**Adaptative Echo Cancellation scheme for hands-free systems based on Fast Block Least Mean Square Algorithm**

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*Abstract*—In this paper an echo cancellation scheme based on Fast Block Least Mean Square (Fast Block LMS) algorithm is implemented. The paper discusses convergence, stability, parameter selection and coefficient tracking of the mentioned echo cancellation scheme. The performance of the designed scheme is evaluated with different in-room movement simulations so as to guarantee the scheme’s functionality and response in diverse acoustic conditions. The performance of the designed scheme is found to be satisfactory and is viable for usage in common rectangular rooms.

Index Terms—Adaptive algorithm, Adaptive Filter, Echo Cancellation, Least Mean Square, Fast Block Least Mean Square

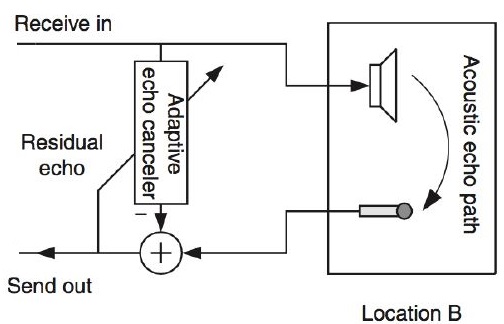
# Introduction

In hands free telecommunication systems talking through a loudspeaker and a receiving microphone, the acoustic echo greatly deteriorates speech quality. This acoustic echo is formed by both the direct path Echo and all secondary path echoes. These secondary path echoes are generated by acoustic wave reflections in different surfaces present inside the room (walls, furniture, people). Moreover, in a real hands free communication situation, furniture can be moved, people can enter the room or even the speaker can wander inside the room making secondary paths variable with respect to time.

In order to solve this problem a system or scheme has to be found that given an input communication signal and an output room filtered signal can estimate main and secondary echo paths in a dynamic or adaptive fashion. In this paper a scheme to solve this pressing problem is proposed and its performance is evaluated taking into account different room impulse simulations.

# Proposed Solution Scheme

A solution for this previously mentioned issue is the implementation of an adaptive echo cancellation scheme so as to suppress the acoustic echo signal’s magnitude. The principle of this scheme is to estimate the impulse response of interfering acoustic paths using an adaptive filter, generating a pseudo echo and substracting it from the original echo.



Let be the transmitted and received signal of a telecommunication system and the same signal filtered by a given room impulse response, in this adaptive scheme an M-order adaptive filter H is obtained so as to obtain an estimation of the room impulse response . This estimated signal is then subtracted it from the original filtered signal so as to obtain an acoustic echo-free signal as an error signal. This scheme’s error signal can be represented as follows:

The scheme’s output form known, what is left to determine are the H adaptive filter coefficients. The proposed scheme utilizes the error signal so as to update the filter parameters in order to improve its performance. There exist many algorithms that can be used to update adaptive filter coefficients, some of them are discussed in the following section, highlighting the algorithm considered to be more suitable related to the desired functionality.

# Adaptive Filter update algorithm

Various adaptive filter update algorithms exist, some of which will presented and discussed in the current section of the paper.

## Least Mean Square Algorithm

Least mean squares (LMS) algorithms are a class of [adaptive filter](https://en.wikipedia.org/wiki/Adaptive_filter)s which update filter coefficients so as to minimize the mean square error between the desired signal and the filter’s output . This algorithm seeks to converge to the same final value as Wiener Filters without having to solve the Wiener-Hopf equations with matrix inversion. This algorithm equation takes the form of:

Where is the k-th adaptive filter coefficient, is the learning rate and is the loss function gradient which is reduced to . Applying this new definition, the algorithm’s update equation resumes to:

Taking into account that this algorithm emerges as a solution for the Wiener-Hopf equation it can be proven that this algorithm’s convergence depends on the relation between the learning rate and the input signal autocorrelation matrix. When is small, this relationship related to filter coefficient convergence can be shown with the following inequality:

Where is the greatest eigenvalue of the input autocorrelation matrix. If the algorithm’s learning rate meets this inequality, algorithm convergence in the mean is guaranteed.

This algorithm’s maladjustment ***M***can be defined as the relation between the stationary minimum loss and the stationary excess mean quadratic error. This can be represented with the following equation:

It can be seen that this algorithm’s stationary quadratic error is related to the filter’s length, the learning rate and the input signal power. Given that in the case of hands free communications, the input signal is human speech and its instantaneous power can vary, the algorithm’s maladjustment can also vary accordingly and that may not be desired.

So as to solve this issue, modifications of the LMS algorithm exist that take into account input signal instantaneous power. One of this modifications is the Normalized Least Mean Square Algorithm that will be explained below.

## Normalized Least Mean Square Algorithm

This LMS variation intends to reduce dependence between input instantaneous power and filter coefficient’s update. So as to solve this issue the Lagrange multiplier method was used resulting in the following coefficient update equation:

It can be seen that in the coefficient update equation input signal power is taken into account. So as to prevent drastic and unexpected changes to occur when input signal power is null or weak another parameter is is added. This parameter ensures that the denominator is never too small, causing undesirable big changes in coefficient’s update.

