Speech Enhancement Techniques for Hearing Impaired People: Digital Signal Processing based Approach

Sami Mohammad Halawani[1], Abdul Rahman Al-Talhi[2], and Abdul Waheed Khan[1]

[1]Faculty of Computing and Information Technology Rabigh, King Abdulaziz University, Rabigh, Saudi Arabia
[2]Faculty of Computing and Information Technology, King Abdulaziz University, Jeddah, Saudi Arabia
awk_career@yahoo.co.uk

Abstract: A great proportion of human population suffers from hearing loss. Hearing loss is a measure of shift in auditory system compared to that of a normal ear for detection of a pure tone. It is very difficult to imitate the behaviour of human auditory system in its entirety and thus compensate for the hearing loss. However, with the availability of modern day technologies and the recent developments in signal processing area, sophisticated artificial hearing aid systems can be designed that relax the job of damaged auditory systems to a great extent and make much of the sound available to the hearing impaired. In pursuit of designing an artificial hearing aid, human auditory system is the best model to start with. Most hearing aids work well in noise-free environments but give poor performances in noisy environments. However with the availability of a variety of noise reduction algorithms, the background noise could be reduced to a great extent. This paper first presents characteristics of human auditory system, which undoubtedly serves as a prototype for the design of an artificial hearing aid. Then various algorithms for noise reduction, frequency-dependent amplification and amplitude compression are presented along with their Matlab simulations. Finally, future trends and expected innovations in the hearing aid industry are discussed.
[Halawani SM, Al-Talhi AR, Khan AW. **Digital Signal Processing Based Speech Enhancement Techniques for Hearing Impaired People.** *Life Sci J* 2013;10(4):3467-3476]. (ISSN:1097-8135). http://www.lifesciencesite.com. 462

Keywords: Hearing Loss, Speech Enhancement Techniques, Noise Cancellation, Speech Amplification, Amplitude Compression

1. Introduction

The introduction of the first digital hearing aid in 1996 has revolutionized the hearing aid industry [1]. Prior to that majority of hearing aids were analogue and there was only a negligible proportion of potential wearers just because of limited relief that those analog hearing aids used to provide. Analog hearing aids are not much different from linear amplifiers and generally do not provide any noise-cancellation mechanism. Whereas digital hearing aids contain a very advanced degree of signal processing that offer significant environmental noise reduction. The analog hearing aids offer a generalized solution to the hearing impairment [2] while we know that everyone hearing characteristics are unique and therefore there should be specialized solutions according to the hearing impairment of the individuals. The digital circuits are more flexible than analog circuits. They can be precisely programmed to match the patient's individual hearing loss, sometimes at each specific frequency/pitch. This signifies the use of human audiogram. To compensate for the frequency dependent hearing loss, hearing aids can be fit to comply with an individual's audiogram so that different gains are applied to different frequency bands. Digital hearing aids can be operated with very less battery power, usually in mW [1]. In short, digital hearing aids offer improved clarity of sound, less circuit noise, faster processing of sound, and improved listening in noise when compared to analog circuits.

2. Normal Hearing Characteristics

The dynamic range of hearing of a normal person is 20 Hz to 20,000 Hz. Human ear also acts like a filter and favors certain frequencies over the others [3]. Human ear is most sensitive to sounds in the range of 1,000 Hz to 5,000 Hz and particularly at about 4000 Hz. The sensations of these frequencies are commonly referred to as the 'pitch' of a sound. The second important property of sound wave is the 'sound intensity or loudness'. Since the range of intensities, which the human ear can detect, is so large, hence the scale frequently used by audiologists to measure intensity, is a compressed scale called a logarithmic scale. The scale for measuring intensity is the decibel scale. Hence, in terms of decibel, the audible sound pressure range is from 0dB – 120dB, as shown in the Figure 1.

Figure 1. Audible Range of Normal Human Ear

3. Classification of Hearing Loss

Hearing loss is generally classified as mild, moderate, severe, or profound. The quietest sounds or softest intensity levels of sounds that can be perceived by people suffering from different hearing losses are summarized in Table 1:

Table1. Hearing Thresholds associated with various categories of hearing losses

Category	Softest Intensity Level
Mild Loss	25 – 40 dB
Moderate Loss	40 – 70 dB
Severe Loss	70 – 95 dB
Profound	95 dB or more

4. Structure of Digital Hearing Aids

Figure 2 shows the block diagram for the Matlab implementation of a digital hearing aid. The input speech signal after digitization passes through several stages: First of all, the digitized input speech passes through a Noise Reduction System to suppress any noise if contaminated with the speech. Then the filtered input speech signal passes through Frequency Shaping System which modifies the spectral content of the speech according to the listening convenience of the hearing impaired. Finally the amplified speech passes through Amplitude Compression System to ensure the overall gain of the amplified speech according to the listening comfort of the hearing impaired, before producing an adjusted output speech.

