCLUSION

Our investigations revealed that the fine structure of DP-grams remains stable across various sweep rates. Significantly, we observed that this stability extends up to a sweep rate of 4 octaves per second. In addition, we found that the noise interference is similar at different sweep rates. Noise levels tend to decrease with increasing sweep repetitions.

ACKNOWLEDGEMENT

This work was supported by the project 23-07621J of the Czech Science Foundation (GACR).

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Machine learning for infinite periodic structure transmission modelling

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1 INTRODUCTION

Currently, the transmission for infinite periodic structures is modelled as follows. The geometry of one cell is parameterised by a radius function $R(x)$. In order to obtain the dispersion relation for the whole infinite periodic structure, firstly, the Webster equation [1] for a single unit cell is solved and via the Floquet's theory for periodic structure, we get the Bloch phase $\xi(f)$. This step is solved numerically. Then, from the Bloch phase, the dispersion relation is obtained as $\arccos(\xi(f)/2)$. Our aim is to obtain an analytical equation relating geometry and dispersion relation.

2 METHODS

It was decided to use the approach of data driven physics. The knowledge of the problem will be used, i.e. we do not have to model the dispersion relation directly, but we can take a step back and model the Bloch phase $\xi(f)$ instead.

In this work, we focus only on the first band gap to see if it is feasible to model the relationship with the use of machine learning.

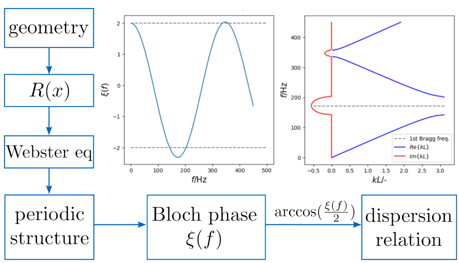

Figure 1: *From geometry to dispersion relation, process diagram.*

3 RESULTS

The geometry was parameterised by a radius function $R(x) = r_0 + A\sin^2[\pi g(x)]$, with $g(x) = ax^2 + (1-a)x$, where $A \in [-0.1, 0.1]$ is amplitude and $a \in [0, 0.5]$ is asymmetry. With the use of the current numerical solution, Bloch phases $\xi(f, a, A)$ for given frequencies f and geometries given by a, A was generated. By principal component analysis (PCA) we obtained from the dataset mean Bloch phase $\xi_0(f)$ and principal components $\xi_i(f)$, that are shared by the entire dataset, and a score $c_i(a, A)$ for each geometry. Note, that we focus on the first band gap, so it was sufficient to use just the first principal component.

The result of PCA is an array of values. In order to obtain the $\xi_0(f)$ and $\xi_1(f)$ in analytical form, the components were fitted with several functions by the Levenberg–Marquardt algorithm. For the scores $c_1(a, A)$, a library of possible expressions was created and LASSO was employed, which performs regularization and variable selection to get a sparse, interpretable equation.

Hence, we have a formula for the Bloch phase $\xi = \xi_0(f) + c_1(a, A)\ \xi_1(f)$, where $\xi_0(f), \xi_1(f), c_1(a, A) = c_1(A)$ are the mean Bloch phase, the first principal component of Bloch phase and the score, respectively. First estimation of these expressions is given in basic SI units as

$$\xi_0(f) = 2.087\cos(0.018f + 0.016) - 0.068 \tag{1}$$

$$\xi_1(f) = 0.072\exp(0.001f)\cos(0.015f + 0.4) - 0.058 \tag{2}$$

$$c_1(A) = -1197A^3 + 305.6A^2 - 1.098 \tag{3}$$

4 CONCLUSION

It is possible to extract an analytical equation relating dispersion relation and axis symmetric geometry parameterised by a radius function $R(x)$ from the data for given problem. This could be further used for proposing an optimized design for a desired band gap width, which was so far possible only with numerical optimization going repeatedly back and forth from geometry to the dispersion relation.

