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Obsah

Biografie prezentujících autorů

Šimon Skvaril

Po dokončeném středoškolském vzdělání na osmiletém gymnáziu v pražských Bohnicích se přesunul na vysokoškolská studia do Brna. Zde úspěšně završil bakalářský studijní program Audioinženýrství na VUT FEKT se svou prací, která se zabývá akustikou uzavřených prostor. Momentálně dokončuje práce na své diplomové práci pod vedením pana doc. Ing. Jiřího Schimmela v navazujícím studijním oboru. Mimo školní vzdělávání působí aktivně jako akustik pod firmou DISK.

Ondřej Klimeš

Ondřej Klimeš se narodil v roce 1996 v Praze. V roce 2021 získal titul Ing. v oboru Audiovizuální technika a zpracování signálů na Fakultě elektrotechnické ČVUT, kde nyní pokračuje v doktorském studiu. Věnuje se tématům spojeným s modelováním kochleární mechaniky.

Miroslava Kozlová

Narozena v roce 1992 v Jindřichově Hradci. Titul Bc. získala v roce 2015 na Přírodovědecké fakultě Jihočeské univerzity v Českých Budějovicích, studijní obor Matematika-Fyzika pro vzdělávání. Titul Mgr. získala v roce 2018 ve stejné instituci v oboru Fyzikální měření a modelování. V současné době studuje na Českém vysokém učení technickém v Praze, Fakultě elektrotechnické, v doktorském studijním programu Akustika a pracuje ve firmě Robert Bosch, spol. s r. o. v Kompetenčním centru pro hluk a vibrace.

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Narozena 4.9. 1996 v Prostějově. Už na základní škole se věnovala zpěvu a hře na hudební nástroje. V roce 2015 začala studovat bakalářský obor lékařská biofyzika na Univerzitě Palackého v Olomouci se zaměřením na hlasivky. Roku 2020 obhájila magisterský titul ve stejném oboru. V současné době je studentkou doktorského studia oboru biofyzika na Univerzitě Palackého v Olomouci pod vedením doc. RNDr. Jana G. Švece, Ph.D. et Ph.D.

Hugo Lehoux

Born 23.4.1990 in Caen, France. After a technical degree in sound engineering and audiovisual business, my deep interest in music acoustics led me to first obtain a Bachelor in Physics at the University of Caen (France), and then a Master’s degree in Acoustics at the University of Le Mans (France). During my Master’s degree, I had the opportunity to do an internship at the University of New South Wales, Sydney (Australia), where I worked on a project dealing with the human voice physiology and acoustics. I am currently a PhD student in the Voice Research Lab at the Palacký University in Olomouc, Czech Republic, where I am working on the broad topic of biomechanics and acoustics of human voice production.

František Blažek
Narozen v roce 1989. V roce 2016 získal titul Ing. v oboru Inteligentní budovy na Fakultě elektrotechnické ČVUT, kde nyní pokračuje v doktorském studiu parametrické akustiky. Věnuje se tématům spojeným se zpracováním signálů.

ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

20. 1. 2022

Design of subwoofer with cardioid radiation pattern

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1 INTRODUCTION

Most of current low frequency speakers (subwoofers) has omnidirectional radiation pattern thanks to large wavelenghts of low frequecy sound waves. This problem is used by using more ludspeaker units, and by using time delay and phase shift of signal more directional response is achieved.

This work proposes a design of single loudspeaker unit, which should, by using open transmission-line enclosure, achieve directional radiation pattern. [1]

2 METHODS

In software MATLAB with the use of k-wave toolbox a virtual model of speaker baffle is created. First simulations revealed, that the whole enclosure is beahving like tube with one closed end, thus unwanted amplification of the signal occurs.

Second model was created with absorbent materiál inserted in the rear part of the enclosure, which dampens the standing waves. The result of these computations shows Figure 1.

Beacause of the computational difficulty a scaled model of the loudspeaker was created, whose frequency response was measured in the close field in front of its membrane and in close field of enclosure opening. Figure 2 shows the differences of frequency responses before and after inserting the damping material and provides good evidence, that damping has significant effect on resulting frequency response. After the damping, the differences between the signals generated from both sides of memrane differ less than 3 dB, which suggests good subtractiuon of signals in the rear side of the loudspeaker enclosure.

