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Improving the Quality of an Audio Signals Using Adaptive Filter

A.O. Afolabi, O. Ekundayo and M. Ogundare
Department of Computer Science and Engineering,
Ladoke Akintola University of Technology, Ogbomoso, Nigeria

Abstract: Several factors are responsible for signal corruption but the most common factors are attenuation distortions, delay distortion and noise. Noise is a major limiting factor in communication system performance and they are caused by several means over various channel. In this research, a model was developed by the use of an application in MATLAB called simulink to simulate the effect of distortion on signals and adaptive filtered was also modeled to filter the noise that constitute the distortion. It was found out that by the use of an adaptive filter noise cancellation could be effected. The results are graphically displayed by the various graphs of the input signal, signal with noise and the filtered output.

Key words: Audio signals, adaptive filter, improving, quality, communication

INTRODUCTION

Research works have shown that digitalized form of data transmission is better compared to the analog form because they are easy to read and manipulate in discrete forms. Though most input signals usually come in an analog form e.g. audio signal, they therefore need to be converted to a digital form using an analog to digital converter.

When sending or transmitting data in an analog or digital form from one source to another, transmission impairments are bound to occur, transmission impairments are factors that cause signal distortions in a communication system. Such factors include attenuation and attenuation distortion, Delay distortion and noise of different types (John, 1979).

This study is geared towards solving the problem of signal distortion in data transmission which involves the use of a filter processor. Two main types of filters exist-Analog and Digital filters. However, digital filters have a better performance on signal regeneration and reconstruction and are therefore, mostly used for filter system designs (Jackson, 1992). An adaptive filter is used for the implementation of the digital filter system because it is very effective for noise cancellation and signal regeneration thus an adaptive filter algorithm called the Least Mean Square (LMS) algorithm is used for this implementation. The effect of this filter on the audio input signal is illustrated on a scope with varying input frequency in the audio bandwidth of 0-4 khz (Oppenheim and Schafer, 1989).

MATERIALS AND METHODS

The analog signal input considered in this research is an audio signal (sensory data signal). However, when it passes through the required medium in a communication system such as a microphone it is bound to be corrupted or distorted by some factors in the transfer medium in which it passes through consisting the equipment, machines and other channel devices thereby constituting a corrupted signal. The major impairment that causes corruption of signal is noise (Oppenheim and Schafer, 1989) and it is the factor considered in this research.

The process that was adopted in carrying out this effect is by studying the analog input signal or frequency range of an audio signal and their effects when subjected to a particular condition represented by a transfer function. Response of the audio input signal to filtration is achieved by using a digital filter suitable to acquire the desired output signal. An application in MATLAB called simulink handles the modeling of a digital filter transmission system, operates on the system and gives out the output result.

Model design function: The model designed has three main functions which it performs:

- Allows a range of frequency input signal from the DSP source block with a specified bandwidth and outputs the original signal through a DSP source (scope).

Corresponding Author: A.O. Afolabi, Department of Computer Science and Engineering, Ladoke Akintola University of Technology, Ogbomoso, Nigeria

- Noise is fed into a FIR low pass noise filter to reduce excess distortion, the noise is mixed with the original signal to produce a "Signal + Noise" signal.
- The noise produced is fed into the adaptive filter which gives a filtered noise output. The adaptive filter uses the reference signal on the "In" port and the error signal on the "Err" port to automatically match the filter response in the Noise filter block. As it converges to the correct filter, the filtered noise should be completely subtracted from the "Signal + Noise" signal and the "Error Signal" should contain only the original signal.

RESULTS AND DISCUSSION

In this project, analog voice signals are transmitted over free air, typically of long distance telecommunications.

ASCII, a telecommunication body has set up a standard for long distance communication. First, all long distance communications should be in digital form because digital communication offers advantages such as lower noise, greater space between signal regenerators and compatibility with digital systems. For every long distance communication, digital communication is the standard.

In digital communication, there has to be a conversion from the original analog signal to be sent to its digital equivalent (Parks and Burrus, 1987). This is usually done by an Analog to Digital Converter (ADC). The conversion process involves the following steps:

Sampling: This is when the analog signal (in form of a sine wave) is taken at equal intervals, depending on the sampling process/method involved. Nyquist, a scientist specified that for a fine sample of an analog signal, the sampling frequency must be at least twice the highest frequency contained in the signal to be sampled. For voice, the frequency varies from 300-3400 Hz. It is approximated, usually to 4000 Hz. Therefore to sample voice effectively, the sampling frequency must be 8000 Hz (8 KHz) at the least. Mathematically, sampling frequency (fs); $fs = 2f_B$, where f_B highest frequency component in base band signal.

