

IMPROVEMENTS IN ACTIVE NOISE CONTROL OF HELICOPTER NOISE IN A MOCK CABIN

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ABSTRACT

The application of active noise control (ANC) to interior cabin noise of helicopters is a challenging problem because of multiple tones and significant broadband frequency content. The most common control approach is to use the standard filtered-x algorithm. For this algorithm, the convergence and tracking speed is dependent on the eigenvalues of the filtered-x autocorrelation matrix, with these eigenvalues being frequency dependent. To maintain stability, the system must be implemented based on the slowest converging frequency that will be encountered, which can lead to significant degradation in the overall performance of the control system. This paper will discuss an approach that has been developed which largely overcomes this frequency dependent performance, in a manner that maintains a relatively simple control implementation but significantly improves the overall performance of the control system. The favorable convergence characteristics are demonstrated through the application of helicopter noise in a mock helicopter cabin.

INTRODUCTION

Helicopter cockpit noise has become an increasingly important problem for designers to address. Research at NASA Langley has indicated that the dominant frequencies of helicopter noise fall in the range where the human ear is most sensitive and that the issue of cockpit noise is a significant concern and needs to be addressed.

Traditionally, noise reduction has occurred through improved design of rotor systems or through passive attenuation devices installed in the cockpit itself. Generally, passive noise cancellation is limited to attenuating high frequency noise where the wavelengths of the sound are relatively short. In the case of low frequency noise, active noise control (ANC) has shown to be an effective approach. One of the most common control algorithms for implementing ANC is the filtered-x LMS algorithm (FXLMS)¹⁻⁴.

One of the limitations of the FXLMS algorithm is its slow frequency dependent convergence properties. An

improvement to the algorithm has been made which largely overcomes this limitation. This development is explained by first giving some necessary background information. Second, a new approach to implementing the FXLMS algorithm is developed. Lastly, the application to helicopter noise is discussed through a brief characterization of helicopter noise followed by a presentation of results using the new control approach.

BACKGROUND

As a basis for understanding the new control approach, a brief discussion of the FXLMS algorithm and its convergence properties are given.

FXLMS Algorithm

The goal of the FXLMS algorithm is to minimize the mean-squared error by adaptively updating $W(z)$, a vector containing control coefficients of an FIR filter. A basic block diagram of the FXLMS algorithm is shown in Figure 1. In Figure 1,

signals are represented in both the time domain and in the frequency domain*. Boucher, Elliot, and Nelson⁵ provide a good reference for deriving the control filter update equation for $W(z)$.

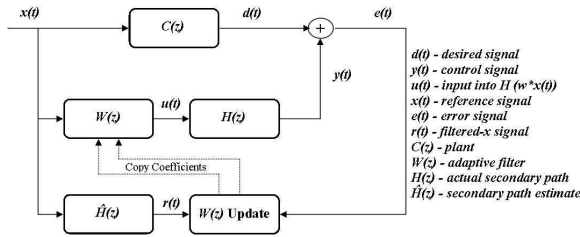


Figure 1. Block Diagram of the FXLMS Algorithm

To adaptively update the control coefficients of $W(z)$, the method of steepest descent is used on the gradient of a quadratic cost function, $\xi(t)$, defined as the mean-squared error. For each iteration of the algorithm, $W(z)$ takes a step of size μ , the convergence coefficient, times the gradient in search of a single global minimum that represents the smallest attainable mean-squared error. The control filter update equation for $W(z)$ can be expressed in the time domain as

$$w(t+1) = w(t) - \mu e(t)r(t) \quad (2.1)$$

where again μ is the convergence coefficient.

One difficulty in implementing the FXLMS algorithm is that the secondary path, represented as $H(z)$ in Figure 1, is unknown. This secondary path is an impulse response that includes the effects of digital-to-analog converters, reconstruction filters, audio

* A lower case letter with a t in parenthesis represents the signal in the time domain, for example $d(t)$, and capital letter with a z in parenthesis represents the signal in the frequency domain, for example $D(z)$. Both are used in the derivation of the control filter update equation.

power amplifiers, loudspeakers, the acoustical transmission path, error sensors, signal conditioning, anti-alias filters, and analog-to-digital converters. In practice it is impossible to get $H(z)$ so an estimate, $\hat{H}(z)$, of the secondary path must be used. This estimate is usually obtained through a process called system identification. Once obtained, the secondary path estimate is used to create $r(t)$, the filtered-x signal, that is used in updating the control coefficients of $W(z)$. The reference signal, $x(t)$, is then filtered with these control coefficients to produce the desired control signal.

