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ELIMINATION OF FEEDBACK ACOUSTIC NOISE IN HEARING AID USING IMPROVED NLMS ADAPTIVE FILTER

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ABSTRACT

Adaptive filtering constitutes one of the core technologies in digital signal processing and has numerous application areas in science as well as in the industries. Adaptive filtering techniques are used in a wide range of applications, including echo cancellation, adaptive equalization, adaptive noise cancellation, and adaptive beam forming. Acoustic echo cancellation is a common occurrence in today's telecommunication systems. The signal interference caused by acoustic echo is distraction to users and causes a reduction in the quality of the communication. The method used in carrying this research work is known as the normalized least mean square. This focuses on the use of the combination of Finite impulse response digital filter and Normalized least mean square algorithms to reduce this unwanted echo, thus increasing communication quality. This research work further did some comparative analysis of Finite impulse response digital filter, Least mean square, Normalized least mean square. But the improved Normalized least mean square adaptive filter response obtained in this research work turns out to be a positive result, with the highest signal to noise ratio and lower mean square error at 0.2 steps and 25 filter size. We are also recommending this method to be used in noise elimination in hearing aids, electrocardiograph systems, communication industries and any area that requires noise cancellation.

KEYWORDS: Adaptive Filter, Finite Impulse Response, Hearing Aid, Noise, Acoustic, Feedback, Low pass Filter, High pass Filter

1.0 INTRODUCTION

A hearing is an electroacoustic tool that's worn in or at the back of the ear. It is designed for the development of speech intelligibility and for correcting impaired listening to as measured through the audiometry. There are essential varieties of listening to aids, namely, the analog and the virtual listening to aides, amongst human disabilities, deafness might be taken into consideration as an extreme handicap, which threatens a big a part of the population. When deafness happens by accident in the course of life, few sufferers document that they cross through quite a few problems from this handicap considering the fact that they had been accustomed with listening to faculty [2][8]. In humans, the declaration aid to impairment is used for humans who've near insensitivity to sound or air vibration with inside the speech frequencies. The circumstance of a hearing loss may be labeled primarily based totally at the boom in quantity that have to be made above the traditional stage earlier than the listener can come across it. Digital kinds the previous operates through processing the electric sound sign via way of means of certainly amplifying each the favored speech in addition to the noise sign [2].

Furthermore, in a listening to-resource tool, the acoustic remarks route among the receiver and the microphone is the supply of instability for the entire closed-loop device. When the gadget is volatile the tool sound pleasant is deteriorated seeing that distortion and excessive-degree self-maintaining oscillations may be made. These phenomenon are very unsightly for the listening to-impaired and restriction the capacity sound excellent of the tool. This observe proposes to introduce a progressed remodel area echo cancellation set of rules the use of an aggregate of virtual fir excessive pass clear out, virtual fir low pass clear out and an NLMS adaptive clear out for noise and echo cancellation in listening to useful resource. Noise presence commonly degrades the nice of offerings and the data content material of acoustic sign. Therefore, listening to-impaired humans is probably experiencing a few problems in information what they heard. Subsequently, to cope with this problem, a technique that could suppress the noise even as keeping the desired speech fine and intelligibility is important to beautify the overall performance of the listening to-useful resource. This task is addressed through virtual listening to aids. With digitization, electric sound sign can go through distinct superior sign processing which include noise discount, filtering, acoustic remarks cancellation, and sound classification. Consequently, digitization improves the overall performance and versatility of virtual listening to aids in addition to improving the listening consolation of the customers [9]. Hence, speech enhancement is a key a part of sign processing aiming at enhancing the sign-to-noise ratio (SNR), intelligibility and sound great of speech sign. There are many answers which have been proposed with inside the literature to cope with the problems of noise estimation and speech enhancement for listening to aids [9].

1.2 Statement of Problem

Over the years hearing aid has been a good assistant to some people with hearing loss conditions, but to some it has not been a good side this is due to the feedback acoustic signal that is been re-amplified again, thereby creating whistling sound(noise), echoes and other forms of noise in the ear which is so unwanted. However, this feedback acoustic signals that is been re-amplified from the head phone back to the condensed mic has actually cause more noise in the ears that has kept the users so un comfortable, this problem also tends to affect the original speech signal that is been amplified and makes it difficult for the user not to get the original message.

