# Acoustic Feedback Cancellation in Hearing Aids Using Least Squares Delay Based Algorithm in Fuzzy Interactive Controller

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***Abstract**: The acoustic feedback is a persistent and annoying issue in hearing aids, which either limits the maximum gain accessible by the user or degrades the sound quality of the device or both. Feedback cancellers are utilized to produce a replica of the feedback signal (FS), in order to remove the original FS before being delivered to the loudspeaker component; however, a bias due to correlation of input-output signals, was introduced into the system, due to spectrally colored nature of input signal. The Least Squares Delay (LSD) based adaptive algorithm was utilized, in which the filter coefficients were updated individually to minimize the bias, however under slow convergence rate. A novel approach of introducing a Fuzzy interactive controller (FIC), evaluates the performance measures of LSD algorithm, by incorporating a Rule-Decision-Table to generate the linguistic rules. The FIC-LSD algorithm adapted the system output in proportion to the magnitude of estimated filter coefficients. The steady-state performance achieved through FIC, were compared with various well-known adaptive techniques and the obtained results proved that the FIC integrated LSD algorithm would provide a significant and robust acoustic feedback cancellation in the presence of varying environmental conditions.*

***Keywords:** Acoustic Feedback, Least Squares filter, adaptive fuzzy controller, Hearing aids, Stable gain*

## 1. INTRODUCTION

The acoustic feedback is a technical occurrence that tends to happen in hearing aids to cause howling or whistling effect that makes unpleasant effect to the consumers. The feedback exists due to high intensity oscillation and make the low-level signal to fall below audibility.

The input sound signal that gets into the microphone, passes through the hearing aid processing and reaches the loudspeaker. Whenever the input signal followed only this forward path, and the system behaves like an open loop. However, at this point, the loudspeaker signal finds an alternate path, termed as acoustic feedback path, reaches the microphone and forms the closed loop. The simple block schematic in Figure 1, shows the forward and feedback path of an acoustic signal in a hearing aid.

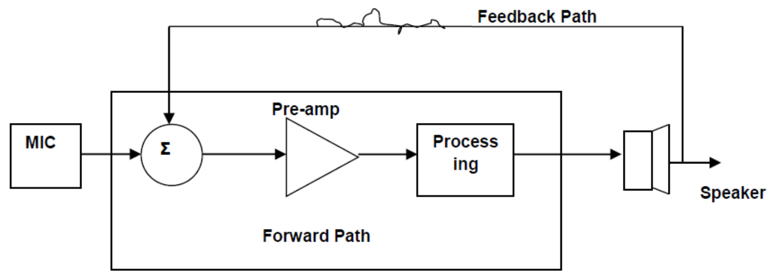


**Figure 1. A Block Schematic of a Hearing aid**

Whenever the frequency-gain of the open loop system is greater than unity and the open loop phase response is multiple of $2\pi$, the acoustic feedback is said to be occurred (Qingyun et al. 2008) and the presence of feedback signal in a hearing system leads to the onset of oscillation and limits the maximum gain that can be provided by the device. Hence the normal hearing aid processing gets interrupted and makes the device unused.

## 2. LITERATURE SURVEY

Number of feedback cancellation algorithms are proposed in recent years (Kates 1991a, Shin et al. 2007, Lee 2017), which follow either non-continuous or continuous techniques. Bustamante et al. (1989) proposed a non-continuous adaptive feedback algorithm in which the feedback filter is operated after the application of delay in the receiver output. It adapts to minimize the error continuously between the filtered output and microphone input (Kates 1991a). But during the estimation of feedback path, the algorithm requires a periodic update schedule or an update request by the user (Kates 1991b), which is time consuming.

In Levitt et al. (1998), a fixed phase shift filter is inserted in the signal path of the time-invariant cancellation filter, to compensate the open loop phase response. Although an adaptive filter is employed on estimating the transfer function of feedback path, to reduce the system difficulties due to the changes in acoustic environment, which in turn causes an alteration in the closed loop system response (Bustamante et al.1989).

Freedb (2008) stated that whenever the feedback signal is much greater than the input of microphone preamplifier, a clipping of feedback signal is offered. It is ensured by the difference between maximum acoustic output of hearing system and attenuation by the available feedback path, which is lesser than the maximum acoustic input to the system, the clipping of feedback signal occurs, that makes the signal to get distorted (Maxwell and Zurek 1995). The effects of non-linearity during the performance of feedback cancellation also modifies the system algorithm, which in turn, react to the sinusoid and cause a re-estimation of the feedback path and increases the processing time (Kates 1990).

