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Decomposition of EMG Signal Using MAMDF Filtering and Digital Signal Processor

Jin Lee, Jong-Weon Kim, Sung-Hwan Kim

=Abstract=

In this paper, a new decomposition method of the interference EMG signal using MAMDF filtering and digital signal processor. The efficient software and hardware signal processing techniques are employed.

The MAMDF filter is employed in order to estimate the presence and likely location of the respective templates which may include in the observed mixture, and high-resolution waveform alignment is employed in order to provide the optimal combination set and time delays of the selected templates. The TMS320C25 digital signal processor chip is employed in order to execute the intensive calculation part of the software. The method is verified through a simulation with real templates which are obtained from needle EMG.

As a result, the proposed method provides an overall speed improvement of 32-40 times.

Key words : Interference EMG signal, MAMDF Filter, Decomposition of EMG signal, High resolution waveform alignment.

Introduction

Superimpositions arise in neuroelectric signals when cells fire independently of one another. The extracellular potentials generated by different nerves and muscle cells sum linearly, so that when motor units discharge simultaneously the net potential equals the superimposition of the individual potentials. Superimpositions of motor-unit action potentials (MUAP's) occur frequently in EMG (electromyogram) signals, particularly during forceful contractions. In clinical applications, such as neuromuscular disorder diagnosis it is necessary to resolve each superimposition into the individual wavelet that make it up.

Several methods for resolving superimpositions have been developed previously in the literature[1~5]. In evaluating their capabilities, it is meaningful to distinguish two different levels of superimposition: partial superimpositions in which wavelets overlap peripherally without their peaks being obscured, and complete superimpositions in which the peaks of the wavelets merge to form a single peak[3].

One approach to resolving superimpositions is the sequential recognition[1, 2], and the other approach[4, 5] is to find the set of templates (i. e., MUAP's) and alignments that gives the best fit to the superimposition waveform. In principle, this requires trying all possible template combinations and for each combinations determining the alignments that give the best reconstruction.

While this approach is more time consuming than sequential recognition, it is optimal and capable of resolving complete as well as partial superimpositions. Therefore, in order to resolve a superimposition when it is not known which templates or even how many templates are involved,

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통신저자: Jeonmong-Dong, 8-3 Dongdaemun-Gu, Seoul, Seoul city Univ. Dept. of Electronics, Tel. (02) 210-2318, Fax. (02) 249-6802

it is necessary to try all possible combinations and pick the best one. Unfortunately, the search takes too long time when there are many templates because of the large num ber of combinations to be checked.

De Figueiredo and Gerber[4] compute the optimal set of alignments much more efficiently by treating the wavefor- ms as continuouse functions and employing an iterative continuous-time optimization procedure, and McGill and Dorfman[5] have developed an accurate algorithm of high-resolution alignment. But time consuming problem still remains to be solved for practical use.

Together with the development of digital signal processing algorithms, in recent year, rapidly growing VLSI (very large scale integrate) technology has revolutionized modern digital signal processing. One of the most important breakthroughs in electronic technology is the high-speed digital signal processor (DSP) chip.

This paper describes a new decomposition method which reduces the sets of admissible combinations based on the MAMDF (modified average magnitude difference function), and provides the optimal combination set and time delays of the selected template through the high resolution waveform alignment by using DSP chip. The TMS 320C25 DSP chip was chosen for the implementation because of its advanced architecture and its high speed of instruction execution, and cost[7, 8].

The proposed method achieves its high level of performance through the combination both of the software and hardware approaches. The main advantages of this technique are: 1) Computational efficiency is increased when there are many templates; and 2) Unlimited resolution.

The efficiency is verified through simulation with real templates which are obtained from EMG signals recorded using a concentric needle electrode.

Decomposition Scheme

A) Software

The initial estimation needed to resolve the superimposition of signals are achieved in following way. Generally, the AMDF, which is a variation on autocorrelation analysis, is defined by the relation[6].

