Analysis of echo cancellation using adaptive system identification

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Abstract - **Adaptive filters on the VLSI implementation can be a long array which usually implies in echo cancellation. Such an array of filters results in more resources and high-power consumption. To minimize the adaptation filtering algorithms, geared are used to improve efficient implementation for echo cancellation application. This paper discusses about the application of adaptive normalized least-mean squares (NLMS) algorithm to reduce the amount of computations in echo cancellation. The algorithm is implemented in a commercial acoustic echo canceller to analyze its fast convergence.**

***Keywords – Adaptive filter, NLMS, hybrid* system, PSTN, PO2, LMS.**

I. INTRODUCTION

An acoustic echo canceller is designed to overcome the acoustic feedback that interferes with teleconferencing and hands-free telecommunication. An adaptive filter identifies the difference in transfer function between a loudspeaker and a microphone and produces an echo replica where the real echo with an attached noise subtracts the replica. Various adaptive algorithms are applicable to an acoustic echo canceller.

NLMS algorithm is one of the most common adaptive filtering algorithms used in echo cancellation, as it uses computationally-efficient techniques for echo cancellation [1].

Error using power-of two (PO2) quantization [2] and Signed-Error (SE) are found in this project. Power –of-two (PO2) error is applied into the NLMS to reduce multiplication by introducing a shift operation, and hence the quantity of computations is decreased [2].

The quantization done is a nonlinear operation and the output is defined in as a binary word using a single “1” bit. In Signed-Error, the sign of the error signal is only kept. It is not robust and is known to diverge when a power-of-two step-size does a shift operation instead of a multiplication. Through this, controlled noise or signal can be reduced, resulting in an improvement in robustness, in the form of error.

II. ECHO CANCELLATION BACKGROUND

Echo cancellation represents one of the most challenging system identification problems. The most used adaptive filter in this application is the popular normalized least mean square (NLMS) algorithm, which must address the classical compromise between fast convergence and low mis adjustment. To meet these conflicting requirements, the step-size of this algorithm needs to be controlled. Echoes which usually occur over telephone calls over the public switched telephone network (PSTN) are basically due to two users connected through a two-wire line to the both phones on their respective local central office and two separate simplex that make a four-wire inter-office link, as shown in Fig. 1.

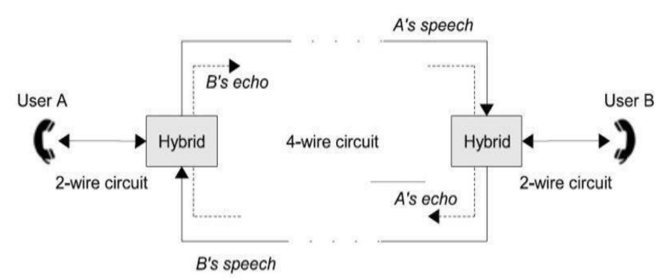

Fig. 1 Network echoes over the PSTN

A hybrid is supposed to transfer all incoming energy signal from the four-wire circuit to the two-wire circuit. However, due to unstable impedance matching, some of the energy is transmitted again to the four-wire branch as a result echo is created. Thus, the echoes in the PSTN arise from the mismatch hybrid devices.

Nowadays, many popular applications like a mobile phone and teleconferencing systems require a loudspeaker-enclosure microphone or Hands-free audio terminal. But the issue during the uses of these devices is the acoustic coupling between the loudspeaker and the microphone.

Due to this coupling, the microphone picks up signal from the loudspeaker as well as signal reflections off surrounding objects and boundaries. This is usually known as the acoustic echo. This phenomenon is influenced by the environment’s characteristics [3], it is presented in the Figure 2.

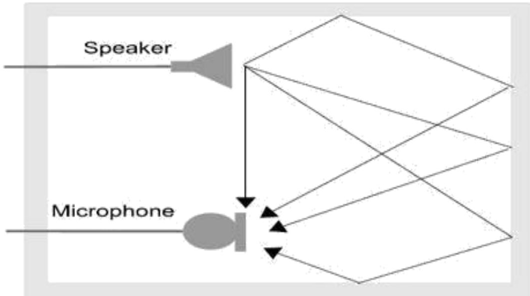

Fig. 2 Acoustic echoes.

