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Abstract:
Voice signal is a sound signal generated by the dynamic activity of human vocal system. It is converted into electrical form by piezoelectric transducer such as microphone. Voice signal in electrical form can be used to pass message from its source to other remote locations through transmission lines and other technological facilities. Example is video conferencing technology where people can stay in different locations in various parts of the world and be contributing to a meeting holding outside their domains. Since this signal travels long distance along its path it is prone to degradation by noise such as Additive White Gaussian Noise (AWGN), Random Noise, power line noise, low frequency noise and high frequency noise, from its source to its destination. If these noise components are not removed, message in voice the signal may be lost or become full of errors and as such activities that rely on such message will become unreliable. In this paper, a height adjustable sine (HAS) window function is used to design a FIR filter for removing high frequency noise from voice signal in a matlab environment. The optimal parameters of the filter are sampling frequency of 44100Hz and filter order of 34. A real voice statement, “Education is the Key to the Development of any Nation” is converted into electrical signal using the system in-built microphone and recorded in windows media audio (.wma) format in one of the files of the system. The signal which in this case is a double column vectors is transferred to a matlab workspace using “audioread” instruction. A noise component of 4500Hz and above is generated with matlab and added to the voice signal to constitute a contaminated signal. When the contaminated signal is applied to the filter, result shows that the window is effective in removing high frequency noise from voice signals.

Keywords: HAS window, voice signal, high frequency noise, power spectral density.

1. INTRODUCTION

Noise components present in communication channels are disturbing and results in signal contamination. When the desired signal in the channel is a voice signal it may experience contamination by such noise components as random noise, power line noise, high frequency noise, additive white Gaussian noise (AWGN) and low frequency noise. Unless these noise components are removed the message in the voice signal is unreliable. In this paper removal of a high frequency noise from a voice signal so as to preserve the message in it is targeted. A voice signal has a frequency range of about 200Hz to 3400Hz and therefore any signal with a frequency of above the upper frequency of 3400Hz associated with it signal is termed a noise and must be removed.FIR filtering is one of the effective means of removing this high frequency noise from voice signal. As FIR filter a window is necessary to improve the smoothness of its impulse response to prevent distortion of the signal being filtered. Some researchers have used various windows to weight FIR filters for voice signal denoising. In [1] Supavit and Thaweesak used rectangular window to design low pass FIR filter for random noise reduction in speech signal with a sampling frequency of 10KHz, cutoff frequency of 5KHz and filter order of 133. The authors also examined the effectiveness of the filter when designed with different cutoff frequencies of 750Hz, 1.0KHz, 1.25KHz and 1.5KHz in terms of reducing random noise from speech signals and the result showed that the filter effectively removed the noise. In [2] Oka for and Mbachu designed a low pass FIR filter with Blackman-Harris window for removing high frequency noise from voice signal. The filter has a cutoff frequency of 3200Hz, sampling frequency of 16000Hz and order of 50. When a voice signal contaminated with frequencies above 3200Hz was applied to the filter result shows that it drastically reduced the noise. Akanksha eta l [3] made designs of FIR low pass and high pass filters with four different windows which are Han, hamming, Bartlett and Blackman windows to remove high and low frequency noises from speech. The order of the filter is 33, sampling frequency of 8000Hz, lower and upper cutoff frequencies of 0.25 and 0.5 normalised, respectively. According to the authors Blackman window illustrated superiority in performance and demonstrated best functionality among Hamming, Hanning and Blackman windows when their magnitudes response, phase response, equivalent noise bandwidth and response in time and frequency domain using matlab simulation were analysed. Saseendra and Rajesh [4] implemented a low pass filter with Kaiser Window technique, which is an adjustable window function as the bandwidth of the main lobe and side lobe amplitude can be varied by changing the value of its adjustment parameter Beta for fixed length of the window. The sampling frequency is 40800Hz, cutoff frequency of 10800Hz and filter order is 20 (21 filter taps) and three different values of Beta equal 0.5, 3.5 and 8.5 are applied in sequence to filter high frequency noise from audio signals. Simulation results shows that in FIR filter Beta=8.5 gives a better performance than other values of Beta. In [5] Pranab and Mohammad designed low pass, high pass and band pass
filters to eliminate both low and high frequency noise components contained in human voice, providing high quality voice. The windows used in the design are rectangular, triangular, Kaiser, Hamming, Hanning and Blackman windows. The sampling frequency value is 3400Hz and the voice signal is applied to ascertain the effectiveness of the filter. Secondly the cutoff frequency of the high pass filter is made to be 600Hz and the voice signal applied to it to ascertain its effectiveness and thirdly the lower and upper cut of frequencies of the band pass filter are made to be 600Hz and 3400Hz respectively and the voice signal applied to it to observe its filtration strength. These procedures are done for the different windows in the design. Results show that each of these filters eliminated the corresponding noise from the voice signal. Furthermore, the cutoff frequency of low pass, high pass and band pass were made variable in a DSP block to determine real time optimum cutoff frequencies. For the low pass the cutoff frequency was varied from 2.6KHz to 4KHz and the quality and the intensity of voice varies a little around 3400Hz. Similarly for a high pass filter the cutoff frequency was varied from 300Hz to 1000Hz and no significant change is observed around 600Hz. Variation of cutoff frequencies for the band pass gives a bandwidth that is between 600Hz and 3400Hz. The filters show sharp cutoff for removing high and low frequency components of noise from voice signal with non real time digital filters and real time digital filters. In [6] Ritesh and Rajesh designed a low pass FIR filter individually with hamming and hanning windows to remove high frequency noise from audio signals. Three different noisy signals were analysed and hanning window provided better results compared to hamming window. Babu et al [7] demonstrated how hamming, hanning and Blackman windows can be used to analyse speech signals. The authors designed low pass and high pass filters with the windows to remove high frequency noise and low frequency noise respectively from speech signals with a sampling rate of 22050, number of bits per sample of 16 and order of 64. The Blackman window out-performs among the three windows. Sangeetha and Kannan [8] in using multirate signal processing for speech signals designed low pass and high pass FIR filters by different windowing techniques such as hamming, hanning, Blackman, Rectangular and Kaiser Windowing. A voice signal of 8000Hz is recorded and stored as a wave file for use in a matlab. An additive white Gaussian noise (AWGN) is added to the speech to form a noisy speech signal. The noisy speech is filtered with the designed filters. Results show that each of the filters provided a good performance. The performance of these windows in processing voice signals are found to be satisfactory to a large extent but no researcher has used height adjustable sine (HAS) window to process voice signals. In this work therefore HAS window will be used to design FIR low pass filter for removing high frequency components from voice signals.

