ČESKÉ VYSOKÉ UČENÍ TECHNICKÉ V PRAZE

ČESKÁ AKUSTICKÁ SPOLEČNOST

4. STUDENTSKÝ AKUSTICKÝ SEMINÁŘ

19. ledna 2023

Praha

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Sborník 4. studentského akustického semináře

Redakce sborníku: Ing. Petr Honzík, Ph.D.
Ing. Václav Vencovský, Ph.D.
Ing. František Rund, Ph.D.

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Obsah

Biografie prezentujících autorů

David Vagner

Po dvou letech na víceletém gymnáziu se přesunul na střední průmyslovou školu v Č. Budějovicích, aby zde rozvíjel své znalosti v elektronice, která ho zajímala již delší dobu. Na FEL ČVUT v Praze úspěšně absolvoval bakalářské studium oboru Elektronika a komunikace, na kterém zahájil práci na projektu sférického mikrofonního pole. V současné době na projektu stále pokračuje při studiu stejnojmenného magisterského programu na stejné fakultě a projekt plánuje použít jako předmět jeho diplomové práce.

Jan Šedivý

Narozen v roce 1998 v Českých Budějovicích. Po úspěšném zakončení studia na českobudějovické střední průmyslové škole nastoupil na České vysoké učení technické v Praze, kde nyní dokončuje magisterskou etapu oboru Elektronika a komunikace. Zajímá se o mikrokontroléry a embedded systémy.

Martin Novotný

Martin Novotný was born in 1997 in Prague. He graduated with honours from the High Technical School of Communication Means in Prague, specialisation in Film and Television Production, in 2016. After that, in 2022, he obtained an Electrical Engineering degree from the Czech Technical University in Prague (Faculty of Electrical Engineering). His field of study was Electronics and Communication, specialising in Audiovisual Technology and Signal Processing. He wrote his Master’s thesis on spatial hearing, specifically about sound localisation in Virtual Acoustic Space. He enrolled in doctoral studies, specialising in Acoustics. He works as a part of the Department of Radioelectronics (FEE CTU) research group that studies human hearing and the application of the findings.

Jiří Bečka

Narozen v roce 1996 v Praze. V roce 2022 získal titul Ing. v oboru Budovy a prostředí na Fakultě stavební ČVUT v Praze, kde nyní pokračuje v doktorském studiu na oboru Akustika. Školitelem je prof. Ing. Ondřej Jiříček, CSc. (FEL), školitelem-specialistou Ing. Jiří Nováček, Ph.D. (FSv). V rámci své disertační práce se Jiří Bečka věnuje využití obnovitelných a recyklovaných materiálů ve zvukově izolačních prvcích. Mimo školu je součástí firmy KONTRAHLUK, s.r.o. – specializované společnosti se zaměřením na oblast hluku a akustiky.

Balaji Ramdas
I am Ing. Balaji Ramdas, currently doing my Doctoral studies in Acoustic Holography at CVUT, Prague. An aspiring Acoustician, working on sound beyond imagination.

Petra Veselá
Studentka 2. ročníku kombinované formy doktorského studijního programu Akustika, navrhované téma disertační práce "Modely binaurálního slyšení". Absolventka programu International Master's Degree in Electroacoustics (2018, Le Mans Université, Francie). Od roku 2018 pracuje ve švédské firmě Dirac Research v oboru zpracování zvukového signálu.

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.

ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

19. 1. 2023

HW for microphone array

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1 INTRODUCTION

The bachelor's thesis [1] and article that covers the progress that followed it [2] describe the project of a spherical microphone array

2 METHODS

The attempt to solve all the electronics-related issues was made by designing a new PCB for the auxiliary electronics, which can be seen in Fig. 1. Signal integrity problem was approached by adding signal buffers for all important signals on the board and reordering the wires. The previous counter that was too slow was replaced by another type (CY74FCT191ATSOC [3]), which should deliver the divided-frequency signal with a much shorter delay. The new PCB was tested on a digital oscilloscope.

