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Minimum Coefficients Digital Input Speech Filtering For Communications Devices Applications

Mutasim I. Malik[1], Mohammed S. Khesbak [2] , Raad Mutar Dhahiri[3]

[1]College of Science , Wasit University, Wasit, Iraq

[2]Al-Rafidain University College, Baghdad, Iraq

[3]College of Science , Wasit University, Wasit, Iraq

ABSTRACT

In most communications devices the input speech is severing from high frequency interference. This interference may be sounds generated from many sources such as industrial machinery and music devices with relatively higher amplitudes than human voice. This paper investigates the design of a digital filter with a minimum number of coefficients suitable for reducing the effect of such high frequency components on human speech in a certain communications device.

1. Introduction

The science of speech processing on a large degree of importance, because speech is the most important means of communication between humans [1]. The speech signal is within the low frequencies up to(4KHZ) [2], while the environment around us produce high frequencies up to (20KHZ) [3], these sounds are interfere with the speech signal in the carrier devices for voice and cause a decrease of speech intelligibility and sometimes cause completely lost [4], So suggested building a digital filter (low-pass filter) has the ability to reduce or prevent high frequencies with the lowest effect on the identity of the speech .

2. Digital Filters

Digital filters are mathematical algorithms implemented in hardware or software or both that operates on a digital input signal to produce a digital output signal for the purpose of achieving a filtering objective [5]. There are two main types of digital filters, Finite Impulse Response (FIR) and Infinite Impulse Response (IIR) digital filters [6]. (FIR) filters are linear phase, high filter order (more complex circuits) and high stability, while (IIR) filters are non-linear phase characteristic, low filter order (less complex circuits), and unstable. The IIR filter is required to be used when the only important requirement is the sharpness of cutoff due to its non-linear phase relation [7]. Moreover, the IIR filter has the advantage of lower resultant coefficients compared to FIR type [8]. The relation between input and output signals of IIR filters is [6]:

$$y(n) = \sum_{k=0}^{N-1} h(k)\, x(n-k) \qquad (1)$$

and the transfer functions is:

$$H(z) = \frac{\sum_{k=o}^{N} b_k\, z^{-k}}{1+\sum_{k=1}^{M} a_k\, z^{-k}} \qquad (2)$$

To design of IIR filter, the popular method is first to design an analog filter with desired specifications then mathematically converting into an equivalent IIR digital filters [9], there are two main methods(bilinear and impulse invariant transformation) to converting analog to digital filter [10].
Bilinear transformation formula is;

$$s = \frac{2}{T} * \frac{1-z^{-1}}{1+z^{-1}} \qquad (3)$$

And impulse invariance transformation is;

$$\frac{1}{s+P_i} = \frac{1}{1-e^{-P_i T} z^{-1}} \qquad (4)$$

3. Proposed system design considerations

Figure (1) shows the two different sources time domain waveforms of the desired human speech and the environmental undesired high frequency interference plotted together for comparison reasons. The comparison will be more self-explanatory when the bandwidth frequencies of each source are plotted in frequency domain as shown in Figure (2). As it is shown, the original desired speech bandwidth is interfered and overlapped with the low frequency components of the high frequency interference bandwidth. Moreover, depending on the type of the interference source, the high frequency components may be in higher frequency range compared with that of the speech and in such case, the effect of interference is less effective and verse versa.

