Proceedings of the 28th IEEE
EMBS Annual International Conference
New York City, USA, Aug 30-Sept 3, 2006

A Labview-Based GUI for the Measurement of Otoacoustic Emissions

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Abstract— **This paper presents the outcome of a software development project aimed at creating a stand-alone user-friendly signal processing algorithm for the estimation of distortion product otoacoustic emission (OAE) signals. OAE testing is one of the most commonly used methods of first screening of newborns' hearing. Most of the currently available commercial devices rely upon averaging long strings of data and subsequent discrete Fourier analysis to estimate low level OAE signals from within the background noise in the presence of the strong stimuli. The main shortcoming of the presently employed technology is the need for long measurement time and its low noise immunity. The result of the software development project presented here is a graphical user interface (GUI) module that implements a recently introduced adaptive technique of OAE signal estimation. This software module is easy to use and is freely disseminated on the Internet for the use of the hearing research community. This GUI module allows loading of the a priori recorded OAE signals into the workspace, and provides the user with interactive instructions for the OAE signal estimation. Moreover, the user can generate simulated OAE signals to objectively evaluate the performance capability of the implemented signal processing technique.**

Index Terms— **Medical devices, hearing research, signal processing, signal estimation, software development**

I. Introduction

Otoacoustic emissions (OAEs) are very low level acoustic responses to audio signals presented to the ear canal and provide objective methods of non-invasive testing of the peripheral auditory function; as such, they are widely used for hearing assessment especially in newborns [1]. A special type of OAEs are distortion-product otoacoustic emissions (DPOAEs) that have been recognized for a number years [2], [3] and are one of the main techniques of newborn first screening presently employed.

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.

This paper presents a Labview-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 hearing research community.

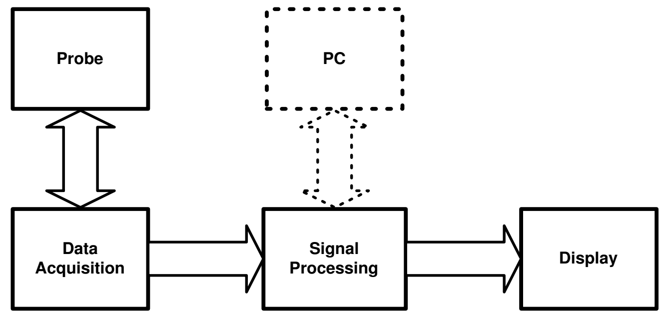

Fig. 1. Generic block diagram of a DPOAE measurement device.

II. Structure of a Generic DPOAE Measurement Device

Figure 1 shows the generic block diagram of a DPOAE measurement device. It consists of three main modules: the data acquisition/transducers module, the signal processing module and the display.

The data acquisition module is the medium between the processing module and the probe which transmits and receives acoustic signals in the audio range. One of the main functions of this module is to convert digital signals produced by the signal processing module to analog signals

This work was supported by U.S. NSF Award BES-0447705.

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which are then conditioned and converted to audio signals. Conditioning of the signals in this case may or may not include filtering. Conversely, the audio signals recorded by the probe are conditioned and converted to digital signals to be processed by the signal processing module.

The heart of the system is the signal processing module which produces the digital form of the stimuli and extracts and measures the DPOAE signal. A DSP, or a microprocessor, can be employed as the hardware platform of this unit. Signal processing is embedded as the software in such a hardware platform. Alternatively, and provided that the complexity of the signal processing algorithm remains low, the signal processing module may be implemented solely in hardware using programmable logic array (PLA) or field programmable gate array (FPGA) technology. In an ideal case, namely when the signal processing algorithm is not excessively complex, the hardware does not require a PC for its operation; however, interfacing to a PC is usually provisioned for data management.

The display module is the interface between the device and the operator. It can be a simple LED/LCD and/or a small printer.

