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Designing and implementing a narrow double notch filter to remove two undesired tones from an audio signal

Maulod Khanan, Mohamed Shaban

Lecturer, Department of Electric and Electronic, Sabratha Higher Institute of science and Technology, Libya
Associate Professor, Department of Electric and Electronic, Surman College Of science and Technology, Libya

ABSTRACT: Digital signal processing (DSP) techniques have rapidly developed in the recent years due to advances in digital computer and integrated circuit fabrication. Notch filters are invariably used in control, communication, instrumentation, and bio-medical engineering, besides a host of other fields, to eliminate power line interferences and noise. Digital notch filters can be designed as finite impulse response (FIR) as well as infinite impulse response (IIR) structures. The aim of this research is going to process the sampled signal to remove the tones while keeping as much of the signal intact as possible. The filter will be implementing by Matlab program to surgically eliminate to high noise frequencies.

KEY WORDS: Double notch filter, IIR filter, FIR filter, noise signal.

I.INTRODUCTION

Digital filters, nowadays, play a major role in sound and image processing. It is considered a mathematical model that performs a certain operations on sampled, discrete-time signals to obtain the required performance. It's a powerful tool and its power is the convenient implementation on computers in real-time application. In this research, an analog signal, which is sampled by period of T=1/8192 second, is corrupted by two continuous-time noise tones of f1=697 Hz and f2=1209 Hz. So the task is how to remove these two tones from the original signal which are represented by the two red lines in Fig. (1). A double notch filter will be designed and implemented on MATLAB program such that the original signal can be conventionally heard.

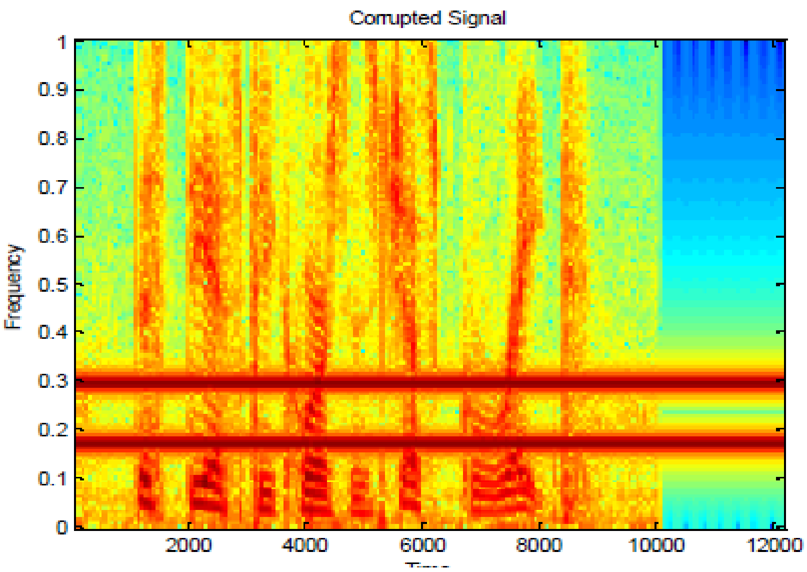

Fig.1: Corrupted Discrete Signal

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The entire process is described in Fig. (2) Where x (t) is the continuous-time signal and the X[n] is the discrete-time signal and X' [n] is the corrupted discrete-time signal by the noise continuous-time signal N (t). The noise corrupts the discrete signal X[n] to produce from the discrete-time system Y' [n] instead of y[n]. Y' (t) and y (t) are the heard-able continuous-time signals where y (t) is the filtered signal and y' (t) is the corrupted signal.

