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EXPERIMENT 6. SYSTEM IDENTIFICATION WITH ADAPTIVE PROCESSING, DESIGN AND IMPLEMENTATION OF LMS FILTER

PART 2

LABORATORY REPORT

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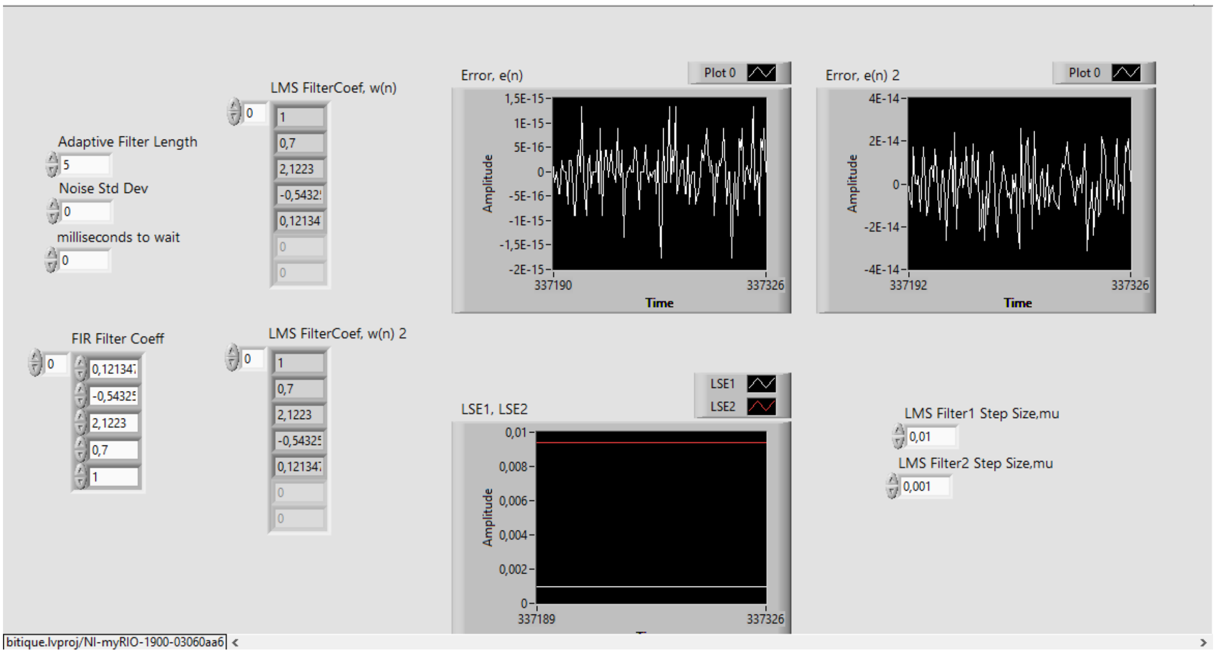
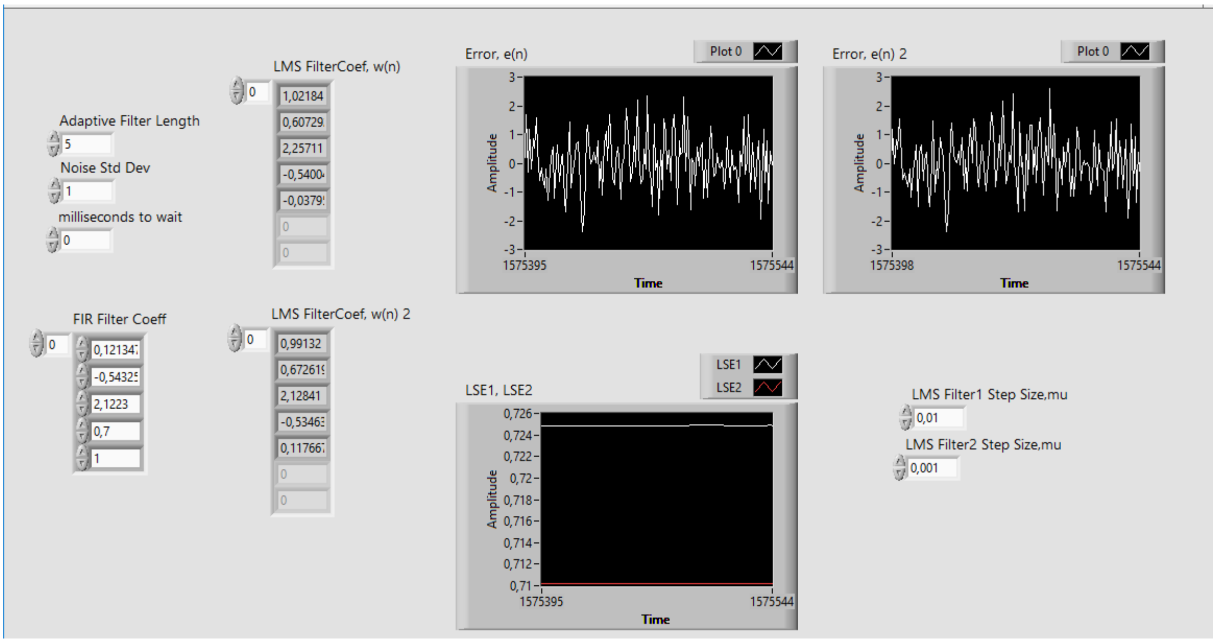
6.6.1. System Identification in MyRIO

