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MINIMIZATION OF ACOUSTIC NOISE IN A BROADCASTING STUDIO USING AN ADAPTIVE NOISE CANCELLER

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Abstract: **This work presents the minimization of acoustic noise in the Nigerian broadcasting industry using the adaptive noise canceller. This was embarked upon to address the problem of poor quality of service due to acoustic noise in broadcasting studio. To achieve this, empirical study to determine the level of noise in the broadcasting signal was conducted at Dream FM radio station, via characterization. A model of adaptive noise canceller was then developed using least square algorithm and noise canceller algorithm. The noise canceller detects the noise in the broadcasting signal, and then the least square algorithm filters the noise using reference noise value and gives throughput of clear broadcast service. This adaptive noise cancelling scheme was deployed on the radio station mixer using Simulink and then simulated for results and evaluations. The result showed that noise was reduced from 0.56dB to 0.22dB.**

Keywords: **Acoustic Noise, Adaptive Noise Canceller, Least Square Algorithm, Broadcasting Signal.**

I. INTRODUCTION

Over the years, noise has remained a major issue affecting the quality of service in all forms of information communication technologies (Rosselti, 2011). Noise is an unwanted signal which degrades the quality of service in communication systems via interrupting the real signal to upset the standard (Lim and Oppenheim, 2017; Gibson et al., 2019; Lakshmikanth et al., 2014).

One of the major sections of communication which has suffered this problem of noise the most is the radio broadcasting studio. This studio are designed be default to mitigate noise from the environment via studio padding, however acoustic noise from the mixer and other components like amplifiers, contributes negatively to the quality of broadcasting signal. This type of noise (acoustic) is an unpleasant signal which affects the quality of other real signals and has remained a very big problem to the quality of broadcasting signal in radio studio (Mohanaprasad et al., 2017; Zhengyou et al., 2016).

To address this problem, many solutions have been proposed such as linear filter, kalman filter among others, but despite the success the tolerance level of noise mitigated by these algorithms still leaves room for improvement (Rudolph, 2016; So and Paliwal, 2011; Greg and Gary, 2015). To achieve this desired quality of service, the use of adaptive noise canceller is proposed. This will be developed using least mean square algorithm and noise filter to improve the adaptivity and reliability of noise filter to mitigate acoustic noise to the lowest gain. This when achieved will go a long way to improve the quality of service in the radio broadcasting industries.

Aim and Objective of this research

The aim is to minimize acoustic noise in broadcasting studio using adaptive noise canceller with the following set out objectives;

i. To perform empirical study of a testbed and determine the acoustic noise level

ii. To develop an adaptive noise canceller to mitigate the acoustic noise

iii. To implement the noise canceller on the testbed using simulation

iv. To evaluate the performance

II. LITERATURE REVIEW

Author	Title	Technique	Work done	Success/limitation
Paliwal and Basu (2017)	A speech enhancement method using Kalman filtering	Kalman filter	The work reviewed various signal processing filters and took a position with kalman type for improving the quality of speech sound	Not reliable in noise conditions
Lim and Oppenheim (2017)	Enhancement and width compression of noisy speech	Linear filters	The work employs linear filter for the processing of noisy speech wave	The filter used was not intelligent and adaptive in nature
Rosselti (2011)	Digital communications processing in Wireless Local Network (WLAN).	infinite state algorithm	In the research, the proposed algorithm was used to improve the quality of data communication over WLAN	Delay computation time
Sujan K and Kuldip K., (2020)	Casual convolutional encoder decoder based augumentated kalman filter for speech enhancement	Kalman filter	The study used linear prediction coefficient of speech and noise signal to estimate Kalman filter performance of speech processing.	Better than the conventional kalman filter
Guthrie et al. (2005)	The replacement of frequency agile bands pass filters in fronts of receivers' low noises amplifier (LNAs) and after the transmitter power	Post filtration technique	This technique stops signal interference by post filtering blocks through the post filtration of any reverse signal that enters the low noise amplifier	The work implements post filtering technique using low noise amplifier

