UNIVERSITY OF SOUTHAMPTON

SCHOOL OF ENGINEERING SCIENCES

Aerodynamics and Flight Mechanics Research Group

Advanced Acoustic Wind Tunnel Measurement Technologies

by

Alexander Carballo-Crespo

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ABSTRACT

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Measurements in hard-walled, closed section wind tunnels are desirable for the development of quiet aircraft and to validate computational results and whilst Open-jet anechoic facilities are a better measuring environment acoustically; closed-section wind tunnels offer high confidence in the aerodynamic characteristics of the testing conditions. Aeroacoustic noise from aircraft continues to be a major issue to government and industry and the accuracy and validity of acoustic measurements in Closed Section Wind Tunnels is of paramount importance.

This project began by building on existing concepts; augmenting and modifying technology to fit various wind tunnel facilities. After successful implementation of the microphone array in an industrial setting further research into improving the physical technology was started. One of the restrictions of such testing is the poor signal-to-noise ratio (SNR) when using arrays of microphones mounted on the wind tunnel wall. This can limit the ability to discern acoustic sources which are near, or below, the background noise level of the facility.

The second part of this study looked to investigate how sensor mounting details can help to improve SNR. Within this report a systematic study of microphone mounting strategies is presented. Results showed that recessing individual microphones by the depth of the microphone diameter (*d*) up to *2d* can provide up to 3dB improvement. Increasing the recess depth beyond *2d* provided up to 10dB improvement, with recessing to *10d* depth providing up to 20dB improvement. The greatest improvements occurred below 25 kHz, although there is improvement across the 0 to 48 kHz range. The effect of countersunk recessing was either no improvement, or an increase in the background noise level of up to 20dB, possibly due to cavity mode oscillations within the recess aperture. Significant differences in SNR were observed between Kevlar cloths of different densities, and with a silk covering. A reduction in background noise level of 5 to 10dB was observed when acoustic foam lining was added to the floor of the recessed array. Overall this study concludes that the use of recessed arrays with acoustic foam lining may significantly improve microphone array SNR in hard-walled wind tunnel testing.

The final part of the study aimed to find ways of improving the microphone array for a given number of sensors, looking at directivity from noise sources from test models in the wind tunnel. The primary concern was to find the range at which the array is a viable tool for source location and to determine the error in sources at the extremes of the range of the array to improve the measurement technologies for the future.

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Nomenclature

d	=	Microphone diameter
dB	=	decibel
gm	=	gram
kHz	=	kilohertz
mm	=	millimeter
m/s	=	metres per second
M	=	Mach number
PSD	=	Power Spectral Density
SPL	=	sound pressure level
U	=	freestream velocity
V_0	=	Reference Velocity (m/s)
M_0	=	Reference Mach Number
Baro	=	Wind Tunnel Barometric Pressure
T_{I0}	=	Wind Tunnel Test Section Temperature
V_{0C}	=	Wind Tunnel Test Section Wind Velocity
ALPHA	=	Angle of Attack in degrees (°)
BETA	=	Sideslip in degrees (°)

Rossiter Mode Formula

St	=	Strouhal number
f	=	frequency of vortex shedding,
L	=	characteristic length
n	=	cavity mode
Uc	=	convection velocity

Beamforming Formula

The Beamforming equations can be found in section 2.4 on page 36.

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1. Introduction

The overall aims are to research and study methods by which current acoustic wind tunnel technology can be improved and implement the advanced technologies in a working wind tunnel. In doing so, also develop current and future wind tunnel testing techniques to improve and accelerate the aircraft design process. An overview of current wind tunnel measurement techniques and technologies will also be carried out.

This research is necessary given the current and projected limitations of computational methods, such as accuracy and the time required for complex calculations; it will continue to be necessary to use wind tunnel testing for use in gathering empirical data for air vehicle design.

Microphone arrays, flush mounted into wind tunnel test sections provide unobtrusive aeroacoustics measurements whilst allowing aerodynamic testing to be performed simultaneously. This reduces time and cost which is beneficial for industry.

To that end this thesis is composed of three studies or parts, split into Year 1, 2 and 3 (found in sections 3, 4 and 5 respectively). Part one covers the acoustic analysis of the Filton wind tunnel as well the subsequent design work that went into designing the microphone array for the site as well as some concept design for point source designs. This first part was validated with a proof of concept test proving that Beamforming could be performed in an existing industrial wind tunnel with off the shelf components, as part of a standard aerodynamics testing suite, the results of which can be found in part two. The study in year 1 builds on the work performed by Benjamin Fenech who initially developed the Matlab code which implements the Beamforming (ref. [70]).

The second part, which forms the main part of this thesis focused on improving the Signal to Noise Ratio (SNR) of the off the shelf microphones used for the work in part one. By experimenting with various microphone mounting techniques it was theorised and determined that significant improvements could indeed be made. The paper produced and presented by the author for the AIAA 2009 conference titled “An Investigation of Microphone Array Installation” forms the basis for section 4.2.

The positive effects from the investigation on array installation were subsequently applied to the array construction and the work in part two carried forward to improve the array technology employed, as well as providing an analysis of the practicality of the various techniques to increase the effectiveness of source location and identification. A number of post-processing techniques were applied to improve the results from the various tests and in combination with the developed array design significant improvements in the beamforming plots were attained.

The third part continued the study into practical applications with the use of a moving array (designed by the author) to ascertain if it would be possible to identify noise source directivity in a closed section wind tunnel at the facility at Airbus. It was determined that it was possible to define directionality based on source strength of the same source viewed from different angles of the wind tunnel model.

Whilst it is acknowledged that there are limitations and assumptions that are made with beamforming, this research focuses on the application of beamforming theory to industrial situations and environments and provides solutions for real world challenges. In this way the focus of the study is the application of known beamforming techniques to the industrial setting. Theories and techniques were combined to complete studies performed in a wind tunnel not optimised for acoustics (Filton, Airbus) in a real-world series of industrial experiments.

This thesis sets out to prove that the implementation of beamforming technology (as of 2008) for noise detection could be executed in a Wind Tunnel that was not designed for acoustic studies. The study also seeks to prove that significant and useful data can be gathered using cost effective off the shelf components. At the time of the studies implementation of these techniques in a closed section wind tunnel was not prevalent using cost-effective components for industrial level testing.

1.1. Primary Objective:

The first objective in this thesis was to develop a microphone array for the locating of sound sources on wind tunnel models during wind tunnel tests. The project involved building on existing technology and augmenting and modifying said technology to fit the wind tunnel facilities at the University of Southampton and the Filton Airbus UK sites. This development of microphone array technologies for use in hard-walled wind tunnels required study into acoustics, aero-acoustics and computational methods to gather and present the data received in a clear and coherent way for aircraft designers.

The basis for this objective is the required reduction in aircraft noise as stated in the 2020 accord. To ensure the sustainability of air transport, aircraft noise must be reduced so that the disturbance to people around airports is minimised. This is to be achieved through implementation of quieter engines and quieter airframe structures, the latter being more important especially when aircraft are landing and their engines are trimmed right back. Modern engines are indeed much quieter than their old counterparts and it is airframe noise that is now being targeted for noise reduction. While at take-off, engines still are the dominant noise source, airframe noise is as important as the engine noise on the approach and at landing. The problem is that when approaching for landing, the aircrafts fly over populated areas at low altitude. Therefore, the high levels of radiated noise have a big impact in community noise. The main components of the airframe noise are the high lift devices and the landing gears. In order to satisfy noise regulations imposed by aviation authorities and some airports, the noise levels needs to be further reduced in future years.

1.2. Secondary Objectives:

The secondary objective was to look into and investigate alternative and novel wind tunnel technologies and how they could be improved to give better performance results. The development and implementation of new untested measurement technologies was also within the scope of this project and how these technologies may be improved to give better performance. This research was done internally at Airbus and will not be included in this report.

Given the limitations on computational methods, such as accuracy and the time required for complex calculations it will be necessary to have wind tunnels that can gather as much information as possible for test models, with a view to have the most accurate representation and the most useful data for use in design and modification of future projects.

It should be noted that wind tunnel such as the Filton wind tunnel were not initially designed with acoustics or many of these other modern techniques in mind and as such provide a challenge in having those techniques implemented within them successfully.

This report aims to give a clear description of the project goals and what has been achieved in the study. All results gathered from the years of study will be represented in this report.

The report will show what techniques have been learnt and mastered and which skills are still require for successful completion of the overall project. In addition this report will identify the proposed timetable for the projects and any unforeseen delays incurred due to software issues or equipment problems. Any project delay will be assessed and new timetables will be produced.

2. Literature Review

2.1. Introduction

This thesis report begins with a literature review. For this review a specific examination is made into acoustic and aero-acoustic investigations to continue with the initial research into acoustic array development.

The main aim of this literature review is to provide a starting point for continuing research into the field of aeroacoustic noise location. Within the general field of wind tunnel technology the focus is on aeroacoustic noise sources and locating them within wind tunnel tests. Thus the initial emphasis will be on literature that fulfils this requirement.

2.2. The Research

One of the aims of this review is to look at what has been done in the field of acoustics and more specifically a look at what has been done within wind tunnels, in recent times, with a view to identify the state-of-the-art and make the distinction between up to date technology, past technology and technology that is still in the theoretical field.

Special interest is made in previously successful acoustic array implementation in any hard-walled wind tunnels as this is directly related to the primary objective of this study. With this is mind a specific distinction is made between studies done in closed section wind tunnels and those performed in open section wind tunnels. Since this study is mainly orientated towards closed section wind tunnels, there is subsequently more interest in the studies done in those areas. Thus in this review four clear sections are revealed; the first two sections look at different wind tunnel set-ups, broadly speaking there are two types of wind tunnel: Open section wind tunnel studies and Closed Section wind tunnel studies. Set one will look at studies performed in open jet facilities, whilst set two will concentrate on facilities with totally enclosed test sections such as the facilities at the University of Southampton and the Filton site.

The third section is set aside for all encompassing papers and documents; this section focuses on papers which describe whole systems and provide overall breakdowns of entire acoustic measurement mechanisms from the theory to experimental implementation. These documents normally encompass a number of papers brought together or a series of research done over a length of time.

The final section of papers are the 'Theoretical' papers, these tend to be from earlier periods, although there are a number of papers that look at completely new techniques. These papers focus entirely on mathematical and theoretical studies.

Papers in either of the first two sets are specifically focused on research and experimentation done in the respective wind tunnel environment. Papers [1-11] describe studies performed in open section (or open jet) air flows whilst papers [12-23] focus on studies performed in closed section wind tunnels.

The third section is reserved for all encompassing papers and documents; this section focuses on papers which describe whole systems and provide overall breakdowns of entire mechanisms from the theory to experimental implementation, such as the study performed at the University of Southampton by Benjamin Fenech, [24]. As expected there are fewer documents in this section as all-encompassing documents that cover every aspect of a subject tend to be somewhat rarer, however one such document that does encompass everything that is required for successful acoustic measurement is the book 'Aeroacoustic Measurements' [25] which contains in itself a number of papers which are referenced separately [26-28].

The final set of papers consisting of mathematical and theoretical publications is in stark contrast to the more focussed publications outlined in the first two sections. These papers, [34-54] involve specific study into one aspect of the research area. Although the scope is limited, these papers do provide a very thorough and mathematical analysis. These papers show the level of research at the very extreme end of theory and calculation; it is included in this review to provide a comparative view point to the non-specific, lesser detailed, overview given by the first two sets of publications. Although by no means are the papers in the first two sets lacking in detail.

Indeed the quality of research in the studies is of the highest level with all the aspects required for industrial replication covered. This is to be expected since the majority of papers are industrially supported and leading companies such as NASA and Audi are behind the research. The Industrial influence is evident as the majority of the papers in the first two sections are often laid out with a view to show how effective this type of acoustic measurement can be on models with a real life value.

Grouping the papers in such a broad way allows a large number of papers to be organised and reviewed together and gives an overview of how the research is progressing with respect to areas of interest to this study. Although there are numerous papers in the field of acoustics a closer inspection, looking at more relevant aeroacoustic papers that deal with noise detection technology, reveals about eighty or so papers as of 2008. For this report a look is taken at the fifty most relevant publications. This literature review looks at a total of fifty papers that were selected for their relevance with respect to the specific problem of detecting noise in a working wind tunnel. The total of fifty-four references also included a single book [24] and the chapters within it which were referenced [26-28]. The research publications that have been selected are all in the specific aeroacoustic field, however, to provide some variety the publications selected cover a wide (relatively speaking) period of time and come in various forms from various mediums, from presentation notes to magazine articles.

Over viewing each individual paper for this report would take far too long, so this subsequent section will look in detail at the different publications collectively, identifying the key findings and conclusions achieved in each of the research document groupings and their relevance to the research area.

The first set of papers looking at open-jet experiments, where for the most part the aim is to develop an entire set of methods. These papers focus their research into studying methods employed in acoustic testing in wind tunnel conditions. These methods for an experimental testing system are analysed and compared; the conclusions of the papers clearly indicating that one method is superior to another. Some of these papers in this set, [1 - 4, 7 – 8 and 11] look at both theory and experiments. In [2] a series of extensive studies are done into the effect that applying leading-edge serrations has on noise produced from airfoils, the paper concluding that serrations do indeed lower the noise produced much as the wing of an owl having serrated edges allows it to fly silently. This paper performs experiments on propellers in a free jet environment and plots the reduction in sound noise measured against different types of serration that was employed. The paper if nothing else categorically proves that noise is produced from structures such as airfoils. Indeed tonal noise is measured off the airfoils, and subsequent serrations application eliminated these tones by changing the character of the wake vortex.

Another example [7] presents a study of flap edge noise looking specifically to the single component and testing at a facility called the Quiet Flow Facility (QFF) designed specifically for anechoic acoustic testing in an open jet environment. The facility allows for speeds of up to 0.17 mach but limitations are evident when the facility only has a 2 by 3 ft open jet nozzle. Usefully the paper shows the amount of data that can be expected from such acoustic testing with an array of 33 B&K microphones (and 2 reference microphones). Results for this paper are supported by CFD modelling and the use of RANS modelling as such this paper is useful as it describes how modelling can be used to support real world wind tunnel testing and seeks to find correlation between the theoretical and the experimental. T. F Brooks excels at studying individual components of the airframe and correlating noise to surface pressure results, this allows T. F. Brooks to produce semi-empirical predictions and new flow and boundary layer calculations. Brooks continues this trend with further studies with directional arrays in [8 and 10].

The majority of these papers conclude with a handful of statements which provide insight into the workings of wind tunnel noise on one object or specific set. By contrast [4] is an example of a study on the entire aircraft, specifically Boeing measuring their new 777-300ER as it flew over a very large phased array of more than 600 microphones. The article proves interesting as it lists all the hardware employed and methodologies executed in timing and synchronising that many microphones for relatively short fly over times.

The area of most interest for this study is the second set of papers which look at closed section wind tunnel experiments that have been accomplished. The best example of which is [13]; an excellent paper looking at a specially designed facility: the full scale aeroacoustic AUDI wind tunnel. A fully operational wind tunnel which produces low background noise of only 60 dB(A) at a wind tunnel speed off 160 km/hr, with a capability of a maximum speed of 300 km/hr. the paper gives a project background and an overview of the entire facility with detailed description of the key technical components. Furthermore direct comparisons are made to 10 other European wind tunnels and excellent diagrams and detail of the specially designed rounded off inlets shows the level of the detail that was employed in the design and construction of the wind tunnel. Details such as the turning corner designed to absorb background noise. Given that the facility went into operation in 1999 it can be clearly deduced that the technology for quiet facilities are available and the current challenge is the converting of a non-quiet facility to perform as well as quiet facilities such as the AUDI wind tunnel.

An example of non-quiet closed section wind tunnel acoustic testing is the paper entitled "Methods to Measure Acoustic Sources in a closed Wind Tunnel" [16] this is a document that matches almost precisely what is required from this research. The study Cross Spectral Matrices (CSM) are described, the same as the methods employed by the Matlab code used in this study. The paper also show that in a closed section environment there is a definite need to reduce boundary layer noise. The paper shows that acoustic sources could be identified more clearly using CSM rather than the other methods described (conventional phased array method - summation method) and as such it verifies the use of the CSM methodology in this project.

The paper [16] also looks at sound to noise ratios making appropriate comparisons at speeds up to 300 Km/hr. The study is conducted using an array of 64 microphones as the University of Southampton did similar studies using an array of 56 microphones. Indeed two arrays are described a large array of 1 m radius that is identified to be effective over the frequency range of 630 to 2000 Hz providing better resolution but not so good with high frequencies at high velocities, for which a smaller array of 300 mm radius was used. The microphones in the arrays are flush mounted and covered with thin film of polypropylene of 16 micro metres thick. Given the publication date of 2005 it is clear that level of research is equivalent to that being conducted at the University of Southampton.

This paper [16] at first glance would appear to be the most relevant of all the publications with respects to the overall aim of this report. It is a study into acoustic measurements in a closed section wind tunnel, with real world application into locating the noise source on a real world transport. The only difference is their focus on trains rather than aircraft, which in itself offers entirely different problems and solutions. Despite the obvious differences there is sufficient in common to validate the use of the methods employed in this study.

Another good example of a paper from the second set of publications (closed section wind tunnels) is; “Airframe noise study of CRJ-700 Aircraft Model in the NASA AMES 7- by 10- Foot Wind Tunnel No. 1” [18]. This is an account of an acoustic study of a 7% scale Bombardier CRJ-700 in a 7 by 10 foot wind tunnel. This is a wind tunnel of equivalent size and as mentioned in the study the test section background noise was recorded between 80 and 90 dB between 63 Hz and 10 kHz, values which are similarly recorded in acoustic tests performed in this study. The paper exactly shows what is required from the tests and experiments are run. Speeds of 0.22 mach and 0.26 mach were utilised in the tests and the study revealed 10 potential sources for sound on the aircraft’s airframe, form which 5 major sources were identified as the slat gap, the main gear, the flap tips, the flap gap and the slat inboard tip. The paper concludes by showing the levels relative to each other shown. The paper also mentions model modifications and noise alleviation devices were employed but they are not described in great detail and it feels as if too much detail was omitted. Although an excellent paper in places and certainly in relevance, it is ultimately limited by there being no extrapolation to full scale model or the publication of any further study. Although the paper is significantly relevant, in that it shows where the relatively ‘loud’ parts can be located on an aircraft model, whilst offering another view on the research field, where the other papers focus on the theory or the mathematics this paper concentrates entirely on an actual experimental research; other papers which follow this arrangement include [7, 13, 14 and 17] These papers have little theory; rather they centre on actual physical work. The papers document what was achieved instead of what could or might be done. It is composed of recorded results and proven methodologies. These papers are a great example of an experimental environment and of the state of the research in the ‘experimental’ and application stage.

The paper, "Airframe Noise Study of a CRJ-700 Aircraft Model…" [18] is in essence such a laboratory report, it shows direct application of current technology to gain results. This paper applies some of the theories discussed in previous papers and performs a series of experiments to obtain a solution for a specific wind tunnel model. Since the paper focuses on only one model (the CRJ-700 aircraft) its conclusions are limited. Nonetheless, the paper proves useful as it describes in adequate detail how the required results were acquired. In this way it would be possible to use the paper to perform similar analysis of other models. Other papers which are similar in this respect, focusing on a single model or experimental application are references [1 - 4, 7 - 9, 11, 12, 14, 15 and 16]. The other papers [5, 6, 10, 13, 17 and 19 - 23] study methodologies for performing aeroacoustic wind tunnel tests in the respective wind tunnel environment.