d) Set the noise standard deviation to zero and run the VI. Check if the FIR filter coefficients are found accurately by the adaptive filter. Note that the coefficients should be the same only their order is different. Take a ***screenshot of the Front Panel after convergence***. Set the step sizes for adaptive filters to **0.01** and **0.001** in order to observe the difference between convergence speeds as well as the misadjustments. **Explain** which filter has faster convergence and which filter has the smallest misadjustment.



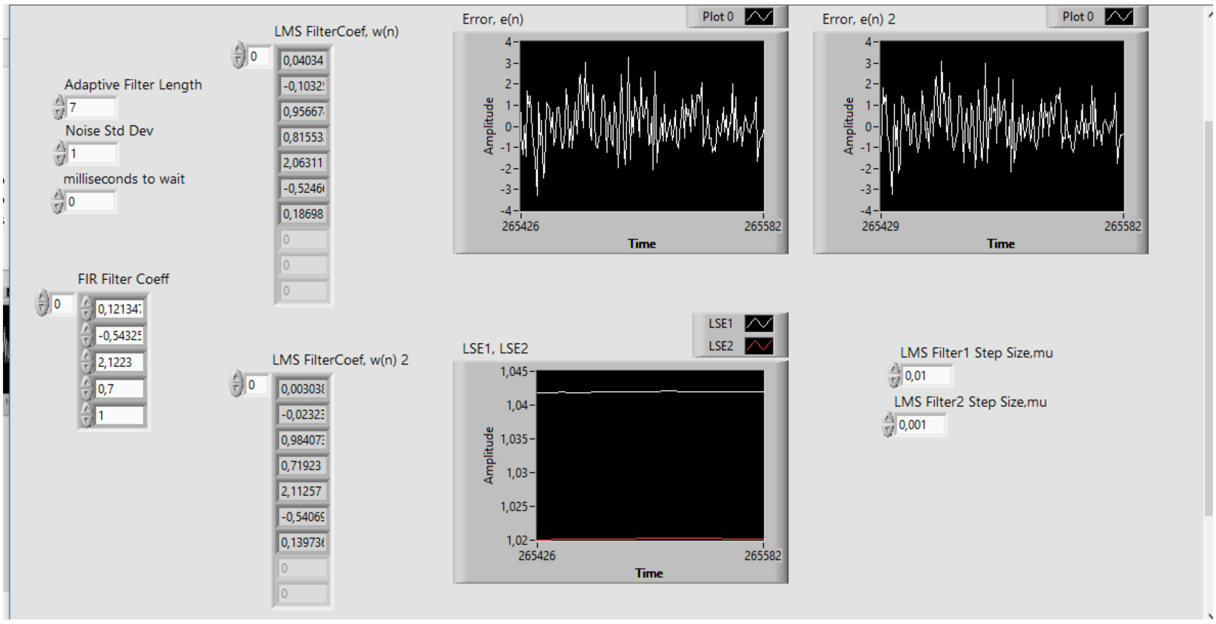
With the higher step size, we observed that the LSE converges faster. Here at the figure, LSE1 converged really fast and the LSE2 were continues to converging.

e) Set the noise standard deviation to 1 and **repeat d**. Wait for approximately two minutes until you observe a significant change in the LSE plots. ***Don't forget*** to include the ***screenshot of the Front Panel after convergence***. **Explain** your results and observations.



With the noise std deviation equal to 1, both LSE1 and2 increased, after waiting sufficient amount of time LSE2 converged to lower values slowly.