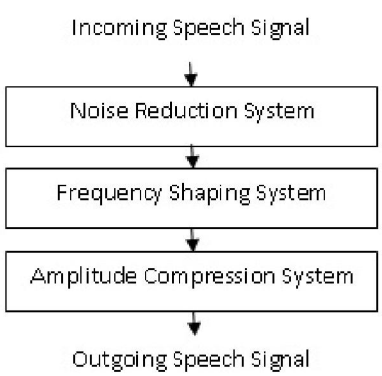

Figure 2. Block Diagram of Digital Hearing Aid

5. Noise Cancellation

The most common problem in speech processing is the effect of interference noise in speech signals. The interfering noise generally masks the speech signals because sometimes they fall in the same frequency range and therefore degrades the intelligibility and quality of speech. Speech intelligibility is greatly reduced by background noise and the greater the level of background noise the greater the reduction in the intelligibility of speech.

Before digging into the details of various noises cancellation techniques, it is worthwhile to briefly describe speech and noise signals.

5.1. Speech

Speech is a very basic way for humans to convey information to one another. Speech has certain properties such as; it is a one-dimensional signal, with time as its independent variable. It is random in nature. It is non-stationary, i.e., the frequency spectrum does not remain constant with time.

5.2. Noise

Noise is any unwanted signal that interferes with the desired signal. In the context of speech processing, speech is the signal of primary interest and there are three types of noises that have direct impacts on the speech intelligibility [4]:

- Random noise with Power-Density Spectrum similar to that of speech.
- Competing speaker(s) noise.
- Room Reverberation.

The more we know about the speech and noise, the more we can do to reduce the effects of noise on the speech.

5.3. Classification of Noise Cancellation Techniques

Most digital hearing aids rely on single-channel systems. Since only a single recording is available and we don’t have an explicit access to the noise so the performance of the noise suppression system greatly depends on the accuracy of the background noise estimate. Speech enhancement techniques must estimate noise characteristics during the non-speech periods when only background noise is present. Therefore an effective and robust voice activity detector (VAD) plays a key role in the single-channel noise-reduction systems.

5.3.1. Voice Activity Detection Algorithms

The process of separating conversational speech and silence is called voice activity detection. VAD algorithm extracts some measured features or quantities from the input signal and compares these values with thresholds, usually extracted from the characteristics of the noise and speech signals [5]. Then, voice-active decision is made if the measured values exceed the thresholds. VAD in non-stationary noise requires a time-varying threshold value.

According to [5], among the various VAD algorithms, following two are the most commonly used:

- Energy and Zero-Crossing Rate (EZCR) Based VAD
- Statistical Model Based VAD

5.3.2. Frequency Domain Noise Cancellation Techniques

There are two ways of analyzing a signal in frequency domain: The first involves the use of various digital filters i.e., lowpass, highpass, bandpass, bandstop etc. The second involves the use of Fourier analysis where the time domain signal is transformed into frequency domain using Fourier series (practically implemented using Fast Fourier Transform-FFT).

a. Noise Cancellation using Fixed Filters

This is the simplest technique to handle the problem of noise elimination/reduction. This method requires prior knowledge of the frequency spectra of noise and is only fruitful when the nature of the noise is stationary and deterministic. As an example, we know that noise components in the frequency region below 1 KHz (especially below 0. 4 KHz) are the most intense. Based on this knowledge, a high-pass filter with a cut-off frequency of 1 KHz can be designed that will attenuate all signals below 1 KHZ and will pass all the signals above 1 KHz without attenuation. We observe that the signal in the area below 0.4KHz is mostly noise as shown in Figure 3, and eliminating these components has the desired effect of reducing the loudness of the noise and improving overall sound quality. However, the high-pass filter eliminates both speech and noise in the frequencies region between 0. 4 KHz and 1 KHz. In this region, the frequency spectrum of the speech is slightly above that of the noise and so a small contribution to the intelligibility of speech is lost.

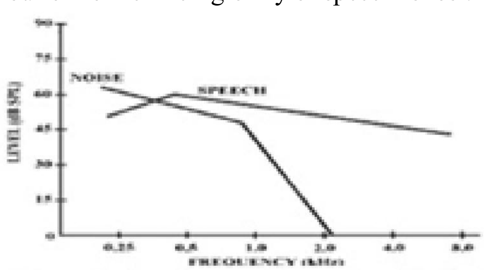

Figure 3. Spectrum of Noisy Speech showing intensities of speech & noise at different frequency bands.

b. Matlab Implementation & Results

We used filters with different specifications to remove different types of noises. We first implemented a notch filter having a notch frequency of 2500 Hz, to remove beep noise. We used an IIR notch filter to give a sharp shape to the notching area and reduce the effect of notching other frequencies. Figure 4, shows the result of notching the beep frequency.