III. EXPERIMENTAL SETUP FOR EMPIRICAL DATA COLLECTION

This work studied the dream FM radio broadcasting studio located in Enugu Nigeria which broadcast at 92.5MHz. This was done to collect broadcasting signal data and analyze the quality for improvement. The materials used were monitor, mixer, laptop, router, processor, microphone, mix engine, etc. The testbed used for the study was presented in the figure 1;

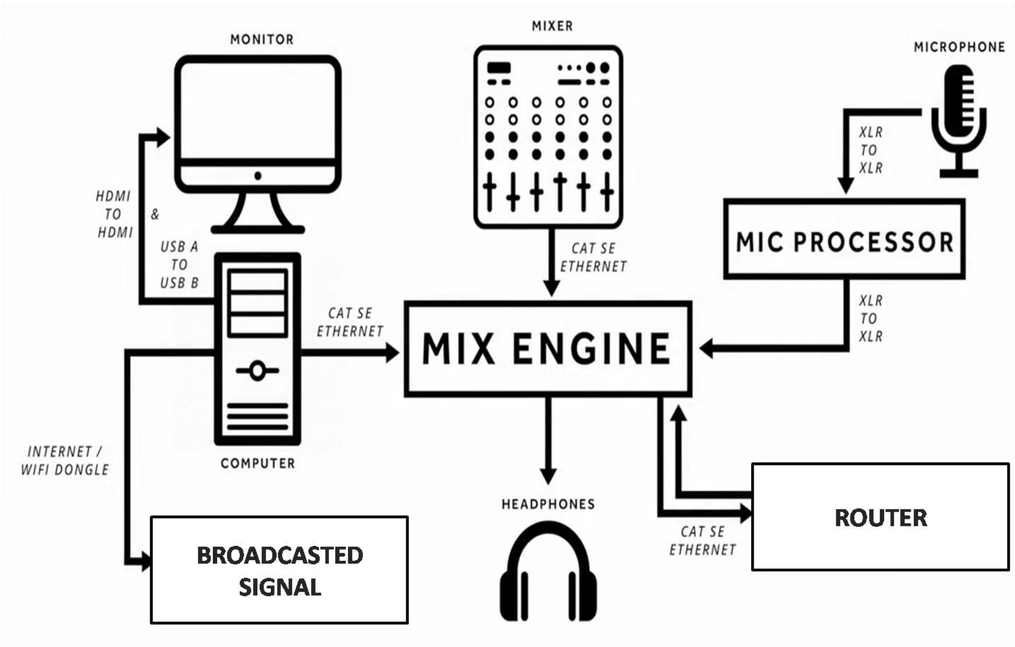

Figure 1: Experimental setup

The figure 1 shows how the main components are interconnected with each other for signal broadcasting. The microphone was connected to the processor which has an analogue to digital converter embed in it, before processing the signal and forward to the mixer engine which employed linear filter for signal processing and then output as the broadcasting signal.

The monitoring PC was installed with HTK speech analyzer software which was used to monitor the data broadcasted via the channels. The model used for the data collection considered Signal to noise ration and noise signal input from the microphone as shown in (Sujan and Kuldip, 2020) and the results presented in table 1;

Table 1: Data Collected from the testbed

Time (s)	Size of voice packet (Kbit)	Noise level (db)
1	1000	0.20
2	1100	0.24
3	1200	0.29
4	1300	0.31
5	1400	0.37
6	1500	0.41
7	1600	0.45
8	1700	0.49
9	1800	0.50
10	1900	0.60
11	2000	0.65
12	2100	0.77
13	2200	0.83
14	2300	0.91
15	2400	0.99
16	2500	1.03
Average	**1750**	**0.565**

The table 1 presented the broadcasting signal collected from the tested for analysis. The result of the table showed that a noise gain of 0.565dB was recorded at an average voice signal of 1750kbit. This research seeks to reduce this noise signal to further optimize the quality of broadcasting signal at the testbed.