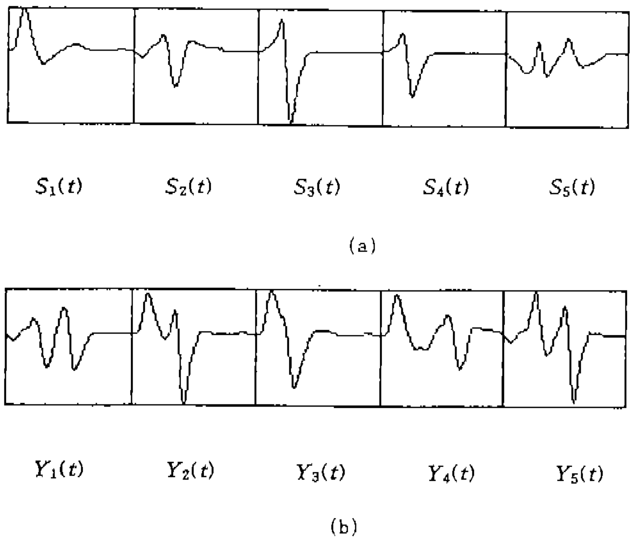

Fig. 1. **a)** The template signals obtained from needle EMG experiments
b) The superimposed waveforms

$$D\tau = \frac{1}{N}\sum_{n=1}^{N} |s(n)-s(n-\tau)|,\ \tau = 0,1,\cdots,\tau_{max} \quad (1)$$

where $s(n)$ = nth sample of the input waveform
$s(n-\tau)$ = delayed sample of the input waveform
N = size of the input waveform
τ = delay value
τ_{max} = maximum delay shift ($\tau_{max} = L$).

In eq. (1), the vertical bars denote taking the magnitude of the difference $s(n)-s(n-\tau)$. Thus a difference signal $D\tau$ is formed by delaying the input signal various amounts, subtracting the delayed waveform from the original, and summing the magnitude of the difference between sample values. Note that the difference signal is always observed to exhibit deep null at delay corresponding to the original.

Specifically, we assume that the superimposed signal $y(t)$ shown in Fig. 1 (b) can be represented by

$$y(t) = \sum_{i=}^{m} s_i(t-\tau_i),\ t \subseteq 1 \quad (2)$$

where s_i, $i = 1,\cdots, m$, are template signals which are involved for the superimposed signal; τ_i is unknown time delay

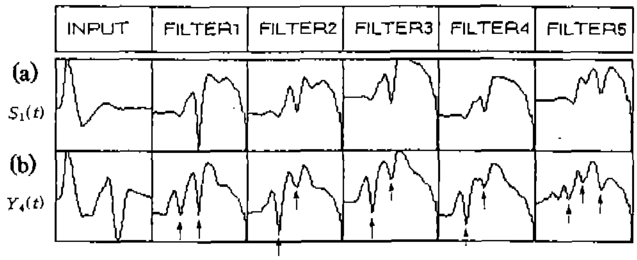

Fig. 2. The output of s1 and y3 through the MAMDF (The arrows indicate the significant dips)

associated with the templates. The template signals are shown in Fig. 1 (a). Also, we assume further that $y(t)$ is observed over a finite window I and any one signal s_i does not occur more than once in the window I.

From this, the output of MAMDF filters, F_j, $j=1,\cdots,m$, which is a extension of AMDF, can be written as[4)]

$$z_j(\tau)=\int |s_j(t-\tau)-y(t)|\ dt. \tag{3}$$

Thus, the output of a given filter shown in Fig. 2 is measured for different shifts τ of s_j as s_j is made to slide across the entire interval I. The point where the minimum of the output of the MAMDF filter occurs corresponds to the most likely location (delay) of s_j and is used to determine the presence of s_j. In this regards, the MAMDF filter is used to obtain initial conditions of relative time shift on templates which may be present in the superimposed signal.

The lowest downward peaks (dips) appear at $\tau=0$ when correct templates, the same as original signal, are detected. Similary, Fig. 2 (b) represents the outputs corresponding to the application of the superimposed signal $y_4(t)$ as input to the filters for example. Waveforms in each row are plotted with the same scaling factor to allow detection by threshold. For a particular set of templates, the possible delays obtained as above are used as starting conditions for the alignment criteria. From this, total search space for all possible combination is reduced to considerable number.

Next, on the basis of this condition, the high resolution alignment is carried out for all possible combinations and one corresponding to the least minimum error is accepted to be the valid one.

For the accurate comparison of the waveforms, they must be precisely aligned. Imprecise alignment makes them appear more different than they really are, and this can lead to comparison error.

The Nyquist rate of waveform equals two times the frequency of its highest frequency component and is theoretically the lowest sampling rate adequate to complete capture the information it contains. However, since the sampling clock was not synchronized with the motor unit the samples are out of phase, causing the two occurences of the same waveform sampled at its Nyquist rate to appear quite different when they are aligned only to the nearest discrete sampling interval. The higher temporal resolution needed to align the waveforms precisely can be achieved by oversampling (by a factor of 5-7) or by reconstructing the continuous waveforms by interpolation.