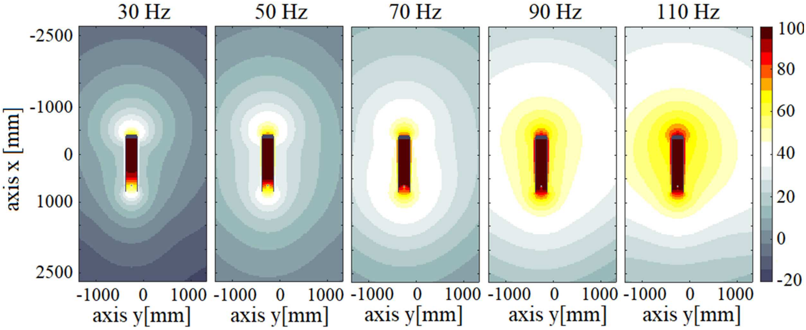

Figure 1. *Spatial distribution of acoustic pressure of damped model [dB spl]*

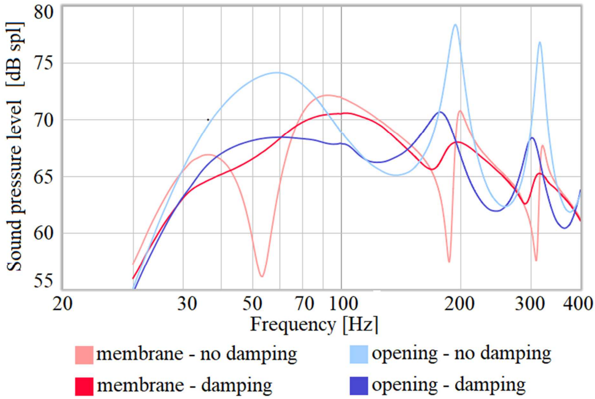

Figure 2. *Comparison of frequency resposes in close fields of membrane and opening before and after inserting damping material*

3 CONCLUSION

Computational model and scaled real-life measurments showed, that the proposed concept enclosure should work as intended. However properly sized enclosure should be built and its radiation pattern should be measured in the appropriate distance.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

20. 1. 2022

Middle Ear Muscle Reflex in Middle Ear Lumped-Element Model

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1 INTRODUCTION

Middle-ear muscle reflex (MEMR) involving the stapedius was thought to be activated by loud sounds (approx. 80 dB SPL). However, recent findings indicate that the threshold may be much lower (around 15 dB less i.e. 65 dB SPL) [2]. This work presents analysis of MEMR effect on forward and reverse transmission, and pressure measured by a probe in the external auditory canal. MEMR is simulated in a middle ear model [1] by increasing the stiffness of the stapedius.

2 METHODS

By using a lumped-element model, we predict the effect of middle ear muscle reflex on forward transmission p_c/p_t, reverse transmission p_t/p_c, pressure changes in the ear canal p'_t/p_t and residue calculated as a difference $p'_t - p_t$ for excitation from within the cochlea. The middle-ear model is coupled with an ear canal model, which is approximated by a thin tube, i.e. a waveguide with a constant cross section.

3 RESULTS

During forward excitation, the transmission function p_c/p_t decreases at frequencies <800 Hz when increasing MEMR. Between 0.8-3 kHz, there is a slight increase in the transmission function compared to the state without MEMR.

During reverse excitation, the transmission function p_t/p_c decreases at frequencies <1.3 kHz when increasing MEMR. Between 1.3-4 kHz, there is a slight increase in the transmission function compared to the state without MEMR.

There is a more significant effect of MEMR on the pressure in the ear canal p_t in the case of reverse excitation. We can observe this effect at frequencies 0.6 kHz $< f <$ 5 kHz. Near 1 kHz, this effect is about two-times higher than for the forward excitation.

The larger MEMR effect on signal coming from within the cochlea can cause that the difference between p_t' and p_t can be dominated by the signal coming from the cochlea.

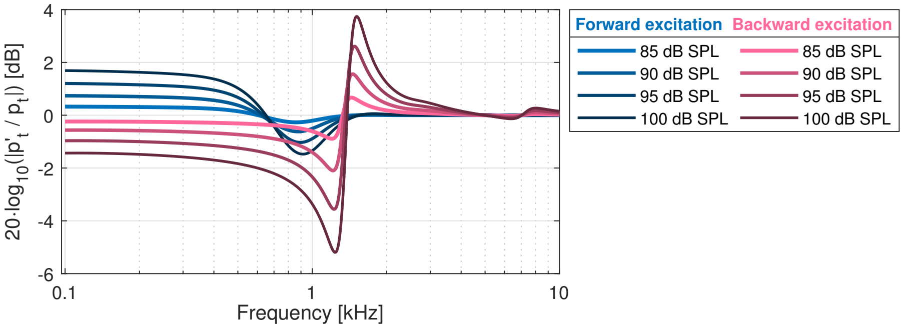

Figure 1: *MEMR effect on the pressure in the ear canal during forward and reverse excitation. MEMR is dependent on L_{st} intensity and is varied from 80 to 100 dB SPL. The blue lines represent the forward excitation, whereas the red lines depicts the reverse excitation.*

