ECG Power line Interference Removal Using Combination of FFT and Adaptive Non-linear Noise Estimator

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Abstract: *Modern ECG instruments have a very high common mode rejection ratio to reduce power-line interference noise as much as possible. Nevertheless, the residual power-line noise should be eliminated. Due to the fact that, the central frequency of power-line may alters about ±1Hz and the desired signals around the central frequency should not be attenuated, applying the ideal notch filter with a fixed frequency or a band stop filter is not practical. Benefitting from the powerful microcontroller STM32F4 in a new brand of Electrocardiograph instrument "Araz630", we present a low cost efficient method of power-line noise removal. This method includes an FFT-based algorithm to find the center frequency of power-line and then subtracting the noise estimated by adaptive non-linear algorithm. In our implementation, we achieve between 30 to 35 dB of cancellation of the fundamental power-line noise component and the central frequency is obtained by the resolution of 0.1 Hz.*

Keywords: Adaptive noise estimation, power-line noise, STM32F4, CMSIS DSP

1. Introduction

Electromagnetic fields caused by a power-line (PL), represent a common noise in all bioelectrical signals recorded from the body surface such as ECG [1]. Such noise overwhelms signals of interest and makes sensitive or low-voltage measurements impractical [1,2]. This sinusoid noise is characterized by 50 or 60 Hz depending on the location, for example, 50-Hz AC power is used in Europe and Asia, whereas 60-Hz power is used in North America [3]. Such narrowband noise leads to more difficult ECG analysis and interpretation, since the delineation of low-amplitude waveform becomes unreliable and spurious waveform may be presented [4]. An example of power line interference in ECG signal can be seen in Fig. 1.

Various precautions can be taken into consideration to reduce the power line noise [5,6]. For instance a recording location with few surrounding electrical devices or shielding and grounding the location appropriately can reduce the power line interference [5,7]. Moreover, data acquisition analog hardware at a very early stage with common mode rejection circuitry

design can weaken the noise as well as the cardiograph right leg drive referencing. Nevertheless, a small residual may remain on the ECG signal that interferes with obtaining the highest quality signal. To this end, signal processing is necessary to remove such interference [3].

Several techniques have been presented for this purpose ranging from straightforward linear band stop filters to advanced methods of adaptive estimation [1]. A common approach to remove power-line interference is a fixed frequency notch filter [3]. The ideal notch filter has a flat frequency response over all frequencies of interest, except at 50 or 60 Hz, with zero frequency response. However, it is impractical because it suffers from two undesirable traits. First, the frequency response will not be flat, especially near the attenuated frequency where ringing occurs and this artifact can be seen as oscillatory behaviour near the sharp transitions in the time domain. [8,9]. In Fig. 2 this artifact is displayed after QRS complex. The second trait is that, the notch frequency cannot vary and, for the reason that the frequencies in 50 or 60 Hz power line references can alter as much as about ±1Hz, thus, the notch filter cannot attenuate noise with frequency shifts and it is unsuitable [8].

In non-ideal cases, the filter attenuates a narrow range of frequencies around the central power line frequency [2,3]. Widening the notch filter to include 49-51 or 59–61 Hz will attenuate some of the desired signals and, thus it is impractical [2].

Several methods such as subtraction, adaptive estimation, different types of adaptive and non-adaptive digital notch filters and so many other methods have been presented to solve the problem of PL noise reduction. Moreover, many researches were specified to improve the efficiency and rectify the current methods deficiencies such as signal distortion, unacceptable transient response time and imprecise computed central frequency [9,10].

Despite various researches performed and presented in a few previous decades, practical tests and experiments accomplished by famous and popular brands of ECG systems in hospital environment have shown the lack of

an efficient and low cost method of PL interference removal. The results have shown that these instruments eliminate almost all PL noise when they are tested by a simulator, whereas they show low efficiency while getting signals from body surface in the same environments. In addition, some instruments present different efficiencies in PL removal at different times in a day, which is likely due to the altering central frequency of PL. Altogether, in this paper we present an effective, practical and low cost method of PL noise removal in our ECG machine Araz630, that benefits from a non-linear filter for PL noise estimation/subtraction and by an adaptive central frequency detector using FFT algorithm.
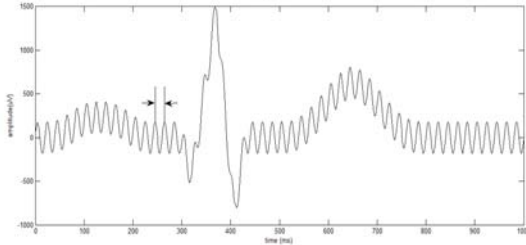

