Cancelling Echoes and Improving Rectal of Telecommunication by using an AFA

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ABSTRACT
In this paper, we present a new approach to cancelling echoes mostly occurrence in today’s telecommunication system due to acoustic coupling between loudspeaker and a microphone. The proposed algorithm is a VSS-NLMS algorithm the popularity of this algorithm due to the solve a conflicting requirement of fast convergence and low misadjustment. Most of the algorithm was design under-modeling scenario; the proposed algorithm doesn’t require any prior information about acoustic environment. Due to the specific characteristic of this algorithm are equipped with good robustness feature against the near end signal variation and has a low computational complexity and low level data storage. So it’s a reliable candidate for real world application.

KEYWORDS
Acoustic Echo Cancellation, Adaptive filtering Algorithm(AFA), UMSI (Under modeling System Identification), NLMS, Variable Step Size NLMS algorithm.

1. INTRODUCTION
Adaptive filter have been successfully applied in a divers field including a communication, control and system identification [1] and is still in the field of the research reason is that it can be satisfactorily applied unknown environment. [2]. the context of acoustic echo cancellation adaptive filter will design a finite impulse response to estimate a echo path between the terminal’s loudspeaker and microphone. The most popular adaptive algorithm is a NLMS algorithm frequently involved in the context of AEC, but NLMS suffer from a slow convergence speed so that it cannot optimized both conflicting requirement like a convergence speed and low misadjustment. To solve this problem number of VSS-NLMS algorithm have been developed. The very famous VSS-NLMS algorithm was derived under modeling scenario is known as a VSS-NLMS-UM case where the length of adaptive filter is less than the length of echo path [3]. It was designed only for a single talk case but in AEC double talk happen frequently. If adaption is not halted during double talk case algorithm will be diverse. The objective of this paper is to compare the VSS-NLMS algorithm with various adaptive algorithms in the context of echo cancellation.

2. CURRENT STATUS
The various type of adaptive filter algorithm(AFA) has been proposed to cancelling echoes in telecommunication system listed below.

1. Normalized least means square algorithm.
2. Variable step size NLMS Algorithms.
4. Double talk robust VSS-NLMS algorithm.
5. LIME approach.

3. BASIC PRINCIPAL OF ECHO
Basic definition of the term using in this paper.
AEC - Device used to cancel to echoes that can be automatically adapt changing the echo environment.
AAEC- Acoustic Adaptive Echo Cancellation. Acoustic echo canceller applied to an acoustic environmental (Loudspeaker to microphone).
NAEC – Network AEC Acoustic echo canceller applied network environment (2-wire 4-wire hybrid).
ERLE - Echo Return Line Enhancement Amount of echo cancelled by AEC. Echo canceller is a one of the most widely device used in a digital device used in computer because each telephone call required a couples of canceller. Transversal filter which is adaptive modeling a echo path impulse response, generate a estimate of echo, with this an echo canceller is created and give a right time to cancel actual echo. Diagram show in below [4].

Algorithm were developed in the under modeling situation due to the fact that echo path highly time variant (movement means people moving around in the room). The residual echoes caused by the part of the system that cannot be modeled
interpret as a additional noise. But it can be remove nonlinear processor is used to removing a echoes by injecting a comfort noise (also known a pink noise). A low level of comfort noise is usually added during a silence periods as an audio psychological comfort effect for the listener.

3. ADAPTIVE TRANSVERSAL FILTER

Adaptive transversal filter is an FIR filter used in acoustic echo cancellation due to,

1. FIR filter is inherently stable.
2. The length of adaptive filter depends on application. Network echoes required typically short length and echoes required mostly long length.
3. Length affects the performance of the algorithm such as a convergence speed, computational complexity and residual error.

Adaptive transversal filter can be classified in two ways and length selection of adaptive filter is most important in the context of the echo cancellation.

1. Short length filter.
2. Long length filter.

**Short length filter**
- 125 to 256 tap or 16 – 32 ms if data is sampled at 8 kHz which is typical for voice.
- Fast convergence but final solution has more residual error since the true response is IIR.
- Less complexity since algorithm complexity depend on the order of the filter.

**Large length filter**
- 512 to 1025 tap or 64-128 ms.
- Slower convergence but final solution has less error.
- Computational complexity has been increased because order o algorithm is N^2.

Literature Review – In today’s telecommunication most of the telephone required couples of the echo canceller but rivalry growing day by day so it is very importance to improve communication quality. Various type of the adaptive algorithm have been developed like a normalized least means square algorithm (NLMS), Variable step size NLMS algorithm (NPVSS-NLMS, VSS-NLMS-UM ) according to the improvement of performance.

The most popular adaptive algorithm is a NLMS algorithm in the context of acoustic echo cancellation. But main disadvantage in NLMS algorithm is that it suffers from a low convergence speed. The performance of the algorithm is given by in terms of the Convergence rate maladjustment and stability is governed by a step size parameter is given by [1],

\[ 0 < \mu < 2 \]

The above proposed algorithm is not able overcome both conflicting requirement like a high speed of convergence and low maladjustment, so it is not real candidate for AEC. Above problem may not be completely overcome because most of the adaptive algorithm tune by adaptive filter parameter like a \[ d(n) \], \[ e(n) \] and \[ x(n) \] but improve the performance of the algorithm.