Reference [19] is interesting as it performs a test in a wind tunnel environment comparing the noise with and without model; in addition this document serves as an excellent starter paper breaking the basics down for an easy introduction to Beamforming; a method used to perform acoustic measurements and reduce the effect of background noise. Indeed this description of beamforming and its application make this paper extremely relevant and useful, as beamforming is used throughout the study.

References [12 and 21] are other interesting papers as they look at acoustic liners to reduce background noise and at surface treatment to reduce microphone self-noise. Both these papers seek to improve noisy environments for acoustic testing. The effect of acoustic liners was recorded to be a reduction of 1.5 to 3 dB. Both these papers are relevant to this study as they describe methods which could be employed for acoustically improving wind tunnel environments.

The academic paper [24] shows the work done by Benjamin Fenech and can be considered as a precursor to the work being undertaken here in this study. This report shows that which is required to assemble and how it can be achieved, it takes theories and shows how they can be made reality. Although arrays of only 56 microphones were ever achieved it is an excellent start to the entire aeroacoustic measurement field in close section wind tunnels.

From the third set of papers it is quite evident that the “Aeroacoustic Measurements” book [24] has had a major impact on the subject as many of the authors within the book re-appear in subsequent years with further writings and a multitude of other publications. Further research into other papers in the field show that the authors involved in the “Aeroacoustic Measurements” book are indeed regarded as ‘experts’, and their credentials are well earned with many years of published research. A good example is Paul T. Soderman who was involved in 5 of the publications in this review. Soderman is generally involved in experimental work and his influence and papers are amongst the more relevant and useful. Specifically [24] is a look at what the current level of research in this particular field was, as it documents all the findings in the field up to the year 2002 when it was published, and although somewhat dated by a few years it is still regarded as the foremost research in the field as well as providing a good basis for any continuing investigation. It is composed by many authors, all regarded as experts, each offering a unique viewpoint into the research area. This book forms the basis for much of the theory employed in the software application. ‘Aeroacoustic Measurements’ allows past accomplishments to be replicated and the point of the book is to give a broad overview of the aeroacoustic subject allowing for the book to be used as a reference point in finding other sources for further examination into a specific area of the acoustic research field.

Reference [29] outlays the importance of aircraft noise and shows location of noise on various parts of the airframe. The discussion does a good job of explaining how air travel growth is limited by noise and how the current program has achieved only half (5 dB) of the noise reduction goal. The reference also shows levels of noise expected from aircraft on approach and identifies the aim of a 10 dB nose reduction on aircraft airframes by 2020. Paper [30] looks at measurements made on turbine rotor blades of a wind turbine, where the blades were treated with different surfaces: one left rough, one having a tripped flow and one turbine blade cleaned. An acoustic array was used to measure the noise at very low speeds of 6 to 10 m/s, this in itself makes the paper not terribly relevant but it does brings up some good points such as the aerodynamically clean turbine was found to be substantially quieter up to 6 dB quieter. This leads on to keeping surfaces inside wind tunnel aerodynamically clean, which might not always be possible but is something which should be kept in mind.

Although [32] is a study not performed in a wind tunnel the paper looks at algorithms applied to sound source application in a noisy environment, it is interesting as it makes claims to have the largest microphone array of 1020 sensors. This study also utilises beamforming with a good description of the technique and a number of other methods which can lead to techniques to improve wind tunnel application by introducing software that can implement some of the methods described. The paper main relevance though is the improvement of signal to noise ratio, in that it is improved from 17.2 dB with one microphone to 30.9 dB with 1020 microphones. The number of microphones also increases the accuracy and improves the error rates achieved; 10% accuracy with one microphone is raised to above 50% with 60 microphones, with accuracies of 90% with all 1020 microphones.

Despite using the simplest form of beamforming (a delay and sum method) the paper shows that the number of microphones has a distinct impact on acoustic acquisition. Signal to noise ratios and accuracy levels for 23 array configurations from 1 to 1020 microphones were measured and recorded and a clear improvement over both was seen as a steady improvement. However, there is a concern raised in the conclusions with the comparisons to past work being vastly different. These differences are attributed to different experimental conditions and methods, a point emphasised by the example of temperature in that it can have an impact with a change in 5 degrees Celsius changing the speed of sound by 3 m/s which affects calculations and results.

The fourth and final publication set of papers include the paper [34], “MEMS-Based Acoustic Array technology”, which look at a specific item, concentrating on a specific piece of equipment. It is designed to present to the reader an analysis of the ‘novel’ technology; in this case it describes how a Micro-electromechanical system (MEMS)-based directional array could be constructed and then implemented. It is in essence a construction and user manual for such a mechanism. This new technology is described in heavy detail and its potential is explored along with how it could be used and implemented. This paper is fairly unique as it describes a specific component in theory but does not provide experimental support unlike papers [2, 8, 9, 12 and 21] which are also papers which focus on one item (or aspect) of aeroacoustic measurement, but also include experimental study to support the research.

A number of papers, which looked at newer technology, were also reviewed in detail for this study. The last set of papers, studied in this review, looks at some specific theories and provides possibilities for further research. Some of the papers tackle different specific problems but they are all brought together by a common theme, namely that of wind tunnel noise, and each offer alternative solutions or possible avenues of continued research [34, 37, 40, 41, 43, 44, 45, 46, 47, 51, 52 and 54]. Brooks appears again working with a new system called DAMAS (Deconvolution approach for the mapping of acoustic sources) in [31 and 44], whilst reference [37] gives a complete tutorial on microphone arrays, describing the basic fundamentals, from sound wave propagation to beamforming techniques and the application of beamforming. A number of beamforming techniques are described in full and their advantages and disadvantages are summarised, this is exceedingly useful when attempting to implement acoustic beamforming from scratch.

Reference [38] is another paper looking at microphone calibration, it describes methods for calculating the channel's gains to within +/- 0.45 dB and involves tests done in a real environment, although the tests not done in a wind tunnel, and so this reference is of limited use in this study. Reference [40] is another paper looking at microphone positioning and gain measurements and calibration; it describes positioning techniques using acoustic techniques, where a series of small speakers are used to produce pulses from which the delay can be used to determine the sensor location. The paper determines that individual channel sensitivity can vary up to 6 dB and proceeds to give a description of new techniques and apparatus for automatic calibration. This papers describes accuracy of gains can be found within 1-2dB and shows potential for fast automatic gain and position calibration which may prove useful in future array implementation.

[43] Refers to another paper on DAMAS describing it as an algorithm for phased array imaging that potentially eliminates side lobes and array resolution effects. Although interesting it is limited in aeroacoustics due to restrictions which assumed in optics are not so good for acoustics. This paper is a very complex paper very mathematically minded and not very easily approached. The implementation of DAMAS also requires the use and design of specialist array design

Reference [45] looks at a mathematical method for improving the results gained from an acoustic array. It shows a method for determining the strength of the acoustic source by inverting the matrix of frequency responses and is another complex, difficult to approach, mathematical paper. Composed almost entirely of calculations and theories these papers shows in intricate detail the mathematics involved in such studies. For example paper [46] looks at a mathematical technique, developed many years ago, known as Classical Multidimensional Scaling (MDS) used to calibrate microphone by finding the sensor co-ordinate points using non-iterative methods and develops it, in to Basis-Point MDS for use with microphone arrays, the paper shows the level of accuracy that can be expected from such calibrations to be 10-20 mm. Papers such as [47] look at the progress of sensor array signal processing; it shows how mathematics and the creation of new faster algorithms have developed the efficiency of such arrays. The paper also names a number of mathematical techniques employed in effective signal processing and how some of the techniques are linked and have developed from each other and the latest techniques on which research is still being done. These latest papers show with determined focus the mathematics behind acoustic source location and in this review tends to be on average the most recent papers to be released. It is a reflection on the current state of research in the world today. The level of research has jumped considerably since 2002 and as such a large number of the 'theoretical' papers looking at possible tangents or alternatives to older (relatively) technologies.

This set of publications included in this review provides a basis for the state-of-the-art as it was in 2006. Although given current trends a subsequent amount of interest has been sparked in this area in recent years and it is expected that there will be more research in the future. However the sources of this continue research will almost certainly be from the same people in industry and facilities, this is to be expected as there is a limited number of working wind tunnels around the world and a limited number of experts in the relevant aeroacoustic field.

With regards for the state–of-the–art and expert opinion, the 'Aeroacoustic Measurement' book [24] reflects the state of the art the up to the minute (as it was then in 2002) situation that existed within the area of aeroacoustic research. This publication takes the form of a book and as with most engineering books it is concerned with facts and figures. The book actually makes a point of advertising that it contains these figures, pictures and photographs as a way of identifying itself as a relevant and realistic application of the research. As such there is little in the way of theory and speculation in the book. However, the Aeroacoustic Measurements book is inherently approachable and offers an excellent path in to the research field. The book is written in such a way that it can be understood easily and efficiently. Although there are complex mathematical sections in the book it is laid out in such a way that it does not deter the reader from pursuing further study. At the time of printing (2002) this book reflected what was considered the most recent and innovative methods of aeroacoustic measurements.

The theoretical publications, form the later sets, those which focus on only one aspect of the research area are directly concentrating on theory and mathematics. These papers are so focussed on the mathematics that at times they can be quite difficult to follow. The papers are sometimes written in a complex fashion and require a lot of previous engineering and mathematical knowledge to be clearly understood.

The other publications, from the first sets are, by contrast to the other papers, totally routed in the experimental frame of mind. The paper is laid out with an abundance of photographs from the experiments and diagrams showing methodologies and set-ups. As a result of this abundance of technical information these papers are very easy to follow and work very well as a guide or experiment manual. These papers [1-23] are more useful when trying to replicate acoustic research, although by necessity some of the theoretical papers are necessary to understand the principles behind acoustic measurement.

The most useful of papers are those papers, like [1, 2, 11, 12, 13, 14, 15, 16 and 24], which focuses on the mathematics and theory initially, then continues onto more practical applications; taking on a form that more resembles that of the experimental papers, these publications describes how to apply the theory, mentioned in their early parts, into practice. With a combination of mathematics and then clear diagrams showing how physical applications of the mathematics can be constructed these papers shows the clear connection between the theory and the applied. The best example of such a paper is [15] a paper describing an aeroacoustic test of a 26% scale model of the Boeing 777 commercial airliner. The paper refers to conditions almost identical to the ones that are the challenge of this study. The 26% scale model acoustic test was performed using 70 microphones in a 244 cm array in a frequency range of 1 to 25 kHz, all values that are similar, if not quite exact, to the conditions of the study to be done at the University of Southampton and at Filton. The paper also describes recessing techniques and their usefulness in reducing background noise by up to 20 dB, but also raises the added problem of accounting for resonance in the recessed gap. This paper is most relevant in the environment, the item being studied and the techniques employed. It is an excellent paper, with high levels of detail, from array design and justification to simulation results to test the array design.

It is this paper which forms the basis for the study conducted in the second year, section 4.2; where recessing techniques were tested with a view to prove these advantages and to quantify the level (and type) of recessing required to maximise or at least match the results quoted in [15].

Papers which remain focused on one aspect, as mentioned before, either in theory or indeed on one aspect of an experiment can be surprisingly limited, but can still prove useful as guides to specific problems or solutions to specific problems.

These publications show quite neatly the need for balance between theory and practical application, especially when replication or further research is required. Papers which combine both these aspects of engineering research are extremely useful and when they are recent to the point of showing the current state of knowledge they prove to be invaluable.

An excellent recent paper [55] is one of the few papers to directly compare Open Jet acoustic array technique and results with closed test section methods. The paper is a good source of validating the Beamforming methods and looking at absolute level acquisition techniques. The paper is co-written by Pieter Sijtsma, whose other work [56] describes the CLEAN algorithm, forming the basis for a major processing modification made to the Beamforming code which when implemented can give superior results from the same data. Further papers made available in the second year, continue to improve on the processing code include papers [57, 58 and 59]; with paper

[57] suggesting that higher-frequency sources are potentially easier to locate. This is a potential area of future study as higher frequency acoustics is more important in aero-acoustics given the scale and size of the models used in wind tunnel testing.

Several internal reports from the University of Southampton, [60, 61 and 62] are also referenced here as they catalogue the processes and basic information that occur within the test regime of an academic facility. They prove useful too as they invaluable in referring to practices that were attempted and listing the techniques that have been mastered in time.

A special mention must be made of paper [58] which makes an analysis of the use of the cheaper electret microphones (similar to those used in the majority of the experiments in this collection of studies) and concludes that the microphones are effective between 250 Hz and 40 kHz, which coincides with the range of frequencies of interest and support the finding made by the internal University of Southampton papers [60 and 61]. Although further study into the use of these cost-effective microphones will be undertaken in the future these papers support the use of them in wind tunnel array testing and validates their use for the time being.

Paper [64] look at developing a 3D acoustic array, this opens up a whole new area of research into the potential of 3D acoustic array technology, which could potentially be an expansion of the current array technology employed. Further investigation is likely beneficial to future array design as requirement grow.

Characteristically a lot of the newest papers are written by well-known authors and aero-acousticians, and they often are a continuation of past experiments, which is to be expected, names such as Pieter Sijstsma, and Paul T. Soderman are synonymous with aero-acoustics, and especially in the field of array measurements in wind tunnels. Their experience is significantly noteworthy and forms a substantial basis for the use of array technology. New avenues of research will need to be explored over the next year as emphasis will move away from acoustic arrays to other wind tunnel technology, but the experience and techniques used in acoustic data gathering can be used in other data gathering and many of the papers listed above will remain relevant and useful.

2.3. Literature Review Summary Concluding Comments

The majority of the references rely heavily on previous mathematical or acoustic knowledge, in most cases at the mathematics behind acoustics and acoustic measurement; as such these publications are aimed at those people who are already knowledgeable in the field. These papers can be less useful to non-acoustic engineers attempting to integrate acoustic technology into established

Discussing the path into the future with regards to all the papers it can be seen that there is a plethora of avenues open to research in this area of study. It is also apparent that there is a growing interest in the continued success in the area of accurate acoustic source location. The growing number of papers in the field in the last few years is indicative of genuine interest in acoustic array technology and application.

Books such as the Aeroacoustic Measurements [24] offer a collection of current knowledge and a wide perspective into the research subject. However, books can be outdated (with respects to current ideas) and as such they are best used as a starting point or a reference point leading to more up to date discoveries. It is the ideal introductory publication into the research field. The Books such as the one analysed in this review will prove invaluable as a resource for proven theories as well as a guide to finding other relevant research publications. Through the book, several specific authors are readily identified as specialists in the field.

In contrast the papers do not provide such a broad prospective of the subject; rather they focus on specific areas within the field. They indicate that whilst something is being done in the field of acoustic source location, it is limited and still in its infancy; the sheer number of theoretical papers compared to experimental papers is indicative of this. It is also clear that the technology required for such acoustic measurement is still in early experimental stages, the sheer volume of theories and mathematical modelling shows that there is great potential for further practical research.

This is not to say that there is a lack of experimental results, a small number (due to limited number of appropriate facilities worldwide) have provided some of the most relevant and useful papers cited in this literature review.

From this review it can be deduced that there are a variety of ways to both approach the research and to display the appropriate results. From a wide, all-encompassing gathering of information such as some of the papers and documents in the third and fourth sets of publications, to the investigation of a specific area, such as the focused theoretical papers and the experimental papers in the first and second sets,

Considering the amount of research completed in comparison to the time scale, the earliest paper being from the early 1990's, it is evident that this area of study is still relatively new and there is indeed scope to continue studies into acoustic source location in winds tunnel tests. Thus there is the option and opportunity to take the research into new areas and levels of sophistication and complexity, as has already begun with new research into alternative microphone set-ups [34, 43, 44, 45 and 47].

The newest papers and references included in this report incorporate a number of papers written by the personnel of University of Southampton, as well as a series of reports written to record and plot the course of the aero-acoustic research undertaken to date. The trend of the latest papers is to take Beamforming and alter or study one aspect, such as [55], which looks at Open Jet vs. Closed section, the most useful feature of the latest papers are those which explain clearly what has been done to the point where it could be repeated and validated as is often necessary. It is papers such as [67] and [68] which go on to validate the results from such studies. Additionally the number of papers in the respective sections from this selection gives an indication of how much of this research is still in the development stage. There are far more papers in the theory selection than there are in the experimental stage. This is not to say that a good level of experimentation and practical application has not already been carried out but it does show the relative 'young' age of the techniques and theories. Despite the attempts to categorise the papers into these four sets it should be noted that this is an oversimplification and that many of the papers cover theory as well as experimentation, nonetheless the theoretical papers whilst they are highly detailed are also limited in the scope of their information and it is found that the experimental papers are more appropriate to this study and the information that is required for successful execution of a new microphone array in an existing wind tunnel can be found within them, allowing a reasonable understanding of previous work in the area to be realised to an excellent extent.

2.4. Beamforming Equations

This section is aimed at providing a starting point for beamforming by providing the initial equations that are used in the Matlab code as described in the code description section 3.2.1. (Page 37) in this report. All the equations were acquired from references [19, 24, 27 and 32].

Beamforming is a signal processing technique used with arrays of transmitters or receivers that controls the directionality of, or sensitivity to, a radiation pattern. When receiving a signal, beamforming can increase the gain in the direction of wanted signals and decrease the gain in the direction of interference and noise. This is ideal for attempting to localize a noise source. Information from different sensors is combined in such a way that the expected pattern of radiated sound (in acoustics' case) is formed to be that which is expected. The beamforming response is plotted as a contour map and maximums are acquired where the focus of the beamformer coincides with source location. Broadly speaking there are two types of Beamforming:

- Conventional or fixed beamformers
- Adaptive beamformers

Conventional beamformers use a fixed set of weightings and time-delays to combine the signals from the sensors in the array, primarily using only information about the location of the sensors in space and the wave directions of interest. Adaptive beamforming techniques, however, combine the information from conventional beamforming with properties of the signals actually received by the array, typically to ascertain unwanted signals from the other directions, and so an adaptive beamformer is able to automatically adapt its response to different situations. The following equations describe conventional beamforming for which the simplest beamformer is known as a delay-and-sum beamformer. This is beamforming in the time domain where all the weights of the elements have equal magnitudes. The beamformer is steered to a specified direction only by selecting appropriate phases.

The beamformers output signal in the case of time domain is:

$$z(t) \equiv \sum_{m=1}^{M} \omega_m p_m (t - \Delta_m)$$

Beamforming in the frequency domain is more relevant to acoustic research and allows the use of several techniques for reducing sidelobes, narrowing the mainlobe and reducing noise and reflection effects. In this case the equation above becomes:

$$z(\omega) \equiv \sum_{m=1}^{M} \omega_m p_m (\omega) e^{-i\omega\Delta_m}$$

Applying a Fourier transform of z(t) and p(t) gives z(w) and p(w), performing a short time Fourier analysis with a time window of duration D gives:

$$p_m(t,\omega) = \int_{t}^{t+D} W(\tau - t) p_m(\tau) e^{-i\omega t} d\tau$$

From this the beamforming output can be determined to be:

$$z(t,\omega) \equiv \sum_{m=1}^{M} \omega_m p_m (t,\omega) e^{-i\omega\Delta_m} e^{-i\omega\Delta_m}$$

Applying beamforming to a spherical wave the pressure field induced by a source at any position $\vec{x}$ is governed by the outward propagating solution to the wave equation:

$$p(\vec{x},t) = \frac{\sigma(t - \Delta_t)}{4\pi \left\| \vec{x} - \vec{x}_o \right\|}$$

In the frequency domain a spherical wave in unbound free space, as above, becomes:

$$p(\vec{x}, f) = \frac{a(f)e^{-2\pi i f \Delta t_0}}{4\pi\tau_0} = a(f)g(\vec{x}, f)$$

Where $g(\vec{x}, f)$ is the steering function also known as the Green's Function which for a point source is given by the inhomogeneous Helmholtz equation:

$$\nabla^2 g + (\frac{2\pi f}{c})^2 g = -d(\vec{x} - \vec{x}_0)$$

In a wind tunnel with uniform velocity $\vec{U}$ the above solution becomes:

$$\nabla^2 g - \frac{1}{c^2}(\vec{U} \bullet \nabla + i2\pi f)^2 g = -d(\vec{x} - \vec{x}_0)$$

Where the solution is given by:

$$g(\vec{x}, f) = \frac{e^{-2\pi i f \Delta t_0}}{4\pi\sqrt{(\vec{M} \bullet (\vec{x} - \vec{x}_0))^2 + \beta^2 \left\| \vec{x} - \vec{x}_0 \right\|^2}}$$

For which $\vec{M} = \vec{U}/c$ is the mach number of the flow and $\beta = \sqrt{1 - \left\| \vec{M} \right\|^2}$ is the Glauert compressibility factor.