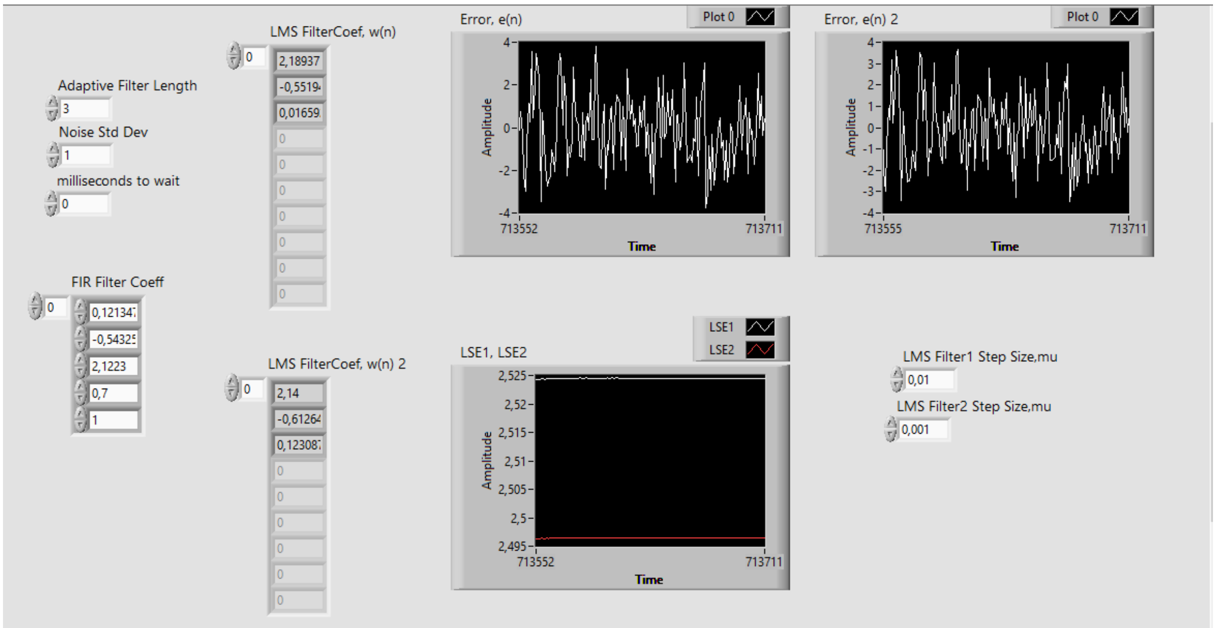
f) Increase the number of coefficients for the first adaptive filter. What is the effect of this? **Explain**.



With the increased adaptive filter coefficient length, we observed slightly higher LSE values due to the added unnecessary non-zero coefficients. This unnecessary coefficient interferes the signal with additional parts and caused slight additive error. If the length is too high it gives unnecessary information, which increases the error.

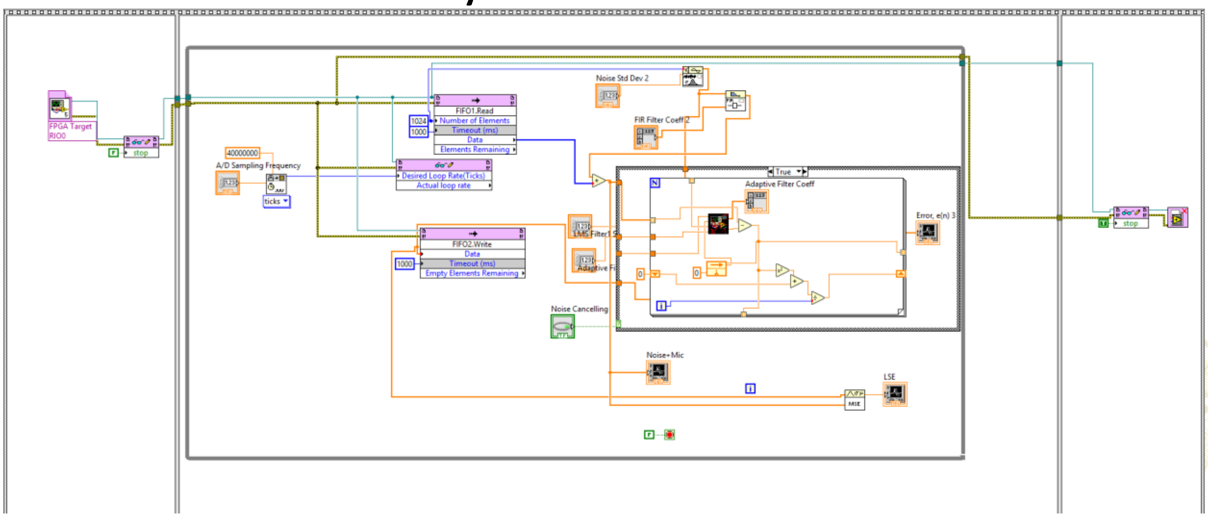
g) Decrease the number of coefficients for the first adaptive filter. What is the effect of this? **Explain**.