In continuous adaptive feedback cancellation method, probe signals are injected for varying purposes, in order to preserve the quality of speech, speed of convergence and to maintain a steady state SNR (Akhtar and Nishihara 2016). Maintaining the level of probe injection is important as; Shorter probe sequence of random noise causes a small reduction in intelligibility (Kates 1990). The usage of continuous probe sequence is avoided as it would mask the desired speech signal (Maxwell and Zurek 1995). When intense probes are applied, the speech intelligibility will get affected significantly (Shin et al. 2007, Kates 1991).

In non-continuous adaptive technique, a standard, predefined notch filter was used in adaptation for initial convergence of the desired response. An adaptation procedure of notch filtering is helpful in achieving the rapid convergence of the system coefficients. Furthermore, it modifies the amplifier's transfer function to make the system stable. However, the presence of interfering signals at microphone, degrade the accuracy of filter estimation.

A notch filter of order 8 is setup, with pass-band ripple as 1, whose magnitude and phase response for the normalized frequency (rad/sample) is achieved at the center frequency at 0.75 and quality factor 10 respectively. When a Notch filter is employed in the adaptive system of hearing aid, the center frequency of feedback signal is adapted to reduce the largest spectral peak (Akhtar and Nishihara 2016), (Maxwell and Zurek 1995). When the microphone signal is passed to the filter and whenever the energy in notched band exceeds a specified threshold, the system output becomes the output of notch filter; else the input is passed unchanged (Akhtar and Nishihara 2016).

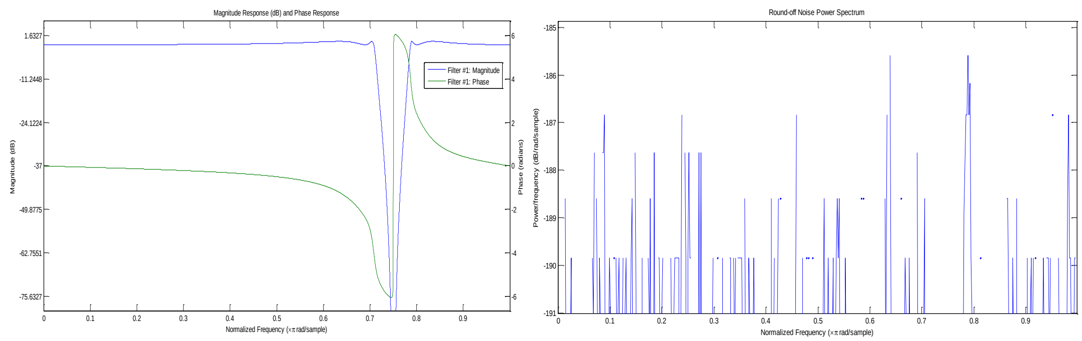


**Figure 2. Characteristics of Notch Filter adaptive system (a) Magnitude (dB) and Phase Response (radians) (b) Round-off Noise Power Spectrum**

In adaptive notch filter, the power of largest sinusoid is estimated and detected by updating the filter coefficients. This is done by finding the ratio of power input to the adaptive filter to the power output from it, which is then compared with a threshold, averaged by 1-pole LPF (Kates 1990). The FIR portion of notch filter is adapted to track the frequency of the sinusoid and the IIR portion of the filter gets copied by the coefficients (Levitt et al. 1988), (Kates 1990). Thus, the notch filter continuously adjusts its coefficients to emulate the impulse response of the feedback path and increases the computational complexity of the system. The Figure 2 (a) and (b) shows the Frequency characteristics of Notch Filter adaptive system and its Round-off Noise Power Spectrum for the notch filter system.

Most of the feedback cancellation techniques have the drawback of performing well in a particular configuration, may not work in another configuration. Hence it is necessary to increase the stable gain in an HA device, to make the device perform well in any hearing aid configuration and to be computationally efficient.

Also, there is a danger of removing the ambient signal like alarms or alerting signals in the user's environment by the feedback reduction systems. Hence, a great care was taken for the system not attenuate to any sinusoidal ambient signals by mistake, as these signals are required to alert the users in the ambient places and maintain environmental awareness.

The main intention of this paper is to identify an efficient algorithm, that will detect and estimate the feedback path and provide a real ear acoustic characteristic. A novel approach of integrating Fuzzy interactive controller (FIC) with Least Squares Delay (LSD) based adaptive feedback cancellation algorithm is proposed to increase the maximum stable gain (MSG) of the system and to achieve required compensation for the feedback effect.

