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An Interactive Software Module for DPOAE Signal Estimation

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Abstract— **This work presents a freely downloadable software module for the estimation of distortion product otoacoustic emission (DPOAE) signals based on a novel adaptive signal processing technique of measurement of signals under large amounts of noise. DPOAE signal estimation is an effective method of testing the human peripheral auditory function and is extensively used in newborn hearing screening. Current technology is based on the averaging of long strings of data and subsequent Fourier analysis, and suffers from the need for relatively long measurement time and acoustically insulated examination rooms. The method presented in this work features structural simplicity which renders it particularly attractive for implementation on both software and hardware platforms. As such, a fully functional software implementation of the proposed algorithm is developed and is made publicly available for free distribution to researchers in the area. The proposed technique offers a high degree of immunity with regard to background noise and parameter variations. Compared to conventional methods, the proposed method offers a shorter measurement time which is of significant value in clinical examinations. Performance of the proposed method is demonstrated with the aid of computer simulation and is verified in laboratory using recorded clinical data. Snapshots of the developed software environment analyzing both simulated and real clinical data are also presented.**

Index Terms— **Otoacoustic emissions, adaptive signal processing, hearing assessment, peripheral auditory function**

I. Introduction

Distortion product otoacoustic emissions (DPOAEs) are very low level stimulated acoustic responses to two pure tones presented to the ear canal. DPOAE measurement provides an objective non-invasive measure of peripheral auditory function and is used for hearing assessment especially in newborns [1]. DPOAEs have been recognized for a number years [2], [3]. However, DPOAE measurement is considered an active area of research because of the challenging nature of the signal processing task.

In this type of otoacoustic test, two pure tones with frequencies f_1 and f_2 are presented to the cochlea. For best results, f_2 is usually chosen as $1.2f_1$. Since the ear is a nonlinear structure, a number of very low level distortion products are generated due to the inter-modulation process within the cochlea. Among various distortion products, the component with frequency $f_d = 2f_1 - f_2$ is usually the strongest. The level of such a distortion product (commonly referred to as the DPOAE signal) is taken as an index of the functionality of the ear. Estimation of such a weak signal buried under two strong stimuli and other inter-modulations in a potentially noisy background is a challenging signal processing problem.

Conventionally, the discrete Fourier transform (DFT) has been used as the main signal processing tool to estimate the level of the DPOAE signals. Application of the DFT to this problem has a number of shortcomings, among which the long measurement time is the most pronounced one [4]. Long measurement time is usually required for the acquisition of a sufficiently large amount of data which, when averaged, will reduce the overall background noise effect. In addition to the need to increase the measurement time, the tests are usually required to be conducted in low noise environments such as sound-proof rooms or other types of sound-proof enclosures.

In an attempt to devise high performance DPOAE estimation techniques, adaptive signal processing techniques and maximum-likelihood estimators have been employed. Such techniques generally offer better performance in terms of measurement time which may be interpreted as their higher noise immunity compared to that of the DFT.

This paper presents a Matlab-based interactive graphical user interface (GUI) implementing a recent method of DPOAE signal measurement which employs, as its main building block, a nonlinear adaptive signal processing algorithm capable of estimating sinusoidal signals buried under large amounts of noise. The formulation, mathematical properties and DSP implementation of the employed signal processing algorithm are presented in [5], [6] where detailed discussions on the stability and convergence issues of the algorithm are also presented. Some of the applications of the employed algorithm in diverse areas of engineering are presented in [7], [8]. The software introduced in this paper is publicly available on the website of the Signal Processing Laboratory at Clarkson University (www.clarkson.edu/~spl) for free distribution to the interested audience.

II. Signal Processing Algorithm

The proposed signal processing method employs three units of the algorithm presented in [5] to construct a high performance DPOAE estimation algorithm. Each unit is capable of focusing on and extracting a pre-specified sinusoidal component of its input signal which may contain other components and noise. More importantly, it can effectively follow variations in the amplitude, phase (and frequency) of the extracted sinusoidal component. Although the underlying mathematics ensuring the stability and performance of such an algorithm is very complex, its structure remains extremely simple. It is found to be very robust with respect to variations in the internal settings of the controlling parameters, as well as external conditions such as the presence of noise, and

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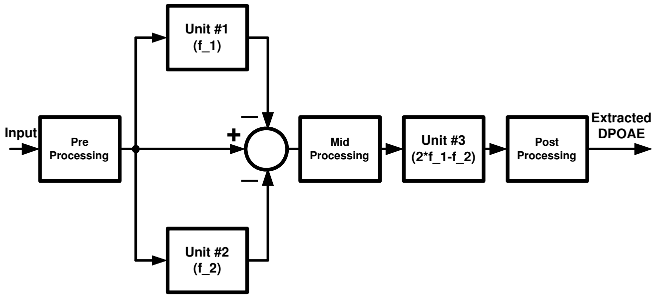

Fig. 1. Block diagram representation of the proposed method of DPOAE estimation.

exhibits superior performance over existing linear adaptive and DFT-based algorithms in terms of convergence speed versus accuracy trade-off [5].

