FREQUENCY-DOMAIN ADAPTIVE NOISE CANCELLATION

RESULTS

1. Calculating MU

In order to determine the value of MU, we run a MATLAB code where we call the function used in the original program.

The original function is called to obtain the SNR values before and after ANC is applied to it. The difference of the two will give us the SNR Improvement. The code also specifies a given set of iterations and iteration number. The product of the two values will give us the range of MU values we are considering. Within this range lies the best MU value, which corresponds to the highest SNR Improvement.

Figures 1 and 2 show the Best Mu value along the X-axis and the corresponding highest SNR Improvement along the Y-axis for N=4.

Random Noise

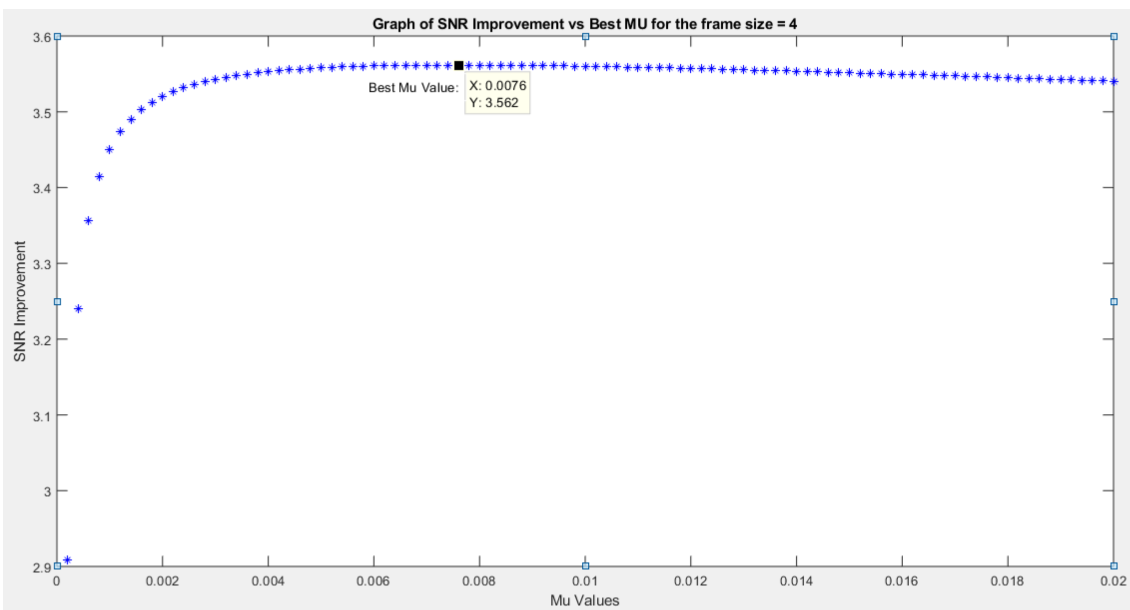


Figure 1

Background Music

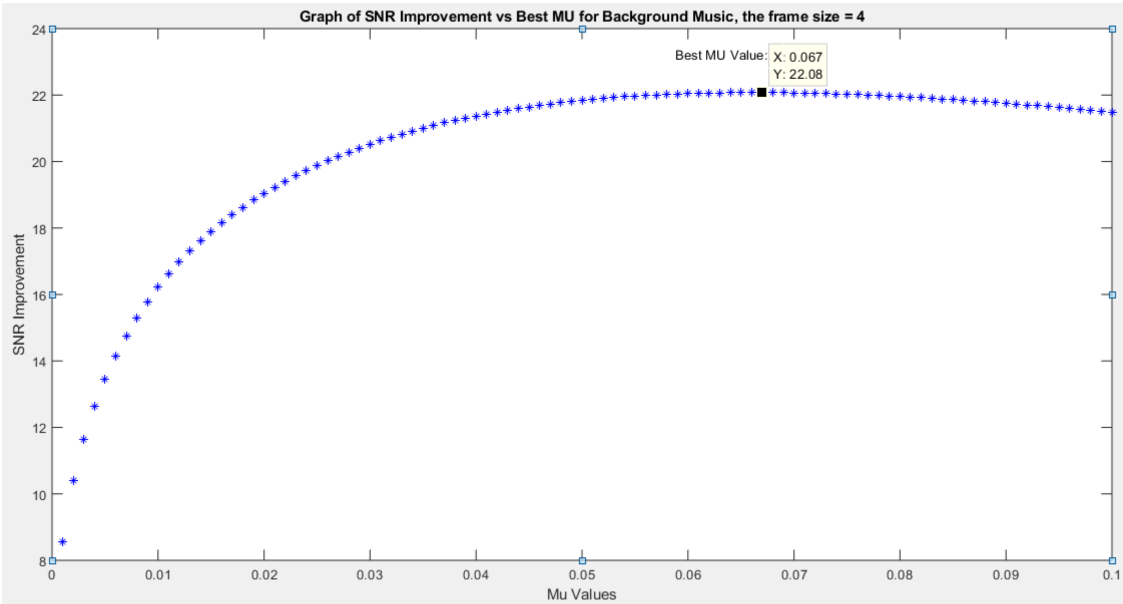


Figure 2

2. SNR Calculation

A subjective measure, which is also called Mean opinion Score (MOS) is a scale to measure the quality of the output based on the judgement of the listener. The range is from 1 to 5, 1 being extremely poor to 5 being excellent.

The improvement in SNR is determined by:

$$SNR_{dB}^{\mathbf{Improvement}} = SNR_{dB}^{After} - SNR_{dB}^{Before}$$

The results are tabulated as follows:

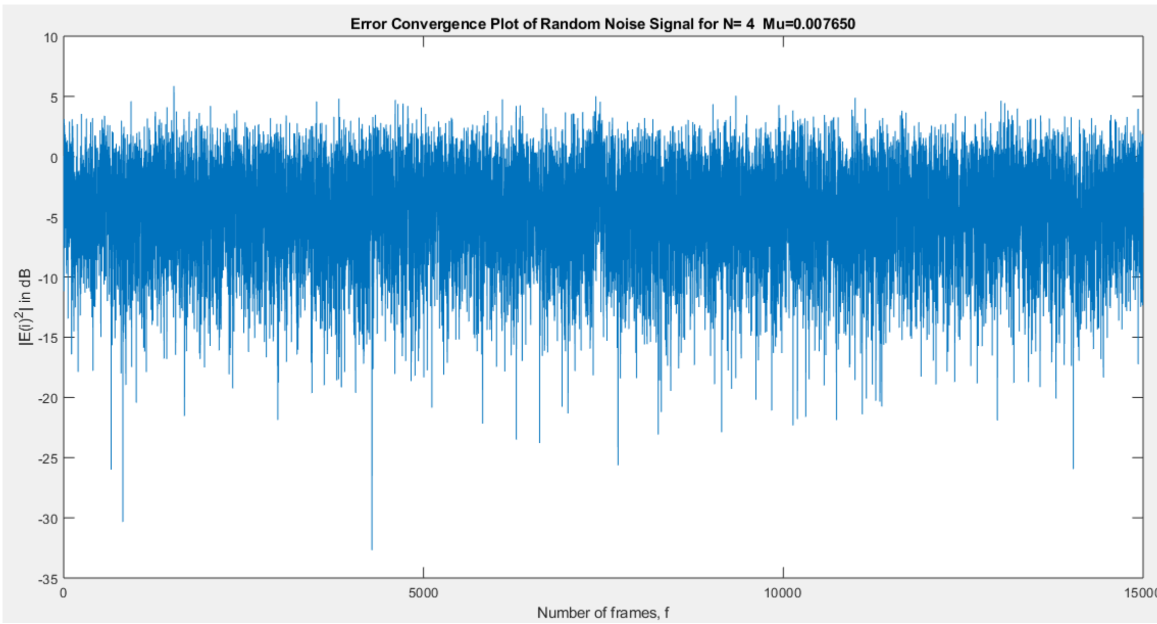
RANDOM NOISE			
N	**SNR_{imp} (dB)**	**Subjective Rating 1-5**	**Best µ**
4	3.562	2	0.00765
16	6.594	2.5	0.00587
64	10.61	3	0.00445
128	12.18	3	0.004
256	12.79	3.5	0.0025