Complex sources can also be solved by using derivatives of the Dirac – Delta function in the equation on the previous page.

Finally to take into account the number of sensors is ever changing the spherical wave in unbound space equation (on the previous page) can be written in vector notation. Giving a general beamforming equation:

$$\hat{a} = \frac{g^{+}p}{\|g\|^{2}} = \frac{1}{\|g\|^{2}} \sum_{m=1}^{M} g^{*}{}_{m} p_{m}$$

Where P is the pressure vector and g is the steering vector, g^{+} is the Hermitian conjugate (complex conjugate transpose) and * is the complex conjugate.

Adaptive beamforming, when applied appropriately gives large improvements in resolution and gain. For adaptive beamforming the weight vectors are not all equal and the general beamforming equation is rewritten with appropriately chosen weighting vectors.

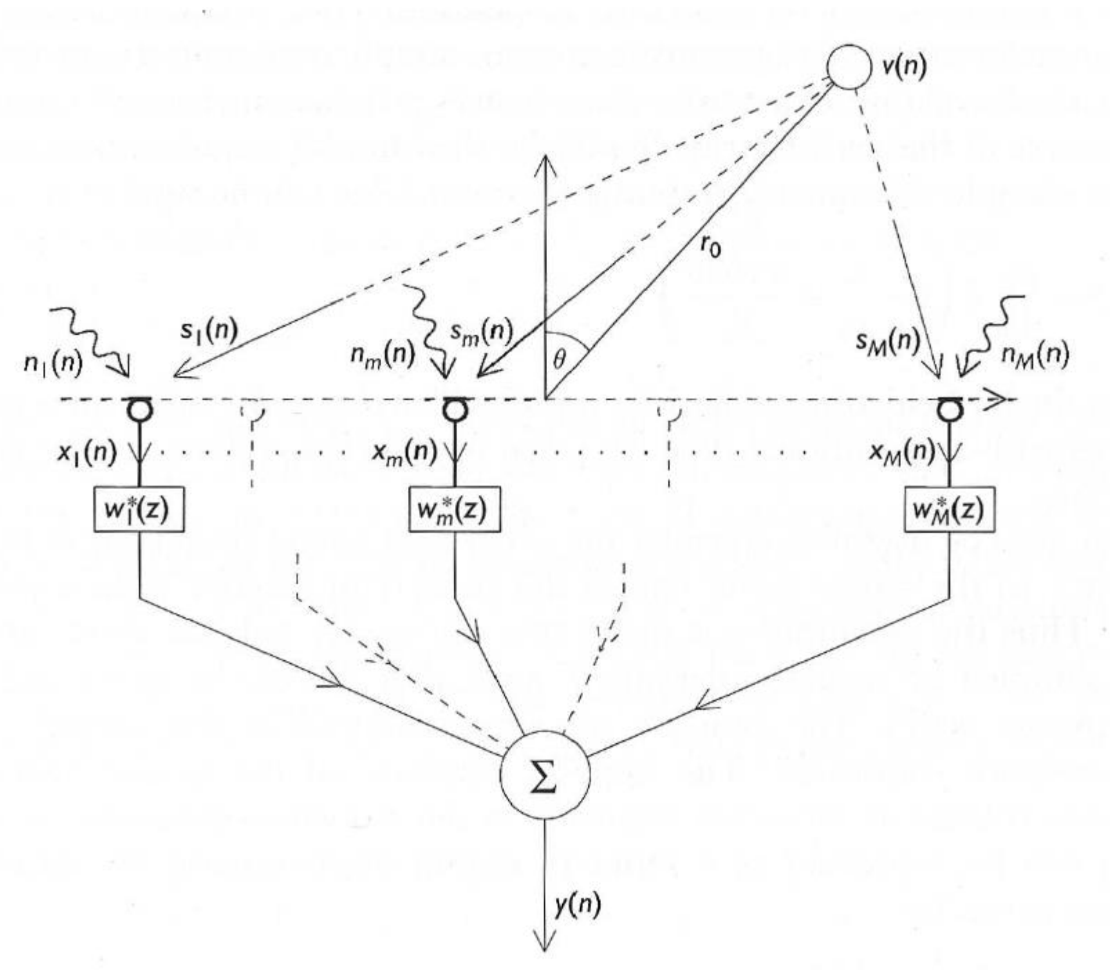


Figure 1 - Diagram showing an array of sensors and a single source. The signals to the sensors consist of the sound from the source and noise. Ref. [69]

Figure 1 on the previous page shows a single point source emitting a sound which propagates to an array of microphones. The signal received at the sensors consists of sound from the source as well as noise. The outputs from the sensors are passed through digital filters to give the beamformer output.

Focused beamforming is based on two planes; the array plane and an imaginary scan plane which is defined at a distance at which the source is expected. In the case of a test in the wind tunnel the scan plane would be defined at the location of the model. The beamformer's output is calculated at a number of focus points on this imaginary scan plane, which is subdivided into a number of grid points. The beamformer's output is at its greatest for the grid point closest to the source position. Figure 2 below depicts these two planes, which are implemented in the Matlab code as defined in section 3.2.1.

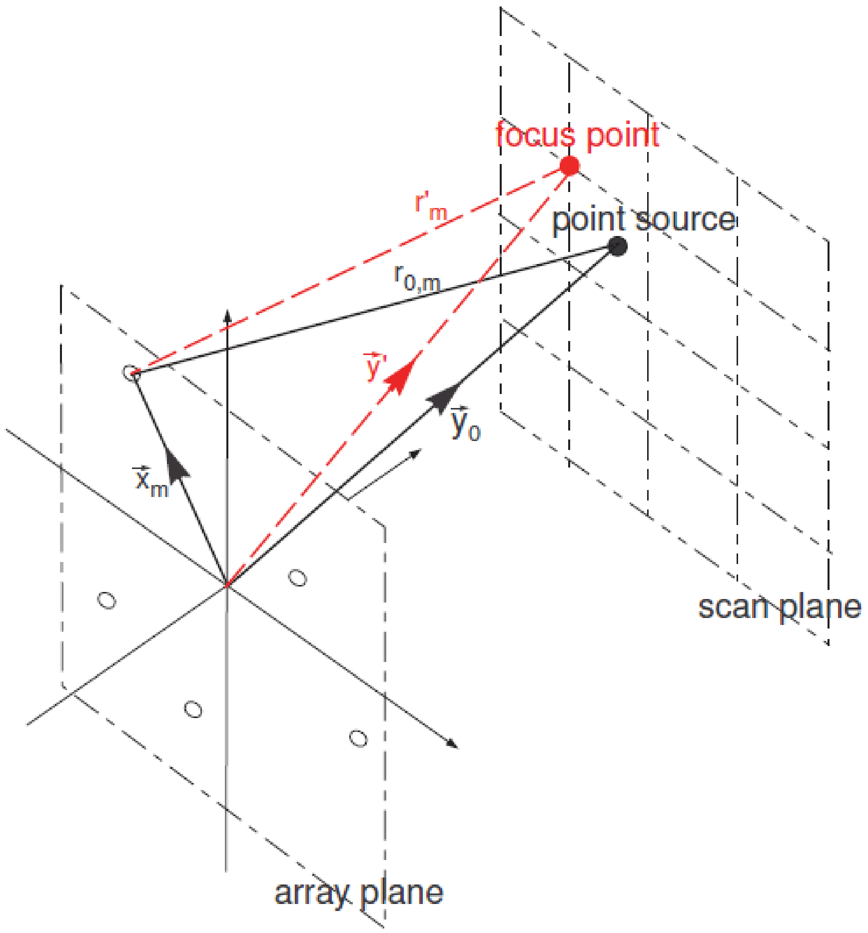


Figure 2 – The focused beamformer is based on two planes, the array plane and an imaginary scan plane defined at the area of interest. Ref. [70]

2.5. General Assumptions and limitations

In this section some general assumptions and clarifications will be made. There are a number of assumptions with the beamforming technique which should be recognised and acknowledged. Beamforming is inherently limited by poor resolution at low frequencies and spatial noise at high frequencies; deconvolution techniques (such as DAMAS [31]) were considered and CLEAN-SC [56] was applied to counter these issues during the second and third years of this study. These post-processing techniques were subsequently applied to data acquired in the earlier tests. Further information on these techniques can be found in section 4.1.1. For closed section wind tunnels two other concerns are raised; the noise due to the turbulent boundary layer and the effects of reverberation on the source location measurements. Reflections in hard walled test sections can lead to additional image sources being generated and an energy build up that can interfere with source strength estimates. In real world wind tunnel tests aeroacoustic sources can occur close to hard surfaces and reflections off wind tunnel boundaries are normally coherent with the source. These factors can cause an error in beamformer output as the example in Holland and Nelson [41] where a single source under semi-reverberant conditions represented by free-space Green's functions led to significant errors compared to 'de-reverberated' results using measured Green's functions. In Aeroacoustic Measurements (ref [25]) the effect of wind tunnel boundaries are described using the concept of image sources. The test section of the wind tunnel is effectively a duct and the sound field within will consist of a sum of the true source and any resulting images sources.

The noise due to turbulent boundary layer interactions with the flush-mounted array is accommodated by removing the cross-spectral matrix (CSM) in the post-processing Matlab code employed as described in section 3.2.1. This technique referred to as diagonal removal (as it involves the removing the diagonal of the CSM) can completely remove noise influence without compromising the capabilities of the array. This is on the proviso that the noise is uncorrelated over the array microphones. With this assumption the noise will average to zero in the CSM except on the leading diagonal. The diagonal removal method is not without its own issues, however, as this technique can cause non-physical negative source powers to appear in the beamforming plots. The other concern of the highly reverberant nature of a typical hard-walled closed sections wind tunnel is dealt with by the work done by Guidati et al [71], Sijtsma and Holthusen [72] and Fenech [70], here the de-reverberation method Image Source Model (ISM) where the enclosure boundaries are replaced by an infinite set of mirror image sources is the most relevant for high frequency studies. It should be noted that de-reverberation techniques can cause distorted array patterns (point spread functions, see section 3.4) which can cause its own problems with source identification.

3. Year 1 - Methodology and Testing

The following section illustrates the work completed in the first 12 months of this study into acoustic wind tunnel technology. Section 4 looks at works completed in the main study of the second year. Whilst section 5 looks at the studies done in the final years of this thesis. The first subsection here looks specifically at the wind tunnel testing performed and the essential skills gained in performing them.

3.1. Wind Tunnel Testing

The first series of tests in the 7 by 5 wind tunnel consisted of a succession of landing gear experiments used to observe the array in operation. This test was primarily used as time to familiarise the use of array technology, and the set-up mechanism that was employed in mounting the array and acquiring the data. The array employed for these tests was the prototype for the 7 by 5 Southampton wind tunnel. An important aspect of microphone arrays was also looked into when Microphone Calibration was performed for a later experiment using the same 7 by 5 wind tunnel array. Further wind tunnel time was spent in the 7 by 5, in a series of further array experiment. For these later experiments the array was subjected to installation and dismantling; this was used as an opportunity to identify the individual array components and how the array was assembled. A number of installation problems were identified and rectified – such as the necessity for clear wiring labels and the need to bundle the cables for ease of assembly.

A New Array Set-up, for the University of Southampton's 11 by 8 wind tunnel, was installed and tested in a series of tests leading to more wind tunnel time this time in the 11 by 8; a considerable jump in size and a move which introduced new problems; the 11 by 8 wind tunnel has a different configuration. Microphone Calibration for this array was also performed and familiarisation with Labview program and code used to calibrate and operate the acquisition was achieved. The tests yielded a vast amount of data which required an extensive amount of post processing. This led to familiarisation with the Matlab processing code. Several weeks of post processing was achieved with data of well over 1000 data sets processed. In this time a number of modifications and improvements were made to the software. This is described in the next section.

3.2. Software Development

A major aspect of the Microphone Array System is the software used. The system employed in use this past year has been controlled and operated using two major pieces software. The first is the software used for data acquisition and calibration of the microphones, LabVIEW is the software supplied with the data acquisition cards and unit used in the system and as such is the software engaged in their use. The LabVIEW code acquires the data and allows variables such as the data block size and the sampling frequency to be set.

For microphone calibration a LabVIEW code is used to acquire and compare a signal from a test microphone and a reference microphone as well as record the signal received from a test microphone subjected to a pistonphone (a device which produces a set tone at a pre-selected frequency). This allows the operation of an individual microphone to be assessed and calibrated accordingly.

Matlab is the other major piece of software used in the microphone array system although this software is used primarily for the data processing. An extensive code is implemented to take the raw data acquired from the array and process it to give a visual representation of the sound recorded from the test.

Additional Matlab codes are used to simulate array conditions and part of the array design process makes use of specially designed Matlab codes. The array design and initial tests and checks are performed using Matlab code as Matlab is a convenient engine for performing quick calculations and providing visual images of the results.

The following section describing the Matlab code is only a description of the Matlab code used in the processing of the data acquired using the array. The majority of the software was initially designed, developed and written by Benjamin A. Fenech [24, 60 and 61].

3.2.1. Matlab Code Description

The next section depicts the Matlab software and how it performs the Beamforming process. The actual program consists of 1 overall run program, 1 main 'master' program and 15 sub-programs. The program produces and incorporates Cross Spectral Matrix (CSM) Calculation and Beamforming (BF) in one script.

The entire program begins by Specify Global Parameters; this is required for Matlab programs composed over a number of sub-programs. The run program begins with the User Inputs. Other types of inputs are then inputted later in the main program. Data inputs for each individual test can be entered in the overall run program

Data inputs for each individual test are also entered at this stage of the program; this includes items such as Wind Tunnel Temperature and Wind Tunnel Speed. Data inputs for each set

For the program to operate correctly the source data File names and locations are specified. The root directory is separately specified. Input variables are next loaded by the program, this includes loading the Microphone co-ordinates, and then loading the Microphones that are used by loading a text file that specifies the Microphone Numbers that were used for the specific test. Other values that must be specified for the program to operate correctly are the Sample Frequency used in the test being processed and the Block Size utilised in the data acquisition. These values are specified during data acquisition in the LabVIEW program.

An additional feature of the latest code introduces an Option to use a saved CSM. This allows the same source data to be processed over and over again under different conditions (such as alternate frequencies) without having to load and process the initial raw data.

The following part of the main program designates where the 'scan' plane (or the area of acoustic analysis) is set. This is done by fixing a reference point on the model with respect to array centre. If the model is rotating, the reference point must be equivalent to a centred pivot point. The Sub-function files 'gen_coor_grid' and 'save_and_plot' have to be modified to make sure that the scan plane is in the right place with every new test. This allows accurate locations of the sound source to be depicted in the plots.

The next selection for the user is the frequencies of interest. Here the frequency in which to perform the analysis can be set. The option exists to switch from one frequency to a selection of frequencies; thereby allowing either a quick analysis of one frequency or plots over a range of frequencies to be made.

The final set of user preferences is then specified, in this part of the code the user can select whether or not to have Frequency averaging as well as choosing the selection of file types to save. Image files can be saved in a number of formats. A switch allows easy selection and a 'Perform' vector is used to store all the choices.

Finally the code selects the output filename; the code is set currently to saving any output to the same root file. The code currently selects a Filename that is dependent on the input values, for Example: 'Trial 34, WT_Speed=0 m/s'. This is useful and with a lot of data being processed a necessary requirement for easy identification of results.

It should be noted that the choices made by the user up to this point affect the speed of code in processing. The more frequency plots requested the slower the entire processing. Asking for additional save files also slows the code down but not significantly so.

The next part of the code is used to describe the data acquisition cards and the microphones that were utilised in the data acquisition. The Microphone index is loaded from the file 'mic_used'. If a different set of microphones are used a different 'mic_used' file can be used. Correct number of microphones must be in the 'mic_used' file in relation to the time data received from testing, this is essential to provide accurate results.

After all the inputs and outputs have been specified by the code the main body the start of the main program begins with the number of microphones taken from the input files and loaded. The code checks for a previous CSM and if a previous CSM is not available then the requested data file is loaded. Currently the code is making a transition from the ASCII format of data to a Binary format. The binary format is faster but the LabVIEW code utilised outputted the raw data to ASCII format. At this time the transition from ASCII to Binary has only just been completed and extensive testing has not yet been performed to verify its full operation.

The code then continues by performing checks for consistency between number of microphones and the loaded data file. This next section only applies if the calibration was requested by the user (if requested in the perform command (number 2)). Here the calibration values are converted (Vpeak to Vrms). The loaded Time data is divided by the microphone sensitivity to give Corrected Time Data. Again it is necessary to have checks for the sizes of the data to be equivalent. Once the checks are complete the files are cleared. At this point one of the sub-codes is activated as it is designed to produce list of calibrated microphones. If the user has requested FRF calibration (again in the 'perform' function) the code loads file 'mic_numbers' and lists the calibration files of the microphones used. This is done by a sub-program. This is useful when large number of microphones will be used as it then becomes possible to select the required microphones.

The following section of the code builds the averaged, optimised Cross Spectrum Matrix from the raw time data. The Sequence in this section is thus:

- Full calibration made
- Code performs FFT (Fast Fourier Transforms) on the data.
- Code builds the cross spectrum matrix (CSM) and averages

The main code performs Discrete Fourier transform (DFT) which are computed with a fast Fourier transform (FFT) algorithm. The use of Fourier transforms is to find the frequency components of a signal buried in a noisy signal. The FFT part of the code uses the corrected time data as its input. At this point the use of a Window Type that was selected earlier is made. Here there is an Optional File to take the FFT data and save it for later analysis, this again can be used to save time in performing repeated analysis using the same data.

Optional decisions (again made earlier in the 'perform' vector include the ability to remove background CSM, optimise the diagonal and also to save the matrix (thus allowing it to be loaded later for faster processing of the same data). Removing the background CSM and optimising the diagonal are mathematical methods for improving the results. The background CSM is the CSM made from the data recorded in an empty wind tunnel; removing it theoretically removes the background noise. However this is not the case in reality, although some improvements in the results can be made. The code is designed to take background CSM and subtract from model CSM is required as well as removing or replacing the diagonal of CSM to optimise the CSM. If optimisation is performed the CSM is replaced by Optimised CSM so the later code calls the correct matrix.

The Main code for Building Cross Spectral Matrix requires the Frequency data as gained from the FFT analysis. Any averages are summed along the way. The cross power spectral density is the distribution of power per unit frequency. The code here is used to compute the Power Spectrum and Auto spectrum of the data.