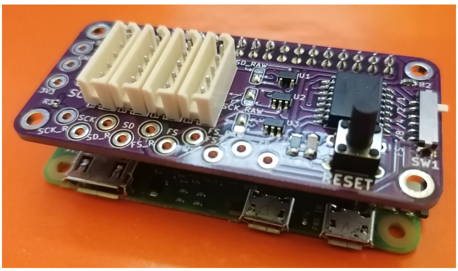

Figure 1: *New PCB with auxiliary electronics for the connection of microphones to the already connected Raspberry Pi Zero 2 W*

3 RESULTS

The measured delay of buffers that can be seen in Fig. 2 was around 1.65 ns, which is much less than one-half of the minimal period of the SCK (Serial Clock) signal (the minimal period is around 40.7 ns). The input-to-output time of the frequency divider, as seen in Fig. 3, is measured to be around 9.5 ns, which is again much less than half of the minimal period of the SCK signal.

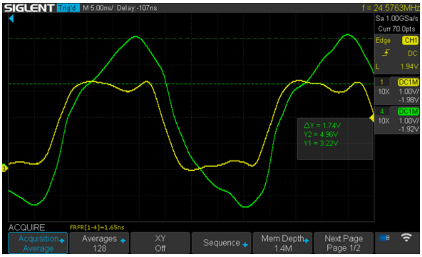

Figure 2: *Input (yellow) and output (green) signals of one of the buffers with measured delay of 1.65 ns.*

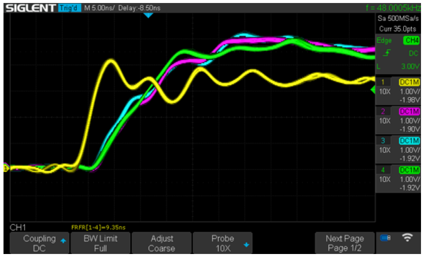

Figure 3: *Input (yellow) and output (other colors) signals of the counter used as a frequency divider with measured delay of 9.35 ns.*

4 CONCLUSION

All of the measured results are within acceptable ranges of delay, which seems to indicate that the board will be functioning well. The board should be soon tested with real equipment (connected to microphones and a Raspberry Pi) and thereby reveal its capabilities.

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

19. 1. 2023

Communication with MEMS microphones using FPGA

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1 INTRODUCTION

MEMS (Micro Electro-Mechanical System) microphones are electronic components that contain not only a microphone, but additional electronic circuits that process the analog signal from it, in one surface-mount package. The goal of this work is to read sound signal from MEMS microphones with TDM or I^{2}S [1] interface using a Field Programmable Gate Array (FPGA) and stream it to a computer through USB (Universal Serial Bus).

2 METHODS

The core parts of this project were realized on the Digilent Nexys 3 FPGA development board. For the USB communication using FPGA, an extra USB microcontroller chip branded FX2LP [2] was used. The FPGA logic for reading the TDM and I^{2}S interfaces, as well as sending the data out via USB FIFO bus of the microcontroller, was written in the VHDL language.

3 RESULTS

A data flow chain fot the sound data from the MEMS microphones to the computer was implemented, as shown in figure 1.

On the computer, a program written in C++ is accessing the USB device (using the `libusb` library). It periodically requests audio samples from the FX2LP. The USB microcontroller's firware, written in C language, is configured so that it sends audio samples via USB bulk transfer.

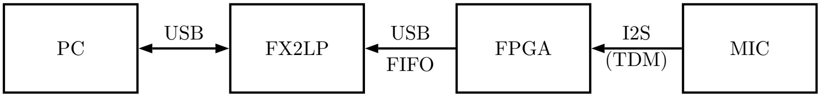

Figure 1: *Communication chain diagram – PC (Personal Computer) communicates with USB microcontroller FX2LP. FPGA writes sound data from microphone (MIC) to USB FIFO bus interface of the FX2LP chip.*

4 CONCLUSION

The I^2S and TDM communications were tested on real MEMS microphones. When the communication chain is assembled, audio from the microphones can be recorded on the computer. There still persists a problem with missing sound samples when using higher sampling frequency (96 kHz and above). The solution of this problem is one of the goals for future work, which is supposed to use isochronous USB transfer instead of bulk transfer.