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

A. Review of the Employed Core Algorithm

Let $u(t)$ denote a signal comprising a sinusoidal component polluted by a number of undesired components and noise. The least squares error between the input signal $u(t)$ and the sinusoidal signal $A \sin \phi(t)$ embedded in $u(t)$ is minimized by the use of the gradient descent method to yield the following set of differential equations. This system presents a nonlinear adaptive algorithm which takes $u(t)$ as its input signal and extracts the buried sinusoid within the input signal which is represented by $y(t)$:

$$\dot{A}(t) = \mu_1 e(t) \sin \phi(t), \quad (1)$$

$$\dot{\phi}(t) = \mu_2 e(t) \cos \phi(t) + \omega_o, \quad (2)$$

$$y(t) = A(t) \sin \phi(t), \quad (3)$$

$$e(t) = u(t) - y(t). \quad (4)$$

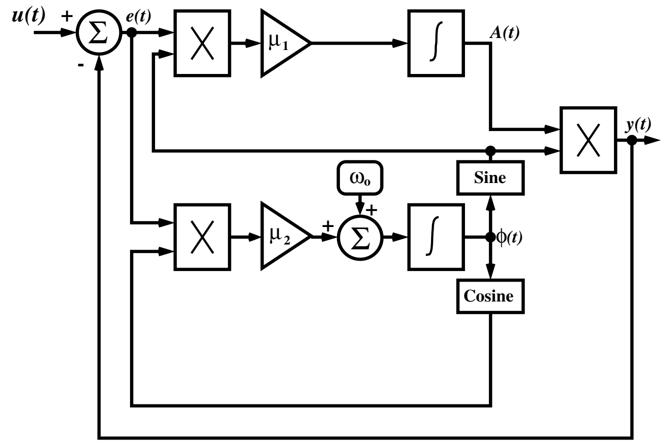

Fig. 2. Block diagram representation of the employed core algorithm.

The dot on top ($\dot{}$) represents the differentiation with respect to time. State variables $A(t)$ and $\phi(t)$ directly provide instantaneous estimates of the amplitude and phase of the extracted sinusoid, respectively. The totality of the undesired components and noise imposed on the sinusoidal component of interest is provided by $e(t)$. The parameters μ_1 and μ_2 are positive numbers which determine the behavior of the algorithm in terms of convergence speed versus accuracy.

The dynamic of the core algorithm, as described by (1) to (4), presents a notch filter in the sense that it extracts (i.e. lets pass through) one specific sinusoidal component and rejects all other components including noise. It is adaptive in the sense that the notch filter accommodates variations of the characteristics of the desired output over time. The center frequency of such an adaptive notch filter is specified by the value of ω_o.

Implementation of the differential equations governing the employed core algorithm is straightforward. Figure 2 shows the implementation of the algorithm in the form of composition of simple blocks suitable for schematic software development tools. It basically consists of a few arithmetic and trigonometric operations which may be easily implemented in any programming language.

B. The OAE Measurement Technique

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 level of noise, one single unit assigned to extract the DPOAE

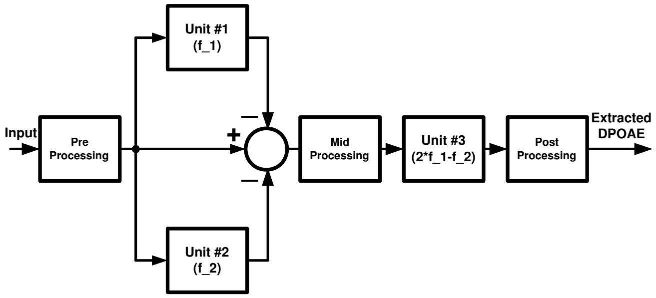

Fig. 3. Block diagram representation of the proposed method of DPOAE estimation.

signal out of the input signal exhibits poor performance in terms of the estimation error. Thus, a more elaborate signal extraction scheme is employed that is delineated below.

In the block diagram of Figure 3, 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 Figure 3. 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.