In this paper, a computationally efficient interpolation method based on the discrete Fourier transform (DFT) was used[5)]. This algorithm is more accurate and requires slower sampling rates, less storage, and fewer computations than discrete-time algorithms based on oversampling.

The optimal alignment that maximizes the cross-correlation between two waveforms $y(t)$ and $s(t)$ also minimizes the squared error between them. The maximum correlation alignment is thus achieved by the offset ϕ that minimizes the error

$$e^2=\sum_{n=0}^{N-1}|y(n+\phi)T)-s(nT)|^2 \tag{4}$$

where T is a sampling interval, and N is a total data number. This error can also be expressed in the frequency domain using Parseval's formula as follow;

$$e^2=\frac{1}{N}|Y_0-S_0|^2+\frac{2}{N}\sum_{k=1}^{\frac{N}{2}-1}|Y_{k\phi}-S_k|^2 \tag{5}$$

where S_k and Y_k, are the DFTs of $s(nT)$ and $y((n+\phi)T)$.

Therefore the DFT representation is well suited for aligning waveforms because time shifts which are fractions of the sampling interval can be computed by simple rotations, i.e.,

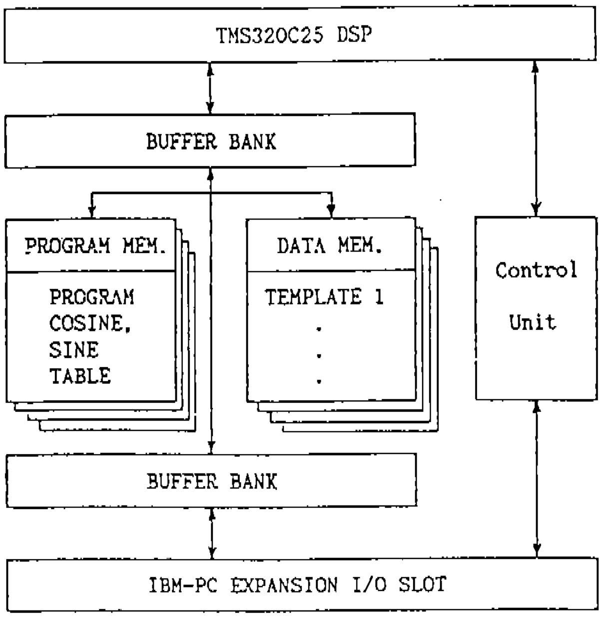

Fig. 3. Schematic block diagram of speed improvement system hardware

$$Y_{\kappa\phi} = Y_{\kappa} \exp(j2\pi\kappa\phi / N) \tag{6}$$

The least-squares (maximum-correlation) alignment algorithm can be extended to solve the problem of aligning more than one template with a superimposition waveform. The mismatch error in this case is

$$e^2 = \frac{1}{N} \sum_{n=0}^{N-1} | Y_k - S_{1,\kappa,\phi_1} \cdots - S_{M,\kappa,\phi_M} |^2 \tag{7}$$

where

$$S_{m,\kappa,\phi_m} \equiv S_{m,\kappa} \exp(j2\kappa\phi), \tag{8}$$

and $S_{m,\kappa}$ and ϕ_m are the κth DFT coefficient and the offset of the mth (of M) template. This error is minimized using the DSP chip with the output of the MAMDF filtering.

B) Hardware

In spite of the enhanced software approach, heavy computational requirement is remained. Therefore, the high resolution waveform alignment which is intensive calculation part of the total processing time is implemented on a TMS320C25 DSP chip.

Table 1. Scaling factors in the TMS320C25

Parameters	Scaling factors
Y_k	2^{-3}
$S_{m,k,\Phi}$	2^{-2}
$Re\{Y_\kappa * \exp(j2\pi kn/N)\}$	2^{-16}
$Im\{Y_\kappa * \exp(j2\pi kn/N)\}$	2^{-16}
e^2	2^{-16}

After reducing admissible template combination by MAMDF, the number of each template and its dip position are transferred to the data memory of the DSP board.

The block diagram of the hardware schematic consists of five parts: 1) The TMS320C25 which performs high speed processing of data (i. e., templates, superimposed waveform) transferred from IBM-PC; 2) The data memory which stores the DFT coefficients of the templates, superimposed waveform,and results of the MAMDF filtering; 3) The program memory for storage of the program of eq. (7) and tables (sine and cosine); 4) The control unit which supervises the system, transfers the results of the execution from DSP to IBM-PC, and; 5) The buffers which are used to eliminate the situation of both buses being active at the same time.