1.3 Aim of the study

The aim of this research work is to eliminate the feedback acoustic noise in hearing aid for improve signal gain quality.

The objectives of this research work are presented as:

i. To Evaluate the finite impulse response high pass filter
ii. To evaluate the finite impulse response low pass filter
iii. To implement the normalized least mean square algorithm for noise reduction
iv. To Conduct simulations using a software tool (MATLAB/SIMULINK)
v. To determine the frequency response of the output signals for investigations
vi. To compare the input and output signal (Frequency) of the aided device to the output of the NLMS adaptive filter frequency response

2.0 LITERATURE REVIEW

According to [1] they both delivered a specialized model of the Kalman clear out suited for ANC which became a smooth theoretical evaluation with the RLS and LMS algorithms become made, which suggests why the Kalman clear out become capable of outperform the formers. For a basic route modeling, a method that gets rid of the consequences of the secondary direction turned into used to put in force the traditional adaptive filters model of the Kalman clear out with right outcomes [3]. For on-line secondary direction modeling a Kalman clear out form of the whole modeling set of rules became used. For this case, the shortage of excitation of all of the classes of the signal may be a trouble [1].

According to [7] they proposed a Fast Block LMS set of rules for speech sign which became simulated and examined via way of means of the use of MATLAB. The effects from the FBLMS suggests that it may get rid of the exceptional degrees of noise extra successfully and successfully and which may also purpose a right away reaction. It has a low computational complexity asset than LMS set of rules [7]. In Low energy noise the machine confirmed SNR development as much as 13.15dB, for Medium noise degree the device confirmed SNR development as much as 20.5dBand in excessive noise degree the device confirmed SNR development as much as 10.75dB. This strategy may be used for noise removal in speech sign and Bio clinical alerts.

According to [5] in 2017, an adaptive multifunction filters out noise for radar signal processing. He went in addition to mention that, an adaptive multifunction FIR clear out has been proposed to carry out distinct tasks, inclusive of filtering of the accrued sign to reject out-of-band interference, conveying goal indication and coupled filtering. MATLAB software is used with inside the layout of every of the man or woman filters and the ensuing multifunction clear out, using the integrated capabilities and the clear out layout and evaluation tools [5]. Simulink turned into used to check an adaptive LMS clear out that acts because the supposed multifunction clear out.

According to [4], the author made it clear that LMS adaptive set of rules on finite impulse reaction filters could be very powerful in filtering out additive white Gaussian noise from voice alerts. The optimized values of the variable parameters including clear out order and the step length rely upon the sort and importance of noise to be cancelled [2]. The noise electricity present with inside the filtered sign from frequency of 8hz and above is as it isn't always a part of lively speech sign segment for the duration of recording, as a consequence creating room for noise to cowl up the segment in the course of contamination [4][9].

3.0 Method

3.1 Description of the Improved NLMS

This entire system eliminates feedback acoustic noise by the means FIR digital filter combination with the NLMS adaptive filter. The high pass FIR, filters the noise and sum it with the original speech which is then filtered by the low pass FIR digital filter. The output signal is then fed into the NLMS desired port while part of the original speech is fed into the input of the NLMS adaptive filter, then the NLMS does the final filtering by comparing the input signal and desired signal coefficient in order to reduce the noise error and improve signal to noise ratio as shown in figure 1.