Quantization: This is the step after sampling. The entire amplitude of the analog signal is broken into a number of finite steps. Each flat top of each sample is then drawn to meet the vertical axis (amplitude). Depending on the number of steps the decimal, coinciding with each flat top of each sample or closest to it is converted into binary number. This is the encoding part. First, the ASCII has set a standard of 8 bits (total) for encoding of every signal. This a affects the number of quantization steps, according to the equation; $m = 2^n$ where m = no. of steps and n = no. of bits

Therefore, for 8 (as a standard of ASCII) the number of quantization steps $m = 2^8 = 256$ steps.

Encoding: According to the ASCII standard, each sample is to be encoded with 8 bits (1 byte). Most of the noise that is produced in a digital signal is as a result of quantization steps. That is why for the greater number of quantization steps, the approximation (of each flat top) to meet a decimal while numbers on the amplitude is minimized. However, the greater the number of steps, the more the number of bits used to encode it and the greater the bandwidth used to transmit it, the more expensive the communication system becomes. That is why ASCII specifies an 8 bit standard.

Filtering: The encoded signal is filtered to remove any signal outside the band so that the signal becomes band limited.

Now, at the receiving end, noise is filtered from the signal using filters as it is converted back to analog form, usable to the end user.To calculate the signal to noise ratio the following equation are needed.

Signal to noise ratio in decibytes (db): (S/N) = 1.761 + 6.02n (where n = 8 bits). This is the standard equation

$$S/N\ (db) = 1.761 + (6.02 * 8) = 49.92\ db$$

$$\text{also; } S/N = 3/2\ m^2;\ \text{where } m = 256$$

$$\text{therefore, } S/N = 3/2 * 256^2 = 98304$$

The signal must be 98304 times greater than the noise in it. This however, implies that

$$S/N = 98304;\ \text{therefore, } S = 98304$$

Calculating signal to noise ratio in percentage: Calculating this in percentage (%); S = 98304/98305 * 100 = 99.99%

$$N = 1/98305 * 100 = 0.001\%$$

These are the acceptable values and this why adaptive filters are useful in reducing the noise level, making noise very small compared to the signal required. The digital filter system model illustrates the ability of a Least Mean Square (LMS) adaptive filter to extract useful information from a noisy signal. The information bearing signal is a sine wave that is corrupted by additive white Gaussian noise. The adaptive noise cancellation system

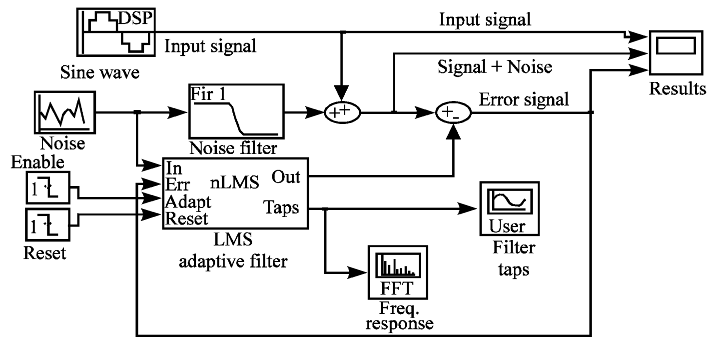

Fig. 1: Diagram showing the adaptive filter system

assumes the use of two microphones. A primary microphone picks up the noisy input signal, while a secondary microphone receives noise that is uncorrelated to the information bearing signal, but it is correlated to the noise picked up by the primary microphone.

The Digital Signal Processing (DSP) block from which the input signal generated contains an Analog-to-Digital (A/D) converter which samples the continuously varying amplitudes of the analog audio signal (Proakis amd Manolakis, 1996). The continuous analog signal was sampled at a rate of twice the highest frequency present in the spectrum of the sampled analog signal in order to accurately recreate the analog audio signal from the discrete samples. The analog audio signal was mixed with noise using a sum block which is bound to occur when the audio signal passes through the communication channel. The noise was however, first low-pass filtered using a finite impulse response filter to make it finite in bandwidth. The FIR noise filter was observed to have little or no significant effect on the “signal with noise” signal, which therefore, implied that a significant amount of noise was still present this is seen from the scope.

Due to this impending problem, an adaptive filter was introduced. The noise was fed into the adaptive filter The LSM adaptive filter uses the reference signal on the "In" port and the error signal to automatically match the filter response in the block representing the "Noise filter". As it converges to the correct filter, the filtered noise is completely subtracted from the "Signal + Noise" signal which generates the "Output Signal” containing the original signal with a minimal amount of noise as shown in Fig. 1. The output signal generated is close to the input signal.