The input DPOAE signal is often assumed to consist of two pure sinusoids with frequencies f_1 and f_2 at a very high level (usually about 50 to 70 dB SPL) and a very low level DPOAE $2f_1 - f_2$ at about -5 to 15 dB SPL. It is contaminated by a noise usually considered to be about 0 to 20 dB SPL. In fact, the noise represents the totality of all undesired signals that may be present in the environment in which the examination is being conducted, the sum of all generated inter-modulations as well as the unavoidable background noise. It has been observed that the estimation error increases with the increase of the amount of background noise. This can be compensated by re-adjusting the parameters of the algorithm to reduce the error at the expense of the convergence speed. Because of the excessive degree of the noise, one single unit assigned to extract the DPOAE signal out of the input signal exhibits poor performance in terms of the estimation error, (or the convergence speed).

In the block diagram of Fig. 1, the first two units are assigned to extract the two stimuli. They effectively do so with very small errors. The extracted stimuli are then subtracted from the input signal to produce a signal, of which DPOAE has a higher relative portion. The third unit is then set to extract the DPOAE signal. To further enhance the performance of the DPOAE estimator, some pre-processing, post-processing as well as intermediary filtering stages have been added.

The stage of pre-processing consists of preliminary normalization and band-pass filtering. The purpose of the normalization process is to amplify the input signal to bring it to a certain level on the basis of which the setting of the parameters of the units may be adjusted. The band-pass filtering is intended to attenuate all components except the DPOAE signal as much as possible to enhance the quality of the input signal. This can be achieved by means of a simple second order band-pass filter, the center frequency of which is set to be that of the DPOAE signal.

The intermediate signal out of which the two stimuli are removed may be directly input to a third unit for the extraction of the DPOAE signal. Since elimination of the two stimuli needs certain convergence time, at the very early initial moments a large portion of the two stimuli exists which will set the initial operational point of the third unit too far away from the true level of the DPOAE signal. To overcome this, a time-gating process may be employed to delay the transfer of the intermediate signal to the third unit. This is accommodated in the mid-processing unit of Fig. 1. The output of this unit is zero and remains zero for a short period of time until a more or less steady state condition for the two units is achieved. The mid-processing may also include some normalization and band-pass filtering just like the pre-processing stage.

The post-processing unit consists of denormalization of the DPOAE signal and its level to restore the original values as well as some (low-pass) filtering to further smooth out the estimate of the DPOAE signal and its level.

Adjustment of Parameters

An important part of the design of the proposed DPOAE estimation method is the adjustment of the parameters. In each application, one has to roughly define the nature of the input signal and the desired speed (or the tolerable error) to be able to appropriately adjust the parameters. For this matter, the level of the two stimuli are assumed to be about 60 dB SPL, the level of the DPOAE signal 15 dB SPL and the noise floor 10 dB SPL. A convergence time of less than one second for each DPOAE level measurement and an estimation error of less than 15% seem to be sufficient from a practical point of view. This definition of the problem is a rough guideline for the design. However, and thanks to the robust and adaptive nature of the employed algorithm, variations of orders of magnitude in these values are observed to be easily tolerated by the system.

The values of the μ-parameters (refer to [5] for details) for the two units assigned to extract the two stimuli are $\mu_1 = 200$ and $\mu_2 = 20000$. The values of the μ-parameters for the third unit assigned to extract the DPOAE signal are $\mu_1 = 200$ and $\mu_2 = 20$.

The mid-processing stage consists of a time-gating (switching at $t = 100$ ms), a gain (or normalization) of 1000 and a band-pass filter. The estimate of the amplitude by the third unit is transferred to the post-processing unit. The post processing is a denormalization factor of $\frac{1}{1000}$ and a smoothing low-pass filter. The output of this filter is the estimated DPOAE level.

III. Results

Performance of the proposed method is demonstrated in this section using both flexibly controlled simulated data and a set of real clinically recorded signal.