Since the TMS320C25 is a fixed point 16/32 bit multiprocessor, numerical operation can cause overflow or underflow. To avoid these problems, the proper scaling factors for each variable and parameter must be selected.

The scaling factors for all the variables are shown in table 1.

In this paper, all of the data were normalized so that the maximum value becomes 32767. Since the dynamic range of the DSP is from -2^{15} to $2^{15}-1$, computation computation results must be greater than -2^{15} but less than $2^{15}-1$. In the case of the multiplication, $Re\{Sk \cdot \exp(j2\pi\kappa n/N)\}$ or $Im\{S_k \cdot \exp(j2\pi\kappa n/N)\}$, the multiplier and the multiplicand can be the maximum value of 32767. Therefore, the results of the multiplication is scaled by 2^{-16}. In the same manner, the squared error(e^2) is scaled by 2^{-16} to avoid overflow.

When Y_k and $S_{m,\kappa,\phi}$ are aligned, the obtained error must not excess the upper range. Thus, they are also scaled by 2^{-3} and 2^{-2} respectively.

Experiment and Simulation

The performance of the proposed approach was evaluated by a computer simulation of template signals which were obtained by automatic signal processing method (ADEMG)[3] for extracting MUAP's from the electromyographic interference pattern. The recorded EMG signal from biceps branchii using concentric needle electrode during a 30 percent-maximal isometric contraction. Raw EMG signal is amplified and band-pass filtered (10Hz-4KHz), and digitized at 8192 Hz. The differential values of the collected EMG signal are used to the position detection of differential spikes that exceed threshold (In this paper used 50% of the maximum value). Each 64 samples of both sides at detected spike position (i. e., 128 samples in the center of the spike position) are stored for using as templates.

After the several MUAP's were identified by the ADEMG, the five templates (as shown in Fig. 1(a)) selected for resolving superimpositions.

The extra 15 templates were also used for the comparison of the execution time (See Fig. 4). From these signal, five cases of superimposed signals (as shown in Fig. 1(b)) were formed.

$$y_1(t) = s_2(t) + s_4(t\text{-}20)$$
$$y_2(t) = s_1(t) + s_3(t\text{-}10)$$
$$y_3(t) = s_1(t) + s_4(t\text{-}5)$$
$$y_4(t) = s_1(t) + s_2(t\text{-}20)$$
$$y_5(t) = s_1(t\text{-}10) + s_2(t) + s_3(t\text{-}20)$$

Next, the time domain data of the superimposed waveform are transferred to the main program written in C-language which is designed for easy control of the total processing routine and so to be convenient adjunct to the automatic qualitative resolving superimpositions. The main program performs the MAMDF filtering and the automatic interface control of IBM PC and DSP board. On the basis of these preparations, the DFT coefficients of the superimposed waveforms and of all the templates are transferred to the memory on DSP board, and computes the alignment error. To evaluate the speed performance of the alignment algorithm, the execution time was checked with each templates 5, 10, 15, 20.

Table 2. The MAMDF results of each superimposed waveforms

Input	Filter number and dip position
$Y_1(t)$	1/26, 1/3, 2/0, 2/16, 3/3, 3/17, 4/19, 4/4, 5/13, 5/-3, 5/-19
$Y_2(t)$	1/10, 1/0, 2/7. 2/-7, 3/10, 4/12, 5/-10
$Y_3(t)$	1/0, 2/-2. 3/3, 4/7, 5/12, 5/-12
$Y_4(t)$	1/24, 1/0, 2/20, 2/-2, 3/23, 3/-2, 4/25, 4/2, 5/18, 5/3, 5/-24
$Y_5(t)$	1/21, 1/8, 2/18, 2/0, 3/20, 3/4, 4/22, 4/8, 5/28, 5/15, 5/-1, 5/-18

Results and Discussion

The significant minima, the outputs of each MAMDF filter (as shown in Fig. 2), are detected by the simple threshold detection method (40 percent of the maximum dip amplitude). The position of the minima of the various filter outputs are listed in table 2 where in the notation denotes the filter number, and the relative time position (dip position).

