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Research Original Article

Matlab Simulation Using Kalman Filter Algorithm to Reduce Noise in Voice Signals

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Abstract

Sound signals polluted by noise are a common problem in various audio applications, including communication, sound processing, and audio recording. In this article, proposes the use of Kalman Filter algorithm as an effective method to reduce noise in speech signals. Simulations are performed using Matlab software to implement the Kalman Filter algorithm on noise polluted voice signals. The study includes several important steps, including the input of noise-polluted speech signals and the implementation of the Kalman Filter to clean the signals. Simulation results are measured using commonly used audio quality metrics, such as Signal-to-Noise Ratio (SNR) and Mean Square Error (MSE), to evaluate the effectiveness of the algorithm. The results from the simulations show that the use of the Kalman Filter algorithm significantly improves the quality of noise-contaminated speech signals. These results indicate that this algorithm can be a potential solution to the problem of noise reduction in audio applications. In addition, the implementation in the Matlab environment allows for easy testing and adaptation of this algorithm for different types of audio applications. This research makes a positive contribution to the development of more efficient noise reduction techniques in speech signal processing, focusing on the use of the Kalman Filter algorithm and its implementation using Matlab software. The implications of this research can be potentially beneficial in improving the quality of sound signals in various audio application contexts.
Keywords*:* Voice signal; Noise; SNR; MSE; Kalman Filter; Matlab

Introduction

A clean sound signal free from noise interference is a critical aspect in many audio applications. Noise, or unwanted sound interference, can affect the quality of communication, speech processing, and audio recording. Therefore, noise reduction in speech signals has become an important research focus in the field of audio technology (Misriana & Kartika, 2018). In an attempt to address this issue, various noise reduction algorithms have been developed. One of the well-known algorithms is the Kalman Filter Algorithm, which has the advantage of reducing noise in a sound signal by utilizing statistical estimation of the signal. This Kalman Filter Algorithm has been successfully applied in various technological applications, including navigation, image processing, and signal processing (Sudaradjat & Rosano, 2020).
In this context, this research aims to investigate and simulate the use of the Kalman Filter Algorithm in reducing noise in speech signals using Matlab software. Matlab is a very useful tool in the development and testing of speech signal processing algorithms, and the Kalman Filter Algorithm is one of the promising methods to achieve this goal. This article will discuss the implementation steps of the Kalman Filter Algorithm on noise-polluted speech signals(Kecepatan & Dc, 2021). Quality measurements of the noise reduction results will be made using commonly used metrics, such as Signal-to-Noise Ratio (SNR) and Mean Square Error (MSE). The results of these simulations will provide valuable insights into the effectiveness of this algorithm in overcoming noise problems in speech signals (Misriana & Kartika, 2018).
This research has great potential to improve the understanding of noise reduction in the context of audio and can positively contribute to the development of more efficient techniques in processing noise-contaminated speech signals (Harahap & Kartika, 2022). Moreover, the implementation of this algorithm using Matlab software allows researchers to easily test and customize this algorithm for different types of audio applications (Iqbal et al., 2010). Thus, this research has significant relevance in the understanding and development of more advanced and quality audio technology.

Literature Review

1. Voice Signal and Noise

A sound signal is a longitudinal mechanical wave consisting of gradual variations in air pressure. A sound signal can be thought of as a complex signal consisting of different frequency components (Fft & Matlab, 2019). Noise, on the other hand, is an unwanted signal that can corrupt or interfere with the original sound signal. Noise can come from various sources, such as electronic sources, the environment, or other interference (Widiarto, 2012), (Syahputra et al., 2018).
The following types of noise are often common in everyday environments:

- White noise: Is a type of noise that has the same energy density at all measured frequencies. white noise sounds like a constant hissing or static sound. Hissing or constant static is a type of noise or sound that appears continuously and does not change in intensity or frequency. Hissing sounds can occur in many types of electronic or audio devices, such as speakers, headphones, or microphones, and are often caused by interference from sources such as electrical noise or electromagnetic interference. A constant hissing sound can interfere with the experience of listening to or recording audio.
- Pink noise: Is a type of noise that has the same energy density at every octave of frequency. Pink noise sounds smoother and lower than white noise.
- Brown noise: Is a type of noise that has a higher energy density at lower frequencies compared to white noise. Brown noise sounds heavier and deeper than white noise.