BACKGROUND MUSIC			
N	**SNR_{imp} (dB)**	**Subjective Rating 1-5**	**Best µ**
4	22.08	5	0.067
16	18.47	4.5	0.026
64	13.3	3.5	0.00518
128	11.81	3	0.0032
256	11.17	2.5	0.00254

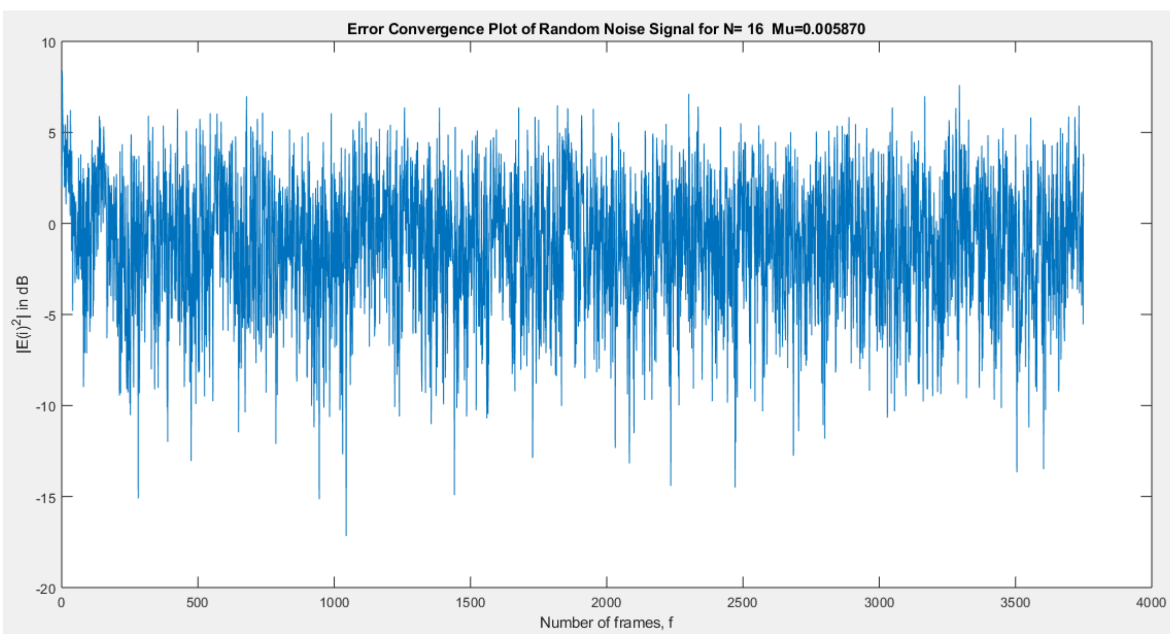
3. ERROR CONVERGENCE CURVES

3.1 RANDOM NOISE