To create the synthetic echo, the unknown time varying uses an adaptive filter, in which both driven by the same zero-mean input signal, ***x(n)***. The two systems are assumed to be finite impulse response (FIR) filters of length *L* of the input signal, ***x(n)***, and *v(n)* (the near-end signal) plays the role of the system noise (assumed to be quasi-stationary, zero mean, and independent of ***x(n)*** that corrupts the output of the unknown system. Echo path impulse response is modeled using an adaptive filter.

As the received signal is pushed back again with the adaptive filter outputs by a synthetic echo. By removing the synthetic echo, the original echo is removed before to returning to the transmission end. Most of the times the near-end signal is assumed to be a noise portion. This estimation because of the double-talk detector (DTD), is usually implemented to pause the adaptive filter's adaptation, to avoid diversity, when both received, and near-end signals are present.

III. FORMULATION

LMS algorithm has a fixed step size parameter in each of their iteration which is seen as a down side of the algorithm. Modifying LMS algorithm we find the NLMS, this modification comes with an added advantage of calculating maximum step size value. This step size is directly equal to the inverse of the total expected energy of the instantaneous values of the coefficients of the input vector ***x(n)***. The addition of the outcome energies of the input signal is also equivalent to the dot product of the input vector with itself, and the trace of input vectors auto-correlation matrix, R

$$t_r[R] = \sum_{i=0}^{N-1} E[x^2(n-i)]$$
$$= E[\sum_{i=0}^{N-1} x^2(n-1)]$$

The recursive math metica for the NLMS algorithm is:

$$w(n+1) = w(n) + \frac{1}{x^T(n)x(n)} e(n)X(n)$$

By implementing the NLMS algorithm in MATLAB. The step size function is selected which is based on the input values; through the NLMS algorithm it is seen that stability with unknown signals has increased. This is a product of excellent convergence speed and simplistic computation making the NLMS algorithm ideal for the real time adaptive echo cancellation system.

As the NLMS is an extension of the standard LMS algorithm, the NLMS algorithms practical implementation is very similar to that of the LMS algorithm. Each iteration of the NLMS algorithm requires these steps in the following order.

1. The output of the adaptive filter is calculated-

$$y(n) = \sum_{i=0}^{N-1} w(n)x(n-i) = w^T(n)x(n)$$

2. An error signal is calculated as the difference between the desired signal and the filter output.

$$e(n) = d(n) - y(n)$$

3. The step size value for the input vector is calculated.

$$\mu(n) = \frac{1}{x^T(n)X(n)}$$

4. The filter tap weights are updated in preparation for the next iteration.

$$w(n+1) = w(n) + \mu(n)e(n)x(n)$$

IV. PROBLEM SPECIFICATION

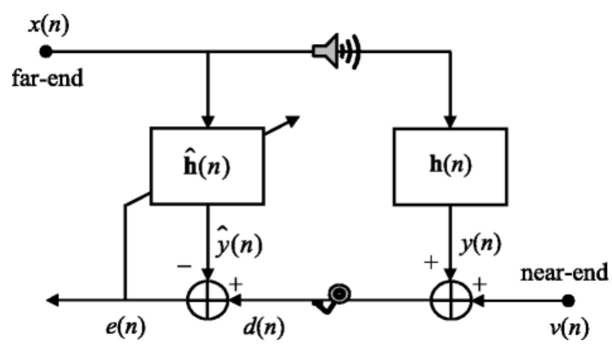

Fig. 3 Echo cancellation system model.