The noise signal waveform graph can be seen in the image below:

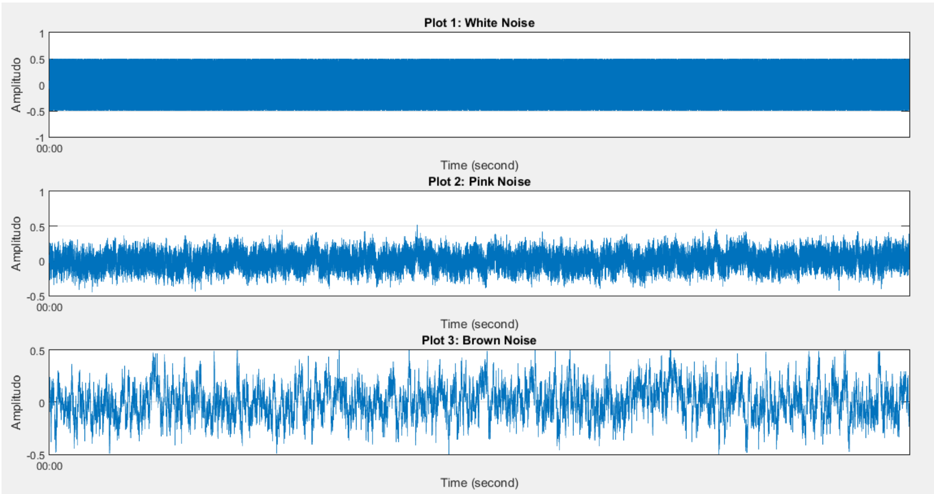


Figure 1. Noise signal waveform graph

In the figure above are three line graphs representing the amplitudes of "white noise," "pink noise," and "brown noise." over time. You can see the blue colored line representing the noise signal in waveform. This plot will have an x-axis label in minutes: seconds format and a y-axis label in units of audio amplitude.

2. MATLAB (Matrix Laboratorium)

Matlab stands for "Matrix Laboratory". It is a programming language and numerical computing environment developed by MathWorks. Matlab is widely used in various disciplines, including engineering, mathematics, biomedical science, economics, and physics. Matlab has a graphical user interface (GUI) that allows users to visualize and manipulate numerical data in the form of matrices, vectors, sound signal processing and data processing (Syarifuddin et al., 2018), (Hasibuan et al., n.d.). In addition, Matlab also provides various functions and libraries that are useful in various types of numerical analysis, such as signal analysis, optimization, image processing, system modeling, and so on (Mata & Per, 2006), (Hasibuan & others, 2019).

Figure 2. Matlab logo software

The advantage of Matlab is its ability to easily implement and test signal processing algorithms through various toolboxes and functions available. In the context of this article, using Matlab software to implement the Kalman Filter Algorithm on noise polluted speech signals, with the aim of reducing noise and improving the quality of the observed speech signals (Pratiwi et al., 2017), (Harianto, n.d.). This simulation will help in testing the effectiveness of the Kalman Filter algorithm in overcoming noise problems in speech signals, as well as understanding its potential use in various audio applications (Safa et al., 2017).

3. Kalman Filter Algorithm

Kalman filter is a signal processing method and statistical estimation technique used to estimate unmeasurable values or values contaminated with noise. The Kalman filter was first developed by Rudolf Kalman in 1960, and has since been used in a variety of fields, including navigation, robotics, signal processing, and others (Juslam et al., 2019). The Kalman filter is a set of mathematical equations that provide a computationally efficient (recursive) solution to the least squares method (Salsabila et al., 2022). This filter is very effective in several aspects. It supports past, present, and even future ratings even when the exact nature of the modeled system is unknown. There are several types of Kalman Filters, such as Standard Kalman Filter, Extended Kalman Filter, Unscented Kalman Filter and Ensemble Kalman Filter. The Standard Kalman Filter is the simplest while the other types are modified for more complicated task (Morgan & Software, 2022).

The Kalman Filter algorithm is an estimation algorithm used to estimate variables that cannot be directly observed by utilizing observable information (Hasan et al., 2023). This algorithm was developed by Rudolf E. Kálmán and is very useful in estimation problems in the signal processing domain. The Kalman Filter algorithm combines prediction and measurement to provide optimal estimation by minimizing the mean square error.

