On The Achievable Amplification of the Low Order NLMS Based Adaptive Feedback Canceller for Public Address System

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Abstract: - In an environment wherein public address system are used to address the general public, acoustic feedback occurs unpredictably which significantly degrades the acoustic quality of the information signal. A low order adaptive filter is then presented, with the objective that is to reduce the effect of acoustic feedback of the public address system that might occur at any point within the area. The system covers a multi-tapped finite impulse response (FIR) low order adaptive filter that is implemented in field programmable gate array (FPGA). The effects of the adaptation constant adaptation constant and filter length was investigated. During the evaluation, the acoustic activity of the environment was observed by a spectrum analyzer and spectrograph. Results show that with large adaptation constant, the adaptive filter was able to quickly reduced before the system exhibit an unstable behavior. However, results also show that the magnitude of the output information signal of the public address system with large adaptation constant is lesser compared to a smaller adaptation constant of the adaptive filter. This shows that the adaptation constant greatly affects how the adaptive feedback canceller eliminate the acoustic feedback but also reduces the amplification of the public address system.

Key-Words: - Acoustic Feedback, Howling, Adaptive Filter, Adaptive Algorithm, Digital Filter

1 Introduction

In communication, the sound quality is of major importance for being able to properly perceive and interpret the correct sound signal, [1]. One way to effectively transmit a sound signal over a distance is to use public address system. Public address system has been in used from the past in orchestras, theatres, cinemas, conference room and much more. It has greatly improved since its first used and keeps on improving from time to time. However, its efficiency has been severely affected because of some problems. One of the problems that plagued public address system is howling, [2-4]. Howling is the undesirable sound that occurs when the loudspeaker signal is being picked up by the microphone and amplified by the power amplifier. In this way, a special kind of positive feedback having a closed signal loop is created, [2]. The resulting effect affects the systems performance, deteriorates the sound quality and limiting the achievable amplification, [2-5]. In literature, howling occurs when a closed loop system satisfies the two conditions of the Nyquist criterion for instability. The Nyquist criterion for instability stated that a system is unstable 1) if the product of electroacoustic forward path and the acoustic feedback path is greater than one and 2) the overall phase angle of the electroacoustic forward path and the acoustic feedback path is an integer multiple of 2π.

For the past five decades, a lot of solutions have been proposed with the objective that is to automatically eliminate or reduce the acoustic feedback. The process solving the acoustic feedback problem by completely eliminating or reducing is referred as acoustic feedback control, [2]. However, technicians still preferred to eliminate or reduce acoustic feedback in a manual manner. The main reason why acoustic technicians prefer manual acoustic feedback control is reliability issues, [2][3]. Acoustic feedback control techniques are categorized into four, 1) Phase Modulation (PM), 2) gain reduction, 3) spatial filtering and 4) room modelling, [3]. In general, the objective of acoustic feedback control is to prevent the system that it will satisfy the two condition of the Nyquist criterion for instability, [6-9]. In phase modulation method, the acoustic feedback controller manipulates the overall phase of the close loop so that it will not satisfy the phase condition of the Nyquist criterion for instability, [10-14]. While gain reduction method, simply reduces the gain of the acoustic forward path in order to prevent the closed loop to satisfy the magnitude condition of the Nyquist criterion for instability, [2][15]. On the other hand, the spatial filter uses a microphone array to manipulate the beam of the microphone array and focused it to the talker, while the null is directed towards the loudspeaker, [2][15]. The objective in spatial filtering is to avoid the existence of acoustic feedback path, hence, it is guaranteed that closed loop will not satisfy the two conditions of the Nyquist criterion for instability, [2][3]. Lastly, room modelling technique models acoustic environment or the acoustic feedback path and the resulting model is subtracted to the signal from the microphone,[16-18]. The result, is a feedback compensated electroacoustic forward path. An adaptive filter with an adaptive algorithm is being used to model the acoustic feedback path. The adaptive algorithm manipulates the coefficients of the adaptive filter to obtain an equivalent model of the acoustic feedback path.

Among the acoustic feedback control techniques, the room modelling method based on adaptive feedback canceller are widely used, [2][19]. Its popularity can be attributed to a large achievable maximum stable gain (MSG), sound quality, and complexity. Room modeling objective is to acquire an equivalent model of the acoustic feedback path. Some approach directly model's the acoustic feedback path, while others obtain the inverse model of the acoustic feedback path. The inverse model can optimally equalize the microphone signal when inserted in the electroacoustic feedback path. This approach is also referred to as adaptive inverse filtering, [2]. Adaptive inverse filtering approach has gained little attention only in the context of AFC. Some works on adaptive inverse filtering for AFC can be found in some published works, [20-23].

2 The Acoustic Feedback

A typical block diagram of a simple public address system is shown in Fig 1., [24][25]. It is composed of a microphone, an amplifier or with an audio processing circuit and a loudspeaker. The microphone picks the acoustic signal v(z) and then convert it to electrical signal d(z). The electroacoustic signal d(z) is then amplified and enhanced by an audio processing circuit. The amplified and enhanced electroacoustic signal is then converted back to acoustic signal through a loudspeaker. Usually, the microphone and the loudspeaker are positioned in the such a way that the loudspeaker sound does not hit the microphone. However, in some cases such as an area constrained environment, part of the acoustic signal from the loudspeaker is being feedbacked to the microphone through the acoustic feedback path. The acoustic feedback path is referred to the environment wherein the acoustic signals from the loudspeaker returns to the microphone. In this case, the acoustic signal from the loudspeaker may unavoidably be reflected by the boundaries of walls, floors or ceilings. Acoustic feedback path couples the loudspeaker and the microphone and as a result, a closed loop system was established. The closed loop system is shown in Fig. 1 greatly affect the performance of the public address system, [2]. Among the different artifacts that are produced by the acoustic coupling between the loudspeaker and the microphone, howling effect is the most characteristic one. The transfer function of the public address system, in consideration to the existence of the acoustic feedback path, is shown in (1).

$$\frac{d(z)}{v(z)} = \frac{G(z)}{1 - G(z)F(z)} \tag{1}$$

In equation (1), the G(z) and F(z) are the mathematical models of the acoustic forward path and acoustic feedback path respectively, [2]. The acoustic feedback path model F(z) is linear, time varying and of finite order. It is assumed linear since the effects of sound propagation and reflections in the acoustic environment are quasi-level and independent. Meanwhile, the product of G(z) and F(z) in the denominator in (1) is often referred to as the loop response of the system. The loop response of a closed loop system plays an important role in the overall performance of the public address system. It is known that a closed loop system may exhibit instability, which may lead to oscillation, that in an acoustic system is perceived as howling. In control system, a closed loop system will exhibit instability if the two conditions of Nyquist criterion for instability are satisfied. The Nyquist criterions for instability are shown below for magnitude and phase condition respectively.

$$|G(z)F(z)| \geq 1 \tag{2}$$

$$\angle G(z)F(z) = n(2\pi) \qquad n \in \mathbb{Z} \tag{3}$$

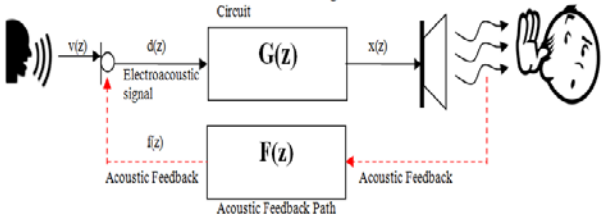


Fig. 1. Block Diagram of a simple public address system

Equation (2) explains that for any acoustic signal of any quantity will gradually increase from time to time as it is being feedbacked because the gain which is magnitude of the product of G(z) and F(z) is greater than one. Furthermore, the gradual increase in acoustic signal as it is being feedbacked can be also attributed to an in-phase relationship of the acoustic input and output signal. The equation in (3) shows that the acoustic input and output aids one another if the acoustic output signal is being feedbacked. Hence, the acoustic signal will gradually increase from time to time, [5]. Magnitude condition in the Nyquist criterion for instability if fulfilled will cause the signal traveling around the loop to further increase and the phase condition in the Nyquist criterion for instability if fulfilled will cause the signal to adds up in phase to the microphone signal.