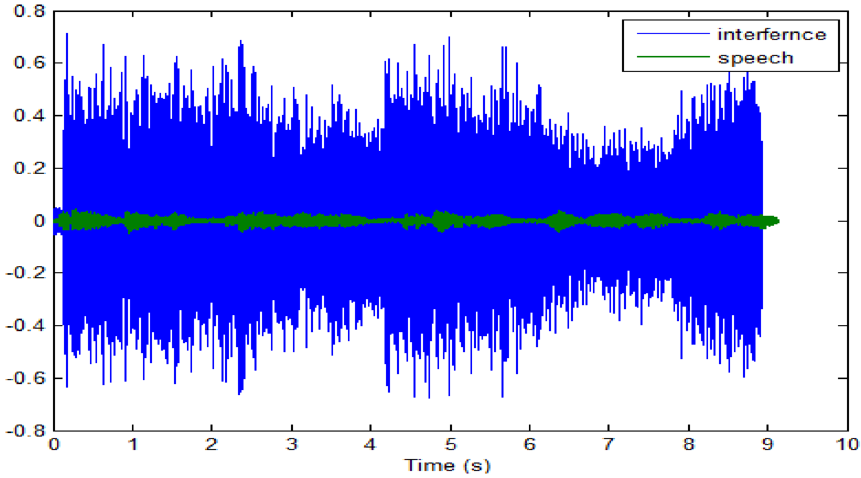


Figure (1) Time domain plot of both the speech and interference.

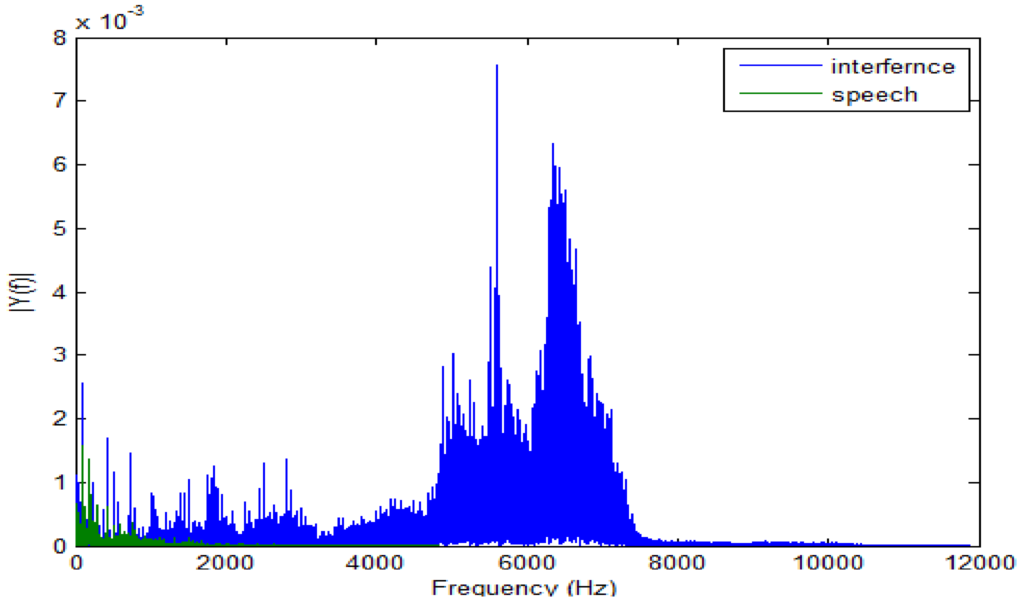


Figure (2) Frequency spectrum of both the speech and interference.

The proposed system is oriented for filtering the human speech bandwidth from the relatively high frequency interference components. The low pass IIR filter is chosen as the most suitable digital filter to be used for such filtering requirements due to unimportant phase needs and less complexity offered. The speech is usually within limited low frequency bandwidth of few kilo-Hertz which makes the low pass filter the nominated type. Butterworth is chosen to be applied which suggest a ripple-free pass-band region.

3.1 Butterworth low pass filter modeling

Butterworth or maximally-flat response. It exhibits a nearly flat pass-band with no ripple. [11]. Table (1) shows the coefficients for the denominators of normalized Butterworth filters of various orders where all of the coefficients correspond to a corner frequency of (1) radian/s [12]. For instant, to design filter coefficients with 2^{nd} order and cutoff frequency (F_C) of (2) KHz, the following procedure is to be followed;

The transfer function H(s) for the normalized coefficients shown previously in Table (1) is given by:

$$H_x(s) = \frac{1}{s^n + a_{n-1}\, s^{n-1} + a_{n-2}\, s^{n-2} + \ldots + 1} \qquad (5)$$

Table(1) Butterworth polynomials.

