which requires that fresh adhesive tape be applied every 5-7 days, a task which the subject as a rule cannot perform by himself. On undulating surfaces such as the shoulder, the junction box tends to lift away from the skin surface during muscle contractions, so that tape replacement must occur more frequently at these locations. A conscientious subject also contributes heavily to achieving lengthy insertion durations by monitoring the quality of tape adherence and maintaining an awareness of those situations in which the junction box is likely to be snagged (e.g., dressing).

The second area requiring development concerns the occasional inability of the electrode to slide freely inside the needle. When this occurs, placement of the electrode in the muscle is not possible. Forcible removal from the needle of such electrodes reveals that they are "sticky," although they were not so prior to autoclaving. Varying the amount of catalyst or silicone fluid added to the Silastic has not affected this unpredictable phenomenon.

In summary, the fine wire coil electrode offers a great improvement in intramuscular EMG recording duration over present methods. There are some associated problem areas which require further development, but even as is, this design could be utilized in many current research areas.

ACKNOWLEDGMENT

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Filtering without Phase Shift

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Abstract—**The influences of phase shifts are extremely important in the determination of filtered waveforms and therefore in the interpretation of many biosignals. Low frequency shifts from high pass filtering, for example, are significant because large time displacements result from small phase deviations in the vicinity of the corner frequency. If a filter with a symmetric time response is employed, no phase shift will ensue. The time symmetry in this example is achieved with a tape recorder, first recording a high pass filtered signal and then playing back with time reversal to a second recorder through the high pass filter again. Real time processing is not achievable.**

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Most recorded physiological signals are filtered before analysis. Examples might include high pass filtering to compensate for drifts in the zero level, or low pass filtering before digitizing a signal to prevent aliasing errors. Filtering, which is usually a necessary first step in data processing, may introduce phase shifts in the data that one cannot afford to ignore. The analysis of maxima and minima and of wave shapes from physiological systems can have significant errors unless such phase shifts are compensated for.

FILTERING TECHNIQUES

A simple way to overcome this phase shift problem while still performing the necessary amplitude reductions is to filter the data twice, first directly, then with time inversion. If the analog signal is recorded on tape, this can be done by first filtering during normal recording, then, reversing the tape and playing back through the same filter while re-recording the result. Now events that occurred at a given time prior to being recorded and filtered will not be shifted in time due to the phase shift produced by the filter. The phase shift caused by the forward pass will be exactly canceled by the phase shift caused by the backward pass.

Let us assume that a given filter is linear in operation, having at each frequency ω an attenuation $\alpha(\omega)$ and a phase lead $\phi(\omega)$. The time lead corresponding to the phase lead is $\tau(\omega) = \phi/\omega$.

Assume that one is recording through a filter an event with a frequency component ω, and that the event occurs at time t_r. Due to the filter phase shift it is actually recorded on tape at a time $t_i(\omega) = t_r - \tau(\omega)$. Consider that the output of the filter is faithfully recorded and that the recorded signal is now used as a source, but the record is run backwards so that $t_o = T - t_i$ where T is the (time) length of the record, t_i is as above, and t_o is the time of the reversed playback recorder output. Due to the initial filtering, the ω component of the signal was recorded at time t_i but actually took place at t_r. Now consider that the playback signal is passed through the filter again (in the filter's forward direction). The filter would attenuate it by $\alpha(\omega)$ again and cause a phase lead $\phi(\omega)$. What goes into the filter at t_o comes out at $t_o - \phi/\omega$.

This output, of course, is in reverse order. To be understood, it must be run backward again, bringing it to the forward direction. We must compare $T - (t_o - \phi/\omega)$ with the timing of actual events. Substituting for t_o yields:

$$T - [(T - t_i) - \phi/\omega] = t_i + \phi/\omega = t_r - \phi/\omega + \phi/\omega = t_r.$$

Thus in the final playback the time relationships of events are the same as the actual happenings. The attenuation, however, is α^2. The mathematics shows that the process is independent of ω and therefore should work for all frequencies. Since all intermediate processes are linear, the total process should be usable for any wave form with frequencies within the capabilities of the tape recorder.

Fig. 1 shows the results using a square wave input. The square wave can be considered as the sum of the fundamental and the odd harmonics. Here a high pass filter is used. The bottom curve is the result of the two pass phase shift free technique. It can be seen to have the fundamental pretty well removed with some removal of the third harmonic. The square wave input is particularly appropriate as the symmetry of the process is then obvious. The middle tracing, made by passing the recorded square wave output through a conventional R-C high pass filter, also shows the removal of low frequency, but the great difference in wave form is due to the phase shifts.