The Kalman filter has two parts to the equation, the prediction part and the update part. The standard Kalman Filter equation is shown in the following equation.

Prediction:

$$xt-1 = F_{t}xt-1|t-1 + B_{t}u_{t} \tag{1}$$

$$P_{t|t-1} = F_{t}P_{t-1|t-1}F_{t}^{T} + Q_{t} \tag{2}$$

Update:

$$xt|t = xt|t-1 + K_{t}(y_{t} - H_{t}xt-1) \tag{3}$$

$$Kt = Pt|t-1\, HtT(Ht\, Pt|t-1HtT + Rt) - 1 \tag{4}$$

$$P_{t|t} = (I - K_{t}H_{t})P_{t|t-1} \tag{5}$$

Where x is the estimated state, F is the state transition matrix, u is the control variable, B is the control matrix, P is the state variance matrix, Q is the process variance matrix, y is the measurement variable, H is the measurement matrix, K is the Kalman gain, R is the measurement matrix, $t\,|\,t$ is the current time period, t - 1 | t - 1 is the previous time period, and t | t - 1 is the intermediate steps (Deng et al., 2007). Equations 1 to 3 can be referred to as the Kalman Filter system model and the goal has not yet been determined. Therefore, the Kalman Filter system model can be modified based on the designed goal and how complex the system is. Algoritma Filter Kalman dapat diterapkan untuk mengurangi *noise* pada sinyal suara dengan menganggap *noise* sebagai gangguan yang dapat diestimasi. Dalam konteks ini, sinyal suara yang tercemar *noise* dianggap sebagai sinyal yang diamati, sedangkan sinyal suara asli yang bersih dianggap sebagai variabel yang tidak dapat diamati. Algoritma ini bekerja dengan menggabungkan prediksi sinyal suara bersih dengan pengukuran sinyal suara yang tercemar *noise* untuk menghasilkan estimasi yang lebih baik pada pemrosesan sinyal (Simbolon & Sani, 2014).

4. SNR and MSE

Signal-to-Noise Ratio (SNR) and Mean Squared Error (MSE) are two terms that are often used in the domain of signal processing and statistics, especially in the context of communications and data processing. Here is a more detailed explanation of both:

- Signal-to-Noise Ratio (SNR):

SNR is a measure used to quantify the extent to which the desired signal (usually represented by the signal component) differs from the interference or noise (unwanted component) in a system. SNR is usually measured in decibels (dB) and is calculated by the following formula (Safa et al., 2017):

$$SNR_{dB} = 10log_{10}\left(\frac{P\ signal}{P\ noise}\right) \tag{6}$$

Definition :

- P signal is the desired signal power.
- P noise is the interference or noise power.

The higher the SNR value, the better the signal quality. A high SNR indicates that the desired signal has considerable power compared to the noise (Susilo et al., 2019).

- Mean Squared Error (MSE)

MSE is a measure used to quantify how well a model or forecast models the actual data. It is a statistical metric that measures how large the average of the squared difference between the predicted value and the true value is. MSE is often used in the context of regression and estimation, and is calculated by the following formula (Pasaribu et al., n.d.):

$$MSE = \frac{1}{n}\sum_{i=1}^{n}(Yi - \hat{Y}i)^2 \quad (7)$$

Definition :
- n is the number of observations.
- Yi is the true value of the observation (reference signal)
- Ŷi is the value predicted by the model for the observation (comparison signal)

The lower the MSE value, the better the model is at modeling the data, as this means the difference between the prediction and the actual data is low. So SNR is used to measure signal quality by comparing signal to noise, while MSE is used to measure how well a model or prediction is able to model the actual data by calculating the average squared error between the prediction and the actual data (Editor, n.d.).

Methods

The research method used utilizes Matlab software to implement the Kalman Filter Algorithm on noise polluted sound signals, with the aim of reducing noise and improving the quality of the observed sound signals. This simulation will help in testing the effectiveness of the Kalman Filter algorithm in overcoming noise problems in sound signals, as well as understanding its potential use in various audio applications. The process carried out in this research can be seen in the following flowchart image of the Kalman filter simulation.