The magnitude and phase condition in the Nyquist criterion for instability shown in (2) and (3) are very important in the design of an effective acoustic feedback control. Acoustic feedback control method will be designed with the objective that is to prevent either one of the conditions in (2) and Equation (3) from being met.

2 The Adaptive Feedback Canceller

The adaptive filter based on LMS adaptive algorithm will serve as a foundation for designing an adaptive feedback canceller in this study. The adaptive filter shown in Fig. 2 will model the acoustic feedback path. It has a discrete-time Finite Impulse Response (DTFIR) structure that is based on tapped-delay-line and a set of N length of adjustable gain w_0 through w_{N-1}.

The adaptive filter output y(k) is the sum of the delayed and scaled inputs that is described in (4). Equation (4) can also be simplified in a form as describe in (5) where **X**(k) is the input vector and **W** is the weight vector.

$$y(k) = \sum_{i=0}^{N-1} w_i x(k-i) \quad (4)$$

$$y(k) = \mathbf{X}^{\mathbf{T}}\mathbf{W} = \mathbf{X}\mathbf{W}^{\mathbf{T}} \quad (5)$$

The error signal e(k) is derived from the difference of the desired signal d(k) and adaptive filter output signal y(k) as shown in (6).The error signal in (6) can also be written in a form as illustrated in (7) and (8), with the use of (4) and (5).

$$e(k) = d(k) - y(k) \quad (6)$$

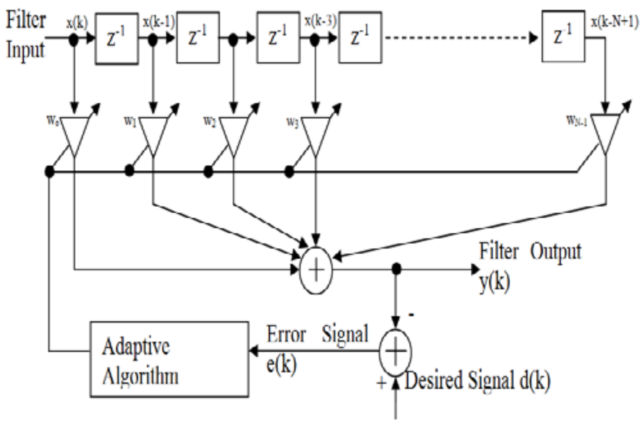


Fig. 2. Adaptive FIR filter structure

$$e(k) = \sum_{i=0}^{N-1} \left[d(k) - w_i x(k-i) \right] \quad (7)$$

$$e(k) = d(k) - \mathbf{X}^{\mathbf{T}}\mathbf{W} \quad (8)$$

The objective of the adaptive algorithm is to find the best impulse response vector **W** which is also called the desired choice of weight vector and it is denoted as $\mathbf{W}^{o}$ in this paper. The best impulse response vector is the choice of the weight vector that makes the summed square difference between d(k) and y(k) as small as possible. The sum of the square difference is the MSE or sometimes called as the performance function which is denoted as ξ.

$$\xi = \sum_{i=0}^{N-1} \left[d(k) - w_i y(k) \right]^2 \quad (9)$$

$$\xi = \sum_{i=0}^{N-1} \left[d(k) - w_i x(k-i) \right]^2 \quad (10)$$

For a given sequence of input vector **X**(k) and scalar d(k), the MSE ξ is said to be a function of a weight vector **W** only. Therefore, the MSE ξ(**W**) is the measure of how well the weight vector **W** execute as a filter's impulse response that produces an output y(k). It also describes the difference between desired signal d(k) and system's output signal y(k).

The choice of **W** that minimizes the MSE ξ(**W**) is the value that produces an adaptive filter's output y(k). To find the $\mathbf{W}^{o}$ that causes the MSE ξ(**W**) to be at a minimum level, first, expand (10) with the use of (5), which defines the adaptive filter output y(k).

$$\xi(\mathbf{W}) = \sum_{i=0}^{N-1} \left[d(k) \right]^2 - 2\mathbf{W} \sum_{i=0}^{N-1} \left[d(k) \mathbf{X}^T(k) \right] + \mathbf{W}^T \mathbf{W} \sum_{i=0}^{N-1} \left[\mathbf{X}(k) \mathbf{X}^T(k) \right] \quad (11)$$

To further simplify (11) the following terms are defined

$$D = \sum_{i=0}^{N-1} \left[d(k) \right]^2 \tag{12}$$

$$\mathbf{P} = \mathbf{W} \sum_{i=0}^{N-1} \left[d(k) \mathbf{X}^T(k) \right] \tag{13}$$

$$\mathbf{R} = \mathbf{W}^T \mathbf{W} \sum_{i=0}^{N-1} \left[\mathbf{X}(k) \mathbf{X}^T(k) \right] \tag{14}$$

From (12) to (14), the term D is a scalar constant because it does not have a vector **W** as its coefficient. The **P** is a cross-correlation of the desired signal d(k) and input signal x(k), which describes the correlation properties of d(k) and x(k) over an N-point choice of shifts and **R** is the autocorrelation of the input signal x(k). The diagonal elements of **R** measure the energy of the signal x(k) and also known as the eigenvalues of the autocorrelation of **R**.

In order to find the minimum Mean Square Error (MSE) we take the gradient of the MSE ξ(**W**) in (11) with respect to the elements of **W** where the weight vector **W** is set at its optimum value $\mathbf{W}^o$. The gradient of the MSE ξ(**W**) is set to zero. The gradient of MSE ξ(**W**) is the same as forming a vector of partial derivatives of the MSE ξ(**W**) with respect to the impulse response coefficients.

$$\nabla_w \xi(\mathbf{W}) = 0 = -2\mathbf{P} + 2\mathbf{RW} \tag{15}$$

$$\mathbf{W}^o = \mathbf{R}^{-1}\mathbf{P} \tag{16}$$

On the other hand, it is very hard to implement the inversion of the autocorrelation of the input signal x(k), that is denoted as $\mathbf{R}^{-1}$ shown in (16), [26]. 1) the inversion of **R** may not be possible and 2) if **R** were theoretically invertible, the numerical precision required to invert **R** properly may be beyond the capability of the hardware or computer used in implementing the adaptive filter. For this reason, in order to find the optimum weight vector $\mathbf{W}^o$, we have to search for the function of the MSE ξ(**W**) from the initial condition to the point where the MSE ξ(**W**) is at its minimum value and this can be done with the use of the steepest descent method. It is also important to note that the MSE ξ(**W**) in (11) is a quadratic function of the weight vector **W.**

Because of complexity, steepest decent was used to approximates the weight vector proportional to the negative of the gradient vector. The goal in here is to decrease the MSE ξ(**W**) and this can be done by moving w(k) towards the optimum weight vector in an iterative process. This can be described as shown in (17).

$$w_i(k+1) = w_i(k) - 0.5\mu\nabla_w \xi(\mathbf{W}) \tag{17}$$

The gradient of ξ(**W**) with respect to the weight vector **W** can be estimated directly from the desired signal d(k) and adaptive filter input signal x(k) by:

$$\nabla_w \xi(\mathbf{W}) = \nabla_w \left[e^2(k) \right] \tag{18}$$

$$\nabla_w \xi(\mathbf{W}) = 2e(k)\nabla_w \left[d(k) - \mathbf{X}^\mathbf{T}\mathbf{W} \right] \tag{19}$$

$$\nabla_w \xi(\mathbf{W}) = 2e(k)\mathbf{X}(k) \tag{20}$$

By (20) and (17) a new weight vector can be approximated and can be written as shown in (21). The result is the LMS algorithm that is being introduced by Widrow and Hoff in 1960.

$$w_i(k+1) = w_i(k) - \mu e(k)\mathbf{X}(k) \tag{21}$$

The estimating equation of the weight vector as shown in (21) can be realized in a practical hardware without the need of squaring, averaging, inversion or differentiating and it is simpler to implement. Through iterative process and with the use of (4), (8) and (21), the weight vector **W**(k) will converge to the best weight vector at a right adaption step size μ.