The adaptive filter algorithm: LMS adaptive filter algorithm is simplest and by-far-the-most-commonly-used adaptive filter algorithm, the information bearing signal is a sine wave of 0.055 cycles/sample (Fig. 2).

signal = sin(2*pi*0.055*[0:1000-1]'); plot (0:199, signal (1:200)); grid; axis ([0 200-2 2]); title ('The information bearing signal').

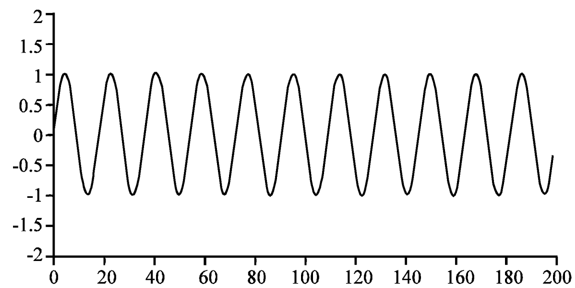

Fig. 2: Plot showing the input signal

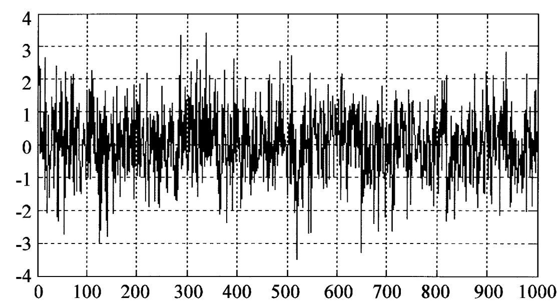

Fig. 3: Plot showing noise picked up by the secondary device

The noise picked up by the secondary microphone is the input for the LMS adaptive filter as shown in Fig. 3. The noise that corrupts the sine wave is a low pass filtered version of (correlated to) this noise. The sum of the filtered noise and the information bearing signal is the desired signal for the adaptive filter.

- Nvar = 1.0; % Noise variance.
- Noise = randn (1000, 1)*nvar; % White noise.
- Plot (0:999, noise).
- Title ('Noise picked up by the secondary microphone').
- Grid; axis([0 1000 -4 4]).

The noise corrupting the information bearing signal is a filtered version of 'noise' as shown in Fig. 4:

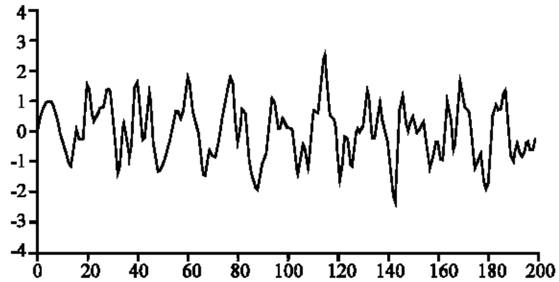

Fig. 4: Noise corrupting the information bearing signal

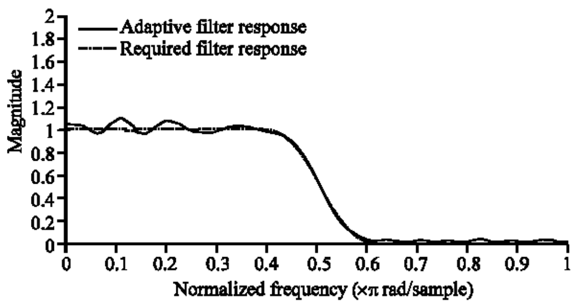

Fig. 5: Convergence of the adaptive filter response to the response of the FIR filter

- Nfilt = fir1(31,0.5); % 31st order Low pass FIR filter.
- Fnoise = filter(nfilt,1,noise); % Filtering the noise "Desired signal" for the adaptive filter (sine wave + filtered noise): d = signal + noise; plot (0:199, d (1:200)).
- Grid; axis([0 200 -4 4]); title ('Desired input to the Adaptive Filter = Signal + Filtered Noise'.

Set and initialize LMS adaptive filter parameters and values:

- M = 32; % Filter order.
- Lam = 1; % Exponential weighting factor.
- Delta = 0.1; % Initial input covariance estimate.
- w0 = zeros(M,1); % Initial tap weight vector.
- P0 = (1/delta)*eye(M,M); % Initial setting for the P matrix.
- Zi = zeros(M-1,1); % FIR filter initial states.
- S = initrls (w0, P0, lam, Zi); Running the LMS adaptive filter for 1000 iterations. The plot shows the convergence of the adaptive filter response to the response of the FIR filter [y,e,S] = adaptrls (noise, d, S); H = abs (freqz (S. coeffs, 1, 64)).
- H1 = abs(freqz(nfilt,1,64)); wf = linspace(0,1,64).
- plot (wf,H,wf,H1); xlabel('Normalized Frequency (\times\pi rad/sample)').