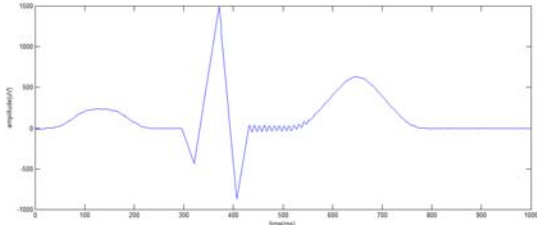Fig. 1: 50Hz power line interference in simulated ECG signal

Fig. 2: The ringing artifact after QRS complex filtered by notch filter. The ringing is due to the abrupt change in the frequency domain spectrum

2.   System Overview

Araz630 is a 6-channel electrocardiograph machine produced by Dahian Pezeshki Pishro Co[1]. In the hardware implementation of this system, neither the reference for PL noise nor the analog filters are not designed; but with a powerful STM32 microcontroller from F4 series, the DSP and the floating-point applications are accessible to perform real time advanced digital filtering methods.

By sampling rate frequency of 1000 sample per second, in each 10 ms, one packet of data is transferred to the microcontroller. Each packet includes 240 bytes that represents for 80 data samples (10 samples from each eight leads). Every sample is consists of three following bytes, combining with each other at the first step of software analysis, they form a 24-bit number. One bit (the MSB) is considered as data sign and the other 23 bits for the amplitude. Then, all samples are converted to the standard 32-bit two's complement (s32-sign32) format. The signals of other four leads are calculated from the current eight ones, after all pre-processing steps.

The first digital filter that is baseline wander removal is turned the s32 data format into a 16-bit signed (s16) one. In other words, by means of drift removal, the data range can be shifted into [-32767 :32768]. Therefore, the format of input data of PL interference removal filter is s16 with the rate of 1000 sample/sec. Three user-defined options of notch filtering are available in this product, including 50 Hz, 60Hz or no notch filtering.

3.   Noise Removal Method

As it is described in introduction part, a notch filter with a fixed frequency cannot be beneficial regarding to the altering frequency of PL noise for about ±1 Hz. In addition, a band stop filter with the cut off frequencies ±1 Hz around the center frequency will attenuate the desired signals around. For these reasons, an ideal PL noise removal would be a method that removes the exact frequency of PL noise while not affecting the other frequencies at all.

Our proposed method consists of an advanced FFT-based method to compute the PL center frequency as well as subtraction the estimated noise from the input signal that is called non-linear filtering method.

3.1   Non-linear Filtering Method

A non-linear filter is designed based on the idea of subtracting a sinusoid, generated internally from the observed signal [9]. The amplitude of the internal sinusoid is adapted to the PL interference present in the observed signal $x(n)$. The adaption process is the key to making the filter less sensitive to transients and avoiding related filter ringing. The internal sinusoid is generated by equation (1).

$$v(n) = w_0 \sin (w_0 n) \qquad (1)$$

Taking into account the fact that, the amplitude w_0 in practice is unknown and changing with time, it is preferable to generate sinusoid recursively. This allows us to update $v(n)$ at every sample, so the amplitude changes can be tracked. The sinusoid can be generated by an oscillator defined by a pair of complex-conjugated poles located on the unit circle at frequency w_0. The transfer function for the oscillator is:

$$H(z) = \frac{V(z)}{U(z)} = \frac{1}{1-2\cos(w_0 z^{-1})+z^{-2}} \qquad (2)$$

Accordingly, the sinusoid is generated by the following difference equation,

$$v(n) = 2\cos(w_0) v(n-1) - v(n-2) + u(n) \qquad (3)$$

Using the initial causal conditions and considering $u(n) = \delta(n)$, an error function $e(n)$ is identified by equation (4) .It indicates how well $v(n)$ predicts the PL interference in signal $x(n)$.

$$e(n) = x(n) - v(n) \qquad (4)$$

Since this error definition suffers from depending on the DC level of $x(n)$, it has to be modified to be insensitive to the baseline; for example by computing the first difference of $e(n)$.

$$\acute{e}(n) = e(n) - e(n-1) \qquad (5)$$
$$= x(n) - x(n-1) - (v(n) - v(n-1))$$