## 3. ADAPTIVE FEEDBACK CANCELLATION

The signal processing of a hearing aid consists of a forward path $G(z)$ and signal amplification components including ADC and DAC. The input to the hearing aid, is the desired signal $s(n)$, processed through microphone, amplified by $G(z)$ and obtained as loudspeaker signal, $y(n)$.

This $y(n)$ is fed back to the microphone, forming a feedback path $F(z)$ as shown in the block diagram of adaptive feedback cancellation system (Figure 3). The elements in feedback path tends to introduce non-linearities that will affect the hearing aid system due to the following reasons: (i) increasing the input level to maximum, raises the quantization noise and (ii) reducing the output to maximum level compromise the benefits of the hearing aid, however the feedback path cannot be predicted with certainty (Freedb 2008).

In the initial step of cancelling the acoustic feedback, an adaptive FIR filter $H(z)$ is introduced and placed in parallel to $G(z)$, to estimate the feedback path $F(z)$. Generally, the adaptive algorithms estimate the transfer function of feedback path, which is then subtracted from the microphone signal, only if the feedback path is linear. To estimate the feedback signal $y_f(n)$, an adaptive estimate algorithm $W(z)$ is implemented in $H(z)$, to generate the filter coefficients $w(n)$ (Akhtar and Nishihara 2016).

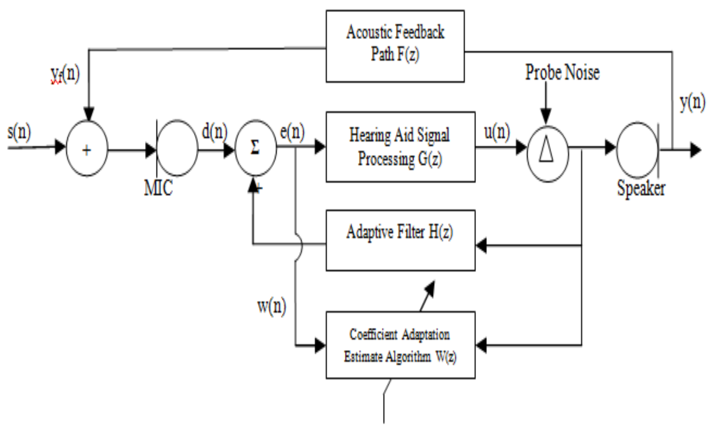


**Figure 3. Block diagram of Adaptive Feedback Canceller**

The signal $y_f(n)$ is due to the loudspeaker output $y(n)$ and $W(z)$ is generated to find the equivalent value of $y_f(n)$, which is then subtracted from $d(n)$, to obtain the desired response.

$$e(n) = d(n) - w^T(n)y(n) \tag{1}$$

In order to achieve, $y(n) = e(n) \approx d(n)$ in eq. (1), the transfer function between $y(n)$ and $s(n)$ of the closed loop system is obtained as:

$$H(z) = \frac{G(z)}{1-G(z)F(Z)} \tag{2}$$

Whenever $G(z)$ is much greater than $F(z)$ and $G(z)F(z)$ becomes 1 at a specific frequency; the hearing system is said to be unstable. The desired signal $s(n)$ and the loudspeaker signal $y(n)$ become correlated, when $s(n)$ is spectrally colored; hence the feedback signal $y_f(n)$ and $s(n)$ cancel each other, the system is said to be biased. The occurrence of biased convergence leads to an oscillation, that would drive the hearing aid at its maximum level and making it unstable.

The resultant biased system is also due to the correlation between the signals $d(n)$ and $u(n)$ with each other (Maxwell and Zurek 1995). The *Wiener* filter technique provides a solution to identify the bias present in the system. And it is a popular method of decorrelating the signals, that finds cross-correlation among the signals to estimate the filter coefficients, which are then computed and copied into feedback algorithm (Kates 1991).

The *Least Mean Squares* (LMS) algorithm is an adaptive filter algorithm, used for filter coefficient calculation, on utilizing different sources of signal. The filter coefficients are updated using LMS algorithm is given in equation (2) as:

$$w(n+1) = w(n) + \mu e(n)y(n) \qquad (3)$$

The coefficients are held fixed after it gets updated, then the hearing system returns to normal operation with the LMS algorithm. However, the rate of convergence of system remain slow.

The NLMS, the *normalized LMS* algorithm, adapts $W(z)$ in order to model $F(z)$, whose weight updation is obtained as:

$$w(n+1) = w(n) + \frac{\mu}{y^T(n)y(n)+\delta} e(n)y(n) \qquad (4)$$

where δ is a small positive constant used in eq. (4) to avoid division by zero complexity. The Figure 4 gives out the error response of an acoustic signal under various filter system. However, the computational burden in Wiener, LMS and NLMS filter algorithms are the major constraints for which the proposed algorithm was introduced.