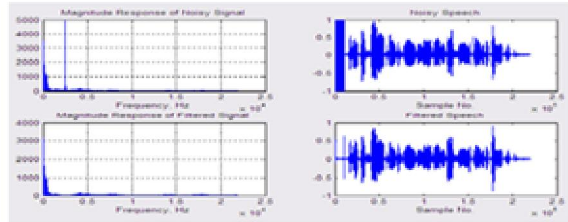

Figure 4. Removal of Beep Noise from Speech via Fixed Filters Method

Next we implemented a bandstop filter to get rid of siren noise. The bandstop region was chosen as 500-2500 Hz, because of the concentration of siren noise in this region. We used 'butter' filter, which is an IIR filter. We set the filter order as 3. Under these settings, we obtained the results as shown in the Figure 5.

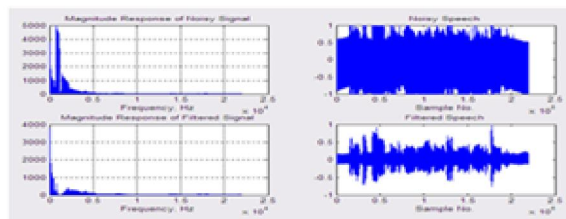

Figure 5. Removal of Siren Noise from Speech via Fixed Filters Method

We observed that the loudness of the siren noise is decreased significantly at the expense of some loss of intelligibility of the speech. This is because; some of the speech frequencies have also been attenuated.

Finally, we tried to remove low frequencies noise from speech by applying a highpass filter. Since, in our example, noise was most concentrated in the frequency range below 2000 Hz, so we used a highpass filter with a cutoff frequency of 2000 Hz. We used an IIR highpass filter of the order 3. Under these settings, we obtained the results shown in the Figure 6. Again here we have removed much of the noise at the expanse of some reduction in the loudness of the speech.

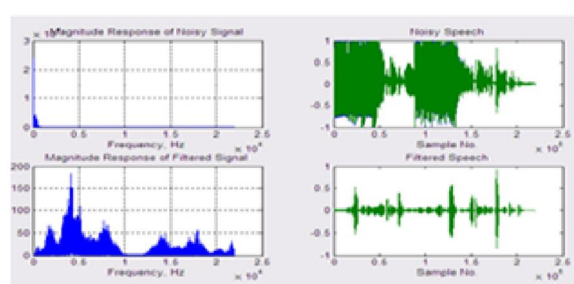

Figure 6. Removal of Low Frequency Noise from Speech via Fixed Filters Method

c. Spectral Subtraction Technique

Spectral Subtraction is one of the earliest and longest standing approaches to noise suppression and was developed by Boll in [6]. This method tries to recover speech signal observed in additive noise through subtraction of an estimate of the noise spectrum obtained during non-speech activity from the noisy signal spectrum. Spectral subtraction is based on the assumption that the noise is a stationary or slowly varying process and that the noise spectrum does not change significantly in-between the update periods. The block diagram of this technique is shown in Figure 7.

Let the speech signal has $s(n)$ been degraded by the uncorrelated additive signal $v(n)$, then the corrupted noisy signal can be expressed as:

$$x(n) = s(n) + v(n) \tag{1}$$

Taking the Discrete Fourier Transform (DFT) of $x(n)$ gives:

$$X(k) = S(k) + V(k) \tag{2}$$

Assuming that is zero-mean and uncorrelated with, the $v(n)$ $s(n)$ estimate of can be $|S(k)|$ expressed as:

$$\left|\hat{S}(k)\right| = |X(k)| - E|V(k)| \tag{3}$$

Given $|\hat{S}(k)|$ the estimate, the speech can be expressed as:

$$\left|\hat{S}(k)\right| = \left|\hat{S}(k)\right| e^{j\theta_x(k)} \tag{4}$$

$$e^{j\theta_x(k)} = \frac{X(k)}{|X(k)|} \tag{5}$$

Where $\theta_x(k)$ is the phase of measured noisy signal. As determined by [7] that for all practical purposes, due to computational complexity of phase of clean speech, it is sufficient to use the noisy speech phase $\theta_x(k)$.

d. Enhancements to the Basic Spectral Subtraction Technique

From equation (3) it is observed that the estimated speech magnitude spectrum is not guaranteed to be positive. A number of techniques focusing on reduction of auditory effect of spectral error have been developed. These methods are spectral magnitude averaging, half-wave rectification, and residual noise reduction [8]. A detailed diagram for spectral subtraction algorithm is illustrated in Figure 8.

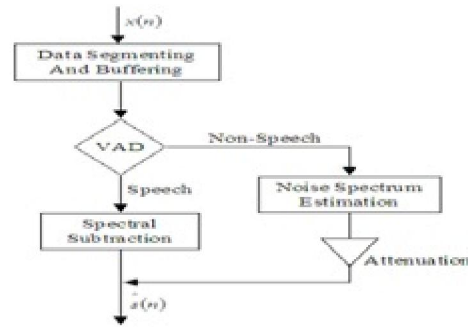

Figure 7. Block Diagram of Spectral Subtraction Technique

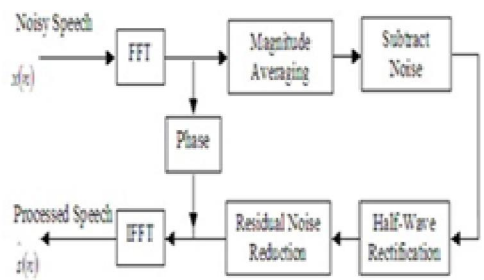

Figure 8. Detailed Diagram of Spectral Subtraction Technique

e. Limitations of Basic Spectral Subtraction Technique

The main limitation of spectral subtraction technique is the assumption that noise is a stationary or slowly varying process otherwise more frequent estimations of noise spectrum would be required which makes the job difficult for most DSPs. Another artifact is phase distortion, caused by the assumption that the ear is insensitive to the phase.

