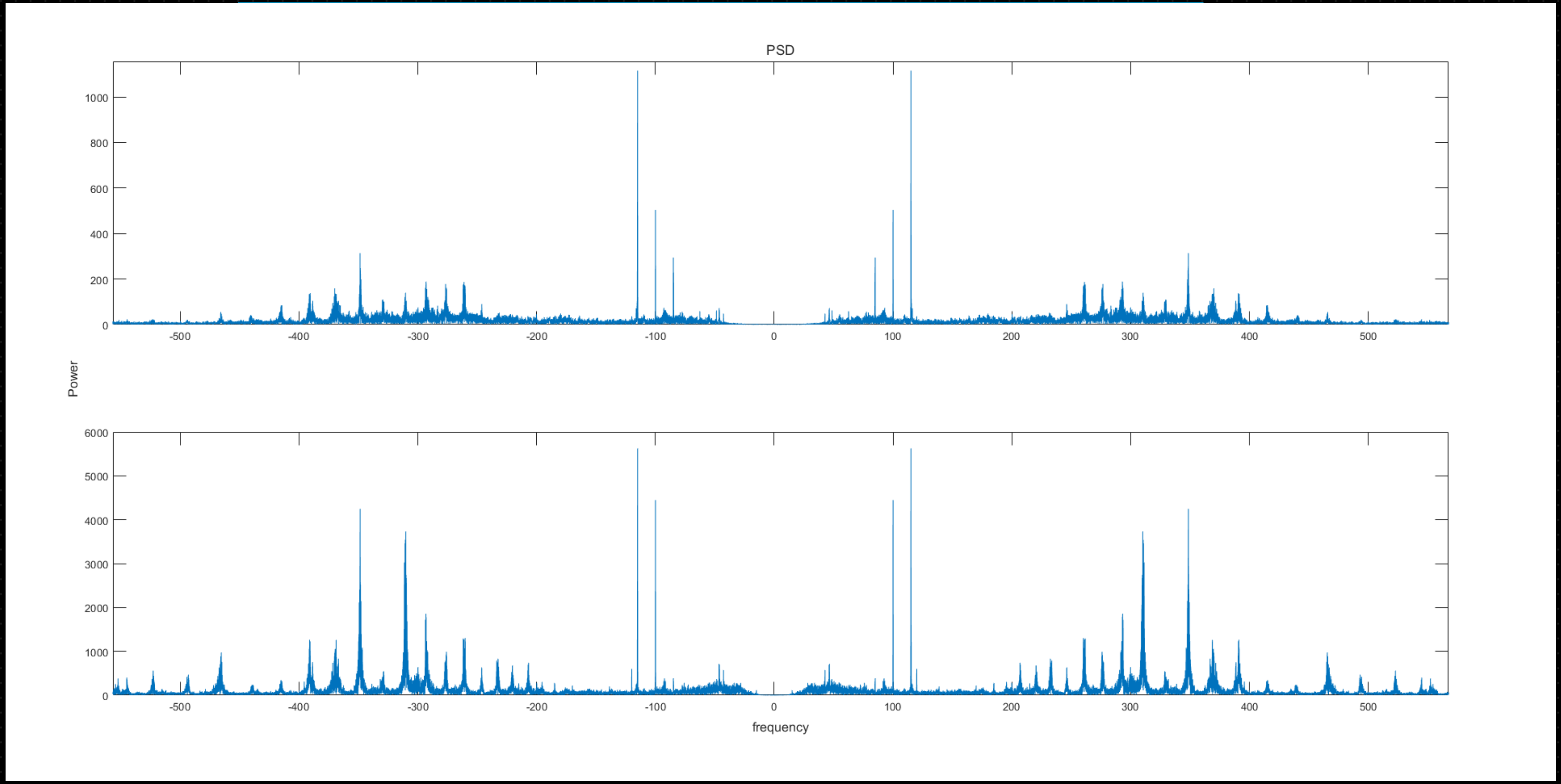
Adaptive Noise Cancellation Using LMS Algorithms

Wiwitthawin Chareonngam 6310501933 Kasetsart University

How can noise which comes along with the wanted signal be canceled?

1. Use Linear Filter



How can noise which comes along with the wanted signal be canceled?

2. Use Adaptive Filter (Nonlinear)

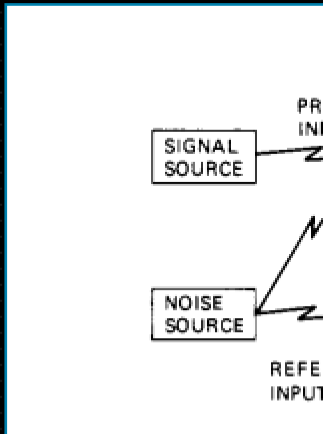
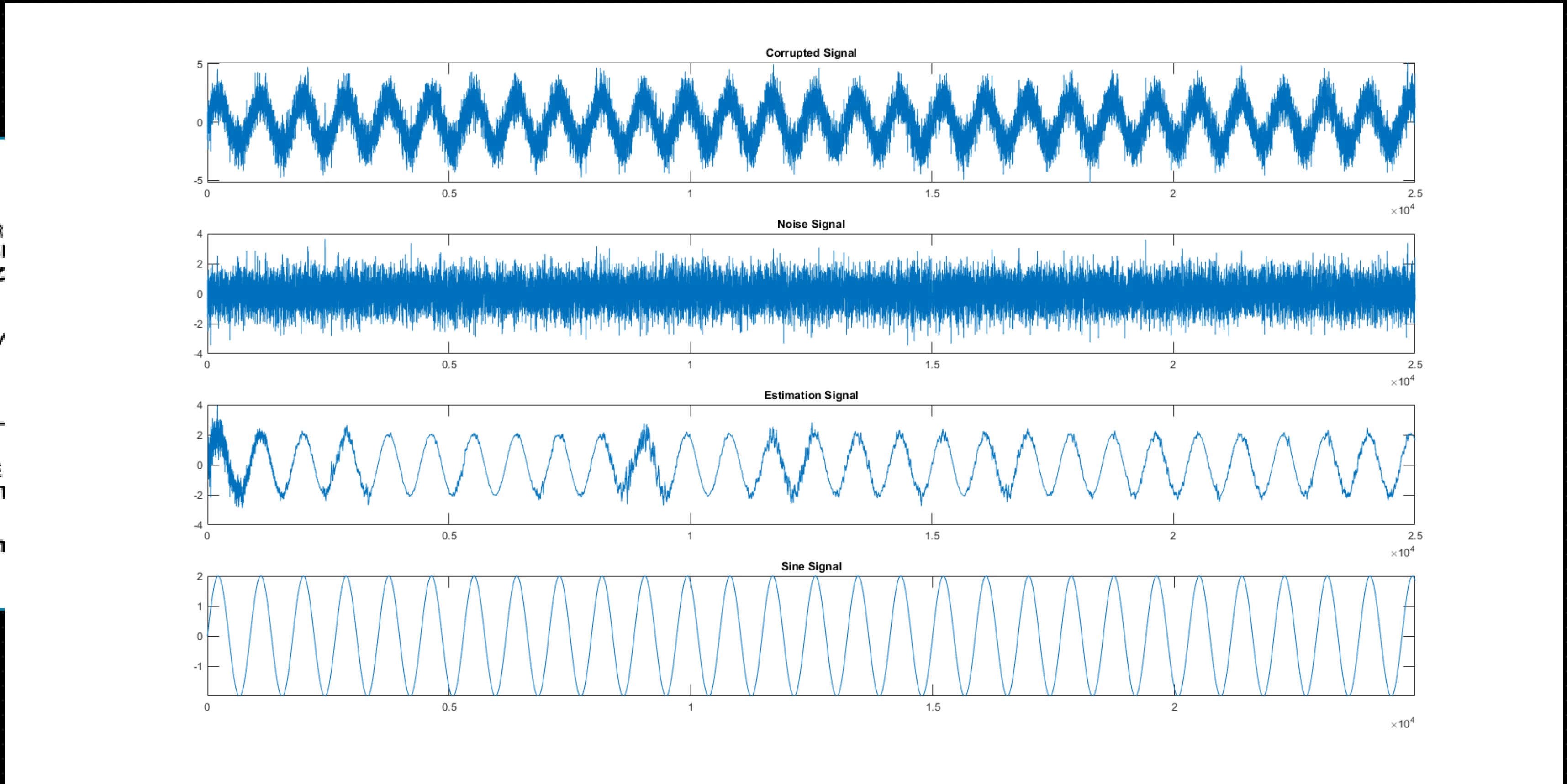
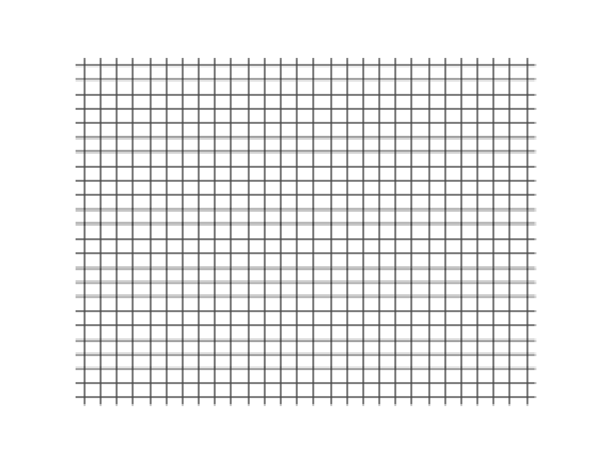


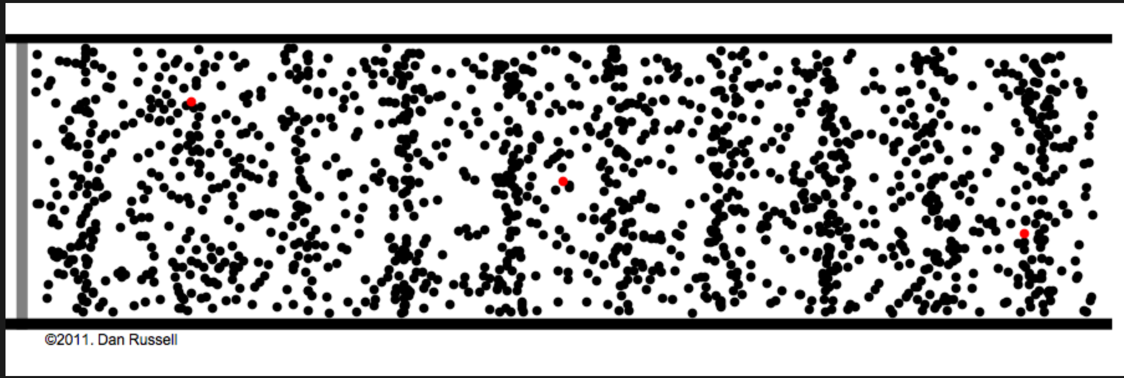
Fig. 1. Th



Sound Wave

- In this Project, we focus on a sound wave that is generated from an object’s vibration through air molecules to a receiver such as an ear or microphone.
- Sound wave is a longitudinal wave or compression wave that carries pressure and velocity through a medium.
- Noise is an unwanted signal such as background noise.





Other’s Research

- Adaptive Noise Cancelling is first published from B. Widrow *et al*., "Adaptive noise cancelling: Principles and applications," in *Proceedings of the IEEE*, vol. 63, no. 12, pp. 1692-1716, Dec. 1975, doi: 10.1109/PROC.1975.10036. with application in electrocardiography, ECG and speech.

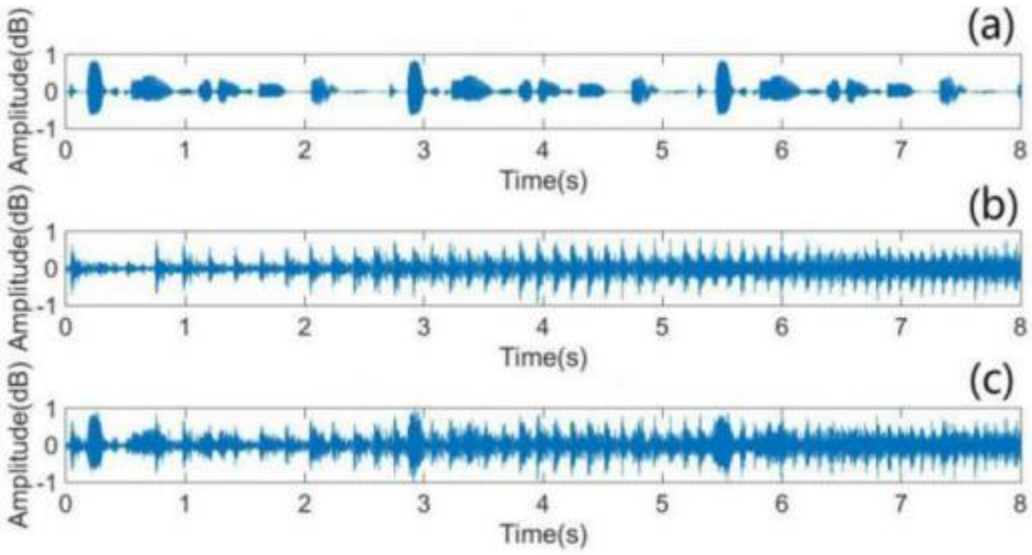


Figure 3(a). Noise-free speech signal **Figure 3(b).** Engine noise signal and **Figure 3(c).** Noise corrupted speech signal.

