

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 Or 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.

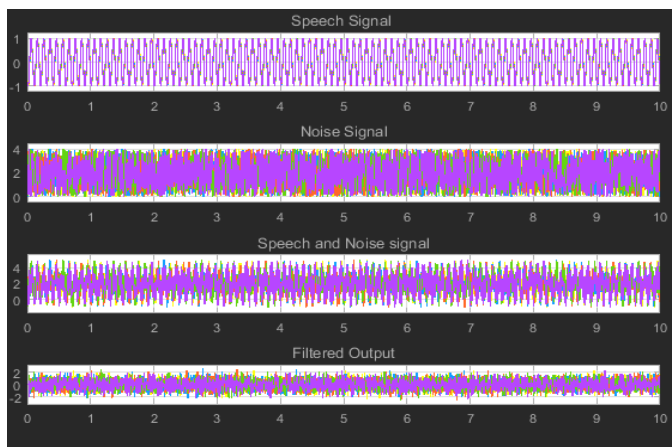


Figure 5: Improved NLMS Adaptive Filter

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.

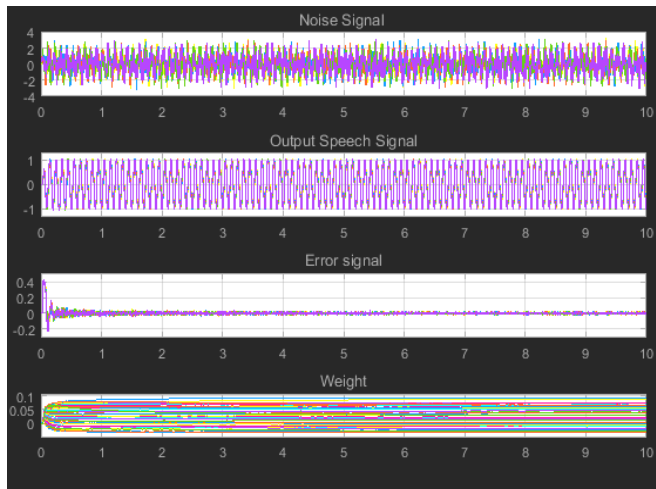


Figure 6: Shows the Results of the Improved NLMS

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.

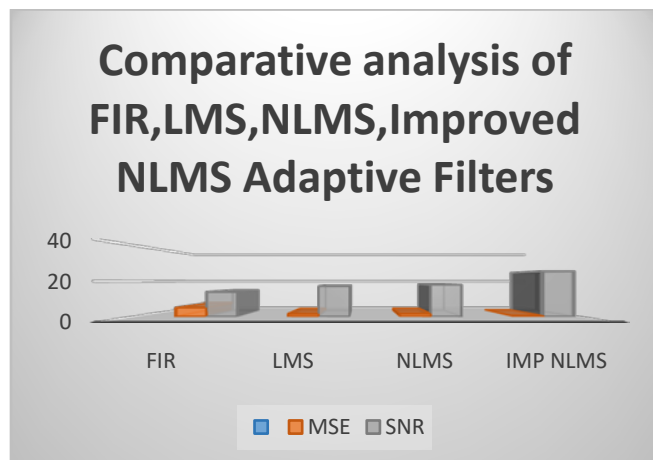


Figure 4.7: Comparative results of FIR, LMS, NLMS and Improved NLMS

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 its high signal to noise ratio.

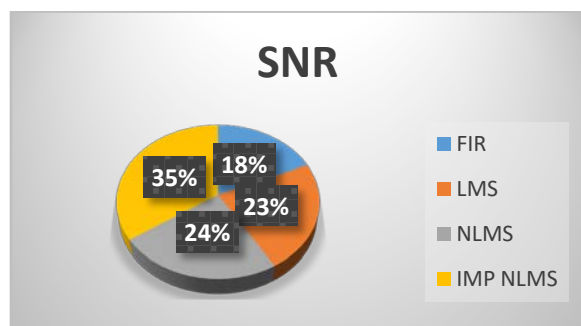


Figure 8: Signal to Noise Percentages of Each of the Digital Filters

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 its 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 used for noise cancellation and feedback noise elimination for hearing aids. The results when compared 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 these results we hereby recommend this method to be applied in digital hearing aids, ECG devices and also other areas that require effective noise cancellations.