The stability and convergence of the LMS algorithm depends largely on the adoption step size μ. This adaption step size is a small constant that controls on how fast the algorithm will converge and approximate the desired or best weight vector. Using (15) in (17) the result is

$$\mathbf{W}(k+1) = \mathbf{W}(k) - \mu\left[-\mathbf{P} + \mathbf{RW}(k) \right] \tag{22}$$

$$\mathbf{W}(k+1) = \left[\mathbf{I} - \mu\mathbf{R} \right]\mathbf{W}(k) + \mu\mathbf{P} \tag{23}$$

where ***I*** is an N-by-N identity matrix. [26] As *k* approaches to infinity, **W**(k) converges to the desired weight vector if the adoption step size is small enough. In addition, it shows that in (23) using a small adaption step size will lessen the effect of the autocorrelation **R** in approximating the weight vector; as a result, a more accurate approximation is expected because the weight vector is being varied not that large to cause instability of the overall adaptive system.

However even if the adoption step size is small it does not mean that the stability of the system is assured, one should also consider the maximum magnitude of the input signal x(k) of the adaptive system. It shows that in (23), the deviation in updating the weight vector also depends on in the input signal x(k) manifested in the autocorrelation **R**. Therefore in selecting an appropriate adaption step size, one should also consider the maximum input signal. As a consequence, the selection for the adoption step size in order to avoid instability is difficult and complex, [27]. The adaptive LMS algorithm is stable if and only if (24) is satisfied and can also be expressed in a form as shown in (25).

$$\left|1-2\mu\lambda_{\max}\right|<1 \tag{24}$$

$$0<\mu<\frac{1}{\lambda_{\max}} \tag{25}$$

The λ_{max} is the largest of Nth eigenvalues manifested in the autocorrelation **R**. In practical application the adoption step size is chosen to be smaller than the upper bound $\frac{1}{\lambda_{max}}$. For this reason, in order to guarantee the stability of the adaptive algorithm, the adoption step size should be varied in accordance with the input signal x(k). Thus, it reduces the effect of the autocorrelation **R** in (23) and the adoption step size will satisfy the limit as describe in (25), [26]. The theoretical limit of the adoption step size in (25) is limited in practical application because the autocorrelation **R** is usually not available and even if it were, computing its eigenvalue is undesirable chore. A reasonable approach is to have some bounds for the largest of the Nth eigenvalues, [26]. The average value of the dot product of the input vector **X**(k) with itself equals to the sum of the eigenvalues of the autocorrelation **R**.

$$\text{Avrg}\left[\mathbf{X}(k)\mathbf{X}^T(k)\right]=\sum_{i=1}^{N}\lambda_i\geq\lambda_{\max} \tag{26}$$

The result of the inner product of the input vector **X**(k) in (26) is the upper limit of a possible maximum eigenvalue λ_{max}. Therefore, the adoption step size shall be in a form of

$$\mu=\frac{\alpha}{\mathbf{X}(k)\mathbf{X}^T(k)} \tag{27}$$

so that μ will stay within the limits as describe in (25). The term α is a positive constant within 0 to 2, [26]. Using (27), (21) can be rewritten as.

$$w_i(k+1)=w_i(k)-\frac{\alpha e(k)\mathbf{X}(k)}{\mathbf{X}(k)\mathbf{X}^T(k)} \tag{28}$$

However if the system's input signal is equal to zero, the dot product of the input vector **X**(k) in (26) is also equal to zero. Hence, the resulting adoption step size in (27) will be excessively large and will definitely cause the entire system to be unstable. Thus, a smallest possible positive constant is added to the dot product of the input vector **X**(k), in order ensure that the updating term of the weight vector does not become excessively large. The resulting equation in updating the weight vector will be in a form shown in (29), where γ is a smallest possible positive value. The resulting equation in (29) is the NLMS algorithm presented by Albert and Gardner (1967).

$$w_i(k+1)=w_i(k)-\frac{\alpha e(k)\mathbf{X}(k)}{\gamma+\mathbf{X}(k)\mathbf{X}^T(k)} \tag{29}$$

To effectively eliminate the howling effect in public address system, the acoustic feedback should be eliminated and prevent the Nyquist criterion for instability from being met. This will require an equivalent model of the acoustic feedback path, in which its output will be subtracted to signal from the microphone. However, the equivalent model of an acoustic feedback path are not directly available and sometimes may vary with time. Hence, this paper uses an adaptive plant modeling scheme using an adaptive filter in obtaining an equivalent mathematical model of acoustic feedback path as illustrated in Fig. 3. The adaptive filter shown in Fig. 3 will imitate the behavior of an acoustic feedback path in which it is considered to be unknown in reality. Both the adaptive filter and acoustic feedback path will be driven by a loudspeaker signal and the objective of the adaptive filter is to match its output signal to the output signal of the acoustic feedback path by simply adjusting the weight vector of the adaptive filter through a LMS algorithm. Hence, the adaptive filter will have an equivalent transfer function to the acoustic feedback path when the weight vector cause the MSE $\xi(\boldsymbol{W})$ to be at its minimum, [27]. Upon convergence the structure and the parameter values may or may not be equal to those of the actual model of the acoustic feedback path but the input-output response relationship will be matched. In reality, the adaptive filter will not resemble an exact model of the unknown plant because of a minimum MSE $\xi(\boldsymbol{W})$ manifested by the adaptive LMS algorithm. This can be illustrated by considering MSE $\xi(\boldsymbol{W})$ in (11) with the use of (12) to (13), and then the MSE $\xi(\boldsymbol{W})$ is

$$\xi(\mathbf{W})=D-2\mathbf{W}^T\mathbf{P}+\mathbf{W}^T\mathbf{R}\mathbf{W} \tag{30}$$

On the other hand, the equation that describes the desired weight vector shown in (11) that causes the MSE $\xi(\boldsymbol{W})$ to be at its minimum value can also be written as

$$\mathbf{P}=\mathbf{R}\mathbf{W}^o \tag{31}$$

Therefore the MSE $\xi(\boldsymbol{W})$ is at minimum when the weight vector has reached its optimum value and can be evaluated as

$$\xi_{\min}(\mathbf{W}) = D - 2\mathbf{W}^{oT}\mathbf{P} + \mathbf{W}^{oT}\mathbf{R}\mathbf{W} \tag{32}$$

$$\xi_{\min}(\mathbf{W}) = D - \mathbf{W}^{oT}\mathbf{P} \tag{33}$$

Thus, the minimum MSE $\xi(\boldsymbol{W})$ described in (35) depends on the energy of the signal d(k), the weight vector **W** and the correlation of signal x(k) and d(k). In addition, random noise in the weight vector also adds misadjustment. If the weight vector were noise free and converged to its desired setting then the minimum MSE $\xi(\boldsymbol{W})$ will be equal to zero. However, because of gradient noise present in the weight vector, the weight vector **W** will be noisy and on the average it will be misadjusted from its desired or best setting and it will also exhibit a Brownian motion in the bottom of the MSE bowl, [28]. As a result, the average MSE $\xi(\boldsymbol{W})$ is greater than the minimum MSE. Therefore, it is impossible to bring down the minimum MSE equal to zero, [27]. Lesser misadjustment can be achieved by letting the adaptive filter adapt slower which can be done by having a smaller adoption step size, [28]. For this reason, the adaptive filter can only provide a closer approximate of the acoustic feedback path and not the exact one.