The sub-programs at this point load microphone coordinates from 'Mics_coor_mat'. A vector of microphone coordinates is generated using the sub-program 'gen_coor_mic', whilst a vector of grid coordinates is generated using 'gen_coor_grid'.

The code then moves on to use the pre-calculated CSM to calculate the main beamforming expression. At this point the data is prepared for plotting. Edited weight vector calculation is done to avoid having to flip the plot matrices; this allows an off-centre grid plane to be defined directly. Here if the user requested it earlier there is an optional frequency averaging over 1/N octave band; averaging in frequency band by mean OR median methods. The code then generates an array of coordinates of sensors used, with the centre of the array as the origin. The code then generates the grid of the scan plane with the co-ordinates of each grid point.
The 'gen_coor_grid' sub-program code then corrects the coordinates to be relative to the array centre.

The main code proceeds to generate the required frequency values. A sub-code is used to define the centre frequencies of the bands, dependant on user input of averaging choice. A Sub-code is used to average the frequency if selected by user to be averaged. Another Sub-code is used to choose the FFT frequency nearest the frequency chosen by user. The main code then calculates the free-space Green's functions relating each grid point with each microphone.

Finally the main code performs the Beamforming expression. Weight vectors are produced and applied to the beamforming grid and all the calculations are applied to the CSM. The final output is 'BF_xpr'. The final few sub-programs are called to finalise the process of the data the Beamforming Frequency Averaging Program is called as a finalisation of the beamforming process.

Finally to produce a plot the clip plot sub-code is applied (to produce an adequate picture and the plot and save sub-function is run to produce the visual output. This is ends the Main Program.

There is a number of Sub Programs to clip the plot at a predefined level and the sub program that is designed to plot and save the required output as requested by the user using the ‘perform’ commands. The averaging process currently uses the Median averaging, but the Mean average process is easily available.

The final few sub-programs are used to determine the titles of the plots. The final image produced is that of the sound with respect to the sound in the area. Additional lines can be ‘drawn’ in to produce and overlay any wind tunnel model that was present.

Further information on the working of the beamforming code and the variables which can be set in the Matlab code can be found in a system guide written for the so-called SotonArray: Southampton University wind tunnel microphone array. [61]

The following section is a visual description of the Matlab code that has just been described in the form of a flowchart.

3.2.2. Visio Diagram

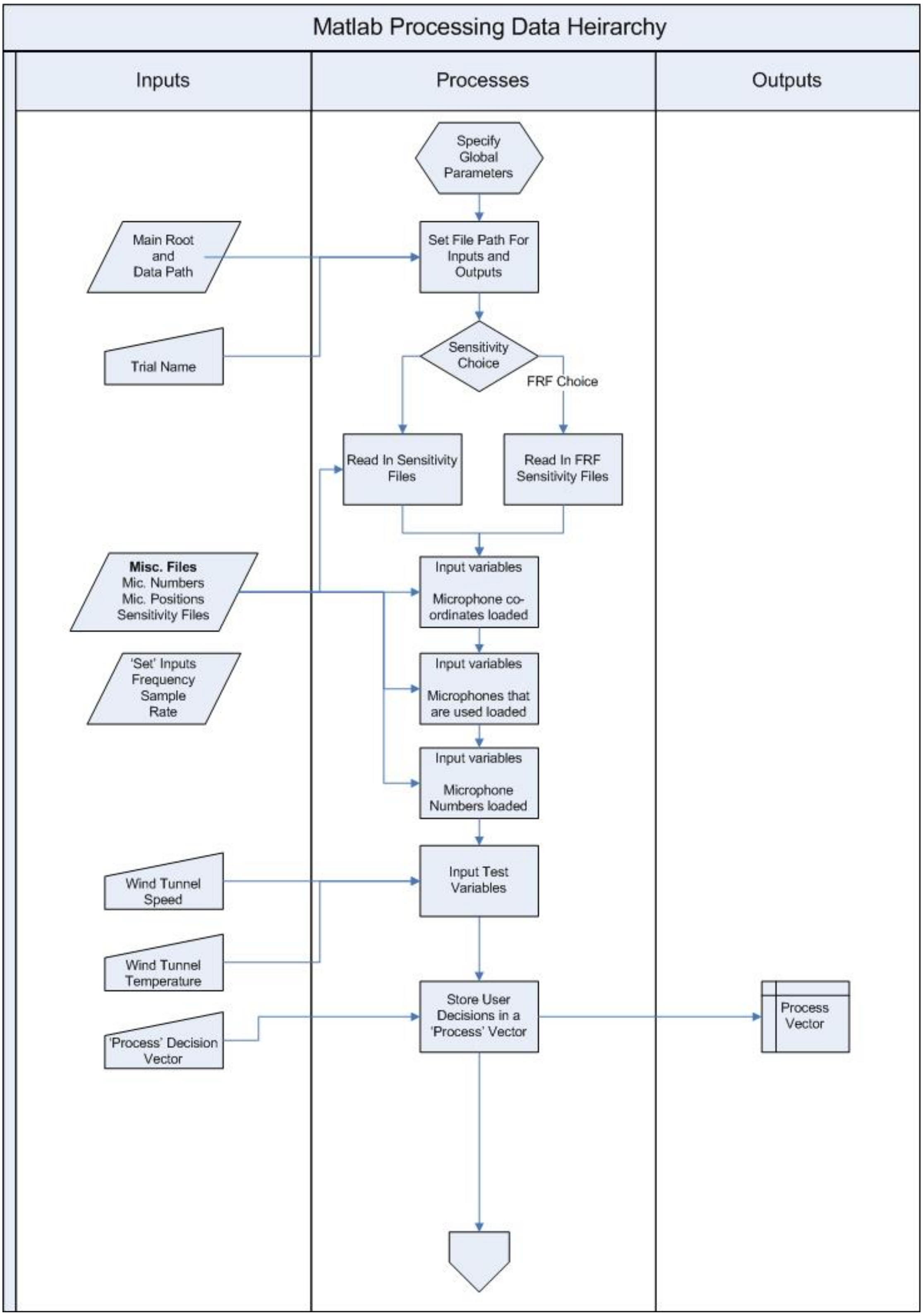


Figure 3.1 - Part 1 of the flowchart showing the Matlab code processing data from a microphone array during an aeroacoustic test.

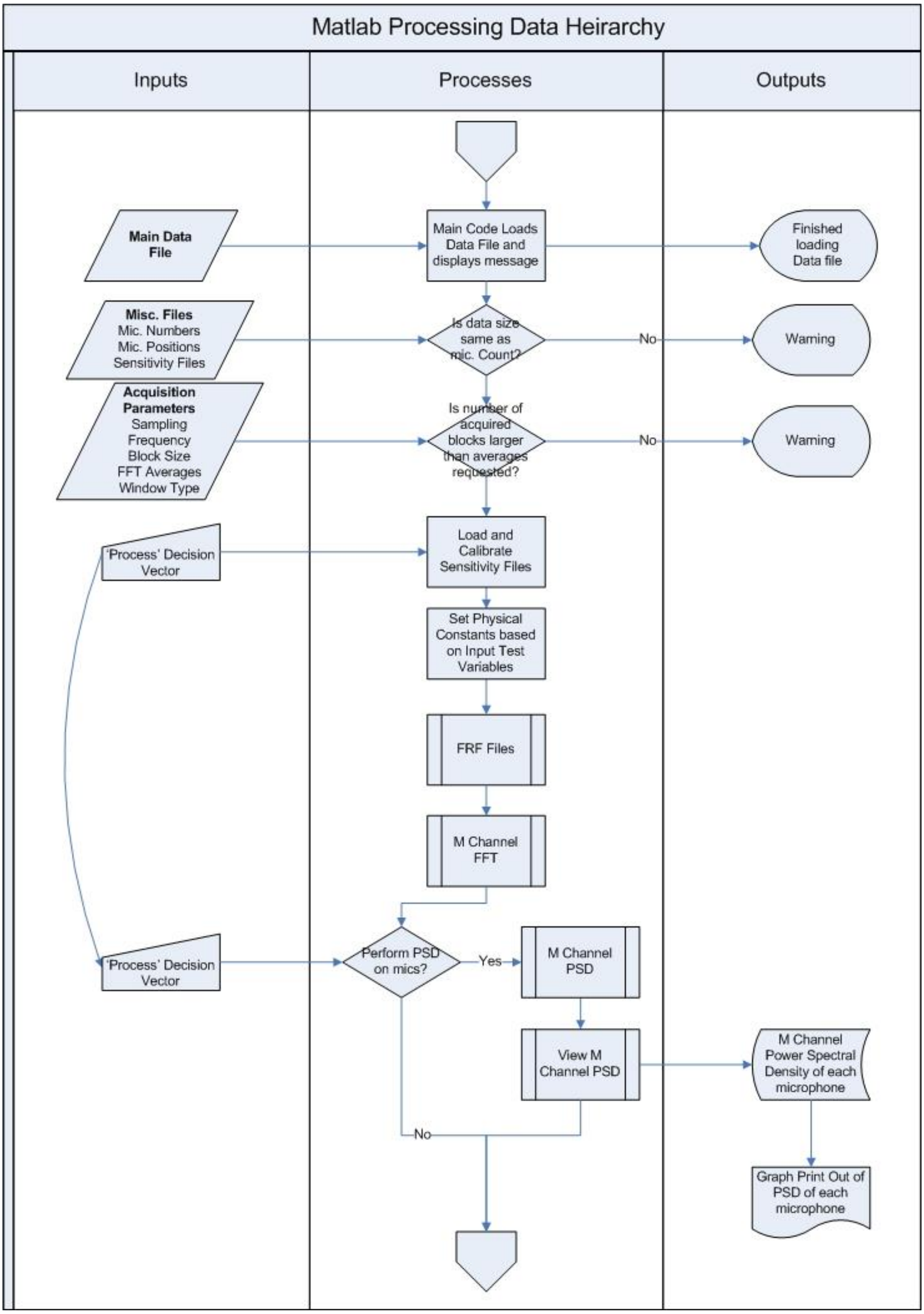


Figure 3.2 - Part 2 of the flowchart showing the Matlab code processing data from a microphone array during an aeroacoustic test.

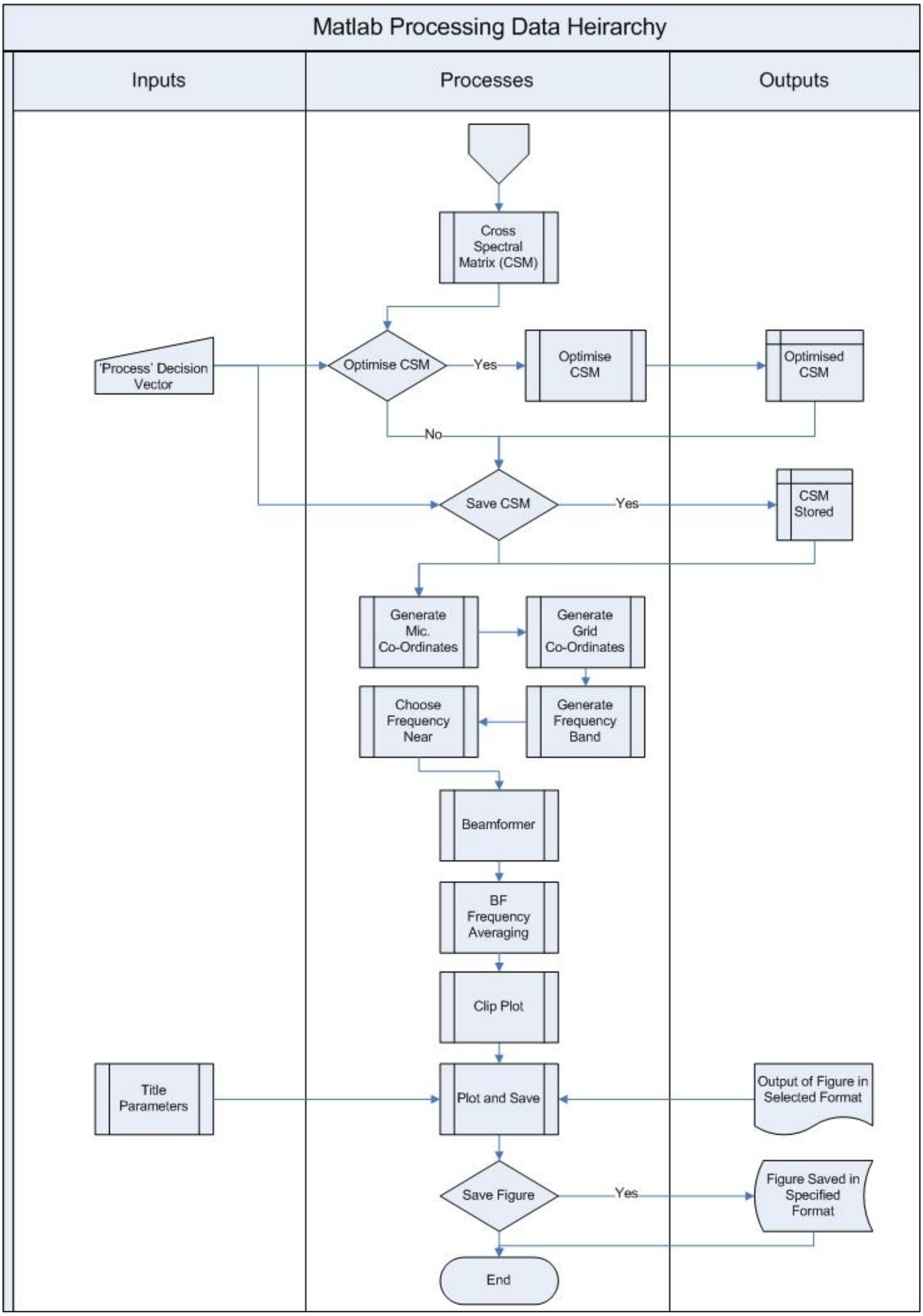


Figure 3.3 - Part 3 of the flowchart showing the Matlab code processing data from a microphone array during an aeroacoustic test.

3.3. Acoustic Survey Report

As part of the initial 12 months of this study an acoustic survey was conducted of the Airbus Filton wind tunnel. As a result of the study a number of recommendations were made to improve the Filton wind tunnel's acoustic testing potential.

3.3.1. Filton Wind Tunnel Acoustic Survey Executive Summary

This following describes results and recommendations of an acoustic survey of the Filton low-speed wind tunnel. The aim was to provide a description of the current operating environment. This is used to suggest improvements that can be made to the tunnel to improve the ability to perform aeroacoustic measurements in the future. The following secitons presents results obtained during the one day noise measurement session performed on the 19th July, 2006 at the Airbus Filton Wind Tunnel. The following questions were addressed:

- What background noise can be expected? How does this compare to other wind tunnels?
- Are there any particular phenomena present in the wind tunnel during operation which generates noise that might interfere with acoustic testing?
- What are the recommended improvements, which are required, to further extend the wind tunnel's acoustic measurement capability?

Testing was performed using a number of microphones mounted in the wind tunnel walls, placed around the working test section and near the fan. Details of the location of the microphones and methods used to partially mask them from boundary layer noise are included in the report. Results reveal that the wind tunnel has a substantial amount of noise produced at the fan section, which then propagates to the test section. Furthermore, evidence is produced showing tones at certain wind tunnel speeds; in particular there is a distinct tone evident at higher wind tunnel speeds. The frequency of the tone is dependent on the wind tunnel speed.

Figure 4 shows one-third octave band sound pressure levels across the wind tunnel speed range, from 5 to 80 m/s in 5 m/s intervals. The data was recorded from a recessed mounted microphone in the test section wall. The z-axis depicts flow speeds, with slower speeds at the front and the fastest speeds at the back. It can be clearly seen that an increase in wind tunnel speed results in an increase in overall background noise, particularly at higher frequency (above 1 kHz). At 80 m/s the sound level at a selection of frequency bands is over 100 dB. Figures 5 and 6 show narrow band pressure level spectrums from flush mounted microphones in the fan section and test section respectively, at 70 m/s (301.93 RPM).

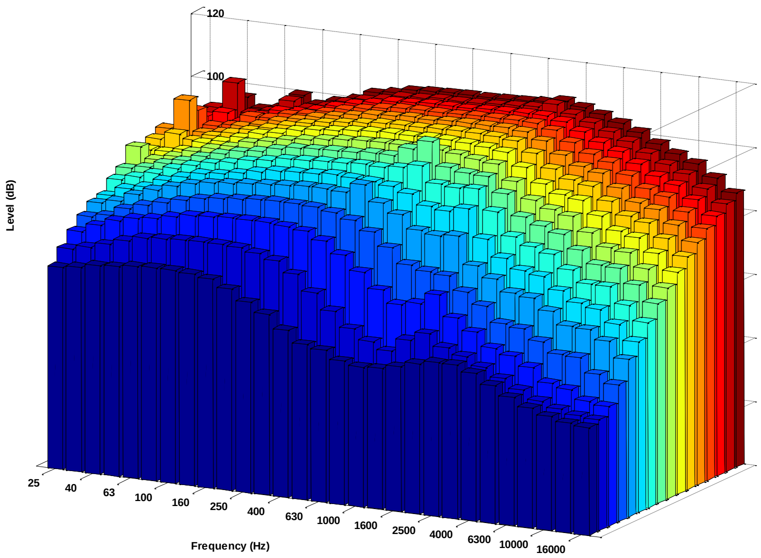


Figure 4 - Test section third-octave spectra across tunnel speed range.

The low frequency peaks evident especially in the section near the fan correlate well with the expected blade passing frequencies. The ‘hump’ evident in the fan section results (figure 5) at approximately 2500 Hz does not change its frequency with wind tunnel velocity and thus is not seemingly of aeroacoustic nature. Given this, it appears evident that the source of this ‘hump’ is airborne noise radiated from the motor driving the fan. This is further corroborated by the fact that the ‘hump’ is more prevalent in the data from the microphones located nearer the fan (and thus nearer the motor source). It is recommended to consider sound isolating or shielding the motor to improve the wind tunnel environment for aeroacoustic testing.

A significant tone, which can be seen at certain wind tunnel speeds, (not shown on these plots) increases in frequency as the speed increases and would appear to be of aeroacoustic nature. This is probably due to physical features of the Filton wind tunnel, such as turning vanes, protruding sections, cavities or other geometric features. The affect these tones have on acoustic testing is dependent on very specific testing conditions and the frequency of interest. Locating and then eliminating the source of the tones warrants further study.

Recommendations include optimising the recessing of the microphones to reduce the level of background noise at the microphone diaphragms. Recessing is proven in literature to effectively reduce the noise level by up to 20 dB. Longer term improvements include acoustic splitters and wind tunnel lining to lower background noise.

Other potential modifications include de-correlation techniques to reduce any 2D vortex shedding from structures in the flow, such as turning vanes, although any modifications made to the wind tunnel structure would need to be performed carefully to minimize any impact on flow quality and speed. Further modelling of the acoustic behaviour of the wind tunnel may be necessary to ascertain the absolute benefits of such modifications.

