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

A Software Module for the Adaptive Estimation of Steady State Auditory Evoked Potentials

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***Abstract*— A graphical user interface (GUI) implementing a novel technique of fast estimation of steady state auditory evoked potentials (SSAEPs) for rapid assessment of the functionality of the human auditory nervous system is presented. The proposed signal estimator has shown great promise in the fast extraction of weak signals buried under large amounts of noise such as the case with SSAEP signals. Currently, the main technical impediment in the widespread clinical use of the SSAEP testing for hearing assessment is the excessively long measurement time needed for the estimation process due to the presence of large amounts of background noise. The presented software module that is publicly disseminated through the Internet is meant to facilitate the use of an efficient signal processing technique by the hearing researchers. The software environment allows for loading previously recorded SSAEP signals into the workspace for analysis. Moreover, it enables the user to add simulated SSAEP signals to the background EEG for the purpose of testing the capability of the underlying signal processing algorithm.**

***Index Terms*— Evoked potentials, auditory responses, adaptive signal processing, software development**

I. INTRODUCTION

The ability to assess the health of a person's hearing without subjective measurements is important for applications such as newborn hearing screening. Each day in the United States, approximately 33 children are born with a significant hearing impairment [1]. Approximately, one in 750 children is born with a hearing impairment significant enough to impede speech and language development [2]. The early identification of children with hearing impairment is an important public health objective in this country. Currently, many of these children are not identified until they are two years old or older. The consequences of a late diagnosis of a hearing impairment may be significant delays in spoken language and literacy. To date, more than 35 states have passed universal newborn hearing screening legislation and five other states have legislation pending [3].

One of the techniques to obtain objective information about the health of the human auditory nervous system is the measurement of the steady state auditory evoked potentials (SSAEPs), which are electrically measured responses (recorded from the human scalp) to acoustically generated stimuli presented to the ear canal [4], [5], [6], [7]. Steady state responses occur when the frequency constituents of a response remain stable in amplitude and phase over time [8]. The acoustic stimulus is an amplitude-modulated sound having a carrier frequency in the audio range and a substantially lower rate modulation frequency. Responses to slow modulation rates (less than 20 Hz) appear to originate largely in cortical structures while responses to faster modulation rates (greater than 70 Hz) appear to reflect brainstem processes. SSAEP signals of frequencies greater than 70 Hz show a great potential for hearing assessment in infants [9]. SSAEP signals are usually unrecognizable in the background EEG noise. Averaging is commonly used to increase the signal-to-noise ratio (SNR) so that evoked potentials can be detected and measured by subsequent discrete Fourier transform (DFT) analysis [10]. This technique, although effective with the availability of long strings of measured data, entails exceedingly long measurement time, which is the main technical obstacle in its widespread adoption in clinical tests.

This paper presents a Matlab-based graphical user interface (GUI) module that implements a novel technique of fast estimation of SSAEP signals. The proposed SSAEP signal estimation method is based on a new nonlinear adaptive signal processing method that has shown great promise in the fast extraction of weak signals buried under large amounts of noise [11], [12]. The user can load previously recorded data into the workspace to perform interactive analysis. The software environment enables the user to add synthetically generated SSAEP components to the background EEG signal for objective evaluation of the efficiency of the algorithm. 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.

II. THE UNDERLYING SIGNAL PROCESSING

From a signal processing point of view, the SSAEP signal is essentially a sinusoid buried under strong background noise [4]; thus, the SSAEP signal measurement involves estimation of nonstationary sinusoids in noise.

The proposed SSAEP estimation technique is based on a nonlinear adaptive method of estimation of nonstationary sinusoids in noise [11], [12]. A sinusoid is a nonlinear function of its phase (and frequency) and its full estimation is inherently a nonlinear problem. Adaptive algorithms are most appropriate means of signal estimation in nonstationary (i.e. time-varying) scenarios such as the case with SSAEP signal estimation. Conventional adaptive methods of sinusoid estimation in noise are based on linear models. The fundamental algorithm presented in [11], [12] is an adaptive algorithm based on a full nonlinear model of sinusoids. Thus, the fundamental algorithm is one of the

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

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most appropriate means of sinusoidal signal estimation in noise. Its successful applications in diverse engineering fields prove the theoretical predictions. Examples can be found in [13], [14]. The mathematical structure and properties of the sinusoid tracking algorithm employed in the proposed SSAEP estimation technique are briefly reviewed here.