Convergence Characteristics

The use of an estimate of the secondary path transfer function, $\hat{H}(z)$, effects the stability and convergence rate¹ of LMS based algorithms. Lower convergence rates and instability are directly related to errors in the estimation of the secondary path transfer function. Estimation errors can be considered in two parts: errors in the magnitude estimation and errors in the phase estimation³. Magnitude estimation errors will alter the maximum stable value of the convergence coefficient through an inverse relationship⁴. It has been shown that phase estimation errors greater than +/- 90 degrees cause algorithm instability⁵. Magnitude estimation errors tend to be less important than phase errors, as magnitude errors can be compensated for in the value of the convergence coefficient or through the adaptive filters.

The convergence coefficient, μ , must be selected for each application.

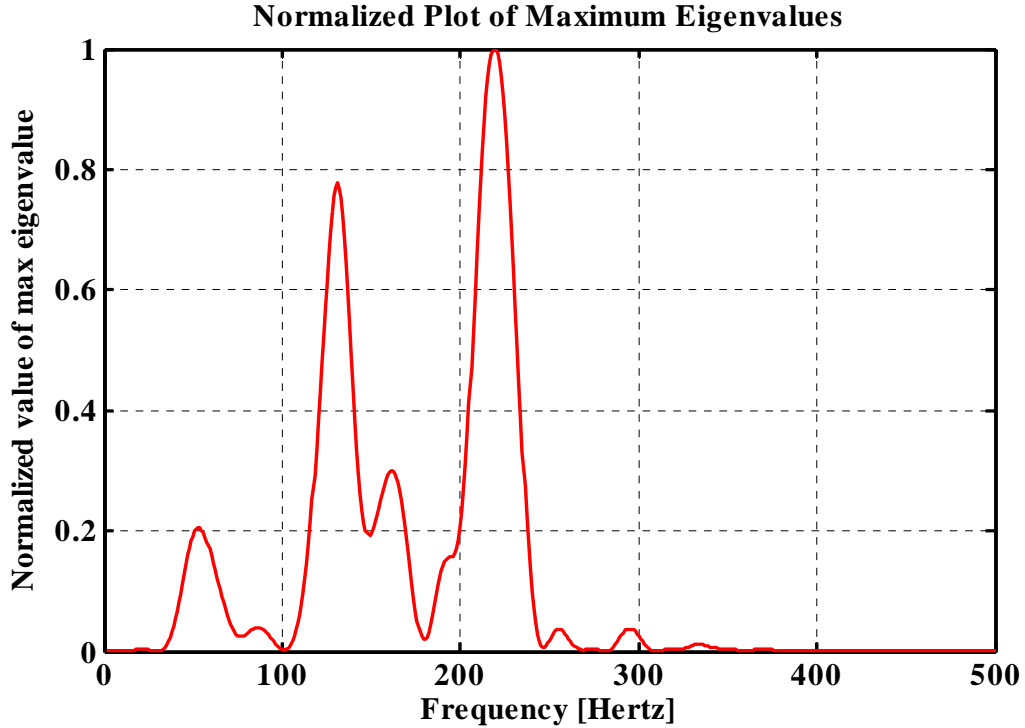


Figure 2. Sample Plot of Eigenvalues for an ANC Application

Several factors affect the choice of μ including: number of control sources and sensors, time delay in the secondary path, digital filter length, system amplifier gains, the type of noise signal to be controlled (random or tonal), and the estimate of the secondary path transfer function¹. An estimate for the largest value of the convergence coefficient that would still keep the system stable is made by looking at the eigenvalues of the filtered reference signal autocorrelation matrix. The autocorrelation matrix is defined in Equation 2.2 where E is the expectation operator and $R(n)$ is a matrix whose column represents the filtered-x signal.