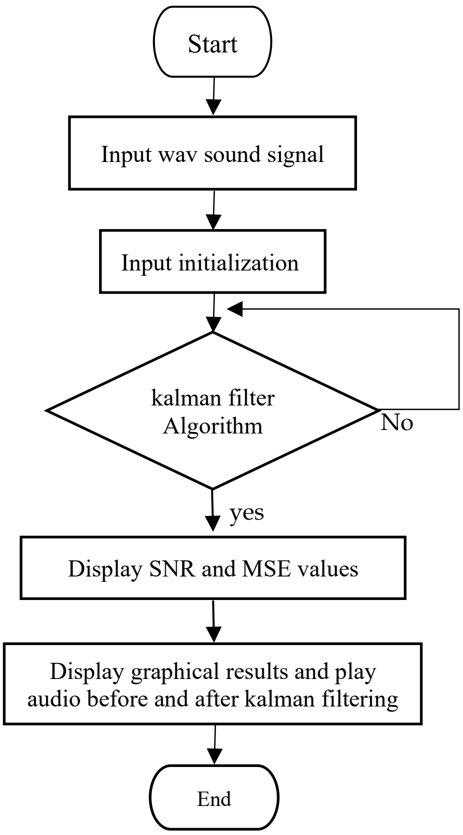


Figure 3. Flowchart of kalman filter simulation

- Input of sound signals and noise signals

In this section, we input the sound signal that will be used to simulate the kalman filter. There is 1 (type) of sound signal, namely the reading of the 1945 proclamation text with a duration of 56 seconds which is polluted by 3 (types) types of noise signals, namely white noise, pink noise and brown noise.

- Input Initialization

At this stage the initialization process is carried out on the sound signal and noise signal. This process is carried out to initialize the input of sound signals polluted by noise signals used for parameters of sound signals and noise signals such as the size of the signal, the sampling frequency of the signal and the signal sampling time.

- Kalman Filter

At the time of the filtering process using the Kalman filter, what is done is the process of filtering noise suppression on the sound signal and noise signal by utilizing feedback from the filtering results performed and then obtaining an error estimate from the signal that is filtered. The Q value is varied from 0.1; 0.01; 0.001; 0.0001; and 0.00001 while the R value of 1 is a significant weight on the actual measurement in the kalman filter update process, but also recognizes a considerable level of uncertainty in the measurement. If in the simulation when the filter process is carried out based on the covariance value given, the results of the noise suppression carried out have not been suppressed properly, thefiltering process will be carried out again, if the noise suppressed gets the results of noise that can be suppressed properly, then proceed to the next stage.

- Show SNR and MSE values

At this stage, displaying the Signal Noise to Ratio (SNR) and Mean Square Error (MSE) values before and after using the kalman filter on the value will be displayed in the command window.

- Display the graph and play the simulated audio

Furthermore, the last process carried out is to display the graph and play audio from the simulated sound signal in the form of a comparison before and after being estimated by the kalman filter.