A. Review of the Employed Sinusoid Tracking Technique

Consider a sinusoidal signal polluted by some noise of unknown frequency composition and expressed by

$$u(t) = A\sin(\omega t + \delta) + n(t) \quad (1)$$

where $n(t)$ represents the totality of the imposed noise, A and ω are potentially time-varying amplitude and frequency of the sine wave, respectively, and δ is the constant phase of the sinusoid. The total phase of the sine wave is $\phi = \omega t + \delta$. If time-variations are sufficiently slow, parameters A, ω and ϕ are constant values A_o, ω_o and ϕ_o within any short time interval.

Least squares error between the input signal $u(t)$ and the sinusoidal signal $A\sin(\omega t + \delta)$ embedded in $u(t)$ may be minimized by employing a gradient descent method. The result (with some modifications presented in [12]) is the following set of nonlinear differential equations to govern the dynamics of a signal processing algorithm aimed at extracting the potentially time-varying sinusoidal signal buried in $u(t)$ without any assumption on the composition of the imposed noise:

$$\dot{\hat{A}} = \mu_1 e \sin\hat{\phi}, \quad (2)$$
$$\dot{\hat{\omega}} = \mu_2 e \hat{A}\cos\hat{\phi}, \quad (3)$$
$$\dot{\hat{\phi}} = \hat{\omega} + \mu_3\dot{\hat{\omega}} \quad (4)$$

where error $e(t)$ represents the difference between the input signal polluted by noise and the extracted sinusoid ($y(t) = \hat{A}\sin\hat{\phi}$), i.e.

$$e(t) = u(t) - \hat{A}\sin\hat{\phi}. \quad (5)$$

In the above equations, $\hat{A}$, $\hat{\phi}$ and $\hat{\omega}$ are estimated values of amplitude, total phase and frequency of the extracted sinusoidal signal $y(t)$, respectively. Parameters μ_1, μ_2 and μ_3 are positive numbers which determine the speed of the algorithm in the estimation process as well as in tracking variations in the characteristics of the input signal over time.

The following theorem, proved in [11], deals with the existence, uniqueness and stability of a periodic orbit for the dynamical system described by (2)-(5):

Theorem 1: Assume that $u(t)$ is given by (1) wherein all the parameters are unknown but bounded. The dynamical system represented by (2)-(5) has a locally unique and hyperbolically stable periodic orbit $\gamma(t)$ in a close neighborhood of $\gamma_o = (A_o, \omega_o, \phi_o)$.

This theorem guarantees (i) the convergence of the solution of the dynamical system to the periodic orbit associated with the sinusoidal signal in $u(t)$ and (ii) the tracking of its variations over time. In terms of the signal processing performance of the algorithm, it extracts a sinusoidal component of its input signal, directly estimates its amplitude, phase and frequency, and adaptively tracks their variations over time.

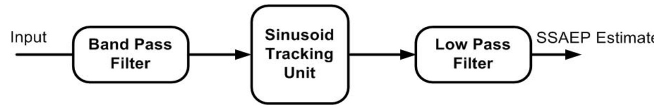

Fig. 1. Block diagram representation of the proposed method of SSAEP estimation.

B. The Proposed SSAEP Estimation Technique

The proposed SSAEP estimation method employs the fundamental algorithm of the preceding section as its main building block. Its block diagram representation is shown in Figure 1. A simple second order band pass filter, passing the SSAEP signal and suppressing part of the background noise, generates a signal with a higher relative portion of the SSAEP signal. Such a signal is then input to a unit of the fundamental algorithm, which estimates the level of the SSAEP signal. The post filtering process (a simple first order low pass filter) is intended for smoothing out the estimation. The structure of the proposed method is very simple, yet it exhibits superior performance over existing DFT-based in terms of speed and reliability. It offers a high degree of robustness with regard to both internal and external conditions. More particularly, it has been observed that the presence of high levels of background noise has little negative effect on the performance of the method and the technique is very insensitive to parameter variations within its internal setting. These features are quite attractive for its clinical application where high degrees of noise and disturbances may be present during the tests.