4 CONCLUSION

Simulations with a middle-ear model showed a more significant effect of MEMR on the pressure in the ear canal in the case of reverse excitation, i.e., a signal presented from within the cochlea.

5 ACKNOWLEDGMENT

This work was supported by the Grant Agency of the Czech Technical University in Prague, grant No. SGS20/180/OHK3/3T/13.

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First part of the psychoacoustic experiment with fuel supply modules

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1 INTRODUCTION

This contribution is focused on the psychoacoustic evaluation of fuel supply module sound. It is a common automotive part producing a specific sound, which is subjectively evaluated as a first part of more complex psychoacoustic experiment. This experiment provides new information about the perception of this product sound and will lead to the development of a psychoacoustic model for the objective evaluation of the fuel supply module sound.

2 METHODS

The first part of this psychoacoustic experiment consists of a Pre-test and a Listening test. The purpose of the Pre-test is to define the verbal space that describes the sound produced by fuel supply modules. This is done by collecting adjective pairs (limited verbalization method [1]). These pairs are then used for the 9-point rating scales definition for the Listening test (the adjective pairs used are listed in Table 1). In the Listening test a combination of two known methods is used – pair comparisons [2] and semantic differential [2]. Participants in this test must evaluate 5 sound recordings of fuel supply module and solve three tasks: paired comparison test (focused on subjective evaluation of loudness), assessment using subjective rating scales (semantic differential), and second paired comparison test (focused on subjective evaluation of pleasantness).

Adjective pair	Representation [%]	Attribute	Antonym
1	25,81	Disturbing	Not disturbing
2	19,35	Rattling	Calm
3	16,13	Low	High
4	12,90	Rough	Soft

Table 1. A*djective pairs from Pre-test used for subjective rating scales definition.*

3 RESULTS

In Pre-test a total of 171 adjectives were obtained and subsequently the adjective pairs were selected for rating scales definition in the Listening test (presented in Table 1). Adjective pairs Pleasant-Unpleasant and Quiet-Loud reached the highest percentage representation, and therefore pleasantness and loudness were chosen as the evaluated attributes for the paired comparison tests. So far, only 15 participants have taken part in the Listening test. Based on this fact, only a brief overview of the partial results of paired comparison tests is presented. Table 2 shows preferences for the evaluated records from the paired comparison tests transformed to order together with the calculated loudness values.

Record No.	Fuel type	Status	Loudness N [sone]		Preferences transformed to order	
			Left channel	Right channel	Loudness	Pleasantness
1	diesel	complaint	5,32	4,71	3rd	5th
2	diesel	complaint	5,72	6,92	1st	4th
3	diesel	serial	5,40	6,44	2nd	3rd
4	gasoline	complaint	3,45	3,88	5th	1st
5	gasoline	serial	4,45	4,69	4th	2nd

Table 2. *Calculated loudness values and order of recordings based on pair comparison tests.*

4 CONCLUSION

Thanks to the output of the Pre-test, a verbal space was defined for subjective evaluation of the fuel supply module sound. The brief overview of the Listening test partial results showed that diesel modules reach higher loudness values compared to gasoline modules, and this tendency is also noticeable in the preferences obtained from the paired comparison test focused on subjective loudness evaluation.

ACKNOWLEDGEMENT

This research was supported by the student grant ČVUT SGS22/160/OHK3/3T/13 “Acoustic measurements and applications”.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

20.1.2022

Empirical rules for deriving the mouth-to-microphone distance during laryngoscopic examination

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1 INTRODUCTION

Laryngoscopy is an essential tool for clinical examination of vocal folds. A rigid laryngoscope with an attached microphone is used to record and relate the vocal fold vibration characteristics[1]. The placement of the microphone is not standardized. This influences the sound pressure level (SPL) of voice which decreases with increasing distance from the mouth[2]. For reproducibility purposes, it is important to know the mouth-to-microphone distance at which SPL measurements were done. Therefore, we aimed at deriving empirical rules allowing clinicians to estimate the mouth-to-microphone distance in patients.