f. Simulating Spectral Subtraction Technique in Matlab

The Graphical User Interface demonstrating the results both in time domain and frequency domain is given as in the Figure 9. To obtain the noise estimate, we assumed the first frame i.e., 1000 samples as pure noise.

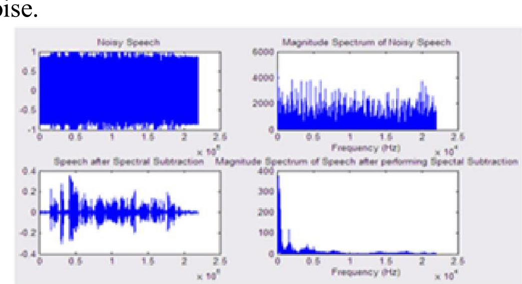

Figure 9. Noise Cancellation using Spectral Subtraction Technique

We also checked the performance of spectral subtraction technique at different signal-to-noise

(SNR) ratios. We observed that this technique even works at 0 dB as shown in Figure 10, but not below 0 dB.

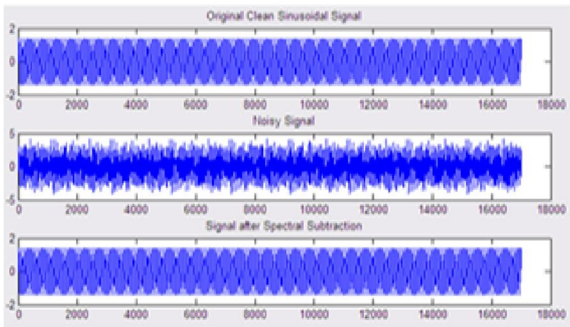

Figure 10. Result of Spectral Subtraction at 0dB SNR tested on Sinusoidal Signal

5.3.3. Adaptive Noise Cancellation (ANC) Technique

Usually the background noise does not keep steady and it changes from time to time, so the noise cancellation must be an adaptive process i.e., it should be able to work under changing conditions, and be able to adjust itself according to the changing environment.

The adaptive noise canceling refers to a class of adaptive enhancement algorithms based on the availability of a primary input source and a secondary reference/auxiliary source. In the very basic model, adaptive noise cancellation system processes signals from two sensors and reduces the level of the undesired noise with adaptive filtering techniques, as depicted in the Figure 11.

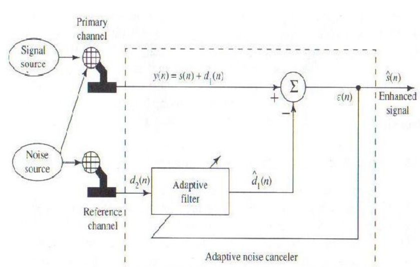

Figure 11. Block Diagram of Adaptive Noise Cancellation

The adaptive noise canceller consists of an adaptive filter that acts on the reference signal to produce an estimate of the noise, which is then subtracted from the primary input. The overall output of the canceller is used to control any adjustments made to the coefficients of the adaptive filter. The criterion for adjusting these weights is usually to minimize the mean-square-error signal $E\left[e^2(n)\right]$, which is strongly dependent on the correlation between noise in the primary input and secondary input. The minimization of $E\left[e^2(n)\right]$ can be achieved by updating the taps/coefficients of the adaptive filter i.e., $b_i(n)$. The two most commonly used algorithms for this purpose are Recursive-Least-Square (RLS) and Least-Mean-Square (LMS). The LMS algorithm is relatively simple to implement and needs fewer computations that is why it is preferred.

The steps involved in evaluation of LMS algorithm are given as follows:

Step 1: Assume the order of the filter (M). Initialize the coefficients $B(k)$ for $k = 0,1,2,\ldots\ldots,M-1$ to zeros.

Step 2: Compute the filter output $\hat{d}_1(n)$, using:

$$\hat{d}_1(n) = \sum_{k=0}^{M-1} b(k) d_2(n-k) \tag{6}$$

Step 3: Compute the error signal, $e(n)$, using:

$$e(n) = y(n) - \hat{d}_1(n) \tag{7}$$

Step 4: Update the filter coefficients using:

$$B(k+1) = B(k) - 2\mu e(n) D2_n \tag{8}$$

Step 5: Repeat from Step 2 sample by sample.

a. Single Channel Adaptive Noise Cancellation

Two different approaches could be used to deal with noise in single channel adaptive noise cancellation systems:

I. Delay Based Approach

According to this approach, the reference signal can be obtained by applying a certain delay to the primary input signal and then the adaptive noise canceller should process on this delayed version of the available measurement i.e., $y(n-T)$, where T is the amount of delay. The block diagram of such is shown in Figure 12.

