Available online at www.elixirpublishers.com (Elixir International Journal)

Advanced Engineering Informatics

Elixir Adv. Engg. Info. 39 (2011) 4693-4696

ISSN: 2229-712X

Implementing cancellation of 50 Hz interference in ECG

N.J.R. Muniraj

Tejaa Shakthi Innovation Centre, Tejaa Shakthi Institute of Technology, Coimbatore.

ARTICLE INFO

Article history:

Received: 19 July 2011;

Received in revised form:

21 September 2011;

Accepted: 27 September 2011;

Keywords

Adaptive filter,

Electrocardiagram monitor,

Adaptive Noise Canceller (ANC).

ABSTRACT

As the world's population grows, the need for health care increases. This necessity has caused the invasion of engineering methods in medical field. A remote sensor is connected to an amplifier via a length of cable in order to measure the required parameter in the medical field. The amplifier output contains not only the sensor signal, but also a 50- or 60-Hz component due to stray pickup from power mains. Pickup of this type is often associated with sensitive high-impedance sensors like microphones and electrodes. Such hum may prove to be merely bothersome in a high-fidelity sound reproduction system, but it can be overwhelming when attempting to measure very small voltages, such as is found when trying to obtain a patient's electrocardiogram (ECG). This paper briefs the implementation of Adaptive noise cancellation algorithms such as LMS algorithm and RLS algorithm using MATLAB 6 (R12) suitable for real time implementation, which can be used during measurements, will be developed.

Introduction

If the spectrum of the sensor single does not include 60Hz, then the hum component can be removed via band pass filtering, but many applications, including ECG measurement, cannot tolerate exclusion of signal energy below 100Hz. Avery narrow notch filter might be appropriate, as long as the frequency of the electric power is steady enough to remain within the notch of the filter. Figure 1(a) illustrates this method.

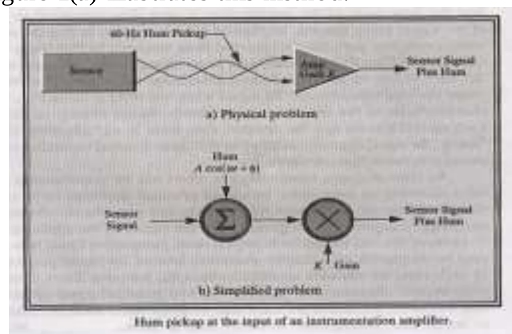

Figure 1.(a)

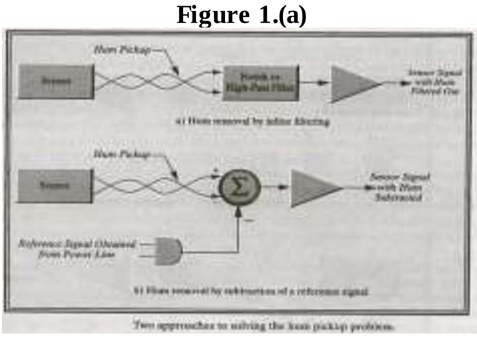

Figure 1.(b)

Another approach to solving this problem is shown in Figure 1(b). Suppose we measure the power main itself and attempt to use that signal to eliminate the 60-Hz component entering the amplifier. Clearly, if that can be done then the amplifier input and output will contain only the sensor signal. Moreover, if the power line frequency changes, then both the hum and the "reference" measurement will change together, allowing good cancellation to continue.

The problem with this cancellation approach is that the amplitude and phase of the 60-Hz reference signal must be carefully adjusted to make it accurately cancel the hum at the input of the amplifier. In some applications the required amplitude and phase settings might be determined once and then remained fixed. In most cases, however, the amount of stray 60-Hz pickup will change with the exact lead placement, the amount of power being drawn in the room, and any number of other factors that either change with time or over which the user has no control.

As a practical matter, then, the gain and phase rotation applied to the reference 60-Hz waveform must be variable and some automatic technique should be variable to adjust to adjust them in real time to sure good cancellation. Figure 1(a) shows a simplified version of how such a canceller might be designed with a variable gain G and phase. However, the variable phase shifter is difficult to implement at low frequencies like 60 Hz. The modified version of Figure 1.3(a), shown in Figure 1(b), is one step closer to practically because it uses two adjustable scaling coefficients w0 and w1 to control the gain and phase of the filter [1].