Meanwhile, the implementation of an adaptive filter based adaptive feedback cancellation for acoustic feedback control is similar to the well-known acoustic echo cancellation (AEC) approach, [2]. The adaptive filter is used to model and track the acoustic feedback path. While the equivalent model will be used to produce a feedback compensated electroacoustic signal. However in AFC, the disturbance signal, and the adaptive filter input signal are correlated. The correlation of the disturbance signal and the adaptive filter input signal will lead to a biased estimate of the acoustic feedback path, [29]. Furthermore, one of the biggest problems in using the adaptive filter for acoustic feedback cancellation is the biased estimation of the acoustic feedback path, [29]. The effect of a biased estimate will lead to a large modeling error and a cancellation of the desired signal, [30]. The resulting effect of the biased estimate is that the adaptive filter does not only predict and cancel the feedback component in the microphone signal but also part of the source signal, [2]. As a consequence, the feedback compensated signal in the electroacoustic path is a distorted estimate of the microphone signal. Hence, it is important to eliminate the correlation of the disturbance signal and the adaptive filter input signal. The concept adaptive feedback canceller is quite simple and similar to AEC however its realization is not straightforward [2]. This is because aside from the main objective of the adaptive feedback canceller which is to eliminate or prevent the occurrence of howling, the adaptive feedback canceller must also have the capability to eliminate the correlation of the disturbance signal and the adaptive filter input signal. Thus, the only way to avoid a biased estimate of the acoustic feedback path is to decorrelate the disturbance signal and the adaptive filter input signal.

When an equivalent model of the acoustic feedback path is achieved by the adaptive filter, the overall transfer function of the public address system shown in Fig. 3 will be equal to the equation shown in (34).

$$\frac{d(z)}{v(z)} = \frac{G(z)}{1 - G(z)\left[F(z) - F'(z)\right]} \tag{34}$$

The term F'(z) is the equivalent model of the acoustic feedback path F(z). It is shown in (34) that acoustic feedback problem will be totally eliminated if the acoustic feedback path and its equivalent model are exactly matched. As a result, the MSG of the public address system is infinite or it would be impossible to experience a howling effect when F(z) and F'(z) are exactly match in any cases. However, as stated previously that it is impossible to obtain an exact match of F(z) and F'(z), then we expect that the overall performance will be improved and the maximum stable gain will be bounded to a certain level higher to the system without adaptive feedback canceller. In this case, in order for the public address system with adaptive feedback cancellation to remain stable and prevent the occurrence of howling effect, then it must avoid satisfying the Nyquist criterion for instability. The public address system must remain in the condition as shown in (35) and (36) in order to avoid howling effect.

$$G(z)\left[F(z) - F'(z)\right] \geq 1 \tag{35}$$

$$\angle G(z)\left[F(z) - F'(z)\right] = n(2\pi) \qquad n \in \mathbb{Z} \tag{36}$$

On the other hand, considering Fig. 3, the microphone signal d(z) is described as the linear sum of the voice signal d(z) and acoustic signal f(z) which is mathematically expressed as shown in (37).

$$d(z) = v(z) + \mathrm{f}(z) \tag{37}$$

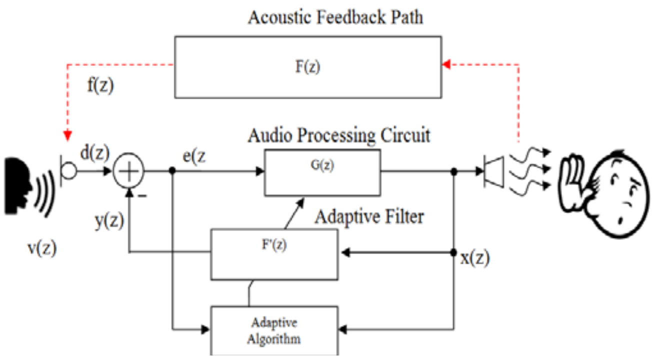


Fig. 3. Block diagram of a single channel public address system with adaptive feedback canceller

If (37) is being used to express the weight vector described in (16) with the microphone signal d(z) as the desired signal of the adaptive filter, then the resulting equation is shown in (39).

$$\mathbf{W}^{o}(z)=\left[\mathbf{X}(z)\mathbf{X}^{\mathrm{T}}(z)\right]^{-1}\left[\mathbf{X}(z)d(z)\right] \tag{38}$$

$$\mathbf{W}^{o}(z)=\left[\mathbf{X}(z)\mathbf{X}^{\mathrm{T}}(z)\right]^{-1}\left[\mathbf{X}(z)\left[v(z)+\mathrm{f}(z)\right]\right] \tag{39}$$

$$\mathrm{f}(z)=\mathbf{X}^{\mathrm{T}}(z)\mathbf{W}^{o}(z) \tag{40}$$

Simplifying (39) and let the feedback signal f(z) be equated with the ideal weight vector $\mathbf{W}^{o}$ and loudspeaker signal $\mathbf{X}(z)$ as shown in (40). Then the ideal weight vector in (41) can be expressed as

$$\mathbf{W}^{o}(z)=\mathbf{W}^{o}(z)+\left[\mathbf{X}(z)\mathbf{X}^{\mathrm{T}}(z)\right]^{-1}\left[\mathbf{X}(z)v(z)\right] \tag{41}$$

In (41) it shows that ideal weight vector is biased by the term $[\mathbf{X}(z)\mathbf{X}^{T}(z)]^{-1}[\mathbf{X}(z)v(z)]$. The biased in the weight vector will lead to a distorted feedback compensated electroacoustic signal. The biased problem shown in (41) in adaptive feedback canceller can be resolved if the term $\mathbf{X}(z)v(z)$ is equal to zero. This can be done by applying a decorrelating technique on either of the electroacoustic forward path or in the adaptive filter. Decorrelation can be achieved by 1) inserting a white noise signal, 2) applying a nonlinear filter or 3) a delay which is shown in Fig. 3. In this study, a unit delay decorrelating technique of 400ms was used.

$$x(z)=G(z)\left[e(z-D)\right] \tag{42}$$

The input signal of adaptive filter in adaptive feedback canceller with delay as decorrelation technique is shown (42). The implementation of a delay based decorrelation technique for adaptive feedback canceller may be simple and straight forward but the delay D should be chosen accordingly. The delay d should be chosen to not so large that it could not cancel the growing feedback signal and not so small enough that it is unable to decorrelate the feedback signal and the input signal. Meanwhile, the resulting correlation effect is similar to having an adaptive filter preceded by a processing delay and when a processing delay is inserted in the electroacoustic forward path, [31].

The use of decorrelation in adaptive feedback canceller exhibit a tradeoff between bias reduction and sound quality. Usually, a perceptible signal distortion is unavoidable because of the decorrelation operation or because of the bias in the acoustic feedback path estimate.

