University of Al Mustaqbal

Biomedical Engineering Department

Biomedical Signal Analysis Lab

Title: Adaptive filter for the power line interference

Variant 2

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Work # 4. Adaptive filter for the power line interference

Objective: examination of adaptive filter for the 50 Hz power line interference suppression.

Theory basics

The 50 Hz (or 60 Hz in the USA and Canada) power line interference represents one of the most widespread noises that are usually present in biomedical signals.

This work suggests an investigation of an adaptive notch digital filter for this kind of noise suppression.

This filter is based on the following well-known trigonometric formula:

$$\mathbf{sin}(\boldsymbol{\theta}+\boldsymbol{\delta})=\boldsymbol{2con\delta}\,\mathbf{sin}\,\boldsymbol{\theta}-\mathbf{sin}(\boldsymbol{\theta}-\boldsymbol{\delta})$$

If the value θ is interpreted as the current time, and the value δ – as the time distance between two adjacent samples of a discreet signal, the above formula can be transferred to:

$$\boldsymbol{a_{i+1}=C\,a_i-a_{i-1}}$$

Where ai-1, ai & ai+1 – successive samples of some sine signal, and C = 2 cos δ – a constant value. In the other words, this formula predicts the next value of a digitized sine signal (ai+1), using two preceding values (ai-1 and ai).

The adaptive filtering algorithm, analyzing the signal in current mode, step by step adjusts the filter parameters to the sine component with certain frequency and then subtracts it from the signal that results in suppression of this sine component. One of the most important parameters of this filtering algorithm is so called adaptation step that determines the speed of the filter adjustment and the filtration quality.

Main tasks of the work

- Development of the program that realizes adaptive filtering algorithm and examination of this program with the use of test signal.
- Adaptive filtering of the real ECG signal, containing the 50 Hz powerline interference and adjusting of the adaptation step optimal value.

Supplement #2: Table W.2

Variant	*Fs*, Hz	*A*	*tmax, s*	File
1	500	50	0.5	W4_01.txt
2	400	10	0.7	W4_02.txt
3	450	15	0.6	W4_03.txt
4	350	40	0.8	W4_04.txt
5	300	7	1.0	W4_05.txt
6	250	1	1.2	W4_06.txt
7	360	12	0.9	W4_07.txt
8	280	60	1.4	W4_08.txt
9	200	150	1.5	W4_09.txt
10	240	80	1.3	W4_10.txt

We will use **Variant Nº 2**

Part 1: Examination of the adaptive filter with the use of test signal

<u>Code (fs50)</u>

```
function y=fs50(x,Fs,da)
a=0;
a1=0;
ax=0;
C=2*cos(2*pi/(Fs/50));
N=length(x);
x1=x(1);
for i=1:N
   dy=(x(i)-a)-(x1-a1);
   if dy~=0
      if dy>0
         a=a+da;
      else
            a=a-da;
      end
   end
y(i)=x(i)-a;
ax=a;
a=a*C-a1;
a1=ax;
x1=x(i);
end
```

<u>Code (P1)</u>

```
Fs=400;
A=10;
tmax=0.7;
F_pli=50;
F1=120;
F2=30;
F3=78;
A1=5;
A2=20;
```

```
A3=15;
t=0:1/Fs:tmax-1/Fs;
PLI=A*sin(2*pi*F_pli*t);
S=A1*sin(2*pi*F1*t)+A2*sin(2*pi*F2*t)+A3*sin(2*pi*F3*t);
x=PLI;
figure(1);clf;
subplot(3,3,1);
plot(t,x);hold on
plot(t,S);
da=0.19625;       %0.25 suprresion
y=fs50 (x,Fs,da);
subplot(3,3,4);
plot(t,y);
B=x-y;
subplot(3,3,7);
plot(t,B);
da=0.3925;        %0.5 supression
y=fs50 (x,Fs,da);
subplot(3,3,5);
plot(t,y);
B=x-y;
subplot(3,3,8);
plot(t,B);
da=0.58875;        %0.75 supresion
y=fs50 (x,Fs,da);
subplot(3,3,6);
plot(t,y);
B=x-y;
subplot(3,3,9);
plot(t,B);
```