n	a_0	a_1	a_2	a_3	a_4	a_5	a_6	a_7	a_8	a_9
1	1									
2	1	1.414								
3	1	2.000	2.000							
4	1	2.613	3.414	2.613						
5	1	3.236	5.236	5.236	3.236					
6	1	3.864	7.464	9.142	7.464	3.864				
7	1	4.494	10.098	14.592	14.592	10.098	4.494			
8	1	5.126	13.137	21.846	25.688	21.846	13.137	5.126		
9	1	5.759	16.582	31.163	41.986	41.986	31.163	16.582	5.759	
10	1	6.392	20.432	42.802	64.882	74.233	64.882	42.802	20.432	6.392

Where (n) is the filter order . The de-normalization process suggests replacing each (S) factor in the filter transfer function by the corresponding cutoff angular frequency. Therefore replacing each (S) by $(s/2 * \pi * F_C)$;

$$H_x(s) = \frac{1}{\left(\frac{s}{\omega_C}\right)^2 + 1.414 * \left(\frac{s}{\omega_C}\right) + 1}$$

Where $\omega_C = 2\pi F_C$ and hence;

$$H_x(s) = \frac{\omega_C^2}{s^2 + 1.414\,\omega_C\, s + \omega_C^2}$$

Using bilinear transformation to converting $H_x(s)$ to $H_x(z)$.

$$H_x(z) = \frac{0.0143 + 0.0285z^{-1} + 0.0143z^{-2}}{1 - 1.6351\,z^{-1} + 0.6921z^{-2}} \qquad (6)$$

Using impulse transformation can be obtain;

$$H_x(z) = \frac{0.3586 * 10^{-2} z^{-1}}{0.0633 - 0.1035z^{-1} + 0.0437z^{-2}} \qquad (7)$$

3.2. High Frequency Interference Distortion HFID

Due to the nature of high frequency interfering in this work, it is required to suggest a filtering measurement criterion reflects the performance of the proposed system in filtering the undesired high frequency components affecting the speech intelligibility.

Accordingly, the following criterion is proposed to reflect such a measurement by dividing the summation of all squared high frequency components (undesired interfering components) by the summation of all squared low frequency ones which . are the desired speech bandwidth waveforms. That is;

$$HFID\% = \frac{\sum_{n>M}^{\infty} Y^2(n)}{\sum_{n=1}^{n\le M} Y^2(n)} \; x\ 100$$

Where (M) represents the harmonic index barrier between the low and high frequency spectrum regions while Y(n) is the magnitude of the frequency component for each value of the harmonic index (n). The value of (M) is assumed here corresponding to a 4KHz to ensure the maintaining of most of the human frequency range bandwidth and reject the high bandwidth which may interfere with the important speech bandwidth due to relatively high magnitude components.

4 Results and discussion

Simulating (in MATLAB) the system proposed is to input the speech and interference (circular saw) sounds for 10 seconds recording . simulation the digital filter using MATLAB programming with specified filter order and type, will be investigated the effect of order on the frequency response of the filter. will be designed the filters using coefficients Tables . The transformation will be through bilinear and impulse transformation in order to test parameters for those filters according to the percentage HFID criteria proposed.

4.1 Digital Filter Frequency Response Investigation

It is worth to investigate the effect of filter type and order on the frequency response of the filter itself. This will reveal the concept of choosing the good filter parameter for a certain case of interference. For instant, Figure (3) shows the effect of changing the order of the digital Butterworth low-pass filter from low order to high order.

The figure shows an enhanced frequency response for the higher Butterworth filter order which may be the choice for sharp filtering in case of near frequency interference interaction with the speech bandwidth. However, such a solution may reflect one important drawback which is the increase of the delay tabs. Such a drawback might be a challenge in some communication devices.

