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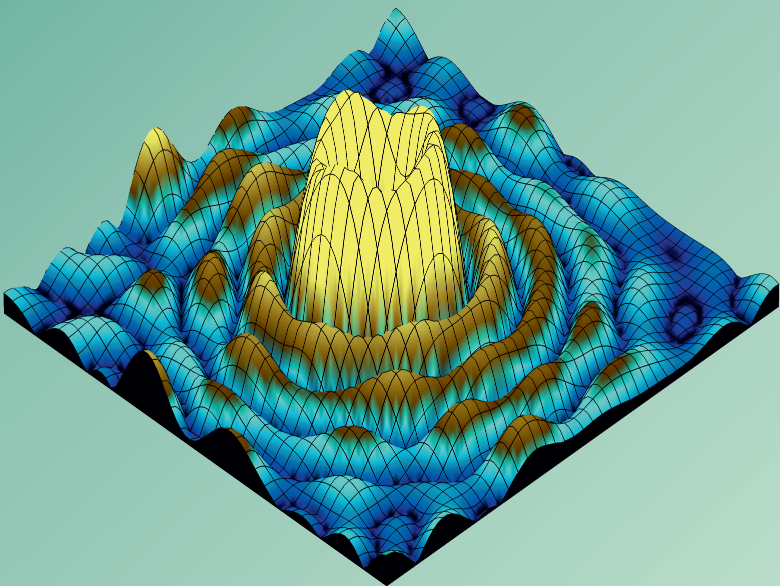

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Przedmowa

Akustyka jest nauką żywą i nieustannie rozwijającą się, a dynamiczny postęp techniczny stworzył dla akustyki olbrzymie możliwości badawcze w skali mikro i makro, dzięki czemu możemy uzyskać wiele cennych informacji o strukturze materii, materiałach, konstrukcjach, a także o organizmach żywych, w tym organizmie człowieka. Należy podkreślić interdyscyplinarny charakter akustyki, której rozwój wymaga nie tylko konieczności ujęcia zjawisk od strony fenomenologicznej, lecz także poszukiwania ich interpretacji matematycznej i fizycznej oraz rozpatrywania zjawisk akustycznych z punktu widzenia innych dyscyplin naukowych. Jest to możliwe przez powiązanie akustyki z osiągnięciami dyscyplin naukowych z zakresu nauk ścisłych, technicznych, biologicznych i medycznych, a także nauk humanistycznych.

Niniejsza monografia zawiera 55 recenzowanych rozdziałów wielu autorów, przedstawiających swoje najnowsze badania z zakresu akustyki biomedycznej, akustyki budowlanej, akustyki fizycznej, akustyki mowy, akustyki muzycznej, akustyki środowiska, akustyki wnętrz, badania materiałów, bioakustyki, elektroakustyki, hydroakustyki, przetwarzania sygnałów, psychoakustyki, ultradźwięków i walki z hałasem. Prace te zostały zaprezentowane 13–16 września 2016 roku na LXIII Otwartym Seminarium z Akustyki w Białowieży organizowanym przez Oddział Warszawski Polskiego Towarzystwa Akustycznego.

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Preface

Acoustics is the active and constantly developing science and dynamic technical progress has created tremendous opportunities for acoustics research at micro and macro scale. Thanks to it we can obtain many valuable information about structure of matter, materials, constructions and also about living beings including human body. The interdisciplinary nature of acoustics must not be forgotten as its development requires not only the approach to the phenomena from the phenomenological point of view but also the search of its mathematical and physical interpretation and the consideration of acoustical phenomena from the perspective of other science disciplines. It is possible by the interrelation of acoustics and the achievements of scientific disciplines in the field of science, technical, biological and medical sciences as well as humanities.

This monograph includes 55 reviewed chapters of many authors presenting the resent research in the field of biomedical acoustics, building acoustics, physical acoustics, speech acoustics, musical acoustics, environmental acoustics, room acoustics, non-destructive testing and evaluation, bioacoustics, electroacoustics, hydroacoustics, signal processing, psychoacoustics, ultrasound and noise control. The above mentioned works were presented at 63rd Open Seminar on Acoustics in Białowieża (13–16.09.2016) organized by the Warsaw Division of the Polish Acoustical Society.