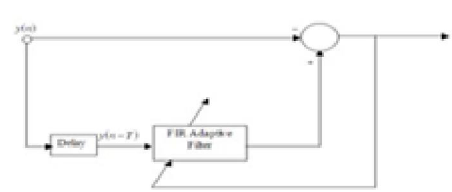

Figure 12. Delay Based Model for Single Channel Adaptive Noise Cancellationn

According to [9], extracting a noise reference from the input signal has some disadvantages such as, possible non-stationarity of the noise and silent/noise decision is not error-free.

II. Sambur Approach

Although it may be very difficult to construct a noise reference channel, it is not difficult to obtain a speech reference channel for some classes of speech. According to [10], due to the quasi-periodic nature of speech during voiced sections, a reference signal can be constructed by delaying the primary data by one or two pitch periods. This reference signal can then be used in the LMS adaptive algorithm. A block diagram of the enhancement system proposed by [10] is shown in the Figure 13.

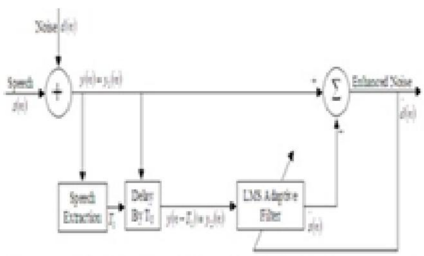

Figure 13. Sambur Model of Single Channel ANC for Removing Speech from Noisy Speech

Sambur in [10] applied the above mentioned approach for additive white noise in the SNR range 0-10 dB. In his work, he obtained improved SNR as the LMS filter length M was increased from 6 to 14. However, the LMS adaptive filter removed some of the "granular" quality of the quantized speech.

The main limitation of Sambur's approach is the requirement of accurate pitch estimation. There are several pitch extraction methods in the literature but almost all of them give very poor performance when extracting pitch of a degraded speech.

b. Limitations of Adaptive Noise Cancellation Techniques

Hearing aids using adaptive noise cancellation are still at an early stage of development [4]. Adaptive noise cancellation requires at least two microphones and, under ideal conditions, at least one microphone must be placed at the noise source. This is not very practical for a person wearing a hearing aid. However, even then we cannot ensure that there will be no crosstalk effect.

c. Simulation of Delay Based Single Channel Adaptive Noise Cancellation

To obtain a reference noise, we delayed the primary input by 10 times. We experimented with different values of step-size 'μ' and filter-order and finally found reasonably good results by setting the step-size 'μ' as 0. 005 and the filter order as 10, as shown in Figure 14.

Since we did not have an explicit access to noise source, but we obtained it by delaying the primary input, that is why little bit distortions can be observed.

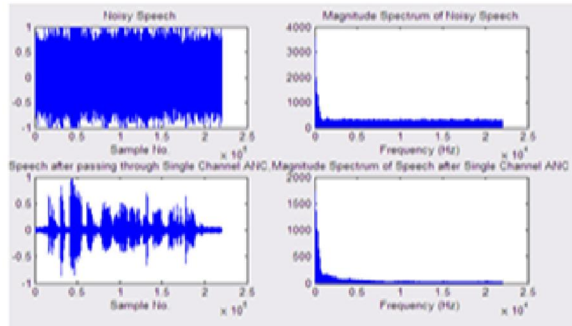

Figure 14. Delayed Based Single Channel ANC

d. Simulation of Dual Channel Adaptive Noise Cancellation

In this part of simulation, we considered an explicit noise signal strongly correlated with the noise in the noisy speech signal. We set the filter order as 10 and the step-size 'μ' as 0. 01. Under these settings, we obtained very good results, as shown in the Figure 15.

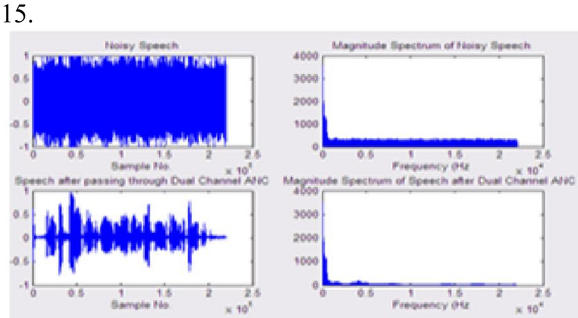

Figure 15. Dual Channel ANC

5.3.4. Signal-to-Noise (SNR) Ratio Based Noise Suppression

Speech understanding is not the only problem experienced by hearing aid users when listening in background noise. Two other factors that need to be considered along with the speech quality are comfort and listening effort. For example, think about a hearing aid user who listens to speech in a noisy environment over a long period of time. The concentration required to follow what is being said can be an exhausting task. However, less effort would be required by the hearing aid user if the background is less noisy.