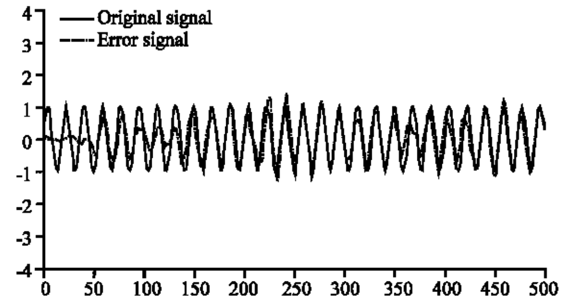

Fig. 6: Plot showing the original signal

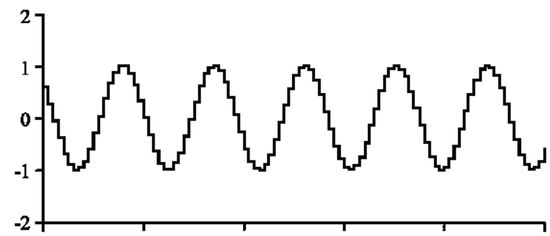

Fig. 7: Input signal

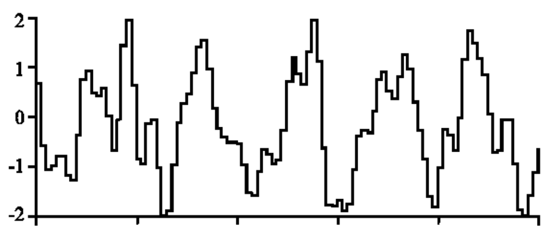

Fig. 8: Signal with noise

- Ylabel('Magnitude'); legend('Adaptive Filter Response', 'Required Filter Response'); grid; axis ([0 1 0 2]).

Figure 5 shows that the adaptive filter converges, the filtered noise should be completely subtracted from the "signal + noise" signal and the error signal should contain only the original signal. Plot (0:499, signal (1:500), 0:499, e(1:500)); grid; axis([0 500 -4 4]); title('Original information bearing signal and the error signal'); legend('Original Signal' ,'Error Signal') (Fig. 6).

Input signal: This is the input digitalized signal with deterministic value of parameters or components such as its frequency and amplitude as shown in Fig. 7.

Signal with noise: Figure 8 shows the distorted signal in the medium, which is assumed that it has been subjected to noise and interference and therefore, the components like its frequency and amplitude are not deterministic.

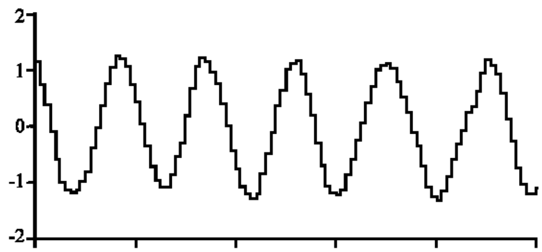

Fig. 9: Filtered output signal

Filtered output signal: This is depicted by Fig. 9 and is the filtered output signal corrected and regenerated to improve the quality of distorted signals having a defined frequency and amplitude at the receiving end.

CONCLUSION

The transmission of an audio signal over a long distance through a communication channel is useful only when the audio signal gets to the receiving end with minimal quantity of noise and distortion.

The study was aimed out to show how the quality of an audio signal transmitted over a noisy communication channel can be improved using an adaptive filter, different models were used and comparison was made.

Therefore, it is concluded that the use of an adaptive filter in improving the quality of an audio signal is encouraging and however, necessary so as to make audio signals transmitted more intelligible to the end users.

REFERENCES

John Wiley, S., 1979. Programs for Digital Signal Processing. 5th Edn. N.Y. IEEE. Press, pp: 56-62.

Jackson, L.B., 1992. Digital Filters and Signal Processing. 3rd Edn. Boston, Kluwer Academic Publishers, pp: 34-56.

Oppenheim, A.V. and R.W. Schafer, 1989. Discrete-Time Signal Processing. Englewood Cliffs, NJ: Prentice Hall, pp: 98-109.

Parks, T.W. and C.S. Burrus, 1987. Digital Filter Design. New York, John Wiley and Sons, pp: 34-56.

Proakis, J.G. and D.G. Manolakis, 1996. Digital Signal Processing: Principles, Algorithms and Applications, Upper Saddle River, NJ: Prentice Hall, pp: 67-80.