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Improved Algorithm of Equation Error Model of Active Noise Control

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***Abstract*—This work attempts an effective alternative algorithm structure for the control of spurious audio signals using the equation error model of active noise control system. Test samples of noise are generated from the greater of a grinding mill and used in validation of the proposed alternative algorithm. The algorithm gained approximately 27dBA noise reduction at about 270Hz frequency which is evident of an improvement in noise control when compared with other conventional active noise control algorithms.**

Keywords— Noise; Algorithm; Equation error; Infinite-impulse response; Filter

I. INTRODUCTION

Acoustic noise in residential and workplaces is a major health concern to the society, it is mainly responsible for hearing damages which impair the efficiency of workers. The control of acoustic noise could be either by passive means or active means but the passive means is less effective. The active means of noise control commonly known as Active Noise Control (ANC) System is mostly deployed at lower frequencies as it guarantees effectiveness and is simple to implement. The ANC involves the production of a signal similar to the noise signal in amplitude and frequency but opposite in phase angle to act as a cancelling signal. The ANC technique operates in two distinctive manners which are the feedforward technique and the feedback technique. The feedforward technique uses both the noise signal intended to eliminate and the counter signal which enables effective cancellation of narrowband and wideband disturbances, while the feedback technique only uses counter signal at the intersecting point. Meanwhile the combination of both feedforward and feedback techniques gives rise to the Hybrid ANC system. The most used of these techniques is the feedforward adaptive filter based ANC. Here the filters adjust their coefficients by varying statistics of the target filter signal. This is usually achieved in two distinct structures which are the Finite Impulse Response Filters and Infinite Impulse Response Filters. Using the poles and zeros the required filter can be modeled with few parameters in the adaptive infinite impulse response filter structure which reduces computational complexity when compared with the Finite Impulse Response Structure. When working with the Infinite Impulse Response, there are two models which are the Output Error Model (OEM) and the Equation Error Model (EEM). The EEM utilizes only zeros and neglects poles there by achieving stability during operation when appropriate step size is applied; while the OEM utilizes both poles and zeros there by making it difficult to achieve stability. The EEM also makes it possible to achieve the global minimum because its mean square error function is a quadratic function. In previous works according to [2] the EEM type of Infinite Impulse Response ANC systems were developed and the conditions of convergence addressed to achieve stability and possible optimal global minimum of the Mean Square Error (MSE). In this work, improved algorithms are applied in the EEM Infinite Impulse Response ANC system in a view to achieving a reduced transformer noise, system convergence and maintainability. Here noise elimination near a grinding mill in a processing plant is taken as a sample to apply this ANC system model.

II. OUTPUT ERROR MODEL OF INFINITE IMPULSE RESPONSE STRUCTURE OF ANC SYSTEM

In acoustic systems, there are always a special kind of loop gain occurring when a sound loop exists between an audio input and audio output known as acoustic feedback. In resolving the problem associated with acoustic feedback, an external feedback structure is introduced to the output channel using FURLMS system of algorithm. Figure 1. Is a block diagram showing the Output Error Model of Infinite Impulse Response Structure of an ANC System employing the FURLMS system of algorithm.

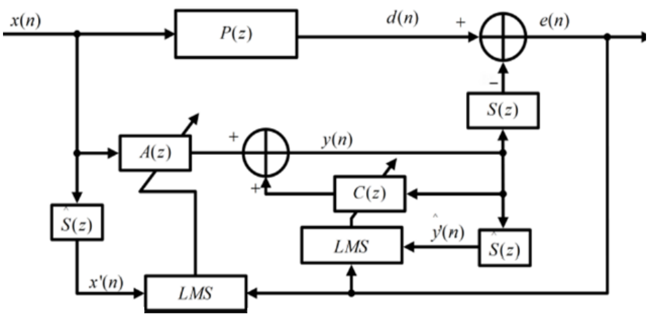

Figure 1: Output error ANC of FURLMS algorithm

Where $P(z)$ and $S(z)$ are the primary and secondary channels respectively

The error signal is expressed as:

$$e(n) = d(n) - s(n) * y(n) \quad (1)$$

Where: $s(n)$is the impulse response of $S(z)$
$d(n)$ is the primary noise signal
* is the convolution operator