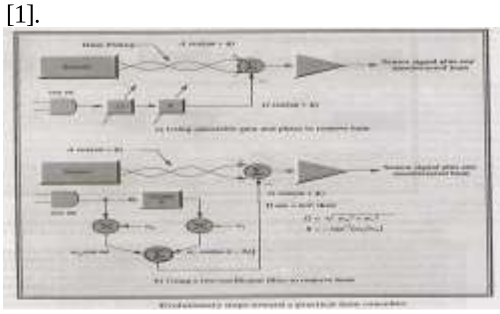

Figure 1.(c)

Adaptive Filters

An adaptive filter is very generally defined as a filter whose characteristics can be modified to achieve some end or objective, and is usually assumed to accomplish this

Tele:

E-mail addresses: njrmuniraj@yahoo.com

modification (or "adaptation") automatically, without the need for substantial intervention by the user. While not necessarily require, it is also usually assumed that the time scale of the modification is very slow compared to the bandwidth of the signal being filtered. Implicit in this assumption is that the system designer could (over any particular substantial time window) in fact use a time-invariant, maladaptive filter if only the designer knew enough about the input signals to design the filter before its use. This lace of knowledge may spring from true uncertainty about the characteristics of the signal when the filter is turned on, or because the characteristics of the input signal can slowly change during the filter's operation. Lacking this knowledge, the designer them turns to an "adaptive" filter, which can "learn" the signal characteristics when first turned on and thereafter can "track" slow changes in these characteristics [5].

Design: hum removal for an electrocardiagram monitor

Measurement of biological activity by means of monitoring electrical discharge, as typified by the monitoring of heart patients, parallels the communications problem: A transmitter (the electrical discharge) radiates energy through a propagation path (the body's tissue) to receiving antenna (an electrode) positioned to maximize energy reception. Because the electrical discharge involves very small potentials, the received energy is very weak and requires care to prevent degradation of the signal content by added noises or filtering. Probably the strongest source of interference is 50/60-Hz pickup and its harmonics emanating from nearby electrical equipment such as lighting and instrument power supplies. The conventional means for dealing with such strong, spectrally concentrated interference is a fixed, low-pass filter, which sacrifices waveform derail associated with spectral components above 50 Hz. Use of a notch filter suppressing the energy in the appropriate narrow spectral band represents an improvement; it still distorts the signal component of interest.

The introduction of the paper described means of removing the additive interference, not by filtering in the signal path, but by coherently subtracting a replica of the interference waveform. This noise –cancellation approach to the problem requires a very accurate match of the replica to the actual interference to achieve adequate suppression. For 30 dB of suppression., for example , the match in amplitude must be better than 3%, with a phase match of better than 2. To account for variations in frequency and amplitude of the stray interference, an adaptive filter can be used to adjust the phase and amplitude of the replica to maximize cancellation. The concept is shown in Figure

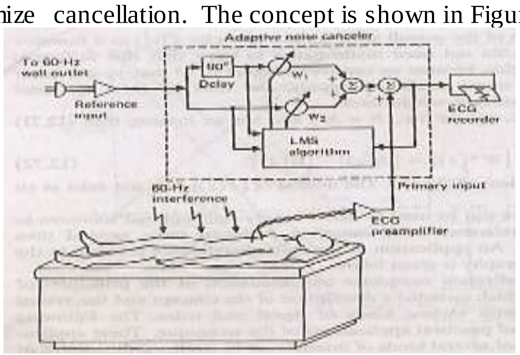

Figure 3.(a) Adaptive noise canceller algorithm

Design Approach

Consider the development of a circuit for use with off-the-shelf medical instrumentation. It must provide significant rejection of the offending 50/60 Hz (nominally) components, say in excess of 30 dB, and provide the enhanced output in real time. The goal is to degrade the real biomedical signal as little as possible. The processed signal is normally displayed using cathode-ray tube (CRT) deflection, so it must be available in analog form, although it would be attractive to have the signal available in digital form as well no adjustment unfamiliar to a competent medial technician. Adequate internal self-testing is necessary to assure the user of proper operation. Second, as such signal enhancement might initially be viewed as a luxury, manufacturing costs should be kept as low as possible. These concerns are very important when developing instrumentation for a no engineering filed; potential for sales may exist, but only if the design is kept attractively simple.