2 METHODS

120 adult clients served as subjects for our study. Two rigid endoscopes (90° and 70°) were used for the laryngoscopic examination. Patients' side photographs focused on the inserted laryngoscope were taken during patients' laryngoscopic examination (Figure 1).

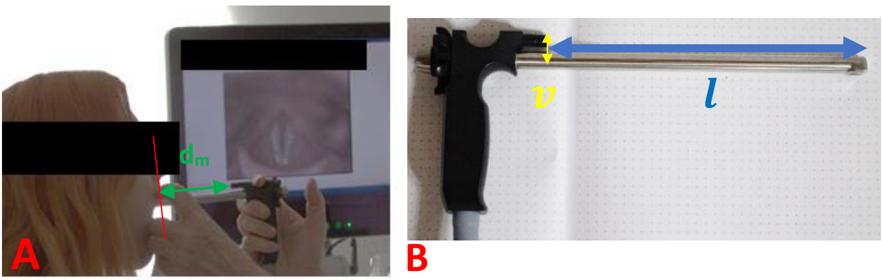

Figure 1 *A) Example of side photograph used for the measurement (d_m is the mouth-to-microphone distance). B) the length of the laryngoscope tube to the attached microphone (l); the distance between the laryngoscope tube and the attached microphone i.e. the height of the microphone placement (v).*

The mouth-to-microphone distance (parameter d_m, Figure 1A green line) and the position of the microphone (parameters l and v in Figure 1B) were measured from sizecalibrated photographs. The depth of laryngoscope insertion was derived from the difference between the parameter l and the measured mouth-to-microphone distance d_m.

3 RESULTS

Differences between males/females and 90°/70° laryngoscopes in the depth of laryngoscope insertion were statistically significant ($p < 0,05$). The average mouth-to-microphone distance and its variability can be calculated from the equation:

$$d = \sqrt{(l-h)^2 + v^2}, \tag{1}$$

where l and v define the position of the microphone and h is the empirically determined depth of the insertion of the laryngoscope into the mouth based on our measurements (Table 1).

	Patients' sex	
Used type of laryngoscope	**Male**	**Female**
90°	9,4 ± 0,6 cm	8,7 ± 0,6 cm
70°	9,7 ± 0,9 cm	8,9 ± 0,9 cm

Table 1 *The empirically determined depth of laryngoscope insertion into the mouth h for males and females and for the 90° and 70°laryngoscopes.*

4 CONCLUSION

The empirically determined values for the depth of laryngoscope insertion into the mouth allow estimating the mouth-to-microphone distance for any laryngoscope-mounted microphone without the need for any further special measurements on patients. Reporting the mouth-to-microphone distance is important for the reproducibility of SPL measurements.

ACKNOWLEDGEMENT

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

20. 1. 2022

Laryngeal and acoustic analysis of chest and head registers extended across a three-octave range: A case study

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1 INTRODUCTION

Voice registers belong to the most controversial topics in the field of voice science. They are assumed to be related to different laryngeal adjustments [1], but objective evidence has been insufficient. While the modal (or chest) register is usually associated with the lower pitch range, and the head (or falsetto) register with the higher pitch range, here we investigated a professional singer and singing teacher who claimed an ability to produce both these registers at every pitch, throughout her entire singing range.

2 METHODS

The singer performed short, separated phonations alternating between the two registers (further called chest-like and head-like) at all pitches from C3 (131 Hz) to C6 (1047 Hz). We monitored the vocal fold vibrations using high-speed videoendoscopy and electroglottography. The microphone sound was recorded and used for blind listening tests performed by the three authors (insiders) and by six “naïve” participants (outsiders).

3 RESULTS

The outsiders were able to correctly identify the registers in 64% of the cases, and the insiders in 89% of the cases, on average. Results from the visual and objective analyses are summarized in Fig.1. They revealed laryngeal, physiological and acoustic differences separating the low-, middle-, and high-pitched phonation regions indicating that the singer employed subtle laryngeal control mechanisms for the chest-like and head-like phonations on top of the traditionally recognized lowpitched chest and high-pitched head register phenomena. A more rapid glottal closure was objectively detected for the chest-like register throughout the entire range (see Fig. 1).