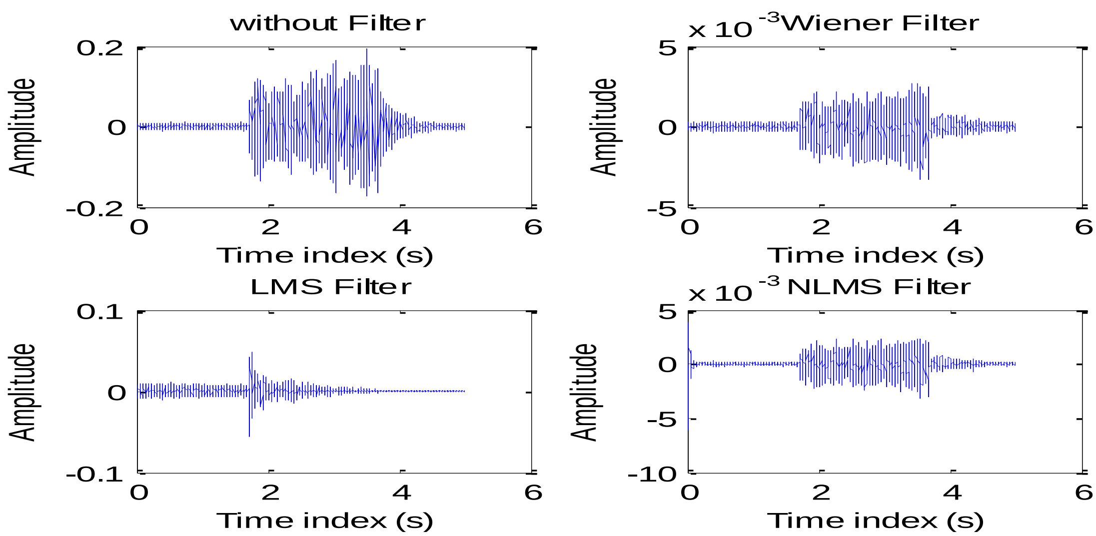


**Figure 4. Error response of an acoustic signal under various filter system**

## 4. METHODOLOGY

An approach of least squares – delay (LSD) based adaptive feedback cancellation algorithm is proposed to increase the stable gain of the system, in which the maximum gain is achieved, without making it unstable (Bustamante et al. 1989). LSD method is performed by injecting a probe signal followed by delay in the feedback path of hearing system.

In LSD analysis, an uncorrelated 'noise' reference signal is generated for adaptation algorithm, which is then added to the receiver. Δ is the delay unit, introduced to de-correlate the input signal $s(n)$ and the feedback signal $y_f(n)$. A probe signal followed by a unit delay to the adaptive filter is integrated to develop the weights transfer, such that the adaptive filter gives a good estimate of $F(z)$. The problem of biased convergence is reduced by choosing the time-varying gain and controlling the level of added probe noise. The gain value is larger in the initial stage for faster convergence, while it is gradually reduced to smaller, to achieve an appreciable steady state SNR. The adaptation process is frozen, once a good solution is obtained.

A pseudorandom random noise burst is injected into the system as a probe signal to adjust the filter coefficients and thereby gives the estimate of feedback path (Freedb 2008, Kates 1990). In the largest sinusoidal component of the microphone signal, whose power is estimated by detecting the feedback, located parallel to the adaptive filter. And the feedback is detected by applying microphone signal as input and the filter coefficients are updated (Kates 1991), based on the decision given by the FIC, which is dealt later.

Whenever the sinusoid in microphone signal is having more power than the preset threshold, the feedback is said to be detected. Once a change in feedback behavior is found, the time variant cancellation filter, updates the estimated feedback path. The impulse Response of sinusoid signal is obtained with respect to its samples and its corresponding magnitude and the phase response are shown in Figure 5 (a) and (b) respectively.