The delay is inserted in the electroacoustic path before the electroacoustic signal is being inputted to adaptive filter input. It is indicated in [32], that a delay of 1ms placed in the cancellation path is sufficient for decorrelating speech signals. Also, a delay of 2ms was introduce in order to reduce the correlation in the speech signal, [3]. However, it is important to note that the delay in the forward path is particularly useful for near-end signals that have an autocorrelation function that decays rapidly such as voiceless speech signals, provided that the delay value D is chosen accordingly, [2]. The selection of the delay D, should be chosen to be large enough such that the speech was largely uncorrelated with itself, while the delay D should be small enough such that the howling could be canceled before it grew too much in magnitude, [3]. Moreover, by making D correspond to the same delay imposed by the cascade of the ADC and DAC, the adaptive filter can be made to converge to a transfer function that models the transfer function of the cascade of the DAC, speaker, room, microphone and ADC, [3]. But it was emphasized in [3] that the delay length in the adaptive filtering circuit should not exceed the initial delay in the acoustic feedback path impulse response.

Moreover, in any given room's impulse response can last up to few seconds, one might imagine the need for adaptive FIR filters implementing tens of thousands of adaptive coefficients, [3]. But because the talker might move quickly, an adaptive filter should adapt quickly. However, the MSE of the least mean square (LMS) algorithm increases with the number of filter taps when the rate of adoption is held constant. This implies that very long filters should have a convergence problem. Adaptive feedback cancellation has been accounted in real time implementation, however, one of its main disadvantages is its computational complexity. Computational complexity in adaptive feedback canceller can be attributed to 1) a very high adaptive filter order is required because the acoustic feedback path is being modeled by its impulse response and 2) the impulse response is densely sampled which requires many coefficients and a large number of iterations has to be performed per second. Thus, it has to be noted that high adaptive filter order implementation has to consider 1) the number of multiplication or DSP blocks to be used and 2) latency of the of hardware were the adaptive filter will be implemented.

3 Methodology

In this study, a Finite Impulse Response (FIR) adaptive filter was used as an automatic acoustic feedback controller. It was implemented in real time using FPGA development board with an embedded audio codec.

An overview of the experimental setup of this study shown in Fig. 4. This was implemented to observe how the adaptive feedback canceller will behave using an FPGA development board. The adaptive feedback canceller was designed in FPGA development board. The acoustic signal from the talker and the acoustic feedback signal was picked up by a microphone. In the microphone, the acoustic signal was converted to electrical signal then it will be feed to the FPGA development board for processing, enhancement and more importantly to eliminate the acoustic feedback. Then the output signal of the FPGA development board is a feedback compensated electroacoustic signal where it will be amplified by a power amplifier and converted back to acoustic signal using a loudspeaker.

In Fig. 4, the FPGA development board will receive the electrical signal from the microphone through its embedded audio codec ADC. The audio codec ADC is responsible for converting an analog electrical signal from the microphone to digital format. Furthermore, the audio codec pre-amplified the analog signal in order to restore its integrity. Then it was sampled at a sampling rate of 48KHz and quantized. After quantization, it is coded to a specified 16 data bits. The audio codec's operational settings are to be configured by addressing its internal register using I2C protocol. The module that will address the internal register of the audio codec through I2C protocol was designed in FPGA. Meanwhile, the data from the audio codec ADC is in serial form and it will be received by a digital audio interface Rx module. The digital audio interface Rx is responsible for converting a serial data to parallel form. The data from the digital audio interface Rx module represents the acoustic signals that have been picked up by the microphone.

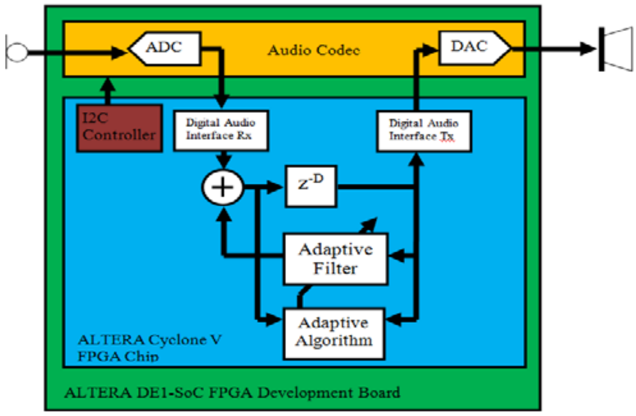


Fig. 4. Block diagram of an FPGA-based adaptive feedback canceller

In order to eliminate the acoustic feedback signal that has been picked up by the microphone, the data from the digital audio interface Rx module was sent to a summer block. In the summer block, the acoustic feedback signal will be eliminated by subtracting the signal from the digital audio interface Rx module with the adaptive filter's output signal. The resulting difference is a feedback compensated signal and it will be delayed by D sample before it will be sent to digital audio interface Tx for parallel to serial conversion. Meanwhile the adaptive filter takes its input from the delayed feedback compensated signal. The adaptive filter through its adaptive algorithm obtained an equivalent model of the acoustic feedback path so that the adaptive filters output signal is an exact match of the acoustic feedback signal being picked up by the microphone. The equivalent model of the acoustic feedback path is obtained by varying the weights of the adaptive filter through its adaptive algorithm. The adaptive algorithm varied the weights of the adaptive filter based on the feedback compensated signal and the delayed feedback compensated signal. The objective of the adaptive algorithm is to match the adaptive filter's output signal and the acoustic feedback signal. When a best match was achieved, the signal that was sent to the digital audio interface Tx module is now a feedback compensated signal. The digital audio interface Tx module converts the parallel form of the acoustic feedback compensated signal to a serial form for digital to analog conversion. The analog signal from the DAC was amplified by a power amplifier and convert it back to an acoustic signal by a loudspeaker.

The adaptive filter composed of multiplier, adder and unit delay blocks is shown in Fig. 5. The adaptive filter in Fig. 5 together with the adaptive algorithm shown in Fig. 6 will model the acoustic feedback path and attempts to cancel the acoustic feedback signal as quickly as possible.

The implementation of the adaptive filter requires 2L multipliers and 2L + 1 adder. One of the multiplier arrays was intended for the multiplication of the weight vector and the delayed input, while the other is for weight vector updating operation. The output of each tap of the tapped delay line is multiplied by the appropriate filter coefficient w(n) and the results are summed up. The response of the adaptive filter y(z) and the error of the overall system e(z) are largely dependent on the adaptive filter input signal x(z) and the desired signal d(z). The desired signal d(z) is the signal that comes from the digital audio interface which is the discrete representation of acoustic signal being pickup by the microphone. While the adaptive filter input signal x(z) is the discrete representation of the output signal of the loudspeaker.

On the other hand, the hardware implementation of the weight updating algorithm is shown in Fig. 6. The input signal x(z) was multiplied by its self first to obtain a squared value of the signal x(z). Then the D-type flip-flop was used in order implement a recursive function of the squared value signal x(z). After obtaining the recursive function, a constant γ was added then it was reciprocated. The constant γ was added in order to avoid an excessively large weight updating function. Afterward, the output of the divisor IP block was multiplied by a constant *a* and the error signal e(z). The constant signal *a* determines the rate of convergence of the adaptive filter.