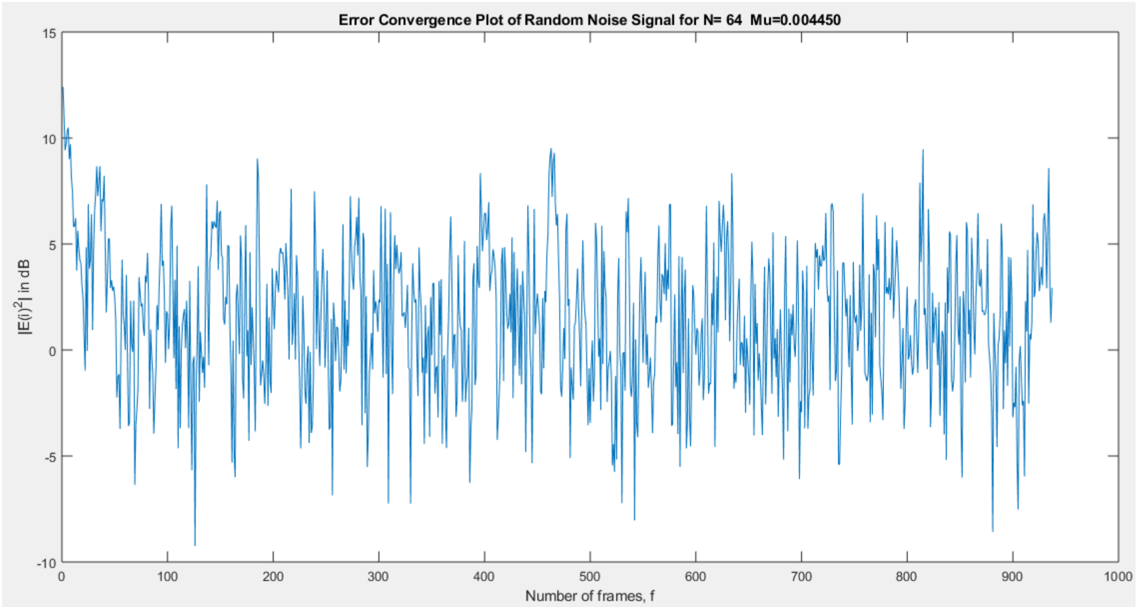
N=4 and mu=0.007650



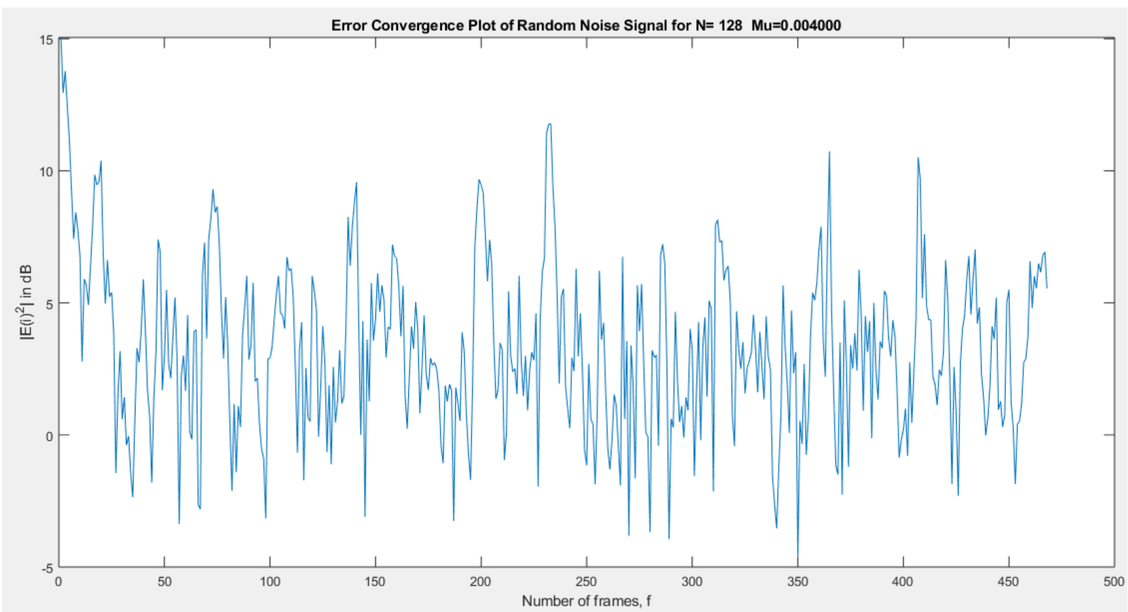
N=16 and mu=0.00587



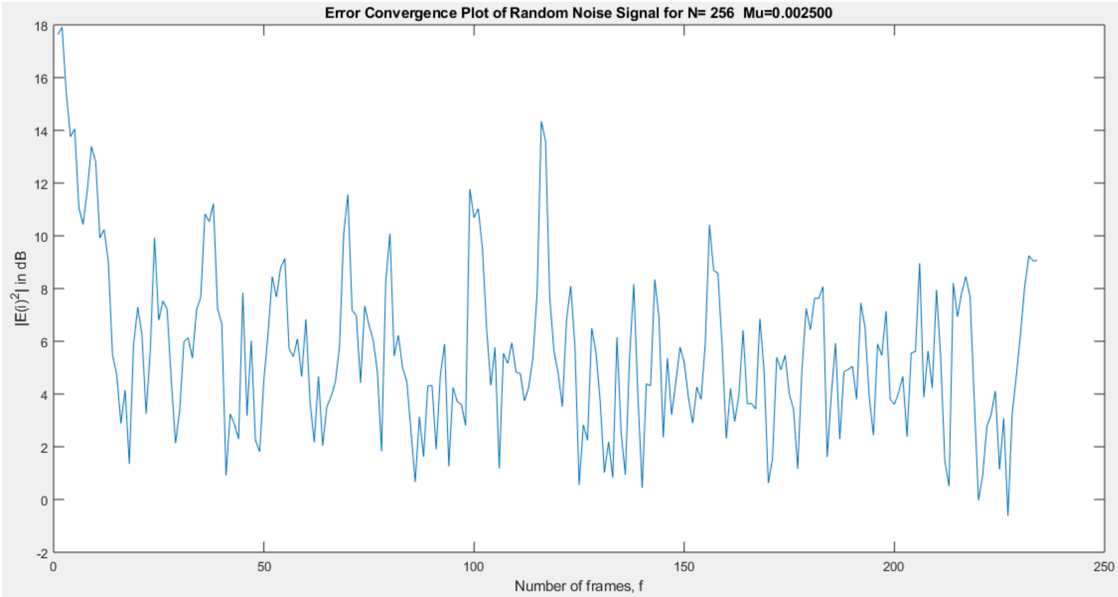
N=64 and mu=0.00445



N=128 and mu=0.004



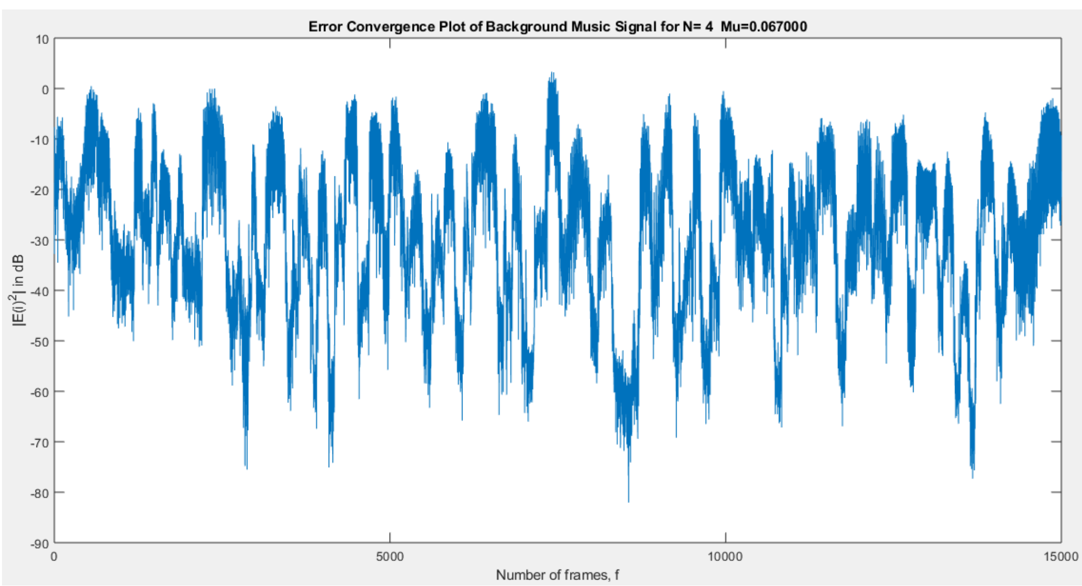
N=256 and mu=0.0025



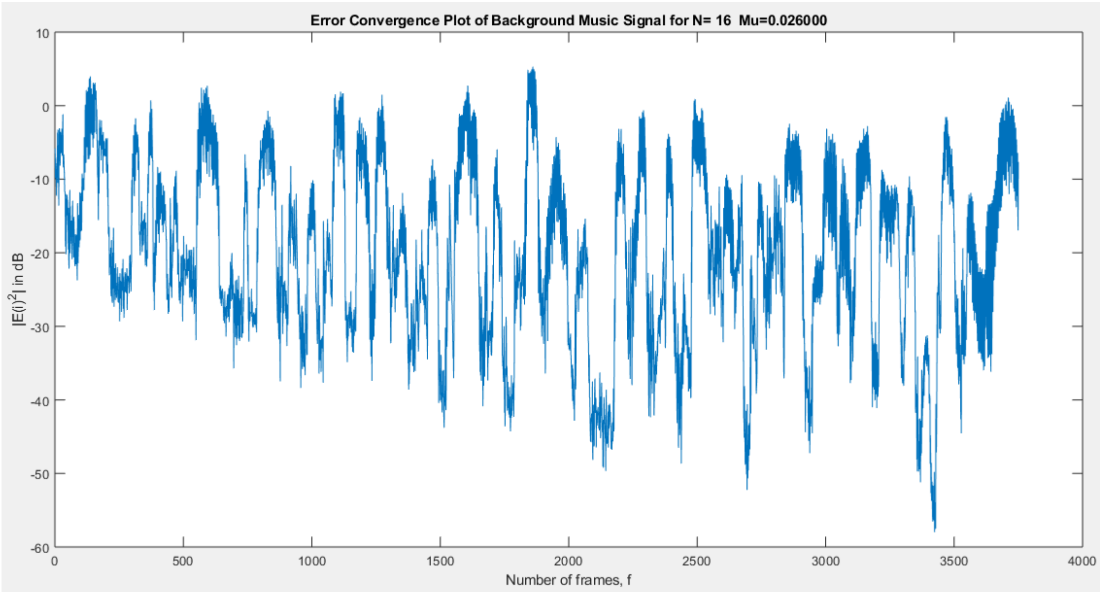
3.2 BACKGROUND MUSIC

Noise is easier to eliminate than music because noise signals exhibit large variations in values, which are easier for a filter to pick up and attenuate. Background music is similar to the original signal itself, making it harder to discern.