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

19. 1. 2023

Sound sources localisation accuracy in VAS for various HRTFs

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1 INTRODUCTION

Spatial hearing is a complex phenomenon that allows humans to use ear signals and their experience to locate sound sources. Sound source position is usually described in polar coordinates, thus using the parameters azimuth φ, elevation θ and distance r. As mentioned in [1], azimuth determination is based on binaural cues and elevation is typically determined using spectral cues of a hearing system. For more efficient description and subsequent work, information about spatial cues for each listener is expressed in the form of Head Related Transfer Function (HRTF), which stands for the set of transfer functions of the head and pinna for various azimuths φ and elevations θ. This function is individual for each listener and could be used in Virtual Acoustic Space (VAS) for a binaural downmix of spatially localised sound sources.

2 METHODS

In the conducted experiment, the emphasis was on sound sources localisation accuracy in VAS for various HRTFs, specifically generic HRTF from the Club Fritz database (HRTF of the Neumann KU 100 microphone, for details, see [2]) and personalised HRTF measured for each listener (85 measurement points) at a designated workplace at the Department of Radioelectronics. Listeners then underwent the localisation test in VAS with described HRTFs. The goal of the test was to select the singing bird (only one sang at the time) in the presented layout (for both static and dynamic sources). In the final phase of the test, acoustic distractors were introduced.

3 RESULTS

The experiment was carried out on a set of 58 listeners. Each listener underwent a total of 14 localisation tests for each HRTF. Each localisation test saved data about the reaction time and angular error in structuralised `TXT` file. The experiment results indicated that personalised HRTFs are partially significant for dynamic sound sources (especially when acoustic distractors are present). For static sound sources, the significance was not so obvious. The test results (test no. 14 with dynamic sound sources and auditory distractor in the form of light rain) are visualised in Fig. 1 (the rest of the data will be forwarded by the author upon contact).

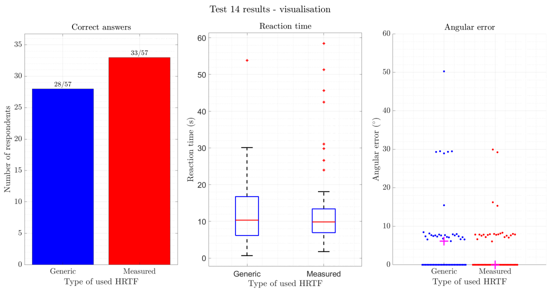

Figure 1: *Results of test no. 14. From left to right - correct answers, reaction time and overall angular error (blue - generic HRTF; red - personalised HRTF).*

4 CONCLUSION

The presented experiment demonstrated that personalised HRTF is significant for dynamic sound sources, a common state of affairs in consumer-oriented applications (such as VR systems). It was also proven that with personalised HRTF, it is possible to achieve comparable results (for sound sources mainly located on the front plane) with a lower spatial resolution of the measurement.

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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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

19. 1. 2023

Effect of floor finishing materials on impact sound pressure level

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1 INTRODUCTION

This experiment, which was conducted for a residential building, demonstrated how a floor finishing.material may affect final values of normalized impact sound pressure level L'_{n} (dB) when placed on a typical floating floor system. Field measurements were performed according to technical standards ČSN EN ISO 717-2 and ČSN EN ISO 16283-2. [1,2] Results were compared with ČSN 73 0532. The requirement in residential buildings is $L'_{n,w} \leq 53$ dB (or ≤ 55 dB before 12/2020). [3]