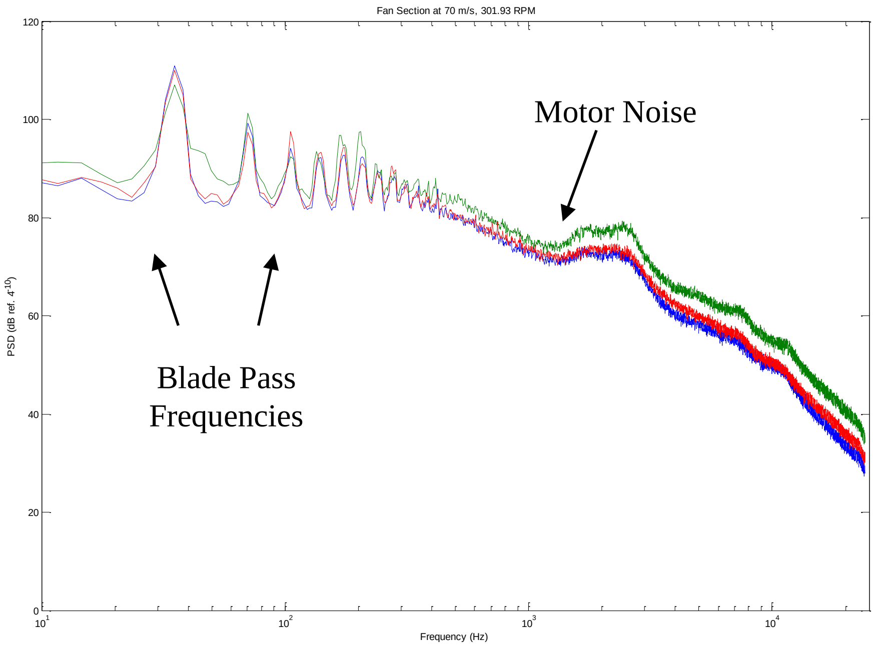


Figure 5. Acoustic spectrum at 70 m/s measured near fan.

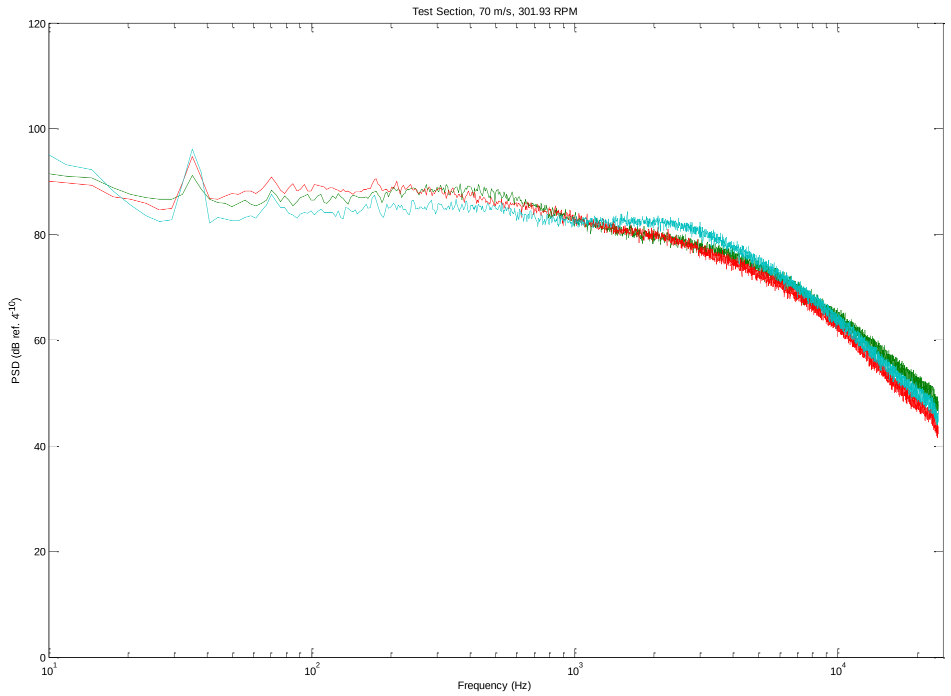


Figure 6. Acoustic spectrum at 70 m/s measured in test section.

3.3.2. Introduction

The Filton low-speed wind tunnel is used extensively for routine aerodynamic testing of aircraft components and models. With the increased importance that noise emissions from aircraft have on the design of aircraft components, it makes sense to introduce acoustic-testing capability to this wind tunnel. The overall aim of this report is to serve as a first step towards researching and studying methods by which current and near-future wind tunnel technologies can be exploited and implemented in the Filton wind tunnel. Specifically, this report presents results obtained during a one day background noise measurements test session performed on the 19th July, 2006.

3.3.2.1 Primary Objective:

To determine the background noise levels in the wind tunnel during operation at different speeds, in particular the standard testing speed (70 m/s). The background noise levels dictate what kind of acoustic testing can be performed, and help in choosing the correct equipment necessary to carry out such tests (for example, a suitable minimum channel count for a microphone array).

3.3.2.2 Secondary Objectives:

- Identify areas where the wind tunnel may be modified to get improvements in results from future aeroacoustic tests. This report provides recommendations for such improvements.

- Determine what level of background noise can be expected from previous studies performed on other similar wind tunnels, and compare them to the results obtained from the testing in the Filton wind tunnel.

- Identify any particular acoustic phenomena present in the wind tunnel during operation, and try to locate the source of these phenomena.

3.3.3. Methodology

The acoustic survey was carried out using nine calibrated electret microphones mounted at various positions in the wind tunnel circuit. The signals from the microphones were fed into preamplifiers which provided power to the microphones and amplified the signal by a fixed amount. Due to the large distances involved, the preamplifiers were located close to the source in order to maintain a high signal-to-noise ratio. Data was acquired using a National Instruments data acquisition systems based on a PXI 1042 chassis and four PXI-4462 cards. Data from all sixteen channels was acquired simultaneously, digitised with a sampling frequency of 48 kHz and streamed to disk. The acquisition process was controlled using a custom-built acquisition program written in Labview, running on a PC-based National Instruments 8350 controller. This controller could be remotely controlled from the wind tunnel control room.

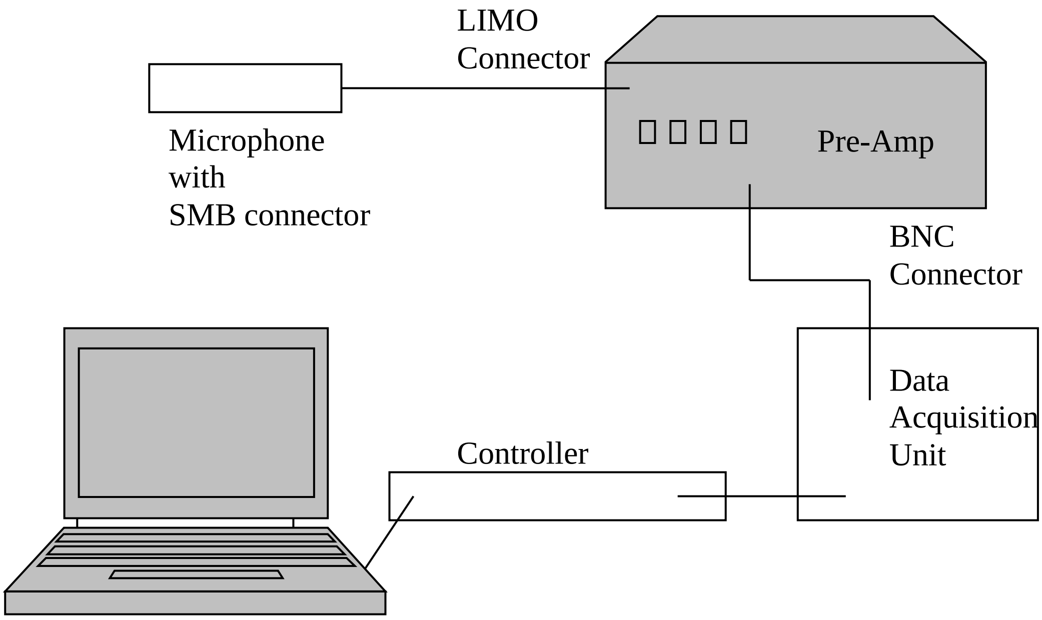


Figure 7 – Schematic of equipment used for acoustic testing.

In order to get data from both the source and receiver sides of the tunnel, microphones were installed at the test section and on either side of the fan. In total nine microphones were installed: three at the fan section, four in the floor of the test section and two in the wall of the test section, as shown in figures 8 and 9. The particular locations of the microphones correspond to available tappings in the tunnel shell. The fan section microphones were located along the centreline of the wind tunnel: microphone 6 was located at a distance of 3.150 m from the fan, microphone 7 at 6.79 m and microphone 5 at a distance of 9.29 m. The working section of the wind tunnel had a total of six microphones located at various points Microphone 16 was located at the centre point of the working section – in the centre of the circular rotating section. Microphone 11 was located 2.4 m downstream from the centre point and along the centre line. Both microphones 12 and 15 were located 0.355 m downstream from the centre point and 1.09 m either side of the centre line. These locations provided sufficient cover for the entire area. In addition two further microphones were located in the test section wall. Microphone 10 was located 1.69 m up the wall and 0.73 m downstream from the centre point. This microphone was recessed by approximately 5mm to allow for direct comparisons with the other microphones. All other microphones were flush-mounted, and any remaining gaps in the original holes were appropriately sealed. All microphones in the test section were covered with a smooth porous cloth in an attempt to smoothen the flow over the microphone diaphragms.

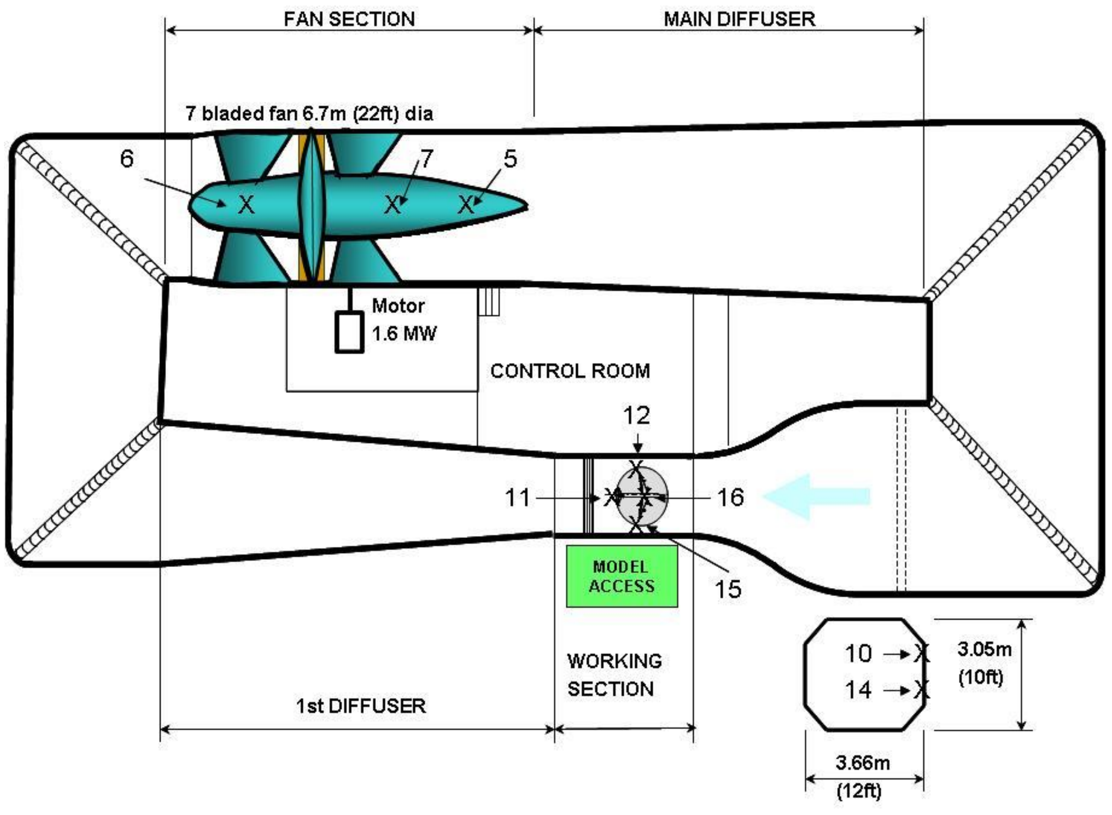


Figure 8 – Plan of the Filton Wind Tunnel with microphone placements indicated.

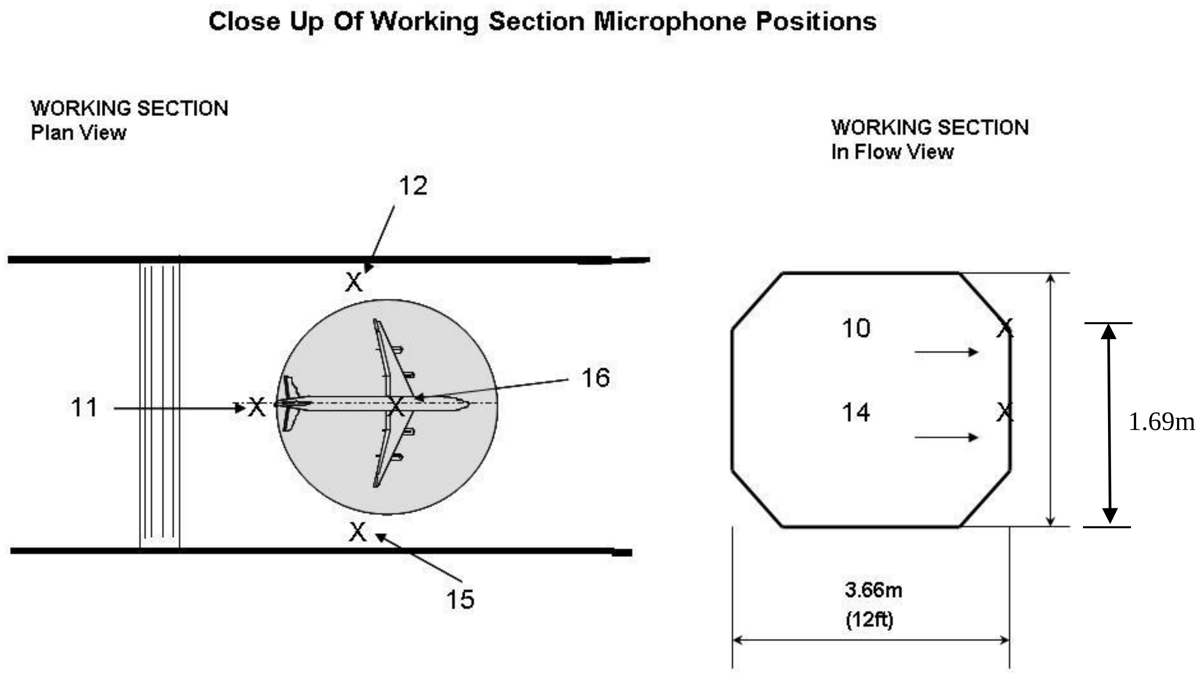


Figure 9 – Plan and cross-section view of the test section of the Filton Wind Tunnel.

3.3.3.1. Testing Procedure

Data was acquired with the wind tunnel running at a range of wind speeds, namely from 5 m/s to 80 m/s at 5 m/s intervals. For each case, a twenty second acquisition was recorded after the speed had settled. The temperature and the barometric pressure inside the tunnel were recorded for each run. Additionally the revolutions per minute (RPM) of the motor was recorded for each of the wind tunnel speeds.

Processing was done at a later stage using custom-written applications in Matlab and Labview. Raw data was converted to engineering units using pistonphone calibration values, and then Fourier transformed to obtain narrow band spectra of the pressure spectrum density. A one-third octave band analysis was also performed, together with an overall A-weighted level analysis to enable comparisons with other wind tunnel data. N-octave band plots are usually more useful for absolute level comparisons, since the values do not change with FFT processing parameters. On the other hand, narrow band spectrums are ideal to identify tonal features in the noise spectrum.

3.3.4. Results & Discussion

3.3.4.1. One-third octave band spectra

The first set of plots show the one-third octave spectrums at two wind tunnel speeds: 40 and 70 m/s. The former was chosen to enable direct comparisons with the 7x5 wind tunnel available at the University of Southampton, where microphone array measurements have already been carried out successfully. On the other hand, 70 m/s corresponds to the standard operating speed at the Filton wind tunnel. It is of interest to see how the Filton wind tunnel compares to the Southampton one at 40 m/s, and then analyse the noise increase from 40 to 70 m/s.

Figures 10 and 11 show the one-third octave background noise spectrums corresponding to a wind speed of 40 m/s recorded at the Filton and Southampton wind tunnels, respectively. Due to the equipment used for the measurements, the data for the 7x5 wind tunnel does not include a value for the band centred at 16 kHz. Furthermore, the two bands on the right correspond to the overall levels (A- and Linear weighted, respectively). These values are not shown in figure 10. Data for figure 10 was taken from microphone 10 (recessed).

The two figures show roughly the same trends, i.e. a peak in the noise levels in the frequency range from 100 – 2000 Hz, after which the levels drop fairly rapidly. The drop-off seems to be smoother in the Filton wind tunnel. The particularly high levels in the 800 Hz and 1 kHz bands for the same tunnel indicate the presence of significant tonal noise, which is not present in the 7x5 wind tunnel. More details about this will be discussed later. On the low frequency side (< 100 Hz), levels are significantly higher in the Filton wind tunnel. This might be because the microphone mounting procedure was completely different when testing in the Southampton 7x5 tunnel, where the microphone was situated outside the wind tunnel boundary, in free field conditions. This setup eliminates the contributions of cavity resonances and most of the hydrodynamic forces which microphone 10 was subjected to.

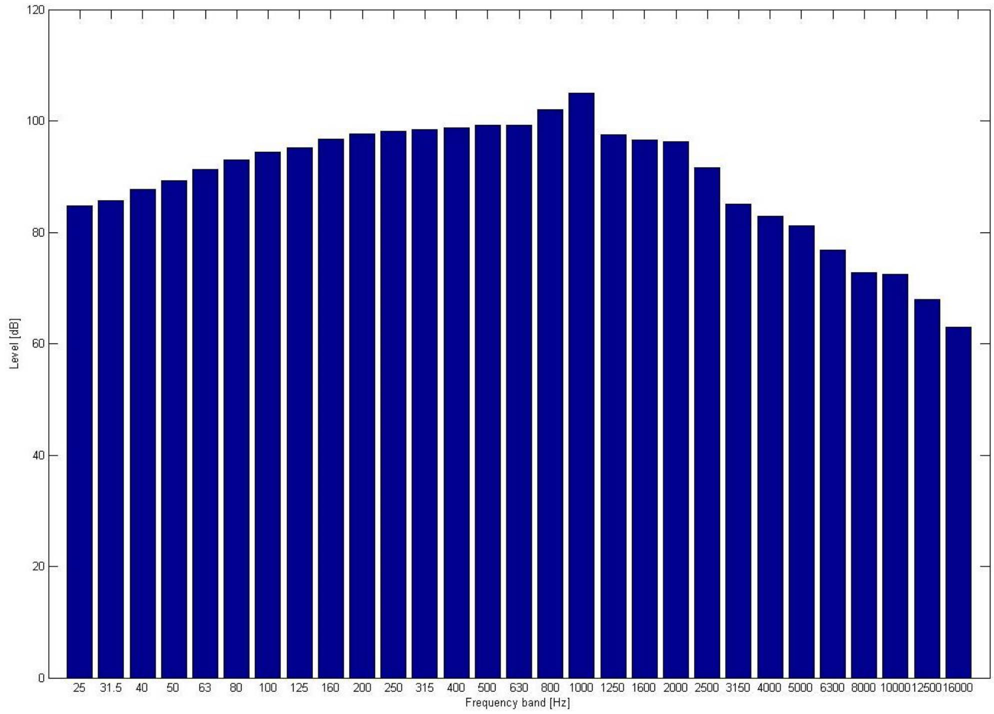


Figure 10 – One-third octave band spectrum recorded in the test section at 40m/s, using the recessed microphone.