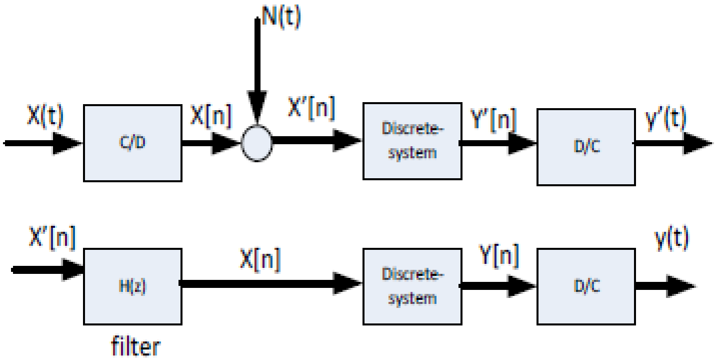

Fig. 2: The Project Process

II. FILTER DESIGN

Identifying the filter transfer function is the main task. First of all, the zeros of the filter are need to be determined from noise frequencies. The noise frequencies are in Hz, so we need to find out the corresponding values in red/sample and that achieved by dividing the original sampling frequency which is Fs=8192 sample/sec . This step is achieved by

$$w_1 = \frac{2\pi f_1}{f_s} = \frac{2\pi * 697}{8192} = 0.534 red / sample \quad (1)$$

$$w_2 = \frac{2\pi f_2}{f_s} = \frac{2\pi * 1209}{8192} = 0.927 red / sampie. \quad (2)$$

After that, the corresponding angels in degrees of these discrete frequencies are calculated by

$$\theta_1 = \frac{180 * w_1}{\pi} = 30.6^0 \quad (3)$$

$$\theta_2 = \frac{180 * w_2}{\pi} = 53.12^0 \quad (4)$$

So, the polar forms of these vectors are the representation of the two pairs of zeros of the filter on the unit circle, where they calculated by

$$x_{Z1} = \cos(\theta_1), y_{Z1} = \sin(\theta_1). \quad (5)$$

$$x_{Z2} = \cos(\theta_2), y_{Z2} = \sin(\theta_2). \quad (6)$$

Hence, the two pair zeros are

$$Z_{1,2} = 0.8605 \mp 0.5095i, \quad (7)$$

$$Z_{3,4} = 0.6000 \mp 0.8000i, \quad (8)$$

Then, identifying the poles of the filter. These poles are chosen arbitrary too close to the calculated zeros on the same angle but with radius less than the unit circle radius, such that they reduce the effect of the zeros on the other frequencies. These poles are $P_{1,2} = 0.825 \mp 0.495i$, $P_{3,4} = 0.5850 \mp 0.770i$,

Figure (3) shows the distribution of the poles and the zeros in Z-plane with unit circle.

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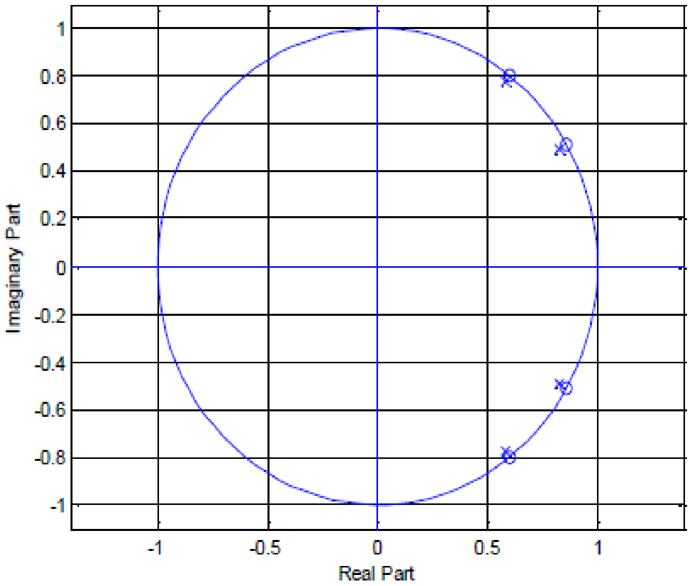

Fig. 3: Zeros: O & Poles: X in Z-plane

In Figure (4), The zeros and the poles of the filter are shown in 3-D of Z-domain, where the zeros located on the unit disc making it goes down to zero at the undesired frequencies and the poles go to infinity in |H (Z)| axis, this value is out of Matlab scope, to reduce the effect of zeros on the other desired frequencies.