IV. PROPOSED SYSTEM

The new system was developed using an adaptive noise cancelling filter which was able to adapt to dynamic frequency response so as to ensuring that the broadcasting signal is filtered against external noise frequencies as it modulates to the receiver destinations irrespective of the broadband outreach. The new system block diagram is presented in figure 2;

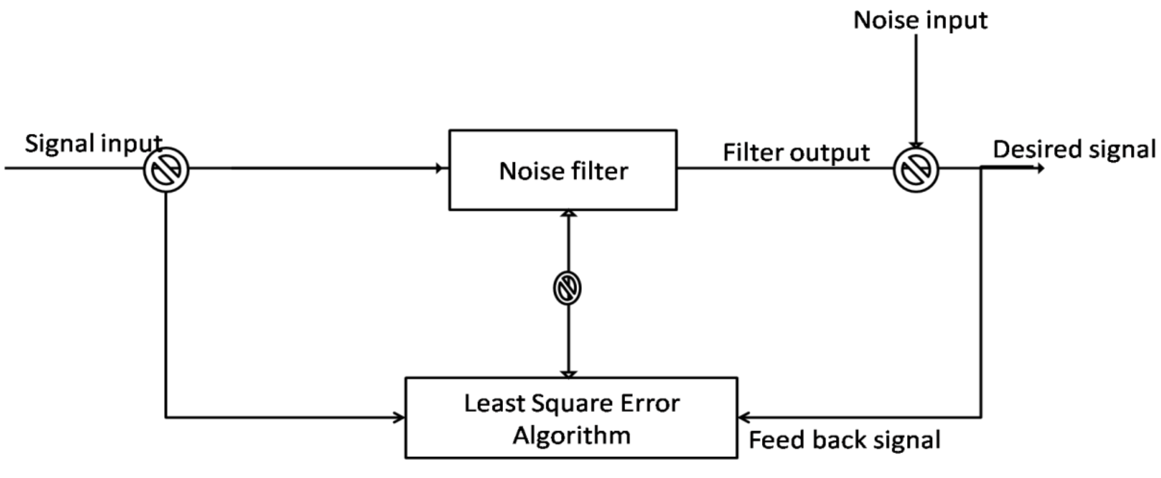

Figure 2: proposed block diagram

From the figure 2, the input signal was processed using the noise filter and then the signal to noise output was feedback to the least square error algorithm for adaptive processing to give out the desired broadcasting signal with limited noise gain.

V. MODELLING OF THE ADAPTIVE NOISE CANCELLER SYSTEM

The adaptive noise canceller was developed as a filter in a closed loop system with the capability to dynamically adjust its error until the desired response was achieved using noise filter and Least Square Error (LSE) algorithm. The LSE was presented in figure 3;

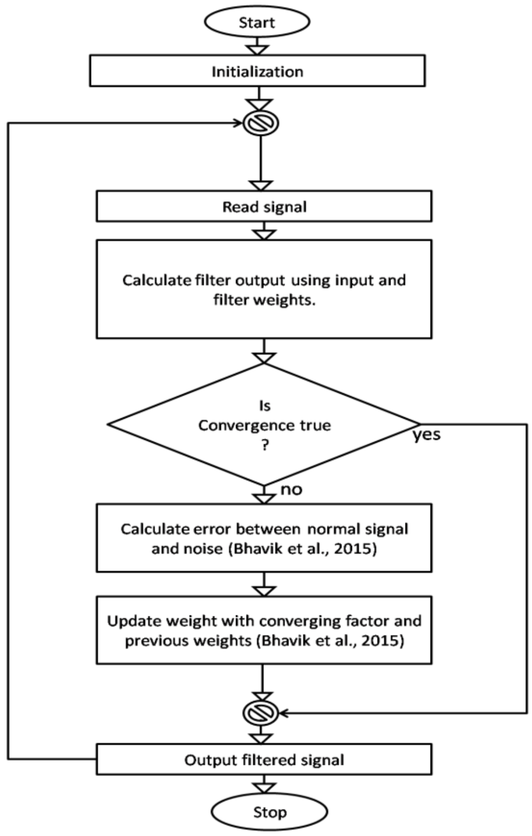

Figure 3: The least square error algorithm flow chart

The least square error (LSE) flow chart in figure 3 was used by the adaptive filter to adjust the weights of the signal with respect to the frequency response using a noise filter. The LSE adaptation algorithm filters the reference input into a replica of the desired input by minimizing residual signal, and when this process is achieved, the output of the filter is estimated as the desired signal to be transmitted. The noise filter is presented in figure 4;

