EE2 Project - PeaceMaker

Imperial College London

Group 17

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Abstract

The project explores how Active Noise Control or ANC technology cancels noise coming out of a designated area, allowing the area to be used as a "noisy room" where people can converse freely without disturbing others. The project aims to improve productivity by allowing office workers in nearby cubicles to work without being disturbed by people in the "noisy room".

The planned system creates an output wave exactly 180 degrees out of phase with the input wave - this causes cancellation at an error microphone placed close to the listeners. The system achieves this through using a Least-Mean Squares (LMS) filter, a type of adaptive filter that uses feedback to adapt filter weights.

The report discusses the problem of conversational noise in offices and formulates 10 design criteria to solve this. The report details the selection of the concept of the "noisy room" from three different concepts based on the central design criteria. A number of high-level implementation details are discussed – these include the acoustic model, algorithms and hardware designs, component selection and costings. A MATLAB model and simulation is constructed to demonstrate how the concept works. A discussion is then made of further work and how the design may be extended. The report concludes that making a full system will require more time available than provided, but that full development of a product is feasible given enough research of the acoustic environment and careful component placement.

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Contents

1 Introduction

Excessive noise is a problem that disrupts life in the 21st Century. Project peacemaker began with the goal of providing a product that could create a portable area of silence. This can be achieved by using a loudspeaker to generate a wave exactly 180 degrees out of phase with the sound waves of noise, causing cancellation at a desired point. This process is known as Active Noise Cancellation or Active Noise Control (ANC). This technology was proposed to be used as a cheaper and more portable alternative to the existing soundproofing technology - which consists of erecting physical barriers which may be costly or simply impossible in some contexts.

Since then, a refinement of the concept has taken place. The ANC technology has several practical limitations which may prevent it from being implemented in a completely portable solution that would be useful in all circumstances. As a result, the group has decided to downscale the project to focus on a particular application of ANC, and to modify our design to make it more practically reliable.

The focus of the project shall be on small offices, in particular open-plan offices. The need for communication is key to any organisation, but holding meetings in small offices can easily disturb other office workers. The alternative of going to a separate public location can be disruptive and risky if sensitive information is discussed. Constructing a meeting room and using traditional passive soundproofing can be a costly and inflexible solution. This forms the basis of the project, which suggests concepts and means of implementing ANC to achieve a virtual meeting room environment where people outside of the meeting area will be undisturbed by those inside it. This project will propose the use of an adaptive filter - the principles of which will be elaborated in later sections in the report.

s The rest of the report shall discuss how the solution will be implemented, the operating principles and how the project was managed and run.

2 Problem Definition

2.1 The Problem

Control of noise is essential for a variety of reasons. While noise is often merely annoying, there are circumstances where it can become disruptive to daily life and even harmful to human health. The American public health campaign Dangerous Decibels [4](DangerousDecibels,2016) considers noise above 85 decibels harmful for human hearing. Noise level hence becomes an important issue to consider in many circumstances.

One particular group of people that can be affected greatly by noise are office workers, especially in open-plan offices. There exists a disturbing dilemma - while there is the need for communication, telephone conversations and conversations can be some of the most disruptive noise in the office. A psychological study conducted [1](Banbury and Berry, 2005) about office noise suggests that conversational noise from other people's conversations and other people's phone conversations ranks among some of the biggest disturbances to concentration.

This leaves office workers with a few alternatives should they need to communicate and do not wish to disturb the rest of the office:

1. The office can build a full-fledged meeting room. This may be impossible due to cost and more importantly, space, constraints.

2. Workers can go to a public location. This may be inconvenient for the worker, and is an unsuitable environment to disclose sensitive information.

3. Soundproofing can be placed between cubicles. Depending on the type of insulation used, this can be costly and may require the tearing down and rebuilding of partitions and walls. The quality of soundproofing may also not be enough.

2.2 The Proposed Solution

The project proposes to provide an active form of noise control to enable conversations and meetings in designated regions without disturbing other workers - isolating sound from the area from its surroundings. This type of solution provides a few advantages:

1. Active forms of control are dynamic and can adjust to all sorts of amplitudes and frequencies of noise, which passive forms of noise insulation cannot do easily.
2. The solution is reasonably portable and flexible in how it is set up. It can be moved from location to location without having to install permanent barriers, and can be used wherever power is available. It also does not require anything to be torn down.
3. The cost of electronics can be a lot less than that of a full soundproofing solution, as shown in section 8 under Cost Analysis. This means that the solution can be more economical.

3 Management Plan

3.1 Roles and responsibilities

There are seven members in the group in total. The roles and responsibilities of the members are as follows:

- Project Manager - Martin Chan
  The role of the project manager is to ensure that the project progresses according to schedule. He assists and manages the other roles to ensure that any roadblocks in progress are cleared, and provides direction to the group, while ensuring that milestones are met.
- Chief Software Developer - Chao Lyu
  The responsibility of the chief software developer is to develop the software and platforms required for the project, which requires a great deal of programming and digital signal processing. He helps to experimentally determine the means of making theoretical concepts into practical reality.
- Algorithmic Designer - Zihao Wu
  The algorithmic designer researches into the ANC and processing algorithms that will be used , along with their alternatives. He then helps to formulate ways of selecting the best ones, and finds ways to practically implement them.
- Acoustics Designer - Jin Lian
  The acoustic designer investigates the physical acoustical problems that the project faces, and finds ways to work around them to make the project a reality. This knowledge in acoustics is then used to see how the project's work can be extended towards future work for more complicated acoustical environments and problems.
- General Engineering Designer - Sida Niu
  The general engineering designer plays an important role in researching any areas not

currently covered by other roles. These include investigating the kind of noise faced in the problem, as well as determining the amplitude and frequency spectra of noise to be cancelled in the problems.

- Marketing Director - Christodoulos Stylianou
  The marketing director tailors the product is towards customer requirements, and assists in making the end product commercially viable and desirable for the market. He conducts market research to ensure the project meets customer needs and wants.

- Webmaster/Editor - Jason Yuan
  This member has both the webmaster and editor roles. As webmaster he is responsible for creating and maintaining the website, and deals with the programming and layout of the website. Additionally, as editor he is in charge of compilation and editing of the report.

3.2 Work Schedule

Weekly meeting were held every Wednesday from 1pm - 2pm to update all roles on the overall progress of the group. Additionally, these meetings allowed for members to request assistance from other member of the group, and to update members on meetings with supervisors or experts. Finally, weekly tasks and role assignments are then set by the project manager to ensure that work progresses at a good rate.
The project was split up into roughly 3 phases as was mentioned in the interim report. Note that the project often involved iterating between phases in response to developments or roadblocks in the project.

- **Phase 1: Waveform Analysis and Parts Sourcing**
  This phase consisted of researching how sound waveforms might appear to the processors, and to select components suitable for the processing of these soundwaves. This phase involved spectral analysis of noise, followed by development of algorithms and components tailored specifically to handling these waveforms. Furthermore, component selection was also carried out, sourcing parts for testing and prototyping to ensure the total cost of the components used are less than traditional soundproofing, while ensuring adequate performance.

- **Phase 2: Algorithm and Circuit Building**
  The main algorithm, and its alternatives, would be developed and tested in this phase. The group looked at how this algorithm would be applied to the product, and how parameters might be adjusted to ensure optimal performance. In addition, plans to implement the hardware configuration such as power, control circuits and housing were made.

- **Phase 3: Prototyping and Testing**
  Within the time frame given, the task of completing a full prototype with a working ANC system was assessed to be unfeasible; the system is complex to implement and has many different factors to consider. However, foundations for further work to implement the system were laid in this phase. An acoustic model of the system environment was made, and simulations of the system were made in MATLAB to observe the theoretical performance of the system.

4 Design Criteria

10 particular criteria from the Product Design Specification (PDS) are selected to address the above problem. These criteria form the central guidelines for the project. More details on the PDS can be found in appendix A. Some criteria have changed from the Interim Report: performance, life in service, product cost, customer and installation. In addition, ergonomics will replace competition as a key design criteria. This is a result of the focusing our product on a group of customers, and as a result of deeper research into the feasibility of the concept.

1. **Performance**
   The following performance criteria are targeted for the project:

   (a) Research has shown that normal conversation levels range from 25-65 dB[9] (Kuwano, Namba and Okamoto,2004). The target is to reduce the noise to a level 20 dB less than the upper limit, at 45 dB. Attenuation below this threshold will be considered a great success.

   (b) The frequency of human speech can vary a lot, with fundamental frequencies in the region of around 80 to 300 Hz [9] (Kuwano, Namba and Okamoto,2004), and overall in the range between 0 to 5 kHz [8](Kutruff, 2009). While the product has to cancel noise particularly well within the region of 0 to 5 kHz, it should cancel noise in other areas of the spectrum within human hearing.

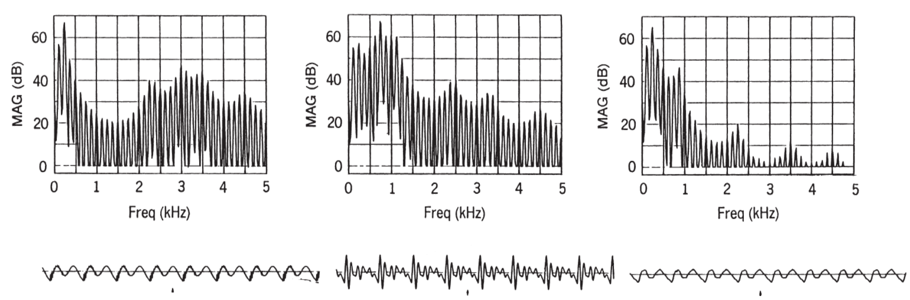


Figure 1: *Amplitude spectrum and time function (sound pressure) of vowels [8](Kutruff, 2009)*

   (c) The processing system must be fast enough to filter the input signal and output a wave that is exactly out of phase at the desired area. A propagation delay can lead to the wave to no longer be exactly 180 degrees out of phase at the desired point.

   (d) The product must be able to run for as long as noise from the noise source needs to be cancelled. This means that the criteria for portability has been relaxed in favour of a solution that can be powered indefinitely, such as through mains.

2. **Environment**
   The product must be able to operate at room temperature in all seasons and in most office environments. Most importantly, the environment should not cause components used in the product to degrade and cause any downgrade in the performance of the product within the targeted life in service.

3. **Life in Service**
   The product must be able to be repeatedly used over a period of time without breaking

down. The target product lifespan is one year on full performance. The device is targeted to run on mains meaning that it can be run indefinitely without charging or replacing of batteries.

4. **Target Product Cost**
   The target market has been shifted towards offices as budget is less of an issue than it is for individuals. The product is aimed at being much cheaper than its physical counterpart without compromising on device performance.

5. **Ergonomics**
   The target audience has little time and little need for complicated controls; a device that provides noise control reliably without the need for the user to spend a lot of time learning how to operate it would be ideal. Operation of the device is aimed at being simple and intuitive.

6. **Customer**
   The target customer are currently office owners who wish to improve the productivity of their employees. Peaceful environments lead to potential increases in revenue and improved working environment. Also, space can be very expensive in cities such as London, leading to insufficient space for meeting rooms. The product aims to construct virtual meeting rooms without the need to erect expensive physical barriers.

7. **Aesthetics, Appearance and Finish**
   This is an important criteria in determining whether or not the product will appeal to offices. The product must be trendy and attractive, and fit in nicely with the office environment. As a result, the end product should be modern, elegant, and should blend in with a variety of indoor environments. Different colours and other forms of customisability can be considered for the final product.

8. **Quality and Reliability**
   Related to life in service and environment is quality and reliability. As customers will rely on the product to carry out activities and socialising without disturbing others, the accuracy and reliability of cancellation is very important. Ideally, the product must be able to reproduce a satisfactory drop in the noise level every time, and cannot break down within the targeted life of service.

9. **Patents, Literature and Product Data**
   ANC is not a new area of research. The group is aware of existing patents such as on "Open-air noise cancellation systems" [10](Nishikawa, 2007) and "Wide area noise cancellation systems"[2] (Christopher A. Brown, 2014) do exist and will avoid infringing on any intellectual property laws. Our product shall be distinguished by portability and cost.

10. **Installation**
    Installation becomes a little more difficult due to the increased complexity of the design. Consequently, set up must be simplified as much as possible. Instruction manuals are provided to make the final product must be intuitive and simple enough for anyone to set up with ease.

5 Adaptive Filtering and Feedback

5.1 Motivations

Before proceeding further, the report shall discuss an important change to the design - the use of adaptive filtering and feedback in control of noise. The previous design used ANC without adapting the controller, also known as a feedforward system. The following diagrams demonstrate how a feedforward system works.