$$E\{R(n)R^T(n)\} \quad (2.2)$$

The eigenvalues of the autocorrelation matrix dictate the rate of convergence of each filter coefficient. The maximum stable convergence coefficient that can

be used for control is the inverse of the maximum eigenvalue of all of the filter coefficients. Disparity in the eigenvalues forces some filter coefficients to converge rapidly and others to converge more slowly⁶. A plot of the maximum eigenvalues at each frequency for a sample ANC application is shown in Figure 2. The data for the graph were computed by calculating the maximum eigenvalue from the autocorrelation matrix for tonal inputs from 0-500 Hz. To generate the filtered-x signal a secondary path from a mock cab was used.

From Figure 2 it is apparent that the maximum eigenvalue varies at each frequency meaning that the system will convergence more quickly at some frequencies and less quickly at other frequencies. While the fastest convergence rate of the system is found at the frequency with the smallest

eigenvalue, it cannot be used or the system can become unstable¹. The only way to assure system stability is to use the convergence rate found at the frequency with the largest eigenvalue. While in reality it is the slowest possible convergence rate for the system, it is nevertheless called the maximum convergence rate for the system because it is the largest rate that still assures system stability.

Degraded convergence and tracking performance is expected in cases where the target frequency is not steady or where several target frequencies are present. One proposed solution for several target frequency applications improves convergence by implementing in parallel a FXLMS algorithm for each frequency to be controlled^{6,7}. Doing so allows individual convergence parameters to be chosen for each target frequency at the expense of computational complexity. A single convergence coefficient that could be optimized over all frequencies to be controlled could lead to improved control performance in terms of the convergence and tracking capabilities of the algorithm without increased computational complexity.

NEW APPROACH

If the variance in the eigenvalues of the autocorrelation matrix was removed, a single convergence parameter could then be chosen that would converge at the same rate over all frequencies. The autocorrelation matrix is directly dependent on the filtered-x signal, $R(n)$, which is computed by filtering the input signal with the secondary path transfer function. The input signal is usually a reference signal taken directly from the sound field to be

controlled and cannot be changed. Changes to the autocorrelation matrix must stem from changes to the secondary path transfer function. As noted previously, variance in modeling the magnitude of the secondary path transfer function can be compensated for by the adaptive filters, but phase errors in excess of 90 degrees lead to system instabilities. Ideally then, changes would be made to the magnitude of the secondary path model while the phase information is preserved.

A relatively simple modification to the magnitude coefficients has led to improvements in the convergence characteristics of the algorithm. The basic procedure is as follows:

1. Get the time domain impulse response of the secondary path transfer function through an offline system identification process
2. Take the Fast Fourier Transform (FFT) of the impulse response
3. Compute the mean value of the FFT
4. Divide each value in the FFT by itself and then multiple by the mean value obtained in step 3
5. Compute the inverse Fast Fourier Transform and use the new modified impulse in the FXLMS algorithm

This procedure flattens the magnitude coefficients of the secondary path model while preserving the phase information. Figure 3 shows the original and flattened secondary path magnitude coefficients and also shows that the phase information has been preserved. Figure 4 shows the system eigenvalues using the original and modified secondary path model.