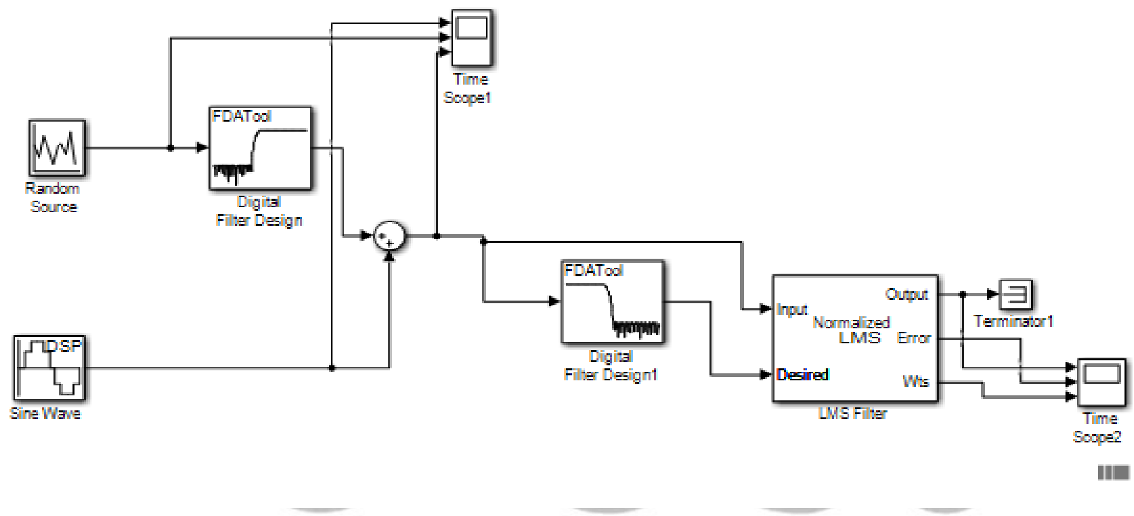

Figure 1: Improved NLMS Adaptive Filter

3.2 Evaluation of Hearing Assessment for FIR High Pass Filter

Knowing that the human ear choices up sound from 20 Hz- 20khz, and sounds beneath the 20 Hz are categorized as infrasound at the same time as sound above 20KHz are labeled as ultrasound. And additionally taking word that from 20db-80dB are sounds below then sounds above the variety are risk to the human ears so the layout of virtual FIR filter of duration N with an input- output dating may be defined with the aid of using the subsequent distinction equation [12]:

$$y(n) = \sum_{k=0}^{N} b_N x(n-k)$$

$$= b_0 x(n) + b_1 x(n-1) + \cdots \dots \dots + b_N x(n-N+1) \qquad (3.1)$$

Where (bk) represents the filter coefficients set. The output y(n) is a function only of the input signal x(n). FIR filter can also be characterized by its transfer function as follows:.

$$H(z) = \sum_{n=0}^{N} h(n) z^{-n} \quad n = 0,1, \dots\dots, N \qquad (3.2)$$

Where the coefficients h(n) constitute the impulse reaction of finite duration and N represents the clear out order. Thus, the range of coefficients will be (N +1). The sort of the filter for a pass or attenuate e.g. excessive pass, low pass, band pass, etc. is decided through the clear out coefficients h(n) which might be to be exact with inside the layout steps. When the coefficients of FIR clear out are symmetrical across the middle coefficient, the FIR filter is a linear-segment in any other word, the coefficients wide variety this is absolutely optimized is (N/2+1), that is same to thirteen on this work. The coefficient vector {h0,h1,...,hN} is represented through the debris positions in (N+1) dimensional seek area of PSO algorithm. Hence, in every cycle of the evolutionary algorithm, those debris locate new positions to be the brand new coefficient vector of the switch function. The following equation describes the FIR clear out frequency reaction [12][10]:

Where, $H_d(e^{jw}) = \sum_{n=0}^{N} h(n)e^{-jwn}$ (3.3)

For ideal HP filter, the following equation defines the filter response

$$H_d(e^{jw}) = \begin{cases} 0 & 0 \le \omega \le \omega_c \\ 1 & \end{cases} \quad (3.4)$$

Where ωc is the cut-off frequency as shown in figure 1.

By sampling the frequency in the range [0, π] with L points, we get the following equations:

$$H_d(\omega) = [H_d(\omega_1), H_d(\omega_2), H_d(\omega_3), \ldots\ldots, H_d(\omega_L)]^T \quad (3.5)$$

$$H_i(\omega) = [H_i(\omega_1), H_i(\omega_2), H_i(\omega_3), \ldots\ldots, H_i(\omega_L)]^T \quad (3.6)$$

Where $H_d(\omega)$ and are$H_i(\omega)$ the frequency response of the designed and ideal filter respectively. So, the error function is defined by the following equation:

$$E(\omega) = [H_d(\omega) - H_i(\omega)] \quad (3.7)$$

Then, the error function above has been utilized for obtaining the fitness function as follow

$$\sum_1^L |(E(\omega)| \quad (3.8)$$

3.3 Evaluation of Hearing Assessment for FIR Low Pass Filter

FIR Filter Design through Windowing In designing FIR filter, given the frequency reaction Hd (ejω) and impulse reaction hd[n] of a really perfect device, we would really like to approximate the infinitely lengthy hd [n] with a finite series h[n], wherein h[n] = zero besides for zero ≤ n ≤ M . Consider a super low pass filter whose frequency reaction is finite and rectangular [13]. A feasible approximation mistakes criterion may be defined as