Obviously, other types of filters can be applied to remove the dc level efficiently and retain the sinusoidal interference, however the first difference filter is extremely easy to implement. Depending on the sign of $\acute{e}(n)$ the current value of $v(n)$ is either updated by adding a positive or negative constant (α), or kept unchanged to produce a new estimate $\hat{v}(n)$ of the power line interference. The update relation is defined by equation (6).

$$\hat{v}(n) = v(n) + \propto sign(\acute{e}(n)) \qquad (6)$$

The output signal of the non-linear filter $(y(n))$ is resulted from subtraction of $\hat{v}(n)$ from $x(n)$.

$$y(n) = x(n) - \hat{v}(n) \qquad (7)$$

Since changes in amplitude are limited by the increment α, the non-linear equation can suppress the noise by tracking the main signal. It is necessary to mention that, if a very small value of α is chosen, the filter can hardly track changes in PL interference amplitude; meanwhile if a very large value is assigned to α, the filter produces extra noise in $y(n)$ due to the large step alteration occurs in $\hat{v}(n)$. Before the next sample at time n+1 is processed, $v(n)$ is replaced by its estimate $\hat{v}(n)$, then it is used in the recursion to generate $v(n+1)$, and so on. In contrast to the IIR notch filter, no ringing artifact happens after the QRS complex by applying this filter. In the ECG signal adaption of the internal sinusoid primarily takes place during the isoelectric line and the T wave, the duration of the QRS complex is short enough to significantly influence the interference estimate $\hat{v}(n)$ [10]. Fig. 3 illustrates the convergence properties of the non-linear filter: a too large value of α cause the sinusoid part remains in the output signal. The frequency characteristics of this filter are not easily analyzed due to its non-linear structure [1].
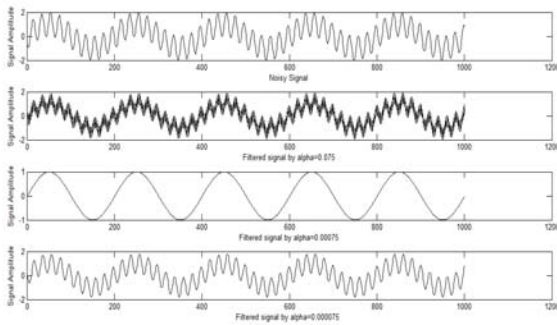

Fig. 3: Contaminated 5Hz sinusoid signal by PL noise with the center frequency of 49.76 Hz at the top figure is filtered by FFT_based adaptive non-linear method. As it is shown in second figure, if α is too large, extra noise will be produced. The variable α is properly set in the third figure, so the main signal is filtered completely. The forth figure is the result of applying an extremely small value of α, so filter could not eliminate the unwanted PL noise.

3.2 Power Line Noise Frequency Detection

The method of non-linear filtering is extremely efficient when the PL sinusoid noise frequency is well defined. Thus, the PL noise frequency had to be distinguished from the input data. The microcontroller used in this system is capable of utilizing the CMSIS DSP software library, a suite of common signal processing functions. DSP_Lib toolbox prepared for Cortex-M processor based devices, is divided into a number of modules each covering a specific category including: Basic math functions, Fast math functions, Complex math functions, Filters, Matrix functions, Transforms, Motor control functions, Statistical functions, Support functions and Interpolation functions.

The algorithm we applied to detect central frequency, benefits from one of the three types of FFT functions available in transform functions of DSP_Lib. The three types include Complex Fast Fourier Transform and its inverse (CFFT/CIFFT), Discrete Cosine Transform (DCT Type IV) and Real FFT function and its inverse (RFFT/RIFFT). CFFT is an efficient algorithm to compute Discrete Fourier Transform. DCT is constructed such that its energy is heavily concentrated in the lower part of the spectrum and is very widely used in signal and image coding applications. RFFT efficiently process real valued sequences with the advantage of requiring low memory and with less complexity.

The Type RFFT is selected to be applied on the signal. Since the PL noise frequency is almost the same in all leads, performing FFT function on the signal of one lead is enough to find the desired frequency. RFFT produces output FFT signals by the size of {128, 512, and 2048} points. Therefore, the number of {256, 1024, and 4096} data samples are needed as FFT input.

