REAL-TIME HEARING ASSESSMENT DEVICE BASED ON DISTORTION PRODUCT OTOACOUSTIC EMISSIONS

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ABSTRACT

Real-time implementation of a recently introduced method of measurement of distortion product otoacoustic emissions (DPOAEs) for hearing assessment is presented. Development of the real-time application begins with the discretization of the DPOAE estimation algorithm in Matlab and then programming in C for final downloading onto a digital signal processor (DSP) platform. Aside from the signal processing task involved in the estimation process, the DSP platform controls a microphone/speaker system, which generates the acoustical stimuli for presentation to the ear canal and collects the acoustical responses recorded in the ear canal. The real-time results are compared with those obtained by a continuous-time off-line model of the DPOAE estimation algorithm. The developed DPOAE measurement system offers a high degree of noise immunity and a short measurement time, both of which are of importance in clinical applications of the DPOAE signals.

1. INTRODUCTION

Distortion product otoacoustic emission (DPOAE) signals are very low-level acoustical responses of the inner ear to the stimuli consisting of two pure tones that are presented to the ear canal. DPOAEs are thought to originate from an active, nonlinear mechanism in the cochlea [1]. Because an unhealthy or damaged cochlea does not produce significant levels of the DPOAEs [2], estimation of the DPOAE signal can provide an objective, noninvasive method of determining peripheral auditory function in humans [3]. Early diagnosis of hearing impairment can greatly reduce long-term deleterious effects on language and cognitive development; DPOAEs have thus gained attention in early stage infant screenings, especially because no behavioral data is required in the process of DPOAE estimation [2, 4].

In order to stimulate the cochlea to produce the DPOAE signals, two pure tones with frequencies f_1 and f_2 (in the audio range) are presented to the ear canal. Maximum levels of the DPOAE signal are observed when the frequency ratio f_2/f_1 is chosen to be 1.2 [5]. Due to the cochlear nonlinearity, low-level distortion products (DPs) will be present at frequencies of $nf_1 + mf_2$ where n and m are nonzero integers [6]. The largest of these DPs is observed to be the cubic DP, located at $f_d = 2f_1 - f_2$ [2, 5, 7]. The level of the cubic DP, which is referred to as the DPOAE, is used to gauge the functionality of the ear. The signal processing problem involves estimation of the weak DPOAE under two strong stimuli and the background noise, which in some cases can be higher that of the DPOAE itself.

Traditionally, the discrete Fourier transform (DFT) combined with averaging has been used to estimate the level of the DPOAE signal. One of the primary drawbacks of using the DFT for this purpose is the long measurement time required to reduce the noise floor to a level where the DPOAE can be effectively estimated. Adding to the inconvenience of a lengthy measurement process, a low-noise environment, such as a sound-proof room, is also often required to further reduce the noise floor.

A recent work ([8]) describes a nonlinear adaptive technique of estimation of DPOAE signals that presents a fast estimation process while offering a high degree of noise immunity. The structure of the proposed signal processing technique of [8] is very simple, suggesting that its real-time operation is feasible even with limited computational resources. This is explored in the present work. This paper reports the successful implementation of the method of DPOAE signal measurement of [8] in real-time. Structure of the proposed technique is briefly reviewed, the steps taken in its DSP implementation are described and the results are presented in a comparison with those obtained by the continuous-time off-line model of the DPOAE estimation algorithm presented in [8].

2. METHODOLOGY

The proposed DPOAE estimation is based on a nonlinear adaptive algorithm presented in [9], which is capable of extracting a specified sinusoidal component of a signal potentially containing other components and noise, even if there are variations in amplitude, frequency and phase [8, 10]. The robust performance of the algorithm with respect to accuracy and convergence speed shows its superiority to the DFT and other linear methods [9].