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:
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Borys KOWALEWSKI*, Michał FERĘCZKOWSKI*, Ewen MACDONALD*, Torsten DAU*

A NEW PROCEDURE FOR AUTOMATIC FITTING OF THE BASILAR-MEMBRANE INPUT-OUTPUT FUNCTION TO INDIVIDUAL BEHAVIORAL DATA

The basilar membrane input-output function (BM I/O) in a healthy cochlea is highly nonlinear. One of the consequences of sensorineural hearing loss (SNHL) is a partial or full loss of this nonlinearity. Behavioral estimates of the individual BM I/O can be useful for modeling the impaired auditory system and, potentially, for clinical diagnostics.

Computational algorithms are available that mimic the functioning of the nonlinear cochlear processing. One such algorithm is the dual resonance non-linear (DRNL) filterbank [6]. Its parameters can be modified to account for individual hearing loss, e.g., based on behavioral, temporal masking curves (TMC) data. This approach was used within the framework of the computational auditory signal-processing and perception (CASP) model to account for various aspects of SNHL [4].

However, due to the computational complexity, on-line fitting of the DRNL parameters is difficult. Until recently, the parameters were manually adjusted and the fitting process was indirect. A new approach is described here, based on a search through a lookup table of pre-computed filterbank input-output functions.

The aim of this approach is to provide a fast, stable, and more objective fitting procedure.

1. INTRODUCTION

It is widely known that a single point on the basilar membrane of a healthy mammalian cochlea exhibits nonlinear, compressive behaviour when stimulated at its characteristic frequency. This is due to the gain provided by the outer hair cells (OHCs).

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Action of the OHCs changes the local mechanical properties of the cochlear partition, leading to sharpening of the cochlear tuning.

Sensorineural hearing loss is associated with a disruption of the OHC function. Major consequences include loss of sensitivity, impaired frequency selectivity and a loss of compression that leads to an abnormal growth of loudness.

Several methods have been developed to obtain psychophysical estimates of the cochlear compression. Oxenham and Plack [8] estimated the growth of masking (GOM) functions that reflect the amount of cochlear nonlinearity. To rule out the influence of suppression, they used a forward masking paradigm. The drawback of the GOM method is an excessive spread of excitation with increasing probe level. Nelson *et al.* [7] proposed an alternative procedure using a sinusoidal target fixed at a low sensation level (SL). They measured the levels of the off- and on-frequency tonal maskers required to just mask the probe at different masker-probe separations (gaps). The resulting functions of masker-level thresholds versus gaps were called temporal masking curves (TMCs). It is assumed that for each gap the off- and on-frequency maskers produce the same response at the characteristic place and that the cochlea responds linearly to the off-frequency stimulation. Pairing the on- and off-frequency TMC thresholds with the same gaps provides an estimate of the input-output function at that place.

1.1. MODELING THE COCHLEAR NONLINEARITY

There exist various computational models of the auditory periphery, based both on the transmission line [11] and filterbank [2,6,12] approaches. One of the most popular is the gammatone filterbank model [1,10], which is relatively simple and computationally fast but incapable of reproducing nonlinearities. As an alternative, Lopez-Poveda and Meddis [5] proposed a dual resonance non-linear (DRNL) human cochlear filterbank. It has been successfully used as a front-end to models of the auditory system, such as the computational signal processing and perception (CASP [3]) model. Jepsen and Dau [4] further modified the front-end to account for individual hearing loss. Behavioural estimates of the BM I/O at 1 and 4 kHz were obtained for each of their hearing impaired listeners and used to adjust the parameters of the nonlinear filterbank. Even though the DRNL is relatively efficient in terms of computation, automatic fitting of the filterbank parameters requires re-estimating the entire simulated BM I/O for each new set of parameters, which takes at least several seconds. Running an optimization algorithm with a large number of iterations would make the time required to perform the fit impractically long. So far, the fitting has been performed indirectly and required manual intervention. A new, direct and automated approach is presented here.

