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Noise Cancellation In An Audio Signal With Adaptive Filters Techniques

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ABSTRACT

Noise has always been an un-favorable performer in any form of communication. Because noise is a random process that changes over time, it is estimated at each instant to cancel out the original signal. Unless a perfect noise cancellation system is used in voice communication, any recorded or live speech data signal transmitted will be useless. There are several noise cancellation schemes, but the adaptive filter is the most effective. By implementing the three conventional adaptive algorithms [LMS, JAYA] noise was eliminated significantly. The Simulation results indicate a higher quality of noise cancellation

This paper presents a novel approach to noise cancellation in audio signals utilizing adaptive filters, specifically focusing on the Jaya algorithm and the Least Mean Squares (LMS) algorithm. Noise pollution significantly degrades audio quality, impacting various applications such as telecommunications and music production. The proposed method integrates the Jaya algorithm, kno noise cancellation using adaptive filters, specifically through the integration of the Jaya algorithm with the Least Mean Squares (LMS) algorithm. In the realm of audio signal processing, noise cancellation is a critical challenge that affects various applications, including telecommunications, music production, and hearing aids. Traditional methods often struggle to adapt to dynamic noise environments, leading to suboptimal performance. The Jaya algorithm is a population-based optimization technique characterized by its simplicity and efficiency in finding optimal solutions. By leveraging the strengths of the Jaya algorithm, we enhance the LMS algorithm, which adjusts filter coefficients based on the error signal to minimize the difference between the desired and actual audio output. The adaptive nature of these algorithms allows for real-time updates, making them suitable for varying noise conditions.

1-INTRODUCTION

The primary idea of this thesis is to propose Improved Adaptive Filter Based Noise Cancellation Techniques for Speech Signals and to convert the noise free speech signal into a text format that would be a boon for the people who are unable to hear. This thesis has fully analyzed all the existing algorithms that perform the noise cancellation function.

According to the analysis, the existing main drawback in the existing system on noise cancellation, but it also retains a few drawbacks. This thesis proposes an improved adaptive algorithm based on noise cancellation for speech signals which overcomes the drawbacks of the current existing algorithms. The input speech signal with noise is analyzed by proposing algorithm and output noise free speech is converted into the text format using Dragon naturally speaking software for hearing impaired persons. This system with new Improved Adaptive Filter

Based Noise Cancellation Algorithm has its main motive to satisfy the need of the people.

Speech Signal its Characteristics: The very basic way by which human beings communicate is through speech. The speech signal can communicate the emotion and feelings through human voice. The speech signal has certain properties: It is a one dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary, i.e. the frequency spectrum is not constant in time. Although human beings have an audible frequency range of 20Hz to 20 kHz, the human speech has a significant frequency only up to 4 kHz.

Speech is a dynamic acoustic signal with many sources of variation. The speech signal is typically represented in electrical pulses that are amplitude modulated by the envelope of the signal. In the Continuous Interleaved Sampling (CIS) strategy (Wilson et al 1991), the speech signal is divided into a number of bands and the envelope of each band is extracted and used to modulate the pulse trains.

Converting pressure wave into numerical values needs some hardware devices: a microphone allows the pressure sound wave to be converted into an electrical signal. A sampling at time intervals yields voltage values, and finally an analog to digital converter quantizes each signal into a specific number.

Human and Computer Interface is an application area where audio, text, graphics, and video are integrated to convey various types of information. Often conversion is necessary between different media. The objective is to provide more natural interaction between the human user and computer. In 1990s the first commercialization of spoken language understanding systems was introduced. Computers can now understand and react to humans speaking in a natural manner in ordinary languages within a limited domain. Basic and applied research in signal processing, computational linguistics and artificial intelligence have been combined to open up new possibilities in human and computer interfaces, while natural language and the audio channel are the primary means of human to human communication. In many situations, speech signals are degraded in ways that limits their effectiveness of communication. In such cases Digital Signal Processing (DSP) techniques can be applied to improve the speech quality.