The following equations describe the dynamics of the employed sinusoid-tracking algorithm [9, 10]:

$$
\begin{aligned}
y[n] &= A[n]\sin\phi[n], &(1)\\
e[n] &= u[n] - y[n], &(2)\\
A[n+1] &= A[n] + 2T_s\mu_1 e\sin\phi[n], &(3)\\
\phi[n+1] &= \phi[n] + T_s\omega_o + 2T_s\mu_2 eA[n]\cos\phi[n], &(4)
\end{aligned}
$$

where u is the input signal, y is the output signal, e is the

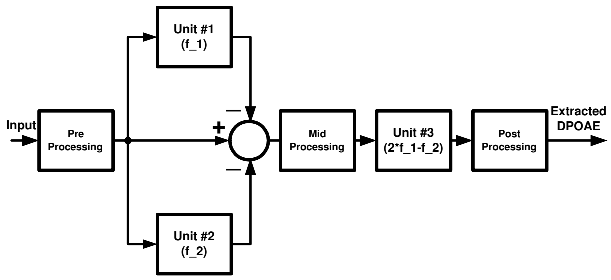

Fig. 1. Block diagram representation of the proposed method of DPOAE estimation.

error signal, A is the estimated amplitude, ϕ is the estimated phase, ω_o is the frequency of the desired sinusoid in rad/s and T_s is the sampling period; μ_1 and μ_2 are the controlling parameters of the algorithm that adjust the convergence speed versus accuracy trade-off.

In order to estimate the DPOAE signal, three units of such a sinusoid-tracking algorithm are used in combination with some pre-processing, mid-processing and post-processing, as shown in Fig. 1. This approach is due to the fact that the input signal essentially consists of two strong (approximately 60 to 70 dB sound pressure level (SPL)) pure tone stimuli with frequencies f_1 and f_2 and a very low-level DPOAE (approximately -5 to 15 dB SPL) located at $f_d = 2f_2 - f_2$. The noise floor is comparable in level to the DPOAE signal (at about 0 to 20 dB SPL) [8].

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. 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.

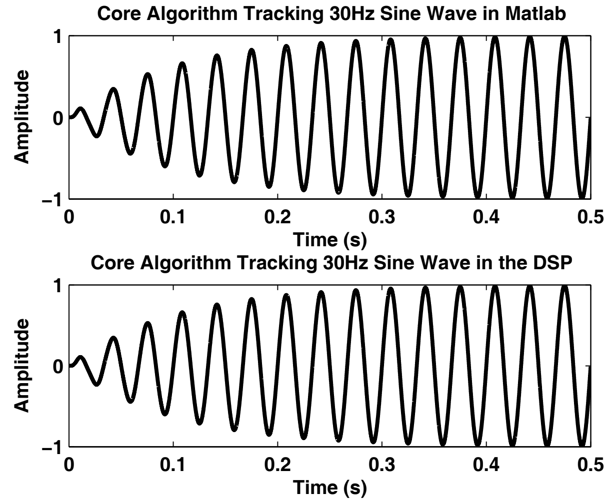

Fig. 2. Performance verification of the sinusoid-tracking algorithm operating in Matlab and in real-time on the DSP platform.

The DPOAE estimation technique of Fig. 1 is first implemented in the Matlab environment. This off-line implementation provides a means of evaluating the real-time implementation described in the following section.

3. DSP IMPLEMENTATION

The Texas Instruments' (TI) C6711 Developers Starter Kit (DSK) is used as the main computational platform in conjunction with the TI's PCM3003 audio daughter card that provides a D/A and A/D interface operable in the audio range. The core of the DSK is the TMS320C6711 floating point DSP. With some necessary syntactical changes, the Matlab code implementing the algorithm can be translated into a C code within Code Composer Studio (CCS), which is the integrated development environment (IDE) accompanying the DSK.