In figure 3 a random signal has been generated or a person is talking which has been denoted as *x (n)*. Here the echo path box is the system with an impulse response to mimic the hybrid subsystem in a network or room which is producing echo. The length of the echo path is varying with a given length hyb2. The signal *x(n)* is going through the adaptive filter and the echo generating path. Figure 1 has another speaker who is not speaking but an ambient noise is getting added from the person's mic. In this project the normalized adaptive filter is taking the 1st speaker's voice and both the echo and noise and after filtering out showing the learning curve of the error. After that, signed error and error with PO2 quantization have been calculated and observed through the responses.

1. Codes for Adaptive system Identification:

```
% ASI.m: a program for Adaptive System
Identification
load hyb2
n=1000000; % no. of sample points
N=98; % filter length
h=zeros(1,N); % initial filter coefficients
%h=[zeros(1,(N/2)-1) .5 .5 zeros(1,(N/2)-1)];
px=zeros(1,N); % dummy vector
mu=0.0002; % convergence factor
x=sqrt(12).*(rand(1,n)-.5); %input as a random
signal
%[x,fs] = audioread('rec_1.wav'); % input as a
voice signal
v =(1/10).*x;%noise
The unknown system
%------------------
%b=fir1(20,.2);
b=hyb2;
N=98;
d=filter(b,1,x);%echo
new=d+v;%echo+noise
[y,h,Er]= ELMSnewQQ(x,new,h,mu,px);
```

2. Codes for Adaptive NLMS algorithm:

```
function [x,h,Er]=ELMSnewQQ(x,d,h,mu,px)
% This function performs adaptive normalized
LMS algorithm
% x--the input signal
% d--the desired response
% h--the coefficients of the adaptive filter
% mu-the convergence parameter

h=h (:);
% px-a dummy vector
N=length(h);
n=length(x);
Er= [];
for k=1: n;
    px=[x(k) px (1: N-1)];
    y(k)=px*h;
    E=(d(k)-y(k));
    ES = sign(E);
    ER=PO2Erroe(E); %Error with power of two
quantizarion.
    Er= [Er E];
    h=h+mu*ES*px'; %Put here, ES for observing
sign error, ER for error with PO2 and E for
error.
end
```

3. Codes for the error with PO2 quantization:

```
function [Q] = PO2Erroe (E)
%UNTITLED3 Summary of this function goes here
%   Detailed explanation goes here
for B=10
    tau=0;
end

if abs(E)>= 1
    Q = sign(E);

elseif power (2, -B +1) < abs(E) && abs(E)<1
    Q = (sign(E) * 2^floor(log2(abs(E))));

else
    Q = tau * sign(E);
end
```

V. RESULTS

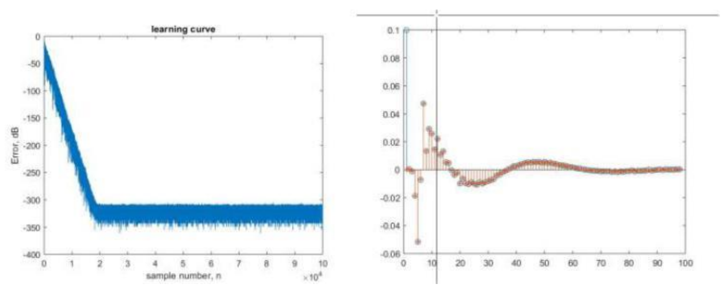

Fig. 4 Error learning curve & Impulse for Random signal with noise.

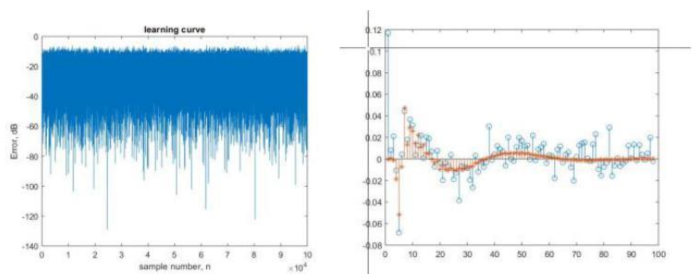

Fig. 5 Sign error learning curve & Impulse for Random signal with noise.