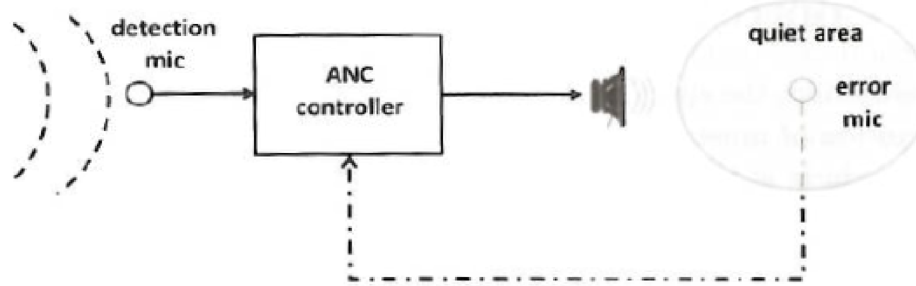


Figure 2: *Feedforward control strategy for active noise control[5](Dimino and Aliabadi, 2015)*

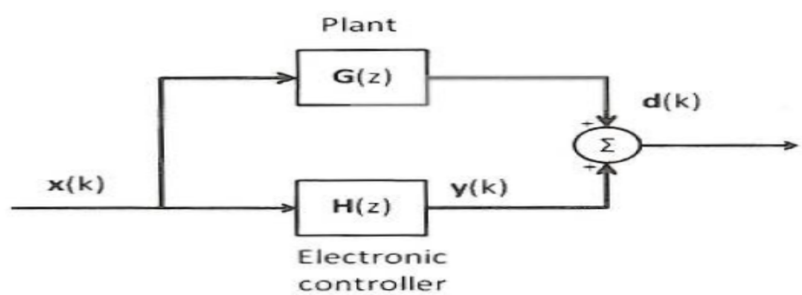


Figure 3: *Block diagram of a general feedforward control [5](Dimino and Aliabadi, 2015)*

Feedforward systems have a number of drawbacks that are described below[5] (Dimino and Aliabadi, 2015) :

- The noise spectrum often consists of many different frequency components that may be shifted in phase differently from each other .The consequence is that the output signal components may not be exactly 180° out of phase with the input. This leads to incomplete cancellation of the signal at some frequencies.
- The filter is very sensitive to changes in the environment.
- The antiphase signal is only effective in cancelling at the loudspeaker. Further from the loudspeaker, the noise may not be fully cancelled and might even exceed the level of the input. While adaptive filters suffer from this same issue at the error microphone, it is easier to adjust the position of the microphone than the position of the loudspeaker.

All of these factors lead to the project using feedback and adaptive filtering over non-adaptive filtering. Feedback allows for the system to cope with time-varying signals and more importantly, changes in the acoustic environment.

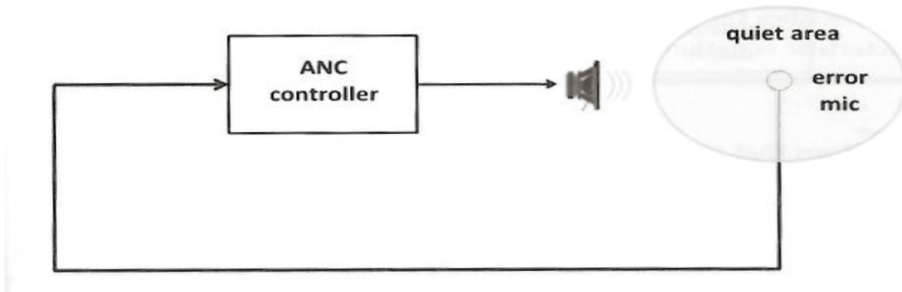


Figure 4: *Feedback control strategy for active noise control [5](Dimino and Aliabadi, 2015)*

5.2 Basic Operation

There are a number of ways that adaptive filtering can be used for ANC. The design used in this project is heavily inspired from ANC systems used in aircraft, and a large part of how the adaptive filter is used to control noise is based on a design featured in Active Control of Aircraft Cabin Noise [5] (Dimino and Aliabadi, 2015).

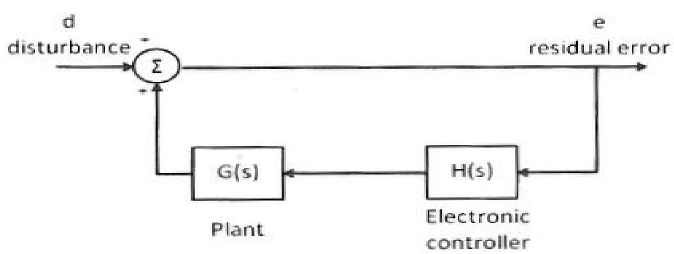


Figure 5: *Block diagram of a general feedback control[5] (Dimino and Aliabadi, 2015)*

The design is a mix of feedforward and feedback designs. One microphone, known as the reference microphone, makes an estimate of the noise. The other microphone, positioned in front of the loudspeaker, is known as the error microphone. The adaptive filter adapts the filter to produce an output at the loudspeaker that will cancel with the original noise to give a zone of silence at the error microphone.
It is clear that the design will require the following:

1. At least one reference microphone and at least one error microphone
2. One loudspeaker
3. One processor to implement the adaptive filter
4. A power supply for the system

6 Concept Selection

Brainstorming yields three concepts that use adaptive filtering to control noise and fulfill the chosen design criteria . All three are discussed and compared in this section.

6.1 The "noisy room"

This proposal suggests the creation of a "noisy room" - an area designated for employees to converse, speak on the phone and hold meetings without disturbing the rest of the office. This area will be outfitted with reference microphones that input noise signals into the filter. The filtered output will then be sent through loudspeakers to surrounding rooms. The adaptive filter will then use input from an error microphone placed close to the listeners to adapt the filter, such that the loudspeaker creates a wave that minimises error at the error microphone. As a result, noise from the room is cancelled and neighbouring workers can work in peace. An assumption made is that sound is significantly attenuated once it reaches beyond neighbouring cubicles, which explains the design choices of only placing error microphones only in these cubicles. However, this assumption means that cubicles that are further away can be disturbed by particularly loud noise, as there is no active control of noise for these cubicles from the noise sources.

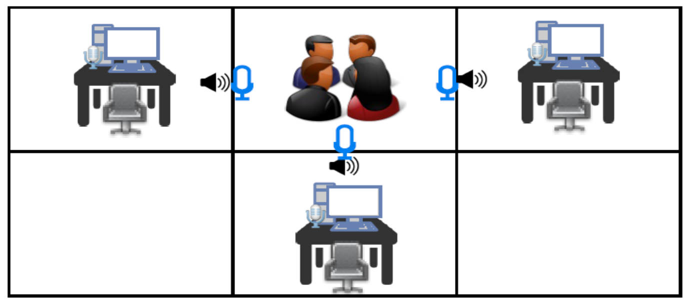


Figure 6: *The Noisy Room*

6.2 Self-Contained Unit

An alternative to the first concept is to attach microphones and loudspeakers to every boundary in the office, and cancel all noise entering into the space. Despite requiring a lot of electronics, this design in theory allows for the office to be divided into many self-contained different areas of isolation that cannot hear each other.

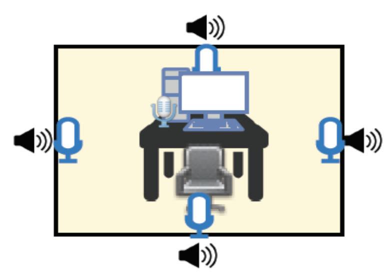


Figure 7: *Self-contained unit*

6.3 Configurable Microphones "Matrix Method"

This concept combines the "noisy room" concept and the self-contained unit concept. Each cubicle is outfitted with a set of microphones and speakers installed on every boundary with another cubicle, as well as an error microphone and a control. The sets of microphones and speakers can be configured at the push of a button to act

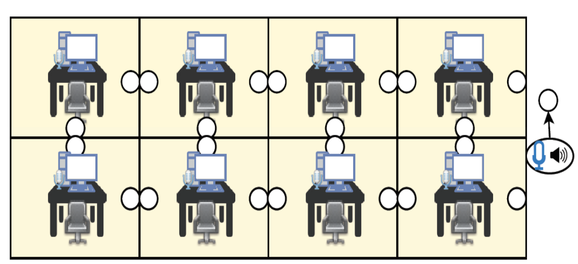


Figure 8: *Configurable Microphones*

either as a reference for noise cancellation as in the “noisy room”, or to act as an ANC device that creates silence in the cubicle.
This grid of microphones and loudspeakers resembles a matrix, giving rise to the concept being dubbed the "Matrix Method".

6.4 Comparison of methods

A comparison table is made comparing the three different concepts:

Criteria	The "Noisy Room"	Self-contained Unit	Matrix Method
Cost	Relatively cheap as only one room is considered.	Considerable expense as every block needs to be outfitted with a new set of microphones, loudspeakers and processing equipment.	Mostly similar to the self-contained unit, as the microphones and loudspeaker configurations will be the same. May require more expensive processing hardware due to computational complexity.
Performance	Works very well for offices where error microphones are installed. System has significantly worse performance for offices where error microphones are not installed.	Works well for cubicles and rooms installed with the system. However, does not work as well for cancelling particular noise sources as the reference microphone is not close to the noise source.	Has the best overall performance compared to the other two methods, due to the reference microphones being close to the noise sources, and the error microphones being close to the intended users. However, this means that different “noisy rooms” can disturb each other, as noise is not controlled in these regions.
Flexibility	The least flexible among the three. The room’s position is fixed until the hardware is reconfigured in a different location.	Very flexible as noise can be theoretically made anywhere without disturbing anyone.	The most flexible as the office can easily be divided into quiet and noisy areas without sacrificing performance.
Design Complexity	The simplest to implement among the three.	Possible with clever arrangement and well written algorithms but very challenging.	Extremely hard to design and implement.

Based on the comparison table, and due to the time constraints of the project, the group has decided on the "noisy room" concept. The concept is the cheapest among the three and is the simplest concept to design and implement. As the performance and flexibility is probably adequate for the purposes of the office, it makes this concept the clear choice among the three.

7 Concept Development

7.1 Noise spectrum of voice

To assess the kind of spectra that may be encountered, a recording was taken in a computer laboratory to simulate the kind of noise in an office environment. A lot of talking and group discussions were observed, alongside people working alone on computers. This approximates an office environment quite well as the main source of noise is from conversations between people. The FFT spectra of the noise is shown below:

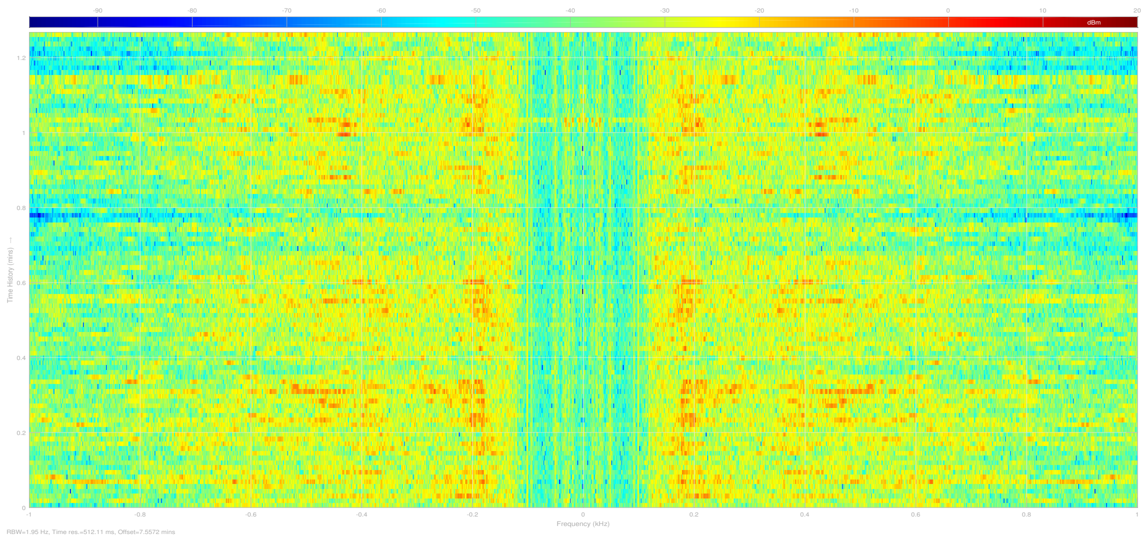


Figure 9: *Nosie Spectrum*

This shows a number of things. The frequency of the noise is measured to be from 100 to 600 Hz. Hence, the microphones and loudspeakers are selected to be sensitive to this range. Similarly, it can be observed that the noise can be time varying and has some nonstationarity. To simplify things, noise is assumed approximately Wide-Sense Stationary (WSS) as time variance is not very great, and step size is assumed to be sufficiently small for approximate stationarity. More advanced models for future work may use methods for tracking nonstationarity of spectra, but this is beyond the planned scope of the project.

7.2 Acoustic Model

Proper implementation will require knowledge of how sound waves propagate in a room. This section shall discuss the acoustic model that has been developed for the project. Much of this section is taken from Room Acoustics, Fifth Edition[8] (Kutruff, 2009).

7.2.1 Spherical Waves

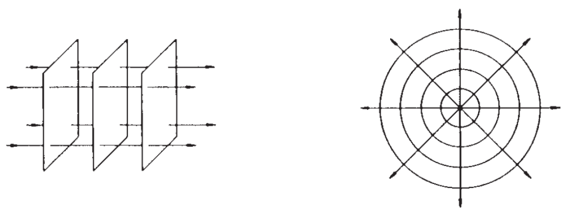
Spherical waves produce pressure according to equation:

$$p(r,t) = \frac{iwp_0}{4\pi r}\hat{Q}\exp\left[i(wt - kr)\right]$$

Where p_0 is static gas density, w is the angular frequency, r represents the distance from sound source, $\hat{Q}$ is the amplitude of the sinusoidal wave and k is the wavenumber. On the other hand, total energy density of the sound wave is given by:

$$w_{total} = \frac{p^2}{p_0 c^2}$$

Where c is the speed of the sound wave. What is observed is that $w \propto p^2 \propto \frac{1}{r^2}$. This shows that energy is inversely proportional to a square of the distance from the sound source, and that the wave's energy decays greatly as far distances from the sound source. This is why the group considers cubicles further away from the noise source as being unaffected by the noise as it will have been greatly attenuated by that point.