$y(n)$ is expressed in an expanded form as

$$y(n) = \sum_{i=0}^{N_a-1} a_i(n)x(n-i) + \sum_{j=0}^{N_c-1} c_j(n)y(n-j) \quad (2)$$

Where: $a_i(n)(i = 0,1, \ldots\ldots N_{a-1})$
$c_j(n)(j = 0,1, \ldots \ldots N_{c-1})$
N_a and N_c are the lengths of the filter

In accordance with the Output Error Model algorithm, the process of updating the weight can be expressed as:

$$a(n+1) = a(n) + \mu x'(n)e(n) \quad (3)$$

$$c(n+1) = c(n) + \mu \widehat{y'}(n-1)e(n) \quad (4)$$

Here $\widehat{y'}(n) = \hat{s}(n) * y(n-1) x'(n) = \hat{s}(n) * x(n), a(n) = [a_o(n)a_1(n) \ldots\ldots a_{N_{a-1}}]^T$
$x(n) = [x(n)x(n-1) \ldots\ldots x(n-N_a+1]^T$
$c(n) = \left[c_1(n)c_2(n) \ldots\ldots c_{N_c}(n)\right]$
$\hat{s}(n)$ = Secondary path impulse response $\hat{S}(z)$
μ = Step size

In ideal condition, the error signal $e(n)$ is minimized by adjusting $A(z)$ and $C(z)$ to satisfy the equation

$$\frac{A(z)}{1-C(z)} = \frac{P(z)}{S(z)}$$

III. EQUATION ERROR MODEL OF INFINITE IMPULSE RESPONSE STRUCTURE OF ANC SYSTEM

In the equation error model of infinite impulse response structure of ANC system, the most prevailing problem is that of pole basically caused by the secondary path impulse response $S(z)$. This problem is eliminated by synthesizing the estimated desired signal $d(n)$ through digital processing. Figure 2 shows the impulse response $s(n)$ of the secondary path.

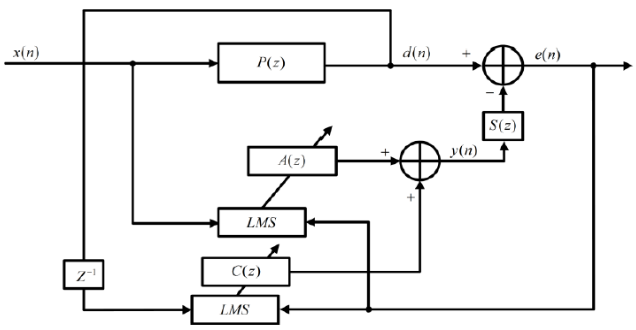

Figure 2: Equation error adaptive IIR-Filter

$$s(n) = [s_0 s_1 \ldots\ldots s_{L-1}] \quad 6$$

The output signal $y(n)$ is expressed as:

$$y(n) = a^T(n)x(n) + c^T(n)d(n-1) \quad 7$$

Here $d(n-1)$ is expressed as

$$d(n-1) = [d(n-1)d(n-2) \ldots\ldots d(n-N_c)]^T \quad 8$$

For the equation error model of infinite impulse response structure of an ANC system, the adaptations $a(n+1)$ and $c(n+1)$ are derived as follows:

$$a(n+1) = a(n) + \mu e(n)[\hat{s}(n) * x(n)] \quad 9$$

$$c(n+1) = c(n) + \mu e(n)\widehat{d'}(n) \quad 10$$

In practice, $\hat{d}(n)$ is synthesized as

$$\widehat{d'}(n) = e(n) + y(n) * \hat{s}(n) \quad 11$$

By this synthesis, the output signal equation is rewritten as:

$$y(n) = a^T(n)x(n) + c^T(n)\hat{d}(n) \quad 12$$

The challenges encountered in the equation error model of infinite impulse response structure of the ANC system are:

a. Slow convergence speed
b. Instability
c. Poor noise reduction

However the challenge of instability is prevented by using non-recursive terms. Figure 3

IV. ALGORITHM OF EQUATION ERROR INFINITE IMPULSE RESPONSE FILTER OF AN ANC SYSTEM

Of the three known challenges encountered in the equation error infinite response filter, on one (instability) can be resolved via physical means. For the remaining two (low convergence speed and poor noise reduction), they can only be minimized by improving the equation algorithm. Here a proposed algorithm is presented where three adaptive filters [$A(z), B(z)$ and $C(z)$] are used instead of the usual two filters [$A(z)$ and $C(z)$]. Figure 3 shows the proposed positioning of the three adaptive filters in an equation error model of an ANC system.