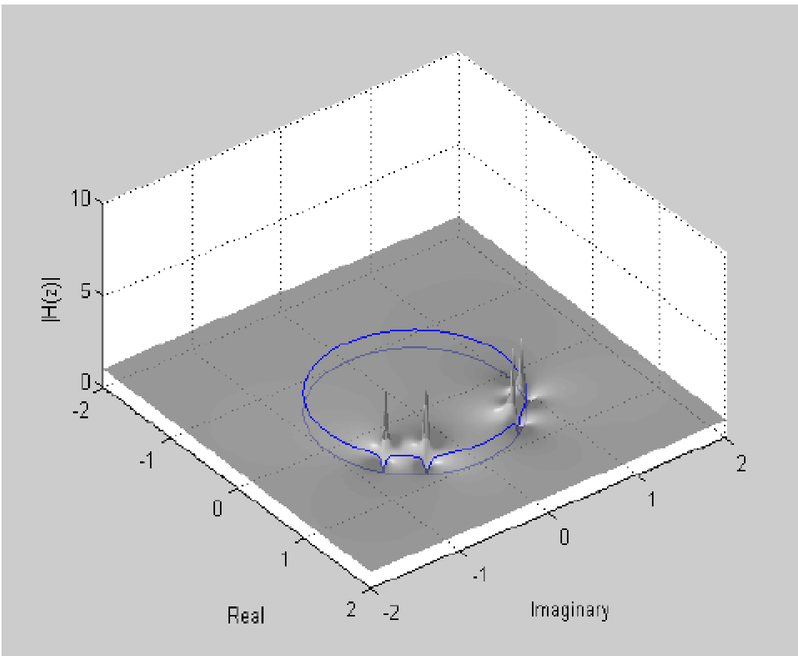

Fig. 4: 3-D zero/pole Locations

Now, the transfer function of the filter has the following form of

$$H(z) = \frac{(z - 0.8605 \mp 0.5095\,i)(z - 0.60 \mp 0.80i)}{(z - 0.825 \mp 0.495i)(z - 0.5850 \mp 0.770i)} \tag{9}$$

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The difference equation of filter will be calculated Form its transfer function. Where $H(z)=\frac{y(z)}{x(z)}$, then by Matlab we can find out the coefficients of the difference equation.

$$H(z)\frac{y(z)}{x(z)}=\frac{z^4-2.8200\,z^3+3.7863\,z^2-2.6202\,z^1+0.8610}{0.9279\,z^4-2.7103\,z^3+3.7720\,z^2-2.7103\,z^1+0.9279} \tag{10}$$

By multiplying the R.H.S by $\frac{z^{-4}}{z^{-4}}$ and then cross multiplication we obtain

$$0.9279\,y[n]-2.7103\,y[n-1]+3.7720\,y[n-2]-2.7103\,y[n-3]+0.9279\,y[n-4]=X[n]-2.8200\,X[n-1]+ \\ 3.7863\,X[n-2]-2.6202\,X[n-3]+0.8610\,X[n-4] \tag{11}$$

$$y[n]-2.9209\,y[n-1]+4.0651[n-2]-2.92093\,y[n-3]+y[n-4]=1.0777\,X[n]-3.0391\,X[n-1]+ \\ 4.0806\,X[n-2]-2.8238\,X[n-3]+0.9279\,X[n-4] \tag{12}$$

In figure (5), the frequency response of the required filter is shown. Where the magnitude of $H(w)$ of the filter is zero at the undesired frequencies and one otherwise.