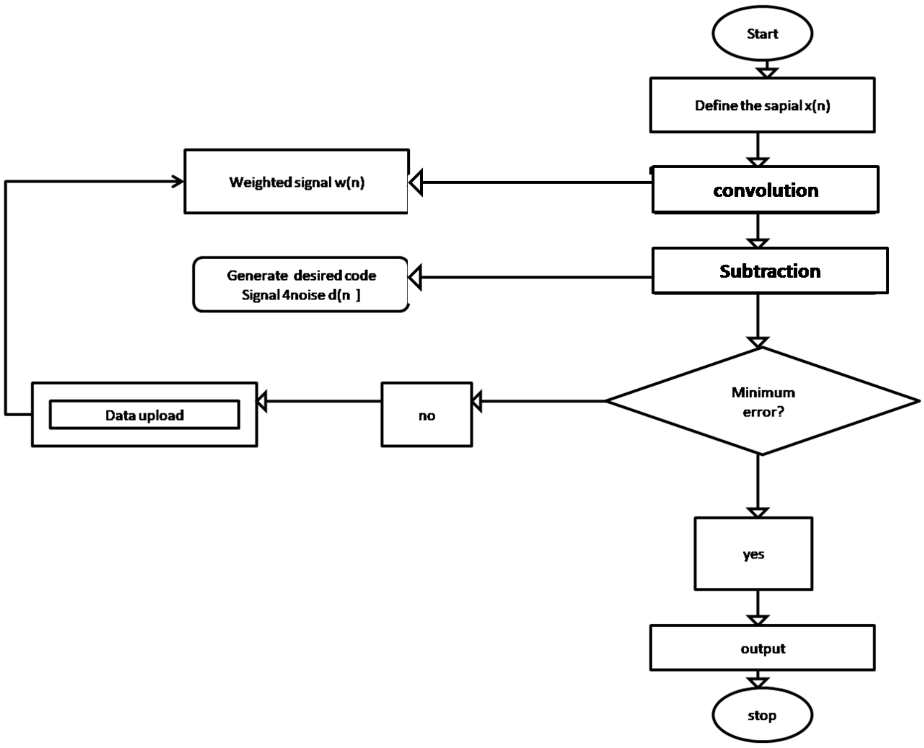

Figure 4: Noise filter

The noise filter was used to eliminate error signal from the voice broadcast made using a convolution of weighted sums determined by the minimization of error signals and achieved a desired signal as shown in the flow chart in figure 5;