A. Simulated Data

For the simulation presented in this section, the frequency of the first stimulus (f_1) is randomly chosen from the range of 500 to 4000 Hz. The frequencies of the second stimulus and the DPOAE are then set as $f_2 = 1.2f_1$ and $f_d = 2f_1 - f_2$, respectively. The initial phases of the simulated stimuli and the DPOAE are randomly chosen within 0 to 2π.

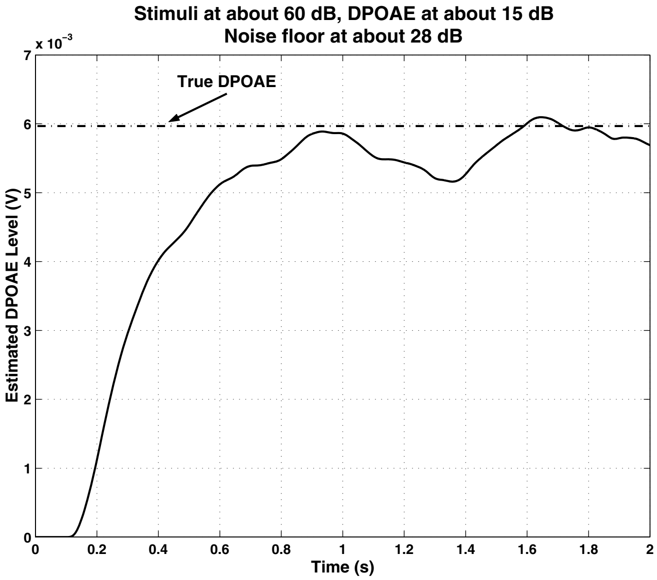

Fig. 2. Illustration of the performance of the proposed DPOAE estimator using simulated data.

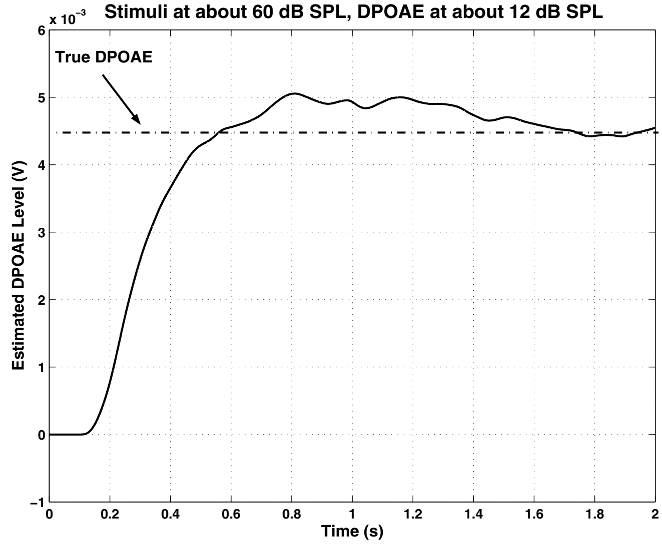

Fig. 3. Illustration of the performance of the proposed DPOAE estimator using a set of clinically recorded data.

Fig. 2 presents the performance of the proposed DPOAE estimator when the levels of the two stimuli are randomly generated within the range of 0.8 to 1 V (roughly corresponding to a relative 60 dB level) while the DPOAE signal has a level of about 6 mV (corresponding to a relative 15 dB level). The noise floor is at about 28 dB. The estimation is achieved well within one second test time with a tolerable estimation error. The present parameter setting easily accommodates noise levels of up to 30 dB, which is believed to be an exaggeration of the actual scenarios. However, if the expected noise floor happens to be even higher, one can sacrifice the speed by re-adjusting the parameter settings. Generally, one needs to take into account the following factors when choosing the values of parameter settings: some idea about the potential background noise, the desired speed of convergence and the tolerable error. Experience of the authors as well as that of the collaborators in companies manufacturing DPOAE measurement equipment confirms the suitability of the suggested parameter settings in practical DPOAE measurement tests.

B. Recorded Data

One set of clinical data recorded at Rotman Research Institute of the University of Toronto is used to verify the functionality of the proposed method. The recording is conducted using specialized otoacoustic probes. About 20 seconds of recording is available. The total length of the recording is used to obtain the frequency spectrum of the signal, which in turn can serve as a means of guessing the true value of the DPOAE level. The frequencies of the two stimuli and the DPOAE are $f_1 = 1618$ Hz, $f_2 = 1797$ Hz and $f_d = 1438$ Hz, respectively. Fig. 3 presents the performance of the proposed method. It is observed that the convergence is achieved within the desired one second test period with a small estimation error.