(a) *Plane wave* **(b)** *Spherical wave*

Figure 10: *Sound propagation [8] (Kutruff, 2009)*

7.2.2 Standing Waves

The above model does not consider reflected waves. For simplicity, an assumption is made that reflected waves are far from the sound source and can be approximated as a plane wave. Another assumption that will be made is that the sound waves reflect off rigid walls in a rectangular room. A plane wave travelling in the x-direction perpendicular to the reflection surface will reflect off the wall and give a backwards wave. The forward and backward waves will combine to give a standing wave. In reality, acoustic impedances of surfaces will absorb energy and attenuate the waveform, but standing waves mean that the energy of sound at points away from the source will actually be louder than simply considering the contributions solely from non-reflective waves. This is a consideration in offices where there are a lot of reflections are present; the noise becomes louder than originally anticipated.

7.2.3 Eigenfrequencies

A peculiar result derived by Kutruff is that acoustic wave equations only have non-zero solutions that correspond to particular eigenvalues. These eigenvalues in turn correspond to some particular values which can be denoted as eigenfrequencies or resonant frequencies. Theoretically, the acoustical nature of the room and its transfer function can be calculated if these eigenfrequencies are known. Calculation of eigenfrequencies is out of the scope of the group's work at this moment, but calculation of at least some of these frequencies will be made if further work is to be done.

7.3 Microphone placement

Where the microphone is placed relative to users is incredibly important in having a good performance for users. Closer placement of error microphone from listeners leads to better noise cancellation; in some cases, far distances from an error microphone leads to sounds appearing louder. Similarly, the closer the reference microphones to the noise source allows stronger correlation of the input with the noise, allowing greater attenuation of noise at the error microphone. Hence, a balance between the proximity of the microphones from users and the comfort and

convenience must be made.
More information on how the acoustic model may be extended can be found later in the section 10.1 under Further Work and Extensions.

7.4 Detailed High-level Implementation

Below is a diagram of the high-level implementation, demonstrating how the different components are linked up:

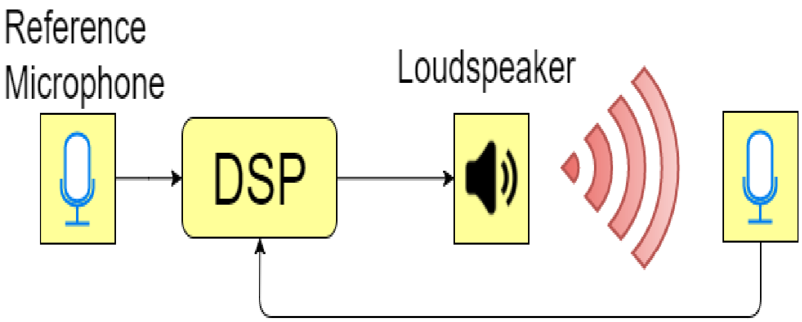


Figure 11: *High-level block diagram*

Each of these systems form a set that will be used for one particular direction. For instance, if the zone of isolation is to be a three-walled cubicle, the system will have 3 separate systems for the three directions.
Due to its complexity, the project uses Digital Signal Processing (DSP) to carry out the adaptive filtering. This requires the signal to be converted into a digital signal by an analogue to digital converter(ADC), processed by the processor and finally outputted through the loudspeaker using a digital to analogue converter(DAC).
The system is not wireless - wires are used to connect all components up. Wired systems generally transmit data quicker and are generally more robust, and are preferred to ensure that delay in the system is reduced as much as possible. In this project's application of DSP phase delays can become very significant, motivating the design to minimise delays as much as possible.

7.5 Components

For each direction, the following components are used:

- One reference and one error microphone. The reference microphone is placed in the "noisy room" while the error microphone is placed close to the listener's head.
- One loudspeaker to provide the anti-phase wave at the error microphone.
- One power supply that converts AC mains electricity into DC power for the device. The choice to use mains is to ensure that the device can run for as long as it needs to without requiring to be charged.
- One processing unit per direction. For processing units, the project plans to use microprocessors purpose-built for DSP. The design choice to use one DSP chip for each direction is to allow for greater ease on the processing and allowing for the DSP's processing power to be focused on one direction.

7.6 Arrangement of Components

Below is a diagram demonstrating an example of how the system can potentially be configured:

The microphones need to be placed close to the user's head without being a hassle to users. This motivates the use of boundary microphones, which are microphones designed to be used on a surface. As most users will probably be working close to walls or partitions, these microphones can be placed on the walls or partitions close to head level.

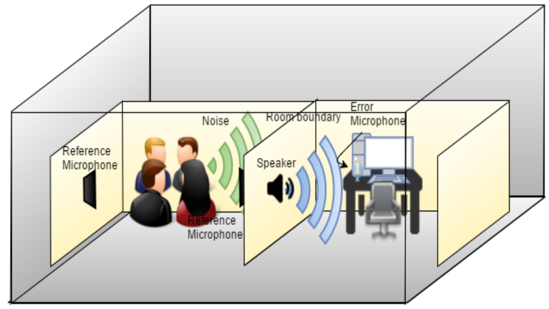
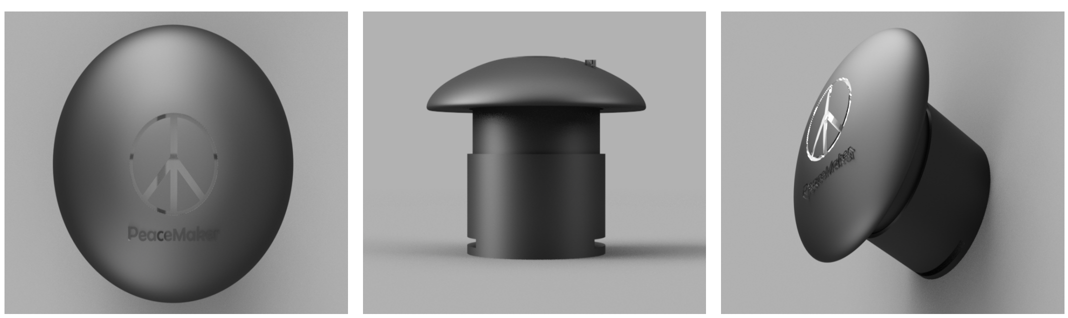


Figure 12: *Provisional arrangement*

- To allow for ease of use, the DSPs and their controls will be centralised in some housing. This housing is designed to be easy to operate, and an easy access point for the DSPs to other components. More details on this can be seen in the housing design section below.
- This arrangement of components is planned to be flexible enough to install in different locations to allow for flexible configurations of the office. Clear documentation is planned to facilitate set up without compromising performance.

7.7 Control Circuit Housing Design

Below are the front and side views of a 3-dimensional render of the control circuit housing design. It acts both as a switch for the circuit and a housing for the DSPs. As can be seen the design is aesthetically pleasing and modern looking; it is designed to be relatively small and no bigger than a desk lamp.



(a) *Top View* **(b)** *Side View* **(c)** *Isometric View*

Figure 13: *Autodesk Render*

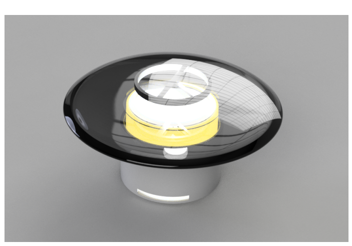


Figure 14: *LED Inside*

The control circuit housing design is designed to hold 3 or more DSP processors as required. The power units and microphones will also feed into this unit. The housing is designed in such a way that the top part can be pushed to turn on and turn off the processing whenever required. This is a simple way of operating the circuit that allows for easy operation of the system. There are plans to install an LED to light up the circuit when it is on to give the user a further visual indication of operation, allowing the user to know if the devices are turned on and off.

7.8 Algorithmic Implementation

This section shall discuss the different algorithms that the DSP controller will use to control the noise. Much of this section has been taken from Adaptive Filter Theory, Fifth Edition [7](Haykin, 2008). Most of the section assumes that filtering is done in discrete time. (More information can be found in Appendix D)

7.8.1 The Least Mean Squares (LMS) Algorithm

The LMS algorithm is a type of finite input response (FIR) filter that works on Wide-Sense Stationary processes. It is based on a type of filter known as a Wiener filter – given a desired response $d(n)$ and actual response $y(n)$, it tries to minimise error $e(n)$ where $e(n) = y(n) - d(n)$. The Wiener filter tries to minimise cost function $J = E[|e(n)|^2]$.
The LMS filter differs in trying to minimise cost function $J = |e(n)|^2$ instead, the instantaneous value rather than expectation value. This makes it more practical for unknown environments like the one faced in the problem. The LMS filter is based on three key equations:

$$y(n) = \hat{w}^H(n)u(n)$$

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

$$\hat{w}(n+1) = \hat{w} + \mu u(n)e^*(n)$$

Where $u(n)$ is the input to the system, and $\hat{w}(n)$ is a filter coefficient which is an estimate of unknown weight vector $w(n)$. The first equation expresses that the LMS filter is a FIR filter – where the output is a weighted sum of several past inputs. The third equation shows how the weights are adjusted – where refers to the step size of the system.

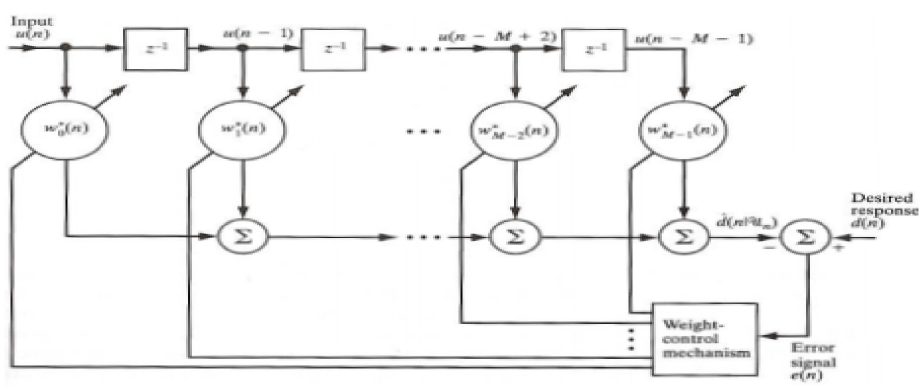


Figure 15: *Structure of an adaptive FIR filter[7] Haykin(2008)*

7.8.2 Stability Analysis and Step-size

It should be noted that there is a trade-off between speed of convergence to a solution, and the stability of the system. As in any system that uses feedback, the LMS algorithm has the possibility of becoming unstable; stability in a system is defined here as tap weights $w(n)$ converging towards the optimal solution.[7] Haykin(2008) gives the stability criterion as:

$$0 < \mu < \frac{2}{\lambda_{max}}$$

Where λ_{max} is the maximum value of the eigenvalues of correlation matrix of input signal:

7.8.3 Correlation matrix

$$R = \begin{pmatrix} r(0) & ... & r(M-1) \\ ... & ... & ... \\ r^*(M-1) & ... & r(0) \end{pmatrix}$$

Given that $r(-k) = r^*(k)$ and $r(k) = E[u(n)u^*(n-k)]$
On the other hand, a larger step size leads to faster convergence to the solution. This means care must be taken to choose a good value of to ensure that the solution is converges quickly enough without becoming unstable.

7.8.4 Normalised LMS

There are a number of algorithms that improve on the performance of the LMS algorithm. One particular algorithm is the Normalised LMS (NLMS). The NLMS algorithm is very similar to LMS, but it normalises step size . The NLMS algorithm is based on the "principle of minimal disturbance", which [7](Haykin,2008) defines as changing the weight vector of an adaptive filter minimally from one adaptation cycle to the next. The NLMS algorithm achieves this by allowing to vary with input $u(n)$:

$$\hat{w}(n+1) = \hat{w}(n) + \frac{\hat{\mu}}{|u(n)|^2} u(n)e^*(n)$$

The NLMS allows greater control over step-size, and allows for better convergence towards the solution and invariance to varying inputs. Due to this, NLMS is often used over the LMS algorithm. The group shall consider a further improvement of the LMS algorithm for ANC - the Filtered-X LMS algorithm.