By applying FFT, the frequency response of the signal is obtained. As a result, the frequency with the highest absolute FFT value in the range of [49-51] Hz or [59-61] Hz is considered as the main frequency of PL noise to be removed. The frequency resolution of the result is completely related to the FFT size, but the decision about the length of applying FFT, is based on the system ability to perform these functions. As our system is unable to carry out FFT size of 2048, we had to benefit from 512-point FFT which desires 1024 sample points of one lead as input data. Considering F_S as the sampling rate frequency, the Nyquist frequency of the system would be $\frac{F_S}{2}$ Hz.

$$F_N = \frac{F_S}{2} = \frac{1000}{2} = 500\ Hz \qquad (8)$$

Therefore, by applying N-point FFT function on $2N$ data samples, the output would be N discrete values covering the range of $[0, \frac{F_S}{2}]$ Hz. Thus, the frequency resolution of

two following values in FFT output can be obtained by equation (9).

$$resolution = \frac{F_N}{N} = \frac{500}{512} = 0.9766\ Hz \qquad (9)$$

This resolution of about 1 Hz is not desired at all; since our filter is so sensitive to the center frequency, we had to specify the frequency precisely. In this regard, down sampling of input data seems to be an effective method. If the signal is down sampled by the rate of M, it means every M^{th} sample is kept starting with the first one, then the new sampling frequency and the new resolution is calculated by the following equations.

$$F_{S\ new} = \frac{F_S}{M}$$

$$resolution_{new} = \frac{F_{S\ new}}{2N} = \frac{F_S}{2MN} \qquad (10)$$

The higher down sampling ratio we select, the more accurate PL noise frequency is achieved. Regarding to the fact that, range of covered frequencies is decreasing by the factor of M, the maximum possible down sampling ratio is calculated by equation (11).

$$M = \max_x\ (x \in \{1,2,3,...\}|\ \frac{F_S}{2M} > max\ desired\ frequency) \qquad (11)$$

Where maximum desired frequency is the maximum frequency required for this study. PL noise frequency would be around 50 or 60 Hz, thus 61~62 Hz is the maximum frequency which is needed to find the PL central noise. Replacing the max desired frequency by 61~62 Hz, M and resolution is resulted by the following equations.

$$M = \max_x\ (x \in \{1,2,3,...\}|\ \frac{500}{M} > 62) \rightarrow M = 8 \qquad (12)$$

$$resolution_{new} = \frac{F_S}{2MN} = \frac{1000}{2*8*512} = 0.1221\ Hz \qquad (13)$$

The resolution of 0.1221 Hz is the most efficient possible resolution we obtain from this system, leading us to distinguish PL noise frequency precisely.

To sum up, down sampling the input signal by ratio of 8, and considering the fact that we need 1024 input samples for FFT, we have to wait $8 * 1024\ ms \simeq 8sec$ to find the main frequency of PL noise and transfer it to the Non-linear filter. So the process of FFT recurs every 8 Sec, when the input buffer of FFT, is filled with new samples again. If an alteration in PL noise center frequency occurs, 8 seconds is needed to follow the changes. Since the central frequency does not change much in a short time, 8-second is not a big deal in online data analysis. At the end, the frequency with the highest absolute FFT value in range of [49-51] Hz or [59-61] Hz found in each

duration (f_c) is used to compute the following equation for the next 8-second.

$$\cos w_0 = \cos(\frac{2\ \pi\ f_c}{Fs}) \qquad (14)$$

4. Experiments and Results

We present a few tests to demonstrate the ability and effectiveness of our PL noise cancellation method.

The first test, applied on a simulated ECG signal that in contaminated by PL noise with SNR=1 and main frequency of 50.4273. The contaminated signal before and after PL noise removal can be seen in Fig. 4 as well as the absolute FFT spectrum of output signal and down sampled input data. As it is shown the PL noise with the main frequency of 50.54 Hz and the amplitude equals to the signal is well removed.

To test the convergence rate of the system—how fast it can achieve suitable cancellation—PL noise is simulated and passed true the system. Although the convergence rate is somehow related to the variable α, but with a moderate value, the system will be stable within 2.5 seconds. Convergence for AC line noise with the center frequency of 49.62 Hz is shown in Fig. 5. It is interesting to note that, in order to reduce the delay of the system in PL noise removal at the start up time, the initial central frequency is set to be 50 or 60 Hz (depending on the setting) to reduce PL noise as much as possible within the first 8 seconds.

We also add 45 Hz signal to the PL noise with the center frequency of 49.76 Hz and apply our method to determine the elimination efficiency of the system. Fig. 6 shows the absolute FFT spectrum of contaminated signal before and after cancellation. As it is displayed, the PL noise is reduced by more than 30 dB while the desired 45 Hz remains unchanged. A typical notch filter implementation achieves only about 15 dB of cancellation [8].