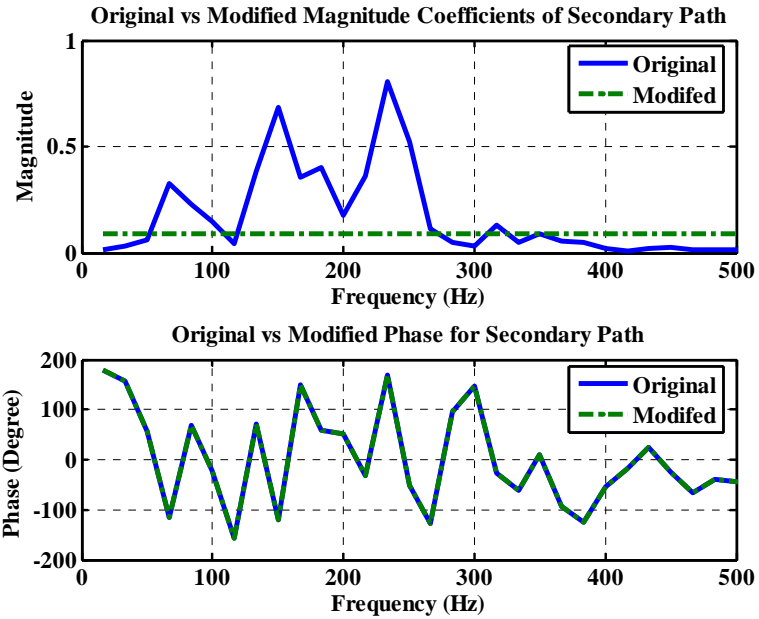


Figure 3. Magnitude and phase of original and modified secondary path model

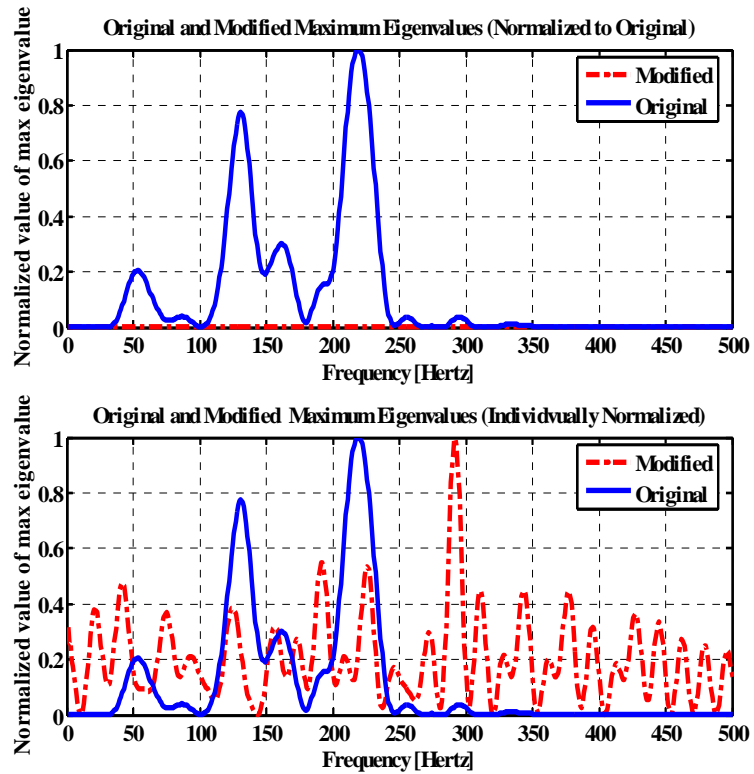


Figure 4. Original and modified maximum eigenvalues

In the top graph of Figure 4 the eigenvalues in each case have been normalized by the largest of the original eigenvalues. In the bottom graph of Figure 4 both the original and modified eigenvalues have been normalized by their individual maximum eigenvalues. It can be seen (top subplot) that compared to the original case the modified eigenvalues are more uniform (“flat”) over all frequencies. It can also be seen (bottom subplot) that the modified eigenvalues are not perfectly flat over frequency. An attempt to quantify the improvement has been made by using the following metrics:

1. *Span* – maximum eigenvalue divided by the minimum eigenvalue. Ideally equal to one.
2. *RMS value* – root mean square. Ideally equal to one.
3. *Crest factor* – maximum eigenvalue divided by the rms value. Gives a sense of how close the rms value is to the peak value. Ideally equal to one.

The comparison of using these metrics can be seen in Table 1.

Table 1. Comparison of modified and original eigenvalues.

Metric	Original	Modified
Span	1506	239
RMS	0.314	0.440
Crest Factor	3.18	2.28

In the Table it can be seen that the modified case has a lower span, higher rms value, and a lower crest factor. In all three metrics, the values for the modified case are closer to the optimum values that would be present if the eigenvalues across all frequencies were exactly the same. While not the optimum, the modification to the

secondary path that gives these improved eigenvalues should make a noticeable improvement in the convergence speed of algorithm at different frequencies.