2 METHODS

The ceiling with a floating floor system was composed of a concrete slab (200 mm), an expanded polystyrene EPS (50 mm), a resilient material (20 mm) and an anhydrite screed (40 mm). In total 9 measurements were conducted – the first one was performed on a subfloor (and was considered as a referential), while the others varied in different floor finishing materials that were laid on the anhydrite screed. These were a laminate flooring (also with two different underlays – Measurement No. 2 to 5), a vinyl flooring (with underlays – Measurement No. 6 to 7), a carpet and a floor tiling (Measurement No. 8 and 9). All these measurements (except for the first and ninth one) were performed with a small sample (with the area at least 1 m^2) of a floor finishing material (this method is allowed in ČSN EN ISO 16283-2). Floor tiling was already firmly attached to the anhydrite screed. A tapping machine was selected as a source of the impact sound. [2]

3 RESULTS

The value of weighted normalized impact sound pressure level $L'_{n,w}$ (dB) generally decreased, but the last measurement ($L'_{n,w,9}$ = 58 dB) didn't come with terms of requirement ≤ 53 dB (or even ≤ 55 dB). An increase of L'_{n} (dB) was probably caused by a rigid interconnection between the ceramic floor tiles and flanking structures.

Measurement No. 1	49 dB	Measurement No. 5	48 dB
Measurement No. 2	48 dB	Measurement No. 6	45 dB
Measurement No. 3	48 dB	Measurement No. 7	45 dB
Measurement No. 4	49 dB	Measurement No. 8	44 dB

Table 1. *Results of all conducted measurements – $L'_{n,w}$ (dB)*

Usage of the vinyl flooring or the carpet as a floor finishing material is more effective in the attenuation of impact sound pressure level than the usage of the laminate flooring. These two materials have a better dynamic stiffness.

4 CONCLUSION

Also a diagram with courses of normalized impact sound pressure level L'_{n} was made. As shown by this diagram, adding of a finishing floor material has usually a positive effect on impact sound pressure level in a mid and high frequency range. This outcome was expected and measurements also proved the suitability of underlays that are laid under many floor finishing materials in order to decrease the transmission of impact sound. Moreover, the experimental measurement proved that the amplification of impact sound pressure level on lower frequencies is an issue which is in many cases unavoidable with present knowledge and technology, because the resonant frequency is increased by these floor finishing materials.
Nevertheless, it needs to be mentioned that mostly all used samples of floor finishing materials weren't attached firmly to the anhydrite screed, so results of this experiment might not turn out that well if they were indeed firmly attached.

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

19. 1. 2023

Study of specific regularization methods on acoustic holography reconstruction holography

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1 INTRODUCTION

This study compares the effects of several regularization techniques on source plane reconstruction using SONAH (Statically Optimized Nearfield Acoustic Holography). A technique of regularization is being utilized to stabilize the reconstruction in order to get over difficulties in forward and inverse reconstruction. Regularization techniques such Tikhonov, Morozov's discrepancy principle, and manually selected regularization parameters and the effect of Tukey window are utilized.

2 METHODS

(a). Statistically optimized NAH (SONAH): To address NAH's Fourier analysis-based limitations, SONAH was developed.

$$\begin{aligned}\tilde{p}(r) &= \tilde{\boldsymbol{a}}^T \alpha(r) = \boldsymbol{p}^T[(\mathbf{B}\mathbf{B}^H + \varepsilon\mathbf{I})^{-1}]^T \mathbf{B}^* \alpha(r) = \\ &= \boldsymbol{p}^T[(\mathbf{B}\mathbf{B}^H + \varepsilon\mathbf{I})^T]^{-1} \mathbf{B}^* \alpha(r) = \\ &= \boldsymbol{p}^T[\mathbf{A}^H\mathbf{A} + \varepsilon\mathbf{I}]^{-1} \mathbf{A}^H \alpha(r) = \mathbf{p}^T c(r)\end{aligned} \tag{1}$$

Where ε is the regularization parameter in the equation (1), which is the vital parameter of this research.