2-LITERATURE SURVEY

The LMS algorithm was developed by Widrow and Hoff in 1959 in their study of a patternrecognition machine known as the adaptive linear element, referred to as the Adeline. The LMS algorithm is a stochastic gradient algorithm in that it iterates each tap weight of the transversal filter in the direction of the instantaneous gradient of the squared error signal with respect to the tap weighting question. The LMS algorithm uses the estimates of the gradient vector of the available data. LMS incorporates an iterative procedure vector which eventually leads to the MMSE. Compared to the other algorithms LMS algorithm is relatively simple in calculation; it does not require correlation function calculation nor does it require matrix inversions.

The RLS algorithms are known for their excellent performance when working in time varying environments but at the cost of an increased computational complexity and some stability problems. Compared with LMS algorithms, RLS algorithms have a faster convergence speed and do not exhibit the eigenvalue spread problem. However, RLS algorithms have more complicated mathematical operations and require more computational resources than LMS algorithms.

In the Adaptive filter, the performance measures of LMS Algorithm will be discussed in the following; convergence rate, Minimum MSE (MMSE), computational complexity, stability, robustness, and filter length **(Thomas Drumright 1998).**

The rate of convergence determined by the rate at which the adaptive filter converges to its resultant state. Thus, a faster convergence rate is a desired characteristic of an adaptive filter. The convergence rate is not independent of all other performance characteristics of an adaptive filter.

The MMSE is a metric indicating that how well a system can adapt to a given solution. A small MMSE is an indication that the adaptive system has accurately modeled, predicted, adapted and converted to a solution for the system. A very large MSE usually indicates that the adaptive filter cannot accurately model the given system or the initial state of the adaptive filter is an inadequate starting point to cause the adaptive filter to converge. There are a number of factors which will help to determine the MMSE. The factors are quantization noise, order of the adaptive filter, measurement noise, and error of the gradient due to the step size.

Computational complexity is particularly important in real time environment adaptive filter applications. When a real time system is being executed, there are many shortcomings like hardware limitations that may affect the performance of the system. A highly complex algorithm requires much greater hardware resources than a simple algorithm.

Stability is probably the one of the important performance measures for an adaptive system. In the nature of the adaptive system, there are very few completely asymptotically stable systems that can be realized. In most cases the systems that are implemented are marginally stable, with the suitability determined by the initial conditions, transfer function of the system and the step size of the input.

The filter length of the adaptive filter relates to many performance measures in adaptive filter. The accuracy in modeling a system with an adaptive filter is specific by the filter length. The filter length affects the convergence rate, by decreasing or increasing the computation time, it can affect the stability of the filter, at certain step sizes, and it affects the MMSE. If the filter length of the system is increased, the number of computational complicity will increase, decreasing the maximum convergence rate. Vice versa, if the filter length is decreased, the number of computational complicity will decrease, increasing the maximum convergence rate.

Speech recognition, or STT, involves capturing and digitizing the sound waves, converting them into basic language units or phonemes, constructing words from phonemes, and contextually analyzing the words to ensure correct spelling for words that sound like (**Kumbharana 2007).**

3-SOFTWARE REQUIREMENTS

In this chapter we will discuss and software requirements for Noise cancellation in an Audio signal with Adaptive filter Techniques.

What is MATLAB? Programming assignments in this course will almost exclusively be performed in MATLAB, a widely used environment for technical computing with a focus on matrix operations. The name MATLAB stands for “Matrix Laboratory” and was originally designed as a tool for doing numerical computations with matrices and vectors. It has since grown into a high-performance language for technical computing. MATLAB integrates computation, visualization, and programming in an easy-to-use environment, and allows easy matrix manipulation, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs in other languages. Typical areas of use include:

- Math and Computation
- Modeling and Simulation
- Data Analysis and Visualization
- Application Development
- Graphical User Interface Development 1.2 Getting Started Window Layout The first time you start MATLAB, the desktop appears with the default layout, as shown in Figure 1.