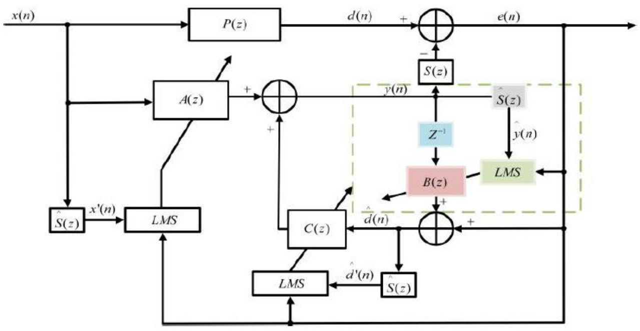

Figure 3: Improved adaptive IIR-Filter

Here the squared error signal is given as:

$$\varepsilon'_{EE} = e^2(n) \quad 13$$

While the gradient of error surface is given as:

$$\nabla\varepsilon'_{EE} = -2e(n)[s(n) * u(n)] \quad 14$$

Here $s(n)$ and $u(n)$ are the secondary path impulse response and adaptive filter input respectively. When the estimated secondary path $S(z)$ is factored in, the equation for the gradient of error surface will become

$$\nabla \varepsilon'_{EE} = -2e(n)[\hat{s}(n) * u(n)] \quad 15$$

For this proposed improved equation error infinite impulse response filter algorithm, the adaptations are derived as:

$$a(n+1) = a(n) + \mu e(n)[\hat{s}(n) * x(n)] \quad 16$$

$$b(n+1) = b(n) + \mu e(n)[y(n) * \hat{s}(n)] \quad 17$$

$$c(n+1) = c(n) + \mu e(n)\hat{d}'(n) \quad 18$$

With these, the output signal $y(n)$ is rewritten as:

$$y(n) = a^T(n)x(n) + c^T(n)[e(n) + b^T(n)\hat{y}(n)] \quad 19$$

$$\frac{A(z)}{1-C(z)B(z)} = P(z) \quad 20$$

From equation 19, the accuracy of the synthesized signal is improved and also instability is avoided since the system is non-recursive.

V. CONSTRAINT OF STEP SIZE

In resolving the issues associated with step size, we take a white noise signal having 0 mean value and 1 as its variance to represent the reference signal $x(n)$. Here the limits of the step size $\mu A\ of\ A(z)$, $\mu B\ of\ B(z)$ and $\mu C\ of\ C(z)$ are set in line with equation 20 as follows:

$$0 < \mu A < \frac{3}{(N_a+3\nabla_{eq})P_{x'}} \quad 21$$

$$0 < \mu B < \frac{3}{(N_b+3\nabla_{eq})P_{\hat{y}}} \quad 22$$

$$0 < \mu C < \frac{3}{(N_c+3\nabla_{eq})P_{\hat{d}'}} \quad 23$$

Where $P_{x'}$ denotes the power of $x'(n)$

$P_{\hat{y}}$ denotes the power of $\hat{y}(n)$

$P_{\hat{d}'}$ denotes the power of $\hat{d}'(n)$

∇_{eq} is the equivalent delay of the secondary path in equation 20

$$\nabla_{eq} = \frac{\sum_{l=0}^{L-1} l s_l^2}{\sum_{l=0}^{L-1} s_l^2} \quad 24$$

The signal powers $P_{x'}$, $P_{\hat{y}}$ and $P_{\hat{d}'}$ in equations 21, 22 and 23 are set to unity (1) while the maximum total step size μ is set as the minimum of μA, μB and μC. Thus only the secondary path delay ∇_{eq} affects the filter length and total step size.