5. Discussion
The noise cancellation system described in this paper produces 35–40 dB of cancellation of a pure PL sine noise in range of 49 to 51 Hz and 59 to 61 Hz. This simple low cost method is presented to be applied for new powerful microcontroller such as STM32F4 series, which is capable of online FFT process and floating-point applications. Using the 512 points FFT functions in DSP_Lib toolbox of CMSIS, and applying it on down sampled input signal we calculated the center frequency of PL noise by the resolution of 0.1 Hz. The efficiency of this method in finding the precise center frequency solved the problem of efficient PL noise removal methods.
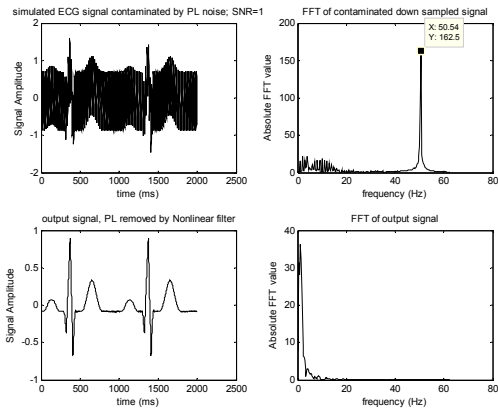
Fig. 4: Contaminated ECG signal by PL noise with the center frequency of 50.42 Hz filtered by FFT_based adaptive non-linear method. The center frequency obtained by FFT is 50.54 and the noise cancellation can be seen in FFT spectrum of output signal
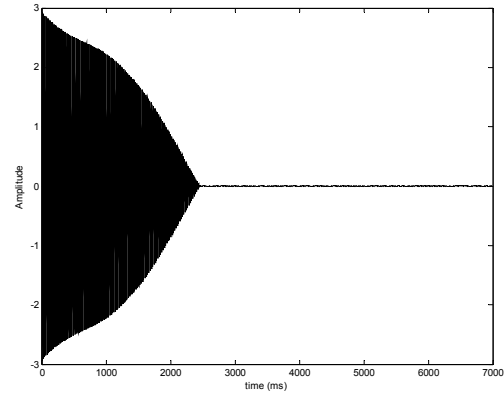

Fig. 5: Convergence of pure PL noise with the center frequency of 49.62 Hz. Convergence occurred in less than three seconds
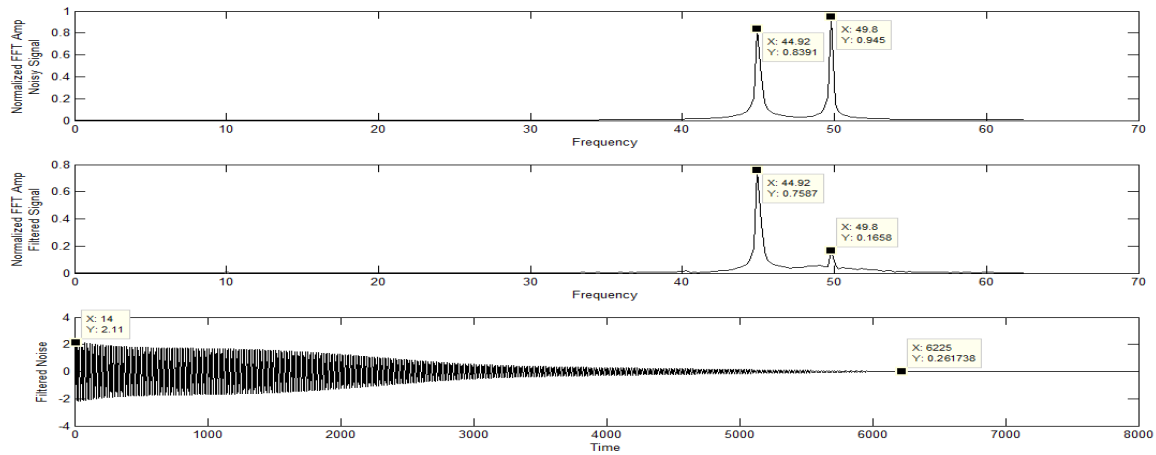

Fig. 6: The above figure shows the normalized FFT amplitude of noisy signal that is combined of a 45Hz sinusoid signal with a 49.76Hz sinusoid noise. The second figured displays the same measure of the filtered signal. The FFT amplitude of noise signal is decreased explicitly while the main signal remains unchanged. The last figure, displays the noise suppression with the attenuation of at least 35 db.

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