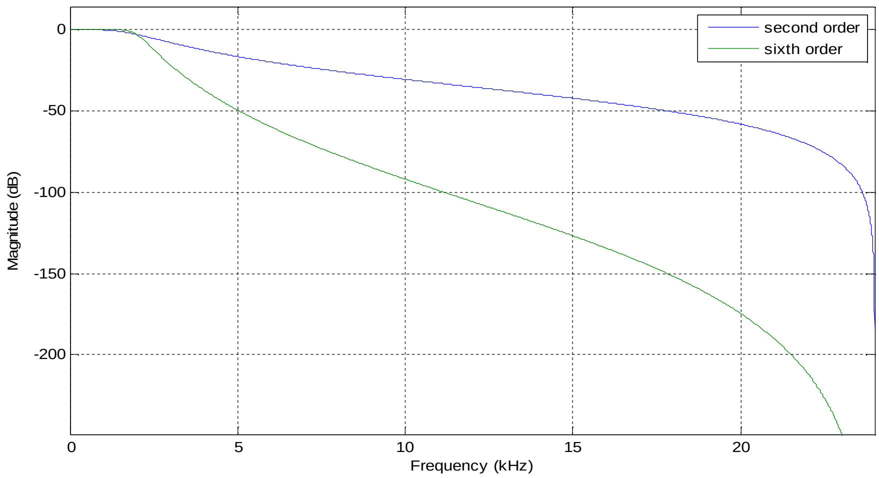


Figure (3) Magnitude frequency response of Butterworth low-pass filter for filter order 2nd and 6th

4.2 Digital Filter Simulation

The interference which is assumed to be generated from one of the environmental or machinery sources such as the circular saw or the angle grinder was also taken under investigation. For instant, the human speech for (10) seconds was recorder under the presence of operating Aluminum cutter in a certain workshop. The human speech was recognized to be highly affected by the interference voice and the frequency spectrum of the resultant recording under such interference is shown in Figure (4).

The spectrum shown in Figure (4) shows the presence of a dominant relatively high frequency components of the interference voice in such a way that the human speech is not totally overlapped and distorted.

The simulation of the digital filter is to be implemented using MATLAB programming with specified filter order and type. The implementation of the digital filter is to be applied to cases as that shown in Figure (4) where the human speech bandwidth is affected by relatively high frequency and amplitude interference components of frequency. Before introducing the full investigation of the filtering process to enhance the speech bandwidth, the program of building the digital filter coefficients using Bilinear and impulse invariance methods are introduced in Figure (5)