Acknowledgement

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.

REFERENCES

- [1] Paulo A. C. & Lopes Moisés Piedade, S. THE KALMAN FILTER IN ACTIVE NOISE CONTROL, IST/INESC, Fort Lauderdale, Florida, USA, pp 7-11, 1999.
- [2] Clark. G. M, "University of Melbourne-Nucleus Multi-Electrode Cochlear Implant," Karger, New York, USA, 1987.
- [3] Levitt H., Bakke M., Kates J., Neuman A., Schwander T., & Weiss M. 'Signal processing for hearing impairment'. Scand Audiol; Supply 75th Street and 30th Avenue, Jackson Heights, NY 11370, USA, 39:7-19, 1993.
- [4] Mbachu C. B & Akaneme S. A, 'LMS-BASED ADAPTIVE FILTERING TECHNIQUE FOR REMOVING NOISE FROM VOICE SIGNALS AND ITS COMPARISON WITH RLS-BASED TYP', Department of Electrical and Electronic Engineering, Chukwuemeka Odumegwu Ojukwu University, Uli, Anambra State, Nigeria, pp 31-38, 2020.
- [5] Moutaman Mirghani Daffalla, 'Adaptive Multifunction Filter for Radar Signal Processing', Department of Electrical and Computer Engineering College – Karary University Khartoum, Sudan pp 5-6, 2017.
- [6] TELAGAREDDI, S. N. & RAMESH, U. V, 'Speech Enhancement in Hands-Free Device (Hearing Aid) with emphasis on Elko's Beamformer', Blekinge Institute of Technology, Sweden April, 2012.
- [7] Jebastine J. & Dr. B. Sheela Rani. DESIGN AND IMPLEMENTATION OF NOISE FREE AUDIO SPEECH SIGNAL USING FAST BLOCK LEAST MEAN SQUARE ALGORITHM, Department of Electronics and Communication, Research Scholar, Sathyabama University, Chennai-600 119. 2Vice-Chancellor & Dean, PG Studies and Research, Sathyabama University, Chennai-600 119, pp 40-49, 2012.
- [8] Hamida. A. B. "Implication of New Technologies in Deafness Healthcare: Deafness Rehabilitation Using Prospective Design of Hearing Aid Systems," in IEEE International Symposium on Technology and Society, pp. 85-90, 2000.
- [9] I.A Alimi and M.O Kolawole, Enhancement of speech communication technology performance a using adaptive filter control factor based spectral subtraction method.' Journal of telecommunication and information technology. Vol2, pp.35-39, 2013.
- [10] Harry Levitt, PhD, 'Noise reduction in hearing aids: a review', The Lexington School for the Deaf/Center for the Deaf, 75th Street and 30th Avenue, Jackson Heights, NY 11370, USA, 2001.
- [11] Radhika, C., Ramkiran, D. S, Khan. H, USHA. M, Madhav. B.T.P, Srinivas. K.P, Ganesh.G.V, 'Adaptive Algorithm for acoustic Cancellation in speech processing', Department of ECE, K L. University, Guntur, AP, India, pp 38-40, 2011.
- [12] Adel Jalal Yousif, Ghazwan Jabbar Ahmed, Ali Subhi Abbood, 'Design of Linear Phase High Pass FIR Filter using Weight Improved Particle Swarm Optimization', Electronic Computing Center University of Diyala, Diyala, Iraq (IJACSA) International Journal of Advanced Computer Science and Applications, Vol. 9, No. 9 2018
- [13] Schafer, R.W, and Buck, J.R, 'FIR Filter Design by windowing', Department of Electrical Engineering and Computer Science Massachusetts institute of technology, Massachusetts, USA, pp 1-5, 2006.