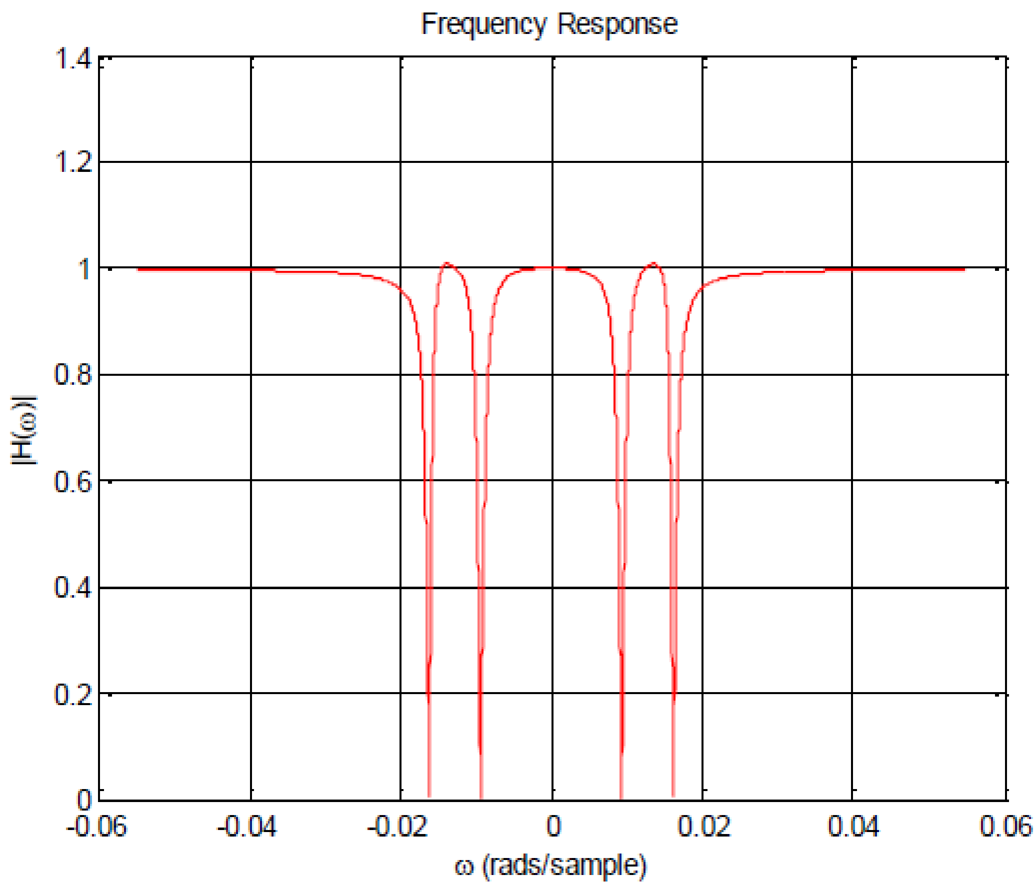

Fig. 5: Frequency Response

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III.EXPERIMENT RESULTS

In this part, the filter is implemented on Matlab as in figure (6). Filtering the corrupted discrete-time signal X'[n] by passing it through the designed filter to fetch X[n] and then into the discrete system to get y[n] then just resembling it to hear the original speech on the phone signal.

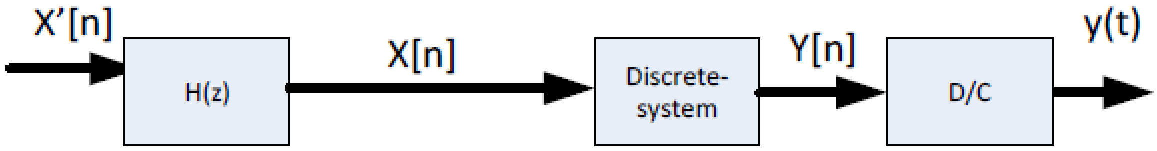

Fig. 6: Applying the Filter

After the task is accomplished, the spectrogram of the corrupted signal and the filtered signal are concatenated with each other on the same figure (6), where the corrupted signal starts first and ends at3 sec., then the filtered signal starts at 3 and ends at 6 sec. we can notes that the two red lines which are the high frequencies noise tones are removed in the filtering part of the figure; however, two tiny yellow lines are lift in their places that the same frequencies of the original signal also gone with the undesired frequencies.