Of course, the same technique may be employed in digital filtering, but not to avoid aliasing. In this case, a physically realizable filter may be used and the time series may be

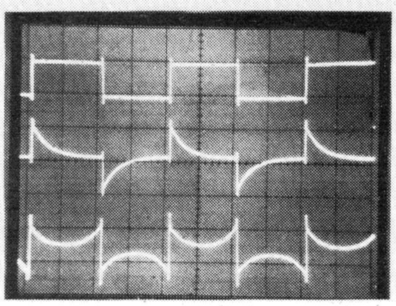

Fig. 1. The top trace is the square wave after a forward and reverse recording. The middle trace is a high pass filtered output made of the square wave input to the scope. The bottom trace is the high pass phase shiftless filtering system output.

passed through it both forwards and backwards, achieving attentuation α^2 and phase shift of zero, except possibly near end points. Another alternative is to use a physically nonrealizable filter that can use data points occurring both before and after the point of interest. If the filter is symmetric in time about the time point being processed, then phase shift due to filtering will be eliminated.

Conclusion

An n-pole filter is used twice, once while recording the input on magnetic tape, and once with the taped output time-reversed. The resultant filtered output has no temporal displacements.

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A Scheme for Biological Model Improvement (A Classroom Approach)

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***Abstract*—This communication presents a classroom procedure for minimizing the error between simulated data and experimental data to obtain the "best" parameters for a given system model. It is hoped that such experience helps in the process of biological model building and brings out the advantages of different types of simulations used for model improvement.**

Introduction

Currently an undergraduate course entitled, "Modeling of Biological Systems" is being offered at Clarkson College of Technology. The article "A simple heart model design to demonstrate biological system simulation" [1] was presented in the class as an illustration of a third order cardiovascular system. The approach was, in essence, given the model, use it to simulate the physiological system on an analog computer and then compare the results to those published. The authors of the above mentioned paper made a number of assumptions justifying the topology, and also gave an analog diagram complete with pot settings. There were some errors in scaling and pot settings. Upon proper scaling, the results given clearly were not consistent with those presented in the article.

The problem with any simulation is the development of the model itself. This reduces to a problem of identification. The present communication describes a classroom procedure for minimizing the error between simulated data and experimental data to obtain the "best" parameters for a given system model.

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Procedure

The procedure described is shown diagrammatically in Fig. 1. An input stimulus is provided for both the model and physiological system. The output of the model is compared to the smoothed data from the physiological system. The smoothing process may take the form of the commonly used parabolic regressive filter. This filtering is not essential to the procedure discussed but is useful in reducing the effects of noise and measurement inaccuracies. The error resulting from the comparison is used to adjust the parameters of the model so as to minimize this error. In this way the "best" parameters for the model can be obtained. If the data are accurate enough, simplifying assumptions can be easily tested. Such a scheme falls into a class of problems generally known as curve fitting and are most conveniently done on a digital computer. There are three main requirements for such problems:

1) An objective function which represents in some way the error between the simulated output and the physiological data.
2) A logical scheme to minimize this error.
3) A program to read in data, and monitor the above.

The objective or error function chosen is the sum of the squared error between each given data point and the corresponding calculated dependent variable for the same independent variable. When matching waveforms, the independent variable is usually time. Thus the error for such a case, is merely the sum of the squared difference between each observed data point and the calculated result at discrete instants of time. If more than one output is matched simultaneously, or the parameter being fit varies over a wide magnitude range, it may be best to express the error as the sum of the percent difference squared. For example,

$$\sum_{i=1}^{n}\left(1-\frac{A_i}{B_i}\right)^2 \quad \text{as compared to} \quad \sum_{i=1}^{n}(A_i-B_i)^2$$

where A_i is the calculated data point, B_i is the actual data point ($B \neq 0$), and n is the total number of data points.

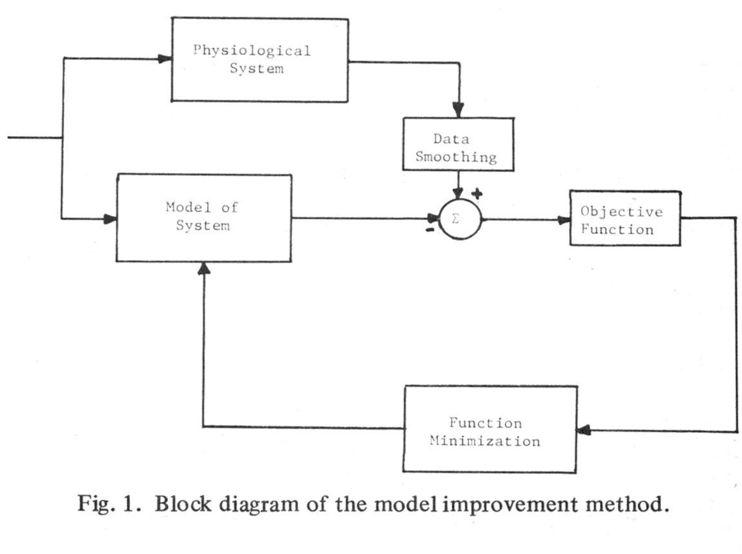

Fig. 1. Block diagram of the model improvement method.