7.8.5 Implementation in Active Noise Control and Filtered-X LMS

It is worth to mention that much of the material used in this section comes from Adaptive Filters: Theory and Applications[6] by Farhang-Boroujenyny (2013).
To practically implement an LMS algorithm in ANC, some modifications to the algorithm need to be done. For one, the cancelling loudspeaker corresponding to y needs to construct a wave that destructively interferes with the noise source at the error microphone. Hence, a particular modification needs to be done [6](Farhang-Boujenyny, 2013):

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

A practical problem is that the reference and error microphones also receive waves from the cancelling loudspeaker, creating secondary paths on top of the problem of the primary path. The Filtered-X LMS

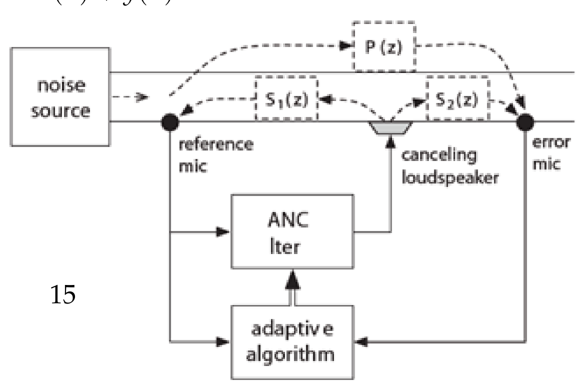


Figure 16: *Model of Active Noise Control with both secondary paths [6](Farhang-Boujenyny, 2013)*

makes an estimate of the secondary paths, and uses these to create a more accurate algorithm that deals with the additional complication(see fig.16).

An assumption is made that the path from the reference microphone and the loudspeaker is relatively long, and the algorithm only considers the path from the cancelling loudspeaker to the error microphone. This path will be known as path S(see fig.17).
The Filtered-X LMS algorithm makes an estimate of the secondary path, known as S. The LMS algorithm can then be re-formulated as[6] (Farhang-Boujenyny, 2013):

$$w(n+1) = w(n) - 2\mu e(n)x'(n)$$

Where $x'(n)$ is the convolution between $u(n)$ and $S(n)$.
After comparing the two algorithms, the group has decided to use the Filtered-X LMS algorithm due to its applicability to our particular problem – it should give better performance of the system due to the inclusion of the secondary path and reformulation towards ANC. However, this does not rule out the advantages of the NLMS algorithm. With enough development time and on a good enough processor, both algorithms could be combined to allow for significantly better performance.

7.8.6 Other Algorithms

Other algorithms that were considered include the Recursive Least-Squares (RLS) and Kalman Filter. However, due to their complexity in practical implementation, the group has decided that there is difficulty in implementing either one within the given time frame. Furthermore, both algorithms require computational complexity that may lead to unacceptable processing delays – both algorithms require matrix operations which are notoriously complex[7](Haykin,2008). This causes it to be deemed inappropriate for use in the project, using the current hardware. However, these algorithms do provide better performance if provided with hardware fast enough to deal with their complexity. These algorithms should be considered in future work, especially when dealing with nonstationary environments[7](Haykin,2008).

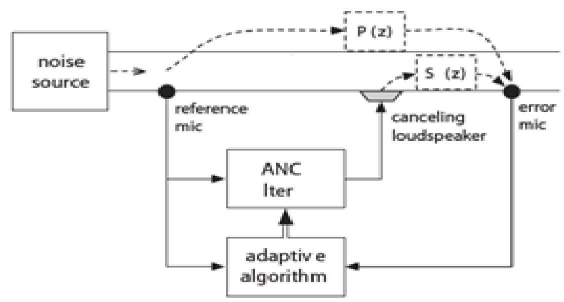


Figure 17: *Model of Active Noise Control with only one secondary path [6](Farhang-Boujenyny, 2013)*

8 Component Selection

Comparison tables of the different components can be found in Appendix F. This section shall just discuss the components selected and the reasons behind their selection. It must be emphasised that these components should be tested and substituted out for better components as hardware prototyping has not taken place yet.

- **DSP processors**
  The TMS320F28069FPNT chip (£11.01) was selected due to the good balance between memory, processing speed, and cost.

- **Loudspeakers**
  The group is tentatively looking at a 10W power level. The likely choice will be the HT-22/8 (£6.15) due to the low cost, as well as the higher voltage which will allow for some headroom.

- **Microphone**
  The most probable choice of microphones in this category will be the 352-2755(£ 15.00), the RS Pro Boundary Microphone, due to its dramatically lower cost. Microphones are the most used components, and using many expensive ones can lead to the cost ballooning. The only disadvantage with this microphone lies with the relatively large impedance, at 1000 ohms, but the group is confident that methods such as amplification can be work around this.

- **Power Supply**
  The final component that needs to be considered is the power supply, which is chosen to be a AC to DC power converter which is connected to the mains. The choice of power supply is motivated by the amount of power it can provide, which is estimated to be between 10-15W per system. The group will likely choose the DA12-120M P-M, due to the low cost, and the fact that it can supply 12W of power, which is within the target range. The power supply should be changed to supply more power should there be more power requirements in the future, in future updates to the design.

8.1 Cost Analysis

An approximate costing will be carried out in the section, along with comparison to traditional soundproofing. The group will be using RS online[3](RS Component, 2016) as a reference for prices, but it is likely that the total cost can be reduced even further if bought in bulk, or direct from the supplier.

Component	DSP chip	Speakers	Microphone	Power Supply
Unit Price (£)	11.01	7.46	15	9.45
Quantity	3	3	6	1
Total (£)	33.03	22.38	90	9.45

The total estimated cost is around £154.86 for a office cubicle with three partitions - with 2 microphones, 1 DSP chip and 1 speaker for each wall.

8.2 Sound Proofing

Type	Acoustical Fire Batts	Acoustics Studiofoam
Unit Area (m^2)	4.32	2.2
Unit Price (£)	25.30	63.23
Cost for 150 (m^2)	885.50	4,362.87

On first glance, the cost of soundproofing may not seem very expensive. Unfortunately, the cost can increase very greatly for larger dimensions, making it difficult to scale up soundproofing. For an office of 5 metres by 10 metres area and 4m height, an approximate calculation for a 3 wall cubicle can be made assuming that the partition only reaches 2m:

$$(5 + 10 + 5) * 2 * 3 = 120m^2$$

This leads to estimated costings of:

	Acoustical Fire Batts	Acoustics Studiofoam
Approximate cost	$28 * 25.30 = £708.40$	$55 * 63.23 = £3477.65$
Percentage increase	$708.40/154.86 * 100 = 457\%$	$3477.65/154.86 * 100 = 2246\%$

It can be seen that even with some price reductions from bulk purchasing, the prices can be very steep. This does not include other costs, such as the cost of installation or teardown. It should be emphasised that soundproofing cannot easily be transferred from office to office, making it difficult and potentially expensive to reconfigure an office.

9 Prototype and 1-D implementation

9.1 MATLAB simulation

A series of 3 MATLAB simulations are carried out to demonstrate the action of the adaptive filter. The MATLAB code for the simulations can be found in Appendix C:

9.2 Time Domain Analysis of NLMS filter

The group measured the effect of using a NLMS filter to do ANC on a random noise waveform, with varying values of step size. The results are shown below at the error microphone:

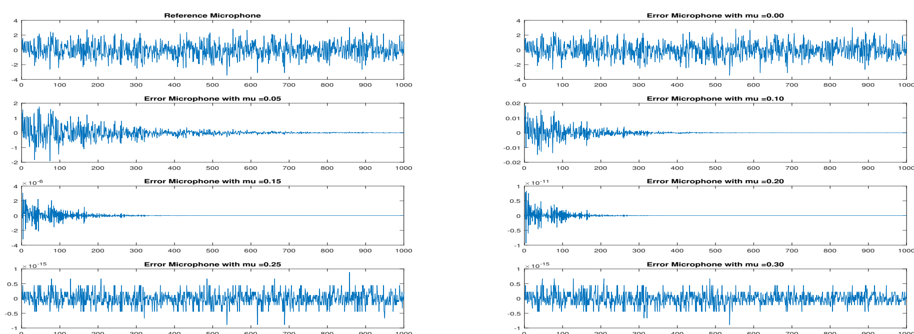


Figure 18: *Time Domain variation with μ*

What is observed is that adaptation to zero happens quicker with increasing values of μ, until a step size of 0.25 is reached, where the system becomes unstable and noise is no longer cancelled. This demonstrates what was elaborated on in a previous section.

9.3 Signal to Noise Ratio(SNR) Analysis of NLMS filter

This example takes a sine wave of frequency 100 Hz and peak to peak amplitude 2 as a signal, and is corrupted by some random noise. This noisy wave is then passed through an NLMS filter to get rid of the random noise. SNR variation with respect to μ was then measured, at the error microphone. The results are shown as figure 19.
The red line represents the SNR of the original noisy signal, while the blue line represents the filtered signal. SNR shoots up very rapidly, followed by a decrease with decreasing rate in SNR as μ approaches infinity and SNR approaches 0.

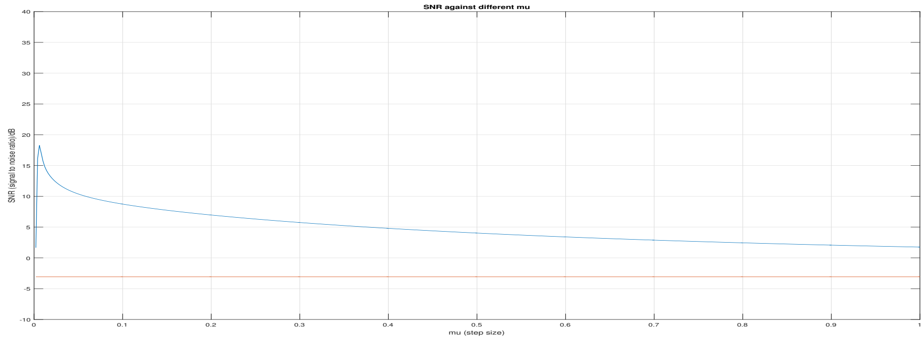


Figure 19: *SNR variation with μ*

9.4 Filtered-X LMS

This example simulates the effect of Filtered-X LMS on the noise waveforms measured in the computer lab. A step size of 0.008 is used, and a FIR filter is used to simulate the secondary path. An amplitude time graph of the action of the filter is shown below:

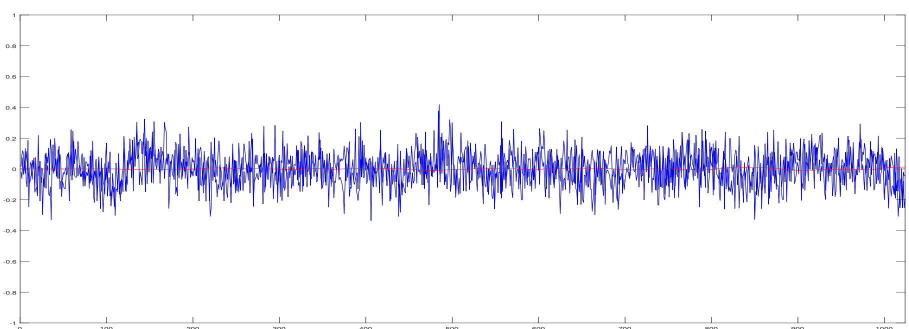


Figure 20: *Simulation of Filtered X*

The blue waveform represents the noisy signal, while the red waveform represents the filtered waveform measured at the error microphone. As can be observed, there is a very notable attenuation in noise. This demonstrates how the Filtered-X LMS filter will be used in controlling noise - the waveform represents what the listener hears and shows that the filter can theoretically be very effective in noise cancellation.

10 Future Work and Extensions

10.1 Accurate Acoustic Model

More work can be done on extending the acoustic model, to make it more robust to conditions that are not covered by the existing model. Firstly, more work can be done on extending the mostly 1-D acoustic model to incorporate more 3-D considerations. Furthermore, studies on the effects of room geometry can allow for more accurate calculations of eigenfrequencies and transfer functions, given more time and budget. This also improves microphone and loudspeaker selection and placement for better performance. Other parameters that can be studied include:

- Specific acoustic impedance
- Reflection constants
- Reverberation time
- Wavenumbers

10.2 Other Extensions

1. Configurable Microphones
   The design can be modified such that microphones contained within the office can be configured as either a error or reference microphone. This means the rooms can be designated as a "quiet" or "noisy" room respectively, depending on user wishes. Consequently, the noise in "noisy" rooms can be more easily profiled, and this allows for more flexibility in how the office is divided. Overall, this leads to better performance in the system as a whole.

2. Wireless
   The concept right now plans to use wires to transmit signals in between components. For a larger scale office, this may be inconvenient and difficult. A proposed extension to the design is to wirelessly transmit data between devices, eliminating the need for wires and allowing for components to be installed and repositioned easily. Using wireless transmission may increase power consumption, and introduce possible lag of 2 output periods[12](San Diego State University,2002). This is why implementing a wireless system requires further research outside of the scope of the report.

3. Predictive algorithm
   The problem of the listener being far away from the error microphone still exists. Something that can be explored in future work is using a predictive algorithm to estimate where the user's position is and generate the anti-phase wave at that point, eliminating the need for the user to be close to the microphone.

4. Wearable
   One last area for consideration installing microphones in wearable form to allow them to be close to the user while being able to be repositioned easily. According to proximity effect,[11](UCSC,1998), the closer the proximity of microphones to users, the better the performance.

11 Conclusion

The project has been a challenging but fulfilling exploration into how engineering and electronics principles can be used to improve our daily lives. The project shows how an ANC system might be implemented in reducing office noise and improving productivity. While no actual hardware has been implemented yet, the group believes that ANC technology can be viable given enough research and careful implementation. If the project is to be taken further, more work can be done on finding a better acoustic model, and implementing a better system based around this model. The problem of office noise will only get worse as more people and machines populate offices, making the task of an effective solution worthwhile to pursue.