In the presence of background noise, the primary speaker's speech is the signal of interest and all other signals are considered as noise. How do we differentiate between the primary speaker and the background noise? One feature is the energy difference between signals coming from different sources or signal-to-noise ratio i.e., the primary speaker's speech is louder than others. Hence a noise suppression scheme could be devised which takes into account the average energy of the signal. The idea is to first calculate the average energy of each frame of the input signal and then check the energy of the incoming sample against the product of the average energy of the frame and SNR value, i.e.,

Check $E_i < Average\_E * SNR$

If the energy of the incoming sample is less than that product, it means it is background noise, and several solutions can be adopted to suppress noise level:

One solution is to zero out the frequency contents at those samples, i.e.,

$$E_i = 0 \quad (9)$$

However, this brings about some unnatural pauses and distortions in the speech.

Since a small amount of noise improves the output speech quality, this could be implemented by using a software constraint, like

$$E_i = 0.005 * E_i \quad (10)$$

Experimental work shows better results using this approach.

A provision of this method in hearing aid can be made by giving the user some control button to set different SNR values. So that whenever the user confronts an environment where there is background noise, he/she can set the control button to a certain value according to his/her listening convenience, to instruct the hearing aid to consider the signal below a certain level as noise and therefore suppress it.

I. Matlab Implementation & Results

We implemented this method of noise suppression in frequency domain. Although this could be implemented in time domain but the advantage of frequency domain is to allow other frequency domain operations, such as frequency shaping and amplitude compression, in tandem with noise reduction. We used the expression

$$E_i = 0.005 * E_i \quad (11)$$

for noise suppression. We checked the performance of the proposed algorithm at different SNR values. Simulation results demonstrated, the more high the SNR value is, the better the results are, in terms of clarity of speech. Figures 16 and 17, 18, show the results at SNR at 2, 3 and 4 respectively.

6. Frequency Shaping

People with hearing loss often have different levels of loss at different frequencies, which leads to sounds being perceived as distorted relative to a normal-hearing individual because the various frequency components are not weighted in the expected manner [11]. Thus, applying a uniform gain does not generally correct for the hearing loss. Therefore, instead of amplifying the entire incoming signal, we need to enhance speech only in the hard-to-hear frequency bands for a particular hearing impaired. Research shows that most hearing impaired people have difficulties to hear high frequency signal, therefore, the frequency shaper needs to apply high gain at those frequencies to correct for the loss of hearing.

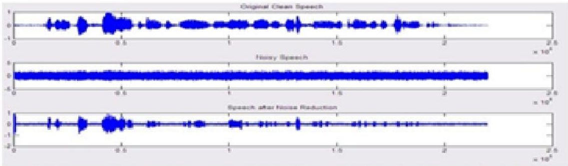

Figure 16. Results of noise suppression at SNR of 2 via SNR based Noise Suppression Method

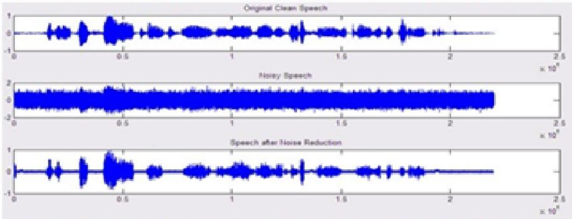

Figure 17. Results of noise suppression at SNR of 3 via SNR based Noise Suppression Method

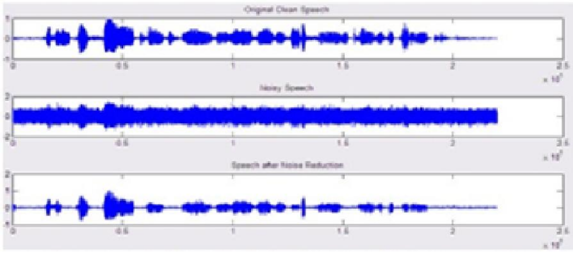

Figure 18. Results of noise suppression at SNR of 4 via SNR based Noise Suppression Method

A frequency shaper provides a natural decomposition of the input signal into frequency bands as shown in Figure 19, which may be processed independently to best compensate for the hearing loss and meet prescriptive targets as identified by the audiologist.

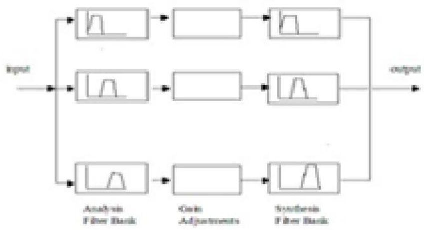

Figure 19. Block Diagram of Frequency Shaper

The frequency shaper can be realized in two ways:

- Digital Filter Bank Approach
- FFT Based Approach

The frequency shaper takes into account the audiogram of an individual hearing impaired. To compensate for the frequency dependent hearing loss, hearing aids can be fit to comply with an individual's audiogram so that different gains are applied at different frequency bands.