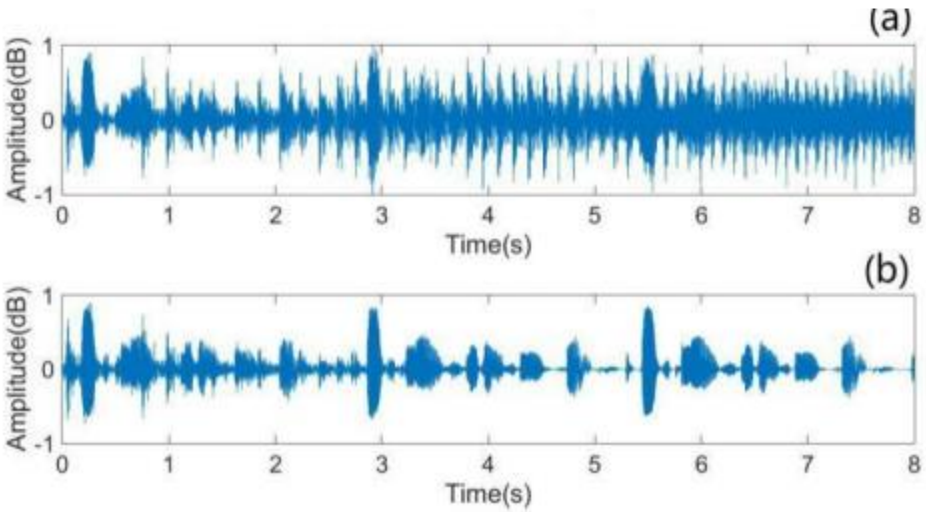


Figure 4(a). Noise corrupted speech signal and **Figure 4(b).** Filtered output.

Collects Signal

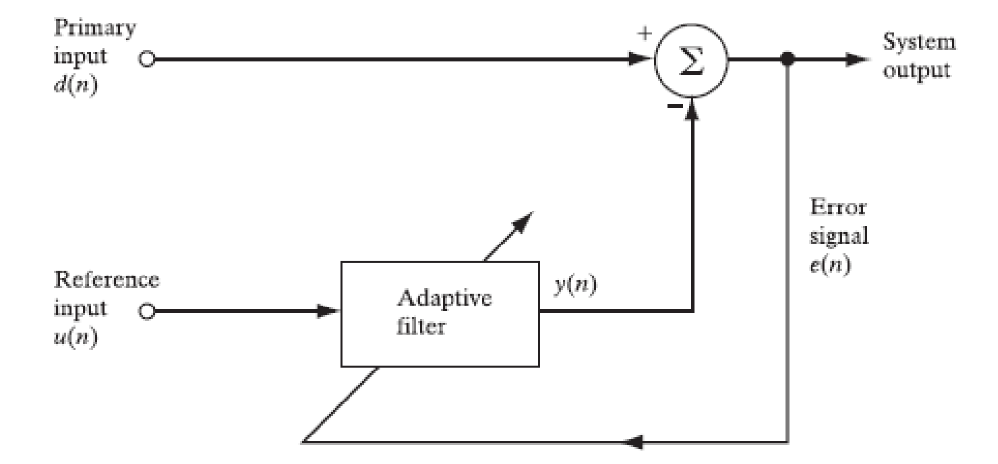


FIGURE 6.6 Block diagram of adaptive noise canceller.

- Signal is collected with 2 microphones; Primary input collects the corrupted sound signal and Reference input collects noise signal and 2 speakers; Noise speaker and sound speaker.
- Signal is collected 3 times for each experiment with a 44.1 kHz sampling rate from a 5x5 m^2 close room.

Collects Signal

Step 1 → Step 2 → Step 3 → Step 4

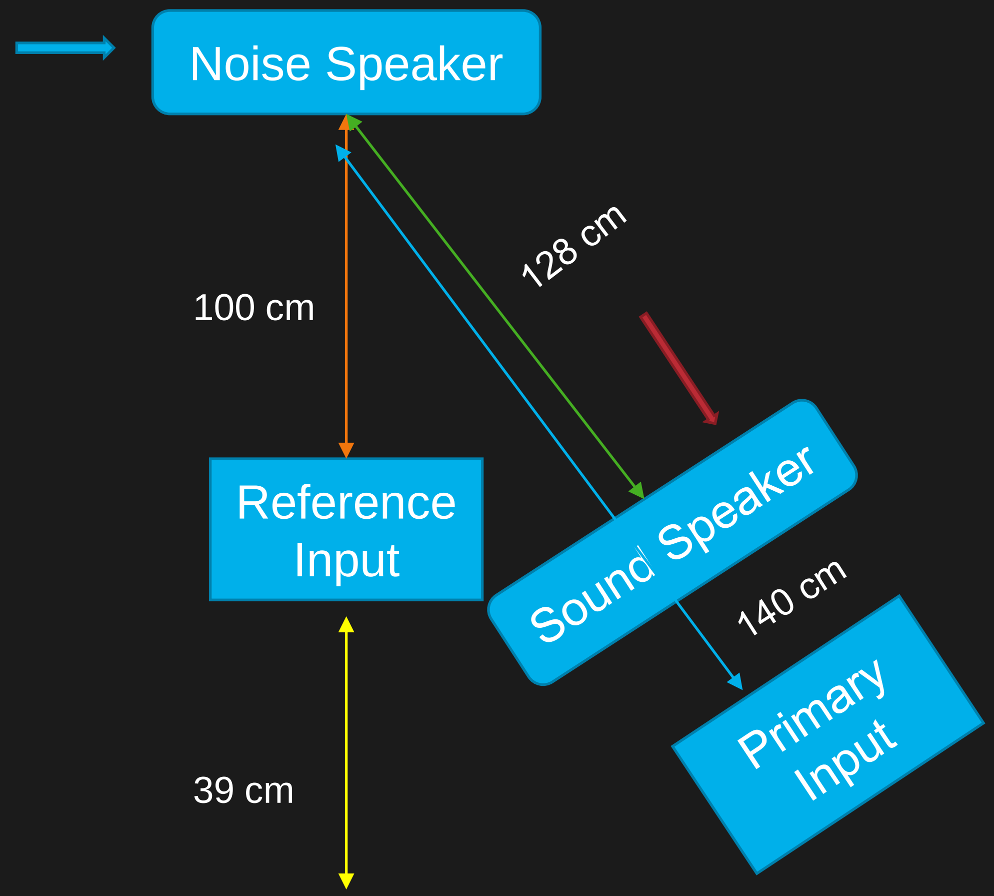
Step 1
- Set up Experiment

Step 2
- Collect Signal from phone’s microphone

Step 3
- Covert from .m4a to .wav with 44.1 kHz

Step 4
- Analyze Signal

Collects Signal



- Simulation picture of experiment setup

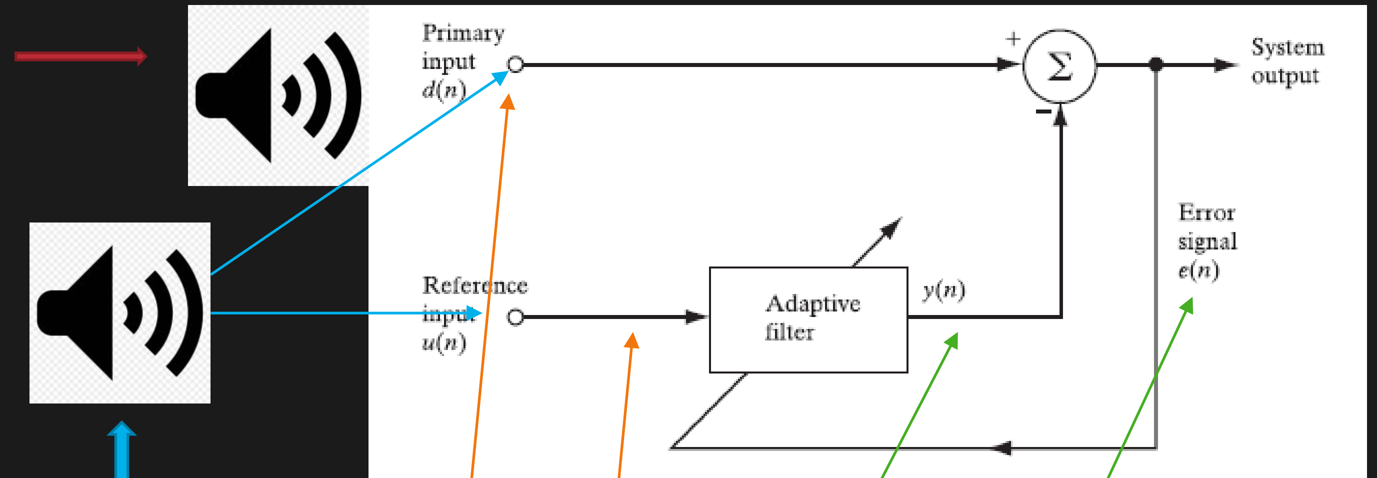


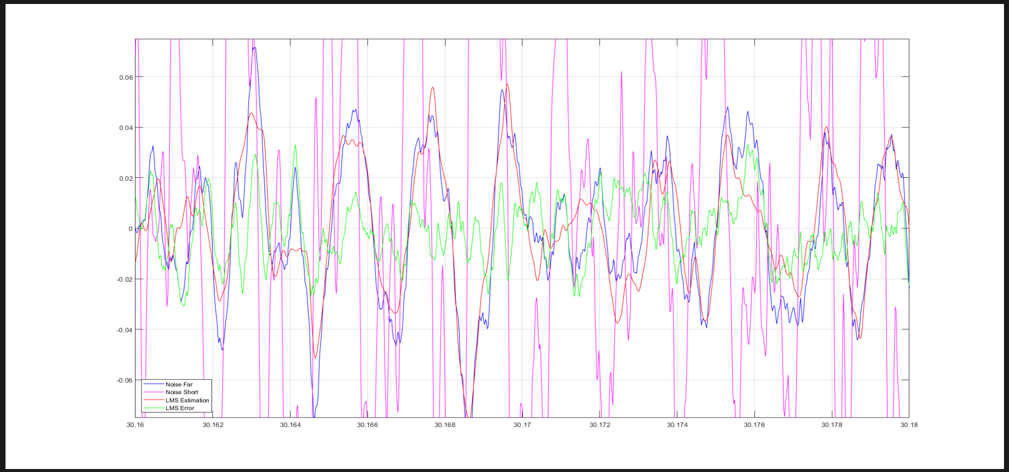
FIGURE 6.6 Block diagram of adaptive noise canceller.

Noise Far = Primary Input
Noise Short = Reference Input

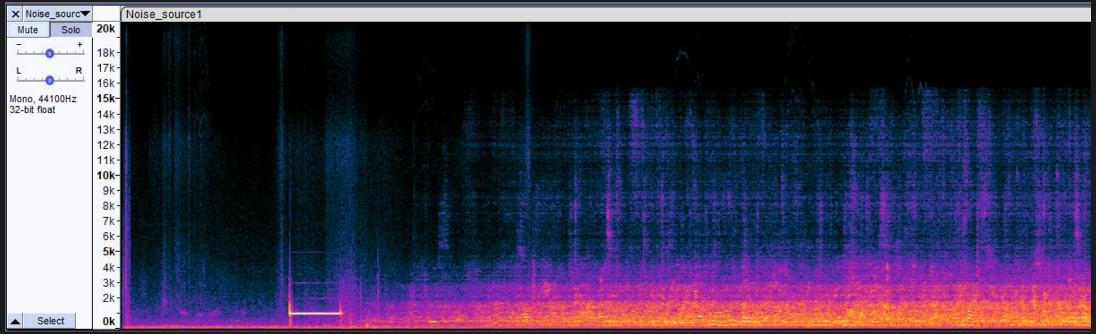
LMS Estimator = Filter Output
LMS Error = Error Signal

Signal Analysis

- Time-domain Analysis: Use in Adaptive Filter Design and to analysis output from LMS filter and correlate 2 signals.



- Time-Frequency domain Analysis: Use to analysis corrupted sound signal, noise signal, LMS Output Signal and LMS Error Signal.



Hypothesis

- If we increase filter length, LMS Estimator can estimate closer to a primary input
- If we increase step size, it can result in underfit and result in overfit for vice versa.
- Noise is reduced from Primary Input

Analysis

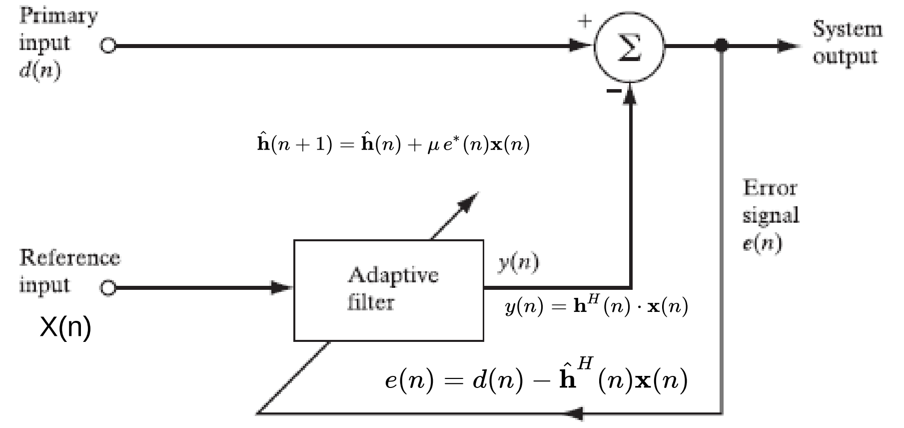


FIGURE 6.6 Block diagram of adaptive noise canceller.