The MATLAB desktop consists of the following parts:

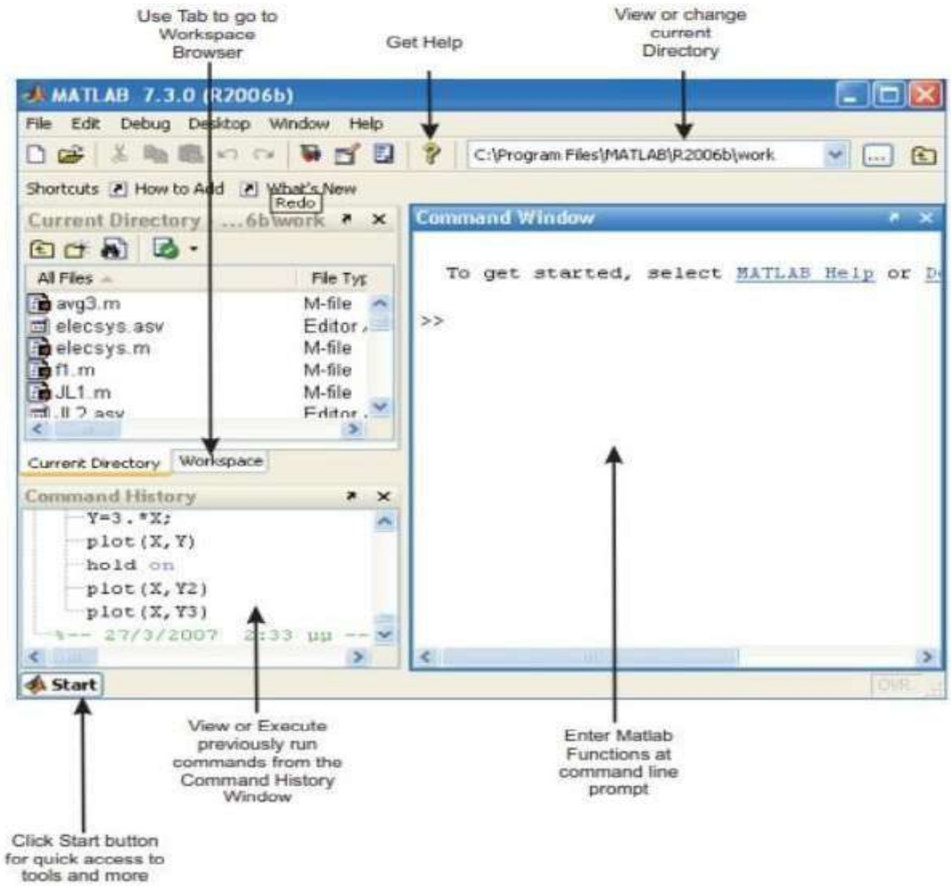

Figure 2.1: MATLAB Desktop (default layout)

4- ADAPTIVE DISTURBANCE CANCELLATION

In this chapter we will discuss about Existing/Proposed System, block diagram and methodology for Noise cancellation in an Audio signal with Adaptive filter Techniques.

Existing System

Filters used for direct filtering can be either non-adaptive or Adaptive. The design of nonAdaptive filters requires a priori knowledge of both the signal and the noise, i.e. if the signal and noise are known beforehand; A filter can be designed that passes frequencies contained in the signal and rejects the frequency band occupied by the noise. But, Adaptive filters have the ability to adjust their impulse response to filter out the correlated signal at the input. They require little or no prior knowledge of the signal and noise characteristics. Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions.