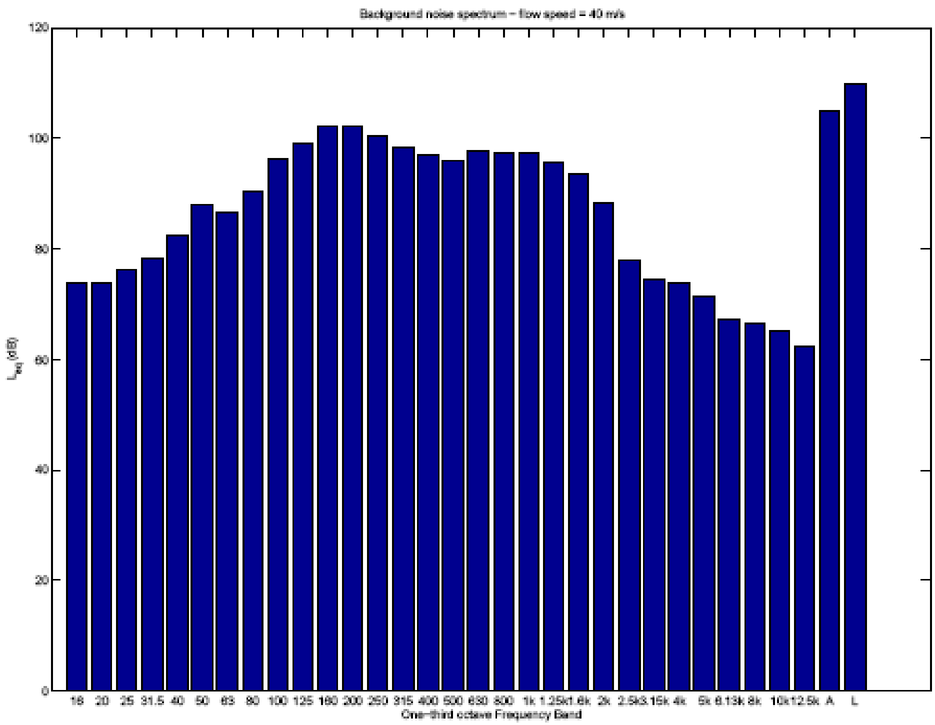


Figure 11 - One-third octave band spectrum recorded in the test section of Southampton's 7x5 Wind Tunnel at 40m/s. From ref. [24].

For microphones mounted on wind tunnel boundaries for use in phased array measurements, data below 500 Hz is not of much use, due to inherent problems in beamforming resolution at such frequencies. As such, data at these very low frequencies is considered irrelevant.

Figure 12 shows the one third octave spectrum corresponding to 70 m/s. The spectrum follows the same trend as at 40 m/s, but is shifted higher in level. Thus, levels peak at around 105 dB at centre frequencies between 200 Hz – 2 kHz, and decrease to just above 80 dB at 16 kHz. At 40 m/s, levels peaked at around 95 dB (ignoring the tones) between 500 and 1250 Hz, and decreased to just above 60 dB at 16 kHz. Therefore, the increase in level is more pronounced at higher frequencies (up to 20 dB) than the mid-frequencies (8 – 10dB). This phenomenon can be seen more clearly in figure 13, which shows the spectrums corresponding to a velocity sweep from 5 to 80 m/s, at 5 m/s intervals. Higher speed runs are at the back whilst the slower wind tunnel speeds are located towards the front of the graph. The plot also shows the presence of strong tonal components for some of the test conditions.

Figures 14 and 15 show the same velocity sweep plots using data from flush-mounted microphones in the test section and fan section, respectively. Figure 14 is very similar to figure 13, except for a slight increase in levels at the high frequency end. A more direct comparison between flush-mounted and recessed microphones is presented later on. Conversely, figure 15 is significantly different from figures 13 and 14. The noise in the fan section is significantly lower than in the test sections, and there is a much stronger presence of tonal components. Noise levels still increase with wind speed, but in a more uniform way. The tonal noise at very low frequencies can be attributed to the fan. The lower overall levels are due to lower flow speeds in the fan section compared to the test section, since the cross-section is significantly larger. This shows the extent of contribution of aerodynamic noise to the overall background noise levels in the wind tunnel.

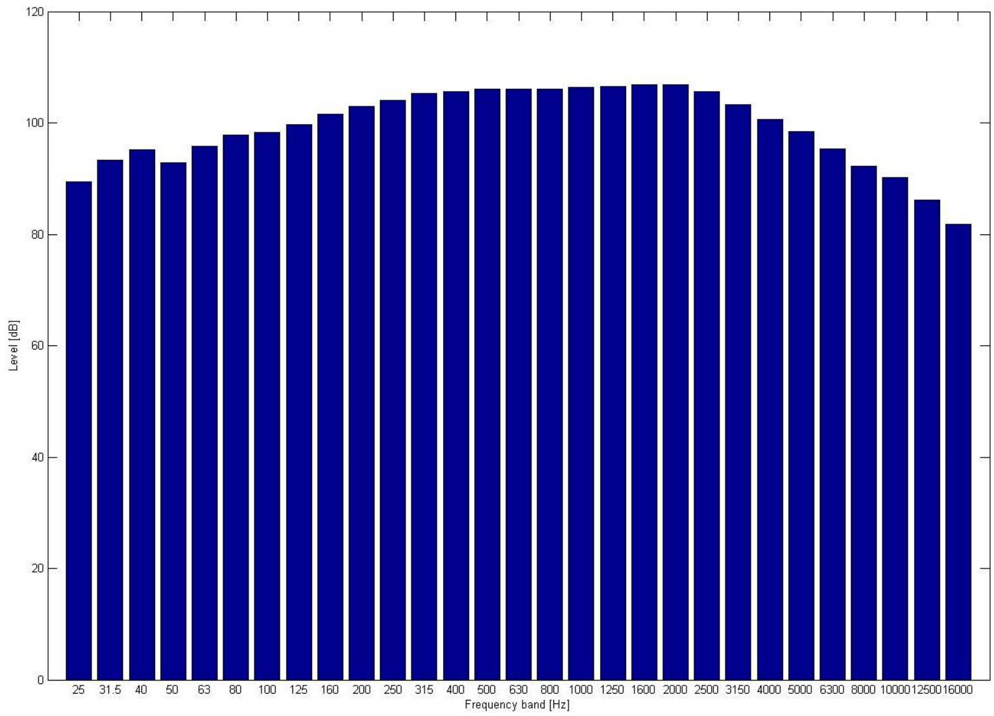


Figure 12 - One-third octave band spectrum recorded in the test section at 70 m/s, using the recessed microphone.

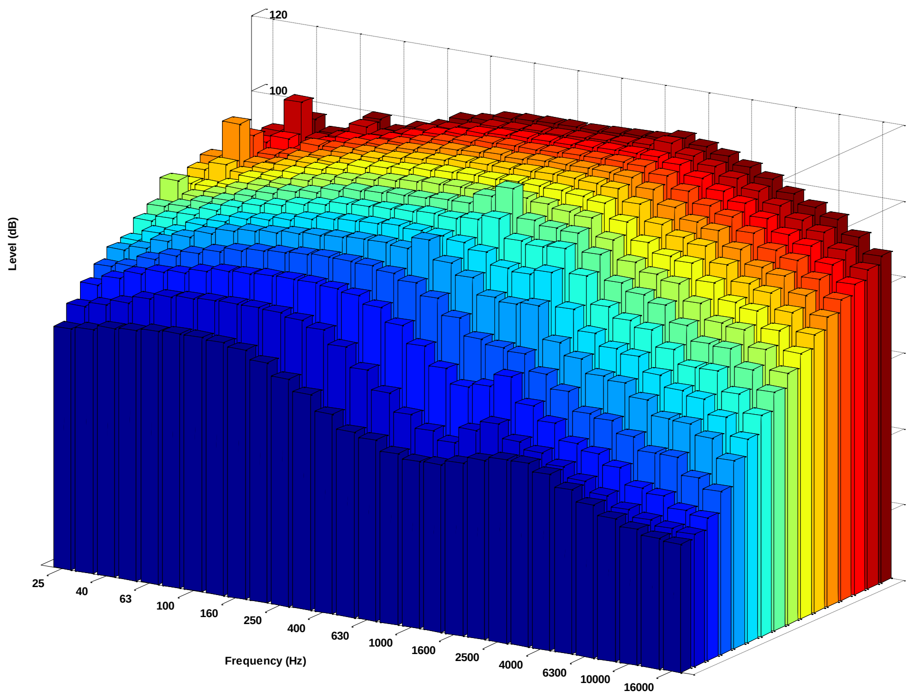


Figure 13 - One-third octave band spectrum recorded in the test section for a velocity sweep from 5 m/s to 80 m/s at 5m/s intervals using a recessed mounted microphone.

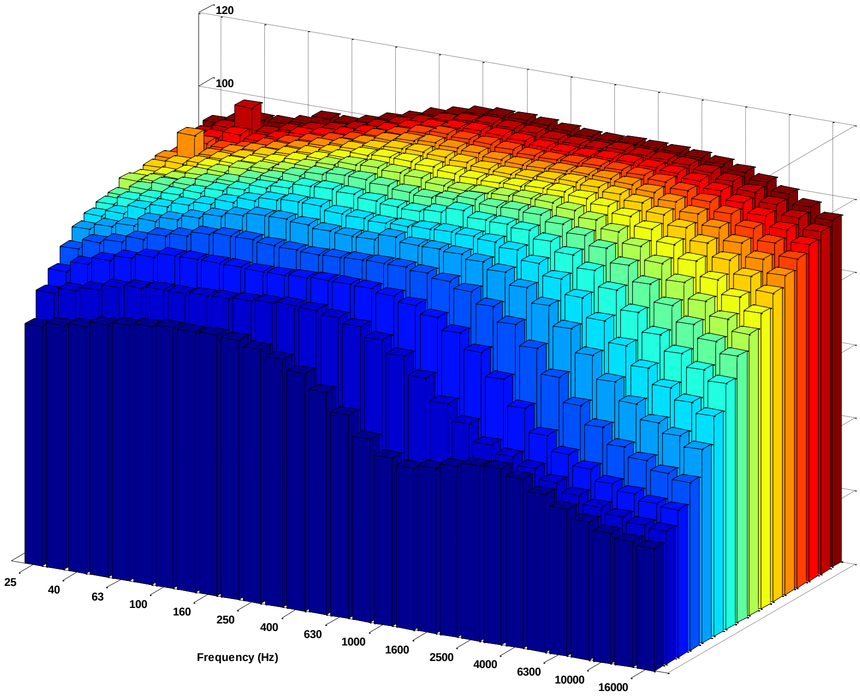


Figure 14 - One-third octave band spectrum recorded in the test section for a velocity sweep from 5 m/s to 80 m/s at 5m/s intervals using a flush mounted microphone.

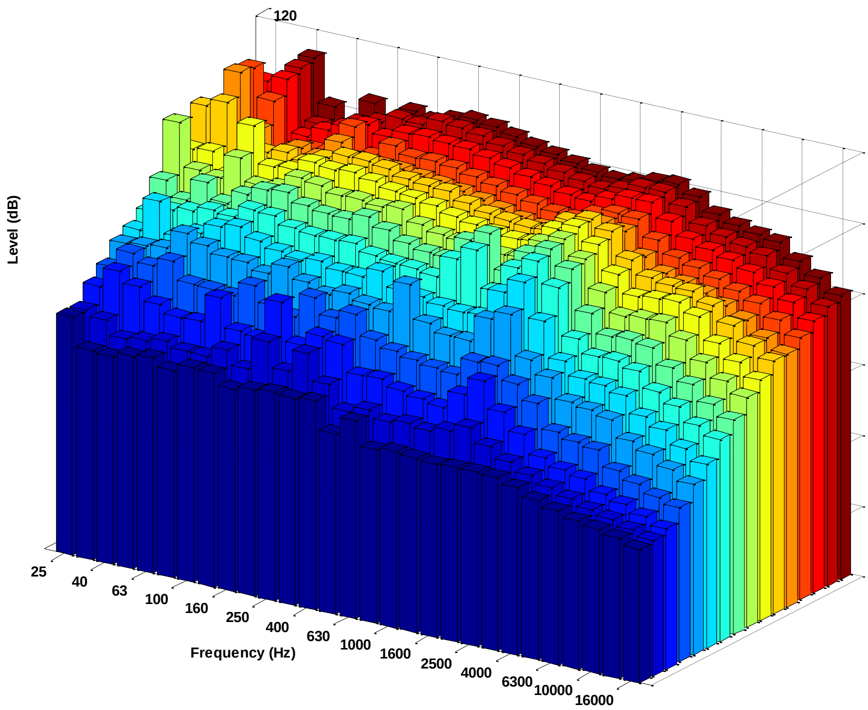


Figure 15 - One-third octave band spectrum recorded in the fan section for a velocity sweep from 5 m/s to 80 m/s at 5m/s intervals using a flush mounted microphone.

3.3.4.2. Narrow-Band Spectra

Figures 16 and 17 show the background noise spectra in the test section at 70 and 40 m/s, respectively. Each of the plots shows a selection of three microphones. Microphones 12 and 15 were both flush mounted, and therefore yield very similar results. The small discrepancies are probably due to localised flow disturbances, caused for example by the fact that the microphones were not perfectly flush mounted. Such minute details start to be an issue at high frequencies. On the other hand, microphone 10 was recessed, and these two plots show that recessing has some obvious advantages at high frequencies (> 2 kHz). The lower background noise recorded arises from the fact that a flush-mounted microphone is subjected to large hydrodynamic forces corresponding to the turbulent boundary layer at the wind tunnel boundaries. The output from such microphones is therefore a combination of the noise present and the effect of such forces. By separating the flow from the microphone diaphragms, the contribution of the latter is diminished. In this case, reductions of up to 15 dB can be seen, although they are not consistent. A properly recessed microphone can record levels up to 20 dB lower, as will be shown in the Recommendations in Section 3.3.5.

Another observation from the same plots is the presence of two tonal noise components: a minor one (5 dB above the average level) at approximately 30 Hz for the 70 m/s case, and a significant tone, nearly 20 dB above the broadband level at just under 1 kHz for the 40 m/s case. Although this tone is particularly evident at 40 m/s, it is also present at other wind speeds between 30 and 60 m/s. The tone increases in frequency as the speed increases, which suggest that it is of aero-acoustic nature. Such tones can usually be attributed to vortex shedding or cavity noise; both produce highly tonal noise in a specific flow range. In fact, a similar tone occurs in the 7x5 Southampton wind tunnel, albeit at a lower speed. In that case, vortex shedding at the turning vanes just downstream of the test section has been attributed as the cause of the tone.

As regards for the broadband levels, they are fairly constant at approximately 90 dB for 70 m/s and 80 dB for 40 m/s up to around 1 kHz, after which a sharp drop-off occurs.

In the fan section, the narrow band spectra look significantly different, as shown in figures 18 and 19. The low frequency end is dominated by a series of strong tones. Such tones are a common feature of fans, and occur at a frequency called the blade pass frequency[1] (BPF), which is a function of fan RPM and the number of blades, and corresponding harmonics. The table in Figure 20 lists the fundamental frequencies corresponding to all the wind tunnel speeds at which tests were performed. Thus, for example, at 70 m/s the BPF should occur at approximately 35 Hz; this corresponds well to the location of the first significant peak in figure 18.

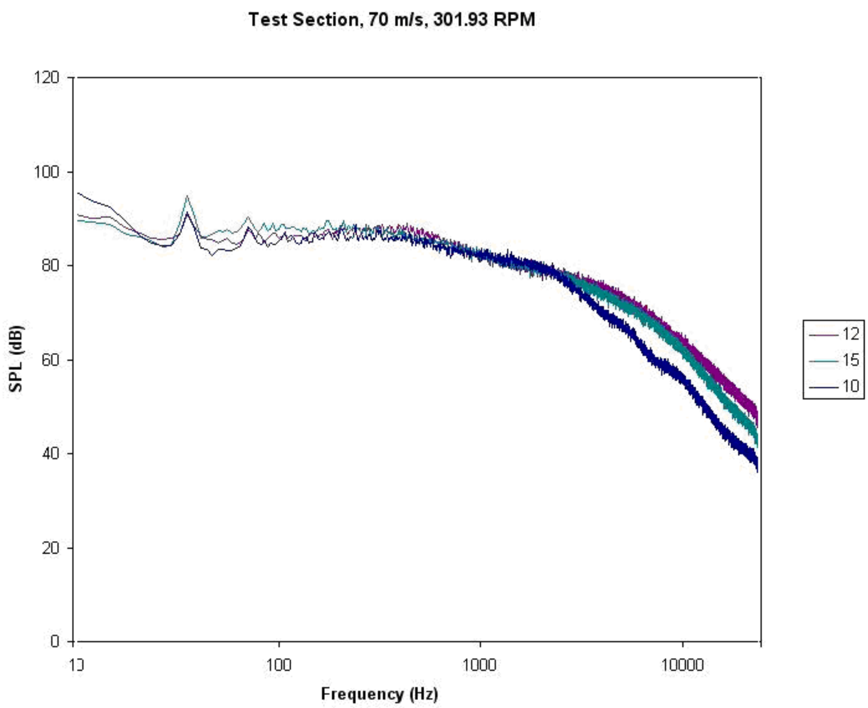


Figure 16 – Narrow band background noise spectrum in the test section at 70 m/s. Comparison between flush mounted microphones (12, 15) and the recessed microphone (10).

[1] Definition from http://www.dliengineering.com/vibman/gloss_bladepassfrequency1.htm

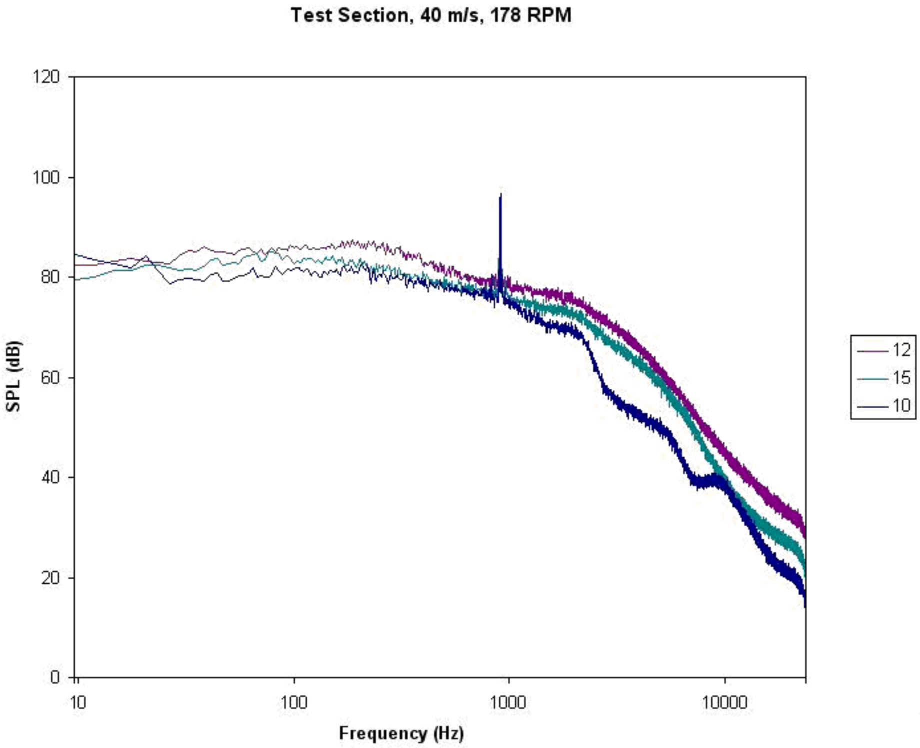


Figure 17 – Narrow band background noise spectrum in the test section at 40 m/s. Comparison between flush mounted microphones (12, 15) and the recessed microphone (10).