Fortunately, this problem lends itself to a simple solution. First, its bandwidth is low, so that we have a wide range of devices over which we can minimize costs and complexity. Arguments can be made that information of interest for the ECG waveform extends no higher than 100Hz. This implies that a sampling rate of 512 Hz would be more than adequate- high enough to satisfy the Nyquist condition for the signal of interest and the interference, yet low enough to keep the processor complexity down. We observe that offending interference is made up of a dominant 50/60-Hz component, a second harmonic (from rectifiers in power supplies), a third harmonic (from nonlinear ties in motors and transformers), and perhaps some low-energy higher harmonics. As implied by discussion the compelling filter requires two degrees of freedom for each sinusoidal component, which can be thought of as one for amplitude setting and one for phase alignment. Thus, at least six to eight active weights are required. The reference to be applied to the filter input is easily obtained. The power supply of the adaptive filter itself can be tapped to provide a 50/60-Hz source that operates at exactly the same frequency as radiated by nerve equipment. Simple harmonic –generating distortion (e.g., with a diode) can provide all of the spectral lines of interest to the filter. The waveform of interest is always present and its amplitude and frequency vary slowly, so speed of adaptation is not a concern. Rapidly converging algorithms are unnecessary, and the simplicity of the LMS implementation is another place where significant cost saving can result.

To first order, the net complexity of this hum-reducing device can be determined by the number of multiplications needed per second. Eight weights, each operating at the sampling rate of 512 Hz, and each updated at the same rate, requires less than 9000 multiplications per second. This low rate permits the hum canceller to be easily implemented in either of two-ways- in a "low-end" digital signal microprocessor, or in high-level language(e.g., C) on a computer's floating point processor as a part of other analytical and display functions.

Fortunately, this particular application permits a trick to be used to mitigate this problem. The signal of interest is a low duty cycle pulse waveform. Once the interference has been reduced by 10 to 20 dB and the pulses of interest are apparent, our processor can sense the pulses and then disable adaptation during the interval containing each pulse. This is termed gating the adaptation process. By adapting the filter when only the interference is present, there is very little weight jitter introduced by the signal of interest. Thus, even with the big step size needed to prevent weight stalling, there is little degradation of the signal of interest. Aside from the front-end analog circuitry, the hardware requirements include two 8-bit A/ D converters, a Processor, and a D/A converter if an analog version of the signal of interest is needed by downstream equipment. Some form of analog automatic gain control (AGC) and low-pass anti-aliasing filters are needed to condition the A/D inputs. Figure 8.2 shows the behavior of a fixed-point simulation of this proposed hum canceller. Note that the "heartbeat" spike is nearly totally masked, and within 2 seconds becomes essentially hum-free. The residual noise is due principally to additive measurement noise at a level near that expected for 8 bit input quantization. Note that better fidelity of the underlying signal of interest might well require higher-precision input sampling.

The adaptive noise canceller (ANC)

The hum remover just described uses an adaptive filter to produce an estimate of an interfering signal and then subtract it away from the corrupted signal of interest. This concept has proved to be very useful in a variety of applications. It is termed the Adaptive Noise Canceller (ANC) {Widrow et6 al., 1975b}. It has two inputs and a single output. The primary input $S_k + n_k$ contains the signal of interest plus one or more interfering signals. The second input, termed the reference input n'(k), is applied to the input of the adaptive filter. This reference input should contain as little of the signal of interest as possible. The objective in adapting the coefficients of the filter is to produce a filter output d(k) that matches, to the greatest extent possible, the exact wave shape of the interference signals appearing in the primary input. The filter output is subtracted from the primary input to produce the system output e(k). If the filter caned be adjusted to achieve a perfect match between filter output d(k) and the interference present in the primary signal $S_k + n_k$, the e(k), the system output, contains only the signal of interest.

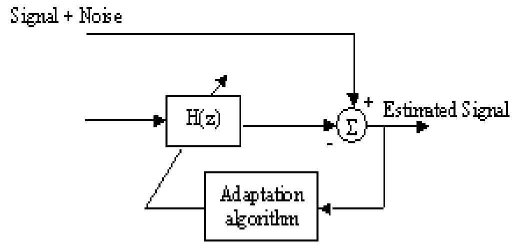

Figure 3.2(a) Adaptive noise canceller

Note that the ANC uses the system output e (k) as the error signal to drive the filter's adaptation. The rationale for this can be seen from the above discussion. When the filter's Coefficients are optimally adjusted, the presence off the interference in the error is minimized. By using an adaptive algorithm that minimizes the presence of the interference, the best coefficients can be found. A reexamination of Figure shows that it is in fact an ANC. The primary signal is that provided by the medical instrumentation containing both the ECG signal of interest and the 50/60-Hz interfering signal; component received from the room's power system. The reference input x (k) is both the system output and the error signal used to drive the filter's adaptive algorithm. As mentioned earlier, the ANC is widely used in practice. While the full range of design possibilities are available in terms of filter structure, performance functions, and adaptive algorithms, the most common designs use FIR filters, least-squares performance criteria, and the LMS approximate-gradient – descent adaptive algorithm.