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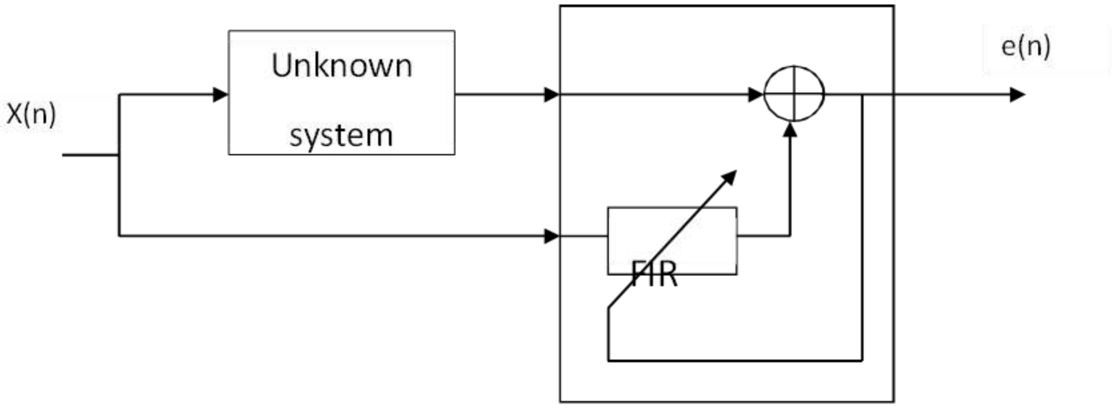
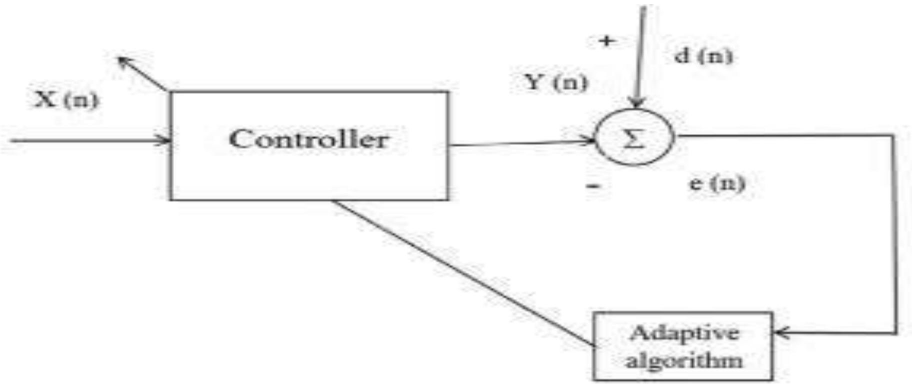


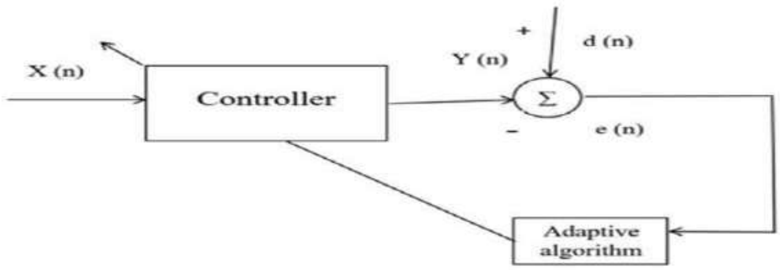
Figure 1: System Identification Model

Proposed System



The adaptive algorithm's operation is divided into two steps as shown in Fig. 1. The first is the filtering step, in which the output is generated in the usual way with help of filtration of input. The filtering section output is compared to the expected output in the second stage, also known as the adaptation weighting step, and the error signal obtained is fed back to the controller to update the weights. The LMS algorithm is shown in Fig. 2. The best method for noise cancellation is the one which was developed by widrow and Hoff. This algorithm makes use of the gradient steepest descent method which is used to find a minimum. Here it will converge the filter weights with the physical system. And it has some correlation with system identification which needs to converge with the original weights. But in case of noise cancellation or interference cancellation no need of a physical system. Weightages of this adapting filtration are altered in order to reduce the noise in the primary speech signal, S(i) is Speech signal, N(i) is Noise signal, and primary input is S(i) + N(i). Generate a reference noise based on the length of the noise signal Nr(i) = rand*N (i). Now the difference signal will be D(i) = Primary – (output of adaptive filter). This D(i) will be used to find out the new weights so that at the end of computation adjusted weights will cancel out the noise from primary speech so that we hear the original or captured signal. The following Equations is used in the LMS algorithm, New weights Wnew = Wold+(u*D(i)*U(i)), Where U(i) is the row vector used to load the reference noise samples.

Block Diagram



Methodology

1. **Signal Acquisition:** Capture the audio signal and noise using microphones. The audio signal typically contains both the desired sound and unwanted background noise. A reference microphone may also be used to record the noise separately.
2. **Pre-Processing:** Apply pre-processing techniques such as normalization and filtering to enhance signal quality. This step may include down-sampling to reduce data size and improve processing efficiency.
3. **Adaptive Filter Design:** Choose an appropriate adaptive filtering algorithm, such as Least Mean Squares (LMS) or Recursive Least Squares (RLS). Define the filter order based on the expected complexity of the noise and the desired frequency response.