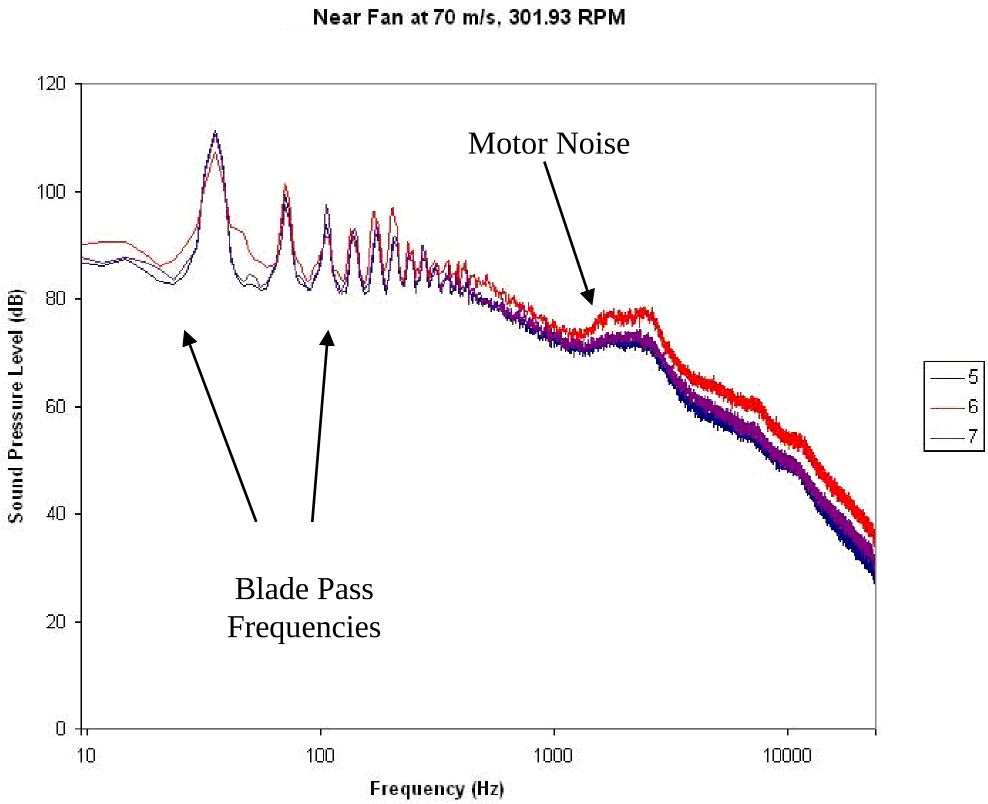


Figure 18 – Narrow band background noise spectrum in the fan section at 70 m/s.

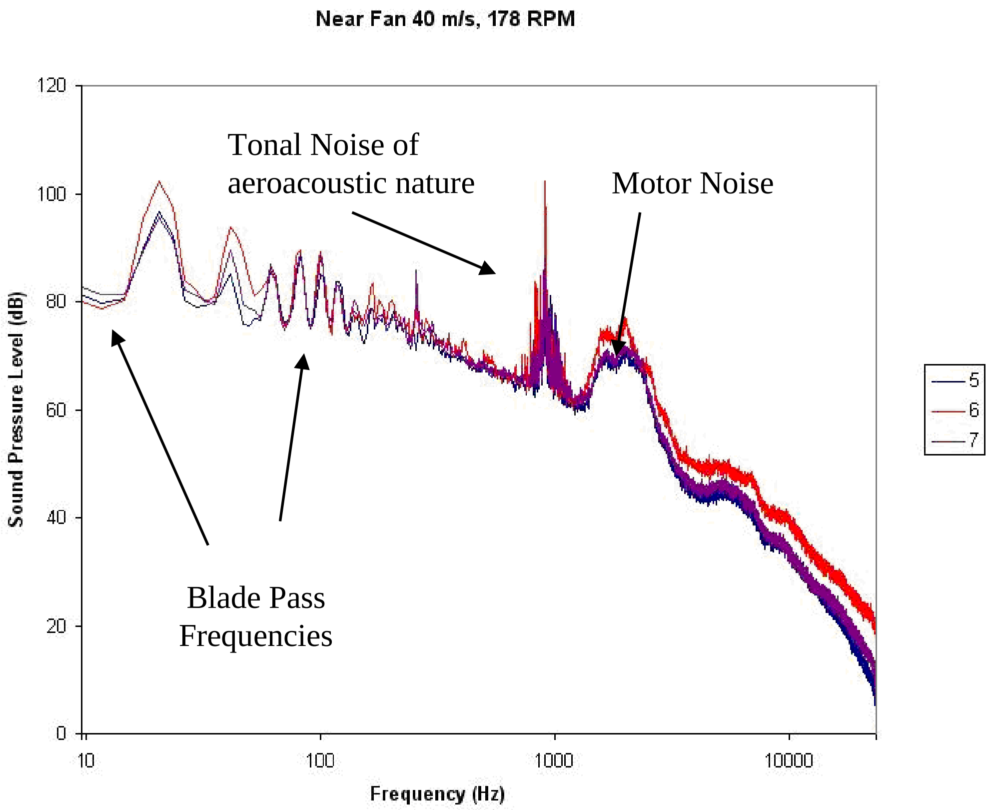


Figure 19 – Narrow band background noise spectrum in the fan section at 40 m/s.

The second feature present in the measurements in the fan section is what can be described as a 'hump' in the spectrums occurring between 1 and 2 kHz, in the form of an increase in level. This 'hump' is clearly evident in the results for the fan section at all speeds, and also in the test section results at the lower speeds. At the higher wind tunnel speeds it is masked by the overall higher background noise. The frequency at which it occurs does not appear to vary significantly with change in velocity, and neither does its level. Therefore it is not seemingly of aeroacoustic nature. The motor driving the fan seems to be a good logical explanation, given that it is closest to the fan section. However, further local measurements are required before the noise source is identified with certainty.

The tonal noise identified earlier at around 1 kHz is also prominent at the same wind speed (40 m/s) in the fan section; furthermore, the level is even higher, peaking at over 100 dB. This indicates that the fan section is closer to the source than the test section.

Speed m/s	**RPM**	**Revolutions per second**	**Blade Pass Frequency (Hz)**
5	32	0.533333	3.73
10	45	0.75	5.25
15	97	1.616667	11.32
20	91	1.516667	10.62
25	115	1.916667	13.42
30	134	2.233333	15.63
35	156	2.6	18.20
40	178	2.966667	20.77
45	200	3.333333	23.33
50	220	3.666667	25.67
55	241	4.016667	28.12
60	261.15	4.3525	30.47
65	282.13	4.702167	32.92
70	301.93	5.032167	35.23
75	319.81	5.330167	37.31
80	337.95	5.6325	39.43

Figure 20 - Table showing Blade pass frequency corresponding to flow speed in the test section.

Figures 21 and 22 show a direct comparison of the level difference between 40 and 70 m/s in the test section with recessed and flush-mounted microphones. Figure 23 shows the same comparison in fan section with flush mounted microphones. The main observations are that recessing is more effective in lowering the levels at lower wind speeds, and that in the fan section lower wind speeds give rise to significantly lower background noise levels even at the medium frequency range (100 – 1000 Hz).

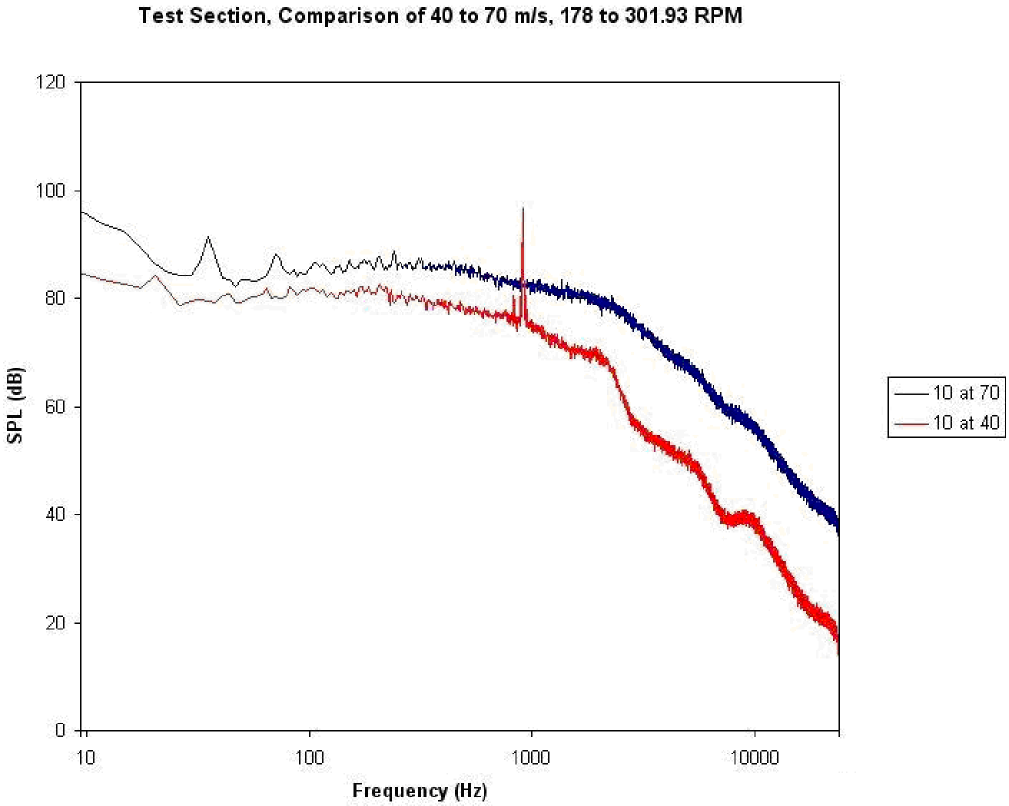


Figure 21 – Comparison between narrow band spectrum levels in the test section at 40 m/s and 70 m/s, using microphone 10 (recessed microphone).

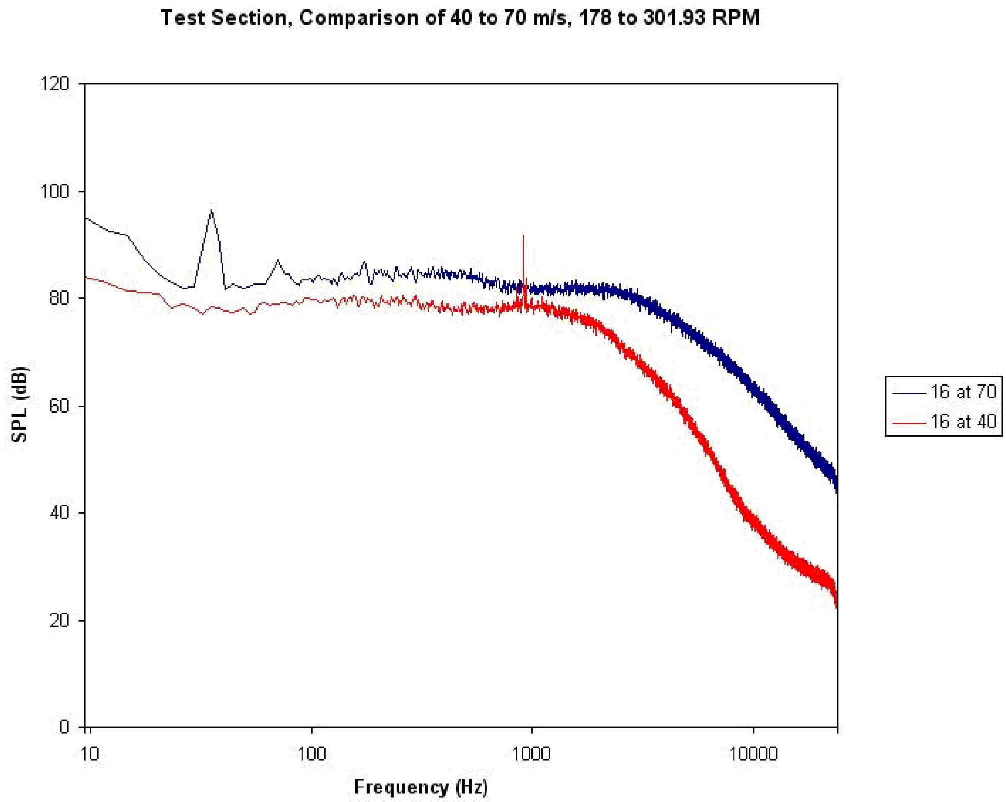


Figure 22 - Comparison between narrow band spectrum levels in the test section at 40 m/s and 70 m/s, using microphone 16 (flush mounted microphone).

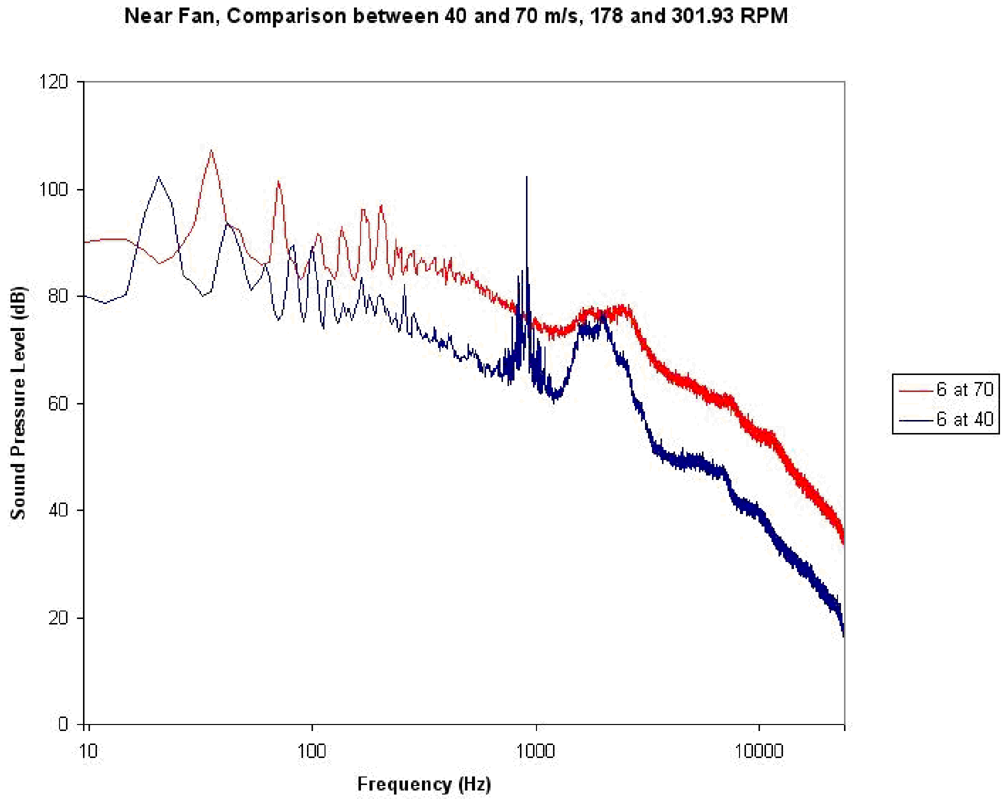


Figure 23 - Comparison between narrow band spectrum levels in the fan section at 40 m/s and 70 m/s, using microphone 6 (flush mounted microphone).

Finally, figure 24 shows the overall A-weighted levels in the test section as a function of wind speed. For the lower wind speeds (< 25 m/s), the recessed microphone yields a roughly constant decrease in level of 5 dB. As the speed is increased, the discrepancy diminishes, and above 70 m/s, it is negligible. This figure can be compared directly to background noise measurements in a number of aeroacoustic wind tunnels, shown in figure 25. Levels in the Filton wind tunnel are up to 50 dB higher when compared to the state-of-the-art (Audi aeroacoustic wind tunnel [13]). This is not surprising, given that this wind tunnel was designed specifically for the lowest possible noise levels. Furthermore, noise measurements in this case are taken outside the flow, which means that boundary layer noise is not present in these measurements. The levels in the Filton wind tunnel do however compare well with background noise measurements in a conventional wind tunnel (Mercedes Benz).

Another comparison can be made with the levels shown in figure 26, corresponding to the NASA Ames 40x80 closed-section wind tunnel [25]. In this case background noise levels are significantly higher than the Filton wind tunnel, however the flow speed is also higher (123 m/s). It is yet unclear how data from in-flow microphones (using fore-bodies) correlates with flush-mounted and recessed microphones, and research is ongoing at the University of Southampton regarding this.

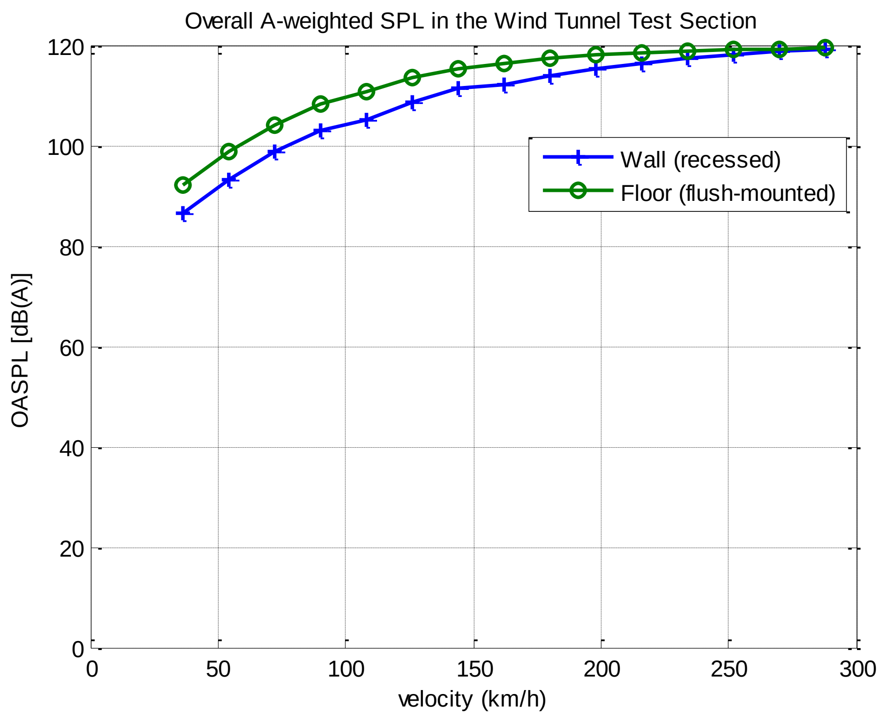


Figure 24 – Overall A-weighted SPL in the test section as a function of wind speed for both recessed and flush mounted microphones.