HELICOPTER NOISE

The control of helicopter noise is a challenging problem in that there are multiple noise sources that contribute to overall sound level observed in the interior of the cabin. In the frequency range where active noise control is most effective (0-500 Hz), three major noise sources have been identified (see Table 2). By using known gearing relationships for the different components, the fundamental frequency for each of these three sources has also been identified.

Table 2. Major sources of helicopter noise and their corresponding fundamental frequencies.

Source	Fundamental Frequency
Main Rotor	13.6 Hz
Tail Rotor	80.9 Hz
Engine	135.8 Hz

As can be seen in Table 2, the fundamental frequencies of these three sources are quite low in frequency; this results in several higher order harmonics below 500 Hz for each source. The significance of this identification is that there are twenty or more tonal components that can be targeted by the active control system.

As previously discussed, multiple tonal noise sources can be a challenge for the control algorithm because it will converge at a different rate for each frequency. The convergence parameter, μ , will be limited by the slowest converging frequency. In order to test the effectiveness of the modified

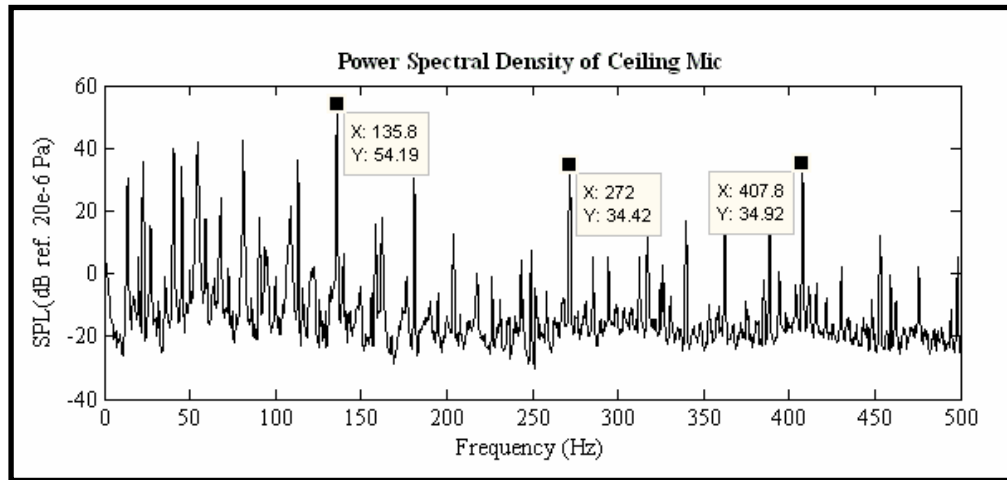


Figure 5. Power spectral density of helicopter noise with engine tones identified.

algorithm in overcoming this limitation for helicopter noise, an ANC simulation was performed.

Recordings of three interior microphones of a Robison R44 helicopter were obtained, in addition to a simultaneous recording of the engine tachometer. The engine tachometer was recorded for use as the reference signal for controlling the tonal components that are created by the engine. Similar reference signals for use in controlling the main and tail rotor tonal components will be gathered for use in later experimentation. For now, the control simulations will be limited to controlling the engine tones. Figure 5 shows the power spectral density from one of these microphone recordings. The three engine tones are identified in the Figure.

RESULTS

Control simulations were made with both the original and the modified FXLMS algorithm. The helicopter recordings were used as the sound source. A secondary path model was obtained by getting an impulse response from a mock helicopter cabin fabricated for use in this and other experiments. The results shown will focus on the convergence speed of the algorithm for

the three engine tones. For both test cases, the maximum stable convergence parameter was used. Figure 6 shows the results when the original algorithm was used. In the Figure, the normalized error signal is plotted as a function of time for the three engine tones. For each subplot, the convergence speed (how long it takes the algorithm to reduce the error signal) can be observed. It is seen that the algorithm converges much quicker at 136 Hz (~0.4sec) and 408 Hz (~0.4sec) than at 272 Hz (~1.2sec). In terms of sound attenuation, less attenuation in practice is expected at 272 Hz.