Obtained graph

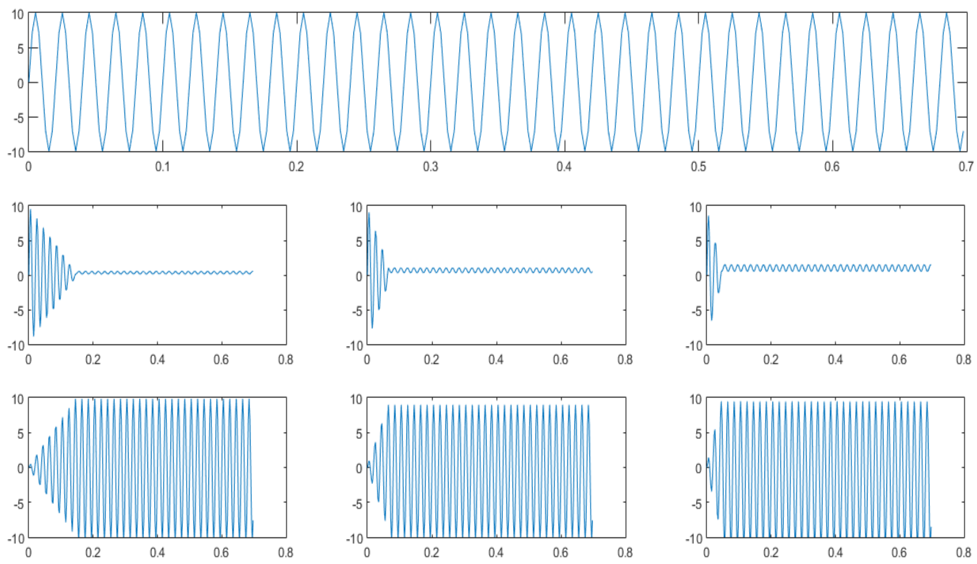

Figure 1: Examination of the adaptive filter with the use of test signal

Part 2: Adaptive filtering of the real ECG signal, containing the 50 Hz powerline interference.

Code

```
Fs=250;
T=1/Fs;
x=load('W4_02.txt');
N=length(x);
tmax=N*T;
t=0:1/Fs:tmax-1/Fs;
da=0.25;
y=fs50 (x,Fs,da);
subplot(3,1,1);
plot(t,x);
subplot(3,1,2);
plot(t,y);
B=x-y';
subplot(3,1,3);
plot(t,B);
```

Obtained graph

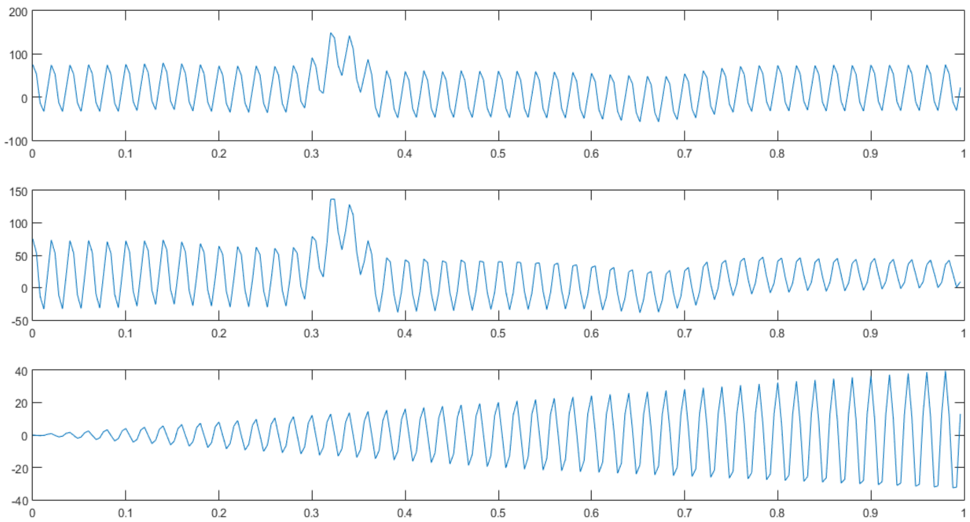

Figure 2: Adaptive filtering of the real ECG signal, containing the 50 Hz powerline interference.

Conclusions

- An adaptive filter is a system with a linear filter that has a transfer function controlled by variable parameters and a means to adjust those parameters according to an optimization algorithm. Because of the complexity of the optimization algorithms, almost all adaptive filters are digital filters. Adaptive filters are required for some applications because some parameters of the desired processing operation (for instance, the locations of reflective surfaces in a reverberant space) are not known in advance or are changing. The closed loop adaptive filter uses feedback in the form of an error signal to refine its transfer function.
- A Notch Filter is also known as a Band Stop filter or Band Reject Filter. These filters reject/attenuate signals in a specific frequency band called the stop band frequency range and pass the signals above and below this band. For example, if a Notch Filter has a stop band frequency from 1500 MHz to 1550 MHz, it will pass all signals from DC to 1500 MHz and above 1550 MHz. It will only block those signals from 1500 MHz to 1550 MHz.
- The quality factor of a notch filter is the entire stopband from the -3dB cutoff points over the null frequency, where the attenuation is the greatest. The quality factor shows how narrow or wide the stopband is for a notch filter.
- A comb filter is a filter implemented by adding a delayed version of a signal to itself, causing constructive and destructive interference. The frequency response of a comb filter consists of a series of regularly spaced notches, giving the appearance of a comb.While, the Notch Filter reject/attenuate signals in a specific frequency band called the stop band frequency range and pass the signals above and below this band.