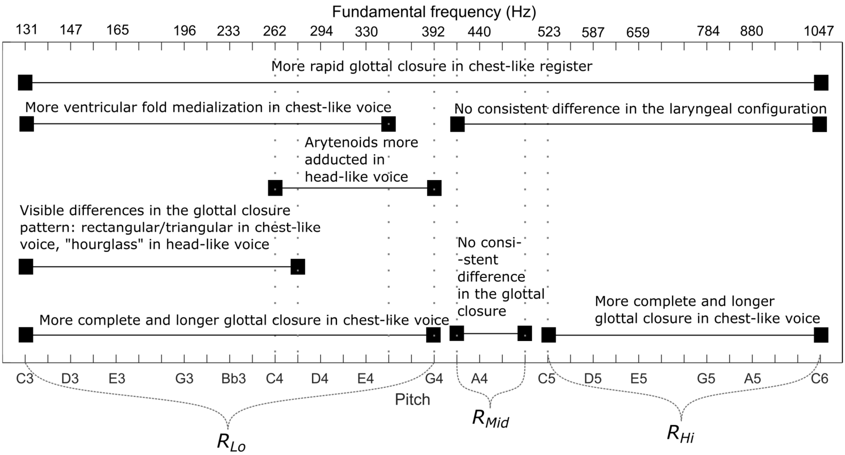

Figure 1. *Diagram summarizing the visual and objective differences observed between the chest-like and head-like phonations in the high-speed videos and the corresponding ranges where those differences appeared.*

4 CONCLUSION

These results suggest that the singer consistently differed the chest-like from head-like register across her whole pitch range by means of more rapid glottal closure. This was usually, but not necessarily, accompanied also by stronger adduction of membranous glottis. These register changes were not always easily perceivable by listeners, however. A more detailed version of this study was submitted for publication to the Journal of Voice.

ACKNOWLEDGEMENT

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Modulation techniques for parametric loudspeakers

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1 INTRODUCTION

Parametric array loudspeakers (PAL) transmit modulated sound waves in ultrasonic range. Due to nonlinear interaction, transmitted waves are then self-demodulated into higher frequency components and difference frequency components, which may be present in audible range. The concept was first proposed by P. J. Westervelt in 1962 [1] and then extended by H. O. Berktay's far field solution in 1965 [2]. In original studies, Dual sideband amplitude modulation technique was used in experiments, however more eligible techniques have been developed in subsequent decades.

2 THEORY AND METHOD

2.1 Berktay's solution

The Berktay's solution is one of the solutions to design modulation and preprocessing techniques for PAL. Primary sound field is expressed as

$$P_1(t) = P_1 E(t) sin(\omega_c t) \tag{1}$$

Where P_1 is primary wave amplitude, E(t) is the envelope function and ω_c is angular frequency of the carrier signal. Secondary wave acoustic pressure is expressed as

$$p_2 = \frac{\beta P_0^2 a^2}{16 \rho_0 c_0^4 \alpha z} \frac{\partial^2}{\partial t^2} E^2(\tau) \tag{2}$$

Where β is the nonlinear coefficient, ρ_0 is the air density, c_0 is the speed of sound in air, z is the distance from source, α is the absorption coefficient of primary waves and τ is the retarded time.

2.2 Modulation Techniques

Several modulation techniques based on Berktay's model were presented in the past. Dual sideband modulation may be simplest to implement but results with highest distortion. Single sideband modulation solves the THD problem but leaving the high IMD. Square root modulation was presented by T. Kamakura in 1984 [3]. In 2010, Modified amplitude modulation (MAM) was presented [4]. When a single input signal g(t) = cos(ωt) is considered, MAM can be expressed as

$$P_1(t) = P_1 \left\{[1 + m\, g(t)]sin\,(\omega_c t)\; + \sqrt{1 + m\, g(t)}\, cos\,(\omega_c t)\right\}, \qquad (3)$$

Where m is a modulation index. Using Taylor series, MAM1 can be expressed as

$$P_1(t) = P_1\; \left\{[1 + mg(t)]sin\,(\omega_c t)\; + \left[1 - \tfrac{1}{2}m^2 g^2(t)\right] cos\,(\omega_c t)\right\}, \qquad (4)$$

Where required bandwidth is limited. This can also be expressed in form of a block diagram which is then implemented in DSP module or simulated in Matlab.

3 RESULTS

A new PAL prototype was developed, modulation techniques were presented both theoretically and experimentally in Matlab and Sigma Studio. Different modulation techniques were analyzed, DSP of presented PAL was programmed and the acoustic output was measured.

ACKNOWLEDGEMENT

This work was supported by 2020 - 2022, SGS20/167/OHK3/3T/1.

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