ACKNOWLEDGEMENT

This work was supported by the Grant Agency of the Czech Technical University in Prague, grant No. SGS24/053/OHK3/1T/13.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Manufacturing of narrowband acoustic silencers based on bound states in continuum for experimental validation

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1 INTRODUCTION

Ventilation systems for industrial and residential buildings always rely on a fan to blow air into the system. However, the airflow past the rotating fan blades, causes noise that can be very unpleasant to humans (e.g., in long exposition). Although the noise produced due to turbulence is predominantly broadband, several peaks in the spectrum might occur due to different mechanisms (e.g., at the blade passage frequency). The peaks correspond to whistling, which is generally more annoying than the broadband noise. If we tried to attenuate the broadband noise with acoustic foam, or other usual means, the resulting hydrodynamic pressure drop could render the whole ventilation system much less efficient (due to increased power requirements and consequently stronger noise emission). Instead, based on the numerical simulations presented in [1], narrowband acoustic silencers with high attenuation and low drag coefficient can be designed to target the most significant peaks in the spectrum. In this text, we present a workflow to create axis symmetric acoustic reactive silencer for the experimental validation of above mentioned numeric simulations.

2 METHODS

Since this was a pilot experiment to validate the numerical results, the production of the silencer had to be relatively cheap. Because the geometry is quite complex, 3D printing was determined to be the best method of production. This decision was made, because 3D printing is known to produce extremely complex shapes with relative ease and is also very cheap, additionally in the event of failure it can be fixed in a matter of hours.

The optimized geometry has the length of 1127 mm. Since the maximum size of produced object on the used 3D printer is 210 mm, the silencer had to

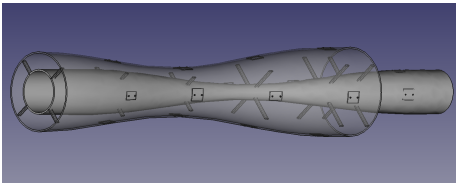

Figure 1: *3D render of the produced silencer.*

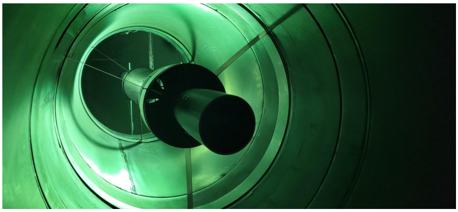

Figure 2: *Photo of the silencer moments before the experiment.*

be split into 6 parts. This means that inserts to connect the parts had to be added. Additionally, spacers to keep the walls of the silencers at the correct locations had to be added.

3 RESULTS & CONCLUSIONS

The resulting model can be seen in Fig. 1 and the photo of the produced silencer in the experimental apparatus is depicted in Fig. 2. Production and assembly of the silencer took approximately 5 workdays, from which about 90% was spent printing the silencer parts.

Subsequently, the silencer was sent to collaborating group at the Friedrich-Alexander University in Erlangen. The pilot measurements showed that the aerodynamical features (low drag) are very good but the acoustical features must be improved since the losses due to material roughness were not captured correctly. However, we can benefit from the work presented here in the future prototyping of corrected silencers.

ACKNOWLEDGEMENT

This work was supported by the Grant Agency of the Czech Technical University in Prague, grant No. SGS24/053/OHK3/1T/13.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Theoretical model of an electrostatic microphone with nonlinearity

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1 INTRODUCTION

This work deals with an analytical model that takes into account the coupling between the displacement of the membrane and the acoustic pressure inside the gap between the membrane and the backplate of a microphone, compared to a model obtained by numerical methods from COMSOL Multiphysics, as well as the influence of electrostatic nonlinearity on microphone sensitivity [1].

2 METHODS

Since the output voltage of the microphone [1]

$$u(t) = U_0 \frac{C_0}{C_\mathrm{p} + C_0} \left(\frac{\tilde{\xi}(t)}{h_\mathrm{g}} - \left(\frac{\tilde{\xi}(t)}{h_\mathrm{g}} \right)^2 + \left(\frac{\tilde{\xi}(t)}{h_\mathrm{g}} \right)^3 - ... \right), \tag{1}$$

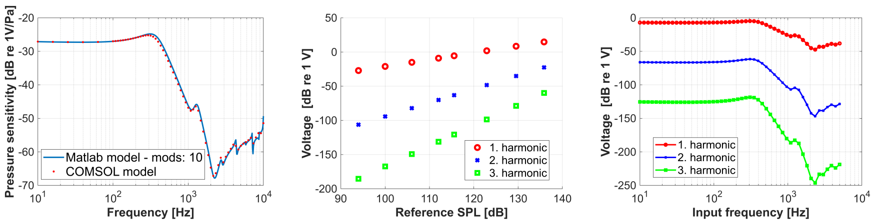
depends on displacement of membrane ξ, we need an analytical model. The model is described by the following system of equations [2],[3]

$$\begin{aligned} T(\Delta_\mathrm{r} + K^2)\xi(r) &= p_\mathrm{inc} - p(r), \\ (\Delta_\mathrm{r} + \chi^2)p(r) &= \zeta\xi(r). \end{aligned} \tag{2}$$

Since we are generally interested in the sensitivity of the microphone, we need to obtain the instantaneous membrane displacement from which the average membrane displacement can be further calculated.