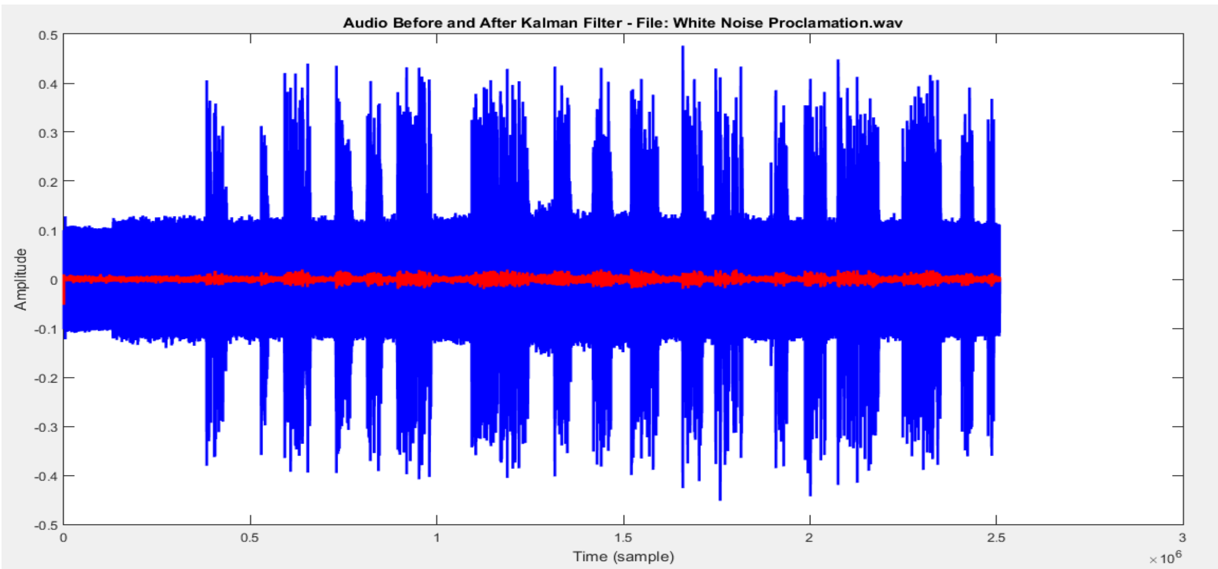
Results and Discussion

1. Simulation Results

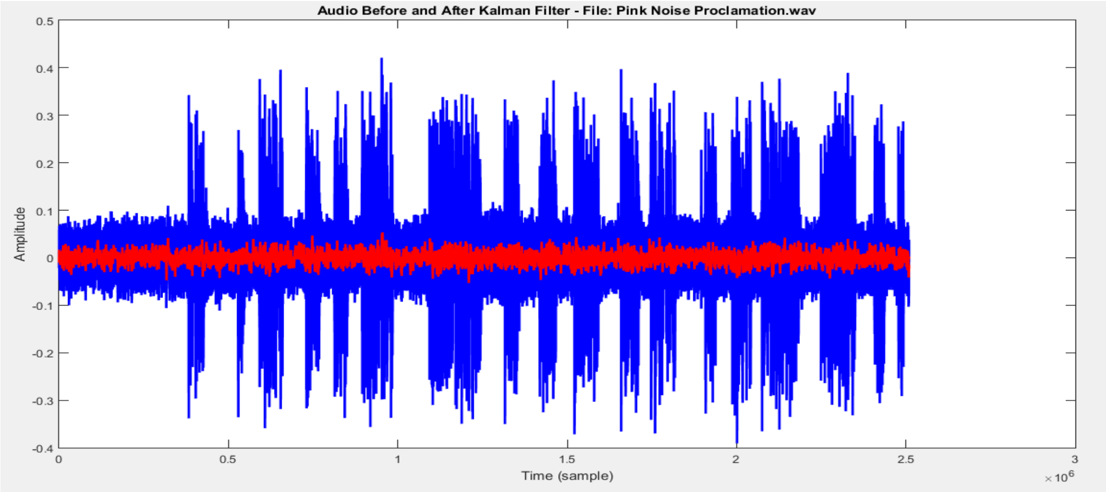
Simulations using MATLAB software to implement the Kalman Filter Algorithm on noise-polluted voice signals resulted in several findings. The following are the results found:

- Noise Reduction

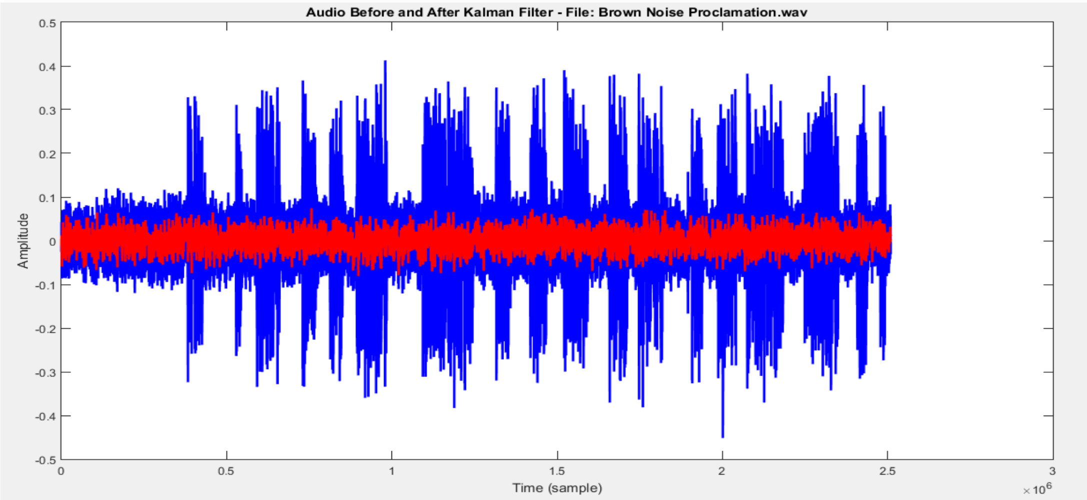
The implementation of the Kalman Filter Algorithm on noise-polluted voice signals successfully reduces the level of noise heard on the signal. The result is a sound signal that is cleaner and free from most of the detected noise interference. The test was conducted with the input of a 56-second proclamation text reading wav sound signal polluted by 3 types of noise, namely white noise, pink noise and brown noise. From the simulation process of noise reduction using a Kalman filter with a Q value that varies from 0.1, 0.001, 0.0001, 0.00001, different filtering graphs are obtained, therefore several graphical displays are selected as representatives. The graphs chosen to represent the results of noise reduction using a Kalman filter with a value of Q = 0.00001 and R = 1 for audio samples of proclamation text readings that are masked by white noise, pink noise and brown noise. The following is a plot of the results of the program simulation. The display of the sound signal polluted by white noise, pink noise, and brown noise signals is marked in blue while the red graph is the signal graph after being estimated by the Kalman filter.