As shown in table 2, a result of the MAMDF filtering of superimposed waveform, there are one and more dip positions for each of the templates and one of them is in the range of ± 5 sample length with respect to the original delay of the templates which are presented superimposed waveform. Therefore in this paper, the allowable range of variations for φ is chosen as ± 5 sample length. This indicates that the dip positions are reasonable conditions which reduce the combinations of the template.

Consider the case of y_1 shown in table 2 to examine the reduced computational requirement. The only deep minimum in y_1 corresponds to 1/0. Thus the incoming signal, y_1, is either the template number 1 (i. e. s_2) at time 0 or it is a combination of more than one template.

The computational procedure by eq. (7) goes as follows:

1. ϕ_1 is rotated by $1/p$ sample length, where $p = 2^n$, n = 1,

2,……

2. Real and imaginary part of the κth sample multiply by $\sin\phi_1$ and $\cos\phi_1$ respectively, and then divide into real and imaginary part.
3. In accordance with the combinations of template, 1,2 are repeated with $\phi_2, \cdots, \phi_m$.
4. The shifted template combination subtracts from the DFT coefficients of the superimposed waveform.
5. From step 1 to step 4 are repeated until each template shifts over data window length.

In the above steps, the second step requires 4 multiplications and 2 additions, and the third step requires iterative computations of the possible combination multiplied by the data window length. If this multiplication procedure is reduced, it can produce the speed improvement effect. If all the possible combination without the MAMDF filtering is examined, the computational requirement is:

$$N_{tp}\sum_{r=0}^{n-1} 4(pN)^{n-r}\, N^{1-sgn(r)} \qquad (10)$$

where $N_{tp}=\sum_{r=1}^{n-1} nCr$, is a number of the examined combination, n means a number of the templates, N is the data window length, p (see step 1) is degree of subdivision, and

$$sgn(r) = \begin{cases} +1 \text{ if } r>0 \\ 0 \text{ if } r=0 \\ -1 \text{ if } r<0 \end{cases} \qquad (11)$$

This computational requirement is reduced by the MAMDF. It requires following multiplication in the range of ± 5 sample length.

$$\overline{N}_{tp}\sum_{r=1}^{n} 4(11p)^{n-r} N^{1-sgn(r)} \qquad (12)$$

where $\overline{N}_{tp}$ is a number of the combination which is depended on detecting threshold level after the MAMDF. In the eq. (10) and (12), the eq. (12) leads the way to even lower-cost computational requirement by maximum rate of $\sum_{r=0}^{n-1}(N/11)^{n-1}$ because the window length N is reduced to 11.

Although the admissible combination is reduced by using the MAMDF, but it is considered computationally burdensome yet. In general purpose computer, addition or subtraction is processed by one instruction, but multiplication, especially sine or cosine multiplication, requires many instructions. To solve this problem, we employed the TMS320C25 DSP chip which requires one instruction cycle to multiply.

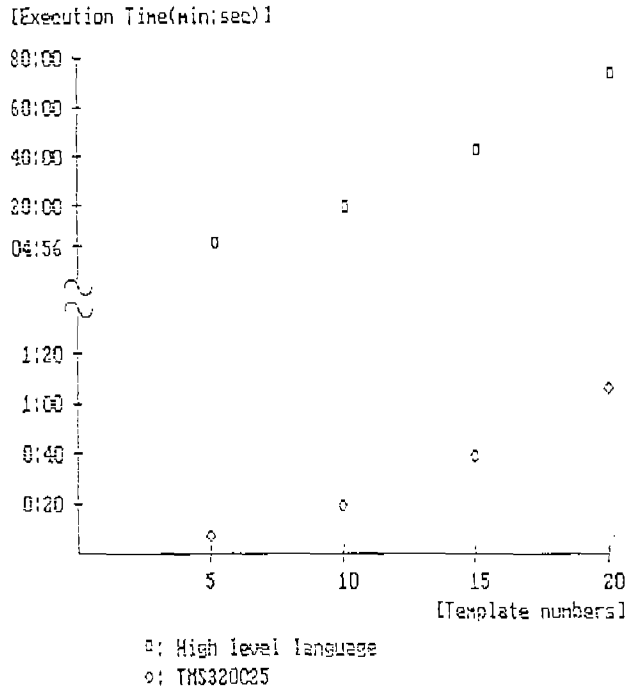

Fig. 4. The time plots of the execution performance using the DSP chip and highlevel language with IBM-PC

Fig. 4 represents results of the execution time comparison for the high resolution alignment between the DSP chip and the high level language after the MAMDF filtering. As shown in Fig. 4 the performance by the DSP chip is more efficient than that by highlevel language as the templates increase. In the case of resolving superimposed waveforms with 20 templates, it shows the very large difference of execution time. The comparison of the execution time between our proposed scheme and the optimization of the eq. (7) by the multidimensional form of Newton's method[5] simulated with highl evel language without the MAMDF filtering is shown in table 3. When there are 20 templates, these enhancements are expected to provide an overall speed improvement of 32-40 times.