References

[1] SP Banbury and DC Berry. Office noise and employee concentration: Identifying causes of disruption and potential improvements. *Ergonomics*, 48(1):25–37, 2005.

[2] Christopher A Brown and John F Schneider. Wide area noise cancellation system and method, May 27 2014. US Patent 8,737,634.

[3] RS Components. Rs online.

[4] DangerousDecibels. Decibel exposure time guidelines.

[5] Ignazio Dimino and Ferri Aliabadi. *Active Control of Aircraft Cabin Noise*, volume 7. World Scientific, 2015.

[6] Behrouz Farhang-Boroujeny. *Adaptive filters: theory and applications*. John Wiley & Sons, 2013.

[7] Simon S Haykin. *Adaptive filter theory*. Pearson Education India, 2008.

[8] Heinrich Kuttruff. *Room acoustics*. CRC Press, 2009.

[9] Sonoko Kuwano, Seiichiro Namba, and Takehisa Okamoto. Psychological evaluation of sound environment in a compartment of a high-speed train. *Journal of Sound and Vibration*, 277(3):491–500, 2004.

[10] Masao Nishikawa. Open-air noise cancellation system for large open area coverage applications, January 18 2007. US Patent App. 11/624,468.

[11] UCSC. Proximityeffect.

[12] San Diego State University. Delays in digital modulation.

Appendix A: Product Design Specification

Product Design Specification

Project: Peacemaker

Date: 13/3/2015

Author: Group 17

1. Performance

The following performance criteria are targeted for the project:

- Research has shown that normal conversation levels range from 25-65 dB. The target is to reduce the noise to a level 20 dB less than the upper limit, at 45 dB. Exceeding this threshold will be considered a great success.
- The frequency of human speech can vary a lot, with fundamental frequencies in the region of around 80 to 300 Hz (Kuwano, Namba and Okamoto,2004), and overall in the range between 0 to 5 kHz (Kutruff, 2009). While the product has to cancel noise particularly well within the region of 0 to 5 kHz, it should cancel noise in other areas of the spectrum within human hearing.
- The processing system must be fast enough to filter the input signal and output a wave that is exactly out of phase at the desired area. A propagation delay can lead to the wave to no longer be exactly 180 degrees out of phase at the desired point.
- The product must be able to run for as long as noise from the noise source needs to be cancelled. This means that the criteria for portability has been relaxed in favour of a solution that can be powered indefinitely, such as through mains.

2. Environment

The product must be able to operate at room temperature in all seasons and in most office environments. Most importantly, the environment should not cause components used in the product to degrade and cause any downgrade in the performance of the product within the targeted life in service.

3. Life in Service

The product must be able to be repeatedly used over a period of time without breaking down. The target product lifespan is one year on full performance. The device is targeted to run on mains; this means it can run indefinitely without charging with a mains power supply.

4. Maintenance

There should be no need for regular system maintenance. The intention is to use materials that don't attract as much dust to ease of the cleaning the exterior of the product.

5. Target Product Cost

The target market has been shifted towards office usage as budget is not much of an issue for large organisation. The aim is to make the product much cheaper than the physical counterpart, without compromising on any previous properties of the device that have discussed before. A comparison between physical soundproofing and the product will be done in a later section.

6. Competition

ANR is not a new idea. There are similar technologies available in the market, such as noise cancelling headphones and noise cancelling systems in cars that can cancel noise very efficiently. The project's use of active noise control as a "noisy room" in offices distinguishes it from its competitors.

7. Shipping

For the time being, the product will only be delivered in place of manufacture to ease shipping.

8. Packing

No special packing for the product required.

9. Quantity

One system will be delivered per noise-cancelling area. 3 modules per system aimed at this point.

10. Manufacturing Facility

Not applicable.

11. Customer

Our target customer are currently office owners who wish to improve the productivity of their employees. A peaceful environment for employees to work in can easily improve efficiency of work causing a potential increase in revenue and improve the working environment of the employees. Also, space can be very expensive in cities such as London; hence offices may have insufficient space to build meeting rooms. Our product is aimed at producing a virtual meeting room without the need of expensive physical barriers.

12. Size

The devices used in the solution cannot be too big; they need to be small and light enough to be portable. As a result, the product should be small enough to be moved around.

13. Weight

Relatively light around 2 kilogram or less

14. Materials

Could potentially use material that to prevent damage in case of household accidents like spilling water or leaks. Materials that affect performance will not be used.

15. Product Life Span

Dependent on company requirements.

16. Aesthetics, Appearance and Finish

This is an important criteria in determining whether or not the product will appeal to offices. Ideally, the product must be attractive, and fit in nicely with the office environment.
The product must be trendy, attractive and appealing enough for potential customers. The end product should be modern, elegant, and blends in with a variety of indoor environments. Different colours and other forms of customisability can be considered for the final product.

17. Ergonomics

The target audience has little time and little need for complicated controls; a device that can provide ANC reliably without the need for the user to spend a lot of time learning how to operate it would be ideal. Operation of the device is aimed at being simple and intuitive.

18. Standards and Specifications

Not applicable.

19. Quality and Reliability

Related to life in service and environment is quality and reliability. As customers will rely on the product to carry out activities and socialising without disturbances, the accuracy and reliability of cancellation is very important. Ideally, the product must be able to reproduce a satisfactory drop in the noise level every time, and cannot break down within the targeted life of service.

20. Shelf Life

As most of the components used will be primarily electrical components that have a relatively long shelf life. 3 years or more predicted.

21. Testing

Testing procedure detailed in report.

22. Processes

Not applicable.

23. Time Scale

Projected 1 year or more.

24. Safety

Minimal safety risk.

25. Company Constraints

Constraint of roughly £50 for product prototype. The cost of the product targeted lower than £40 to produce to ensure a good profit margin.

26. Market Constraints

Due to the daily evolution of technology, constant product feedback is required to ensure the product does not fall out of the market.

27. Patents, Literature and Product Data

ANC is not a new area of research. The group is aware of existing patents such as on “Open-air noise cancellation systems” (Nishikawa, 2007) and “Wide area noise cancellation systems” (Christopher A. Brown, 2012) do exist and will avoid infringing on any intellectual property laws. Our product shall be distinguished by portability and cost.

28. Legal

Not applicable

29. Political and Social Implications

Not applicable

30. Installation

Installation becomes a little more difficult due to the increased complexity of the design. Consequently, the design must be made to simplify set up as much as possible. Instruction manuals are provided to make the final product must be intuitive and simple enough for anyone to set up with ease.

31. Documentation

A full specification of the product including voltage, current and power will be included on the product's documentation. A user's manual will be provided to ensure customers can use the product with ease.

32. Disposal

Proper disposal of the product is documented in the user manual.

Appendix B: Minutes

Minutes: (20/1/2016 – 12/3/2016)

20 Jan 2016: Discussion with Mr Mike Brookes

<u>Attended:</u> Martin, Chao

- Difficult to cancel noise from a large area
- Fortunately, many existing solutions
    - Airplanes (see Southampton University)
    - Cars
    - Living rooms
- Important to determine source and direction of noise
- Simplify problem
    - Infinite duration
    - Single frequency
    - Uni-directional (1 dimensional)
    - Long distance
    - Fixed location of source
- Recommended algorithm NLMS and use FIR adaptive filter
    - Stadnard LMS poor at choosing parameters
- Recommended to use sampling frequency of 8kHz
    - Consider 10-15 kHz if really required
- 3 stages
    - Fetch
    - Process
    - Outputs
- Time delay approximately 2 chunks
    - Minimise as much as possible
- Sampling frequency proportional to computation speed^2

21 Jan 2016: Discussion with Dr Naylor

<u>Attended:</u> Martin, Chao

- Pipe 10 cm-diameter
    - Converting sound wave into a plane wave
    - Effective reduce the problem to almost one-dimension
- Reverberation
    - Microphone close to ears giving a huge advantage
    - Can be modelled as a virtual source
    - Soft surface absorb it. Largely depend on environment (e.g. living room, theatre,etc)
- MATLAB Demo should be the objective.
- DSP learning curve takes more than what we possibly have.
- Problem formulation
- Practical/Commercial application
    - Museum example,

- Good reference books:
  - Adaptive filter theory by Haykin, Simon S
  - Room acoustics by Kuttruff
  - Advanced signal processing – EE3 module

26 Jan 2016: Group Meeting

<u>Attended:</u> Whole group

- Weekly group update.
- Discussed findings from meetings with Dr Naylor and Mr Mike Brookes.
- Assigned research on algorithms, mathematics and physics of the project.
- Assigned Programming Tasks

3 Feb 2016: Group Meeting

<u>Attended:</u> Whole group

- Weekly group update
- Time management
- Discussed findings from research areas being studied
- Updated on the progress of programming area
- Assigned work for MATLAB simulations
- Website discussion

10 Feb 2016: Group Meeting

<u>Attended:</u> Whole group

➢ Challenges:
Need to determine

- Speaker
- Microphone
- Step Size
- Buffer Size

➢ Design:
Determine which one of the two options

➢ Programming side:
Separate Functions

Problem with GUI (Chao)

➢ Zi finished 1st chapter for maths
- Write down notes
- Share them with the rest of the group
- Write conclusions

➢ Extend the main idea (Christodoulos)
- More than one microphones
- Applications

    - Write something about
- Website
    - Send logo to the rest of the group
    - Update photos
    - Write Descriptions (Martin)
    - Roles:
        - Software development -> Chao
        - Algorithmic Design -> Zihao
        - Marketing Director -> Christodoulos
        - Webmaster -> Jason
        - Project Manager -> Martin
        - Acoustic Designer -> Jin
        - Founder & Software Development -> Sida

17 Feb 2016: Group Meeting

Attended: Whole group

- Weekly group update
- Review of the interim report
- Discussion of next steps

24 Feb 2016: Group Meeting

Attended: Whole group

- Brainstorming session
    - Brainstorming topic: "What application are we looking for most in our product?"
        - Ended up with Encryption for Voice signals, ANC for skype, office use
        - We chose to create a device for office use since we felt that is both an innovative and useful product for the industry.
        - Encryption was rejected for its complexity and ANC for skype because it was too common.
- Photos for the website

26 Feb 2016: Group Meeting

Attended: Whole group

- Finalize concept and application (office use)
- Revised Design Criteria
- Concept Selection
    - Noisy room (Selected)
    - Self-contained unit (the cube)
    - Matrix region
- Assigned market research
- Assigned research for microphone and speaker placement
- Challenges involved
    - Noise spectrum of voice
    - Microphone and speaker placement and types

1 Mar 2016: Group Meeting

<u>**Attended:**</u> Whole group

- Finish concept
  - Ergonomic/aesthetic design
  - Algorithmic Implementation
  - Detailed high level implementation
  - Possible future work/extensions
    - Movable Matrix
    - Wireless
    - Predictive algorithm
    - Multiplex
    - Self-contained
    - Wearable
- Update on website
- Update on programming scripts and simulations

5-6 Mar 2016: Group Meeting

<u>**Attended:**</u> Whole group

- Update on algorithmic design and research
- Update on market research
- Prototype design
- Research on acoustics
- Research on microphone and speaker types
- Costing estimation (both for our product and current soundproofing solutions)
- Started on report writing (draft)
- Small filming session

7 Mar 2016: Discussion with Mrs Perea

<u>**Attended:**</u> Whole group

- Discussion about report structure
  - Design criteria and what is not covered squarely by the PDS
- Clarifications on report content and design criteria
- Discussions of how to develop concept

9 Mar 2016: Group Meeting

<u>**Attended:**</u> Whole group

- Weekly update session
- Discussion of findings from all research areas
- Decided final steps before report deadline

12 Mar 2016: Group Meeting

<u>Attended:</u> Whole group

- Report writing
- Finalize website

Appendix C: Matlab code

1.Noise Analysis

It takes a input audio file and displaces its frequency spectrum.

```
noise = dsp.AudioFileReader('complab1.wav');

hfft = dsp.FFT('FFTLengthSource', 'Property', ...
'FFTLength', 1024);

AP = dsp.AudioPlayer('SampleRate',noise.SampleRate, ...
           'QueueDuration',10, ...
           'OutputNumUnderrunSamples',true);

while ~isDone(noise)
 audio = step(noise);       % taking inputs and convert them into a 1024*2
                            %double array


 bar(44100/2*linspace(0,1,512), 2*abs(fft(1:512)/1024));
 axis([0 1024 0 0.2])
 title('Single-sided amplitude spectrum of CompLab Noise signal');
 xlabel('Frequency (Hz)'); ylabel('|Y(f)|');

  drawnow
  nUnderrun = step(AP,audio);
end

release(noise); % release the input file
release(hfft);
release(AP);
```