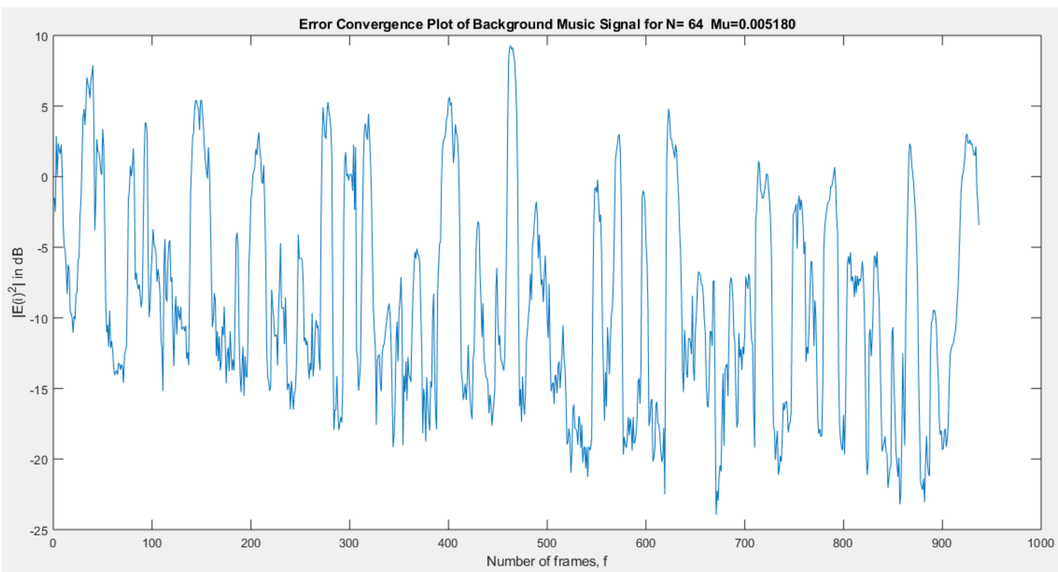
N=4 and mu=0.067



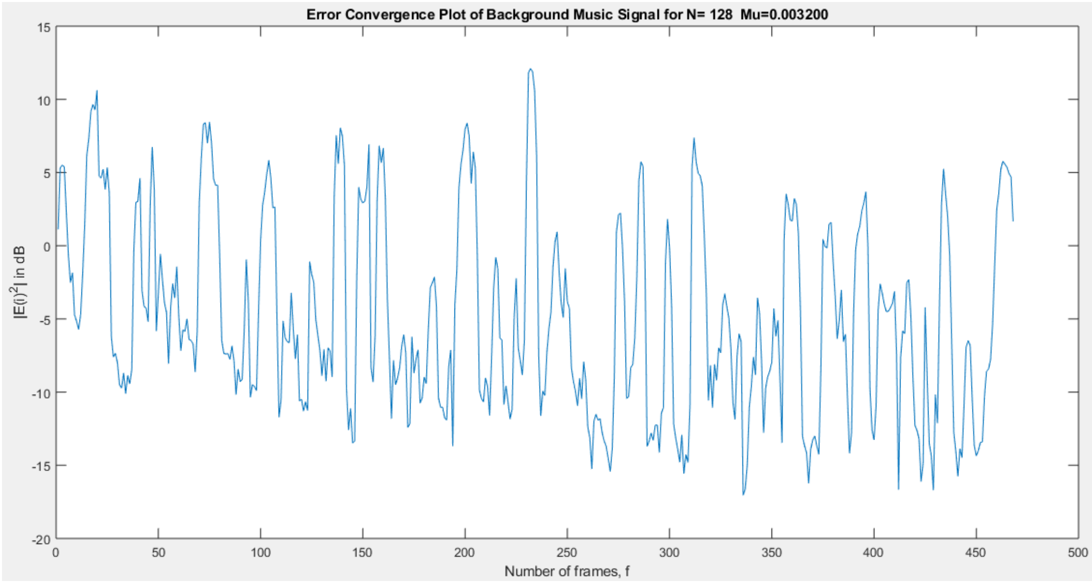
N=16 and mu=0.026



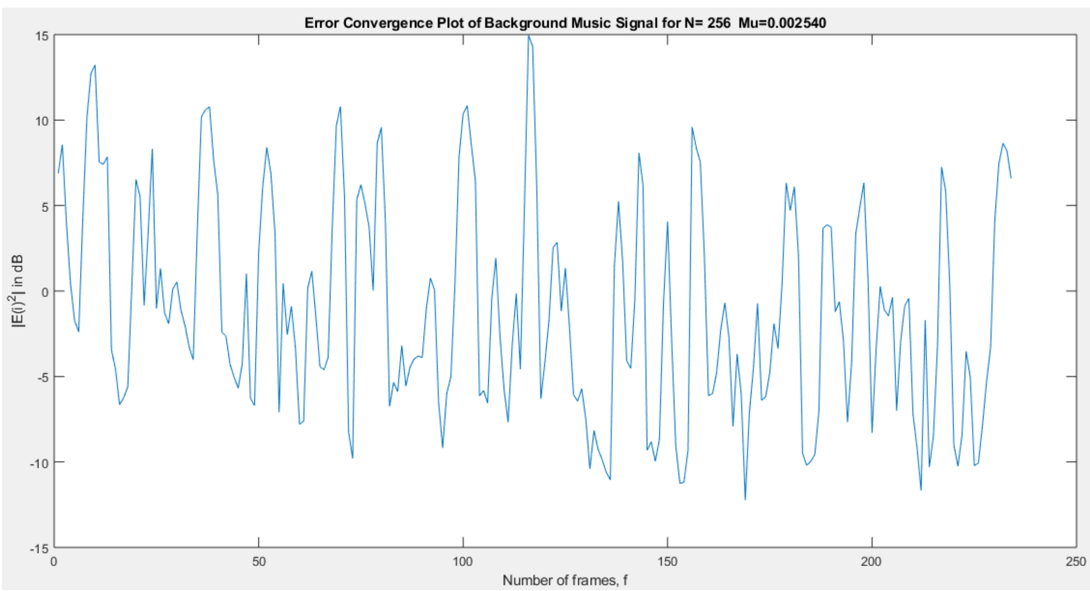
N=64 and mu=0.00518



N=128 and mu=0.0032



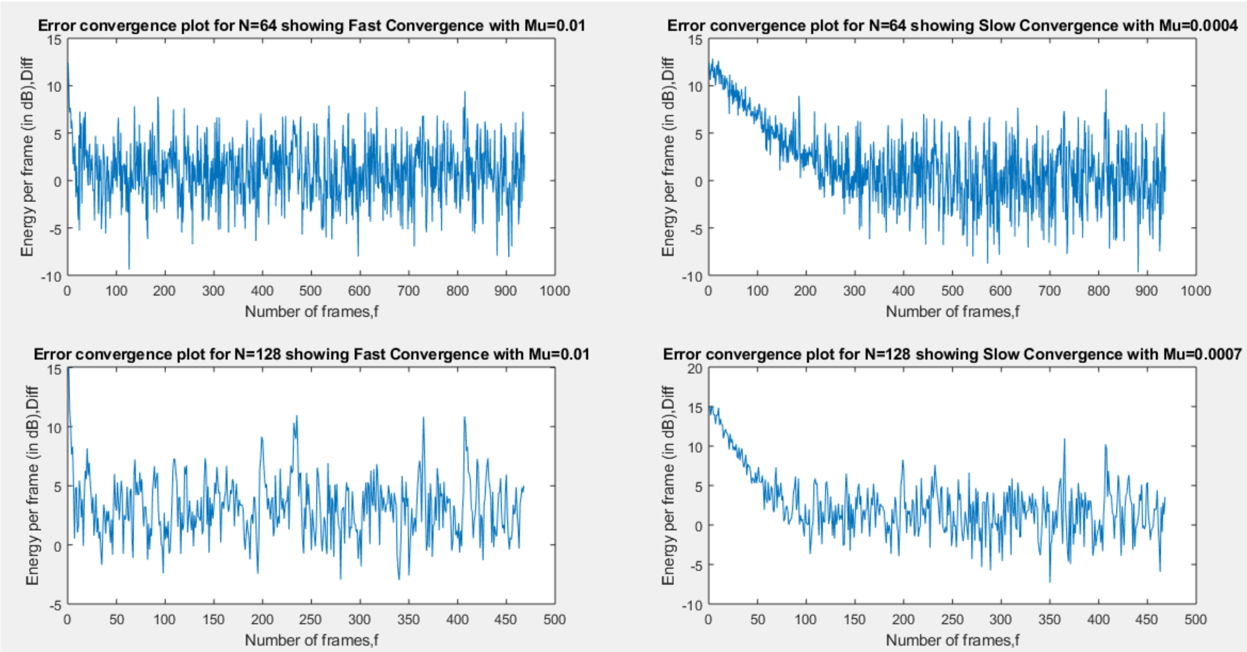
N=256 and mu=0.00254



4. CONVERGENCE CURVES FOR RANDOM NOISE

The purpose of plotting convergence curves for a randomly selected (fast) high Mu value and (slow) low Mu value is to examine the output graphs for these values. For a high Mu value = 0.01, we can see that convergence occurs almost immediately, but energy per frame of this graph exhibits more fluctuations when compared to the same frame size, but with the best Mu value.

For slow convergence, we can see that while convergence occurs and the mean value is fairly constant, the adaptation time is significantly longer, which is disadvantageous.



CONCLUSION

This project gives a fundamental understanding of Adaptive Noise Cancellation and how to employ FFT in a real-world application. The reason we use FFT is because the ANC requires high order FIR filters, and FFT can perform fast convolutions. Additionally, frequency domain allows for simpler computation as it performs multiplication unlike time domain which performs convolution. Frequency domain representation of input signals is obtained using the FFT Function. We then use an LMS Algorithm which can perform adaptive operations in the frequency domain.

Mu represents the convergence coefficient which is used to modify the weights of the channel filter to produce more accurate values. Mu controls the rate of convergence for the adaptive filter. For large Mu values, gradient estimate becomes a determining factor on which weight change depends. It might also result in instability of the algorithm and the frequency domain filter coefficients can become unbounded (overflow). On the other hand, if Mu is chosen to be too small, time to converge to the optimal weights will be too large. A large value of mu will give an output that is moderate in quality as all the noise elements are not removed (owing to faster convergence time). A smaller Mu value might have a longer convergence time, but the output is of improved quality as a greater number of noise samples are eliminated.

From the tabulations, it is observed that SNR increases as N increases for Random noise, and SNR decreases when N increases for Background Music. Noise is easier to remove than background music because background music has a lower convergence rate and hence requires greater number of iterations to converge.

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