6.1. Test Case

As test case, we consider the audiogram of a person who has moderate hearing loss, also known as Ski-Slope hearing loss, as shown in Figure 20.

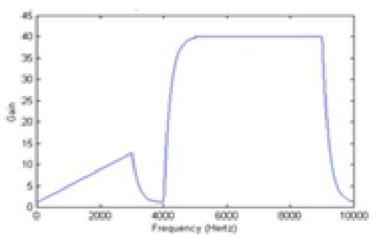
Figure 20. Transfer Function of a person having moderate hearing loss

From this audiogram, we observe the following information:

- A Threshold of Hearing at 40 dB.
- A Threshold of Pain at 90 dB.
- A Saturation-Level (Psat) of 75 dB, where sounds begin to become uncomfortable.
- Difficulty to hear high frequency sounds.

6.2. Filter-Bank Approach Matlab Results

We implemented the Filter-Bank approach using both Finite-Impulse Response (FIR) and Infinite-Impulse Response (IIR) filters. FIR filters are preferred due to their stability and linear-phase properties. The Graphical User Interface for frequency shaper is shown in the Figure 21.

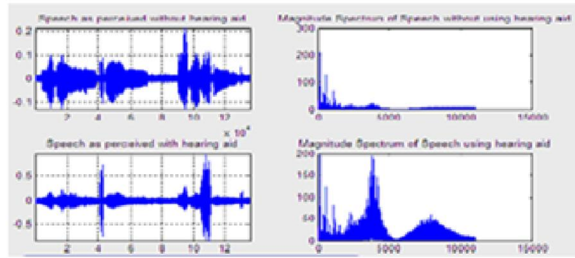

Figure 21. Spectral Shaping via Filter Bank Approach

6.3. FFT Based Approach Results

The plot for the frequency shaper via FFT approach is shown in the Figure 22.

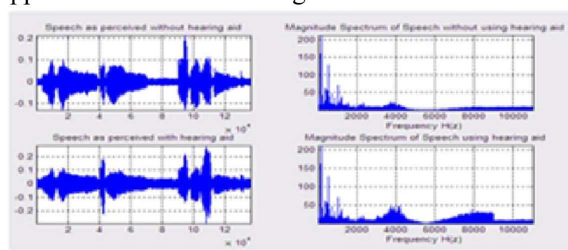
Figure 22. Spectral Shaping via FFT Approach

Frequency shaping using this method gives us more control over the band classifications but this also incurs some delay in adjusting every single frequency individually.

7. Amplitude Compression

The basic function of hearing aid is to make sounds audible, yet not uncomfortably loud for the user. The Frequency Shaper amplifies the sound in the hard-to-hear frequency regions. During this process, some sounds may exceed a certain level and add to the listening discomfort of the user. Therefore the hearing aid must imply some amplitude compression mechanism to control overall gain of the speech amplification system according to the listening comfort of the hearing impaired.

Amplitude compression is based on the average power in the signal. As long as the input power to the compressor is less than a preset threshold, no compression takes place and the input is equal to the output. Some systems use **Peak-Clipping** to keep high-level signals from exceeding listener's threshold of discomfort. This brings about some distortions in the output signal, which have a negative impact on sound quality and intelligibility [12]. Modern amplitude compression systems, also known as **Compression-Limiting** hearing devices, use a completely different approach to handle the high level signals from exceeding listener's threshold of comfort. In these systems, the input signals with power greater than the threshold are less amplified than those occurring below the threshold as opposed to removing portions of the output signal as in peak clipping instruments [13]. Limiting the output in this fashion avoids the waveform distortions associated with peak clipping devices [13], thereby resulting in greater listening comfort. Many modern hearing aids split sound into several frequency bands and apply compression separately in each band. This is known as 'multi-band compression'.

7.1. How does the Amplitude Compressor work?

We know that the Frequency Shaper applies different gains to different frequency bands of the input speech signal that the user has difficulty in hearing. The job of Amplitude Shaper is to process the input signal sample-by-sample and ensure that output power does not exceed a given Saturation-Level Psat, as shown in Figure 23.

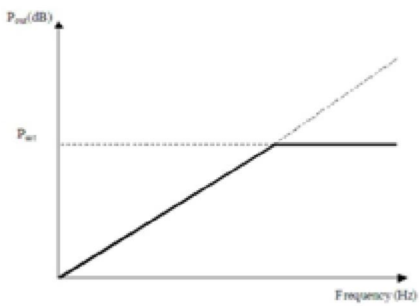

Figure 23. Magnitude Response after Amplitude Compression

The Saturation-Level Psat is where the sound signal begins to become uncomfortable.

7.2. Matlab Implementation & Results

Again we take the example of a person who has a moderate hearing loss and has a Saturation-Level (Psat) of 75 dB. We also know that this person has difficulty in hearing high frequency sound. So whereas, the Frequency Shaper raises the hard-to-hear frequencies within his dynamic range of hearing of the user, the Amplitude Shaper has to check, bit by bit, that output power does not exceed the given saturation level, Psat, which in this case is 75 dB. Output power is set to Psat for the levels above Psat. The Graphical User Interface for such Amplitude Compressor is shown as in Figure 24.