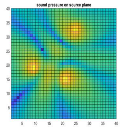
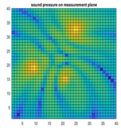
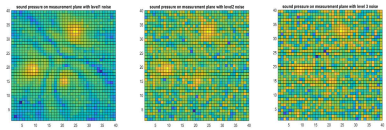
(b). Tukey Window: It is clear from the Tukey window's graphic that all plane edges are brought to zero seamlessly. Because the transition region of the taper is maintained to a minimum, the measured pressure over the source is unaffected.

(c). Manually chosen regularization parameters: For the purpose of comparison, regularization parameter was chosen manually without any formulation.

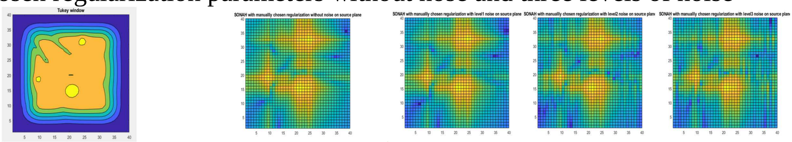
(d). Morozov discrepancy principle and Tikhonov method : Techniques for regularizing poorly posed issues. The problems of multicollinearity in linear regression, which frequently occurs in models with several parameters, can be very effectively mitigated.

2 RESULTS

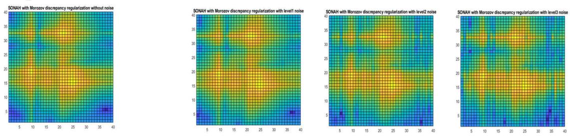
1. Distribution of sound pressure on source plane(a), measurement plane (b) with three levels of noise

(a)  (b)  (c) 

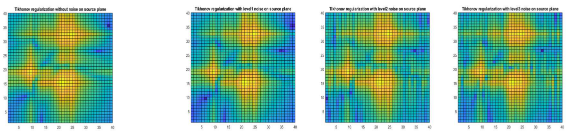
2. Reconstruction of sound field on source plane using Tukey window and manually chosen regularization parameters without nose and three levels of noise



3. Reconstruction of sound field on source plane using Morozov Discrepancy principle without noise and influence of three levels of noise



4. Reconstruction of sound field on source plane using Tikhonov regularization method without noise and influence of three levels of noise



CONCLUSION

Tikhonov's regularization endured best results comparatively. However, there are several regularization methods, and it cannot be generalized that Tikhonov regularization method is the best among all, as it depends upon the method used for reconstruction and also the sound field.

ACKNOWLEDGEMENT

This work was supported by CTU project No. SGS22/160/OHK3/3T/13 "Acoustic measurements and applications".

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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

19. 1. 2023

Models of Binaural Unmasking

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1 INTRODUCTION

Binaural hearing provides two main benefits – ability to localize sounds in horizontal plane and an improved performance in sound in noise detection. While the former phenomenon can be explained by a number of existing computational models, the latter still lacks a widely accepted understanding of underlying principles. Following is a brief introduction to the topic and a description of the prospective work.

2 THEORY

Binaural unmasking refers to an improved ability of detecting a useful signal in a noise background, in the presence of interaural differences in one of the mixture components. The quantitative measure of such improvement is termed Binaural Masking Level Difference (BMLD) and it can reach up to 15 dB when detecting a phase-inverted dichotic low frequency tone in a diotic broad-band noise, so called $N_0S\pi$ scenario[1]. Currently accepted theory is that the binaural unmasking stems from our ability to decode interaural phase differences (IPD) fluctuations generated by the mixture components interaction (see Fig. 1).