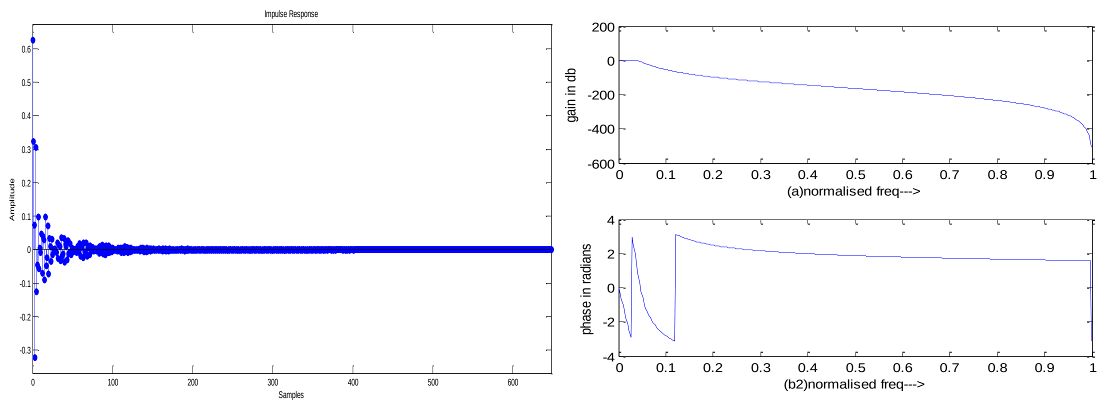


**Figure 5. (a) Impulse Response of actual sinusoid (b) Magnitude and Phase response of given sinusoid**

After the convergence of filter, the delay $\Delta$ was applied before the loudspeaker output, to identify the effective path of $H(z)$, in order to perform de-correlation of $d(n)$ and $u(n)$; and track the convergence-status of $H(z)$ (Levitt et al. 1988). In both the cases, the continuous signal is shown impassive and require a stringent action for implementation.

Lotfi Zadeh (1965) developed the principles of fuzzy logic and the Fuzzy sets, on which the proposed approach of Fuzzy interactive controller (FIC) was developed. The FIC consists of three main processors, namely: *Fuzzifier* for fuzzification of input variables, the *Rule-Decision-Table* for fuzzy inference and a Centroid *De-fuzzifier* for defuzzification of desired response. The input variables were mapped for the weight adaptation of LSD system, in order to handle the linguistic concepts under weighted summation (Özen 2010). However, FIC could adjust the system input by observing the outputs from the history of the information provided by the filter system.

## 5. SIMULATION

The stable gain was obtained from the integrated approach by evaluating the different audio signals (speech and sound signals) with a real-time impulse response. The system performance was simulated for four set of real-time sinusoidal signals: Ambulance sound, Telephone sound, speech segment of a female voice ('Hello how are you'), recorded in a sound-proof room and Babble sound, acquired at various environments, whose time domain representations are shown in Figure 6. These signals were considered to be important for a hearing-aid user at different stages, hence well thought-out and carefully measured for real-time implementation.

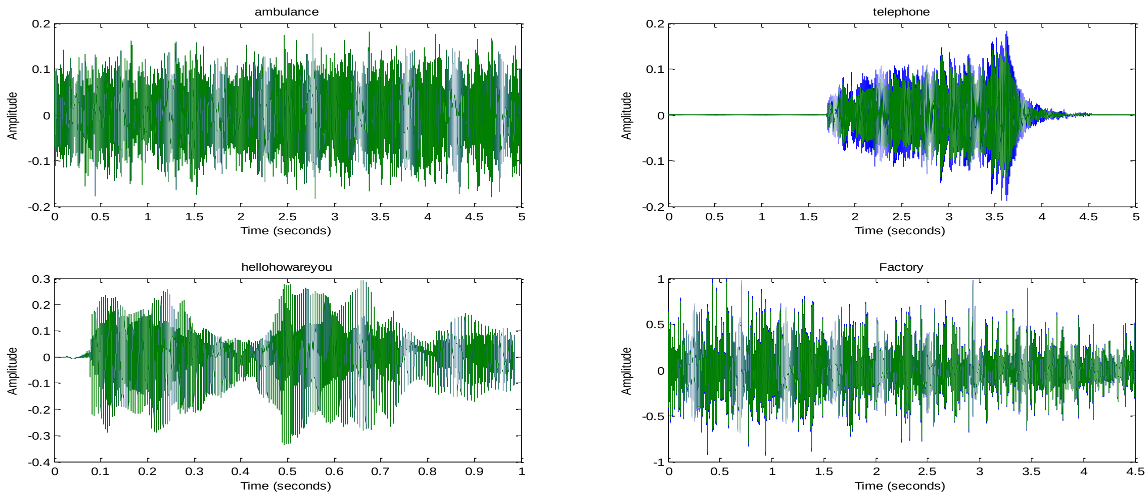


**Figure 6. Time domain response of ambient signals at varying environmental conditions**

The audio wave files were loaded in MATLAB file and a set of selected signals were measured for its magnitude with respect to its samples per second. The test files were converted into .wav format and trimmed accurately for 5 seconds and the bit-rate was of 16 bits/sec. The input sinusoidal signal with amplitude of 0.7 which was added with a random noise of amplitude 0.08 of each part of the microphone signal. The obtained mixed signal gives the actual value of sinusoid, before the application of adaptive LSD algorithm.