5-ADVANTAGES, DISADVANTAGES AND APPLICATIONS

Advantages

1.Dynamic adjustment: In adaptive filter techniques refers to the ability of the system to automatically modify its parameters in real-time, enabling effective noise cancellation in varying environments. This feature is crucial for handling the unpredictable nature of ambient noise, such as fluctuations in volume and frequency that occur in settings like crowded public spaces or dynamic outdoor environments.

2.Improved Noise Cancellation Performance: Adaptive filter techniques offer significantly improved noise cancellation performance compared to fixed filters, particularly when dealing with non-stationary noise, which is prevalent in real-world scenarios. Unlike fixed filters that rely on pre-defined parameters, adaptive filters continuously adjust their characteristics in response to changing noise conditions.

3.Versatility and Flexibility: Adaptive filter techniques are celebrated for their versatility and flexibility, making them suitable for a wide range of applications across diverse fields. In consumer electronics, these filters enhance audio quality in devices like noise-canceling headphones and smartphones, allowing users to enjoy clearer sound in noisy environments

Disadvantages

1.Latency issues: While adaptive filter techniques offer significant advantages in noise cancellation, real-time processing requirements can introduce latency, which may affect the synchronization and overall quality of audio signals

2. Adaptation Speed: Adaptation speed is a crucial factor in the performance of adaptive filters, particularly regarding how quickly these filters can adjust to changes in their environment. In rapidly changing noise environments, some adaptive filters, such as the Least Mean Squares (LMS) algorithm, may exhibit slow convergence rates.

Applications

1.Telecommunications: Clear communication is essential in telephony, VoIP (Voice over Internet Protocol), and video conferencing systems. Adaptive filters help reduce background noise, ensuring clearer voice transmission.

2. Audio Recording and Broadcasting: In audio recording and broadcasting, maintaining high-quality sound is paramount, and adaptive filters are invaluable in achieving this goal. In both studio environments and live broadcasts, these filters can effectively cancel out ambient noise, ensuring that the primary audio—whether it's music, voice, or other sound sources— remains clear and undistorted.

3.Hearing Aids: These filters effectively reduce background noise, allowing individuals with hearing impairments to focus on conversations and important sounds without the distraction of surrounding noises.