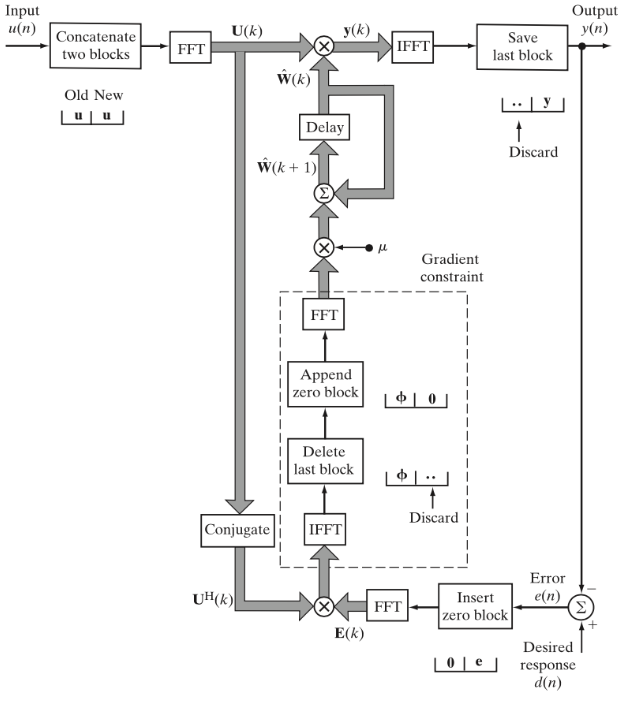
Since input signal power is taken into account for filter coefficient update, it is expected that filter convergence and maladjustment will depend less from this power.

Despite this variation solving the dependence of filter convergence speed and input signal power, this LMS variation does not solve another problem that the algorithm presents; large computational complexity for large adaptive filters. So as to solve this issue another LMS variation is explored: Fast Block LMS with Convergence Optimization, that will be explained and discussed below.

## Fast Block Least Mean Square with Convergence Optimization Algorithm

This basis of this algorithm lies in dividing the input signal in L length blocks and applying the M length adaptive filter to each block. filter coefficients are updated block by block. Relating to computational complexity it can be shown that optimum block length is the adaptive filter’s length M.

So as to improve computational complexity this algorithm uses the Fast Fourier Transform to implement the adaptive filtering equation and the correlation between the input signal and the error signal. This algorithm, can be shown with the following diagram:



This algorithm’s complexity ratio in relation to the basic LMS algorithm can be shown below:

For example, for an M=4000 filter this equation shows that Fast Block LMS is 307 times faster in relation to the normal LMS algorithm.

So as to improve convergence speed, a learning rate can be assigned to each i-th frequency bin of the FFT used.

This parameter can be defined as follows:

Where is a constant and is an estimation of the instantaneous power of the i-th frequency bin. This power can be estimated with an autoregressive equation as follows:

is a forgetting whose range of value lies between 0 and 1. This equation implements a 1 order low-pass filter, where the forgetting factor controls the cut-off frequency of the filter.

This algorithm improves computational complexity in relation to the LMS algorithm while also taking into account input signal power in the coefficient update equation, maintaining convergence speed improvements seen in the NLMS algorithm.

Given the needed application and this algorithm’s benefits, Fast Block Least Mean Squares with Convergence Optimization was chosen.

# Test Conditions and Room Selection

Performance and audio quality of hands free communications should independent of room conditions, allowing flexibility and user comfort. So as to ensure this it would be desirable to find algorithm parameters and test, algorithm’s performance for rooms with different dimensions and impulse responses. Below this selection process is shown.

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## Test Room Selection

Since it is difficult

The intention of this paper is to make an adaptative filter that works on at least 99% of the time, but the key question is how can someone guarantee that is going to work on most cases?

First of all, as we don’t own all conference rooms in the world, we have to generate a way to test our adaptative filter.

To generate a room impulse response the gpuRIR was used (<https://github.com/DavidDiazGuerra/gpuRIR>).

## Selecting the adaptative filter

To determine which adaptative filter is the best, we have to define our inputs and outputs. The signal we send is recorded by a microphone which has several acoustic paths and introduces the echo generated by the room and all kind of things inside it. The signal we receive is the voice of the other person we want to communicate with.

The sound is played by a speaker, which has some non-linearities and can contribute to the input of the microphone.

What we want intuitively is to filter the echoes inside the microphone in order to send my voice to the other person.

Up to now, we have two signals:

-Received signal (voice of the other person)

-Microphone recording

Of those signals we want to obtain our voice, and that is the component that is in microphone recording but not in received signal. We must remember that the microphone also receives the received signal as an input.

-Received signal: voice1

-Microphone recording: voice1+echoesofvoice1+voice2

If somehow, we got the part that has minimum correlation between Received signal and Microphone recording we would obtain voice2. That is assuming blablabla

This leads us to one of the classic adaptative filter schemes which is Adaptative Echo Cancellation.

## Selecting the algorithm to run tests.

As every audio subarray power changes along the complete array, the first naive test was with NLMS algorithm. The results were acceptable but took approximately X seconds to process X so in terms of computational complexity, it was not the best option.

As a result, we looked into another algorithm called Fast Block LMS, which processes the input by blocks and saves time according to the increment of the block length.

# Simulation Implementation and Parameter Selection

# Results and anaylisis

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