(a) White Noise



(b) Pink Noise



(c) Brown Noise

Figure 4. Simulation result graph

From the results of the noise reduction simulation process using the Kalman filter algorithm in simulation using matlab software. From the graphical display above, it can be seen that the sound signal is polluted by noise, the amplitude is greater than the graph after the Kalman filter. The blue graph accompanies the original sound signal that is polluted with noise so that it affects the output of the original sound signal, From the simulation results after the filtering process using the Kalman filter, it can be seen that the clean signal estimated by the Kalman filter is red with a smaller amplitude.

- SNR Improvement

Measurement of sound signal quality using the Signal-to-Noise Ratio (SNR) metric shows a significant improvement. SNR is the ratio between the original sound signal energy and the noise energy. After the application of the Kalman Filter Algorithm, the SNR increases substantially, demonstrating the effectiveness of the algorithm in reducing noise. At varying Q values ranging from 0.1, 0.001, 0.0001, 0.00001 and R is 1, the relationship between the filter order, Q and R values to the noise reduction results (in terms of SNR value) can be determined. From the simulation results, the highest SNR value is obtained from the filter with a value of Q=0.00001. This result is shown in the SNR value table after being estimated by the kalman filter. The SNR (Signal-to-Noise Ratio) value before the kalman filter estimation is expressed as "Inf" or infinity dB before the Kalman filter process indicates that at that time, the original signal energy (the signal component to be measured) is much greater than the noise energy (the signal component to be eliminated). In other words, the original signal has a significant amplitude while the noise has a very small amplitude in comparison. As a result, the SNR becomes very large and approaches infinity. This is a common situation in the original audio signal. Before the filter or purification process.

Table 1. SNR value after kalman filter

Value Q	White Noise	Pink Noise	Brown noise
0.1	6.0358 dB	3.7019 dB	3.4810 dB
0,01	8.6383 dB	5.7460 dB	5.0765 dB
0.001	14.3313 dB	10.2302 dB	8.0247dB
0.0001	22.2251 dB	14.4646 dB	10.2400 dB
0.00001	30.2809 dB	17.1118 dB	12.3938 dB

The SNR value in the simulation is displayed in the command window with the following command:

```
fprintf('SNR before filter Kalman: %.4f dB\n', snr_after);
fprintf('SNR after filter Kalman: %.4f dB\n', snr_before);
```

It can be concluded based on the SNR value from the simulation results, the kalman filter can work optimally to identify and recognize noise, especially in white noise samples.

- MSE Reduction

The Mean Square Error (MSE) metric is used to measure the error between the noise-polluted speech signal and the clean original speech signal. After the application of the Kalman Filter Algorithm, the MSE decreased significantly, indicating that the estimate produced by the algorithm is closer to the original speech signal. At varying Q values ranging from 0.1, 0.001, 0.0001, 0.00001 and R is 1, the relationship between Q and R values to the results of noise reduction (in terms of MSE value) can be seen. In the simulation results, the lowest MSE value is obtained from the filter with a process noise covariance value of Q=0.1 for each type of noise. These results are shown in the table of MSE values before and after being estimated by the kalman filter. can be seen in the table below.