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega \quad (3.9)$$

To minimize E, use Parseval's theorem:

$$E = \frac{1}{2\pi}\int_{-\pi}^{\pi} |H_d(e^{j\omega}) - (e^{j\omega}|^2 d\omega$$

$$= \sum_{n=-\infty}^{\infty} |h_d[n] - h[n]|^2 = \sum_{n=0}^{M} |h_d[n] - h[n]|^2 + \sum_{n=z[0,M]} |h_d[n] - h[n]|^2 \quad (3.10)$$

$$h[n] = \begin{cases} H_d[n] & 0 \le n \le M \\ 0 & otherwise \end{cases} \quad (3.11)$$

The optimal FIR approximation using the mean square error criterion gives the truncation of the ideal impulse response. We can represent h[n] as the product of the ideal impulse response with a finite-duration rectangular window w[m]:

$$h[n] = h_d[n]\omega[n] \quad (3.12)$$

$$\omega[n] = \begin{cases} 1 & 0 \le n \le M \\ 0 & otherwise \end{cases} \quad (3.13)$$

$$H_{(e^{j\omega})=\frac{1}{2\pi}\int_{-\pi}^{\pi} H_d(e^{j\theta})W(e^{j(\omega-\theta)}d\theta} \quad (3.14)$$

this convolution procedure implied via way of means of truncation of an appropriate impulse reaction. The ensuing value approximation because the due to the fact that pulse W (ej(ω−θ)) slides byskip an appropriate frequency reaction Hd(ejω). When W (ej(ω−θ)) actions throughout the discontinuity of Hd(ejω), a transition band consequences and ripples arise on each sides. The major lobe of the window frequency reaction controls transition bandwidth Δω ≈ 2π(2/(M + 1)). The primary lobe is defined because the place among the first 0 crossings on both aspect of the origin. It is applicable to have W (ejω) as focused in frequency as viable. On the opposite hand, aspect lobes manage pass band and forestall band ripples. The large the location below the facet lobes, the bigger the ripples. Pass band and prevent band ripples are about identical over a extensive variety of frequencies [13].

3.4 Implementation of the Improved NLMS Algorithm

With each iteration of the improved NLMS algorithm, the filter tap weights of the adaptive filter are updated according to the following formula. $w(n+1) = w(n) + \mu e(n)x(n)$ (3.15)

Here x(n) is the input vector of time delayed input values, x(n) = [x(n) x(n-1) x(n-2) .. x(n-N+1)]T. The vector w(n) = [w0(n) w1(n) w2(n) .. wN-1(n)]T represents the coefficients of the adaptive FIR filter tap weight vector at time n. The parameter μ is known as the step size parameter and is a small positive constant [11].

One of the primary disadvantages of the LMS algorithm is having a fixed step size parameter for every iteration. This requires an understanding of the statistics of the input signal prior to commencing the adaptive filtering operation. In practice this is rarely achievable. Even if we assume the only signal to be input to the adaptive echo cancellation system is speech, there are still many factors such as signal input power and amplitude which will affect its performance [11]. The normalized least mean square algorithm (NLMS) is an extension of the LMS algorithm which bypasses this issue by calculating maximum step size value. Step size value is calculated by using the following formula. Step size=1/dot product (input vector, input vector) This step size is proportional to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector x(n)[6]. This sum of the expected energies of the input samples is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, [11].

$$tr = [R] = \sum_{i=0}^{N-1} E[x^2(n-i)]$$

$$= E\sum_{i=0}^{N-1}[x^2(n-i)] \quad (3.16)$$

The recursion formula for the NLMS algorithm is stated in equation

$$w(n+1) = w(n) + \frac{1}{x^T(n)x(n)} e(n)x(n) \quad (3.17)$$

The NLMS algorithm has been implemented in Matlab. As the step size parameter is chosen based on the current input values, the NLMS algorithm shows far greater stability with unknown signals. This combined with good [11].