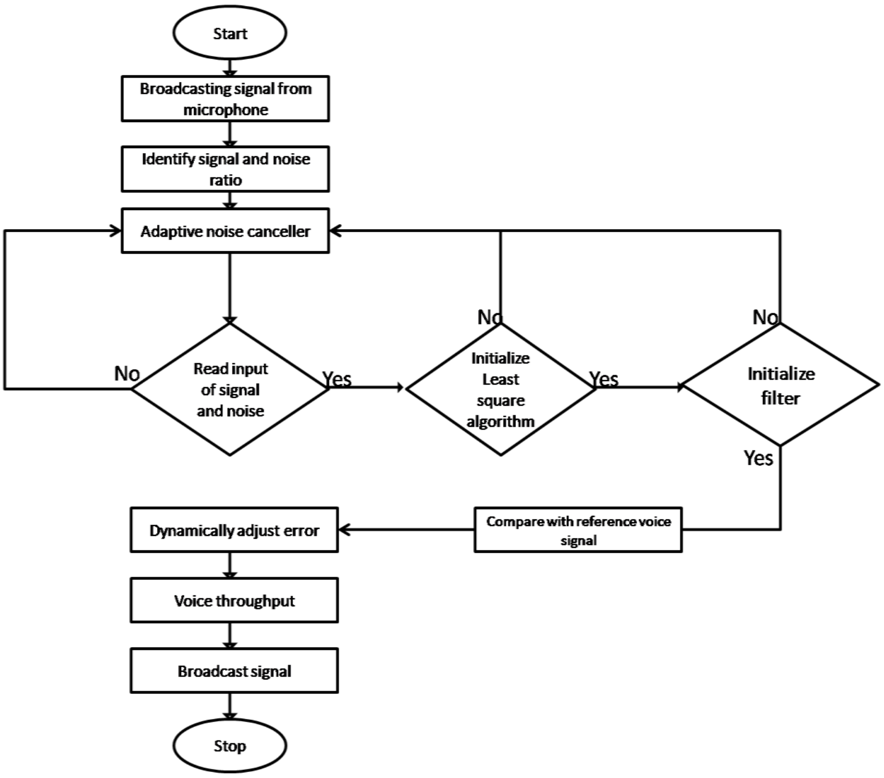

Figure 5: Flow chart for adaptive noise canceller design

The figure 5 presented the data flow chart of the adaptive noise canceller proposed. The system was developed using the LSE algorithm and the noise filter to achieve the desired quality broadcasting signal need for quality of broadcasting services.

VI. IMPLEMENTATION

The implementation of the algorithm was done using Simulink, signal processing toolbox and communication toolbox. The simulink was programmed using the parameters in table 2 which was collected based on the setup of the testbed a used to develop the simulink model in figure 6;

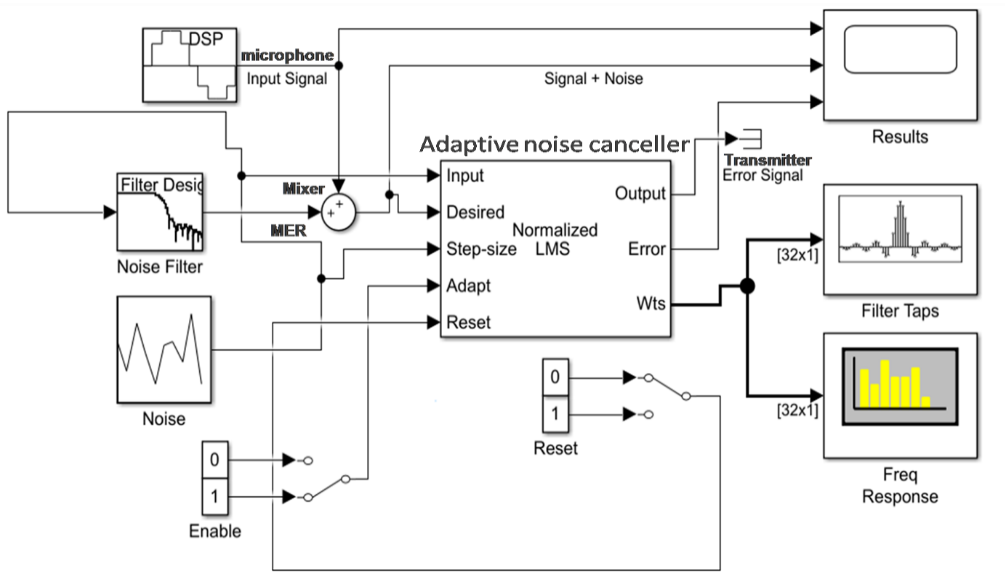

Figure 6: Simulink model of the improved system

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Table 2: Simulation parameters of Dream FM Mixer

Mixer Parameters	Values
Power	25kw
Min packet size	1000kb
The Resolution bandwidth	92.5MHz
Max packet size	2500kb
Packet increase rate	100kb/s
Packet type	Audio

VII. RESULTS AND DISCUSSIONS

The result presented the performance of the simulated testbed using the simulation parameters and the signal to noise ratio model in (Sujan and Kuldip, 2020) to presented the quality of signal input from the mixer without the adaptive filter developed as shown in the figure 7;