VI. SOLUTIONS OF THE GLOBAL MINIMUM

To obtain the Global Minimum based on the improved algorithm of Equation Error model of the ANC, the mean square function is expressed as:

$$\xi'_{EE}(n) = E[d^2(n)] + a^T(n)R_{x'x'}a(n) + b^T(n)R_{\hat{y}\hat{y}}b(n) + c^T(n)R_{\hat{d}'\hat{d}'}c(n) + 2[a^T(n)R_{x'\hat{d}'}c(n) + a^T(n)R_{x'\hat{y}}b(n) + b^T(n)R_{\hat{y}\hat{d}'}c(n) - a^T(n)P_{dx'} - b^T(n)P_{d\hat{y}} - c^T(n)P_{d\hat{d}'}] \quad 25$$

Here $R_{x'x'} = E[x'(n)x'^T(n)]$

$R_{\hat{y}\hat{y}} = E[\hat{y}(n)\hat{y}(n)]$

$R_{\hat{d}'\hat{d}'} = E[x'(n)x'^T(n)]$

$R_{x'\hat{d}'} = E[x'(n)\hat{d}'T(n)]$

$P_{dx'} = E[d(n)x'(n)]$

$P_{d\hat{d}'} = E[d(n)\hat{d}'(n)]$

From equation 25, it is seen that there is a global minimum for improved algorithm of Equation Error model of an ANC system by calculating the function gradient of equation 25, thus we have

$$\frac{\partial \xi'_{EE}}{\partial a(n)} = [\frac{\partial \xi'_{EE}(n)}{\partial a_0(n)} \frac{\partial \xi'_{EE}(n)}{\partial a_1(n)} \ldots \frac{\partial \xi'_{EE}(n)}{\partial a_{N_a-1}(n)}]^T = 2R_{x'x'}a(n) + 2R_{x'\hat{d}'}c(n) + R_{x'\hat{y}}b(n) - 2p_{dx'} \quad 26$$

$$\frac{\partial \xi'_{EE}(n)}{\partial b(n)} = [\frac{\partial \xi'_{EE}(n)}{\partial b_0(n)} \frac{\partial \xi'_{EE}(n)}{\partial b_1(n)} \ldots \frac{\partial \xi'_{EE}(n)}{\partial b_{N_a-1}(n)}]^T = 2R_{\hat{y}\hat{y}}b(n) + 2R_{x'\hat{y}}a(n) + 2R_{\hat{y}\hat{d}'}c(n) - 2p_{d\hat{y}} \quad 27$$

$$\frac{\partial \xi'_{EE}(n)}{\partial a(n)} = [\frac{\partial \xi'_{EE}(n)}{\partial c_0(n)} \frac{\partial \xi'_{EE}(n)}{\partial c_1(n)} \ldots \frac{\partial \xi'_{EE}(n)}{\partial c_{N_a-1}(n)}]^T = 2R_{\hat{d}'\hat{d}'}c(n) + 2R_{x'\hat{d}'}a^T(n) + 2b^T(n)R_{\hat{y}\hat{d}'} - 2p_{d\hat{d}'} \quad 28$$

From equations 26, 27 and 28 the prime weight vectors $a^0{}_{EE}(n)$, $b^0{}_{EE}(n)\ and\ c^0{}_{EE}(n)$ is derived by assuming that the gradient functions are numerically equal to zero.

VII. EXPERIMENTATION

The noise generated by the greater of a grinding mill which is characterized by a strong periodicity and high input level is taken to be $x(n)$. The audio signals are collected by the use of two sensors and transmitted to the MATLAB software on a computer system via a double channel audio signal analyzer. The Matlab software function is to process the algorithm verification. The primary path is measured to be 1.5m which is the distance of separation between the first sensor and the computer system; while the secondary path is measured to be 7.1m which is the distance of separation between the second sensor and the computer system. From the computer measurement system, the time-frequency graph of the noise signal is acquired as shown in figure 4. From the graph, it is seen that the sound pressure level is higher at the low frequency range than at high frequency range.
The secondary path of the system is set at a delay of 24 samples with 64 filter length to create similar conditions used in [3] so as to effectively compare the improved algorithm with the existing algorithms.