IV. Development of the Matlab-based GUI

As part of the ongoing activities at Signal Processing Laboratory of Clarkson University, an integrated software environment has been developed to provide the hearing researchers with a user-friendly DPOAE analysis software based on the technique described in the preceding section. The software available at www.clarkson.edu/˜spl includes a Matlab-based GUI module together with a sample DPOAE recording that can be loaded and analyzed within the GUI. The user can synthetically generate a DPOAE signal for the sake of evaluating the algorithm under various controlled conditions, or alternatively can load previously recorded signals into the GUI for analysis. The GUI accepts recorded data in a number of formats including plain ASCII. All the controlling parameters of the algorithm are conveniently preset at some default values but can be easily altered by the user.

Fig. 4 shows a snapshot of the GUI in generating a synthetic signal that can be used for algorithm evaluation purposes. Fig. 5 shows a snapshot of the DPOAE estimation GUI in estimating a clinically recorded signal. It is observed that the convergence is achieved in a fraction of a second. This convergence speed is quite advantageous over the current commercially available DPOAE testing devices that require several seconds of recording for convergence. Quantitative studies presented in [6] present comparisons of this technique with existing DFT-based or maximum likelihood estimation algorithms (such as that presented in [9]) for DPOAE estimation.

V. Conclusions

A GUI module embodying a method of measurement of DPOAE signal level employing a recently introduced nonlinear adaptive signal processing technique is presented. Performance of the proposed method is demonstrated using both simulated and real clinical data, and snapshots of the

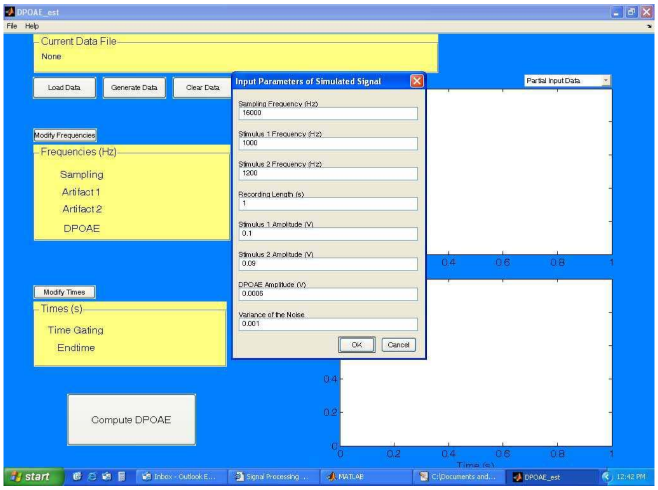

Fig. 4. A snapshot of the interactive GUI module in estimating a synthetically generated signal.

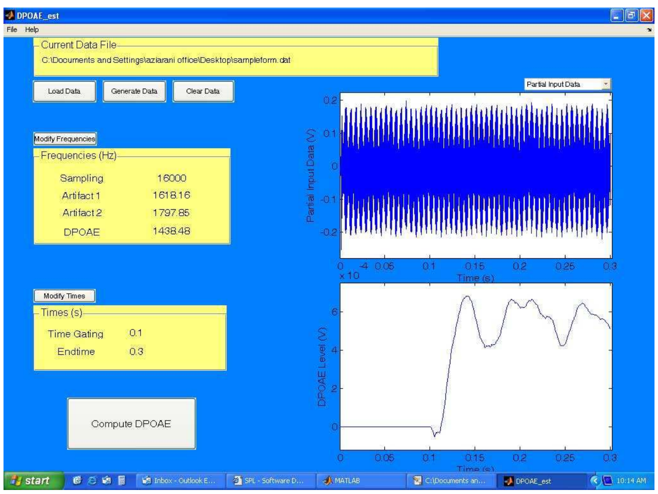

Fig. 5. A snapshot of the interactive GUI module in estimating a clinically recorded signal.

software environment are presented. The publicly available software presented in this paper provides a user-friendly environment for hearing researchers. Given the low complexity of the proposed method, it requires low level of computational resources, which in turn translates into less expensive equipment. High noise immunity and robustness of the proposed method render it suitable for practical clinical examinations which may be conducted in highly noisy backgrounds, perhaps without involving sound-proof examination rooms. This again translates into less expensive medical equipment. Also, given that the reduction in the level of the stimuli translates into a higher relative degree of background noise, the high noise immunity feature of the proposed method may be used to reduce the level of the stimuli for a more patient friendly examination. High speed of convergence of the proposed method is useful in reducing the examination time which again results in a more patient friendly and time effective clinical examination.

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