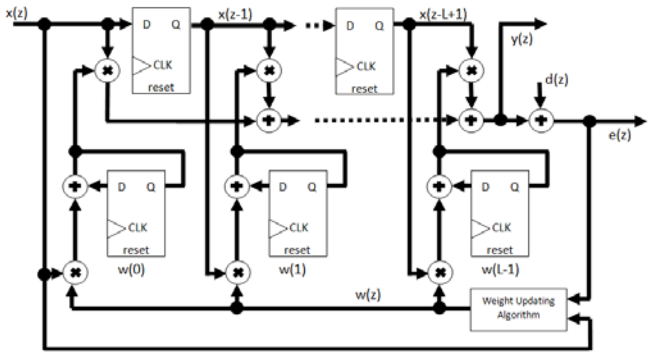


Fig. 5. Hardware implementation of adaptive filter

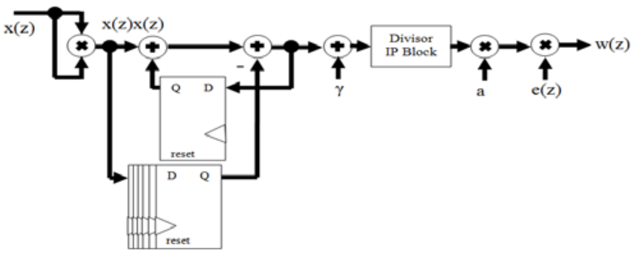


Fig. 6. Hardware implementation of NLMS adaptation step size

Meanwhile, part of the output of the squared function block was delayed by N+1 sample time and the other part was added to the previous value of $x^2(z)$. The N is an integer that represents the length of the tapped delay line of the adaptive filter. Then $x^2(z\text{-}N\text{+}1)$ was subtracted from the sum of $x^2(z)$ and $x^2(z\text{-}1)$ in order to limit the recursive function up to N-1 samples.

The implementation was evaluated using a microphone, a speaker with an embedded power amplifier and an FPGA for the realization of an NLMS based acoustic feedback canceller. The microphone was placed in the direction of the acoustic signal from the loudspeaker. Then, the acoustic environment of the room was observed by an external observer for the presence of howling effect. The gain of the public address system at which the audible ringing or howling first occur is said to be the MSG of the public address system. To quantify the improvement of MSG of the public address system with adaptive feedback canceller, first a public address system was used without adaptive feedback canceller and the gain was slowly increased until an audible ringing or howling was observed. Then the adaptive feedback canceller was employed and the gain was slowly increased until an audible ringing or howling occur. The difference between the two gains is said to be the improvement of MSG, [2]. The advantage of using this kind of evaluation is that the evaluation results are directly linked to the system stability.

The loudspeaker was placed at the centre front of the room and the microphone was placed in the direct path of the acoustic signal coming from the loudspeaker. Meanwhile, the loudspeaker and the microphone was connected to an FPGA development board in which it acts as an adaptive feedback canceller. An acoustic observer in a form of a voice recorder was placed at the centre of the room fronting the loudspeaker. Having a voice recorder that serves as an acoustic observer allow to visualize how a person perceived the sound coming from the loudspeaker. The signal from the acoustic observer will then be used for analysis and evaluation on the performance of the system.

The setup will be run without employing the acoustic feedback canceller and the gain of the amplifier will be increased until howling will occur. When howling occurs, the acoustic observer records the acoustic signal for analysis and evaluation. Also, the gain of the amplifier when howling first occurred is noted so that it will serves as a reference in quantifying the improvement of acoustic feedback canceller.

Then the set up was re-run, this time the adaptive feedback canceller was employed. The gain of the amplifier was also slowly increased until howling occur. When howling occurs, the acoustic observer records the acoustic signal for analysis and evaluation. Also, the gain of the amplifier when howling first occurred was noted and the difference of the gains with and without adaptive feedback canceller is said to be the improvement of ∆MSG.
The procedure was also re-run with different filter lengths of 10, 20, 30 and 40 filter taps and adaptation constant of 0.1, 0.01, 0.001 and 0.0001 in order to obtain a relationship with MSG and filter length, MSG and adaption step size. Results will determine how the filter length and adaption constant will affect the MSG of the adaptive feedback canceller.

Furthermore, the achievable amplification of public address system at different parameters of the adaptive filter was evaluated. An acoustic signal in a form of music was applied at the microphone of the public address system without adaptive feedback canceller. The power amplifier's gain of the public address system was slowly increased, until howling occurs and the output was observed. Then the gain of the power amplifier of the public address system was decreased to 3dB lower and the music signal was replayed and the output was also observed and recorded. An adaptive feedback canceller was then employed in different operational settings and the evaluation was re-run. The adaptive filter with 400ms decorrelator was set to adapt for approximately 20s at a gain 3dB lower to which howling occurred for public address system without acoustic feedback canceller before the gain was slowly increased by 1dB/s until 1dB lower of the power amplifier's gain to which howling occurs. Then the acoustic output signal of the public address system was recorded for analysis and comparison. This kind of method for the evaluation of the of public address system with acoustic feedback canceller was used in [2] [3].

Because it is quite difficult to compare the result in time domain, the comparison of the recorded results was done in the frequency domain. All recorded results were compared in the frequency domain to quantify which of the recorded results achieved greater amplification at a frequency were in the spectrum of the information signal is significant and without the presence of howling effect. At the significant band of the recorded results, the magnitudes were averaged and accounted for a comparison on the achieved amplification of the public address with different operational settings of the adaptive feedback canceller.

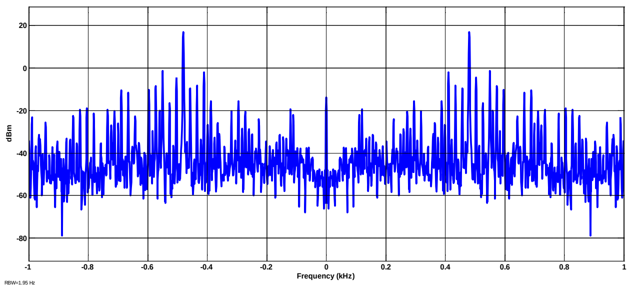
4 Results

Without an acoustic input signal, the gain was slowly increased until an audible howling occurs. The acoustic activity of the room was observed using spectrogram and spectrum analyzer. The results of the spectrum and spectrogram of the observed acoustic is shown in Fig. 7. The results in Fig. 7 happened when the gain of the public address system is -3dB. In Fig. 7, a narrow band of frequency at approximately 480Hz dominates for a specific period of time. Based on the results, this shows that an acoustic feedback path exist which have a close loop resonant frequency of approximately 480Hz.

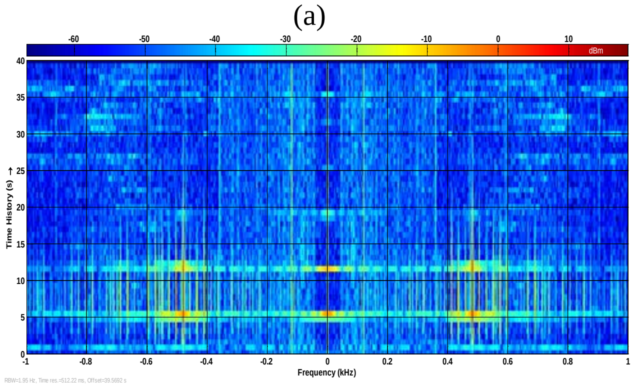
Results shown in Fig. 7 illustrates that the observed acoustic activity of the room had a single frequency component whose magnitude dominates the audible frequency band. Furthermore, spectrograph also shows that the frequency component that dominates remains the same throughout the existence of acoustic feedback. Also the frequency of the acoustic feedback signal may vary as the talker transfer from one location to the other and it might be difficult to address the existence of acoustic feedback with the use only of fixed parameters notch filter.
With music as an input signal, the gain was slowly increased until an audible howling occurs. The acoustic activity of the room was observed using spectrograph and spectrum analyzer. The spectrum and spectrograph results of the observed real-time acoustic activity is shown in Fig. 8 for music signal without howling that happened at -6dB of the public address system and Fig. 9 for music signal with howling at -3dB of the public address system. In Fig. 8, spectrograph show that dominant magnitudes were distributed on the significant band of the information signal. It can be also seen that there are no frequency components that exhibit similar characteristics of the dominant frequency shown in Fig. 7. Thus, the observed signal shown in Fig. 8 had no audible annoying monotone signal.