III. Results

Performance of the proposed method is illustrated in this section using a set of real clinically recorded signals. Figure 2 shows an example of the frequency spectra of typical SSAEP signals in background EEG. This graph is obtained by averaging/DFT analysis of about 5 minutes of recording on an adult subject. Multiple auditory evoked potentials are used. Three tones at frequencies of 84.96 Hz, 85.94 Hz and 86.91 Hz are used in modulating the carrier frequency to form the acoustic stimulus. The electrical signal recorded on the scalp is digitized at a sampling frequency of 16 kHz and is used in an off-line processing. The detected SSAEP signals are visible on the graph. The test is conducted under controlled condition with minimal environmental noise. The levels of the SSAEP signals are about 0.3 mV in this case.

Since there is no time information available on a frequency spectrum, exact time-dependencies of the levels of the SSAEP signals are unknown. Neurological interpretation of the SSAEP signals supports a steady state nature for the signals. This is, however, a hypothesis from a signal processing point of view: a sinusoidal signal of time-varying amplitude has the same amplitude in the frequency domain (at the associated frequency bin) as that of a sinusoidal signal of fixed amplitude having the mean value of the amplitude

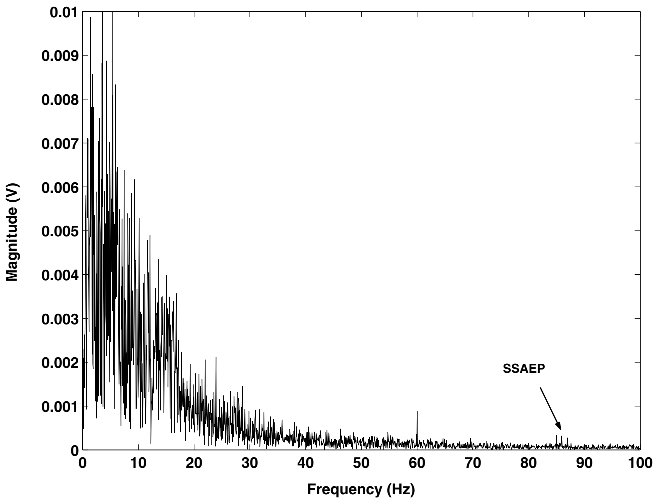

Fig. 2. The frequency spectrum of a multiple SSAEP recording.

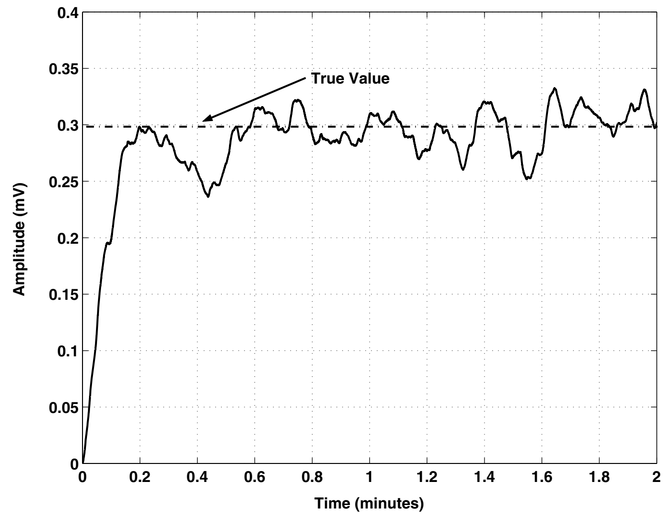

Fig. 3. Illustration of the performance of the SSAEP estimation technique in estimating a controlled signal level.

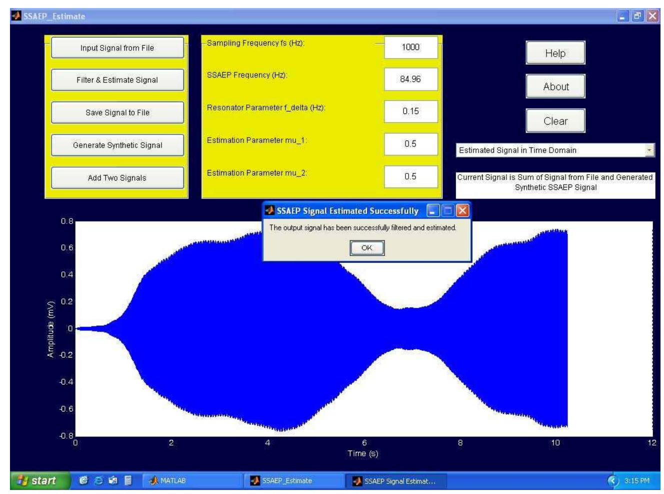

Fig. 4. A snapshot of the interactive GUI module - the estimated signal in the time domain.

of the fluctuating sinusoid. To perform an objective signal processing test, a synthesized sinusoidal signal of 0.3 mV fixed level oscillating at 80 Hz is added to the data.