2. different_mu

```
lms2 = dsp.LMSFilter('Length',11, ...
   'Method','Normalized LMS',...
   'AdaptInputPort',true, ...
   'StepSizeSource','Input port', ...
   'WeightsOutputPort',false);
filt2 = dsp.FIRFilter('Numerator', fir1(10,[.5, .75]));
t = 1:1000;
x = randn(1000,1); % Noise

a = 1; % adaptation control
 % step size
f = 200;
 j=1;
 figure;

 s = sin(2*pi*t'/f);
 d = x + s; % Desired

 i = 1;
%figure;
%plot(d);title('Original Signal');


snr_before_db = zeros(1,500);
snr_after_db = zeros(1,500);

for mu = 0.002:0.002:1

    snr_before = mean( s .^ 2 ) / mean( x .^ 2 );
    snr_before_db(i) = 10 * log10( snr_before ); % in dB

    [y, err] = step(lms2,s,d,mu,a);
    residual_noise =  y - s;
    snr_after = mean( y.^ 2 ) / mean( residual_noise .^ 2 );
    snr_after_db(i) = 10 * log10( snr_after );

    i = i+1;

end
mu = 0.002:0.002:1;
plot(mu, snr_after_db);hold on;axis([0 1 -10 40]);plot(mu, snr_before_db);
grid on, title('SNR against different mu');
xlabel('mu (step size)');ylabel('SNR (signal to noise ratio)/dB');
```

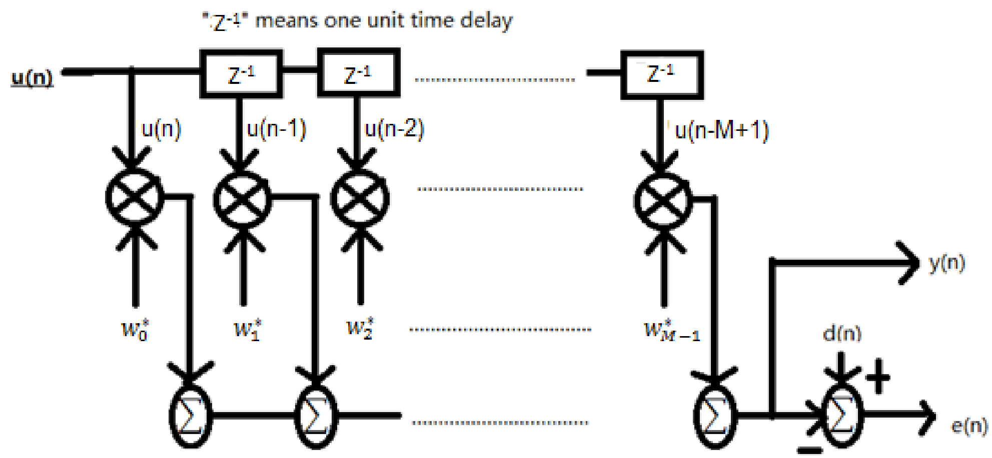
3.filteredX

```
noise = dsp.AudioFileReader('complab1.wav');
AP = dsp.AudioPlayer('SampleRate',noise.SampleRate, ...
            'QueueDuration',10, ...
            'OutputNumUnderrunSamples',true);

figure;
title('Active Noise Control of a Random Noise Signal');
legend('Original','Attenuated');
xlabel('Time Index'); ylabel('Signal Value');  grid on;


while ~isDone(noise)
x  = step(noise);
g  = fir1(47,0.4);
y = filter(g,1,x);

n  = 0.1*randn(1024,2);
d  = y + n ;
b  = fir1(31,0.5);

mu = 0.008;
ha = dsp.FilteredXLMSFilter(32, 'StepSize', mu, 'LeakageFactor', ...
     1, 'SecondaryPathCoefficients', b);

[y,e] = step(ha,x(:,1),d(:,1));

plot(1:1024,d,'b',1:1024,y,'r');axis([0 1024 -1 1]);
drawnow
step(AP, x);
end
```

Appendix D: Algorithm Research

Information in this section mainly taken from (Haykin, 2008).

Before the LMS algorithm, I would introduce the method of steepest descent which is deterministic in a known environment.



This is wiener filter

Where **u(n)** is a time series of input signal to the system **u(n)**=$[u(n), u(n-1) \dots \dots u(n-M+1)]$; **w** is a series of tap-weight assigned to each corresponding input signal, **w**=$[w_0^*, w_1^* \dots \dots . w_{M-1}^*]^T$; y(n) is the output of the adaptive filter,

y(n)= $\sum_0^{M-1} \mathrm{w}_\mathrm{K}^* \, \mathrm{u}(\mathrm{n}-\mathrm{k})$ (1)

and M is order of the filter, or y(n)=$\mathbf{w}^H$ **u(n)**, H means Hermitian i.e. do transpose and take conjugate; e(n)=d(n)-y(n) is called estimation error calculated by difference between desired output d(n) and real output y(n); For steepest descent, the cost function

J=E[|e(n)|^2] (2)

, is used to estimate the performance of the filter; Based on the feedback from the cost function, the weight controller would update the tap-weight to minimize the cost function.

The method of steepest descent demonstrates a feasible way for updating tap-weight:

w(n+1)=**w(n)**-0.5 μ **g(n)** (3)

where n is adaption cycle, keeping track of times of updation; μ is step-size parameter, it is a positive value and affects the rate of convergence to the optimum tap-weight vector $\mathbf{w}_o$; 0.5 is for mathematical convenience; **g(n)** or **∇J** is gradient vector of cost function with respect to the tap-weight vector.

Proof: why this method works (page 217-218)

Define the weight adjustment **δw(n)**=**w(n+1)**-**w(n)**=-0.5 μ **g(n)** (4)

Now we could prove why method of steepest descent works i.e. J(**w(n+1)**)<=J(**w(n)**)

Do 1st order Taylor series on J(**w(n+1)**):

Recall $f(x+a) = f(x) + f'(x)a$

J(**w(n+1)**)=J(**w(n)**)+$\mathbf{g}^H$**(n) δw(n)**

=J(**w(n)**)+ $\mathbf{g}^H$**(n)**(-0.5 μ **g(n)**)

=J(**w(n)**)-0.5||**g(n)**||^2<=J(**w(n)**)

Applying method of steepest descent(3) to the wiener filter(1)(2), we get:

w(n+1)=**w(n)**+ μ [**p**-**R** **w(n)**]   (5)

where **p** is cross-correlation vector between the input **u(n)** and desired output d(n), p(-k)**=** E[u(n-k) d^*(n)] so **p**=[p(0) p(-1) p(-2)….p(1-M)]^T (page 116); **R** is the correlation matrix of the input **u(n)**

$\begin{pmatrix} r(0) & \cdots & r(M-1) \\ \vdots & \ddots & \vdots \\ r(1-M) & \cdots & r(0) \end{pmatrix}or\begin{pmatrix} r(0) & \cdots & r(M-1) \\ \vdots & \ddots & \vdots \\ r^*(M-1) & \cdots & r(0) \end{pmatrix}$ given that r(-k)=r^*(k),

r(k)=$E[u(n)u^*(n-k)]$, **R**=E[$\boldsymbol{u^H u}$]

proof of the updating function (page 219-220)

We need to find gradient vector of cost function J first, recall J=E[|e(n)|^2]=E[e(n)e^*(n))](1)

e(n)=d(n)-y(n)=d(n)-$\sum_0^{M-1} \mathrm{w}_k^* * \mathrm{u}(n-k)$(2):

$$J=E[d(n) \times d^*(n)]-\sum_0^{M-1} w_k^* E[u(n-k)d^*(n)]-\sum_0^{M-1} w_k\ E[u^*(n-k)d(n)]$$
$$-\sum_0^{M-1}\sum_0^{M-1} w_k^* E[u(n-k)u^*(n-i)]w_i$$
$$=\sigma_d^2-\sum_0^{M-1} \mathrm{w}_k^* \mathrm{p}(-k)-\sum_0^{M-1} w_k p^*(-k)+\sum_0^{M-1}\sum_0^{M-1} w_k^* r(i-k) w_i$$

or =σ_d^2-**w*** **p**-**w** x **p***-**w**^{H} **R** **w**$_i$ (6)

where σ_d^2 is variance of desired response d(n)

Define gradient operator ∇_k=$\frac{\partial}{\partial ak}$+$j\,\frac{\partial}{\partial bk}$

Where w=a_k+jb_k

Recall e(n)=d(n)- $\sum_0^{M-1} \mathrm{w}_k^* * \mathrm{u}(n-k)$, J=E[e(n)$e^*$(n))]

$$\frac{\partial e(n)}{\partial a_k}=-u(n-k), \frac{\partial e(n)}{\partial b_k}=ju(n-k),$$
$$\frac{\partial e^*(n)}{\partial a_k}=-u^*(n-k), \frac{\partial e^*(n)}{\partial b_k}=-ju^*(n-k)$$
$$\nabla_k J=-2E[u(n-k)e^*(n)]=-2p(-k)+2\sum_0^{M-1} w_i r(i-k) \qquad (7)$$

∇J=-2**p**+2**Rw(n)**   (8)

Where **p**=[p(0) p(1) …..p(1-M)]

Final updating formula becomes:

w(n+1)=**w(n)**+ μ[**p**-**Rw(n)**]   (9)

Also for **∇J**=0 i.e. **w**=**w$_o$**:

p=**Rw$_o$**   (10)

NB: we won't use (10) to find **w$_o$**, because time complexity is high to do matrix inverse

Stability of the algorithm (page 222-226)

Actually, the formula is derived from 1st order Taylor series, we need to set a constraint on step-size parameter to guarantee the stability for which the new tap-weight converges to optimum tap-weight **w$_o$**:

Define weight-error vector **c(n)**= **w$_o$**-**w(n)**   (11)

to estimate the performance of each tap-weight vector with other 2 formula to construct a simultaneous equations:

w(n+1)=**w(n)**+ μ[**p**-**Rw(n)**]i.e.(9)

p=**RW**$_o$(10)

get:

c(n+1)=(**I**-μ**R**)c(n)

where **I** is identity matrix

Diagonalisation of **R** and rearrange:

R=**Q** **A** **Q^H** where **A** is composed of eigenvalues of **R** on diagonal

Q^H **c(n+1)**=(**I**-μ**A**)**Q^H** **c(n)**

Define **v(n+1)**= **Q^H** **c(n+1)**

v(n)=[v_1(n) v_2(n)….. v_k(n)] where k is number of eigenvalues of **R**

So **v(n+1)**=(**I**-μ**A**) **v(n)** (12)

$v_k(n+1)=(1-\mu\lambda_k)\, v_k(n)=(1-\mu\lambda_k)^n\, v_k(0)$

where λ is eigenvalues of **R**; **v(0)**= **Q^H** **w_o**

For stability,

|1-μλk|<1

$$0<\mu<\frac{2}{\lambda\max} \quad (13)$$

Unfortunately, correlation matrix R and cross-correlation p is not available for method steepest descent (deterministic) in practice. Thus, we need an algorithm that has capability to adapt to statistical variations in an unknown environment. We call this adaptive filter.

To implement adaptive filter, we have two ways: 1.method of stochastic gradient descent
2.method of least squares

We would talk about the first one for our project and one of the method of stochastic gradient descent is LMS (least mean square) algorithm by Widrow and Hoff.

•simple
•unlike the Wiener filter, it doesn't need knowledge of statistical characteristics of the environment
•robust i.e. good performance in face of unknown environment
•no requirement for inversion of correlation matrix

From Wiener filter to Adaptive filter implemented by LMS, simply modifying cost function from J=$E[e(n)e^*(n)]$ to J_S= $e(n)e^*(n)$ because in practice, the use of ensemble averaging is not feasible. We need to ignore expectation value and use the instantaneous value.

Do similar derivation like $\nabla_k J=-2E[u(n-k)e^*(n)]$(7),

$\nabla_k J_S=-2u(n-k)e^*(n)$ (14)

With gradient-descent formula **w(n+1)**=**w(n)**-0.5 μ **g(n)**(3), we get:

w(n+1)=**w(n)**+μ**u(n)**e^*(n)(page252-253) (15)

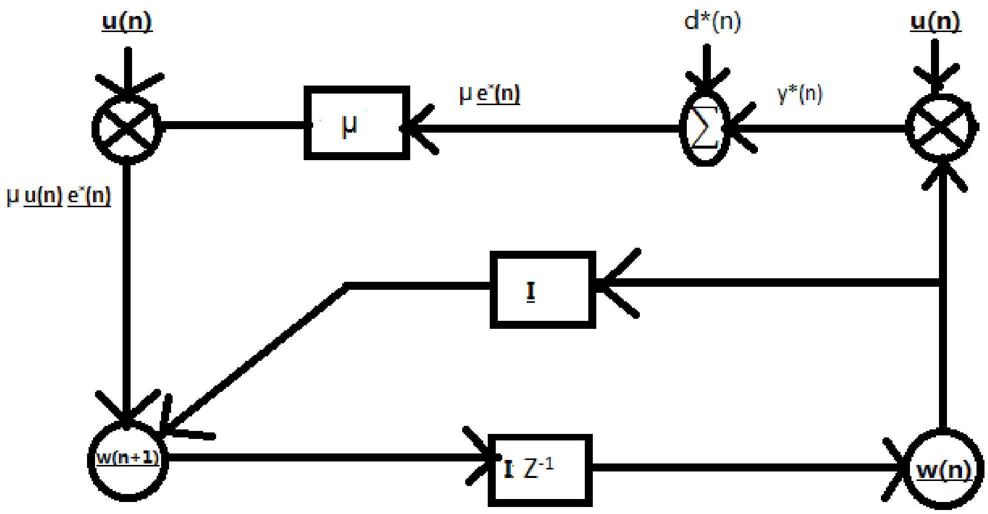
NB: no necessity of calculating correlation matrix, that is why LMS is simple and fast

Same condition of step-size parameter μ for stability: $0<\mu<\frac{2}{\lambda\max}$(13)