3.1. Considerations on Sinusoid-tracking Algorithm

The creation of the sine and cosine functions required for the sinusoid-tracking algorithm defined by (1)-(4) is achieved using look-up tables prior to the real-time operation. This is due to the observation that the large number of clock cycles required for the external function calls to sine and cosine functions provided in the mathematics library of C (i.e. math.c) prevent the algorithm from running in real-time. In implementing these trigonometric functions, each array is set to hold 360 elements: one entry for each degree of an entire cycle of a sine and cosine wave. In the beginning of the operation, the look-up tables are created and their values are stored in the memory on the DSK; while performing in real-time, references to each array can be made efficiently without hindering the real-time operation.

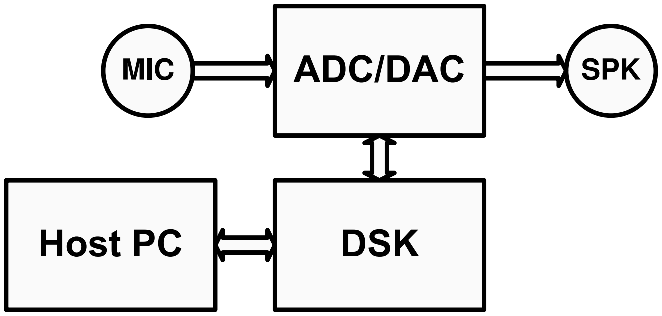

Fig. 3. Block diagram of the hardware components of the real-time system and the data flow.

Fig. 2 compares the performance of the real-time DSP platform with that of the off-line Matlab implementation. Perfect conformity between the two is observed which confirms proper implementation of the algorithm on the DSP.

3.2. Considerations on DPOAE Estimation Algorithm

With reference to Fig. 1, the DPOAE estimation algorithm consists of six functional blocks, three of which are the sinusoid-tracking units operating at frequencies $\omega_o = 2\pi f_1$, $\omega_o = 2\pi f_2$ and $\omega_o = 2\pi f_d$ to extract the two stimuli and the DPOAE component, respectively. The additional blocks of pre-processing, mid-processing and post-processing essentially consist of low-pass and band-pass filters in conjunction with some basic arithmetic operations. The continuous transfer function for each band-pass/low-pass filter is used as the starting point for the design. Coefficients for the discretized systems (i.e. difference equations representing the filters) are derived interactively using the Filter Design and Analysis (FDA) tool of Matlab Signal Processing Toolbox.

Conversion of the Matlab code implementation of the DPOAE estimation technique to a C code for downloading onto the DSP platform is done much like the sinusoid-tracking algorithm conversion discussed earlier. Filter coefficients for the pre-processing, mid-processing and post-processing stages are stored in their corresponding arrays and the filters are implemented using difference equations. Each unit of the sinusoid-tracking algorithm is placed in its sequential position in the code, representing the real-time flow of data throughout each module.

3.3. Stimuli Generation and Data Acquisition

Aside from the signal processing task of estimating the DPOAE signal, the DSP platform is responsible for i) the generation of the acoustical stimuli that are presented to ear, and ii) the collection of the responses from the ear canal. Fig. 3 shows the interaction within the components of the system. The employed speaker is the ER-1 Tubephone and the employed microphone is ER-7C Probe Microphone System, both of which are products of Etymōtic Research Corporation. The stimuli are created based on the developed look-up table for the sine function. The sum of two sine waves located at f_1 and f_2 is generated numerically, is multiplied by 5000 to generate a signal of about 65-70 dB SPL and is stored as a 16-bit integer. This variable is then sent through a multi-channel buffered serial port (McBSP) to the digital to analog converter (DAC) and subsequently to the output port of the PCM3003, which is connected to the speaker. The collected signal is received by the microphone and is sent to the input port of the analog to digital converter (ADC) of the PCM3003. The McBSP returns the samples to the DSP, where they are stored as the DPOAE estimation system input signal that is subsequently processed by the DPOAE estimation algorithm.