2. WINDOWING OF IMPULSE RESPONSE OF FILTERS

The windowing of impulse response of filters is the element by element multiplication of the desired impulse response h_d(n) (that is impulse response without window weighting) by a window function w(n) to obtain a modified or windowed impulse response h(n). That is, the windowed impulse response is a dot product of h_d(n) and w(n) as in (1) [9].

h(n) = h_d(n).w(n)  (1)

The desired impulse response h_d(n) is impulse response obtained from the sudden truncation of infinite impulse response sequence of FIR filter. The essence of the window application is to make the truncation gradual so as to prevent Gibbs oscillation that arises from the sudden truncation which manifests in form of distortion of complex or compound signals when such signals are applied to FIR filters designed with un-windowed FIR filter.

3. HAS WINDOW

The HAS window which is a variable sine window function is presented as (2) below.

\[
w(n) = \begin{cases} 
\alpha + \sin \left( \frac{2 \sin^{-1}(1-\alpha)}{L} n \right) & 0 \leq n \leq \frac{M-1}{2} \\
\alpha + \sin \left( \frac{(L-n)2 \sin^{-1}(1-\alpha)}{L} n \right) & \frac{M-1}{2} \leq n \leq M-1
\end{cases}
\]  (2)

where \(\alpha\) varies from 0 to 1. \(\alpha\) varies the amplitude or height of the window for a fixed length of the window. M is the length of the window and L=M-1.

3. DESIGN OF LOW PASS DIGITAL FIR FILTER WITH HAS WINDOW

In this design HAS window function is used on FIR filter. The function is depicted in fig. 1. Using the window on a FIR filter of order 34 implies that the corresponding window length is M=35 and for such length the HAS window of (2) becomes as in (3) below.

\[
w(n) = \begin{cases} 
\alpha + \sin \left( \frac{2 \sin^{-1}(1-\alpha)}{34} n \right) & 0 \leq n \leq 17 \\
\alpha + \sin \left( \frac{(34-n)2 \sin^{-1}(1-\alpha)}{34} n \right) & 17 \leq n \leq 34
\end{cases}
\]  (3)

Figure.1. Amplitude Response of HAS Window

With specification of filter order as 34, cutoff frequency as 3200Hz and sampling frequency as 44100Hz, six different values of height adjustment parameter \(\alpha\) = (0.00, 0.01, 0.02, 0.03 and 0.04) are considered and in each value the impulse, magnitude and phase responses of the filter are obtained and are depicted below. Note that at \(\alpha = 0.0\), the function becomes a complete sine window function. The sampling frequency value
chosen is because the voice to be used is recorded in windows media audio (.wma) format. The impulse, magnitude and phase responses of the filter at different values of $\alpha$ are shown below.