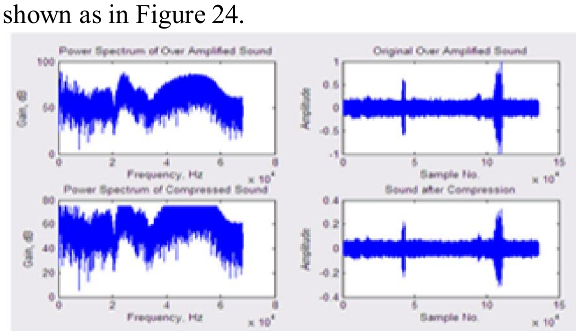

Figure 24. Amplitude Compressor

8. Conclusion and Future Work

Hearing aid technology has seen significant improvements over the past ten years. This mainly came with the introduction of digital signal processing (DSP) in the hearing aid industry. The implementations of advanced signal processing algorithms in hearing aids have overcome much of the deficiencies associated with analog hearing aids. Noise reduction techniques and independent multiple band processing are the key features of digital hearing aid which make it dominant to its predecessor.

New technologies have been developed and are constantly evolving to meet the unmet needs of the consumers. We observe that digital wireless technology is not very common in hearing aids today just because of power consumption issues. However as digital wireless chips continue to be designed smaller and lower in power, these limitations will disappear and by then the majority of hearing aids will have wireless receivers embedded in them in the same way that the majority of hearing aids today have DSPs.

Algorithms that currently exist in the hearing aids will be improved. The need for this refinement is because most of the algorithms in hearing aid industry have been borrowed from other industries where their implementations need different set of resources. As the speed and memory of hearing aid chips increase, the more sophisticated versions of current hearing aid algorithms will be developed by the hearing aid companies, utilizing complex signal processing techniques.

The science of auditory perception is a mature field. However, little research contribution has been made to the hearing aid design and hearing aid fitting. The future will see the successful application of hearing science to DSP technology innovations. The most direct application of hearing science to new digital technology in the future will be the application of auditory models to hearing aid signal processing.

Acknowledgments:

This project was funded by the Deanship of Scientific Research (DSR), King Abdulaziz University, Jeddah, Under Grant no. 525/830/1431. The authors, therefore, acknowledge with thanks DSR technical and financial support.

Corresponding Author:
Abdul Waheed Khan
Faculty of Computing and Information Technology Rabigh, King Abdulaziz University, Rabigh, Saudi Arabia
E-mail: awk_career@yahoo.co.uk

References

1. B. Edwards, "*The Future of Hearing Aid Technology*", Trends in *Amplification*, 11(1), pp. 32-45*,* 2007.
2. T. Stetzler, N. Magotra, P. Gelabert, P. Kasthuri, and S. Bangalore, "*Development of a low power digital hearing processor*", Int'l Hearing Aid Research Conf., August 2000.
3. J. Rodman, "*The effect of bandwidth on Speech Intelligibility*", Polycom inc., White paper, 2003.
4. H. Levitt, PhD, "*Noise reduction in hearing aids: a review*", J of Rehab Research and Development, Vol. 38, No. 1, pp. 111-121, January/February 2001.
5. S. G. Tanyer and H. Ozer, "*Voice Activity Detection in Nonstationary Noise*", IEEE Transactions on Speech and Audio Processing, Vol. 8, No. 4, July 2000.
6. S. Boll, "*Suppression of acoustic noise in speech using spectral subtraction*" Acoustics, Speech and Signal Processing, IEEE Transactions on27, no. 2 (1979): 113-120.
7. D. L. Wang and J. S. Lim, "*The unimportance of phase in speech enhancement*", IEEE Transactions on Acoustics, Speech, and Signal Processing, Vol. 30, pp. 679-681, August 1982.
8. M. Sen, Kuo and B. H. Lee, "*Real-Time Digital Signal Processing: Implementations, Applications and Experiments with the TMS320C55x*", Willey. com, 2006.
9. J. R. Deller, J. H. L. Hansen, and J. G. Proakis, "*Discrete-Time Processing of Speech Signals*", Wiley-Blackwell; Reprint edition, 1999.
10. Sambur MR, "*Adaptive noise cancelling for speech signals*", IEEE Transactions on Acoustics, Speech, and Signal Processing, Proc ASSP-26, pp. 419-423, 1979.
11. S. E. Voss, "*Method and device to optimize an audio sound field for normal and hearing-impaired listeners*", 2006.
12. J. J. Demsey, "*Circuitry Options: Signal Processing at Average Conversation Levels*". In Sowell, T.T. (Ed.), Practical Hearing Aid Selection and Fitting (pp. 17-34). Baltimore, MD: Department of Veteran's Affairs – Veteran's Health Administration, 1997.
13. H. Dillon, "*Compression? Yes, but for low or high frequencies and with what response times*?" Ear and Hearing 17 (4), pp. 287-307, 1996.

12/23/2013