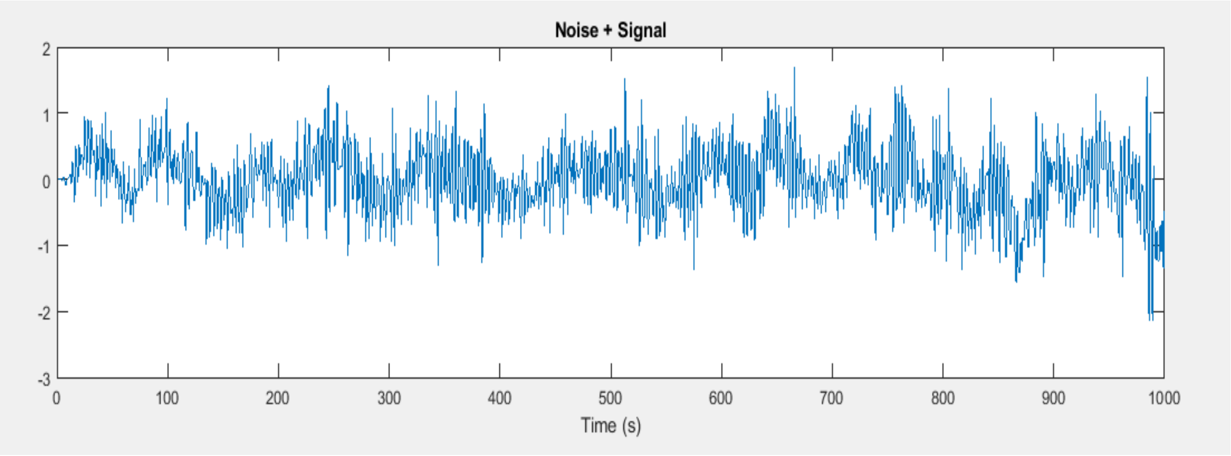

Figure 7: Signal to Noise ratio Result of the Input Broadcasting Signal

The figure 7 presented the sinusoidal wave form of the Signal to Noise Ratio (SNR) in the input broadcasting data from the microphone. The result showed that the signal contains noise and required an adaptive filter to improve the service quality. This was achieved using the adaptive noise canceller developed with the least square and noise filter in figure to process the SNR and output a desired quality of service as in the figure 8;

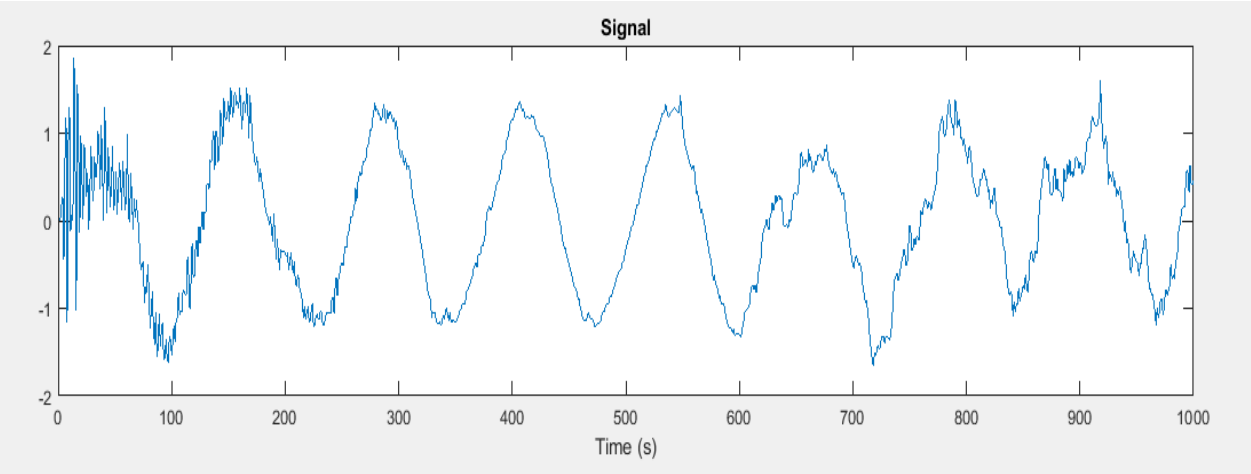

Figure 8: Processed Signal output

The figure 8 presented the result of the processed signal with the adaptive noise canceller developed. The noise canceller feedback the SNR to the LSA for update with the weight and convergence factor as modeled in figure 3 to mitigate the noise from the signal and improved quality of broadcasting output. The level of signal to noise ratio measured from the signal was presented in the figure 9;