Table 2. MSE value before kalman filter estimation

Value Q	White Noise	Pink Noise	Brown noise
0.1	0,0064	0,0036	0.0040
0,01	0,0064	0,0036	0.0040
0.001	0,0064	0,0036	0.0040
0.0001	0,0064	0,0036	0.0040
0.00001	0,0064	0,0036	0.0040

Table 3. MSE value after kalman filter estimation

Value Q	White Noise	Pink Noise	Brown noise
0.1	0.0044	0.0020	0.0022
0,01	0.0053	0.0026	0.0027
0.001	0.0061	0.0032	0.0033
0.0001	0.0063	0.0034	0.0036
0.00001	0.0064	0.0035	0.0038

The MSE value in the simulation is displayed in the command window with the following command

```
fprintf('MSE before filter Kalman: %.4f\n', mse_before);
fprintf('MSE after filter Kalman: %.4f\n', mse_after);
```

It can be concluded based on the MSE value of the simulation results, the kalman filter can work optimally to identify and recognize noise, especially in white noise samples.

2. Discussion

The use of the Kalman Filter Algorithm in noise reduction of voice signals has proven effective in this simulation. Here are some important points that can be discussed:

- Effectiveness of Kalman Filter Algorithm

This simulation proves that the Kalman Filter Algorithm is an effective method for reducing noise in speech signals. This is due to its ability to combine good predictions about the original speech signal with the information provided by measurements of the signal polluted by noise. Some parameters of the Kalman filter that affect the MSE and SNR values in the noise reduction process using the Kalman filter are the process noise covariance (Q), and the measurement noise covariance (R). Increasing the order of the adaptive system will affect the system to produce minimum MSE and greater output SNR. The selection of the Q value should be very small while the selection of the R value should be larger, with the aim that the measurement is more adaptive to produce a minimum MSE value and an output SNR value that is greater than the input SNR value.

- Relevance for Audio Applications

The results of this simulation have significant relevance in audio applications such as telecommunications, music recording, and sound processing. Effective noise reduction can improve the quality of voice communication and audio recording.

- Use of Matlab
  The use of Matlab software simplifies the testing and implementation of the Kalman Filter Algorithm. Matlab provides various tools for speech signal analysis and algorithm modeling, making it a powerful platform for the development of noise reduction techniques.
- Challenges and Limitations
  This simulation may have some limitations depending on the noise model used and the Kalman Filter parameters chosen. Therefore, it is important to consider that the results may vary depending on the actual context of use. In conclusion, this research shows that the use of Kalman Filter Algorithm in reducing noise in speech signals is a promising approach. This simulation provides a basis for further development in noise reduction in audio applications and can be a positive contribution in improving the quality of sound signals in various contexts.
  Therefore, this study shows that the Kalman Filter Algorithm is a promising method for reducing noise in speech signals. Its use in audio applications has the potential to significantly improve audio quality. This simulation can serve as a foundation for further development in noise reduction in audio applications, and further research can deepen our understanding of the potential and limitations of this algorithm in the processing of noise-polluted sound signals.

Conclusion

This research explores the use of Kalman Filter Algorithm with the help of Matlab software to reduce noise polluted voice signals. The simulation results show some important findings that can be taken as conclusions:

1. The Kalman Filter Algorithm proved to be effective in reducing the noise level in the polluted speech signal. Its use produces a sound signal that is cleaner and closer to the original sound signal.
2. The simulation resulted in a significant increase in the Signal-to-Noise Ratio (SNR) metric and a decrease in the Mean Square Error (MSE), indicating an improvement in the quality of the speech signal after the use of the algorithm.
3. These results have great relevance in various audio applications, including telecommunications, music recording, and sound processing. Effective noise reduction can enhance the audio experience in these contexts.
4. Matlab software facilitates the implementation of the Kalman Filter Algorithm and its performance testing. Matlab is a powerful tool for the development of noise reduction techniques in sound signal processing.
5. It is important to note that the results of these simulations may be affected by the noise model used and the Kalman Filter parameters chosen. Users should consider the limitations and suitability of the algorithm to the actual application context

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