### 3.1. Responses When $\alpha=0.00$

![Figure 2a. Impulse Response When $\alpha=0.00$](image1)

![Figure 2b. Magnitude Response When $\alpha=0.00$](image2)

![Figure 2c. Phase Response When $\alpha=0.00$](image3)

### 3.2. Responses When $\alpha=0.01$

![Figure 2d. Impulse Response When $\alpha=0.01$](image4)

![Figure 2e. Magnitude Response When $\alpha=0.01$](image5)

![Figure 2f. Phase Response When $\alpha=0.01$](image6)
3.3. Responses When $\alpha=0.02$

Figure 2g. Impulse Response When $\alpha=0.02$

Figure 2h. Magnitude Response When $\alpha=0.02$

Figure 2i. Phase Response When $\alpha=0.02$

3.4. Responses When $\alpha=0.03$

Figure 2j. Impulse Response When $\alpha=0.03$

Figure 2k. Magnitude Response When $\alpha=0.03$

Figure 2l. Phase Response When $\alpha=0.03$
Analysing the responses above it can be seen that all the impulse and magnitude responses exhibit stability in that there are no sustained oscillations but the phase responses for exhibit linearity. However the phase response when $\alpha=0.00$, 0.01, 0.02 and 0.03 indicate some degree of non-linearity whereas for...
α=0.04 and 0.05 the phase responses are linear. It can therefore be said that the optimum value of α in this circumstance is 0.04 because the impulse and magnitude responses are stable and the phase response is very linear.

4. RESULTS

In order to ascertain the quality of the designed filter a voice statement “Education is the Key to the Development of any Nation” is converted into electrical voice signal using the system in-built microphone and recorded in windows media audio (.wma) format and stored in one of the files of the system. The signal is transferred to a matlab environment using “audioread” instruction after which it is made noisy by adding sine wave of 4500Hz and above it as noise. The voice signal is shown in fig.3 while the fig.4 depicts the contaminating noise, and the noisy voice signal depicted in fig.5. The noisy voice signal is filtered with each of the designed low pass filters and the output recorded. Figures 6, 7, 8, 9, 10 and 11 show the voice signal after filtering. Comparing the clean voice signal of fig.3, the noisy voice signal of fig.5 and the filtered voice signals of fig6 to fig.11 it can be seen that the filter largely removed the noise contained in the noisy voice signal. Listening to the noisy voice signal and the filtered signals also prove the effectiveness of the filter.
4.1. Signal Power Level

The performance of the filters can be analysed by considering the power levels of the filtered signals [10, 11]. Fig. 12 is the spectral density of a clean voice signal while the spectral density of the noisy voice signal is depicted in fig. 13. Fig.14 to fig.18 are the power spectral densities of the filtered voice signal at different values of $\alpha$. Choosing a normalised frequency of 0.875 for the analysis, table 1 below shows the summary of the power levels in dB of the filtered signals at different values of $\alpha$. 

![Figure 8: Voice Signal Filtered With Low Pass Filter When $\alpha=0.02$](image_url)

![Figure 9: Voice Signal Filtered With Low Pass Filter When $\alpha=0.03$](image_url)

![Figure 10: Voice Signal Filtered With Low Pass Filter When $\alpha=0.04$](image_url)

![Figure 11: Voice Signal Filtered With Low Pass Filter When $\alpha=0.05$](image_url)

![Figure 12: Power Spectral Density of Noise Free Voice Signal](image_url)

![Figure 13: Power Spectral Density of Unfiltered Voice Signal](image_url)
From table 1, power level of the contaminated voice signal is +27.37dB. Comparing it with the clean voice signal power level of -91.20dB which can be deduced from fig. 12 shows that noise component added a lot of noise power of 27.37-(-91.20)
=118.57dB to the voice signal. It can also be seen from table1 that the signal power decreases as $\alpha$ increases. Recall that in the impulse, magnitude and phase responses, the optimum value is $\alpha =0.04$ because at this value the phase response is very linear. The corresponding value of power level at this value of 0.04 is -27.79dB.

<table>
<thead>
<tr>
<th>Power level of unfiltered voice</th>
<th>+27.37dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>Power level of filtered voice when $\alpha=0.0$</td>
<td>-33.8dB</td>
</tr>
<tr>
<td>Power level of filtered voice when $\alpha=0.01$</td>
<td>-28.81dB</td>
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<tr>
<td>Power level of filtered voice when $\alpha=0.02$</td>
<td>-28.10dB</td>
</tr>
<tr>
<td>Power level of filtered voice when $\alpha=0.03$</td>
<td>-27.88dB</td>
</tr>
<tr>
<td>Power level of filtered voice when $\alpha=0.04$</td>
<td>-27.79dB</td>
</tr>
<tr>
<td>Power level of filtered voice when $\alpha=0.05$</td>
<td>-27.70dB</td>
</tr>
</tbody>
</table>

5. CONCLUSION

It can be concluded that HAS window can to some extent be used to design FIR filters for voice signal filtration but it doesn’t seem to be a very efficient window for the purpose considering the qualities of the power spectral densities of the filtered signals when compared with that of clean voice signal. For the voice signal, the noise type and level in this circumstance, the optimum value of $\alpha$ is 0.04. This value may vary if a different type of signal, noise type, filter length or sampling frequency is under consideration.

6. REFERENCES