3 RESULTS

Analytical results were calculated for an experimental transducer with a membranes radius of 18 mm. Figure 1(a) shows that the analytical model very closely matches the model obtained by numerical methods.



(a) (b) (c)

Figure 1: ***a)*** *The pressure sensitivities (as a function of frequency) of analytical model and numerical model* ***b)*** *harmonic components of the output voltage dependent on incident pressure ($f_0 = 1kHz$)* ***c)*** *harmonic components of the output voltage dependent on frequency ($p_{inc} = 10Pa$).*

When it comes to nonlinearities, their impact has thus far been addressed solely theoretically using eq.(1). For example the effect of frequency on the microphone's output voltage, figure 1(c), at some level of an incident pressure.

4 CONCLUSION

New analytical model has been developed and it matches closely with the numerical model. Using this model, the nonlinearities to the third harmonic frequency have been predicted.

ACKNOWLEDGEMENT

This work was supported by grant No. SGS23/185/OHK3/3T/13 of the Czech Technical University in Prague.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Undamping force in a nonlinear hydrodynamical cochlear model

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1 INTRODUCTION

Mammalian cochlea is an active system, this phenomenon can be demonstrated by the undamping feedback force, which is caused by the outer hair cells located on the basilar membrane (BM). This study shows that the undamping force could be obtained from the complex wavenumber.

2 METHODS

Data for this study were acquired from MATLAB simulation. Two-dimensional wave equation model created by Nobili and Mammano [1] was used. BM displacement was simulated for input harmonic signal at levels 0, 20, 40, 60, 80 and 100 dB *in vivo* and 100 dB *post mortem* (dead cochlea is linear system without gain). According to [2] and [3], complex wavenumber k can be obtained as the function of the density ρ, angular frequency ω and BM admittance Y_{BM}:

$$k = -2i\rho\omega Y_{BM}(x) = \frac{-2i\rho\omega}{Z_{BM}(x)} \qquad (1)$$

where i is imaginary unit. If we assume that ρ and ω are constant, k is equivalent to the BM admittance, which is reciprocal to BM impedance Z_{BM}. If the real part of the complex BM impedance is negative, the energy is added into the system, so the cochlear amplifier is functioning. This means the imaginary part of the relative wavenumber Δk is greater than zero [3]. Relative wavenumber is the difference between active and passive cochlear model complex wavenumber k.

3 RESULTS

Fig. 1 shows the waveform of the relative wavenumber for each simulated sound level, depending on distance from stapes.

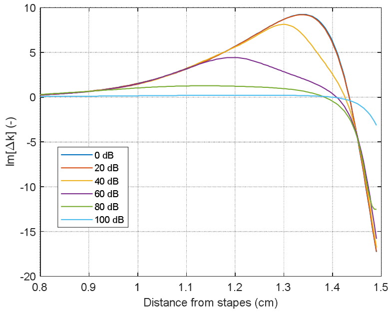

Fig. 1. *Imaginary part of the relative wavenumber*

The highest amplification is noticeable at low sound pressures. At 100 dB the force is still active, gain is very low, but not equal to zero. After the peak the energy from the system is dissipated, so the curves go down.

4 CONCLUSION

Method described in this paper can effectively show the location on the BM, where the undamping force is active and the energy is pumped into the system. Most important role of the cochlear amplifier is to help with hearing of the low-level sounds. Function of the cochlear amplifier at high sound pressures has been not explained yet and this as well as the energy dissipation in the system are the topics for our further research. The role of the amplifier at higher sound levels illustrates the fact, that people with hearing loss have problems with recognition of the speech in noise background despite the same speech without noise could be recognized normally.