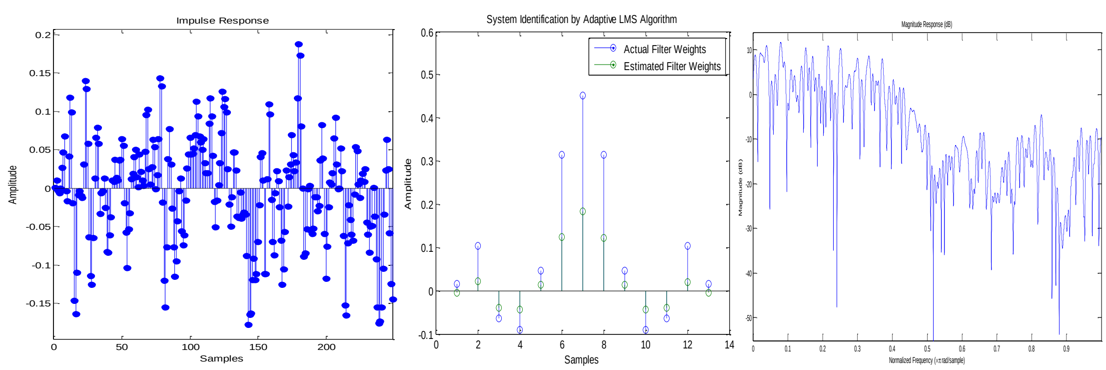


**Figure 7. Characteristics of Feedback signal (a) Impulse Response (b) Actual and Estimated Filter weights (c) Magnitude response**

The impulse response of feedback path was measured and re-sampled for the length of 1000 samples and the sampling rate is chosen to be 16KHz. It was represented as finite impulse response of both microphone and loudspeaker signals. Therefore, the filter gain could exceed 0 dB (Freedb 2008). The characteristics of feedback on an ambient signal was represented with respect to its impulse Response (Figure 7 (a)), their actual and estimated filter weights (Figure 7 (b)) and its magnitude response (Figure 7 (c)).

The Fuzzy interactive controller was implemented by applying the acoustic input signals to *fuzzifier*, to individually attain their membership functions for fuzzification process. The Rule-

Decision-Table of Fuzzy controller provided the *four* levels (gradual, short, quiet and continuous) in terms of linguistic variables as subsets for each input and the resultant values represented the control outputs in terms of crisp values (Özen 2010). The four 4 levels of probe signals considered for analysis were under 'gradual' level (Ambulance sound), 'short' (Telephone sound), 'quiet' (speech segment) and 'continuous' (Factory sound).

The fuzzy variables represented the mean error, determined from the output control signals as shown in Figure 8 (a) and the frequency response obtained from the values of linguistic subsets provided the uniform distribution of gain and phase values among the output variables of the system as shown in Figure 8 (b).

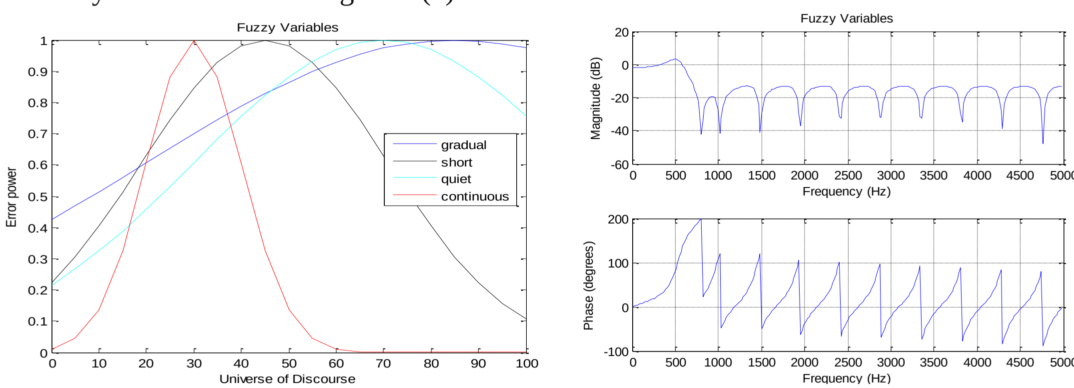


**Figure 8. (a) Membership functions using FIC (b) Frequency response of FIC**

The stable gain estimate was obtained by inserting the delay T for 1 to 5 seconds. The algorithm run on the given signal to obtain the feedback-compensated signal. When the magnitude of feedback signals $y_f(n)$ is closer to $s(n)$, the feedback cancellation system immediately reacted to the sinusoid ambient signal. This leads to the re-estimation of feedback path $F(z)$, for the ambient signal to reappear at its proper intensity. Then the feedback signal estimate $y_f(n)$ was subtracted from $d(n)$, to compute the filter coefficients using least squares – delay (LSD) algorithm.