2. METHODS

2.1. MODEL DESCRIPTION

The model is a digital time-domain implementation of the dual resonance non-linear (DRNL) filterbank [5] with modifications suggested by [3]. The input to the model is a digital signal corresponding to pressure in pascals. It is then transformed to stapes velocity by outer- and middle-ear filters. Subsequently the signal follows two independent paths: a linear path and a nonlinear path, whose contributions are added at the output. The linear path consists of a linear gain g, a gammatone filter and a low-pass filter. The nonlinear path consists of a gammatone filter, followed by a broken-stick nonlinearity, another gammatone filter and a lowpass filter. The broken stick nonlinearity is governed by the following relationship:

$$y[i] = sgn(x[i]) \cdot min(a|x[i]|, b|x[i]|^c), \tag{1}$$

where parameters a and b control the location of the compression kneepoint and parameter c, the compression exponent, controls the slope of the compressive section. Together with the gain in the linear path, g they govern the level-dependent properties of each filter, including the input-output function. To account for individual hearing loss, these four parameters are adjusted, as described by [4]. Other parameters, such as the relative bandwidths of the gammatone filters or the number of cascaded filters in the linear and nonlinear path, are not subject to immediate change due to the simulated hearing loss and are described in [5].

Due to interactions between the two signal paths, given a parameter set, the filter I/O cannot be described functionally and has to be estimated using the paradigm described by [5]. In short, a tonal signal at filter center frequency and a low off-frequency masker are passed separately for each filter. It is assumed that, for each input signal level, the masking threshold occurs when the ratio between the peak output amplitudes of the signal and the masker is equal to (or just exceeds) 1.

2.2. DATA

The TMC data were taken from [4]. TMCs were measured at the two target frequencies 1 and 4 kHz. In each case, the masker was either at the same frequency as the target (the on-frequency condition) or at 0.6 times the target frequency (the off-frequency condition). The BM I/O estimates were obtained by plotting the off-frequency masker thresholds against the corresponding on-frequency thresholds, as suggested by Nelson et al. [7].

The original procedure for fitting the DRNL BM I/O to the data was as follows: First, a piecewise linear fit with one, two or three sections was fit to the data [9]. This served as a reference to estimate the kneepoint (if measurable). The DRNL parameters were then adjusted manually. First, *a* and *b* were adjusted so that the kneepoint of the simulated function matched the kneepoint estimated from the data or the lowest data-point. Subsequently, parameters *c* and g were modified to reflect, respectively, the estimated compression exponent and the extent of the compressive region.

The new procedure is based on pre-computed lookup tables of DRNL BM I/Os. The parameter space (range of values of *a, b, c* and *g*) was chosen based on the values that were previously encountered in the literature (see [4,5]). For each combination of the parameters, the filter output was measured for input levels from 0 to 100 dB SPL in 5 dB steps.

The individual behavioural estimates were compared with each entry of the lookup table and the best-fitting function (parameter set) was chosen using the minimum root mean square (RMS) error criterion. The original best-fit parameters from [4] were used to estimate the RMS error, which was then compared to the error obtained using the new method.

3. RESULTS

Figure 1 shows the input-output function estimated from the TMC data, together with the best-fitting model I/Os suggested by [4] and the I/Os obtained using the new method.

Table 1 compares the root mean square errors (in dB) of both fits.

Table 1. RMS errors of the BM I/O fits shown in Figure 1, expressed in dB

		HI1	HI2	HI3	HI5	HI7	HI8	HI9	HI10
1 kHz	Jepsen & Dau	2.5944	4.444	3.2524	2.9951	3.7486	2.2711	1.5935	1.9989
	New method	2.0589	4.3238	2.4654	2.0037	2.3766	1.1261	1.3677	0.9657
4 kHz	Jepsen & Dau	3.6919	1.4476	4.9660	5.5234	2.0570	4.0050	2.4814	4.2873
	New method	1.6220	1.3214	3.8110	4.4412	1.1024	0.7577	1.3316	1.3154