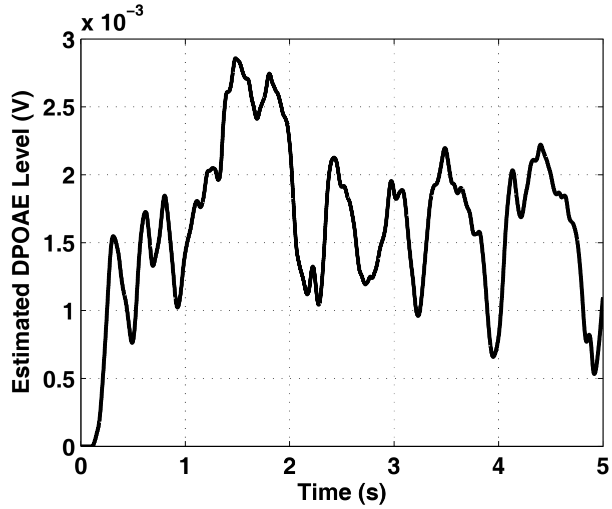

Fig. 4. Performance of the real-time system with no filtering provision.

4. EXPERIMENTAL VERIFICATION AND FINAL ADJUSTMENTS

To evaluate the results provided by the real-time system, the collected data are processed both in real-time using the DSP platform and off-line within the Matlab environment. Unlike traditional DPOAE testing, which is usually performed in sound-proof environments, the tests are performed in a noisy laboratory setting.

Fig. 4 shows the performance of the real-time system. It is observed that the presence of the high level background noise hinders the convergence to the DPOAE level in a timely fashion as expected by the simulation experiments in Matlab.

To decrease the noise floor, a fourth order active high-pass analog filter is employed between the microphone and the PCM3003 input port. The nature of the background acoustical noise is low-frequency pink noise, with the largest content being more or less below 200 Hz. The 3dB point of the filter is set to be about 600 Hz. To use the dynamic range of the speaker/microphone system, the filter is set to have an overall gain of 4. This assures that components below 300Hz undergo an attenuation greater than 24 dB, and components above 1000 Hz are amplified by at least 10 dB. Fig. 5 shows the performance of the system with the analog filter in the loop. It is observed that the DPOAE estimation effectively reaches convergence in less than one second.

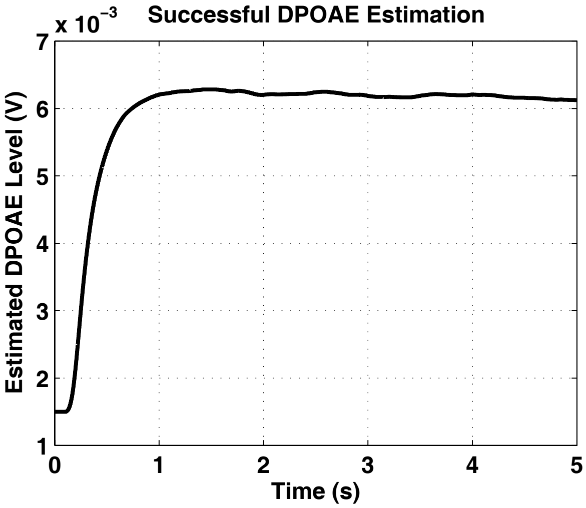

Fig. 5. Illustration of the satisfactory performance of the real-time system when an external analog filter is employed.

This is in conformity with off-line numerical experiments within the Matlab environment.

5. CONCLUSIONS

Real-time implementation of an adaptive method of estimation of DPOAE signals on a DSP platform is presented. DPOAE estimation is of value in assessing the functionality of the human auditory system and finds application in newborn hearing screening. The main features of the developed DPOAE estimation technique are its structural simplicity (that allows its real-time operation), its relatively high immunity to background acoustical noise (that allows hearing testing in normal hospital environments), and its high speed of convergence (that allows convenient testing of infants who may remain cooperative only for short periods of time). Real-time implementation of the technique using low-cost DSP hardware is of practical importance for its clinical applications. It is shown that the technique can be used in highly noisy environments with the provision of some simple analog filtering prior to signal processing.

6. REFERENCES

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