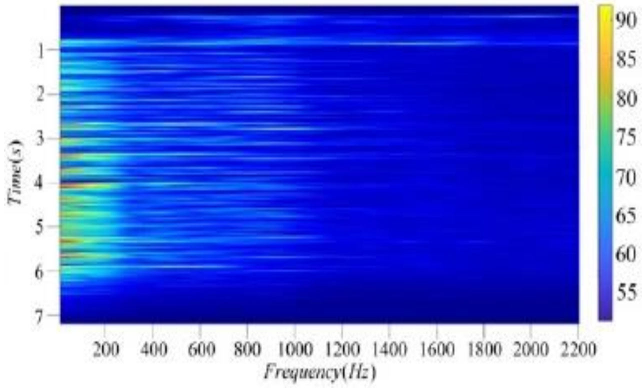

Figure 4: Time-frequency graph

VIII. RESULTS

Using equations 16,17 and 18, the learning process of the weight coefficients are calculated as seen in figures 5,6 and 7. The adaptations are also calculated to be

$a_0^0(n) = 0.69$

$b_0^0(n) = 0.58$

$c_0^0(n) = 1.20$

On the other hand, according to equations 21,22 and 23, the limits of the step size are calculated to be

$\mu_A < 0.14$

$\mu_B < 0.35$

$\mu_C < 0.22$

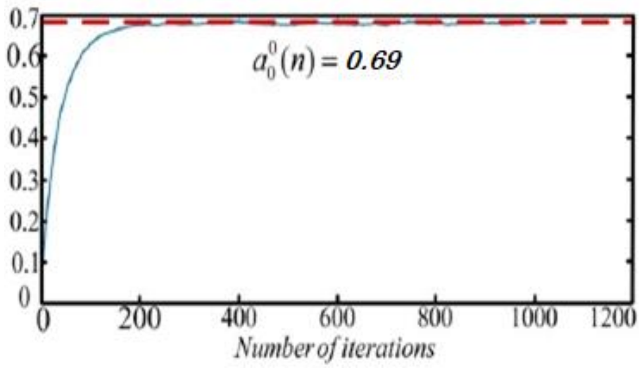

Figure 5: Convergence of weight coefficient $a_0^0(n)$

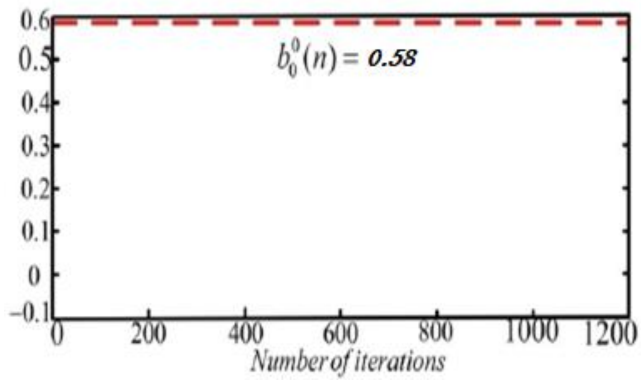

Figure 6: Convergence of weight coefficient $b_0^0(n)$

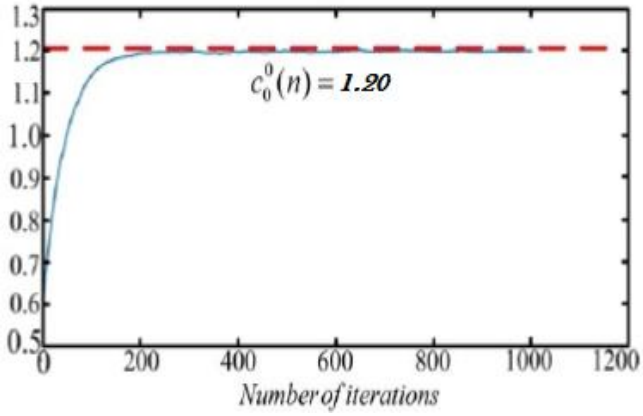

Figure 7: Convergence of weight coefficient$c_0^0(n)$

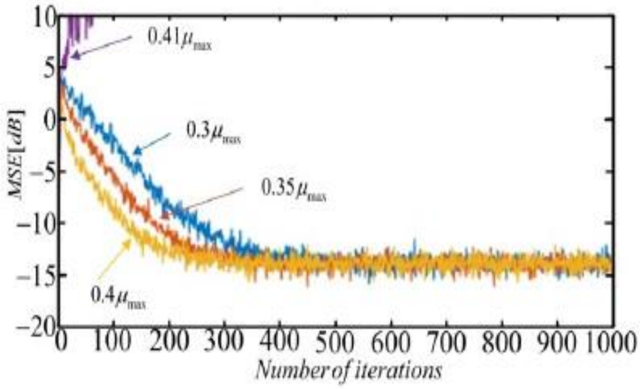

Figure 8: Improved Algorithm of ANC in different step factors

Using different step sizes, the learning curve is generated from the equation

$$\partial(n) = \gamma\partial(n-1) + (1-\gamma)e^2(n) \quad 29$$

Here, $\gamma = 0.96$ is constant known as the forgetting factor. The improved algorithm of Equation Error model of the ANC system is found to converge at step sizes$< 0.4\mu_{max}$.