The graphical representation of magnitude and phase response of an ambient signal (here 'gradual' level was considered), were shown in Figure 9. Similar charts were prepared for the remaining set of signals by calculating the frequency components of signals and their impulse responses after applying LSD were found, which could be compared constructively.

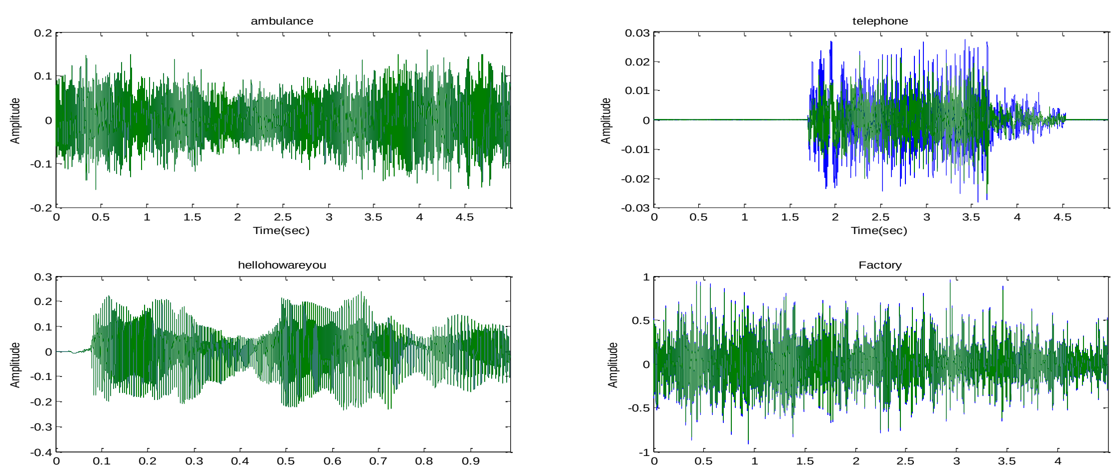


**Figure 9. Time domain response of an ambient signals after applying LSD algorithm**

During the onset of oscillation, the characteristics of ambient signal such as the impulse Response, the actual and estimated Filter weights and their corresponding magnitude response were obtained. The probe noise signal was injected, whenever it is required to adjust the set of filter coefficients. The stable gain was computed from obtained error signal $e(n)$ using $s(n)$ as reference signal.

### 5.1. Measuring SNR

Then the SNR estimate was obtained by subtracting the gain value obtained due to unfiltered AFC, from that of filtered AFC algorithm and those values were recorded. Whenever the obtained SNR value is greater than or equal to T, a small increment Δ, say 0.1 was added gradually. The same procedure was repeated for number of iterations; and when it was found less than T, the previous Δ value was used to estimate the SNR.

The SNR ratio exemplifies the noise control efficiency of the hearing aid. In objective assessment, higher the SNR, lower the mixed signal noise and better the noise reduction performance of hearing aid. Calculating and computing the SNR of output signals of hearing aid were tabulated and the rise in the SNR indicates that the hearing aid reduced the noise (Zhang et al. 2015).

$$SNR = 10log_{10}\frac{\sum_{n=0}^{N-1} s^2(n)}{\sum_{n=0}^{N-1}[\hat{s}(n)-s(n)]^2} \quad (5)$$

Where, $s^2(n)$ is the power of pure tone and $\hat{s}^2(n)$ is the power of output signal of hearing aid in eq. (5). The calculations in Table 1 are acquired when the pure tones are mixed with noises in Noise X-92 (Zhang et al. 2015). The SNR value was intended to provide 22 dB for a 12-tap adaptive filter in a sound-proof room.

**Table 1. Estimation of SNR ratio**

| Pure tones are mixed with noises | LMS | NLMS | LSD |
|---|---|---|---|
| Ambulance sound | 19.0355 | 19.7654 | 19.5243 |
| Telephone sound | 20.5528 | 21.9399 | 21.8104 |
| Babble sound | 17.2441 | 17.6322 | 17.6567 |
| In a sound-proof room | 21.8732 | 21.9494 | 22.0157 |

The time delay occurred between hearing aid output and input signal, will impact the calculation of signal-to-noise (SNR) ratio. Therefore, the shift in time is calculated and synchronized prior to study.