But this is not enough for stability. When we look at the signal-flow graph of LMS algorithm, we will see a problem regarding localized optimality: (page268-269)

w(n+1)=w(n)-0.5μ **u(n)**[d^*(n)-y^*(n)]

=w(n)- 0.5μ **u(n)**[d^*(n)-$\sum_0^{M-1} w_k\, u^*(\mathrm{n}-\mathrm{k})$]



Current estimation error:
e(n)=d(n)-$\boldsymbol{w^H(n)u(n)}$
Posterior estimation error:
r(n)=d(n)-$\boldsymbol{w^H(n+1)u(n)}$ (16)

For the localized perturbation to be small, we require that the step-size parameter satisfy:
$|1-\mu||\mathbf{u(n)}||^2|<1$
or $0 < \mu < \frac{2}{||u(n)||^2}$ (17)

proof

The problem could be interpreted as that "Find the optimum value of **w(n+1)** that minimizes the Euclidean between **w(n+1)** and **w(n)**, given a constraint $0 < \mu < \frac{2}{||u(n)||^2}$ (18)

To solve this problem, we use Lagrange multipliers:

Set a multiplier λ(n), now the Lagrange function is
$L(n)=0.5||\mathbf{w(n+1)}-\mathbf{w(n)}||^2+\lambda(n)(r(n)-(1-\frac{\mu}{2}||\mathbf{u(n)}||^2)e(n))$ (19)
Its gradient vector is
$\frac{\partial L(n)}{\partial \boldsymbol{w^H(n+1)}}=\mathbf{w(n+1)}-\mathbf{w(n)}-\lambda(n)\mathbf{u(n)}$
For gradient vector is 0,
$\mathbf{w(n+1)}=\mathbf{w(n)}+\lambda(n)\mathbf{u(n)}$ (20)
Substitute(19) into (18), we get:
$\lambda(n)=\mu e^*(n)$
The constrained optimization problem's solution is:
$\mathbf{w(n+1)}=\mathbf{w(n)}+\mu\,\mathbf{u(n)}\,\mathbf{e^*(n)}$(15)

The two methods mentioned above indicate a common feature of simplicity. However, both of them have an inevitable problem for choosing a proper step-size parameter μ in light of stability that is sensitive to input signal **u(n)**. Normalized LMS algorithm is capable of overcoming this problem by normalization of input signal with respect to its power $||\mathbf{u(n)}||^2$ or $\boldsymbol{u^H(n)u(n)}$

NLMS adaption mechanism(page 334-337)

$$\mathbf{w(n+1)}=\mathbf{w(n)}+\frac{\mu}{||\boldsymbol{u(n)}||^2}\mathbf{u(n)}e^*(n) \qquad (21)$$

proof

NLMS is to achieve principle of minimum error i.e. if the influence of input signal to the step-size parameter is removed, the difference between two successive tap-weight vector should be minimum, under a constraint desired response d(n)=$\underline{\boldsymbol{w}^{\boldsymbol{H}}(\boldsymbol{n}+\mathbf{1})\boldsymbol{u}(\boldsymbol{n})}$ (22)

Define tap-weight difference between two successive tap-weight vectors
$\underline{\boldsymbol{\delta w}(\boldsymbol{n}+\mathbf{1})}$=**w(n+1)**-**w(n)** (23)
NB: both equations are under posterior condition

Set a complex Lagrange multiplier λ^* and the Lagrange equation is

L(n)=$||\underline{\boldsymbol{\delta w}(\boldsymbol{n}+\mathbf{1})}||^2$-Re{$\lambda^*$(d(n)- $\boldsymbol{w}^{\boldsymbol{H}}(\boldsymbol{n}+\mathbf{1})\boldsymbol{u}(\boldsymbol{n})$)}

=(**w(n+1)**-**w(n)**)(**w(n+1)**-**w(n)**)^H+Re{λ^*(d(n)- $\boldsymbol{w}^{\boldsymbol{H}}(\boldsymbol{n}+\mathbf{1})\boldsymbol{u}(\boldsymbol{n})$)}

The gradient vector of the Lagrange function is

∇L(n)=2 $\boldsymbol{\delta w}(\boldsymbol{n}+\mathbf{1})$- λ^***u(n)**

=2(**w(n+1)**-**w(n)**)- λ^***u(n)**

We need gradient vector be 0 to get minimum tap-weight difference so

w(n+1)=**w(n)**+0.5λ^***u(n)**

Substitute (24) into constraint (22)

w(n+1)=**w(n)**+$\frac{1}{||\boldsymbol{u}(\boldsymbol{n})||}$**u(n)**e*(n)

(25)

In order to have a free degree to the adaption strategy, we need a control μ, called adaption constant, the optimum adaption constant $\mu_{op=}1$ (not considering environment effect). And also to avoid numerical difficulty when power of input signal $||$**u(n)**$||^2$ close to 0, we add a small positive constant δ to the denominator, the final formula is

w(n+1)=**w(n)**+$\frac{\mu}{\delta+||\boldsymbol{u}(\boldsymbol{n})||}$**u(n)**e*(n) (26)

<u>Stability analysis given additive noise v(n)</u>(page 337-338)

Desired response and error estimation become

d(n)=$\underline{\boldsymbol{w}_{\boldsymbol{o}}^{\boldsymbol{H}}}$**u(n)**+v(n) (27)

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

where v(n) is additive noise

The joining of v(n) get us to consider stochastic processing for estimating convergence performance of weight error.

Define weight error vector

c(n)=**w$_O$**-**w(n)** (28)

(27)(28) gives:

c(n+1)=c(n)- $\frac{\mu}{\delta+||\boldsymbol{u}(\boldsymbol{n})||}$**u(n)**e*(n) (29)

We base the stability analysis of NLMS on mean-square deviation

D(n)=E[$||$**c(n)**$||^2$] (30)

i.e. we need D(n+1)-D(n)<0

(29)(30) gives:

D(n+1)-D(n)=$\mu^2 E\left[\frac{|e(n)|^2}{||\boldsymbol{u}(\boldsymbol{n})||^2}\right]-2\mu E[Re\{\frac{\varepsilon(n)e^*(n)}{||\boldsymbol{u}(\boldsymbol{n})||^2}\}]=f(\mu)$<0 (31)

Where $\varepsilon(n)$=$\underline{\boldsymbol{c}^{\boldsymbol{H}}(\boldsymbol{n})\boldsymbol{u}(\boldsymbol{n})}$

It is just a quadratic inequality, $f(\mu)=A\mu^2-2B\mu<0$

$\frac{2B-2B}{2A}<\mu<\frac{2B+2B}{2A}$ $\rightarrow$ $0<\mu<\frac{2B}{A}$

So $0<\mu<2\frac{Re\{E[\varepsilon(n)e^*(n)/||\boldsymbol{u}(\boldsymbol{n})||^2]\}}{E[|e(n)|^2/||\boldsymbol{u}(\boldsymbol{n})||]}$ (32)

Also we can most negative $f(\mu)$ by differentiation

$$\mu_{op} = \frac{Re\{E[\varepsilon(n)e^*(n)/||\boldsymbol{u}(\boldsymbol{n})||^2]\}}{E[|e(n)|^2/||\boldsymbol{u}(\boldsymbol{n})||]} \qquad (33)$$

Recursive least square algorithm(RLS) has a property of rapid rate of convergence to optimum tap-weight but of higher complexity than LMS.
Unlike LMS that sets cost function to investigate the performance of a single error estimation, RLS sets a cost function of averaging error estimation within a time limit:

J=$\sum_1^n \beta(n,i)|e(i)|^2$ (34)

where $\beta(n,i) = \gamma^{n-i}$ is called forgetting index; γ is a positive constant close but less than unity

NB: $0 < \gamma < 1$, as i goes far from current time n, the factor tends to 0, because this Factor intends to reduce influence of distant past input data.

In order to make the estimation well-posed i.e. give more information to reconstruct the input-output mapping, we need to add a regularizing term to the cost function:

J=$\sum_{i=1}^n \gamma^{n-i}|e(i)|^2 + \delta\gamma^n||\underline{\boldsymbol{w}(\boldsymbol{n})}||^2$ (35)

Where δ is a positive and real number called the regularization parameter.

The RLS gives updating mechanism as:

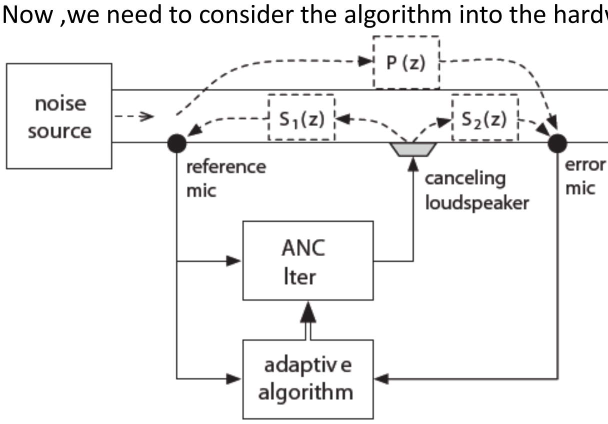
w(n)=**w(n-1)**+**k(n)**$\varepsilon^*(n)$ (36)

where k(n)=$\underline{\boldsymbol{\varphi}^{-1}(\mathbf{n})\mathbf{u}(\mathbf{n})}$; Correlation matrix $\underline{\boldsymbol{\varphi}(\mathbf{n})}$ =$\sum_{i=1}^n \gamma^{n-1}$ $\underline{\boldsymbol{u}(\boldsymbol{i})\boldsymbol{u}^{\boldsymbol{H}}(\boldsymbol{i})}$+$\delta\gamma^n\boldsymbol{I}$;

$\varepsilon^*(n)$=d(n)-$\underline{\boldsymbol{w}^{\boldsymbol{H}}(\boldsymbol{n}-\mathbf{1})\boldsymbol{u}(\boldsymbol{n})}$

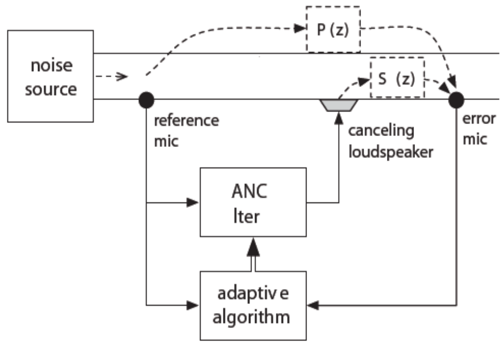
NB: RLS requires do matrix inversion and obviously its has higher complexity than LMS with **w(n+1)**=**w(n)**+μ**u(n)**$e^*(n)$

Now ,we need to consider the algorithm into the hardware setup in practice:

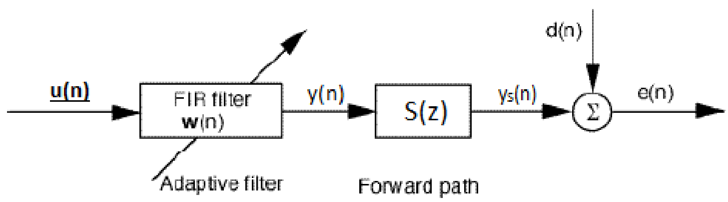


where P(z) impulse response or primary path means the acoustic path between the input signal and error microphone; Similarly, $S_1(z)$ is acoustic path between reference microphone which receive input signal, and cancelling loudspeaker; $S_2(z)$ is path between canceling loudspeaker and error microphone.

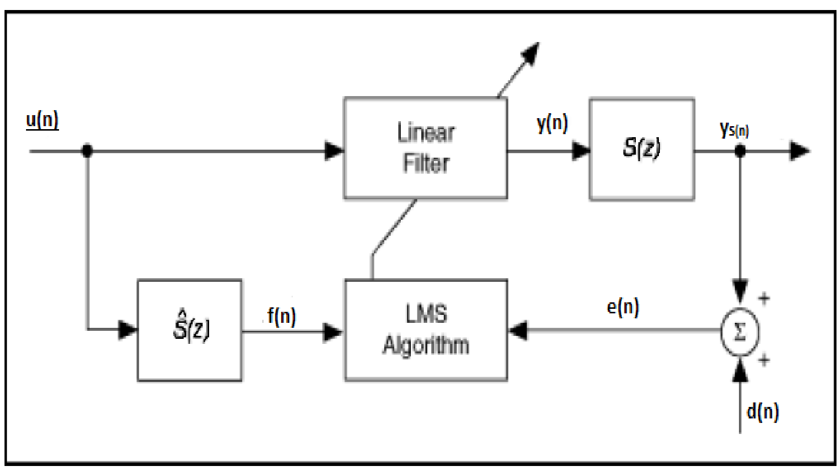
In our project, we will assume $S_1(z)$ is negligible by putting reference mic sufficiently distant from cancelling loudspeaker, thereby leaving a new application graph and forward path



Now, we need to reform some parameters:
e(n)=d(n)-$\sum_0^{I-1} s_i \sum_0^{M-1} w_k(n-i)x(n-i-m)$
J=e(n)e*(n)



∇J=$-2u_s(n)e^*(n)$
where $\underline{\boldsymbol{u_s(n)}}$ is convolution between u(n) and **s**
w(n+1)=**w(n)**+μ$\underline{\boldsymbol{u_s}}$e*(n) (37)



Thus, we have to add a compensating block to update $\underline{\boldsymbol{u(n)}}$ to $\boldsymbol{u_s(n)}$ as illustrated in the signal-flow graph

For stability analysis:
$0 < \mu < \frac{2}{||\boldsymbol{u_s(n)}||^2}$ (38)
NB: $\hat{S}(z)$ is an estimation of S(z) which could be found using knowledge of control system or by knowledge of acoustic wave.