If it is not too low, decreasing the coefficient number was okay. Gave slightly better results. However, if the length is too low, it does not give the desired results. Since it just uses a few samples, it cannot obtain enough statistical information to eliminate the noise.

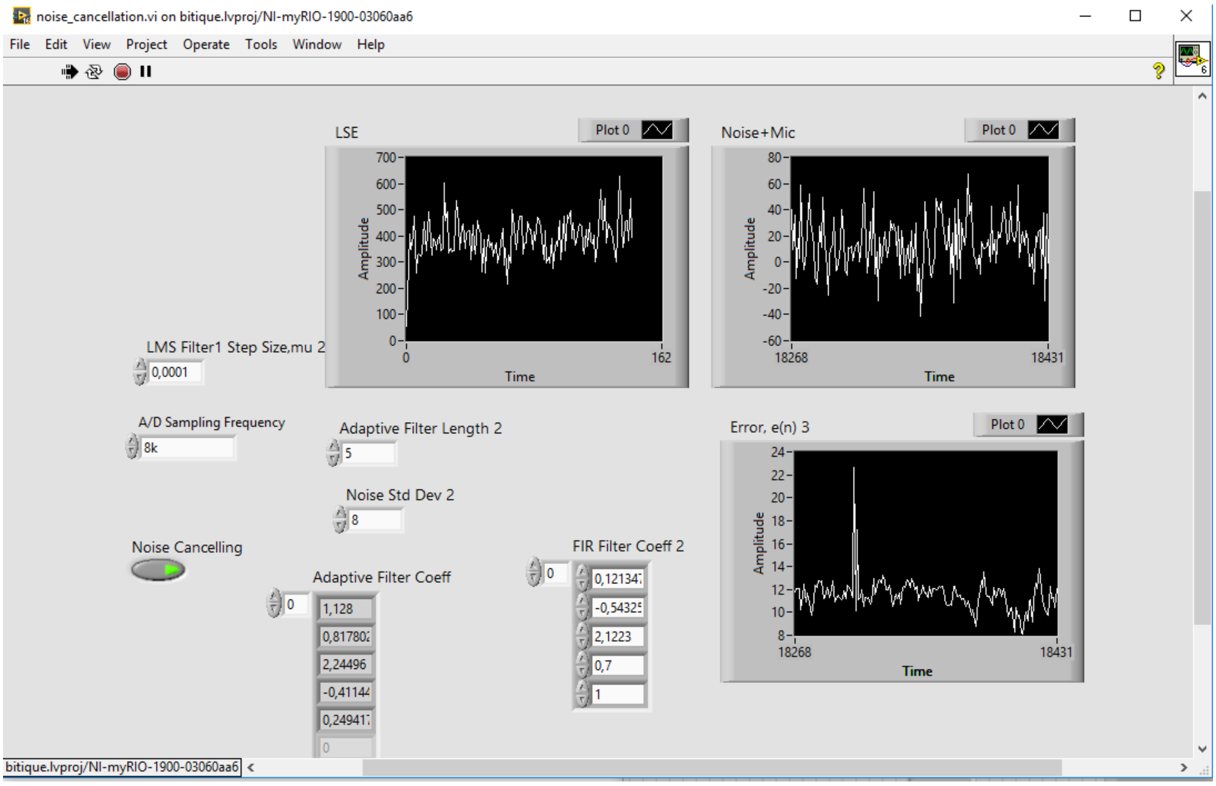


When the coefficient length decreased, both LSE values are increased due to the insufficient amount of statistics taken by insufficient amount of coefficients. When we compare this result with the previous part, we observed higher effect on the LSE

6.6.2. Noise Cancellation in MyRIO



e) Adjust the parameters of the program as shown in Fig. 9 and run your program. Check the convergence of the adaptive filter coefficients. **Don't forget to plot** $(e[n]-x[n])^2$ **in the Front Panel.** Take a ***screenshot of the Front Panel after convergence.*** While you speak to the microphone, press the Noise Cancellation button on and off to identify the function of the noise cancellation. **Explain** your results, **comment on** the function of the noise cancellation button.