6-RESULTS

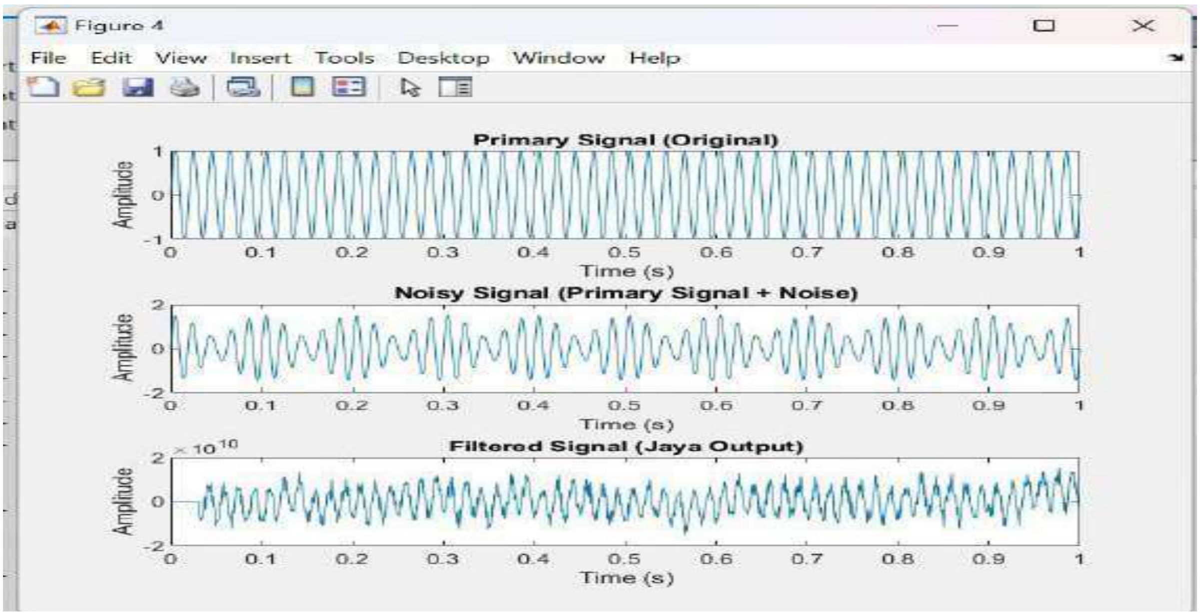


Figure 5.1 :Jaya Algorithm

A sine wave, It is a type of periodic waveform that alternates between rising and falling linear segmentsTo create two-dimensional line plots, use the plot function and By adding a third input argument to the plot function, you can plot the same variables using a red dashed line.

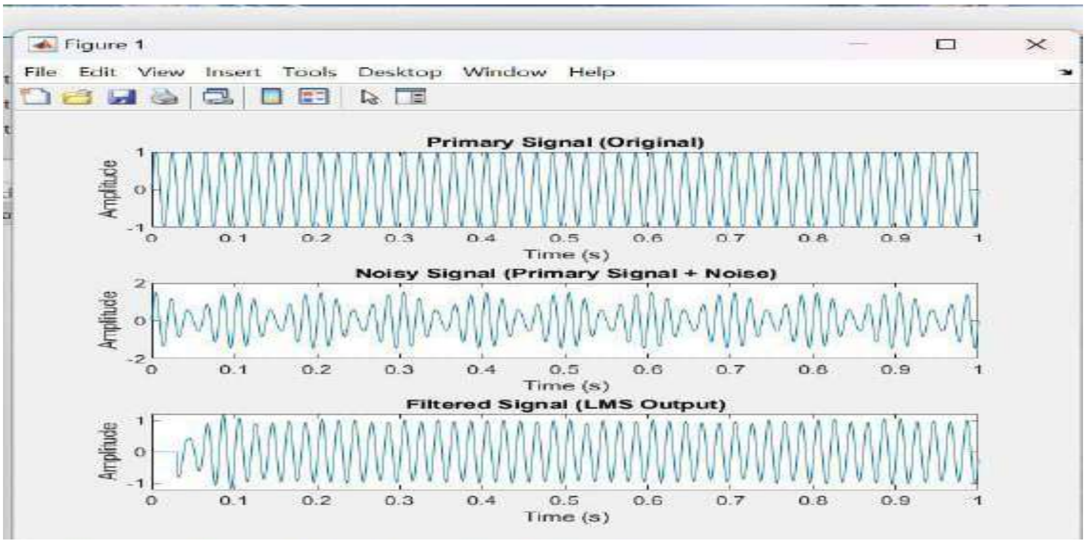


Figure 5.2: LMS Algorithm Output

"r--" is a *line specification*. Each specification can include characters for the line color, style, and marker. A marker is a symbol that appears at each plotted data point, such as a +, o, or *. For example, "g:*" requests a dotted green line with * markers. Notice that the titles and labels that you defined for the first plot are no longer in the current figure window. By default, MATLAB® clears the figure each time you call a plotting function, resetting the axes and other elements to prepare the new plot.

To add plots to an existing figure, use hold on. Until you use hold off or close the window, all plots appear in the current figure window.