ACKNOWLEDGEMENT

This work was supported by the project 23-07621J of the Czech Science Foundation (GAČR) and internal grant of the Czech Technical University in Prague SGS23/185/OHK3/3T/13.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Relationships between the vocal fold vibration parameters and voice intensity

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1 INTRODUCTION

Increase of voice intensity is related to changes in vocal fold oscillations. Knowledge about these changes has been insufficient. Small amount of previous studies report the increase of vibration amplitude, prolongation of closed phase and more prominent mucosal waves with rising intensity [1-3]. This study aimed at investigating the quantitative changes in vocal fold oscillations in a healthy female subject when continuously changing the intensity of voice at a constant pitch (i.e., the crescendo phonation).

2 METHODS

For purpose of this study, we recorded the phonations through the whole pitch range of the subject by high-speed videolaryngoscopy (7200 fps) synchronized with calibrated microphone signals. From the high-speed videos, we created videokymograms from which we obtained the parameters of vocal fold vibrations such as the vibration amplitude, open quotient Q_o, opening quotient Q_{og}, closing quotient Q_{cg}, speed quotient Q_s, normalized amplitude quotient NAQ, vertical phase difference VPD between the lower and upper margin of the vocal folds, and phase difference between the vibrations of the left and right vocal folds LRPD. The voice intensity was monitored through the sound pressure level (SPL) at the distance of 19.2 cm.

3 RESULTS

The changes of parameters and SPL in time were analysed simultaneously for every phonation at different pitches. An example is provided in Fig. 1. With increasing intensity there is the tendency of increasing vibratory amplitude and VPD, and decreasing Q_o, Q_{og}. The other parameters do not show a clear trend.

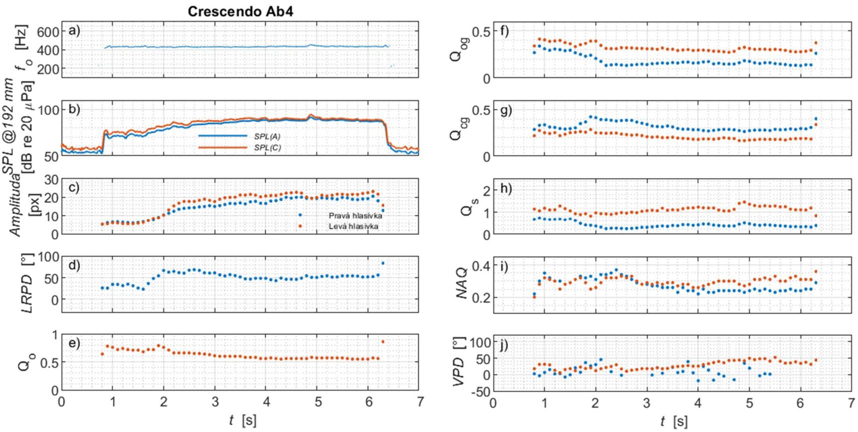

Figure 1. *SPL and vocal fold oscillatory parameters during crescendo phonation at the pitch Ab4 (415 Hz)*

4 CONCLUSION

The presented investigation of a healthy woman showed that most of the presented vocal fold vibratory parameters change with SPL of voice. Clear trends are not bvious in all the phonations, however, revealing on a complexity of human voice production [4]. The acquired unique set of data offers deeper insights into the kinematics of vocal fold oscillations.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Comparisional study of sound power measurement from sound intensity and LDV

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1 INTRODUCTION

Strength of any sound source radiating sound power in the near or far field can be calculated using the sound pressure and particle velocity on the desired surface or region. This method is a conventional way of sound power measurement and most of the standards were also designed based on this sound pressure and particle velocity measurement. While sound pressure based measurement is used very extensively, it has many limitations such as inaccuracy in low frequency region, requirement of known acoustic environment and selection of measurement surface etc. Not all but most of these standards provides the survey or engineering grade results. In the current study we proposed a sound power measurement technique via vibrational velocity of the sound radiating surface, which have the potential to improve the existing methods.

2 METHODS

Radiated sound power from a vibrating structure can be calculated using the Rayleigh's integral method (RIM)[1]. In the current study we have extended the RIM using the Cunefare and Koopman's theory [2] of sound power measurement by discretizing the sound radiating surface and than replacing the sound pressure in terms of the vibrational velocity using the impedance relationship.

$$\bar{P}(\omega) = \ Re[(v_e)^H[R](v_e)] \tag{1}$$

Where $\bar{P}(\omega)$ is sound power, v_e is complex vibrational velocity obtained experimentally and R is the radiation resistance matrix (RRM) calculated using the geometry of the discretized elements and the surrounding medium.