3 PLANNED WORK

Our team has previously developed a physiologically motivated computational model of a human superior olivary complex, which proved to be successful at predicting sound localisation in horizontal plane [3]. As a recent study showed[4], a two-channel model encoding the IPDs can indeed account for the BMLDs described by the experimental literature. Encouraged by such finding, we plan to extend the existing model by implementing a decoding component

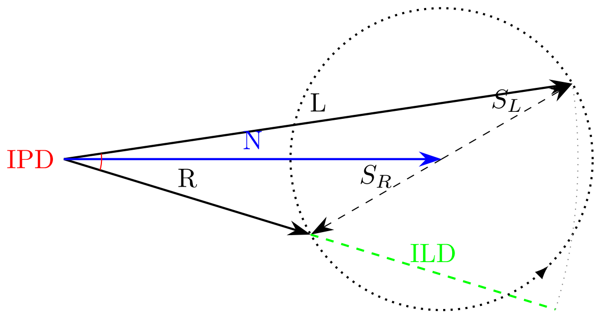

Figure 1: Interaural differences in $N_0S\pi$ scenario. Vectors S_L and S_R represent the relatively phase-inverted tonal components and vector N the broad-band diotic noise. Their vector sums L and R represent the sound mixture at the two ears. Rotating phase of the tone induces fluctuating phase and level difference. Adapted from [2].

so that the model can be used for predicting the BMLDs. Such a model could be valuable for hearing loss diagnostics or improved design of hearing aids.

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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ČESKÁ AKUSTICKÁ SPOLEČNOST

CZECH ACOUSTICAL SOCIETY

19. 1. 2023

Frequency jumps in excised larynges in anechoic conditions: A pilot experiment

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1 INTRODUCTION

Sudden fundamental frequency (f_o) jumps between chest and falsetto registers are recognized as bifurcation events and have been assumed to result from nonlinear-dynamic properties of the vocal folds interacting with acoustic resonances of the vocal tract and of the subglottal tract (ST) [1]. Modelling studies predicted that sudden frequency jumps might also occur without any influence of subglottal or vocal tract resonances [2], but these predictions have never been verified empirically with real larynges.

2 METHODS

We explored a novel anechoic (resonance-free) ST [3] and investigated these frequency jumps in two excised human larynges under two conditions: (a) anechoic and (b) subglottally resonant without a vocal tract.

3 RESULTS

When smoothly elongating the vocal folds, we observed consistent jumps in both anechoic and resonant conditions. The presence of a resonant ST did not result in more numerous jumps compared to anechoic conditions, however the resonant ST slightly altered the initial and terminating frequencies of the jumps in upward jumps (see Fig. 1).

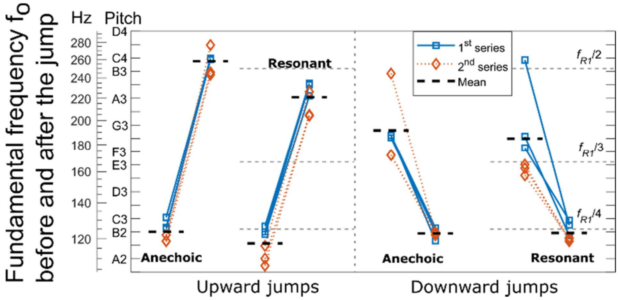

Figure 1. *Fundamental frequency (f_o) before and after each frequency jump for larynx #2 (69 years). The blue squares correspond to the first, and the orange diamonds to the second series of frequency sweeps. For the resonant cases, we also indicate the integer divisions of the first subglottal resonance frequency ($f_{R1} \approx 500$ Hz), $f_{R1}/2$, $f_{R1}/3$, and $f_{R1}/4$.*

4 DISCUSSION AND CONCLUSION

This study presents the very first experiments on f_o jumps in excised human larynges conducted in anechoic conditions. We prove that the jumps occur in excised larynges in anechoic conditions, confirming that those jumps are primarily caused by inherent laryngeal properties. These inherent properties differ among larynges: we found differences between our two larynges especially during upward jumps. These experimental findings should be considered when validating mathematical models simulating the nonlinear-dynamic behavior of the human vocal apparatus.

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