However when the gain of the public address system was increased from -6dB to -3dB, an audible monotone signal was observed. When the output of the public address system was visualized using spectrum analyzer and spectrogram, the howling signal observed in Fig. 7 is seen again in Fig. 9. Fig. 9 shows that the howling signal exists for a period of 2.5 minutes at 480Hz. It is also observed that the frequency of the howling signal did not change for a period of 2.5 minutes. Thus, results in the real-time characterization of howling signal in public address system with music as an input signal support as described in equation (37). Therefore acoustic feedback signal is an audible periodic monotone signal and when an information acoustic signal is applied the result of the public address system output would be a linear sum of the acoustic feedback signal and information signal. With these, it is important to only eliminate the dominant narrow band howling signal shown in Fig. 9.

Using the information of the characteristic of acoustic feedback, the adaptive filter was then employed to effectively eliminate or reduce the effect of acoustic feedback in a public address system. For this reason, a low order adaptive filter was employed and its performance was investigated. The performance of the adaptive filter was evaluated based on the MSG and achievable amplification of the public address system. The effect of the filter length and adaptation constant on the MSG and achievable amplification was presented.



(a)



(b)

Fig. 7 Spectrum and spectrograph of acoustic feedback in real time

With the use of NLMS based adaptive feedback canceller, the acoustic feedback with public address gain at -3dB was eliminated. However, because of the limitation of the adaptive filter, acoustic feedback may again exist at a higher gain. The difference between the two gains is the ΔMSG. The ΔMSG was measured at different adaptive filter length and adaptation constant. Results are then presented in Table 1 in decibels.

It can be seen in Table 1 that the improvement of MSG is at 5 to 6dB for NLMS adaptive algorithm, [2] [33]. The work of Goertz also observed a 5dB MSG increase in a severely undermodelled adaptive feedback canceller with noise injection, [2]. Goertz work is said to be undermodelled because the length of the adaptive filter was only 1/15 of the acoustic feedback path length or at 2646 Filter taps, [2] [34]. But the work of Rombout have reported a 14dB increase in MSG with frequency domain adaptive filter of the order of 2048, [3]. Meanwhile, because howling signal only consists of one sinusoidal signal as also illustrated in this paper, theoretically only two taps are required for an adaptive filter as an adaptive feedback canceller.

The result in this study, the effect of the filter length on the ΔMSG during evaluation may be minimal or negligible. But, it is observed that ΔMSG increases proportionally with adaptation constants. This means that adaptive filter with a faster rate of adaptation offers higher ΔMSG. This is because adaptive filter quickly cancels the acoustic feedback before it grows uncontrollably. However, the disadvantages of having a large adaptation constant are it approximate the acoustic feedback signal less accurate and precise. Thus, the adaptive filter at higher adaptation constant may severely affect the public address system's achievable amplification and sound quality when an acoustic signal in a form of speech or music is applied.

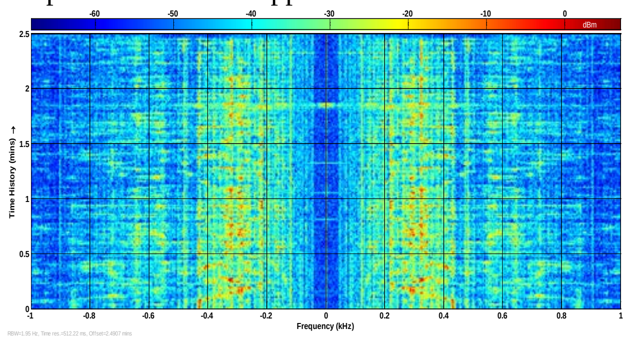
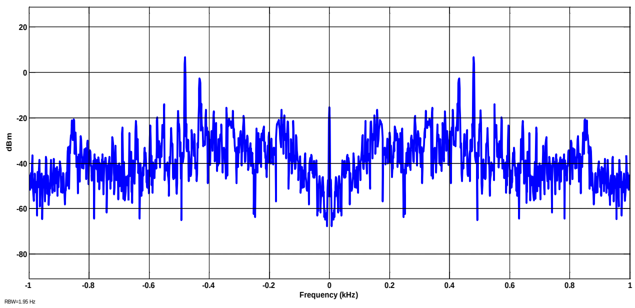


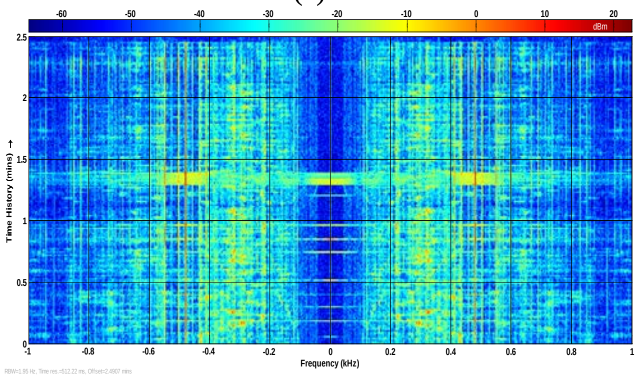
Fig. 8 Music without Howling Signal with PA at -6dB

Table 1 ΔMSG(dB) of NLMS based adaptive feedback canceller at different adaptation constant and adaptive filter length

NLMS		Adaptation Constant				
		0.0001	0.001	0.01	0.1	AVG
Filter Taps	10	5	5	6	6	5.50
	20	5	5	6	6	5.50
	30	5	5	6	6	5.50
	40	5	5	5	6	5.25
	AVG	5	5	5.75	6	



(a)



(b)

Fig. 9 Music without Howling Signal at with PA at -3dB

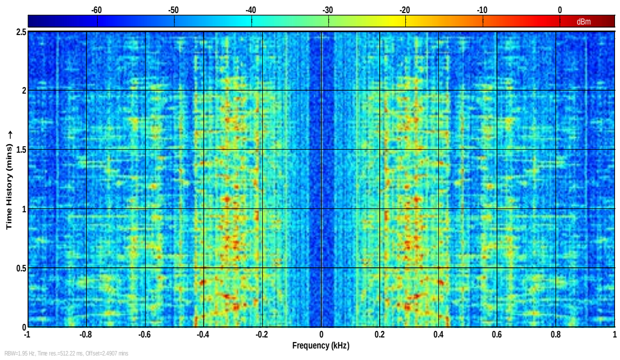


Fig. 10 Music of PA with NLMS-AFC 20Taps & u=0.0001 (2dB)

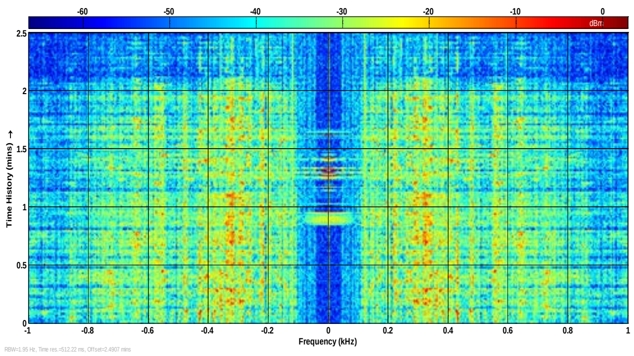


Fig. 11 Music of PA with NLMS-AFC 20Taps & u=0.1 (2dB)

To visualize the effect of the adaptation constants and filter length, a music signal was applied to the public address system with NLMS based adaptive feedback canceller. The adaptation constant being considered is 0.0001 and 0.1. Also, the length of the adaptive filter being considered is 10, 20, 30, and 40. The gain of the public address system with the acoustic input signal and adaptive feedback canceller was slowly increased until howling occurs. Then the system was reset and the gain of the public address system was again slowly increased up to 1dB lower to which howling occurred. Then the acoustic activity of the room was observed by a spectrograph and then recorded for analysis.