The measurement noise is compared with the performance of other existing algorithms and the improved algorithm. Figure 9 below shows the average sound pressure level of the noise in the different algorithms with step sizes 0.017, 0.014, 0.0015 and 0.0006 respectively. Although all the existing algorithms compared in this work is found to effectively converge and achieve a level of noise reduction, the improved algorithm of Equation Error model of the ANC shows a better convergence speed for the noise of a grating mill as seen from figure 10. A close look at figure 9 shows that the improved algorithm of Equation Error model of the ANC can reduce the noise of a grating mill up to 27dBA at frequencies below 2.7 KHz

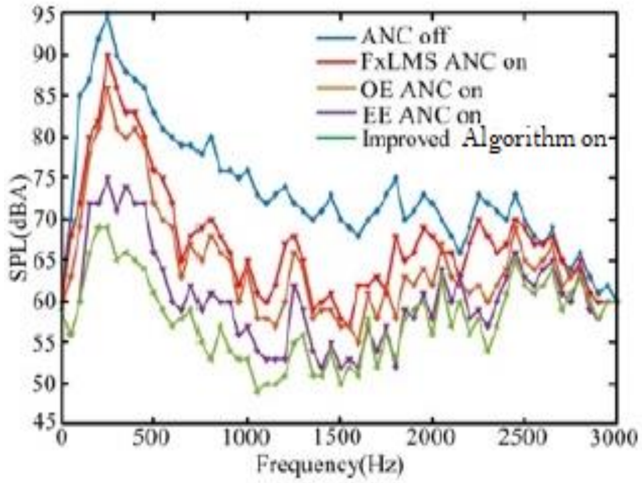

Figure 9: Average sound pressure level of noise

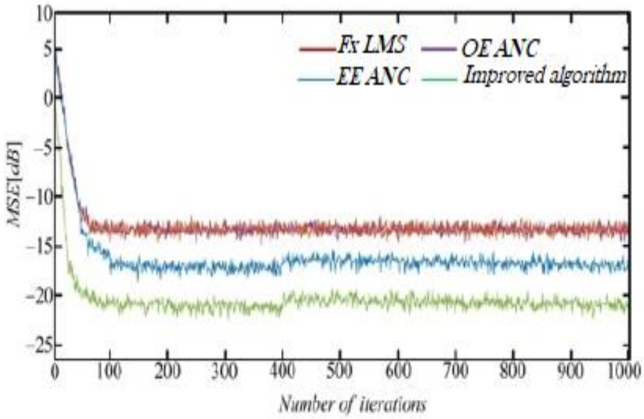

Figure 10: Learning curve

IX. CONCLUSION

This study presents an adaptive IIR filter based on an enhanced EE model that uses an offline secondary path modeling method to improve convergence speed and noise reduction. In order to replicate the system's performance, the greater noise is used as input noise data. It has a good noise reduction impact between 0 and 2.7 kHz, according to the data. However, the system's computing complexity will increase. The model suggested in this study is ideal for processing low-frequency noise signals, which are common in industrial equipment such as engines, graters, and compressors that generate strong noise signals.

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