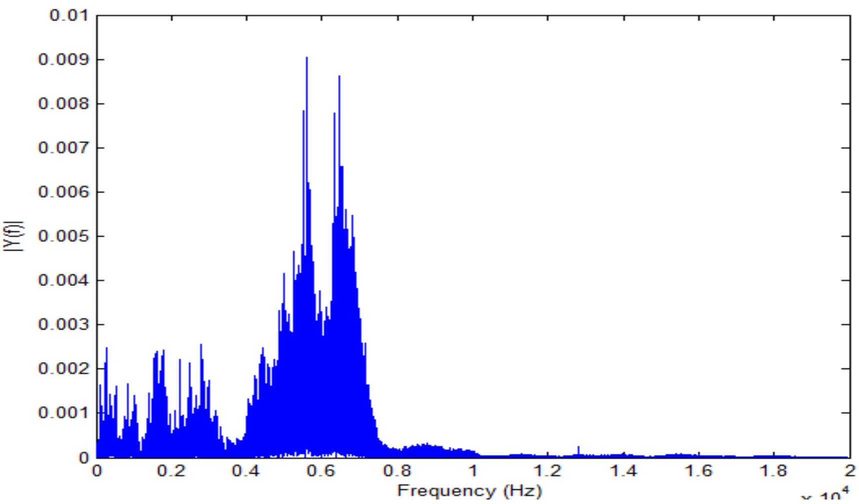
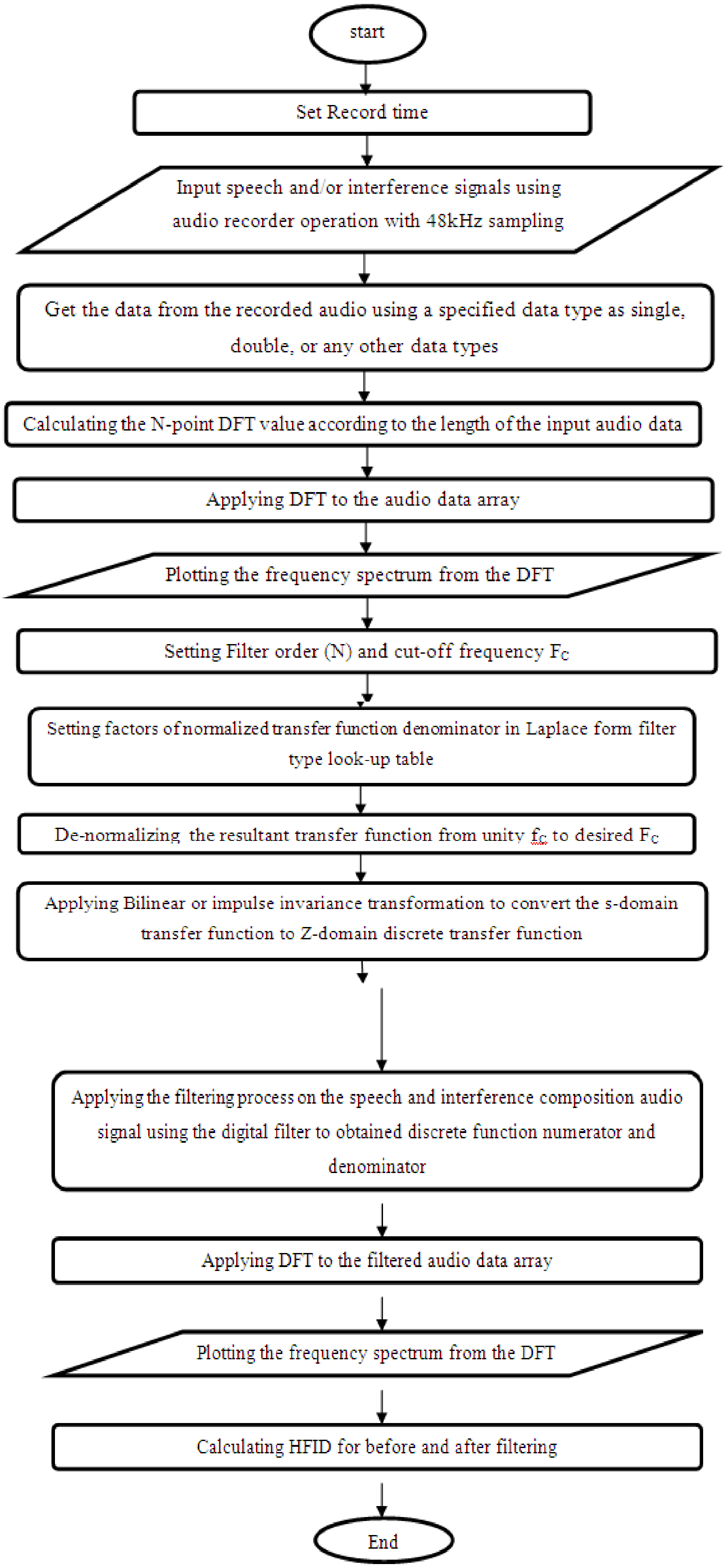


Figure (4) Frequency spectrum of a (10) seconds speech sample under interference condition.



a) **Figure (5)** Flowchart describing digital filter coefficients construction using Bilinear & impulse invariance methods.

Using the program of constructing a digital filter with flowchart shown in Figure (5) the type of filter was set as 6th order Butterworth filter with cut-off frequency 2KHz. However, applying the digital Butterworth filter to enhance the spectrum previously shown in Figure (4) which contains low frequency components (useful bandwidth) and high frequency components (interference) and the results of the filtering process is shown in Figure (6).