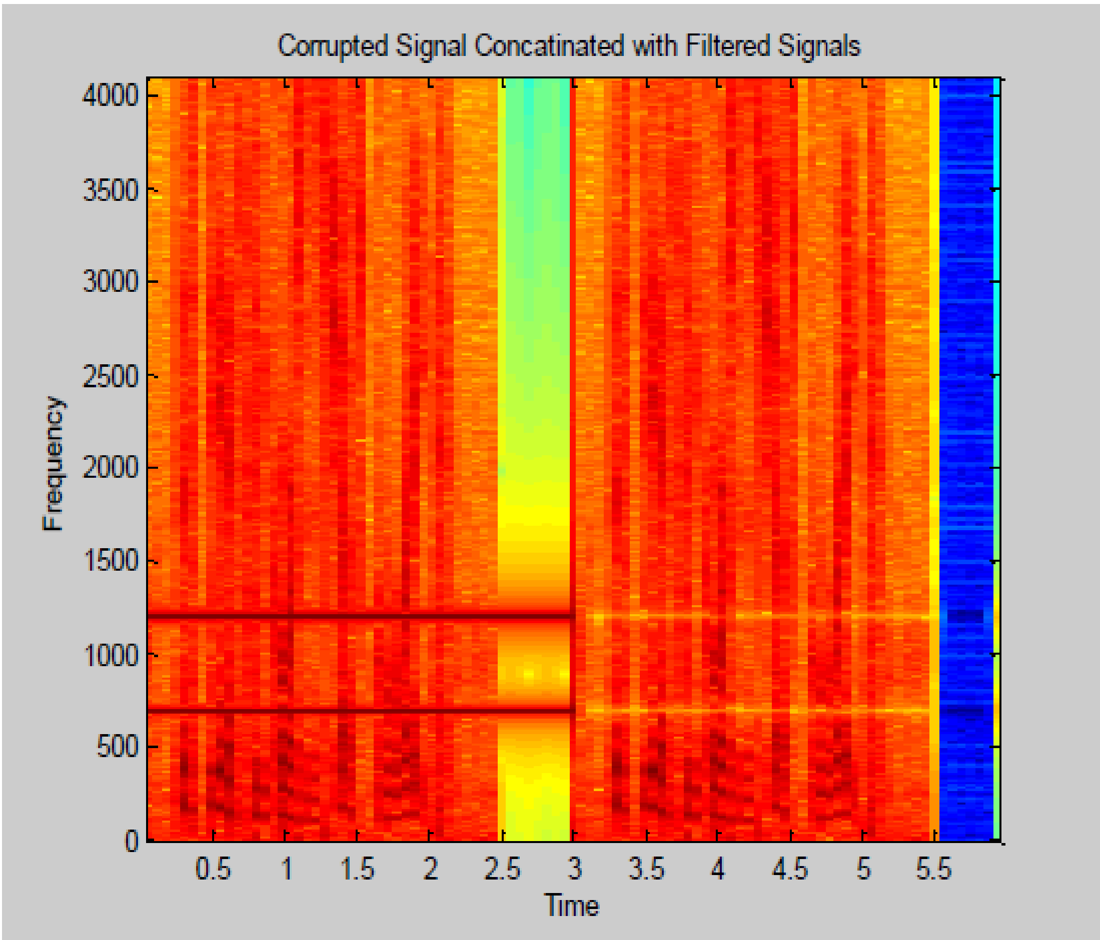

Fig. 7: Corrupted & Filtered Spectrogram Signals

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IV. CONCLUSION

In this research an IIR filter has been implemented by Matlab program to surgically eliminate to high noise frequencies. The zeros of the filter were calculated according to those undesired frequencies such that they will be eliminated from the original signal. The poles of the filter were arbitrary chosen close to the undesired frequencies location to reduce the effect of the zeros on the other frequencies.

Finally, the experiment was successful and the undesired noise was gone after filtering the corrupted signal by the designed filter.

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