The proposed SSAEP estimation algorithm is implemented in the Matlab Simulink environment. Figure 3 shows the performance of the proposed technique in estimating the level of such a controlled SSAEP signal. To clearly show the transient performance of the algorithm, only the first two minutes of output data are shown in this graph. It is observed that the convergence is achieved within a small fraction of a minute, which as such is a substantial improvement over conventional DFT-based techniques in terms of estimation time. The estimation error is a function of the parameter settings of the algorithm and is fully controllable in a trade-off with speed: the more accurate an estimation is desired, the longer it takes to achieve convergence. In the example presented here, the error is less than 10%, which is sufficient for all practical purposes. This example is an objective evaluation of the proposed algorithm since the true shape of the inserted sinusoid is known. To complement this experiment, further experiments have been conducted on this set of data to estimate the actual SSAEP signals at frequencies of 84.96 Hz, 85.94 Hz and 86.91 Hz. In all three cases, the proposed algorithm yields (in less than one minute and within an estimation error of 10%) the same estimates as those by averaging/DFT analysis of five minutes length of data. The human subject tested in this example experiment responds positively to the SSAEP tests. It has been observed that obtaining reliable estimates of SSAEP levels on an average subject takes longer. Whereas the conventional DFT-based techniques require at least several minutes of recording, the proposed technique usually achieves convergence in less than 1 minute. This testing time renders the SSAEP testing quite feasible for clinical use.

IV. Development of the Matlab-based GUI

An integrated software environment has been developed to provide the hearing researchers with a user-friendly SSAEP 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 SSAEP recording in the background EEG that can be loaded and analyzed within the GUI. The user can synthetically add SSAEP components for the sake of evaluating the algorithm under various controlled conditions. All the controlling parameters of the algorithm are conveniently preset at some default values but can be easily altered by the user. Figures 4 and 5 show a snapshot of the GUI.

V. Conclusions

An integrated software environment embodying a method of measurement of SSAEP signal level is presented and performance of the proposed method and the associated GUI module is demonstrated in a case study. The main features of the proposed method of SSAEP signal measurement are its 1) high speed of measurement, 2) structural

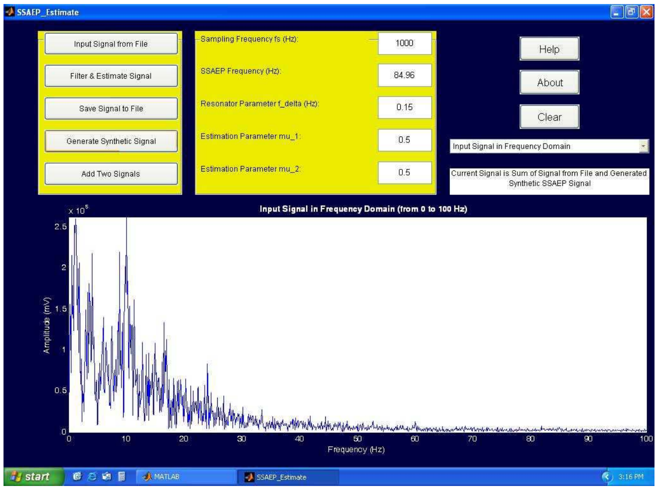

Fig. 5. A snapshot of the interactive GUI module - the input signal in the frequency domain.

simplicity, 3) high noise immunity and robustness. High speed of convergence of the proposed method is useful in reducing the examination time which results in a more patient friendly and time effective clinical examination. 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.

VI. Acknowledgment

The authors would like to thank Dr. David Purcell, Dr. Sasha John and Professor Terence Picton of the University of Toronto for providing the clinical recording used in this paper.

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