An adaptive Filter is essentially a digital filter with self-adjusting characteristics.

When,

a) There is a spectral overlap between the signal and noise,
b) Band occupied by noise is unknown or varies with time,

adaptive filters are sought for. Adaptive Filter can be modeled as a 'NOISE CANCELLER ' in this regard [2].

How 50/60 Hz interferes with the ECG

The above figure shows the application of adaptive noise canceling in electrocardiography. The primary input is taken from the ECG preamplifier, the 50/60 Hz reference input is taken from a wall outlet either proper attenuation. The adaptive filter contains two variable weights. The two weights versions of the reference are summed to form the filter's output, which is subtracted from the primary input. Selected combination of the values of the weights allows the reference waveform to be changed in magnitude and the phase in any way required for cancellation. The two variable weights or two "degrees of freedom" are required to cancel the single pure sinusoidal signal.

Adaptive algorithms used for noise cancelation:

Adaptive algorithms are used to adjust the co-efficient of digital filter such that the noise is minimized. The algorithms widely used are

•LMS *(Least Mean Square Algorithm)*

•RLS *(Recursive Least Square Algorithm)*

Simulation using Matlab

The MATLAB program responded for both original ECG as well as manually generated ECG. However the response of the program for the original ECG was more apt. Although the ECG is uncorrelated with the 50/60 Hz line signal, the adaptive technique proved worthy. The 50/60 Hz line Signal (i.e.)., interference is cancelled from the Heart ECG Signal. ECG Signal was given as an input and was simulated using MATLAB 6. The result of the program is shown in the figure below.

For LMS Algorithm

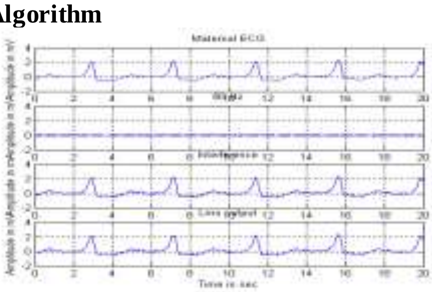

Figure 5.1(a) LMS output

For RLS Algorithm

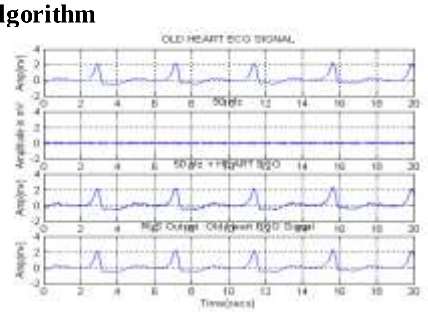

Figure 5.2(a) RLS output

Conclusion

The cancellation of 50 / 60 Hz interference from the ECG is more important in case of further processing of the signal because this causes the ECG to be dispersed and distorted. This same algorithm is used for cancellation of the same for any biomedical signals such as VAG, EEG etc,. The above concepts can also be extended to other algorithms, which can obtain enhanced results.

Reference

1.Bernard Widrow, Samuel D.Stearns, "Adaptive Signal Processing", Pearson Education Asia, second edition, 2002.

2.Emmanuel C.Ifeachor, Barrie W.Jervis, "Digital Signal Processing –A practical approach", Pearson Education Asia, Second Edition, 2002.
3.www.tmc.edu (Texas Heart Institute)
4.Igatavivus.D.D, Bayne. M.V, "Medical Surgical Nursing: A Nursing Process Approach", Philadelphia WB Saunders, 1191,Pg 2176.
5.Treichler, Johnson, Larimore, "Theory and design of Adaptive Filters", Prentice Hall of India Pvt. Ltd., Pg 269.

Authors Biography

N.J.R.Muniraj is presently working as a Principal of Tejaa Shakthi Institute of Technology, Coimbatore. He has more than 22 years of teaching and five years of industrial experience. He has presented more than 40 National and International papers and published fifteen international journal papers. His research area includes VLSI Signal Processing, Neural Networks, Image Processing and MEMS. He is also heading the Tejaa Shakthi Innovation centre. He expresses his sincere thanks to his chairman Mr.T.N.P.Muthu Natarajan and the secretary Ms.A.Tharalakshmi for their support and encouragement.