The output of the adaptive filter is calculated

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = w^T(n)x(n) \quad (3.18)$$

An error signal is calculated as the difference between the desired signal and the filter output

$$e(n) = d(n) - y(n) \quad (3.19)$$

The step size value for the input vector is calculated.

$$\mu(n) = \frac{1}{x^T(n)x(n)} \quad (3.20)$$

The filter tap weights are updated in preparation for the next iteration

$$w(n+1) = w(n) + \mu(n)e(n)x(n) \quad (3.21)$$

3.5 Implementation Means square error of the improved NLMS

$$X(k)=X(K) = \sum_{n=0}^{N-1} x(n)e^{-j2k/N} \quad (3.22)$$

Where N is the number of samples, and n and k are integer values that vary from a reference point to equivalent number of samples in N. That is, n and k can vary from 0 to N-1, 1 to N, 2 to N+1, etc.

$$N0=S-SF \quad (3.23)$$

where S is the power of the original voice signal and SF, the power of the filtered voice signal.

By comparing the power magnitudes for the original, contaminated and filtered voice signals at the nine frequency positions, it clearly shows that the adaptive filter drastically reduced the noise at each of the frequencies. [4]

$$MSE = \frac{\sum_{k=1}^{N}(A(k)-Af(k)^2}{N} \tag{3.24}$$

A small MSE implies an powerful and green clear out and algorithm. The system for computing the MSE is given via way of means of (3.24) [4], Where A(k) is the amplitude of the unique voice sign and 'Af(k) ', the amplitude of the filtered voice sign as 'k' varies from 1 to the wide variety of samples N. The unique and filtered amplitudes have to be measured on the equal or equal factors with inside the device and with inside the identical device bandwidth.

In (f) the sign to noise ratio (SNR) is a determine of benefit that measures the percentage of noise gift with inside the filtered sign. A excessive determine manner that small noise is gift with inside the filtered sign and the best of the filtered sign is excessive. The components for computing sign to noise ratio is given as [4].

$$SNR = 10log\frac{\sum_{k=1}^{N}A^2(k)}{\sum_{k=1}^{N}N_a^2(k)} \tag{3.25}$$

Where A(k) is the amplitude or electricity of the unique voice sign and Na(k) the amplitude or energy of the noise gift with inside the filtered voice sign as 'k' varies from 1 to the variety of samples N. The unique and filtered sign electricity or amplitudes have to be measured on the identical or equal factors with inside the device and with inside the identical device bandwidth. The noise amplitude is acquired from

$$Na(k) = A(k) - Af(k) \tag{3.26}$$

$$N_0 = S - S_F \tag{3.27}$$

The noise power present in the filtered voice signal is calculated from (3.27) [4].

3.6 Overall Error Evaluation of the Improved NLMS Adaptive Filter

From equation (3.7(,(3.9) and (3.19) the overall error estimation of the improved NLMS adaptive filter can be expressed as the subtraction of the FIR high and low pass digital filter mean square error from the NLMS filter as in the equation

$$E_O = E(\omega)_{hp} - E(\omega)_{lp} - e(n)_{nlms} \tag{3.28}$$

4.0 Finite Impulse Digital Filter Result Presentation

From the result in the result in the graph we can see the speech, noise and the filtered signals. This implies that noise reduction in a system with the digital filter is more significant to analogue filter. But however, the results in figure 2 shows that the noise in the system has been reduced but still not enough to have a complete audible signal in the hearing aids when compare to other advance methods.

Table 4.1: Shows the Results of the FIR Filter

Algorithm	Step size	MSE	SNR
FIR	0.025	$5.12e^{-003}$	12.54
FIR	0.2	$5.32e^{-003}$	12.47
FIR	0.025	$5.19e^{-003}$	13.68
FIR	0.2	$2.07e^{-003}$	15.12