The results of resolving superimposed waveforms (see eq.

Table 3. Execution time comparison between the DSP and Newton's method when the templates are 5, 10, 15, and 20

Template numbers	Newton's method (min : second)	Proposed method (min : second)
5	01:11	0:07
10	06:11	0:19
15	14:54	0:39
20	37:31	1:06

Table 4. The separation results by the proposed method

Input	Template number and dip position
$Y_1(t)$	2/0, 4/20
$Y_2(t)$	1/0, 3/10
$Y_3(t)$	1/0, 4/5
$Y_4(t)$	1/0, 2/20
$Y_5(t)$	1/10, 2/0, 3/20

(9)) is given table 4, which agree with actual values.

Consequently resolving superimposed waveform by the DSP chip and the MAMDF filtering has following adventages: Partial superimpositions are easier to resolve because the distinct peaks give an indication of the number of wavelets involved and provide accurate alignment points for the templates. For complete superimpositions on the other hand, not only it is impossible to tell how many wavelets are involved,but the peaks of the wavelets may be as much as a sampling interval away from the peak of the superimposition. In this case, even if the correct templates are known, peak-to-peak alignment will result in a poor reconstruction of the superimposition waveform. This method in this paper apparently works well in this regards.

By subdividing rotation angle as $^1/_2$, $^1/_4$, $^1/_8$,……,$^1/_p$, high resolution alignment of superimposed waveforms is also possible, that is this method has theoretically unlimited resolution, because the rotation of ϕ in frequency domain means the shift of the sample in time domain. Moreover, in order to achieve reliable identification of MUAP's from the superimposed EMG waveform, the decomposition should be performed under the condition when many templates being existed as possible.

Conclusion

Methods for decomposition of the interference EMG signals are time-consuming for practical use, and most previous template-matching schemes have been content to recognize superimpositions without attempting to resolve them. In this regards, we have described a new decomposition method which contains the enhanced software and hardware approachs. The software approach make a contribution to reduced search space and the high resolution alignment is achieved.

The efficient computation speed is achieved via DSP chip.

Finally, the proposed method demonstrates the accurate resolving of partial and complete superimpositions, and the overall speed improvement of 32-40 times to the others. In the proposed method the templates are obtained from the automatic signal decomposition. Therefore, this has always optimal template number and is applicable to several signal decomposition applications.

Reference

1. R. S. LeFever and C. J. De Luca, "*A procedure for decomposing the myoelectric signal into its constituent action potentials,*" IEEE Trans. Biomed. Eng., vol. BME-29, pp. 149-157, 1982.
2. P. Guiheneuc, J. Calamel, C. Doncarli, D. Gitton, and C. Michel, "*Automatic detection and pattern recognition of single motor unit potentials in needle EMG,*" in Computer-Aided Electromogr aphy, J. E. Desmedt, Ed. Basel, Switzerland: Karger, pp. 73-127, 1983.
3. K. C. McGill, "*A method for quantitating the clinical electromy ogram,*" Ph. D. dissertation, Stranford Univ., Stanford, CA, 1984.
4. R. J. P. De Figueiredo and A. Gerber, "*Separation of superimposed signals by a cross-correlation method,*" IEEE Trans. Acoust., Speech, Signal Processing, vol. ASSP-31, pp. 1084-1089, 1983.
5. K. C. McGill, L. J. Dorfman, "*High-resolution Alignment of sampled waveforms,*" IEEE Trans. Biomed. Eng., vol. BME-31, pp. 462-468, 1984.
6. M. J. Ross, H. L. Schaffer, A. Cohen, R. Freudberg, and J. L. Flanagan, "*Average magnitude difference function extraction,*" IEEE Tra ns. Acoust., Speech, Signal processing, vol. ASSP-22, pp. 353-362, Oct. 1974.
7. TMS320C25 *Simulator User's Guide,* Texas Instruments Inc., 1986.
8. TMS320C25 *User's Guide,* Texas Instruments Inc., 1986.