- Least Mean Square (LMS) Algorithms based on stochastic gradient descent are used to adapt coefficient inside adaptive filter to produce least mean square of error signal.

p = filter order, μ = step size, n = 0,1,2, …

$$\hat{\mathbf{h}}(0) = \mathrm{zeros}(p)$$

$$\mathbf{x}(n) = [x(n), x(n-1), \dots, x(n-p+1)]^T$$

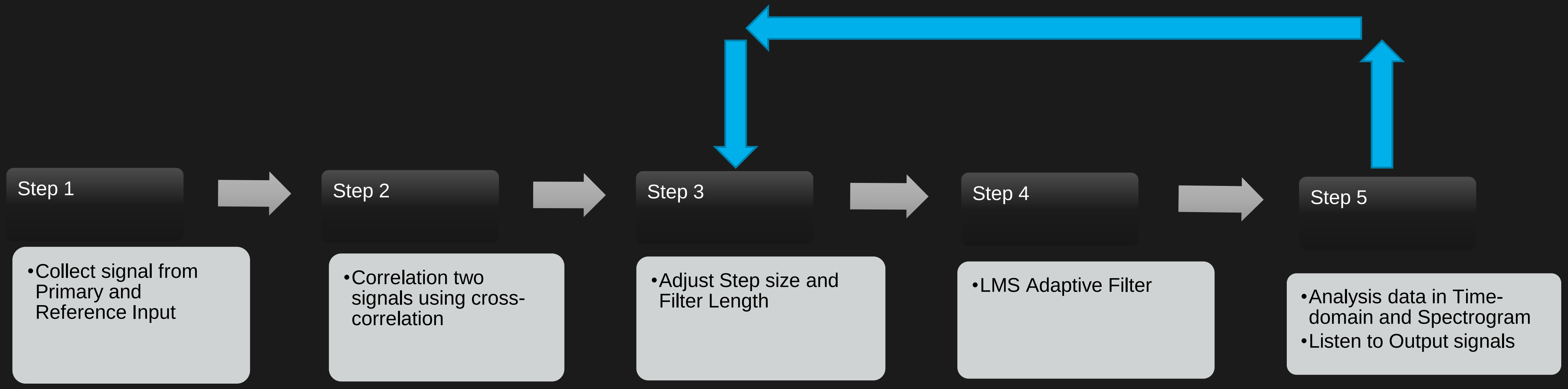
$$y(n) = \mathbf{h}^H(n) \cdot \mathbf{x}(n)$$