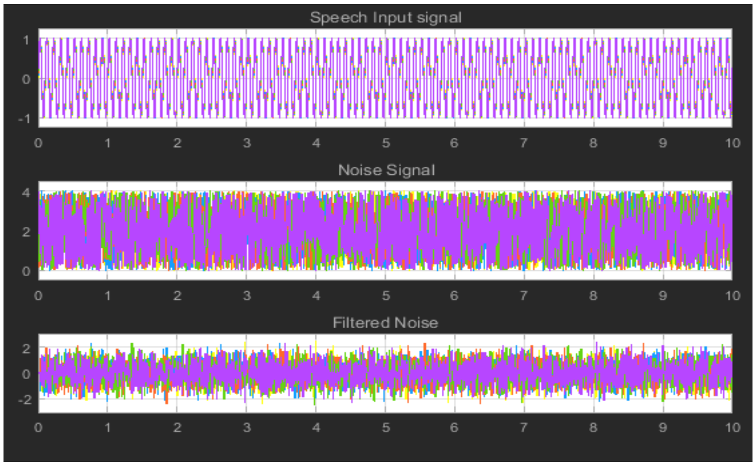

Figure 2: FIR Digital Filter

4.2 Least Mean Square Adaptive Filter Response

The least mean square adaptive filter is an advance digital filter with an impressive algorithm that makes a lot of changes in echo cancellations. However, the results from figure 3 shows that even as super as the LMS adaptive filter but it still has a higher error rate though still superior to the convention digital filter. The error rate signifies that there are still noise presence in the hearing aid.

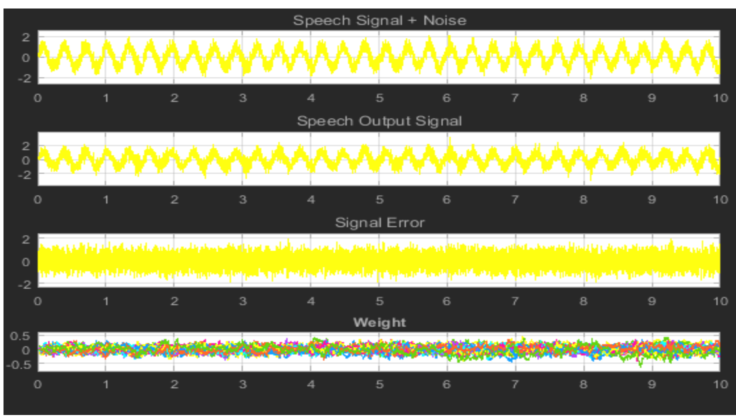

Figure 3: LMS Adaptive Filter Response

Table 4.2: Displays the Result of an LMS Adaptive Filters

Algorithm	Filter size	Step size	MSE	PSNR
LMS	10	0.025	$4.09e^{-002}$	13.87
LMS	10	0.2	$4.22e^{-002}$	13.74
LMS	25	0.025	$4.68e^{-002}$	14
LMS	25	0.2	$1.58e^{-002}$	17.02

4.2 Normalized Least Mean Square Adaptive Filter Response

The results from figure 4 explains that the NLMS has a lower error rate when compare to LMS algorithm this makes it to be superior over the LMS, but as beautiful as the error rate the result shows that as the error rate is reducing the output signal is also distorted.

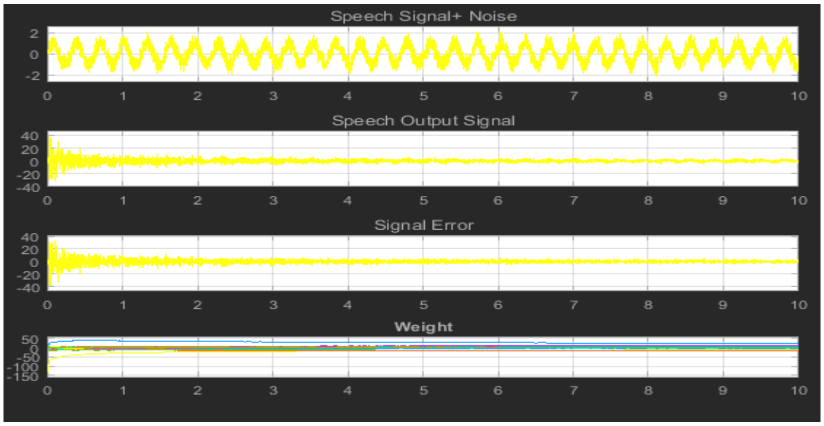

Figure 4: NLMS Adaptive Filter Response

Table 4.3 Indicates the Normalized Least Mean Square Results

Algorithm	Filter size	Step size	MSE	PSNR
NLMS	10	0.025	$1.62e^{-002}$	17.88
NLMS	10	0.2	$1.69e^{-002}$	17.71
NLMS	25	0.025	$1.64e^{-002}$	17.70
NLMS	25	0.2	$1.69e^{-002}$	17.83