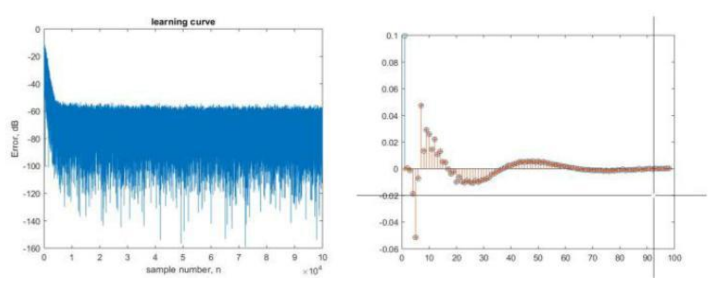

Fig. 6 Po2 error learning curve & Impulse for Random signal with noise.

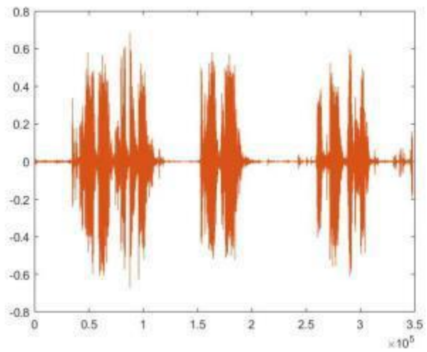

Fig. 7 Generated speech signal.

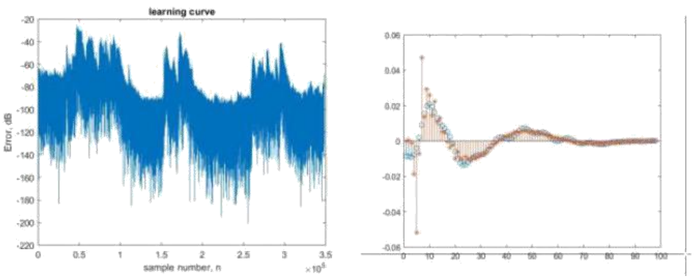

Fig. 9 Error learning curve & Impulse for Speech signal with added noise.

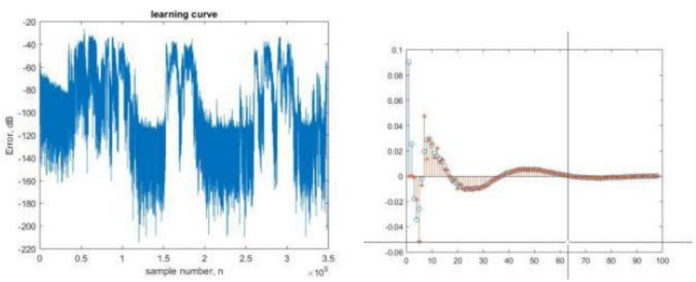

Fig. 10 Sign Error learning curve & Impulse for Speech signal with added noise.

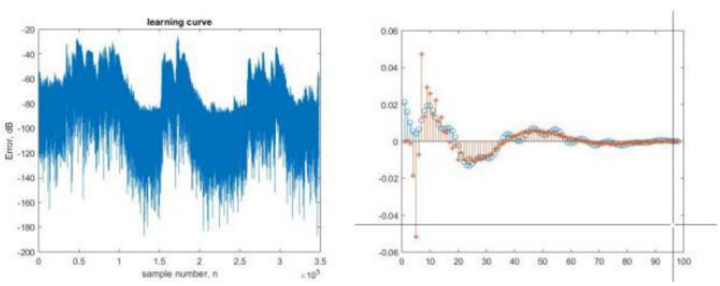

Fig. 11 PO2 Error learning curve & Impulse for Speech signal with added noise.

VII. CONCLUSION

In this project the echo of the system has been removed by using adaptive system identification with normalized least mean square algorithm and found learning curves and impulse responses of error, signed error and PO2 quantization error with a generated random signal. Furthermore, a voice speech signal has been generated and observed the overall responses instead of the random signal which are shown in the results section and the corresponding echo cancellation has been analyzed.

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