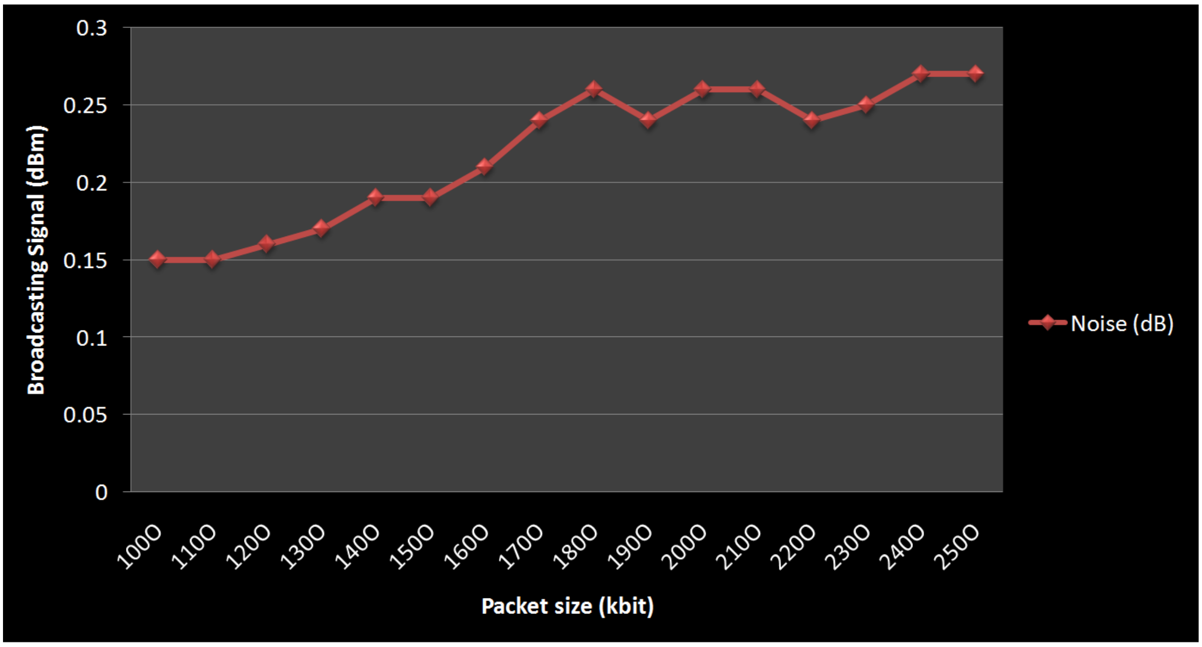

Figure 9: The noise gain filtered

The figure 9 presented the average noise gain filtered out of the SNR from the acoustic noise canceller developed. The result showed that the average noise ratio is 0.22dB as against 0.56dB in the empirical study performed earlier. The implication of the result showed that the adaptive noise canceller was able to mitigate the noise from the SNR and achieved quality of service output.

VIII. CONCLUSION

This work has successfully improved the quality of broadcasting signal in radio studio using an adaptive noise canceller. The noise cancelling device was developed as an adaptive filter using the least mean square algorithm and noise filter to eliminate noise to the lowest level with respect to a reference sampled signal. The work was implemented with Matlab and tested. The result showed that the adaptive filter was able to mitigate noise in the signal from 0.56dB to 0.22dB.

CONTRIBUTION TO KNOWLEDGE

- The research developed an adaptive noise canceller filter for the Dream FM radio studio.

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