$$e(n) = d(n) - \hat{\mathbf{h}}^H(n)\mathbf{x}(n)$$

$$\hat{\mathbf{h}}(n+1) = \hat{\mathbf{h}}(n) + \mu\, e^*(n)\mathbf{x}(n)$$

Analysis

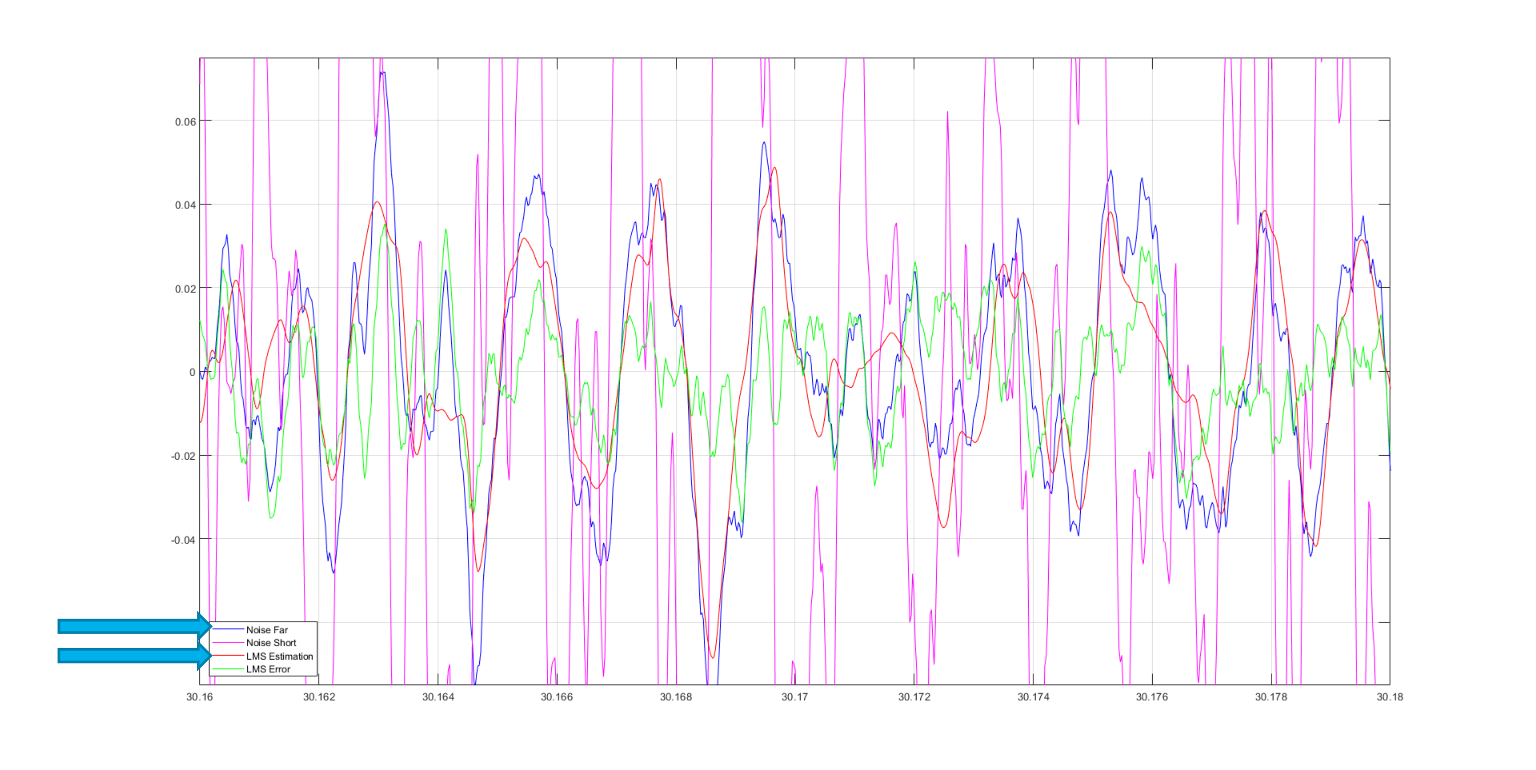
Workflow



Analysis

Experiment 1: Only Noise signal

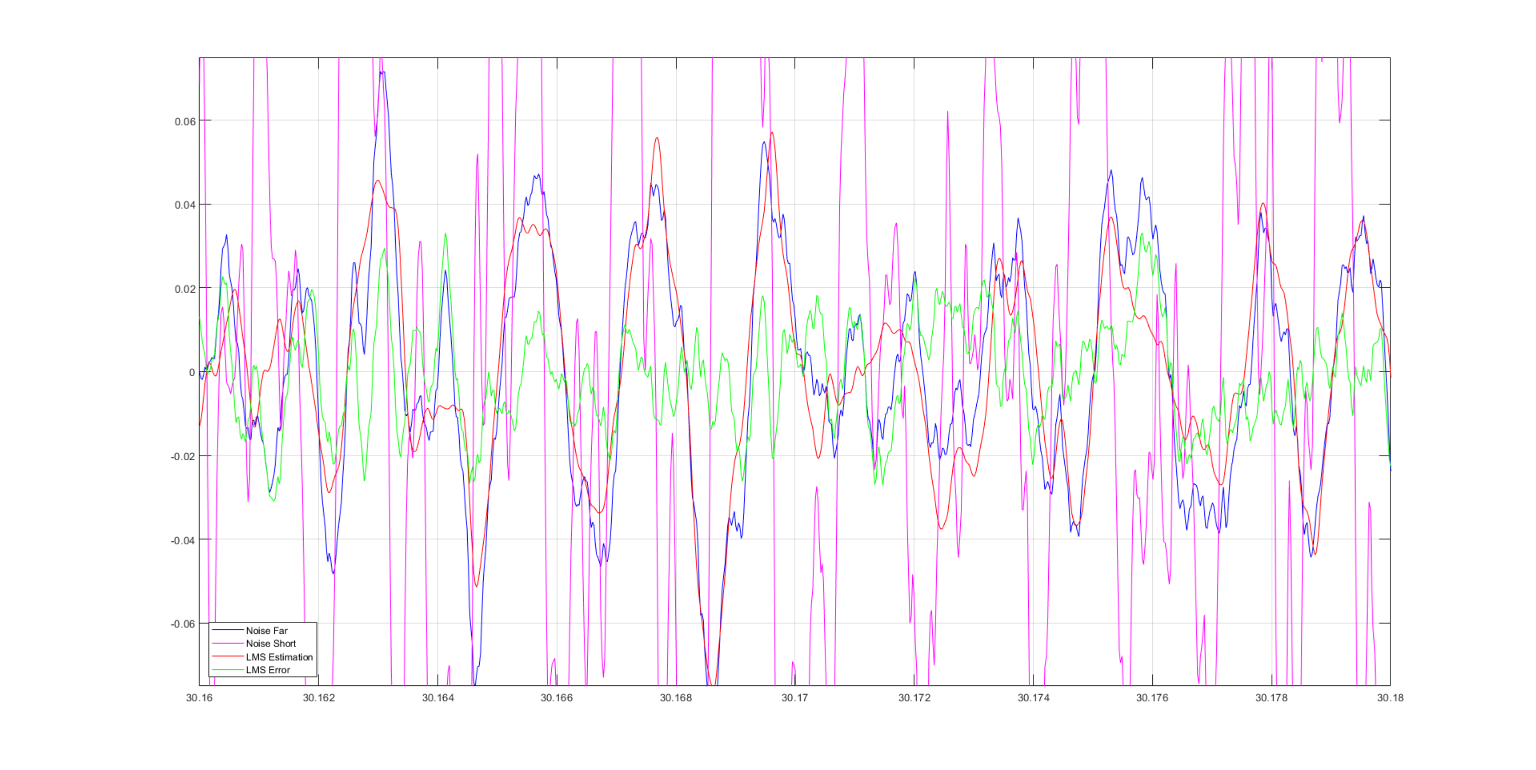
- Filter length = 1500 μ = 0.0002



Analysis

Experiment 1: Only Noise signal

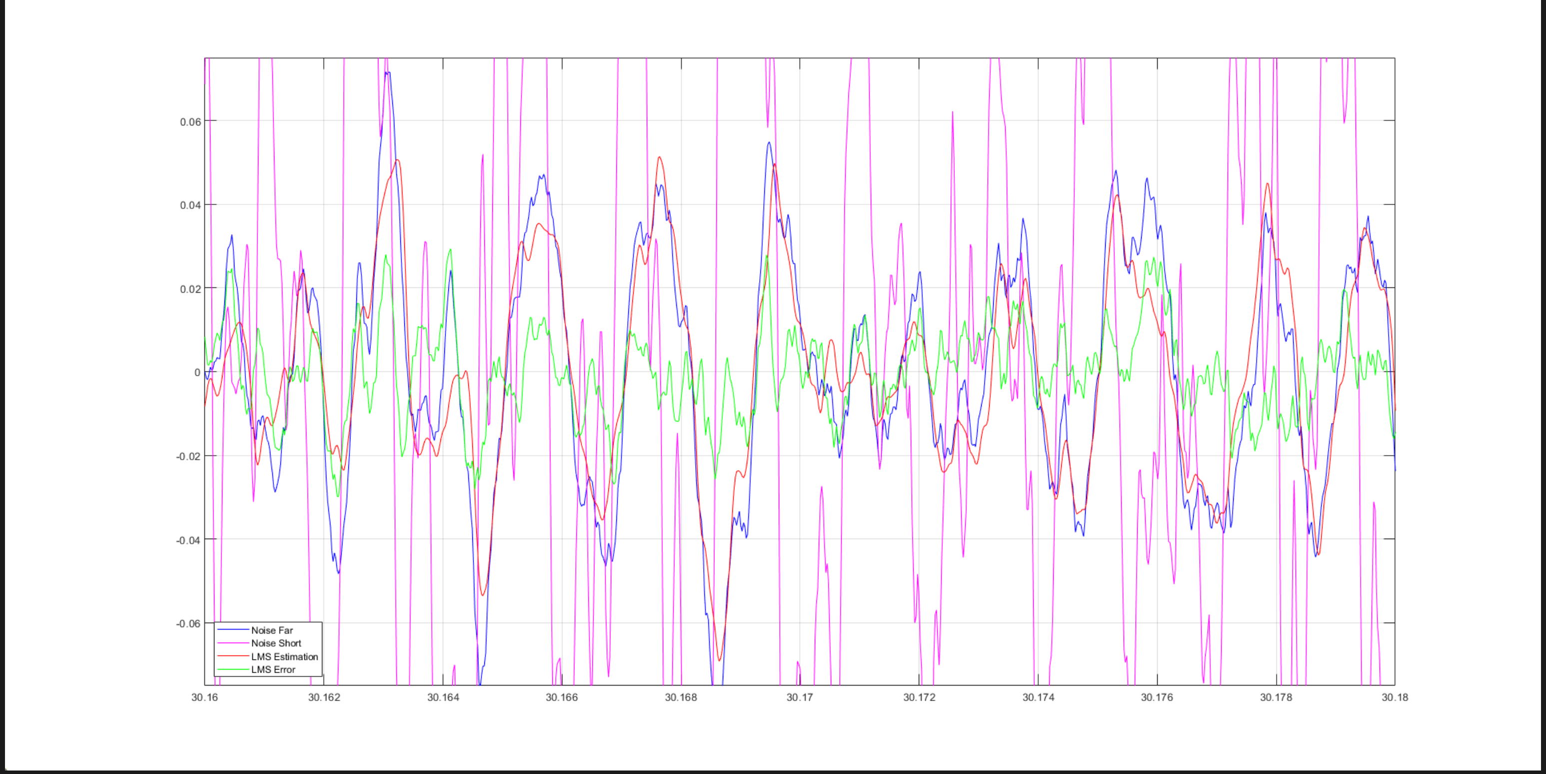
- Filter length = 1500   µ = 0.001



Analysis

Experiment 1: Only Noise signal

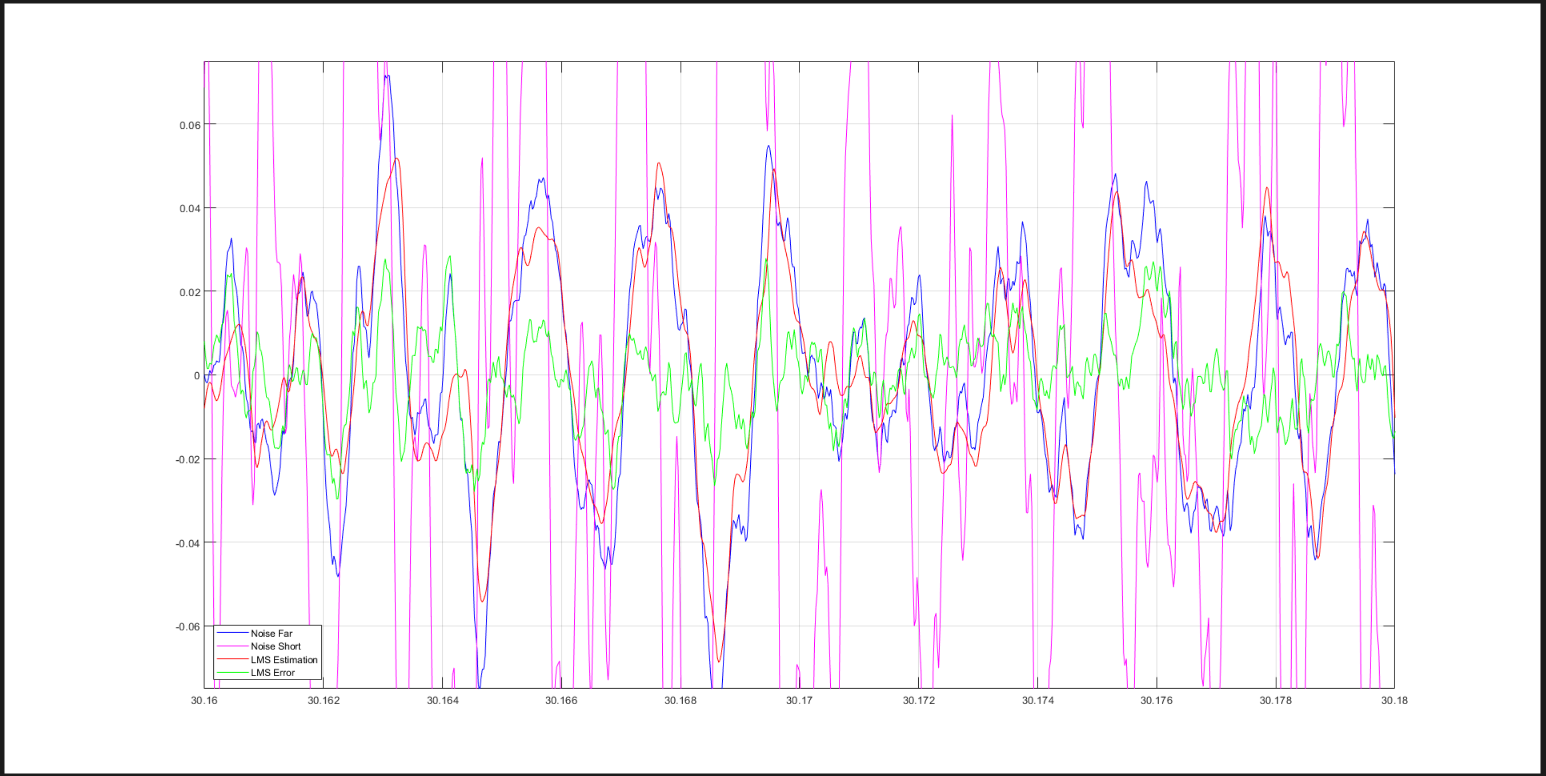
- Filter length = 1500   μ = 0.007



Analysis

Experiment 1: Only Noise signal