The process was terminated and the steps were repeated for both the cases with unfiltered and filtered signals to obtain the stable gain of same audio file. However, a major variation in their magnitude response in time domain characteristics after applying LSD was obtained, whilst the continuous signal shown unaffected. The above steps were repeated for multiple audio files and the resultant SNR values were averaged over to obtain the final stable gain estimate.

Sound files of 1 kHz to 8 kHz ‘gradual’ level sinusoid was computed, sampled at 16 kHz of order 64 by adding white noise with variance 0.001. When the input frequency range was of 1000 Hz to 7999Hz, the system tended to provide a stable gain for N at 64. And it was observed that exactly after 8kHz, the gain was getting reduced, and started providing negative values.

The simulations proved that the Least Squares delay (LSD) based algorithm outperformed for the highly time-varying sound signal such as speech and music and the mismatch between $\hat{F}[z]$ and $F[z]$ decreased, by increasing the forward path gain. The Figure 10 shows the frequency response of ambient signals before and after applying FIC integrated LSD adaptive algorithm. Also, the error signal $e(n)$ is shown to approach $x(n)$, which relatively increase the sound quality.

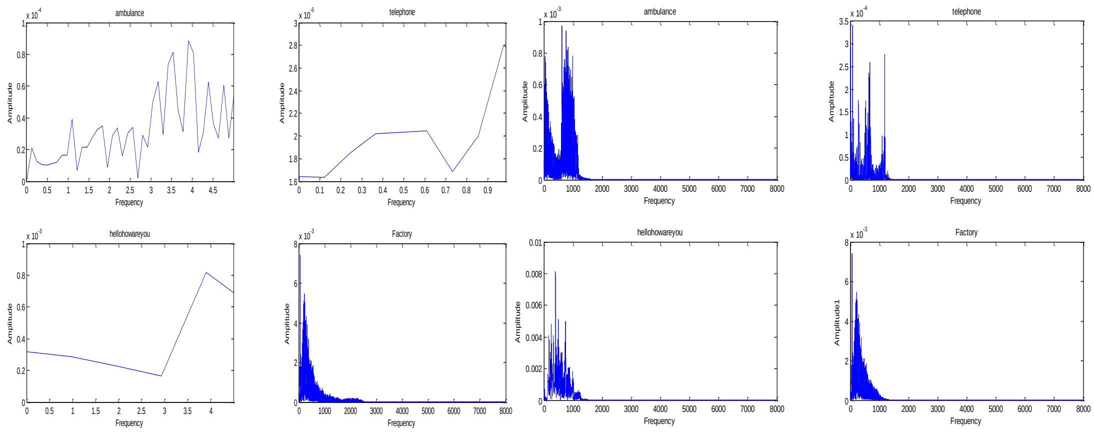


**Figure 10. Frequency response of ambient signals (a) before LSD adaptive algorithm (b) after LSD adaptive algorithm**

Although, the amplifier gain was unstable during the presence of feedback, the receiver output of LSD algorithm was same as that of LMS or wiener filter; hence the proposed algorithm suffered from system bias, due to high correlation between desired and feedback signal in the hearing system (Kates 1991), (Freedb 2008). However, the algorithm was able to track the given input sinusoid of about 1kHz to 8kHz for 1000 samples (50 ms), whose SNR value exceeds 50dB SPL, during which the normal hearing aid processing gets disengaged. However, it was observed that there was trade-off between the convergence rate and the computational complexity, although, the mean square error value was low at the initial stage of training, the steady state error was gradually become very less (≈ 0) after adaptation.

## 6. CONCLUSIONS

In this paper, the proposed method was intended to obtain the feedback signal estimates, neither by interrupting normal speech nor by affecting speech intelligibility of signal. The feedback path was estimated and their experimental results were measured over time and compared with standard filters such as Wiener, LMS and NLMS algorithms. Among all, the Least Squares Delay (LSD) based algorithm showed improvement in SNR value after an extra few iteration carried out. These results could be improved further for a greater number of iterations, that reduces the convergence rate. The results obtained from the Fuzzy interactive controller (FIC) incorporated with LSD algorithm proved that the proposed method would provide a significant cancellation of feedback in varying environmental conditions. However, the variability based on same subject, same ear mold and in different acoustic environments were to be deliberated for much more closer gain values.

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