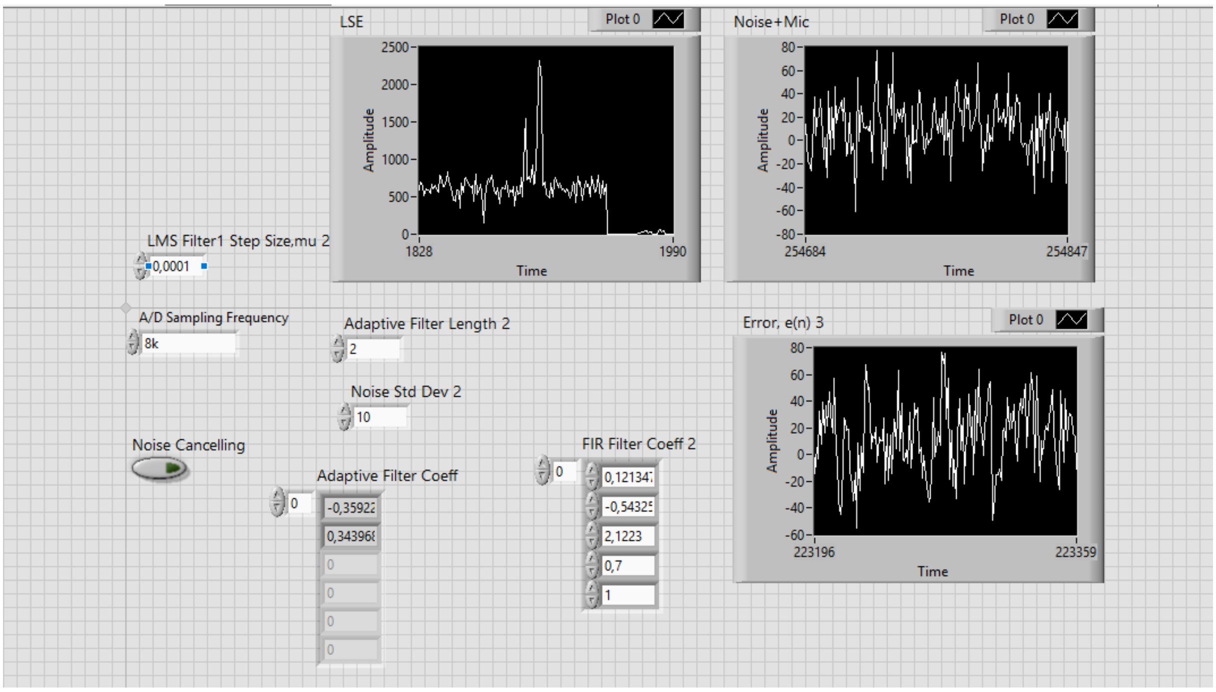
With the noise cancellation is on, we were able to filter the background white Gaussian noise and achieve cleaner sound. The adaptive filters mimic the noise as channel. When we subtract it, we can eliminate the noise.

f) Change the adaptive filter length to see the effect on the quality of the noise cancellation. **Comment on** the effect.

When the filter length is increased, we couldn't observe much difference on the headset and observed slight change in the error, but it was highly similar to the previous part.

However, when the filter length is decreased, we observed that the effectiveness of the filter is significantly decreased. And when we decreased the length to the 2, we observed almost no effect of the noise cancellation on the headphones.

Also increasing the length up to a point gives better results. However, after some point, a lag occurs on the processed signal and the result does not get any better.



g) Change the A/D sampling rate to **2000 Hz**. **Repeat e**. What are the differences between case **e and g**? **Explain the reasons**.

The quality of the voice got quite worse. The signal was lagged a lot. Due to slower sampling rate, higher frequencies in my voice could not be sampled.

Also there is a noticeable delay in the processed signal. Since sampling frequency is lower, the processing speed is slower.

The sampling rate must be at least 2 times higher than the highest frequency in my voice for perfect processing.

h) Set the A/D sampling rate to **8000 Hz** again. Increase the step size of the adaptive filter until divergence. **Note** the value of this step size. **Explain the reason** for divergence.

The convergence speed increases unnoticeably. However, after some point it cannot converge anymore. This value is around 0.1. This step size represents how much we penalties the error and how much it should affect the coefficient updates at each iteration.