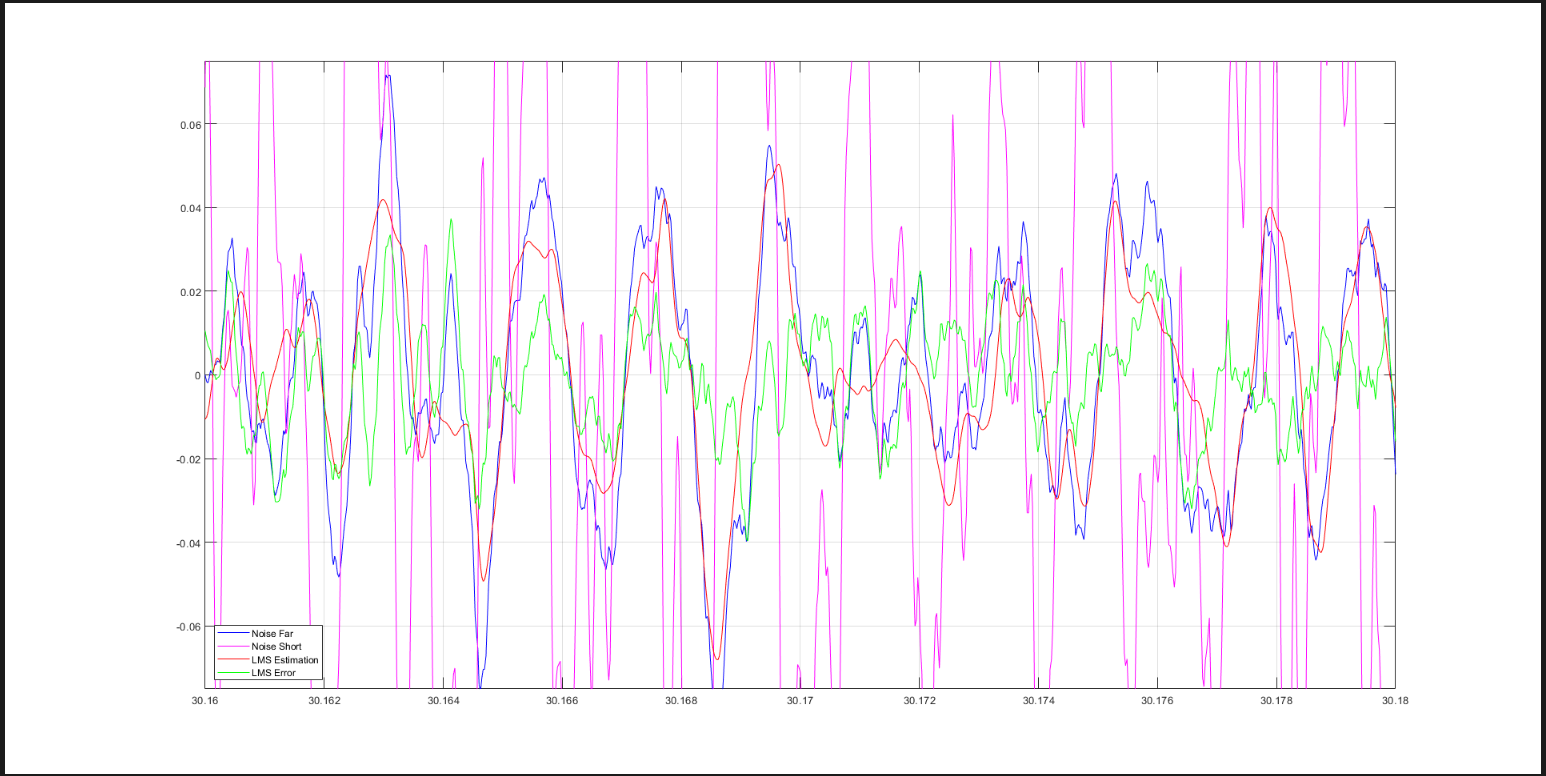
- Filter length = 1500 μ = 0.008



Analysis

Experiment 1: Only Noise signal

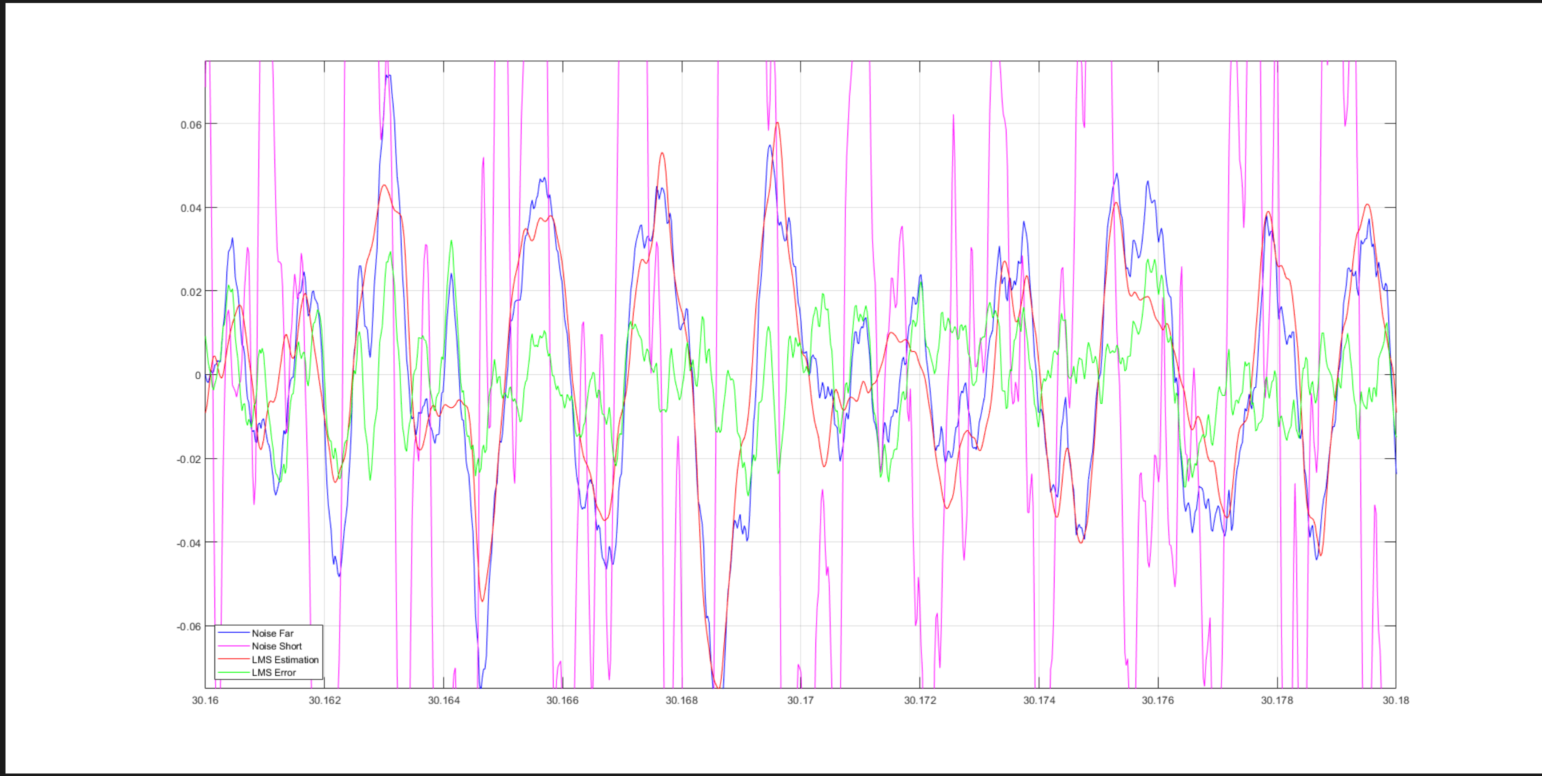
- Filter length = 2000 µ = 0.0002



Analysis

Experiment 1: Only Noise signal

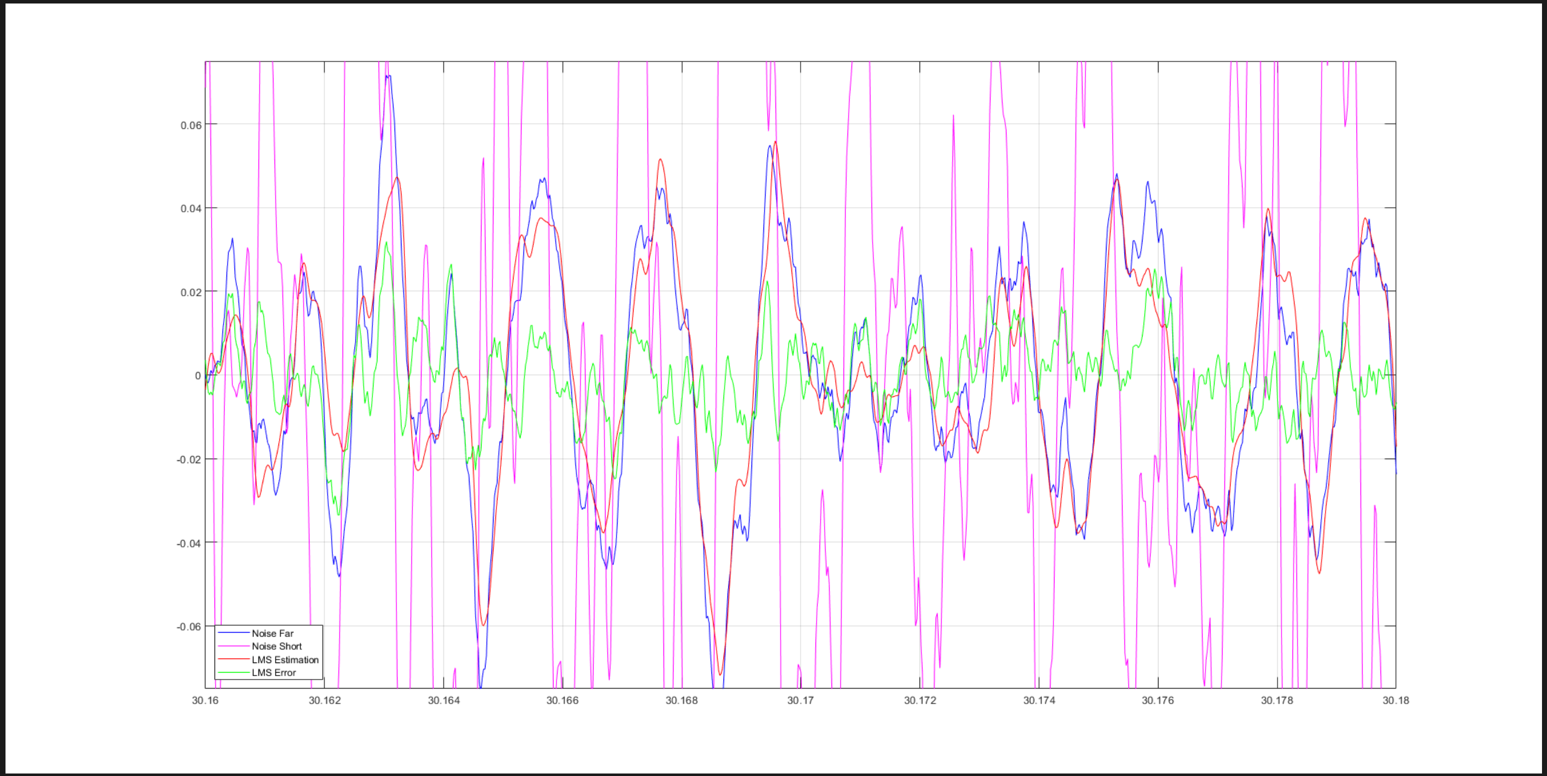
- Filter length = 2000 μ = 0.001



Analysis

Experiment 1: Only Noise signal

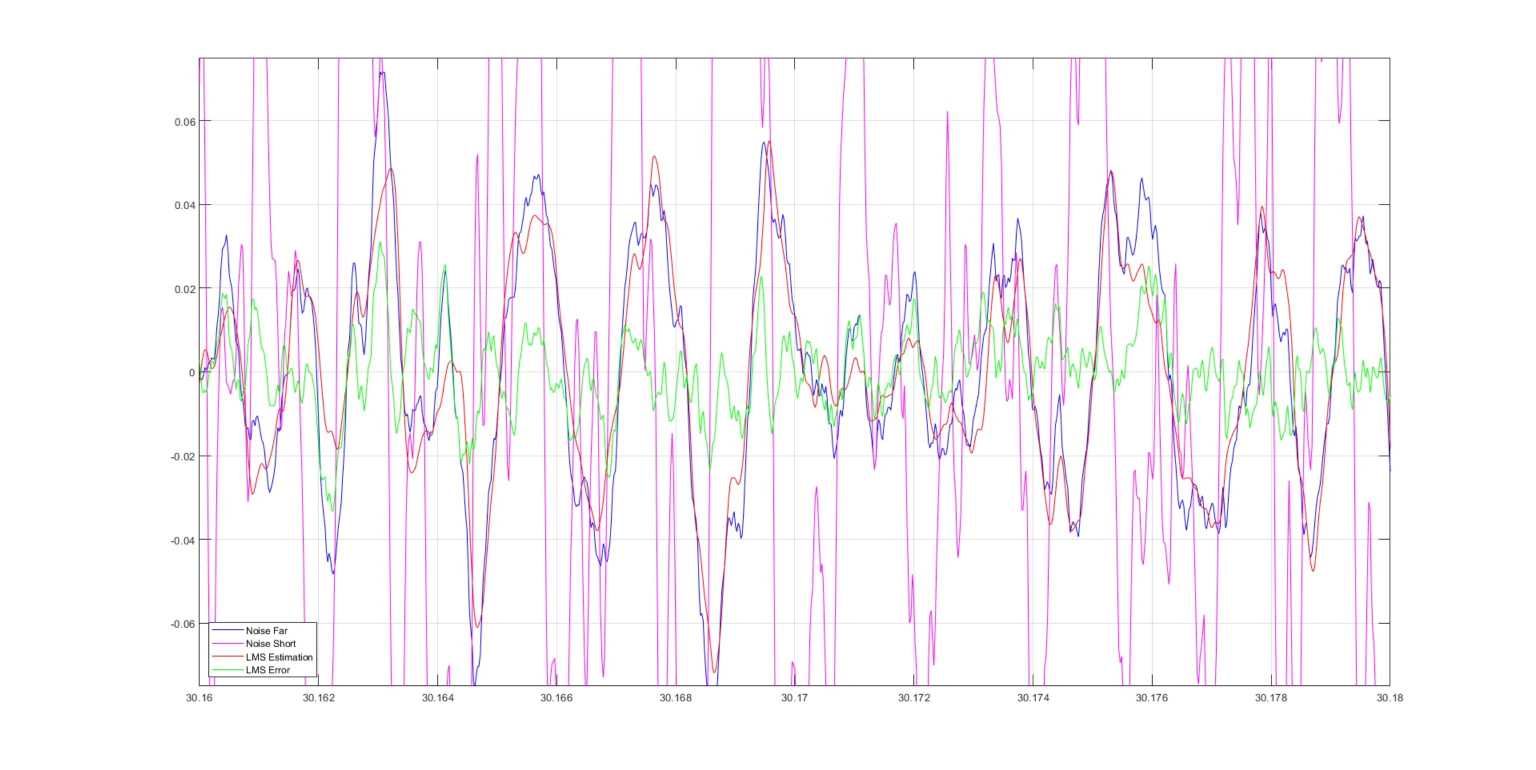
- Filter length = 2000   µ = 0.007



Analysis

Experiment 1: Only Noise signal

- Filter length = 2000   µ = 0.008



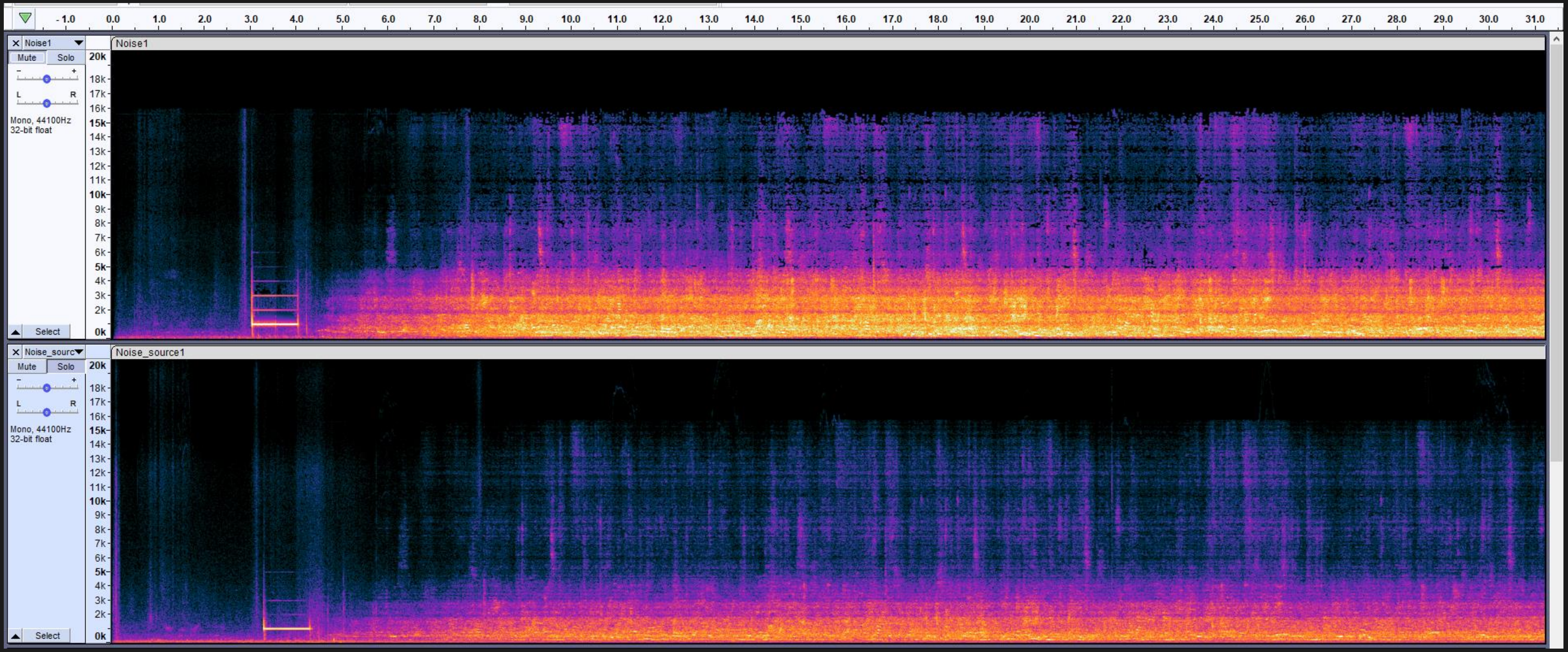
Analysis

Experiment 1: Only Noise signal

- Noise1 Spectrogram Graph