IV. Development of the Labview-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 Labview-based GUI module together with a sample DPOAE recording that can be loaded and analyzed within the GUI. The freely disseminated module contains all the necessary Labview components for a stand-alone operation of the software. In other words, the software does not need a full version of the Labview already installed to run properly and will automatically install the Labview Run-time Engine that is a free distribution of National Instruments. 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. All the controlling parameters of the algorithm are conveniently preset at some default values but can be easily altered by the user.

Figure 4 shows a snapshot of the DPOAE estimation GUI in estimating a clinically recorded signal. It is observed that the convergence is achieved in less than 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 stand alone Labview-based GUI module implementing a method of measurement of DPOAE signal level employing a recently introduced nonlinear adaptive signal processing technique is presented. The publicly available software presented in this paper provides a user-friendly computational environment for hearing researchers, hence obviating the need for programming on their part. It contains all the necessary components for its operation. Given the low complexity of the proposed method, it requires low levels 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 soundproof 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.

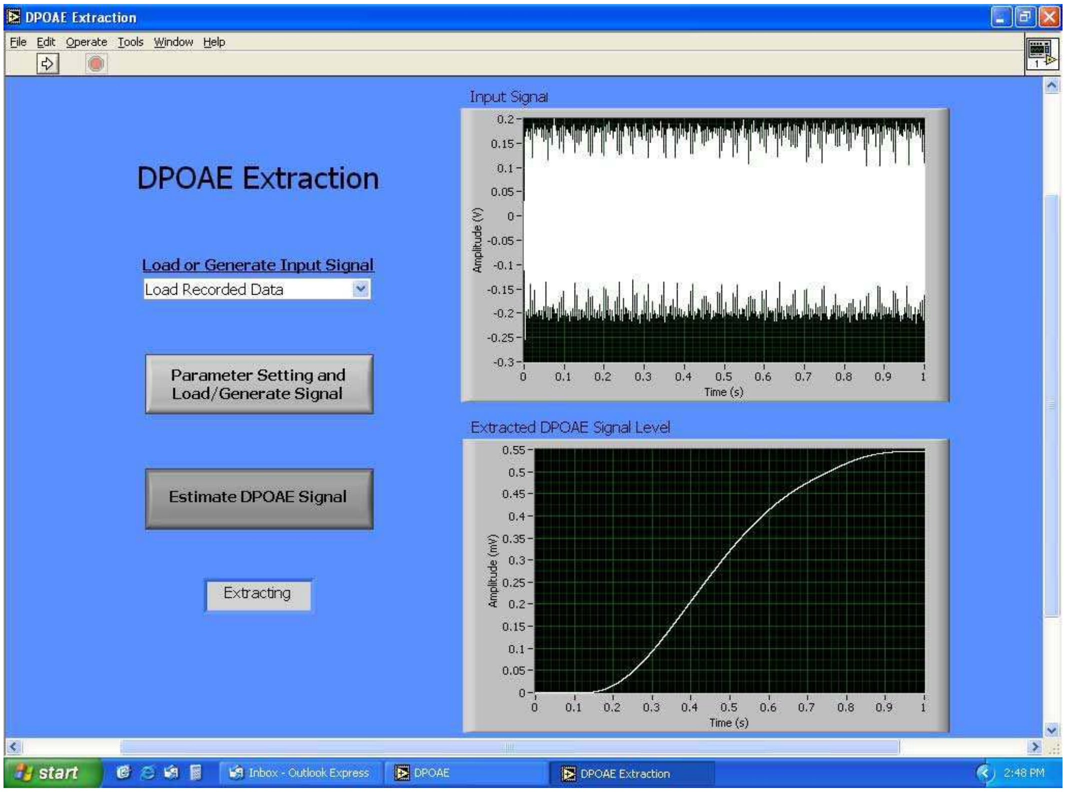

Fig. 4. A snapshot of the interactive GUI module in estimating a clinically recorded signal.

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