The new proposed algorithm is a Nonparametric VSS-NLMS algorithm [5]. It is give better performance in the context of acoustic echo cancellation, step size parameter is given by

\[
\mu_{NPVSS}(n) = \frac{1}{\delta + x_t^2(n)\hat{\mu}(n)} \left[ 1 - \frac{\sigma}{\delta + \hat{e}(n)} \right]
\]

\[ = \mu_{NLMS}(n) \alpha(n) \]

Where \( \alpha(n) \) in the range from 0 to 1 and is known as a normalized step size.

Weight updating equation is given by,

\[ h(n) = h(n-1) + \mu_{NLMS}(n) \alpha(n) \]

We notice that before convergence from equation (1), if \( \sigma_t(n) \) is greater than the \( \sigma_t(n) \), the step size of the NLMS algorithm is equal to the step size of a new proposed algorithm. From equation (2) if \( \alpha \) is replaced by \( \alpha_{sm}(n) \) then NPVSS-NLMS algorithm is known as a set membership (SM) NLMS algorithm [6].

\[
\alpha_{sm}(n) = \left\{ \begin{array}{ll}
1 - \frac{\gamma}{|e(n)|} & , \text{if } e(n) > |\gamma| \\
0 & , \text{Otherwise}
\end{array} \right.
\]

\( \gamma \) being bound with the noise, there is no averaging on \( |e(n)| \). It is clear that SM-NLMS algorithm is not better as compare to the NPVSS-NLMS algorithm but it improve the performance as compared to the NLMS algorithm. In simulation use length of acoustic impulse response and length of adaptive filter are same like a 512. The sampling rate is 8 KHZ. and the input signal is a white Gaussian noise. The independent white Gaussian noise added with the output at 30 dB, we also assume that noise power is given. Theorically, we calculate a misalignment will give below,

\[ MA (dB) = 20 \log_{10} ( ||h - h(n)||_2 / ||h||_2 ) \]

Now we will compare the performance of the NLMS algorithm with two different step size with NPVSS-NLMS algorithm and MS-VSS algorithm. The performance of the proposed algorithm is better than the NLMS and MS-VSS algorithm in terms of both conflicting requirement.

But NPVSS-NLMS is apply if we have knowledge about noise power but most of the adaptive algorithm designed in the under
modeling case. It doesn’t require any priori information about acoustic environment. The proposed a new algorithm is known as a VSS-NLMS-U [7]. Step size of this algorithm is given by,

$$
\mu(n) = \frac{1}{\sum_{i=1}^{L} \gamma_i(n) x_i(n)} \left[ 1 - \frac{1}{\sum_{i=1}^{L} \gamma_i(n) x_i(n)} \right] \ldots (3)
$$

The VSS-NLMS –UM use only the parameters that are available from the adaptive filter i.e. d(n), y'(n), e(n) and all the information concerning the change in the acoustic environment. Experiment result show, the acoustic echo path measured using 8 kHz sampling rate and also consider a length of echo path N = 1000 and length of adaptive filter is 512. The input signal x(n) is either white Gaussian noise or speech signal. Independent white Gaussian noise added with the output of echo path with 20db SNR, and also assume that noise power available for NPVSS algorithm. Theoretically misalignment defined by,

$$
\text{MA (dB)} = 20 \log_{10} \left( \frac{\|h - h^T(n) O^T N_{L}(n) \|}{\|h\|} \right)
$$

Now comparing the performance of the NLMS algorithm with two different step sizes with the NP-VSS and VSS-NLMS-UM algorithm. Initial convergence rate same to the entire algorithm, but we know that echo path is a time variant. The misalignment of VSS-NLMS-UM is lower as compare to the NPVSS algorithm. We found one important thing is that if we increase step size of NLMS algorithm up to few iteration and after that step size will reduced then we see NLMS algorithm have a good performance but echo path will change it will react a slower.

Another situation is that if ambient noise increase but our case we derived algorithm in salient period then new estimation of noise power will not available for NP-VSS algorithm. Therefore proposed algorithm is tactless about these changes.

### 4. CHALLENGES

In telecommunication to cancelling a choose have a more challenge given below,

- Acoustic echo impulse response are on the spick and span of 125 ms.
- Long filter are needed approximately 1000 tape which increases complexity and convergence time.
- Near end noise signal.
- Near end noise will corrupt the echo, decreasing the cancelling ability.
- Near end speech is especially difficult from of near end noise.
- Acoustic echo path can change rapidly.
- More difficult to acoustic echo to remain converged.
- None liner echo component.
- Introduced by speaker driven beyond linear region.
- Longer delay.
- Longer delay required more echo cancellation since the ear is more sensitive to longer delay.
- More difficult to find beginning of echoes.

### CONCLUSION

In the NLMS algorithm, we need to find a negotiation between both the conflicting requirements especially in the case of acoustic echo cancellation. But this compromise may not be possible in most of the cases, so we have proposed a VSS-NLMS algorithm little bit improve the performance of the system. So further improvement of the performance of the system we have a proposed a new NP-VSS-NLMS algorithm in the context of acoustic echo cancellation but some issues are in this case the required a signal noise power estimate and misalignment was not improved. This is overcome by using a new proposed algorithm with modified under-modeling VSS-NLMS algorithm. That algorithm will be giving the better performance as compare to the other algorithm in the form convergence rate and maladjustments and obviously suitable for a real world application.

### FUTURE SCOPE

We have reviews the various papers in the context of acoustic echo cancellation but the better results found in VSSNLMS –UM algorithm but in the case of under-modeling MSE is too large that was big challenge. So have modified this algorithm and give better results in the context of acoustic echo cancellation and improving the rectal of telecommunication and better compatible with 4-G.

### REFERENCES