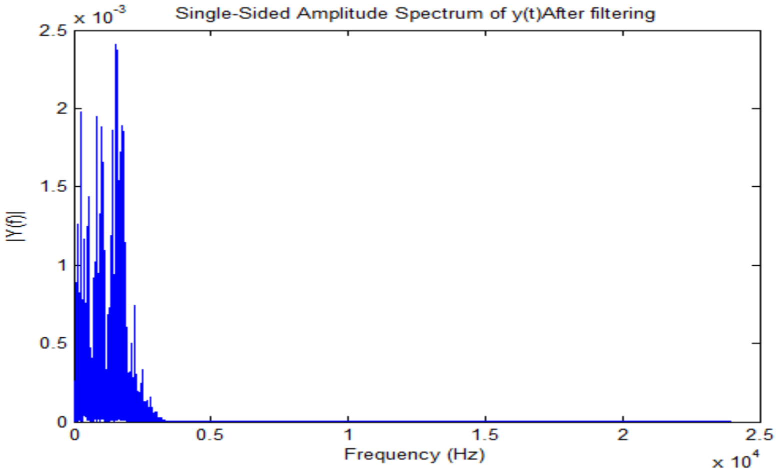


Figure (6) Frequency spectrum represents the filtering results of the speech and interference spectrum shown in Figure (4.3) using 6^{th} order Butterworth digital filter.

4.3 Filter Simulation Using Coefficients Tables

Figure (7) shows the effect of changing the cut-off frequency on the HFID% for the Butterworth low pass filter under circular saw (to Aluminum cutting) source of interference. The effect is measured for two methods of filter design; the Bilinear and the Impulse Invariance methods and for three values of the filter order as the 2nd, 4th, and 6th orders. All the design methods of filters considered in this simulation survey are based on tabular coefficients in S-domain. As expected, the HFID% is decreased with the increase of filter order due to increase in -3dB attenuation drop . Although the curves in Figure (7) reflects small difference between filter performance using Bilinear and Impulse Invariance methods, it is still worth to mention that the Bilinear method is better than the Impulse Invariance especially at low filter order values.

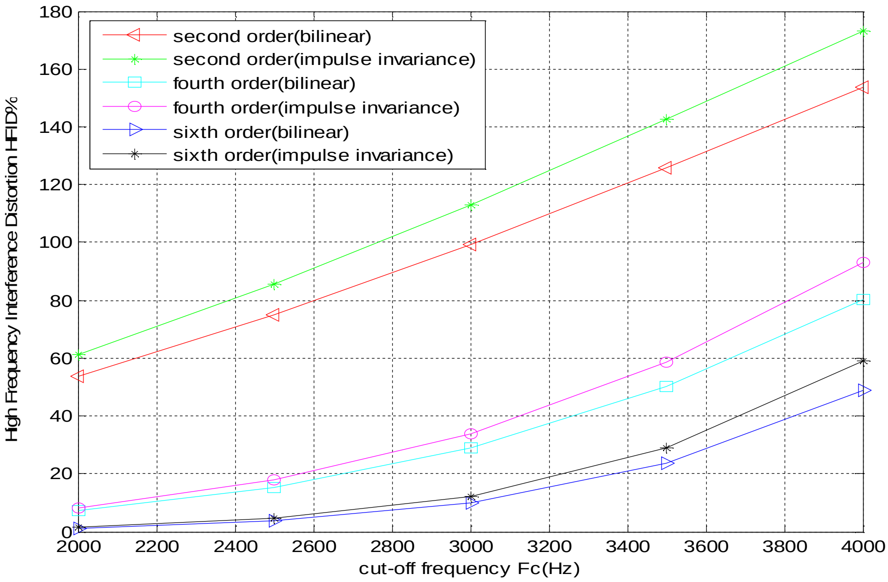


Figure (7) Effect of changing filter cut-off frequency on the HFID% of Butterworth digital filter under two design methods

5. Conclusions

As the aim of this paper is to introduce a simple method and procedure to enhance the human speech embedded in a high frequency interference voices, the simulation shows the investigation and achievement of such aim. This achievement was obvious using a live speech and interference samples and using the stated HFID% criterion which makes the filter application more feasible to be applied practically using a specific microcontroller especially that the minimum filter order does enhance the speech in acceptable manner.

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