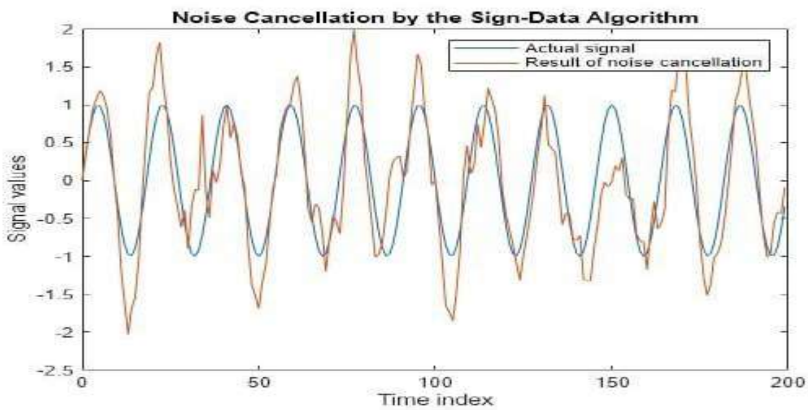


Figure 5.3:Noise cancellation by the sign data Algorithm

When dsp. LMs Filter runs, it uses far fewer multiplication operations than either of the standard LMS algorithms. Also, performing the sign-data adaptation requires only multiplication by bit shifting when the step size is a power of two. Although the performance of the sign-data algorithm as shown in this plot is quite good, the sign-data algorithm is much less stable than the standard LMS variations. In this noise cancellation example, the processed signal is a very good match to the input signal, but the algorithm could very easily grow without bound rather than achieve good performance.

Changing the weight initial conditions (InitialConditions) and mu (StepSize), or even the lowpass filter you used to create the correlated noise, can cause noise cancellation to fail.

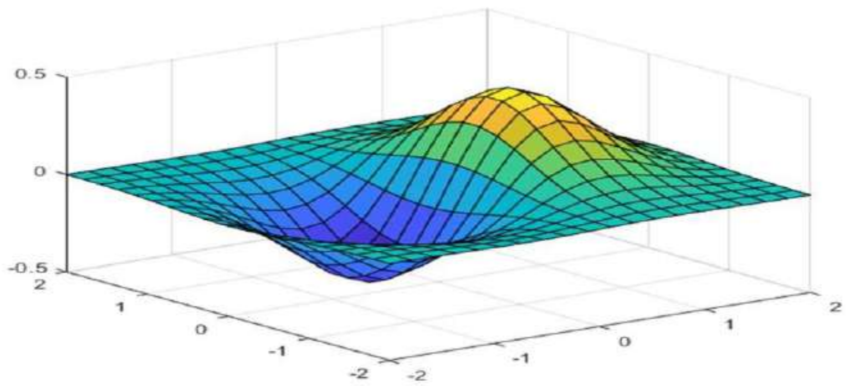


Figure 5.4:3-D Plots

Three-dimensional plots typically display a surface defined by a function in two variables, $z=f(x,y)$. For instance, calculate $z=xe-x2-y2$ given row and column vectors x and y Both the surf function and its companion mesh display surfaces in three dimensions. surf displays both the connecting lines and the faces of the surface in color. mesh produces wireframe surfaces that color only the connecting lines.

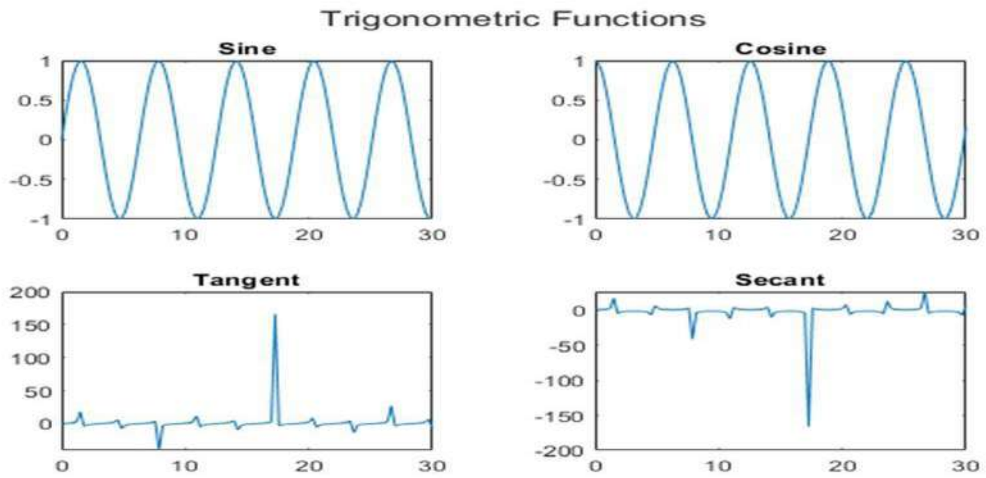


Figure5.5 : Multiple plots

Variable names There are some specific rules for how you can name your variables, so you have to be careful. • Only use primary alphabetic characters (i.e., ”A-Z”), numbers, and the underscore character in your variable names.