3 RESULTS

Steel plate of dimension 0.56×0.56 m^2, uniformly discretized into total 81 elements was used as a test surface. This plate was excited with the loud speaker on the back separated by air medium. Sound power radiating from this steel

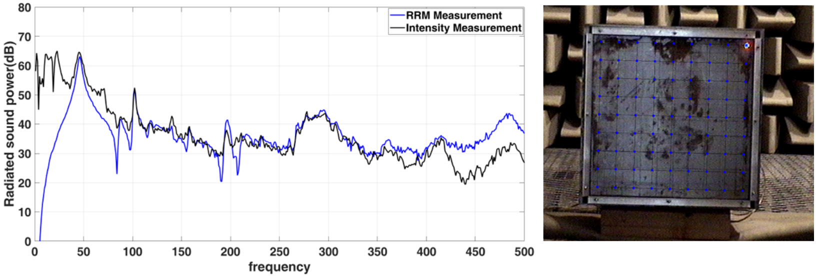

Figure 1: *Comparison of RRM with intensity method (left) and experiment (right).*

plate was calculated using the RRM method and intensity based sound power measurement according to ISO 9614-2 [3], as shown in fig.1. Vibrational velocity of the test surface for RRM method was measured using the laser doppler vibrometry and intensity based measurements were performed using the intensity probe. Analytical modelling of the RRM method was done in the MATLAB.

4 CONCLUSION

Comparison results from the two methods shows the good agreement which conforms the feasibility of the RRM method. While the results agreed well for low frequencies say below 400 Hz, discrepancies were found for higher frequencies. This overestimation of the sound power by the RRM method is expected due to the violation of the assumption made for the pressure calculation on disrcetized points such as the model stands well only when the dimensions of the element is comparatively small to the acoustic wavelength of our interest.

ACKNOWLEDGEMENT

This work was supported by HORIZON–MSCA–2021–DN, No. 101072598 and CTU Project no: SGS22/160/OHK3/3T/13 .

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Subjective evaluation of FSM sound using a pairwise comparison test

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1 INTRODUCTION

Pairwise comparison methods are well-known methods used for subjective evaluation of various product sounds. Also in the automotive industry, the product sound quality is an important topic and this method is usually included in so-called “jury testing” where manufacturers collect subjective data about the sounds of their car components. This contribution focuses on a pairwise comparison test of fuel supply module (FSM) sounds, which is a part of a more comprehensive psychoacoustic experiment. This experiment should lead to the development of a psychoacoustic model for evaluating the sound quality of the fuel supply modules.

2 METHODS

Two pairwise comparison tests were performed in a more complex listening test with FSM sounds. The first test was focused on subjective perception of loudness and the second on pleasantness. The entire listening test was created in the software module SQala (HEAD Acoustics). A set of pairs of FSM audio recording was played to each participant using a PC and headphones, each lasting 8 seconds. His/her task was to choose which of these two sounds he/she perceives as louder or more pleasant. A total of 10 sound pairs per test were evaluated (a total of 5 FSM sounds were used) and the calculation of Kendall’s coefficient [1] was used to check the consistency of the answers received. The subjective results of these tests, the scaled order of the 5 rated FSM sounds, were calculated according to the method of Guilford and Thurstone [2] and the correlation with the psychoacoustic metric loudness (ISO 532-1) was verified by the statistical *F*-test and *t*-test.

3 RESULTS

50 participants took part in both pairwise comparison tests. Unfortunately, 14 of them did not meet consistency in their answers, so only 36 valid answers were used for further analyses. This high number of excluded participants was probably due to the small difference between some of the FSM sounds related to loudness and pleasantness perception. Based on this subjective data, a scaled order of the tested FSM sounds was derived, as shown in Figure 1, and correlation analysis was performed between the results of both tests and loudness values (ISO 532-1), shown in Table 1.