Figure 7 shows the results when the modified algorithm is used. It is seen that the convergence speed for each of the three frequencies is more uniform. In comparison to the results for the original case, the modified algorithm converges faster at each of the three frequencies; 136 Hz (< 0.4sec), 408 Hz (< 0.4sec), 272 Hz (< 0.4sec). The important implication of these results, is the additional sound attenuation that should result from a faster convergence at each frequency. Though not quantified in this report, the difference in attenuation will be largest in an actual helicopter where the target frequencies change with time as the engine speed changes for different flight conditions.

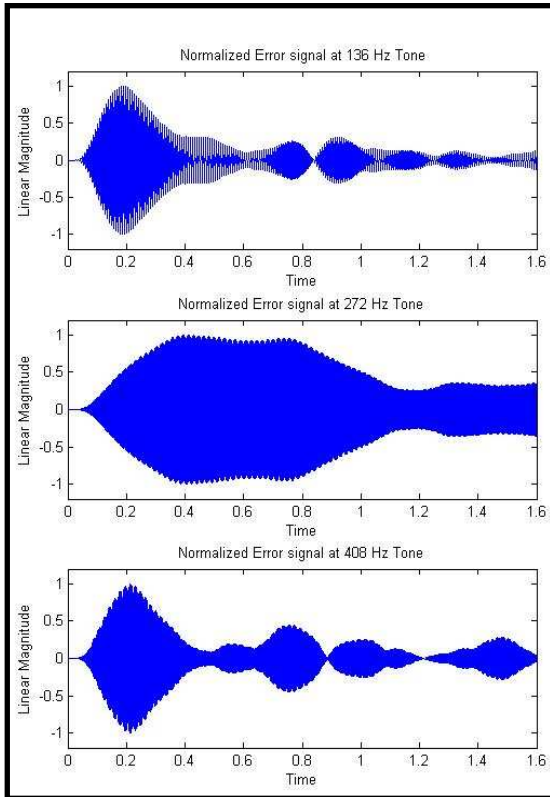


Figure 6. Convergence speed of engine tones using original algorithm

CONCLUSIONS

A new modified FXLMS algorithm has been developed which offers improved convergence characteristics compared to the original FXLMS algorithm. This modified algorithm has been applied to controlling the engine tones of helicopter noise and has shown to converge as much as three times faster than the original algorithm at the frequencies tested.

The greatest advantage of the new algorithm is its simplicity. It can be implemented in only a few lines of code and adds essentially no computational complexity to the algorithm.

Further development and experimentation of the new modified algorithm is planned to fully explore its potential in increasing the performance of the FXLMS algorithm for ANC.

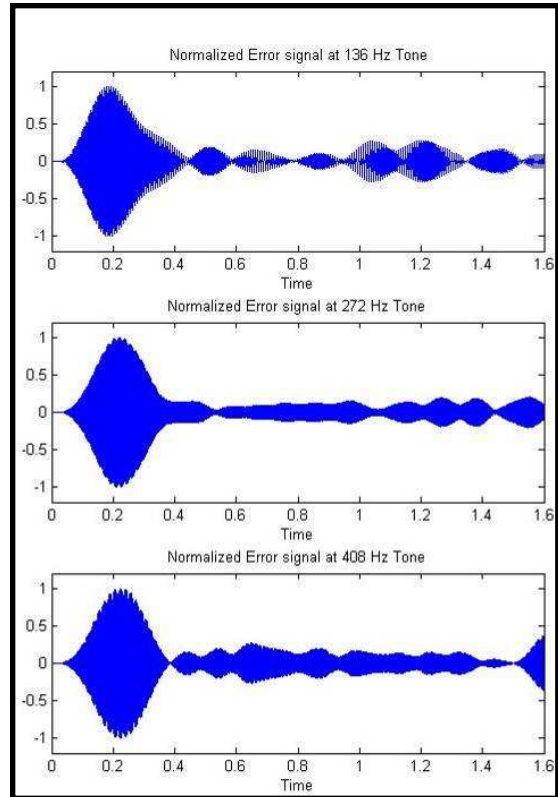


Figure 7. Convergence speed of engine tones using modified algorithm

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