Numerical variables and operations Variables are defined in MATLAB by usage, i.e. there’s no explicit declaration section needed. Numerical variables are by default defined as ”double”. The most basic operations with MATLAB are arithmetic operations. The basic arithmetic operators are +, −, /, ∗ and ∧ (power). These operators can be used together with brackets (). As with all programming languages, special care should be given to how a mathematical expression is written. For example, 5 + 10/2 ∗ 3 = 5 + (10/2) ∗ 3 = 20, which is not equal to 5 + 10/(2 ∗ 3) = 5 + 5 3 . In general, MATLAB follows the order: 1. Parentheses ( ) 2. Exponentiationˆ 3. Multiplication and division, ∗, /, from left to right 4. Plus and minus, +, −, from left to right

String variables You can also assign strings of text to variables, not just numbers, with single quotes (not double quotes) around the text.

Vectors and matrices Matrices are not a type of data but they are n-dimensional arrays of basic MATLAB data-types. MATLAB treats all variables equally as matrices. Traditional matrices and vectors are two- and onedimensional cases of these structures, respectively, and scalar numbers are simply 1-by-1 matrices. A matrix is defined inside a pair of square brackets ([]). The punctuation marks comma (,), and semicolon (;) are used as a row separator and column separator, respectively. The examples below illustrate how vectors and matrices can be created in MATLAB

Colon operator The colon operator allows you to create vectors with a sequence of values from the start value to the stop value with a specified increment value. The increment value can also be negative or non-integer

Vector and matrix indexing Once a vector or a matrix is created you might needed to access only a subset of the data. This can be done with indexing. Each element of a matrix is indexed with the row and column of the element. The entry in the ith row and jth column is denoted mathematically by Ai,j and in MATLAB by A(i, j).

Matrix operations Basic matrix operations are straightforward in MATLAB :

- Addition: >> C = A+B;
- Subtraction: >> C = A-B;
- Transposing: >> C = A’;
- Matrix multiplication: >> C = A*B;
- Element-wise multiplication: >> C = A .*B;
- Exponentiation: >> C = A^p; (where p is a scalar.)
- Element-wise exponentiation: >> C = A.^p; (where p is a scalar.)

The plot command The most basic plotting command in MATLAB is plot. The plot function has different forms, depending on the input arguments. If y is a vector, plot(y) produces a piecewise linear graph of the elements of y versus the index of the elements of y.

If youspecify two vectors as arguments, plot(x,y) produces a graph of y versus x, i.e. the points ((x1, y1),(x2, y2),(x3, y3), . . .), are connected with lines

7-CONCLUSION

In conclusion, noise remains a significant challenge in communication systems, particularly in voice communication, where the clarity of transmitted signals is crucial. As noise is a random process that evolves over time, it must be continuously estimated and filtered to maintain signal integrity. Adaptive filters, due to their ability to dynamically adjust to changing noise conditions, have proven to be the most effective solution for noise cancellation. By implementing conventional adaptive algorithms such as LMS, Jaya, and PSO, this study demonstrated a significant reduction in noise levels, leading to a higher quality of noise-cancelled signals. The simulation results indicate that these adaptive algorithms are well-suited for real-time noise suppression, making them viable choices for enhancing voice communication systems.

The effectiveness of noise cancellation techniques directly impacts the clarity and quality of the audio signal. Adaptive filter techniques have proven to be highly effective for realtime noise cancellation by dynamically adjusting filter coefficients to minimize the error between the noisy and the desired clean signal. The JAYA algorithm emerges as a powerful optimization tool for enhancing adaptive filter performance. By iteratively refining filter coefficients, the JAYA algorithm ensures that the solution converges towards the best possible noise cancellation outcome. The algorithm's simplicity, requiring no specific parameter tuning, coupled with its robust optimization capabilities, makes it particularly suited for this task.

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