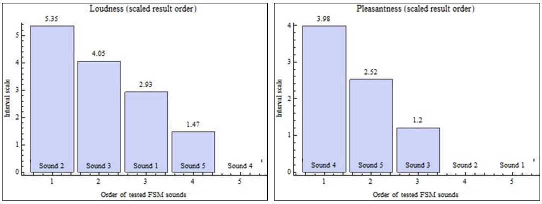

Figure 1. *Scaled result order of FSM sounds*

Sound No.	FSM type	Loudness N [sone]
1	diesel	3,59
2	diesel	4,43
3	diesel	4,14
4	gasoline	2,51
5	gasoline	3,08

Table 1. *Loudness (ISO 532-1)*

As can be seen from the charts above, sounds of gasoline FSMs were rated as more pleasant and quieter compared to diesel FSMs. The correlation between subjective results for loudness and pleasantness was confirmed at the 0.05 significance level, and the correlation between the loudness metric (ISO 532-1) and scaled result order from the pairwise comparison test focused on loudness was confirmed at the 0.01 significance level. These results showed that loudness perception was significantly related to overall pleasantness perception.

4 CONCLUSION

Performed pairwise comparison tests provide new information on FSM sound perception and support the development of conceptual model for FSM sound quality evaluation.

ACKNOWLEDGEMENT

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

25. 1. 2024

Implementation of Tests for Researching Spatial Hearing Errors in Virtual Acoustic Space

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1 INTRODUCTION

This paper presents a design of a subjective experiment that focuses on investigating spatial hearing errors using Virtual Acoustic Space (VAS). The most fundamental way to obtain information on the position of the sound source from acoustic stimuli is using binaural cues [1], [2]. Unfortunately, for some positions of the sound source, these binaural cues are equal. The equidistant points of the left and right ear lie on a cross section of a cone, commonly referred to as the cone of confusion [2]. Using this cone, we can describe many spatial hearing errors, such as front-back confusion, a phenomenon of locating a sound source in a mirror-rotated quadrant of the horizontal plane. These errors can be avoided, e.g. by rotating the head. In our research, we investigate the errors mentioned above and determine the influence of their occurrence in VAS on various parameters of the setting (such as sound source positioning or reaction time) by developing spatial hearing tests for existing application for sound source localization in VAS [3].

2 METHODS

For the purpose of the described research, a new set of 29 tests was created. The purpose of each created test is to select a singing bird based on binaural acoustic stimuli (which is a broad-spectrum signal that represents bird chirping) calculated using a generic HRTF. The set created focuses primarily on the analysis of cone of confusion errors and front-back confusion errors, so the sound sources are placed all around the listener (both with random and given placement). A test scene with highlighted sound sources is presented in Fig. 1. To increase the statistical reliability of the results, a training sequence and randomization of the tests and sound sources were implemented.

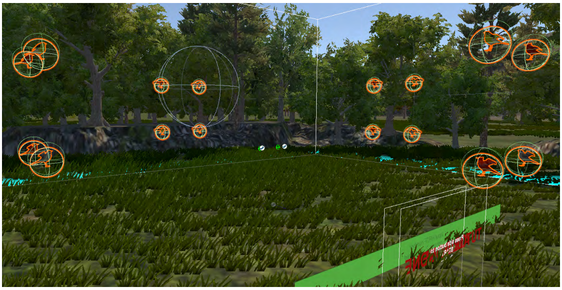

Figure 1: *Detail of implemented test case with highlighted possible sound sources*

3 RESULTS AND CONCLUSION

The test sequence was already carried out by a group of 8 people. The data obtained from this set of listeners clearly show the correlation between angular error and response time. Specifically, the previously mentioned errors (front-back confusion specifically) are more evident with shorter reaction times. From this it can be concluded that when fast reaction times are required, localization errors are much more likely to occur. But due to the complexity of spatial hearing, for general conclusion, it is necessary to perform subjective measurements on a larger set of listeners and perform a deep statistical evaluation of the results. As a part of the follow-up research, we would like to study deeply the effects on spatial localization error occurrence made by the response time of the listeners, the spectral distribution of the input signal, and usage of a personalized HRTF. To obtain complex information about the listener's behavior in the VAS, we would like to collect data about head and eye movements. The results obtained (from a larger group of people) could be used to model spatial hearing behavior.

ACKNOWLEDGEMENT

This work was supported by the Grant Agency of the Czech Technical University in Prague, grant No. SGS23/185/OHK3/3T/13.

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