4.3 Improved NLMS Adaptive Filter Response

The human ear actually pickup sounds from 20 0r 30 KHz to 20khz which is the highest pitch. All sounds that are below 20 KHz are classified as infrasound and all sound above 20 KHz are classified as ultrasounds. From 20 dB to 80 dB is a safe hearing environment for humans which is within the conversation terrain. The results below from figure 5 indicates the input response of the improved NLMS adaptive filter. This filter consists of a combination of high and low finite impulse response digital filters and a NLMS adaptive filter. The input signal of 5 KHz with a random noise.

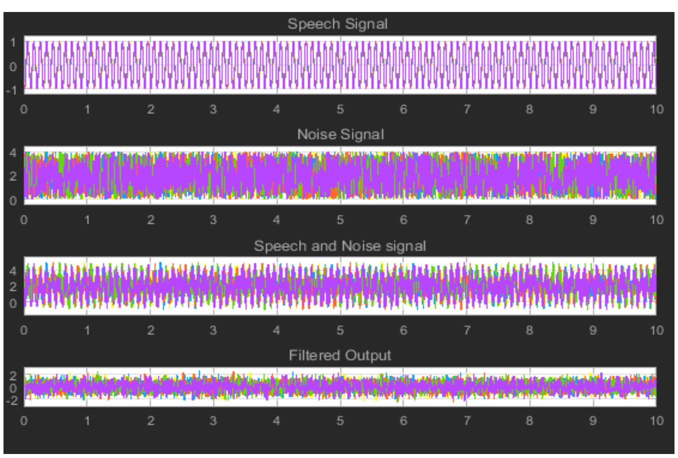

Figure 5: Improved NLMS Adaptive Filter

From the figure 6 below displays the output result of the improved NLMS adaptive filter. The result shows that the improved NLMS adaptive filtering methods is superior to the others aforementioned filters in this research work.

The input signal of the hearing aid was successfully produced at the output with a very low error rate when compare to others. That is to say the novel combination posed more improvement in feedback noise cancellation in hearing aids. However, this method suppresses the noise better and improved the signal gain, that is to say that the more the feedback acoustic noise is reduced the more the output gain will be realized. This method has successfully reduced the noise error to a very low percentage.

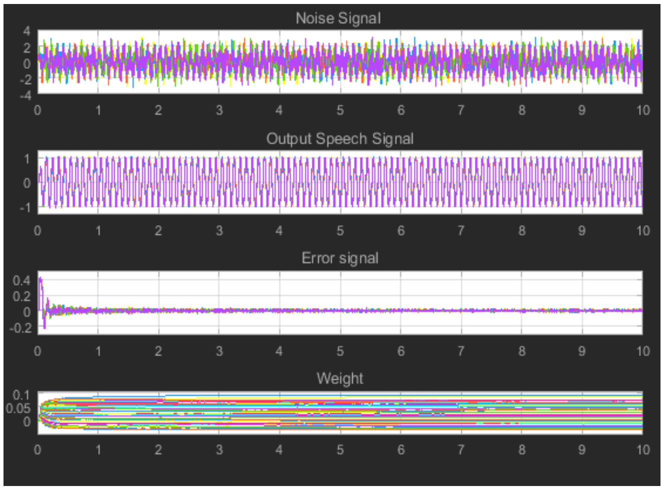

Figure 6: Shows the Results of the Improved NLMS

As you can see from the table the means square error is reduced and the signal to noise ratio increase. When compare to others in this research work u can see that the improved NLMS is better and superior to others with a higher signal to noise ratio making the feedback acoustic noise to almost 100 percent eliminated.

Table 4.4: Results of the improved NLMS

Algorithm	Filter size	Step size	MSE	PSNR
Improved NLMS	10	0.025	$0.45e^{-002}$	25.64
Improved NLMS	10	0.2	$0.48e^{-002}$	25.58
Improved NLMS	25	0.025	$0.49e^{-002}$	25.55
Improved NLMS	25	0.2	$0.48e^{-002}$	25.62

4.4 Comparative Analysis of FIR, LMS, NLMS and Improved NLMS Adaptive Filters

The comparative results here in figure 7 show that the FIR has the lowest signal to noise ratio and the highest least mean error, then the LMS adaptive filter has a better performance ahead of the conventional FIR, then the NLMS adaptive is the superior to the LMS and FIR when it comes to noise cancellation. Lastly the Improved NLMS happens to be the best and the superior over due to it high signal to noise ratio and very low least mean error. This attributes has made the improved NLMS to be more relevant in cancelling of feedback acoustic noise in hearing aids.