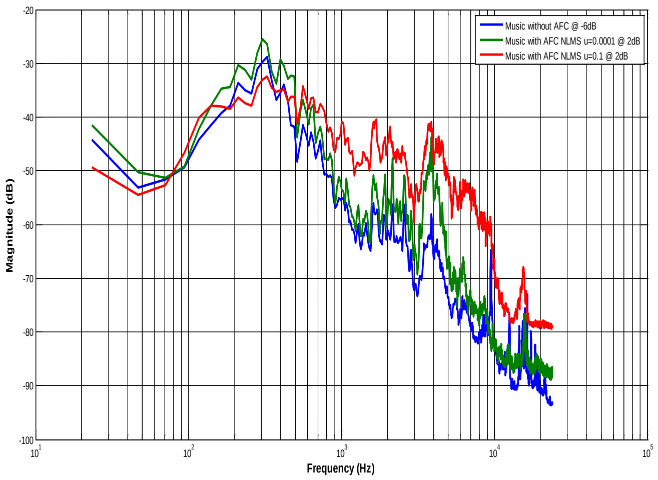


Fig. 12 Music Spectrum with and without NLMS-AFC @ 20 Taps

Table 2 Observed average magnitudes of public address system's output signal with NLMS based adaptive feedback canceller with different adaptive filter settings

		Music (Ref. Signal @ AVG=-38.14dB & MAX=-28.87dB)	
		AVG (dB)	
		u=0.0001	**u=0.1**
Filter Taps	**10**	-34.08	-35.27
	20	-33.90	-36.54
	30	-35.64	-31.89
	40	-34.14	-35.28

The spectrograph of the public address system with adaptive feedback canceller is shown in Fig. 10 and Fig. 11. In Fig. 10 and Fig. 11, the adaptive filter had a length of 20 filter taps and the adaptation constant are 0.0001 and 0.1 respectively. It can be shown in Fig. 10 and Fig. 11 that acoustic feedback at 480Hz previously described was minimized with the use of an adaptive feedback canceller. This means that the adaptive filter was able to adapt in order to effectively reduce the effect of the acoustic feedback. However, it is quite difficult to determine which of the signals presented in Fig. 10 and Fig. 11

offers an amplification to the acoustic input signal and determine by how much it was amplified. The signal presented in Fig. 8, Fig. 10 and Fig. 11 was compared in frequency domain and the results are presented in Fig. 12.

Taking the frequency spectrum of the signals shown in Fig. 12 allows us to quantify the magnitudes of each frequency components. Hence, the difference in magnitudes further determines the achieved amplification of the public address system with adaptive feedback canceller. This will also show the difference in the achieved amplification of the public address system with adaptive feedback canceller at an adaptation constant of 0.0001 and 0.1.

In reference to the signal illustrated in Fig. 8, it shows that the significant frequency band of the information signal is at approximately 100Hz to 600Hz. As a result, the magnitudes of the frequency spectrum of the signals in Fig. 12 was only compared at a frequency band of 100Hz to 600Hz.

In Fig. 12, results show that the public address system with an adaptive feedback canceller at an adaptation constant of 0.0001 had a greater magnitude as compared to the public address system without adaptive feedback canceller and public address system with adaptive feedback canceller at an adaptation constant of 0.1. Also, results in Fig. 12 shows that the magnitudes of the adaptive feedback canceller at an adaptation constant of 0.1 are similar to the public address system without adaptive feedback canceller at the frequency band of 100Hz to 600Hz. Taking the average of the magnitudes of the three signals from 100Hz to 600Hz, the public address system without adaptive feedback canceller had an observed magnitudes of -38.14dB. While the public address system with adaptive feedback canceller at an adaptation constant of 0.0001 and 0.1 have an observed magnitude of -33.9dB and -36.64dB respectively. Hence, this shows that on average the public address system at an adaptation constant of 0.0001 had an achieved amplification of approximately 4.24dB and the amplification achieved is greater than the amplification achieved of the public address system with adaptive feedback canceller of 0.1 by 2.74dB, [2]. The simulation work of Waterschoot have reported a 6-9dB of amplification with speech as an input signal and an amplification of 5-9dB using an NLMS based adaptive filter, [3]. While the work of Berdahl reported a real-time observable amplification of 1.5-2dB depending on the orientation of the microphone, speakers and other objects in the room using a 20-taps LMS based adaptive filter. The results of the work of Berdahl are comparable to results on this work. But Berdahl did not introduce a 3dB gain margin, this is why the results of this study are slightly greater than 3dB to the work of Berdahl.

Moreover, at higher frequency components to which are insignificant to the information signal, the public address system with an adaptive feedback canceller at an adaptation constant of 0.1 had an average magnitude comparable to the average magnitude of the public address system with adaptive feedback canceller at an adaptation constant of 0.0001. Thus, it shows that adaptive filter with an adaptation constant of excessively large for an adaptive filter that act as an adaptive feedback canceller may not only lead to higher ΔMSG but also causes an attenuation at neighboring frequency component of the howling signal. This is because the adaptive filter fails to accurately and precisely eliminate the howling signal. As a result, it causes signal distortion which tends to degrade achievable amplification and sound quality.

To compare the effect of adaptation constant of the adaptive filter and adaptive filter length to the achievable amplification of the public address system, the average of the observed acoustic signal's magnitudes at the significant audible band from 100Hz to 600Hz were tabulated. Tabulated results are shown in Table 2. It shows in Table 2 that public address system with adaptive feedback canceller and with lower adaptation constant of the adaptive filter have a greater achievable amplification compared to the adaptive filter's higher adaptation constant. This is because the adaptive filter fails to approximate the acoustic feedback with large adaptation constant. As a result, the adaptive filter attenuates and distorts the information signal. Thus a lower achievable amplification is observed for those adaptive filter with higher adaptation constant. It is also observed in Table 2 that adaptive filter length may have less significant impact on the achievable amplification of the public address system with adaptive feedback canceller.

Results in Table 2, may show that the key property of adaptive filter as acoustic feedback canceller is its adaptation constant. Thus, the adaptive filter must adapt as quickly as possible before the acoustic feedback signal grows uncontrollably. In addition, the adaptive filter must also approximate the acoustic feedback signal accurately and precisely in order to achieve greater amplification without a noticeable audible howling signal and superior sound quality. With this, adaptive algorithm's rate of adaptation and precision plays an important role in achieving the optimum performance of adaptive feedback canceller.

5 Conclusion

In this paper, an adaptive filter using NLMS adaptive algorithm was employed as an acoustic feedback canceller for public address system. The public address system with adaptive feedback canceller was evaluated in real time environment using Alterra DE1-SoC. The adaptive filter using NLMS adaptive algorithm was implemented in an Altera Cyclone V FPGA with the objective that is to classify and eliminate howling signal when it occurs. The implementation was done at different filter taps and adaption constant and its effect was analyzed.

Results show that the adaptive filter using NLMS adaptive algorithm successfully classify and automatically eliminates the howling signal. The result also shows that with the incorporation of adaptive filter as adaptive feedback canceller allows the public address systems to have an additional gain of 5dB to which it still exhibits a stable behaviour. In addition, the acoustic signal of public address system at a fixed gain shows that the one with adaptive feedback canceller is greater than 3dB to 4dB on average. However, the effect on the selection of the adaptation constant is far more significant than the length of the adaptive filter. Results show that, with music signal applied on the microphone, the adaptive feedback canceller with higher adaptation constant quickly reacts to address the presence of acoustic feedback. But higher adaptation constant does not only attenuate the howling signal but also attenuates the music signal being applied at the microphone. This shows why the adaptive feedback canceller with smaller adaptation constant achieved larger amplification compared to adaptive feedback canceller with large adaptation constant

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