Noise Near

Noise Far



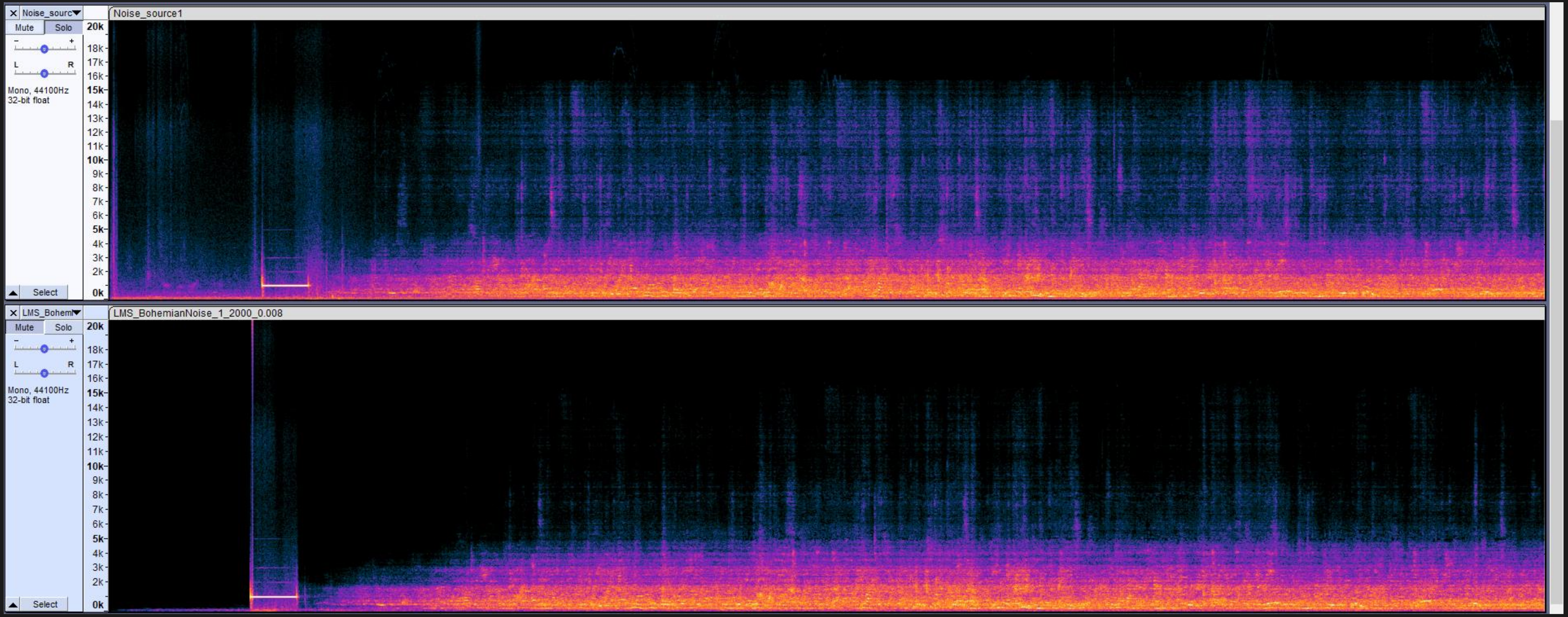
Analysis

Experiment 1: Only Noise signal

- LMS Error Spectrogram Graph: Filter length = 2000 μ = 0.008

Noise Far

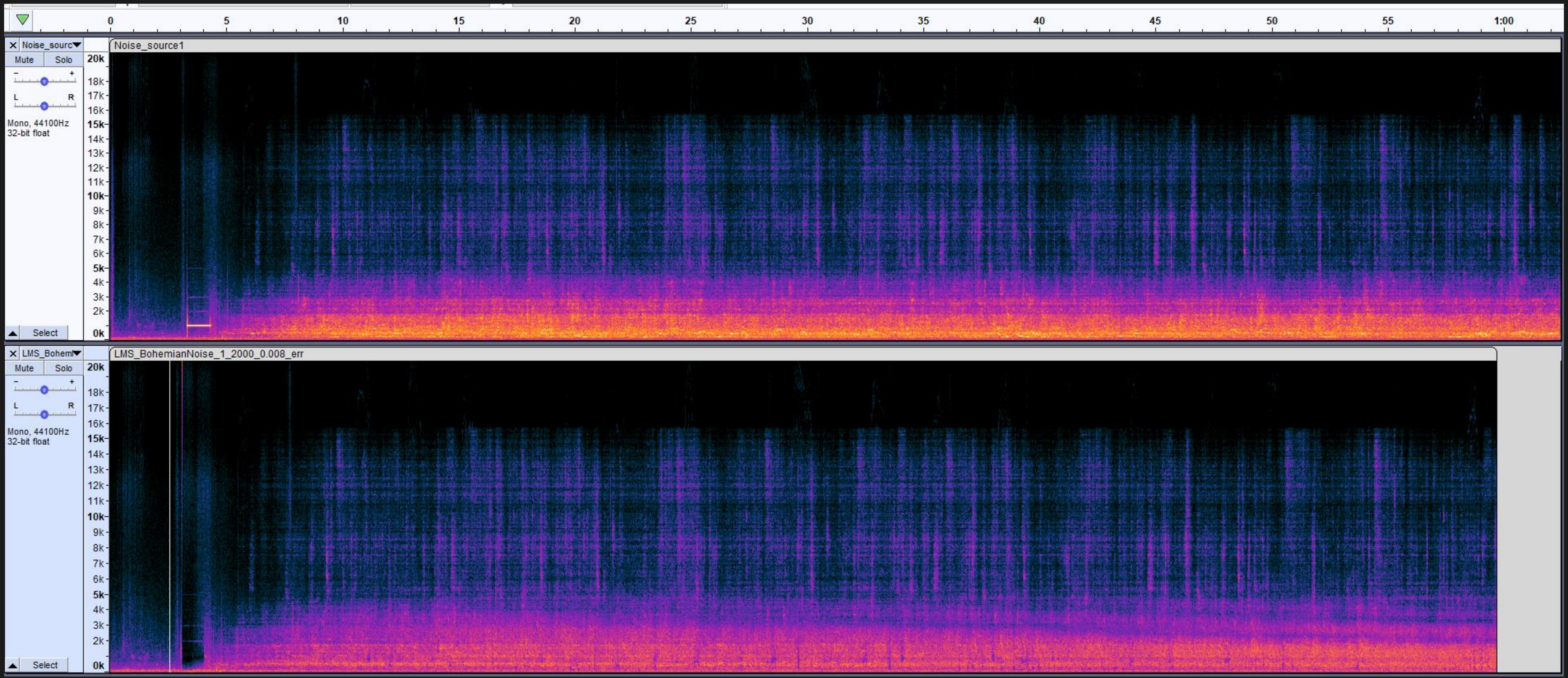
LMS Estimator



Analysis

Experiment 1: Only Noise signal

- LMS Error Spectrogram Graph: Filter length = 2000 µ = 0.008



Noise Far

LMS Error

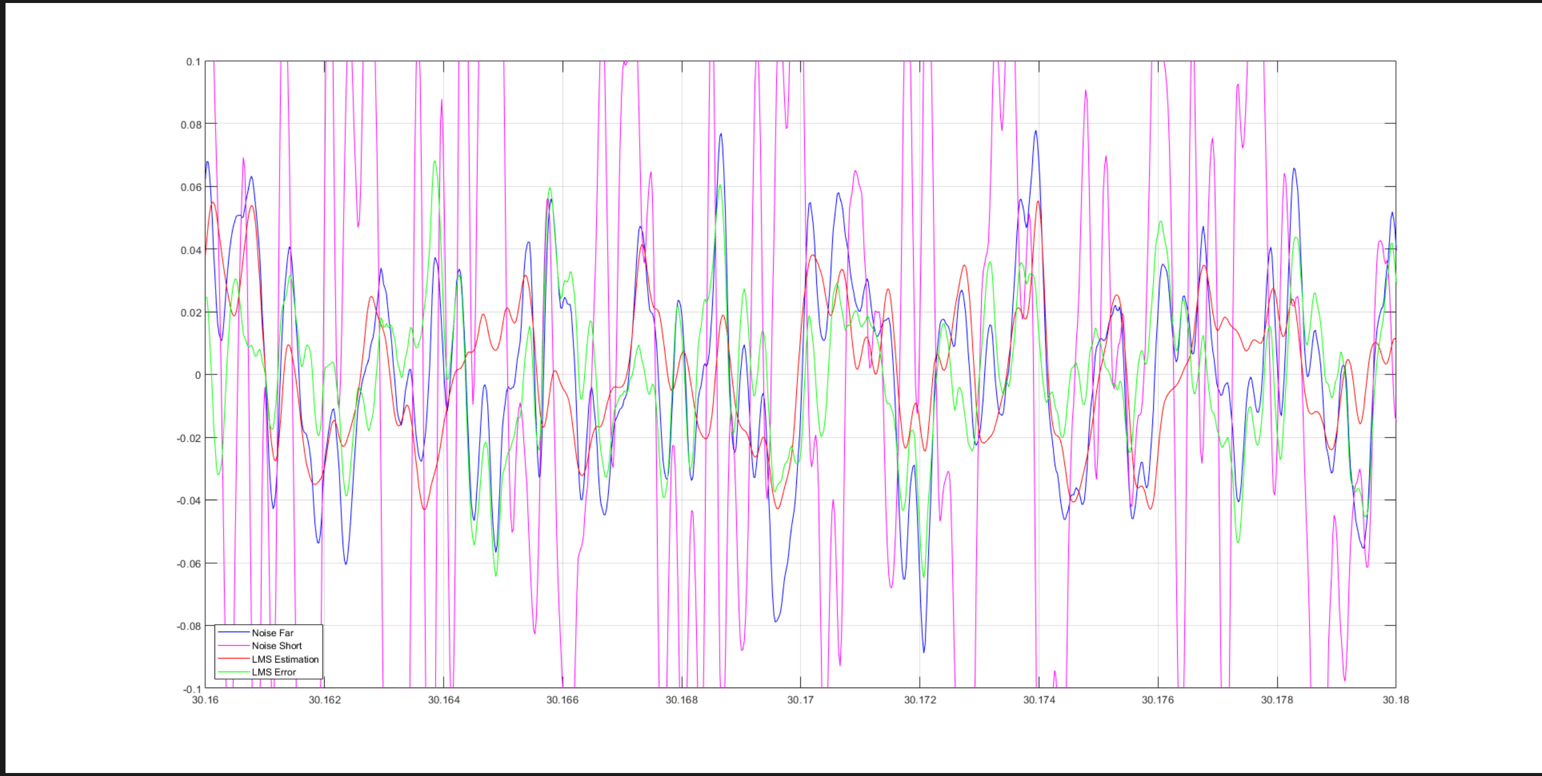
Conclusion

- Filter length: 1500-2000
- μ = 0.007 - 0.008
- Noise is greatly reduced from primary input which is satisfy our hypothesis.
- Noise is reduced in both high and low frequency.