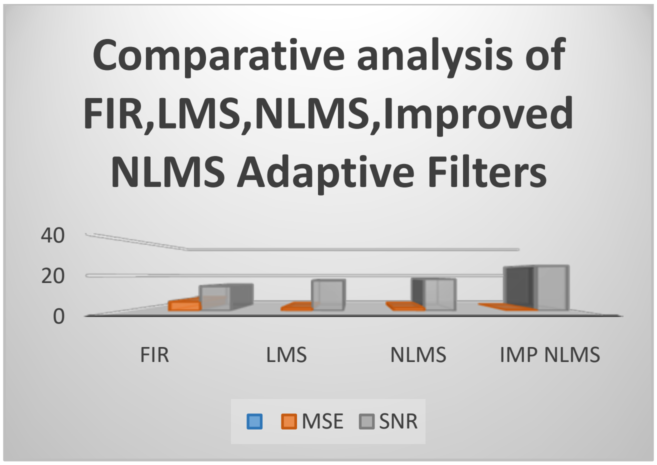

Figure 4.7:Comparative results of FIR,LMS,NLMS and Improved NLMS

The pie chart here in figure 4.8 has expressed that when looking for a method to eliminate feedback acoustic noise in a hearing aid, that the improved NLMS is more reliable due to it high signal to noise ratio.

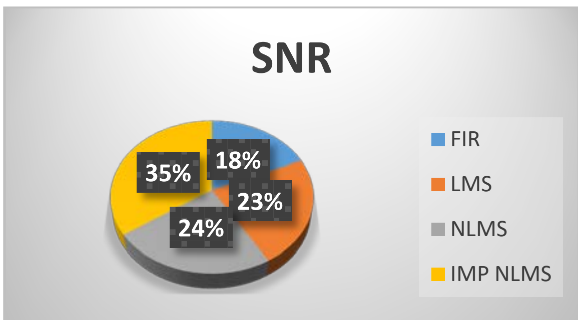

Figure 8: Signal to Noise Percentages of Each of the Digital Filters

5.0 Conclusion

Feedback acoustic noise has been the major problem of hearing aids due to the noise being produced as a result of feedback. The improved NLMS adaptive filter has proven itself to be superior over other filters being compared with by tackling the issues of feedback acoustic noise by reducing the noise to a very low rate. In this research work we evaluated the error in the FIR high pass filter and low pass filter also evaluated the error in the NLMS filter. Governing equation was formed to evaluate the mean square errors of the finite impulse response digital filters and also the signal to noise ratio of the system.

With the SIMULINK we were able to design the feedback acoustic noise cancellation system, the result was absolutely positive, the improved NLMS had the lowest mean square error with the highest signal to noise ratio, then followed by the NLMS adaptive filter then the LMS filter before the FIR digital filter which came last in the result when the comparative analysis was carried out.

The improved NLMS is very suitable and reliable for noise cancellation because of it high signal to noise ratio and low mean square error rates

5.1 Recommendations

Normalized Least Mean Square adaptive filter is a type of digital filter that cancels noise by comparing the coefficient of the signal with the error. However, in this research work we have been able to introduce an improved normalized NLMS adaptive filter which is use for noise cancellation and feedback noise elimination for hearing aids. The results when compare to others were superb to other adaptive filters. The improved NLMS has a very high signal to noise ratio with a very low Mean square error, which implies that the sound output quality is tremendous with a very low or almost no noise. From this results we hereby recommend this method to be applied in digital hearing aids, ECG devices and also other areas that requires effective noise cancellations.

Aknowledgement

Permit me to use this moment to give thanks to the almighty God and also to appreciate the effort of my father Texas O. Ogbondamati, Allwell worlu, Joyce Ukpom, Egnr Bakare, Dr Amadi, Dr Elechi and my fantastic Rivers State University colleagues for their unending support to make this research work a success.

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