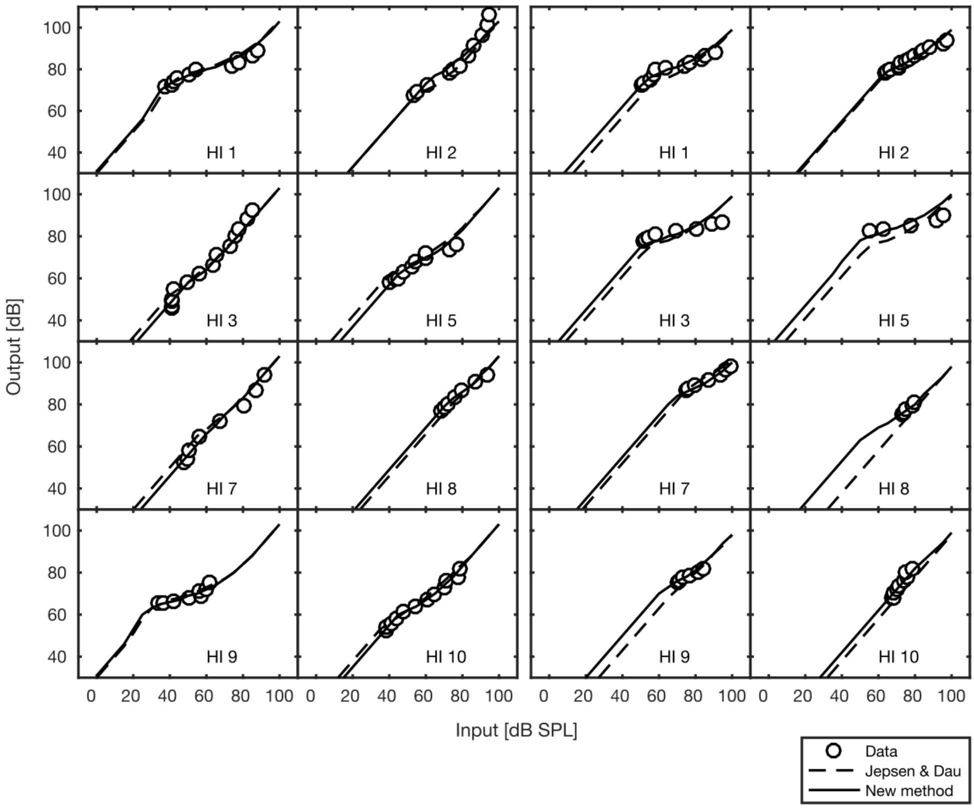

Figure 1. Behavioral estimates of the BM I/Os (circles), original fits from [4] (dashed lines) and fits obtained with the new method (solid lines). The left and right panels show the data and fits at 1 and 4 kHz respectively. The original numbering of subjects is preserved, with subjects HI 4 and HI 6 excluded.

4. DISCUSSION

Upon visual inspection, both methods yield similar BM I/O estimates. The estimated locations of the kneepoint are relatively close to each other within each subject and frequency. Most of the differences occur below the lowest measurable point (below the lower kneepoint). Nevertheless, the RMS error is systematically lower for the fits obtained with the new method, as indicated in Table I. The improvements range from fractions of dB (e.g. HI2) to several dB (HI8). As would be expected, bigger differences correspond to a greater mismatch between the curves shown in Figure 1.

There are two main reasons favouring the new method. First, the error (hyper)surface is non-convex and exhibits multiple local minima. This means that there may exist multiple combinations of parameters that yield a very similar, low-level error. Second, due to the interactions between parameters, the local minima can reside inside long valleys. The above factors make it difficult to find the globally optimal parameter set in a manual way. Concerning time-efficiency, the computation of the lookup table used here took approximately 6 hours of operation on a 24-core machine. Having computed the lookup table, fitting the parameters to each new data set takes only several seconds.

5. SUMMARY

A new method for automatic fitting of a nonlinear cochlear filterbank to behavioural data has been presented and evaluated on a dataset from the literature. The BM I/Os obtained with the new method are visually similar to the original fits but provide a systematic improvement in terms of the RMS error.

The greatest advantage, however, is the reduced time and effort needed to perform the fitting. Previously, the process required manual intervention. After each adjustment of parameters, the model BM I/O had to be calculated anew, which is time consuming. Moreover, it is difficult to accurately predict the outcome of changing the DRNL parameters on the resulting function due to the way they interact. Fitting for one subject at one frequency took several minutes at best, if a good “educated guess” was made for the initial set of parameters. Currently, fitting the model to new data can be realized in a matter of seconds, in a fully automated manner. Nevertheless it requires access to a pre-computed lookup table at the given frequency. Then, the accuracy of fit depends only on the resolution of the parameter space used to compute the table.

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