For a non-stationary environment, we need to introduce Kalman filter of which the measurement tracks statistical noise and other inaccuracy. It is a linear, discrete-time, finite-dimensional system with a key property of minimum mean-square estimator of the state of a linear dynamic model. The Kalman filter is composed of two key part, system equation and measurement equation, formulating a state-space model. Therefore, we find some basic knowledge of state-space model from control engineering:

Given a system with input u(t), output y(t), and there is a differential equation between y(t) and u(t) such that $y^{(n)} + a_{n-1}y^{(n-1)} + \cdots + a_1 y = a_0 u(t)$, we define state $x_1 = y, x_2 = y^{(1)}, \ldots, x_n = y^{n-1}$ we could find $x_k^{(1)} = x_{k+1}, and\ x_{n+1}^{(1)} = +a_{n-1}y^{(n-1)} - \cdots - a_1 y + a_0 u(t)$

Define state derivative vector $\dot{x} = [x_1^{(1)}, x_2^{(1)}, x_3^{(1)} \ldots, x_n^{(1)}]^T$, and state vector $x = [x_1, x_2, x_3, \ldots x_n]$, now we can construct two equations:

$$\dot{x(t)} = Ax(t) + Bu(t)\ state\ equation$$
$$y(t) = Cx(t) + Du(t)\ output\ equation$$

Where A is nxn matrix, called system matrix

B is nxr matrix, called input matrix

C is pxn matrix, called output matrix

D is pxr matrix, called direct feedthrough

To solve this equation we need one initial condition $x(0) = x_0$

In Kalman filtering problem, we rename the state equation with 'system equation', and output equation with ' measurement equation':

x(n+1)=**F(n+1)x(n)**+**v_1(n)**

where **v_1(n)** is system noise, modeled as 0 mean white noise

y(n)=**C(n)x(n)**+**v_2(n)**

Kalman filter as the unifying basis for RLS algorithm

Kalman filtering algorithm, or covariance filtering algorithm.

Before talking about the algorithm, we define an expression a(n|b_n), meaning estimate of a(n) given measurement b_n, b_{n-1}, b_{n-2}, ……

Define $\begin{cases} \boldsymbol{x(n)} = \boldsymbol{w_o} \\ y(n) = d^*(n) \\ \boldsymbol{F(n+1,n)} = \gamma^{-1/2}\boldsymbol{I} \\ \boldsymbol{C(n)} = \boldsymbol{u^H(n)} \\ v(n) = e_o^*(n) \end{cases}$

x(n+1|y_n)=$\gamma^{-1/2}$**x(n|y_{n-1})**+**g(n)**$\alpha(n)$

y(n)=**u^H(n)x(n)**+v(n)

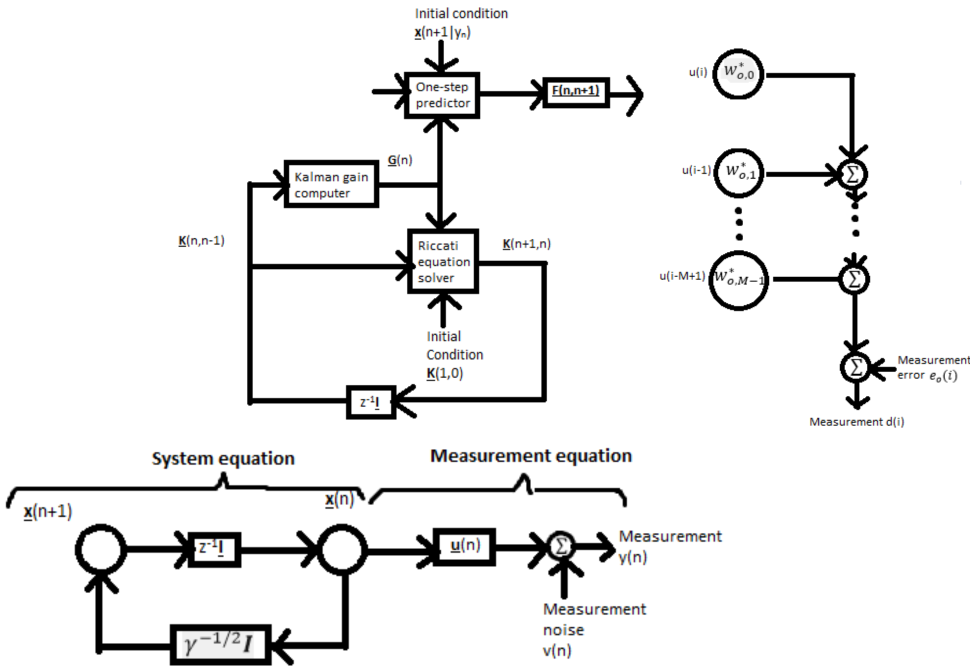
where γ is scaling factor, $0 < \gamma < 1$;

innovation information $\alpha(n)$ =**y(n)**-**y(n|y_{n-1})**=y(n)-**u^H(n)x(n|yn-1)**,represents new information in the measurement **y(n)**;

Kalman gain **g(n)**=$\frac{\gamma^{-\frac{1}{2}}\boldsymbol{K(n-1)u(n)}}{\boldsymbol{u^H}(\boldsymbol{n})\boldsymbol{K(n-1)u(n)}+1}$;

Error in state prediction **K(n)**=γ^{-1}**K(n-1)**- $\gamma^{-1/2}$**g(n)u^H(n)K(n-1)**;

Initial conditions:$\begin{cases} \boldsymbol{x}(1|y_0) = E(\boldsymbol{x}(1)) \\ K(1,0) = E[(\boldsymbol{x}(1) - E[(\boldsymbol{x}(1)])(\boldsymbol{x}(1) - E[\boldsymbol{x}(1)]^H] \end{cases}$



These three graphs are prediction, measurement, state-space model of RLS algorithm respectively

Computation complexity table:

Method	Applies to	Produces	Cost per step	Convergence	Description
Power iteration	General	eigenpair with largest value	$O(n^2)$	Linear	Repeatedly applies the matrix to an arbitrary starting vector and renormalizes.
Inverse iteration	General	eigenpair with value closest to μ		Linear	Power iteration for (A - μI)−1
Rayleigh quotient iteration	Hermitian	eigenpair with value closest to μ		Cubic	Power iteration for (A - μiI)−1, where μi for each iteration is the Rayleigh quotient of the previous iteration.
Preconditioned Inverse iteration[5] or LOBPCG algorithm	Positive Definite Real Symmetric	eigenpair with value closest to μ			Inverse iteration using a preconditioner (an approximate inverse to A).
Bisection method	Real Symmetric Tridiagonal	any eigenvalue		linear	Uses the bisection method to find roots of the characteristic polynomial, supported by the Sturm sequence.
Laguerre iteration	Real Symmetric Tridiagonal	any eigenvalue		cubic[6]	Uses Laguerre's method to find roots of the characteristic polynomial, supported by the Sturm sequence.
QR algorithm	Hessenberg	all eigenvalues	$O(n^2)$	cubic	Factors A = QR, where Q is orthogonal and R is triangular, then
		all eigenpairs	$6n^3 + O(n^2)$		

					applies the next iteration to RQ.
Jacobi eigenvalue algorithm	Real Symmetric	all eigenvalues	$O(n^3)$	quadratic	Uses Givens rotations to attempt clearing all off-diagonal entries. This fails, but strengthens the diagonal.
Divide-and-conquer	Hermitian Tridiagonal	all eigenvalues	$O(n^2)$		Divides the matrix into submatrices that are diagonalized then recombined.
		all eigenpairs	$\frac{4}{3}n^3 + O(n^2)$		
Homotopy method	Real Symmetric Tridiagonal	all eigenpairs	$O(n^2)$		Constructs a computable homotopy path from a diagonal eigenvalue problem.
Folded spectrum method	Real Symmetric	eigenpair with value closest to µ			Preconditioned inverse iteration applied to (A - µI)2
MRRR algorithm[8]	Real Symmetric Tridiagonal	some or all eigenpairs	$O(n^2)$		"Multiple Relatively Robust Representations" - Performs inverse iteration on a LDLT decomposition of the shifted matrix.

(Dhillon, Inderjit S.; Parlett, Beresford N.; Vömel, Christof, 2006)

(Neymeyr, 2006)

Operation	Input	Output	Algorithm	Complexity
Matrix multiplication	Two n×n matrices	One n×n matrix	Schoolbook matrix multiplication	$O(n^3)$
			Strassen algorithm	$O(n^{2.807})$
			Coppersmith–Winograd algorithm	$O(n^{2.376})$
			Optimized CW-like algorithms	$O(n^{2.373})$
Matrix multiplication	One n×m matrix & one m×p matrix	One n×p matrix	Schoolbook matrix multiplication	$O(nmp)$
Matrix inversion*	One n×n matrix	One n×n matrix	Gauss–Jordan elimination	$O(n^3)$
			Strassen algorithm	$O(n^{2.807})$
			Coppersmith–Winograd algorithm	$O(n^{2.376})$
			Optimized CW-like algorithms	$O(n^{2.373})$
Determinant	One n×n matrix	One number	Laplace expansion	$O(n!)$
			LU decomposition	$O(n^3)$
			Bareiss algorithm	$O(n^3)$
			Fast matrix multiplication[17]	$O(n^{2.373})$
Convolution	two nxn matrix	one number	Fast convolution algorithm	$O(n \log n)$

(Henry Cohn, Robert Kleinberg, Balazs Szegedy, and Chris Umans, 2005)

(Raz, 2002)

(T.H. Cormen, C.E. Leiserson, R.L. Rivest, C. Stein, 2009)

Appendix E: Links to videos, reports and websites:

Promo Video: https://www.youtube.com/watch?v=GfreOpo29bM

Project website: http://www.ee.ic.ac.uk/jason.yuan14/yr2proj/index.html

Interim Report: http://www.ee.ic.ac.uk/jason.yuan14/yr2proj/our_progress_interim_report.html

Appendix F

11.1 DSP Processors

Part Name	TMS320F28 335ZJZA	TMS320F28 069FPFPQ	TMS320F28 069MPNT	TMS320F28 069FPNT	TMS320F28 054FPNT	TMS320F28 052FPNT
Memory (kB)	512	128	128	128	64	32
Processing Speed (MHz)	150	90	90	90	60	60
RAM (kB)	68	50	50	50	8	8
ADC Number	16	12	12	12	16	16
ADC Resolution	12bit	12bit	12bit	12bit	12bit	12bit
Package Type	PBGA	HTQFP	LQFP	LQFP	LQFP	LQFP
Typical Operating Supply Voltage (V)	3.135 - 3.465	3.3	3.3	3.3	3.3	3.3
Cost (£)	15.35	12.86	12.71	11.01	6.89	6.31

The likely choice will be the TMS320F28069FPNT, due to the good balance between memory, processing speed, and cost.

11.2 Power Supply

The final component that needs to be considered is the power supply, which is chosen to be a AC to DC power converter which is connected to the mains. The choice of power supply is motivated by the amount of power it can provide, which is estimated to be between 10-15W per system

Mfr. Part No.	5311123	ECP-15-12U	8C94081	DA12-120UK-M	DA12-120MP-M	5311103
Output Voltage (V)	3 - 12	12	3 - 12	12	12	3 - 12
Power Rating (W)	18	15	14	12	12	3.6
Cost (£)	19.40	16.05	11.68	13.00	9.45	10.65

The group will likely choose the DA12-120M P-M, due to the low cost, and the fact that it can supply 12W of power, which is within the target range. The power supply should be changed to supply more power should there be more power requirements in the future, in future updates to the design.

11.3 Loudspeakers

Mfr. Part No.	FRS 7 4 OHM	TEBM130 H10-8	SC 5.9 8 OHM	HT-22/8	FR 9.15 4 OHM	FR 12 8OHM	2008
Power Rating(W)	8-15	10	10-15	10-20	15-30	15-40	10-15
Impedance (Ω)	4	4	8	8	4	8	8
Power Supply(V)	7.74596	6.3245	10.9544	10.9544	10.9544	17.8885	10.9544
Cost (£)	10.57	11.56	7.46	6.15	15.66	12.89	8.68

The likely choice will be the HT-22/8 due to the low cost, as well as the higher voltage which will allow for some headroom.

11.4 Microphones

RS Stock No.	352-2755	819-9476	819-9470
Type	Electret Condenser	Condenser	Condenser
Frequency Response	60 Hz - 16 kHz	70 Hz - 16 kHz	70 Hz - 16 kHz
Directionality	Omnidirectional	Cardioid	Omnidirectional
Cost (£)	15	113.62	113.62

The most probable choice of microphones in this category will be the 352-2755, the RS Pro Boundary Microphone, due to its dramatically lower cost. Microphones are the most used components, and using many expensive ones can lead to the cost ballooning. The only disadvantage with this microphone lies with the relatively large impedance, at 1000 ohms, but the group is confident that methods such as amplification can be work around this.