Analysis

Experiment 2: With Music

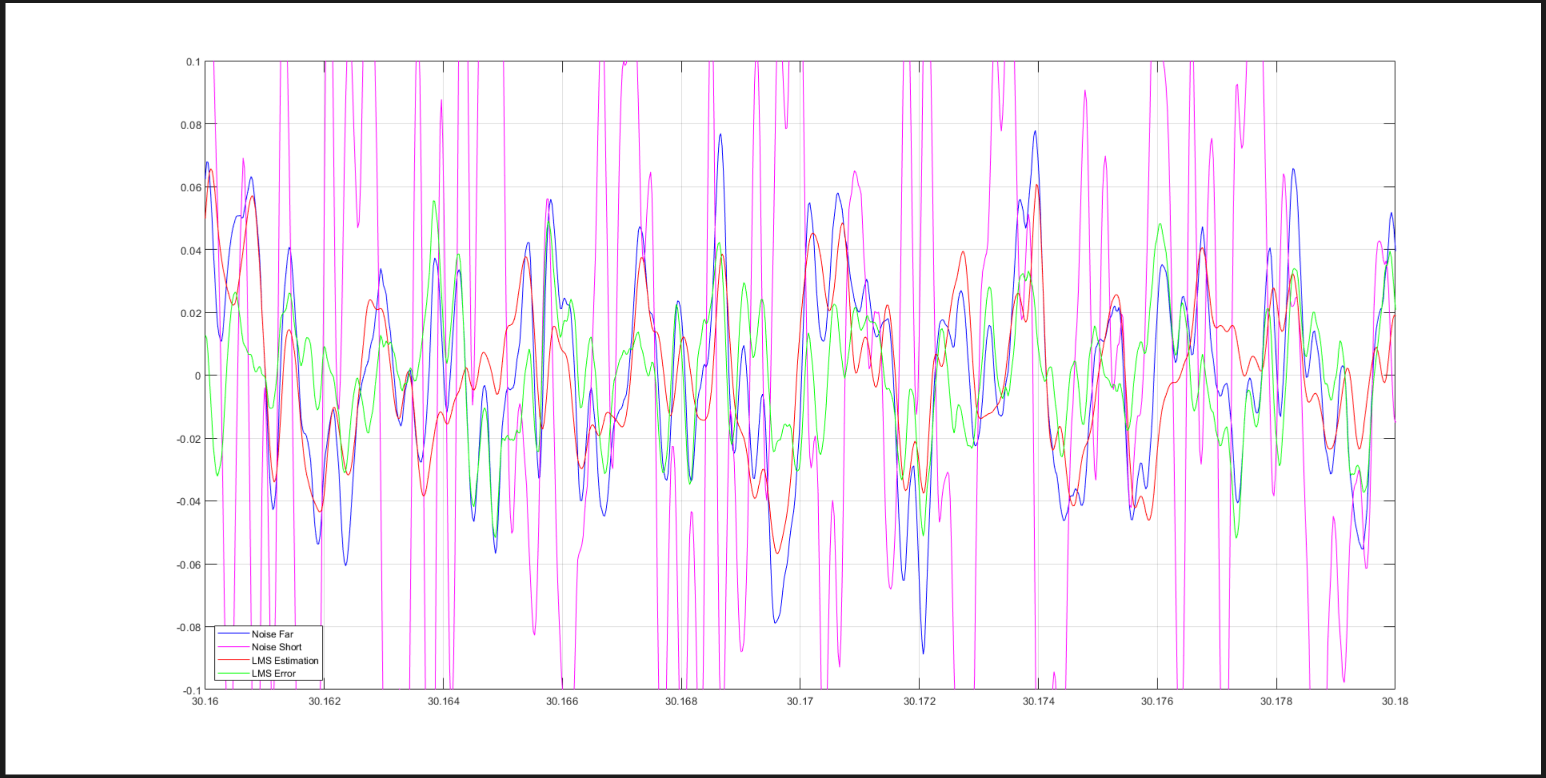
- Filter length = 2000 μ = 0.001



Analysis

Experiment 2: With Music

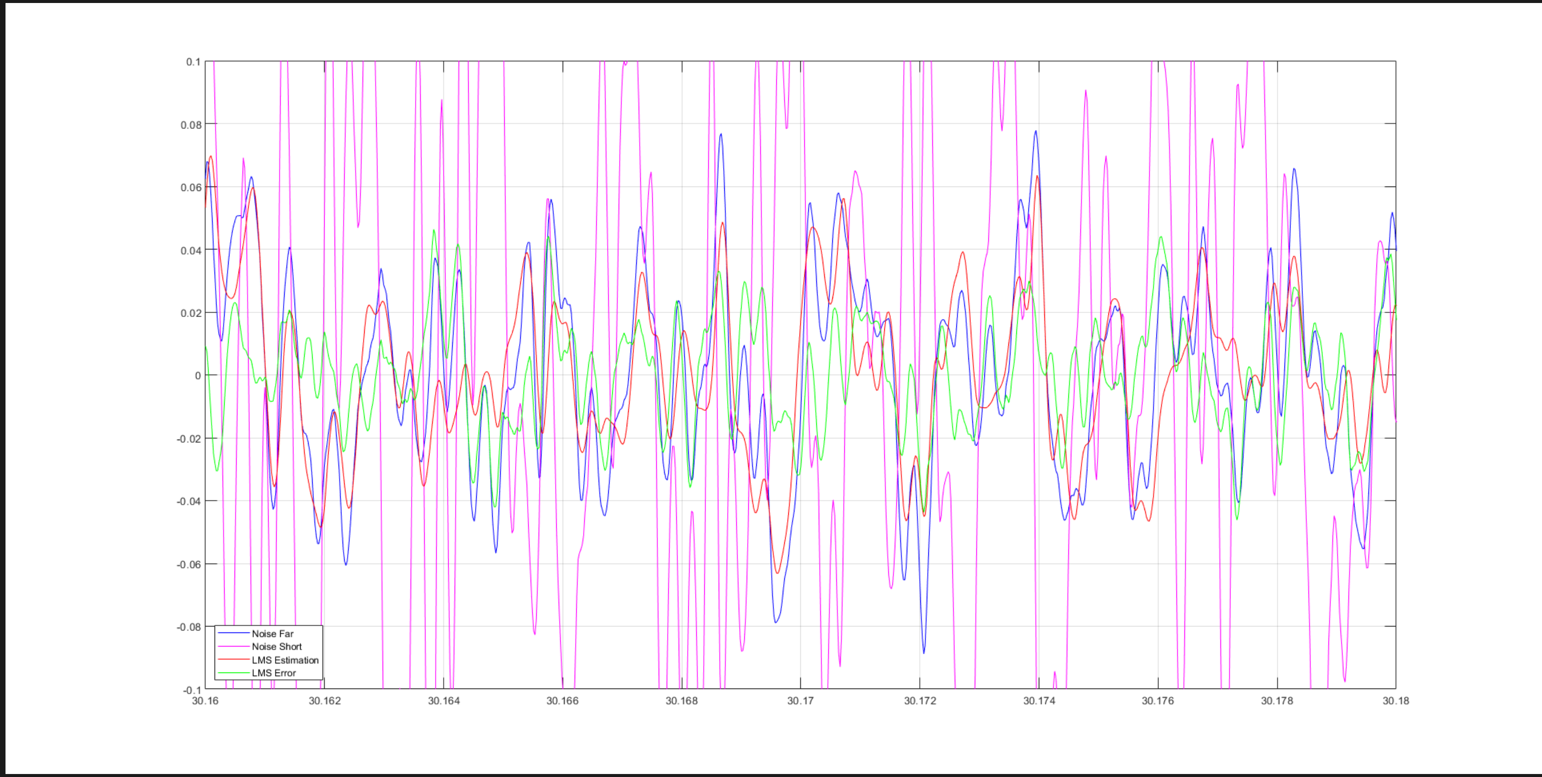
- Filter length = 2000   µ = 0.003



Analysis

Experiment 2: With Music

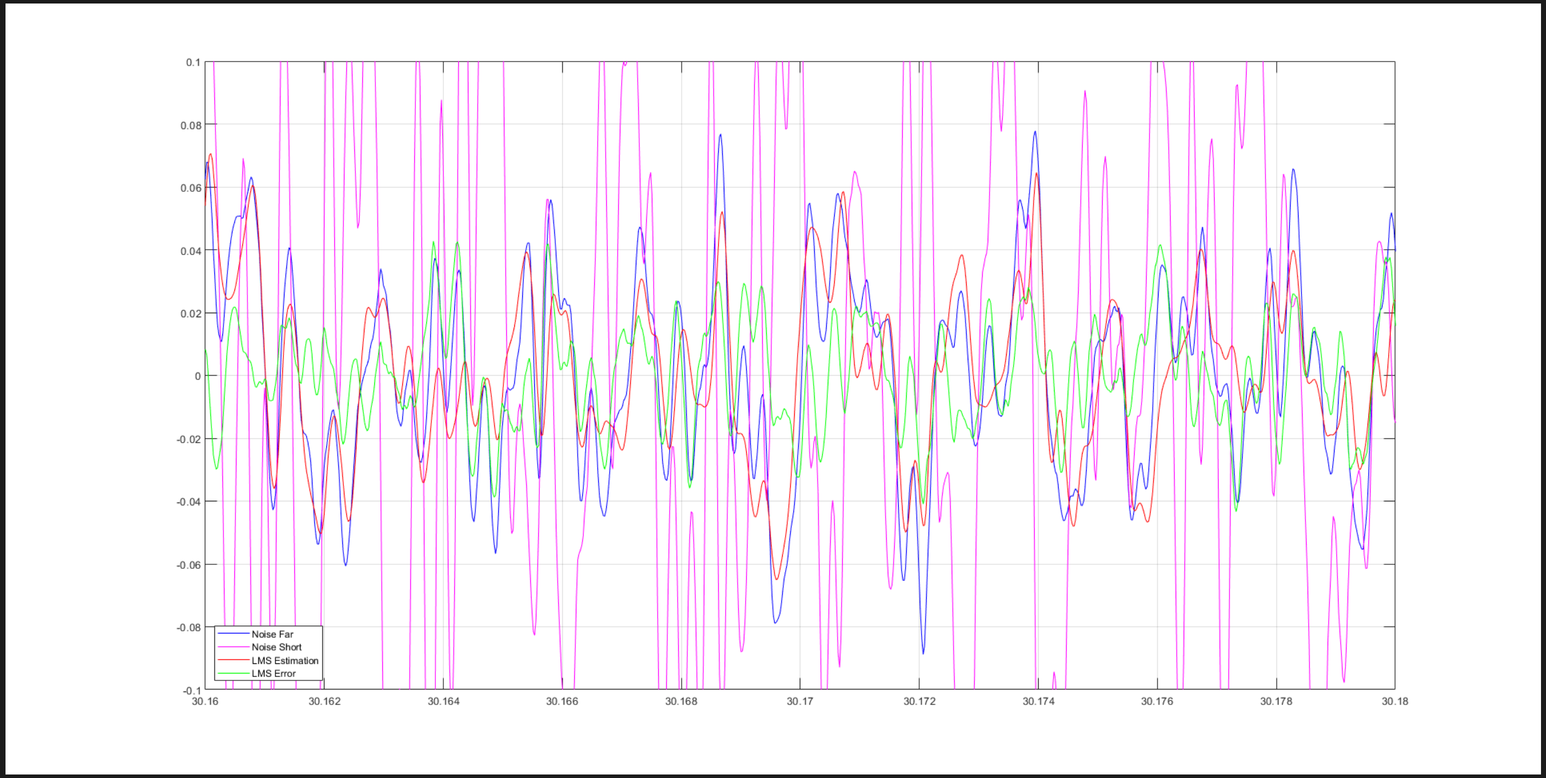
- Filter length = 2000   μ = 0.005



Analysis

Experiment 2: With Music

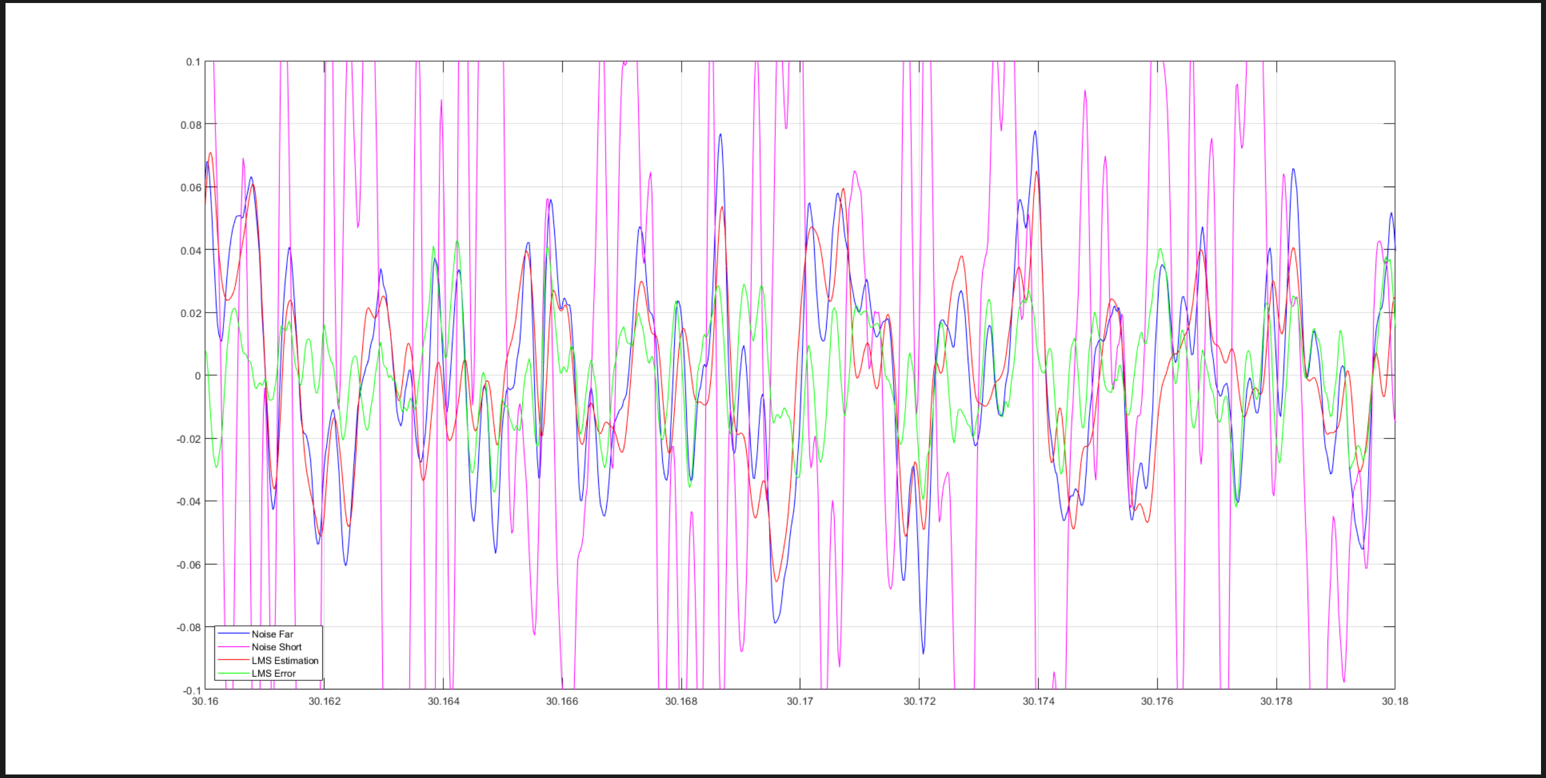
- Filter length = 2000   μ = 0.006



Analysis

Experiment 2: With Music

- Filter length = 2000   µ = 0.0065

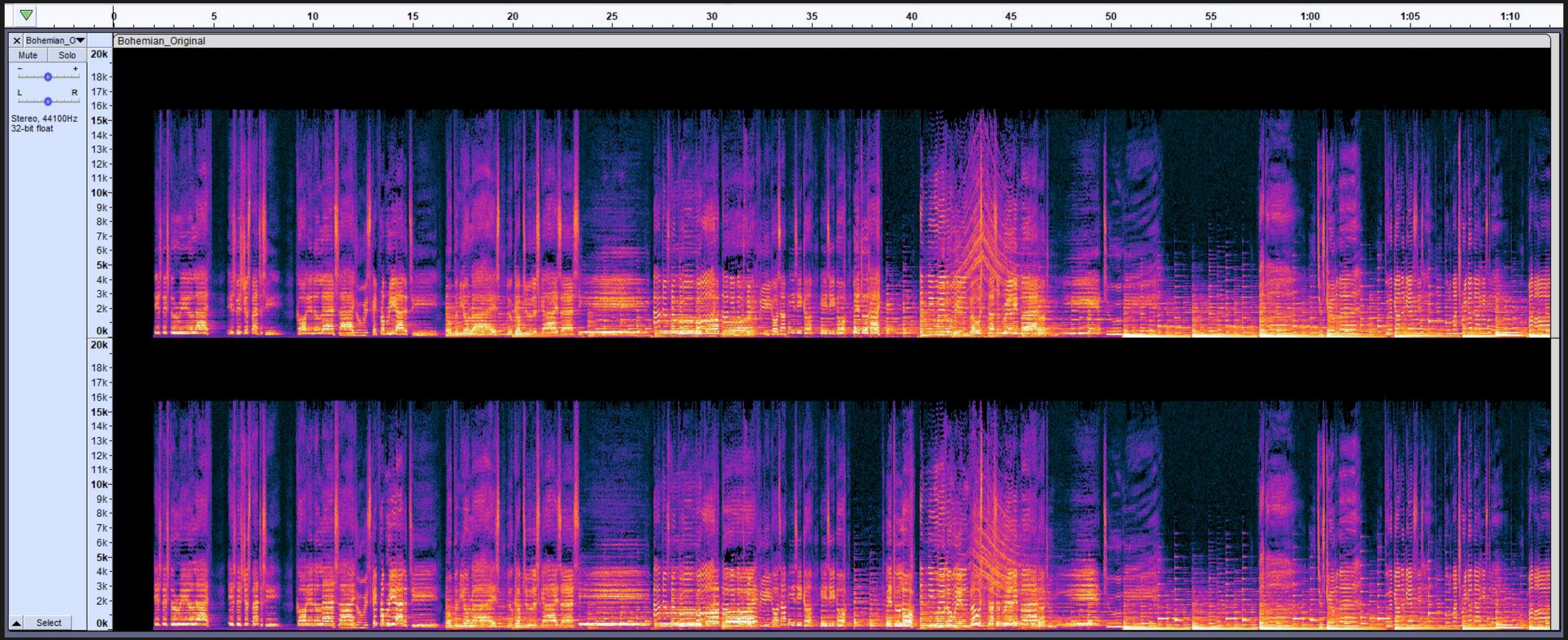


Analysis

Experiment 2: With Music

- Bohemian1 Original Spectrogram Graph

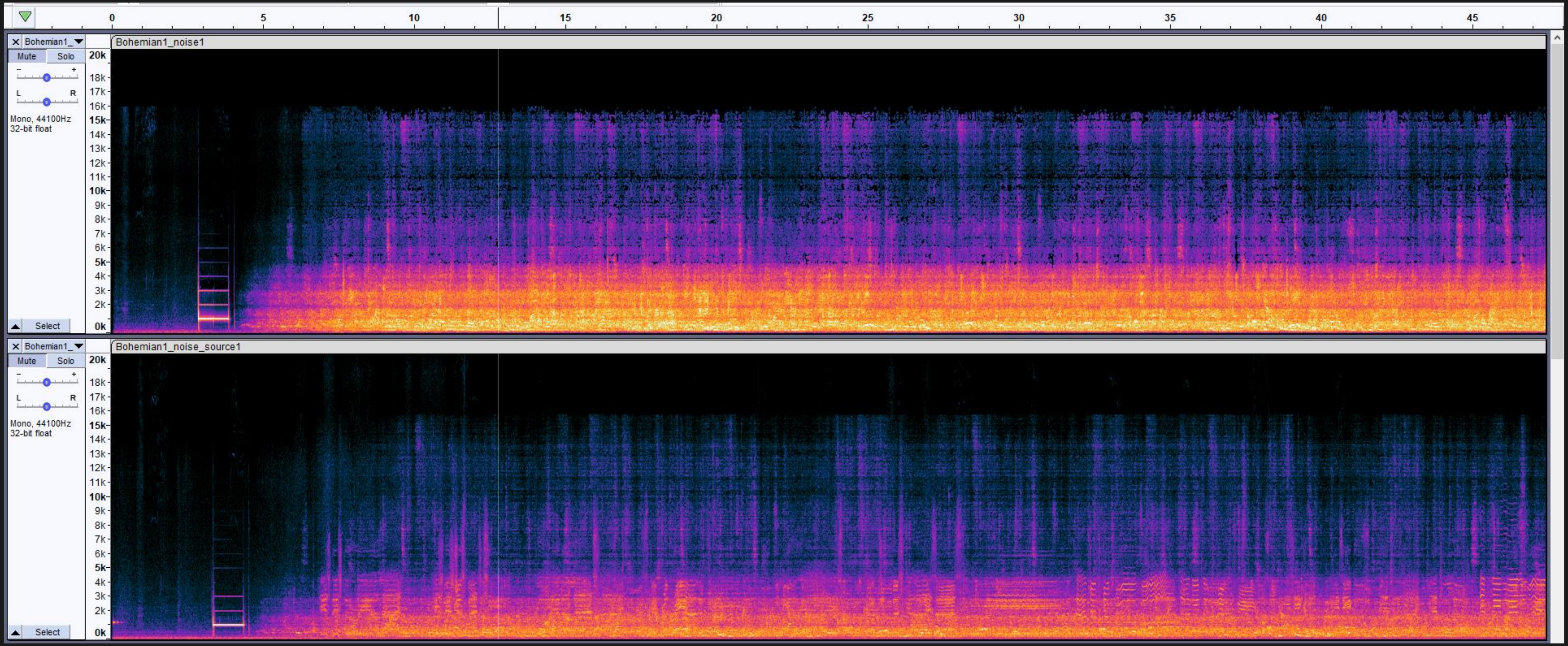
Original Song



Analysis

Experiment 2: With Music

- Bohemian1 Spectrogram Graph



Noise Near

Noise Far

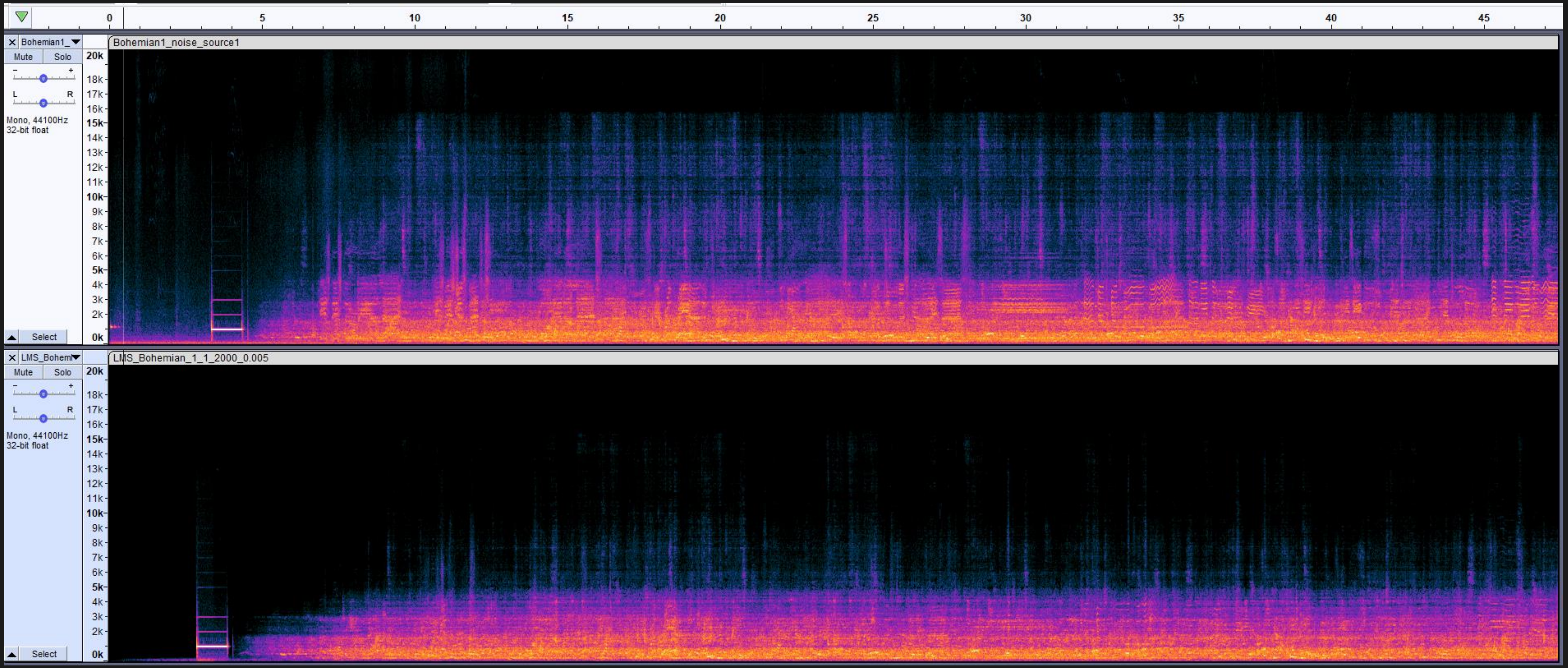
Analysis

Experiment 2: With Music

- Bohemian1 LMS Error Spectrogram Graph: Filter length = 2000 µ = 0.005

Noise Far

LMS Estimator



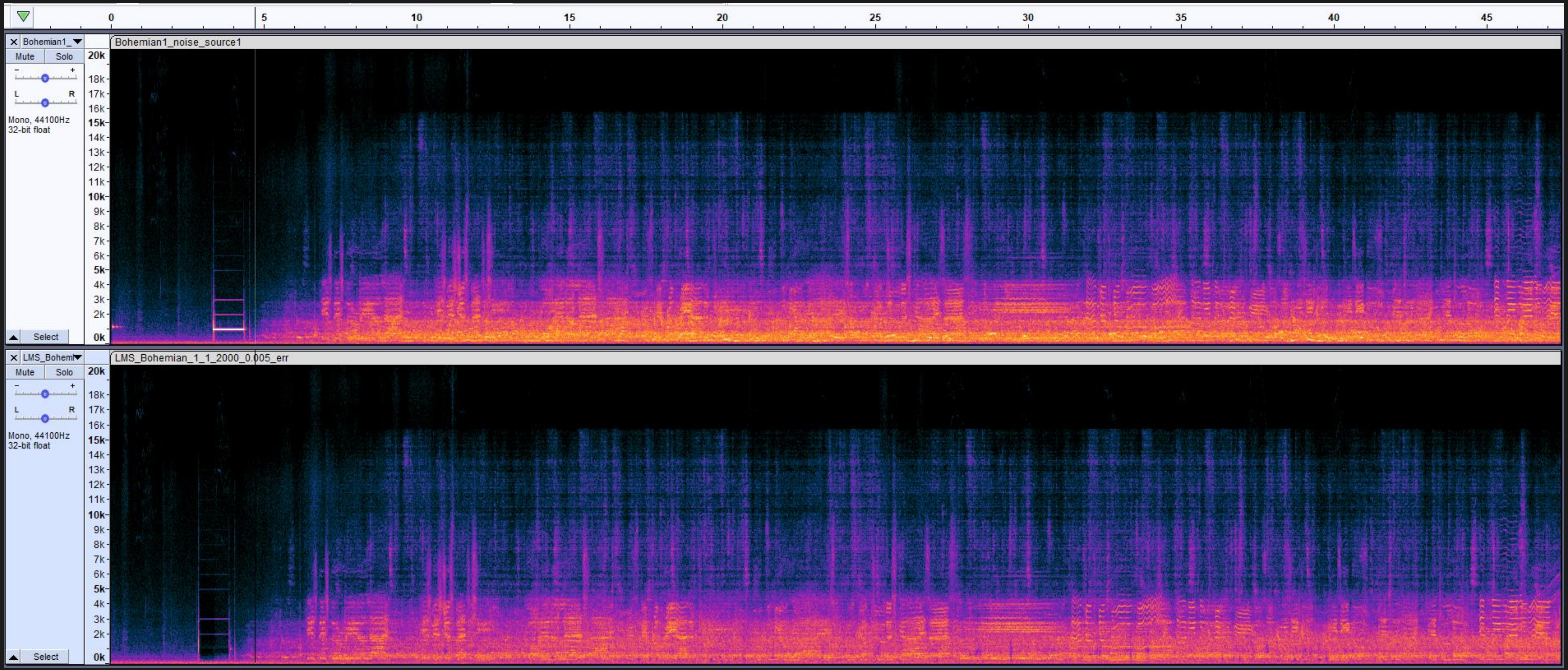
Analysis

Experiment 2: With Music

- Bohemian1 LMS Error Spectrogram Graph: Filter length = 2000 µ = 0.005

Noise Far

LMS Error



Conclusion

- Filter length: 1500-2500
- μ = 0.005
- Noise is greatly reduced from primary input which is satisfy our hypothesis.
- Noise is reduced in both high and low frequency.

Conclusion

- LMS Filter can cancel noise signal from primary input

- Filter length and step size is crucial for each system;

If Filter length is too large, LMS Output will be overfit due to noise in the exceed area of filter length and if Filter length is too small, LMS Output can never be close to Noise Far (Primary Input) because of not enough information in that filter length.

Step size tells how fast it comes close to the local minimum, so small step size results in slow learning rate but signal is time-varying which means it has certain length of step size. Big step size will always cannot find local minimum.

- Noise signal from Primary Input and Reference Input need to be correlate.

Conclusion

- To Increase Efficiency of Adaptive Filter

Restrictive bound of step size cand determine by

$$\textit{Transversal filter:}\quad 0 < \mu < \frac{1}{(L+1)(\text{signal power})}$$

In set up experiment, efficiency can be increased by shorten the distance of Primary Input and Reference Input for the better correlation of noise signal. For the small distance of microphone, reference input must not receive music signal

Future works and comment

- In this project, we can reduce noise from experimental signal
- There are lots of thing that can be adjusted:

Distance of microphone

Synchronization of microphone

Interference of 2 sources

Correlation of signals

Future: Implement on board with synchronization.

Thank For Listening