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Extraction of distortion-product otoacoustic emission source components and its relevance for objective audiometry

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Abstract

Distortion-product otoacoustic emissions (DPOAEs) arise as byproduct of nonlinear amplification in the cochlea in response to two stimulus tones of different frequencies. They are assumed to comprise mainly two components, a nonlinear-distortion component and a coherent-reflection component, which may cause wave interference in recordings with conventional, continuous stimulus tones, hampering the accuracy of DPOAEs when evaluating cochlear function. Here, DPOAEs are recorded with short stimulus pulses enabling the separation of the two source components in the time domain. Comparison with conventional stimulation endorses the importance of reliable source separation when using DPOAEs in clinical audiometry or research.

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1. Introduction

Otoacoustic emissions (OAEs) are sound waves emitted by the cochlea in response to acoustic stimulation which can be measured in the ear canal using a sensitive microphone [1] and are widely used in clinics and research to objectively investigate the functional state of the cochlear amplifier [2]. When stimulating the inner ear with a pure tone, a part of the stimulus is coherently reflected from a region close to its tonotopic place on the basilar membrane (BM), putatively due to impedance perturbations along cochlear partitions [1,3]. This mechanism gives rise to so-called stimulus-frequency otoacoustic emissions (SFOAEs) [4].

Stimulation with two simultaneously presented tones of frequencies f_1 and f_2 yields an overlap region of the two traveling waves close to the f_2-tonotopic place in case of suitable stimulus parameters, such as $f_2/f_1 = 1.2$ and $L_1 \geq L_2$. Intermodulation occurs in this region of overlap due to a nonlinearity in the opening probability of the mechanoelectrical ion channels in the outer hair cell (OHC) stereocilia. As a result, the electromechanical transduction of the OHC soma induces distortion components into the cochlear fluid [5], which become measurable in the ear canal

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as so-called distortion-product otoacoustic emissions (DPOAE). The most prominent DPOAE in humans occurs at the cubic difference frequency $f_{dp} = 2f_1 - f_2$ [3,5,6] and is assumed to comprise two components from different sites on the BM. Close to the f_2-tonotopic place the so-called nonlinear-distortion component emerges from the nonlinear interaction of the two overlapping traveling waves. This component induces a third traveling wave propagating to the tonotopic place of the distortion-product frequency f_{dp}. There, the second component arises due to coherent reflection, similarly to the generation of SFOAEs [4].

Spectral analysis represents a common approach to estimate the DPOAE amplitude at the cubic difference frequency f_{dp} from the averaged microphone signal. However, depending on the relative amplitude and phase difference between the two source components, wave interference may occur which becomes visible as a quasi-periodic variation of DPOAE amplitude with varying frequency [6]. A plot of DPOAE amplitude as function of stimulus frequency is typically referred to as a *DP-gram*. This variation is commonly known as DPOAE fine structure and has been shown to be a major limitation when estimating auditory thresholds by means of DPOAE input-output (I/O) functions [7]. Here, two recently developed methods utilizing short stimulus pulses instead of continuously presented pure tones are used to extract DPOAE source components from time-domain measurements and to reduce DPOAE fine structure in DP-grams.

2. Methods

2.1. Study design and DPOAE signal acquisition

DPOAEs were recorded unilaterally from six normal-hearing subjects between 23 and 53 years (mean age: 33.2 ± 11.1 years). The hearing status of each investigated ear was checked by otoscopy, clinical pure-tone audiometry from 0.125 to 10 kHz (Audiometer AT 900, Auritec, Medizindiagnostische Systeme, Hamburg, Germany), and standard tympanometry (Madsen-Zodiac 901, GN Otometrics, Münster, Germany). All pure-tone thresholds were better than 20 dB hearing level (HL) for each subject. Additionally, spontaneous otoacoustic emissions (SOAEs) were recorded. Pure-tone audiometry and OAE measurements were performed in a sound-proof chamber (Industrial Acoustics Company, Niederkrüchten, Germany). The study was approved by the Ethics Committee of the University of Tübingen in accordance with the Declaration of Helsinki for human experiments.

OAE recordings were made using an ER-10C DPOAE probe-microphone system (Etymotic Research, Elk Grove Village, IL) connected to a PC equipped with a 16-bit analog output card and a 24-bit signal acquisition card (NI PCI 6733 and NI PCI 4472, National Instruments, Austin, TX). The sampling frequency was 102.4 kHz. Data acquisition was controlled by custom-built software implemented in LabVIEW (Ver. 12.0, National Instruments, Austin, TX). The ER-10C speakers were calibrated by in-ear calibration throughout the recording in 120-s intervals. Further details on the acquisition setup and the calibration process are specified elsewhere [8]. DPOAEs were acquired using both short-pulse and continuous stimulation in a frequency range of $1 \leq f_2 \leq 4$ kHz with $\Delta f_2 = 20$ Hz ($f_2/f_1 = 1.2$). The stimulus level of the f_2 tone was held fixed at $L_2 = 45$ dB SPL, whereas the level of the f_1 primary tone was chosen according to a frequency-specific paradigm $L_1(f_2, L_2) = a(f_2)L_2 + b(f_2)$ to optimize maximum overlap between the primary-tone traveling waves at the f_2-tonotopic place [8].

Data acquisition blocks were of 60-ms length for short-pulse stimulation. The f_1 pulse was of 30-ms duration and started 5 ms after block onset with cosine-shaped rising and falling edges of 2.5-ms length (Fig. 1a). The f_2 pulses started 7.5 ms after block onset and exhibited frequency-specific full-widths at half maximum [8]. Suitable phase changes of the primary tones during the acquisition process resulted in their cancellation after ensemble averaging [9]. Continuous stimulation was done with an acquisition block length of 100 ms and at primary-tone frequencies, which exhibited an integer number of periods within the acquisition block. Recording was performed until a signal-to-noise ratio (SNR) of 10 dB was met or a maximum number of 25 or 100 blocks was reached for the continuous or the short-pulse stimulation, respectively. Acquisition blocks not enhancing the SNR were excluded from the averaging process. Signal post-processing and analysis were done using a custom-built toolbox in Matlab (Ver 9.0, Mathworks, Natick, MA). After band-pass filtering with a zero-phase FIR filter (order $N = 1200$) and frequency-dependent bandwidths, the averaged short-pulse recordings yielded a DPOAE time signal, $p_{sig}(t)$, for each investigated stimulus-frequency pair. By contrast, the data analysis of the continuous recordings resulted in a complex amplitude, $P_c = \hat{P}_c e^{i\varphi_c}$, extracted from the complex spectrum at the particular distortion-product frequency f_{dp} after averaging.