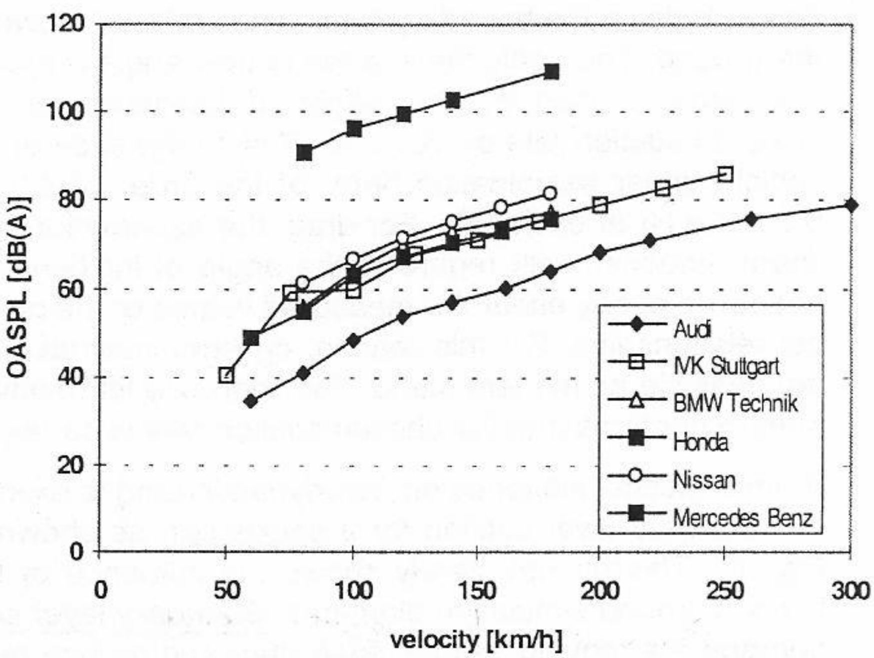


Figure 25 – Overall A-weighted SPL in the test sections of a number of aeroacoustic wind tunnels used in the automobile industry, together with levels in the Mercedes Benz wind tunnel, which is a conventional closed-section wind tunnel. For the aeroacoustic wind tunnels, measurements are outside the flow. From ref. [13].

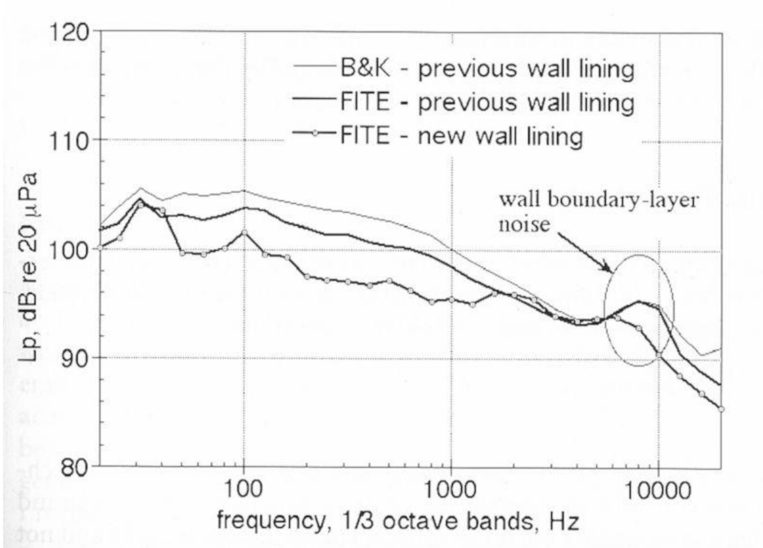


Figure 26 – Background noise measurements in the NASA Ames 40x80 WT, in-flow measurements. Data acquired at flow speed of approx. 123 m/s. From ref. [25].

3.3.5. Recommendations

The main observation from the plots presented in the previous section is that the levels at high frequencies are significantly high at 70 m/s. Although this compares well with data from other similar wind tunnels, care must be taken when designing a flush-mounted microphone array to ensure that successful results can be obtained from such a setup. The most obvious recommendation is to have a high channel-count system (> 100 channels). The more channels in an array, the better the capability to detect noise sources which are otherwise buried in the background noise.

Another strong recommendation is to design a properly recessed array. The measurements presented here already indicate a minor improvement by simply recessing individual microphones by a few millimetres. However, a properly recessed array [15, 21] can give much better results – up to 20 dB lower levels at 5 kHz and 12 dB at 10 kHz, as shown in figure 27. The most important feature is a tightly-stretched porous cloth (such as Kevlar) which is flush with the wind tunnel boundary, and behind which the array is installed. The cloth helps to separate the wind tunnel boundary layer from the microphone diaphragms.

An additional approach is to introduce acoustic liner inside the wind tunnel. This can have two objectives: lining a significant part of the wind tunnel and including baffles and splitters to lower the overall background noise levels by attenuating noise propagating in the duct [12], and lining the wind tunnel test section to reduce unwanted reflections which might "confuse" the array for source identification [14,24]. Figure 28 shows the reduction in background noise levels achieved in the Lockheed Martin low speed wind tunnel by lining parts of the diffusers downstream of the test section. Other wind tunnels, such as IVK-Stuttgart and NASA-Ames 40x80 have also been modified by introducing liners and splitters [12]. Any potential impact on overall wind tunnel performance must be evaluated carefully before considering implementation of any of these solutions.

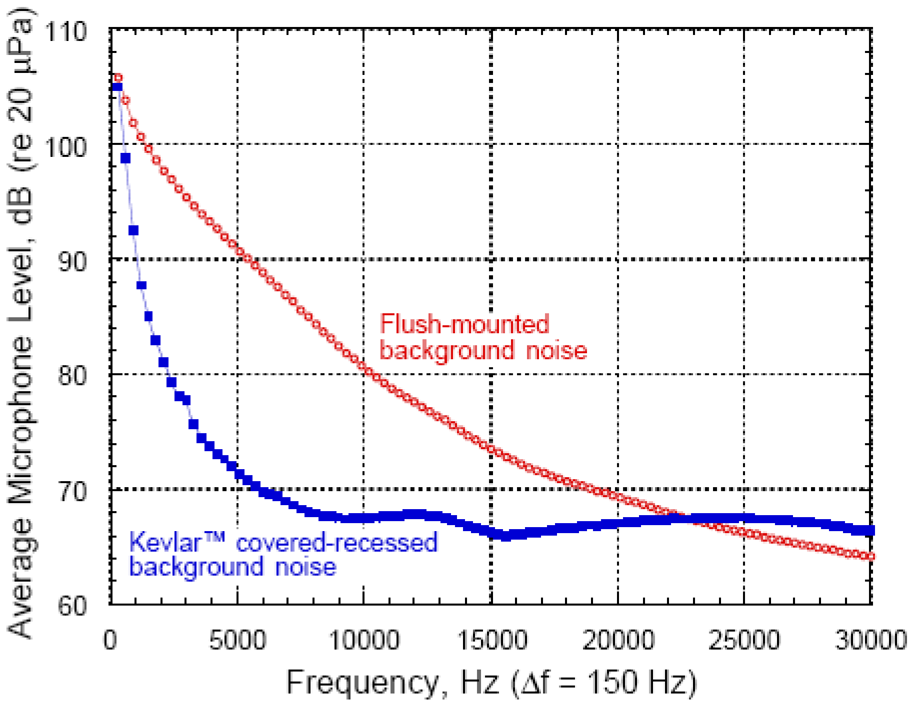


Figure 27 – Background noise levels recorded by a flush mounted microphone and one in a carefully recessed array. From ref. [15].

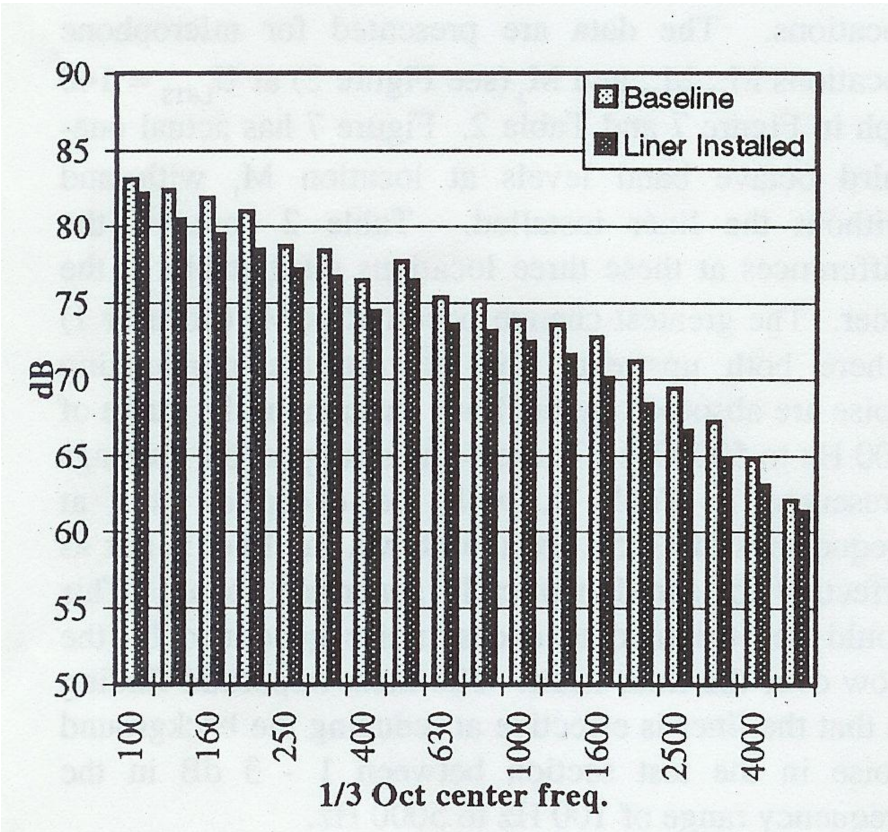


Figure 28 – Noise reduction achieved by installing acoustic liner in the Lockheed Martin low speed wind tunnel. From ref. [12].

The significant tone occurring at around 1 kHz should not be an issue during normal tests, since it is not visible at 70 m/s. However, if tests at lower speeds are required, or if future noise control techniques bring the noise levels down enough for the peak to be evident, the source of this tone has to be identified. If the source is indeed vortex shedding from turning vanes (such as is the case in the 7x5 wind tunnel at the University of Southampton), then a potential solution is to decorrelate the 2D vortex shedding phenomena by using, for example, serrations on the turning vanes themselves [2].

The same arguments apply to the "hump" noise which has been attributed to airborne noise from the motor driving the fan. In the present state this noise is not apparent in the background noise spectrum in the test section at 70 m/s. However this does not mean that this noise will not be an issue if noise abatement techniques are introduced. Suggestions for controlling this noise source include mounting the motor on vibration isolators and enclosing it in an acoustic isolation enclosure.

3.3.5.1. Future Work

The following is a summary of the main possible actions that aim to improve the performance of the tunnel in terms of aeroacoustic measurement capability.

- Identification and mitigation of the source of the tonal noise
    - Aeroacoustic sources and minimize effects – de-correlate 2D vortex shedding flow
- Identification of the 1 – 2.5kHz noise, investigating motor as likely source
    - Sound proofing/dampening of the motor
- Investigating splitter designs to attenuate background noise, and predict performance impact on tunnel operations
- Optimising microphone recessing
- Optimising microphone coverings
- Lining solutions for the test section (especially directly opposite the microphone array)
- Identifying and eliminating local sound sources, like small protrusions and discontinuities
- Isolate frequencies that are particularly problematic – map them to avoid in future testing

3.3.6. Conclusions

An acoustic survey of the Filton low-speed wind tunnel has been carried out and major sources of noise identified. The overall background noise level is approximately 100dB (A-weighted) at an operating speed of 70 m/s, however this is dominated by noise in the frequency range from 100 – 1000 Hz. Scaling of the background noise with flow speed shows the expected trend, although there is a more significant increase in level above 2 kHz.

The wind tunnel has typical characteristics of a closed-section hard-walled design. The major source is the fan, with blade passing frequencies being apparent in the measured spectra. An aeroacoustic tone is particularly prevalent at certain flow speeds, and is indicative of turning vane or cavity noise. Identification and mitigation of this noise source will significantly improve capability, as it lies within a frequency range of interest. A noise source at approximately 1 – 2.5 kHz that varies little with flow speed may well be from the motor. Further investigation of this is necessary before it can be reduced.

While the noise levels of the Filton tunnel are comparable with the Southampton 7x5 tunnel at the same speed, the Filton tunnel typically operates at a much higher speed (70m/s) than the Southampton tunnel (40m/s). The use of a properly-designed recessed microphone array to reduce boundary layer noise should allow an improvement of up to 20 dB, which should provide significantly enhanced measurement performance. Further improvements can be obtained by a high-channel count (> 100 channels).

Acoustic lining of the test section opposite the microphone array is desirable to avoid direct reflections from degrading the array processing. The inclusion of acoustic splitters may provide some level of overall noise reduction, but the practicalities of installation and effect on overall wind tunnel performance must be examined in detail.

In summary, aeroacoustic measurements in the Filton low-speed wind tunnel should provide useful data, and there is scope for improving the facility to allow more detailed measurements to be carried out in the future.

3.3.7. Recommendations

The following is a summary of the main possible actions that aim to improve the performance of the Filton wind tunnel in terms of aeroacoustic measurement capability. These recommendations were born after result graphs revealed a number of acoustic anomalies. These are some recommended improvements that would allow the wind tunnel to further support acoustic technology?

- Identification and mitigation of the source of the tonal noise
  - Further study into the source of the tonal spike
  - Aeroacoustic sources and minimize effects – de-correlate 2D vortex shedding flow

- Identification of the 1 – 2.5kHz noise, investigating motor as likely source
  - Sound proofing the motor
  - Possibility of looking into sound dampening the motor - measuring the levels off it during a run.

- Investigating splitter designs to attenuate background noise, and predict performance impact on tunnel operations

- Consider the effect modifications will have on the flow or physical structure of the wind tunnel

- Out of flow measurements can have improvements of the order of 10 - 20 dB
  - Optimising microphone recessing
- Recessing solution
  - Optimising microphone coverings

- Lining solutions for the test section (especially directly opposite the microphone array)

- Possible Lining solutions elsewhere in the wind tunnel

- Identifying and eliminating local sound sources, like small protrusions and discontinuities

- Isolate frequencies that are particularly problematic – map them to avoid in future testing

- What would be required for effective use at the operational speed of 70 m/s – compare to slower speeds – lower background noise so fewer microphones required at lower speed

- Utilising more microphones to lower the difference requirements to isolate sound sources from background noise

- Further study into the effect of recessing and covering the microphones – specifically in isolating the noise generated by the presence of the microphones themselves.

As well as these recommendations the acoustic survey revealed the relative levels of the background noise in the empty wind tunnel at different wind tunnel speeds.

3.4. Array Design

The following section outlines the processes used in designing a microphone array. The microphone arrays produced for the University of Southampton's 7 by 5 and 11 by 8 wind tunnels were a 64 microphone system arranged in a specially designed spiral [25, 26 and 27] with a radius of 0.35 m. The entire system is composed of an array of microphones linked (in this design's case) to pre-amplifiers to power the microphones. From the pre-amplifiers there is a connection to the data acquisition unit which takes the signal from the microphones in the array and saves the data to a hard drive. The entire system is controlled by a personal computer. With the microphone array design for the Filton wind tunnel there were two primary concerns: The microphone layout and the physical design of the array.

3.4.1. Microphone Placement Design

The microphone placement is determined by a number of factors. The initial concept was to produce an array with many more microphones than the arrays previously built for the University of Southampton. Based on the limitations set by the number of available channels and the cost of each additional channel in terms of time and cost (wiring, pre-amp and microphone cost) the number of microphones was determined to be around 120 for an initial prototype.

The second aspect includes such concerns as building the array around the physical structure of the wind tunnel given the limited nature of the working section with its other measurement and mounting equipment. The decision was made to build the array into the floor of the test section. For convenience the array would be designed to replace floor panels that were easily removed from the test section. The two concerns with the idea that have subsequently become apparent are the limitations this imposes on the overall size of the array and the rails which run into the array. These rails effectively reduce the number of microphones in the array, or at the very least the area in which the microphones could exist. The initial microphone count had to be set higher than the microphone count that was going to be used in the preliminary tests. This was for two reasons; firstly to account for the microphones that would be lost due to the constraints from the wind tunnel structure and secondly to also to give the array the ability to be expanded in the future with more microphones as it is generally agreed that the more microphone in an array the better the results [27 and 28].

The first task after determining the requirements was to plot the microphone layout. The microphone array that was designed was a 253 Microphone array using the Underbrink Spiral design [26, 27], this spiral design was the one used for the design of the University of Southampton arrays [24]. The Underbrink spiral is used as it produces a multi-arm spiral that is particularly effective at sidelobe control and over a broad range of frequencies with a limited number of microphone sensors. A 253 channel array was designed with a view to use only 120 of the microphones initially. The 120 value was selected as this is the number of channels available from the pre-amps and acquisition cards (64+56) as these factors make up the majority of the cost. As well as the two versions of the microphone array, the '253' version reduced to '120' working microphones, a number of other configurations were designed based on the '253' design to explore and maximise the array potential and look into how different configurations affect the results of acoustic tests.

A number of numerical tests can be made on the microphone layout to determine the effectiveness of the design selected. The first is the co-array, a vector spacing view of the array. The Co-array of the microphone layout shows the vectors covered from the microphones layout. It allows easy determination of appropriate microphone layout as it can show any gaps or holes in the array's ability to receive data. The Point Spread Function (PSF) is another test that can be made to microphone array designs to determine their suitability. As the array design was modified to fit around wind tunnel structure several problems became evident with results from the point spread function. The point spread function is a mathematical analysis of a microphone array layout which can be used to determine how good the mainlobe response will be compared to the sidelobe. The point spread function also shows the distribution of the sidelobes with respect to the mainlobe. Better arrays have lower sidelobes that are spread far away from the mainlobe. The PSF also allows the resolution requirement to be determined, if the distance between the test model and the wall of the wind tunnel is larger than the beamwidth (the width of the mainlobe in the PSF) then the array will be unable to separate the real source of the noise from any image or reflected source. [26]. The PSF shows levels at certain frequencies from the microphones layout. The figures on the following pages show the different frequency point spread functions in comparison to each other; showing how the original unaltered design returns excellent results and how the multiple numbers of microphones set up in to different configurations. Although a wide range of frequencies are used in this report only 1, 5, 10 and 20 KHz are shown, as this provides a good range over the expected frequency range that will be measured. The higher and lower values also mark the expected maximum and minimum capabilities of the array due to microphone quality and the overall size of the array.

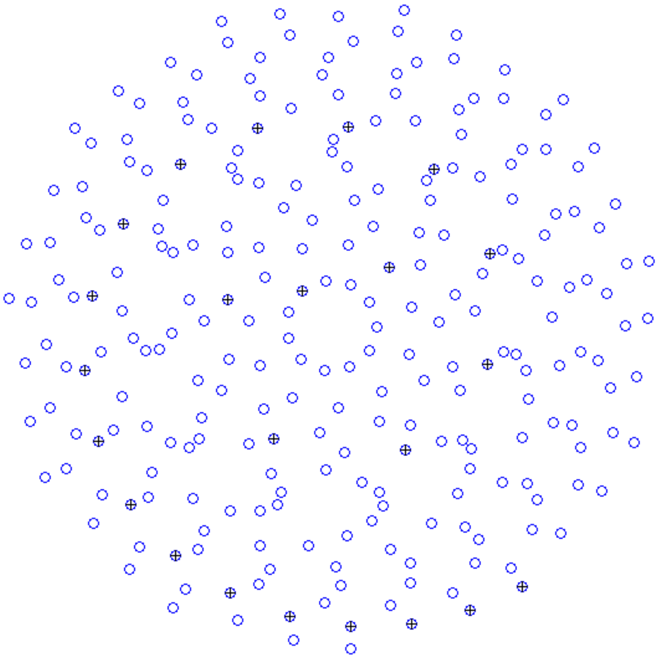


Figure 29 – Array design of 253 microphone positions arranged in an Underbrink Spiral pattern.

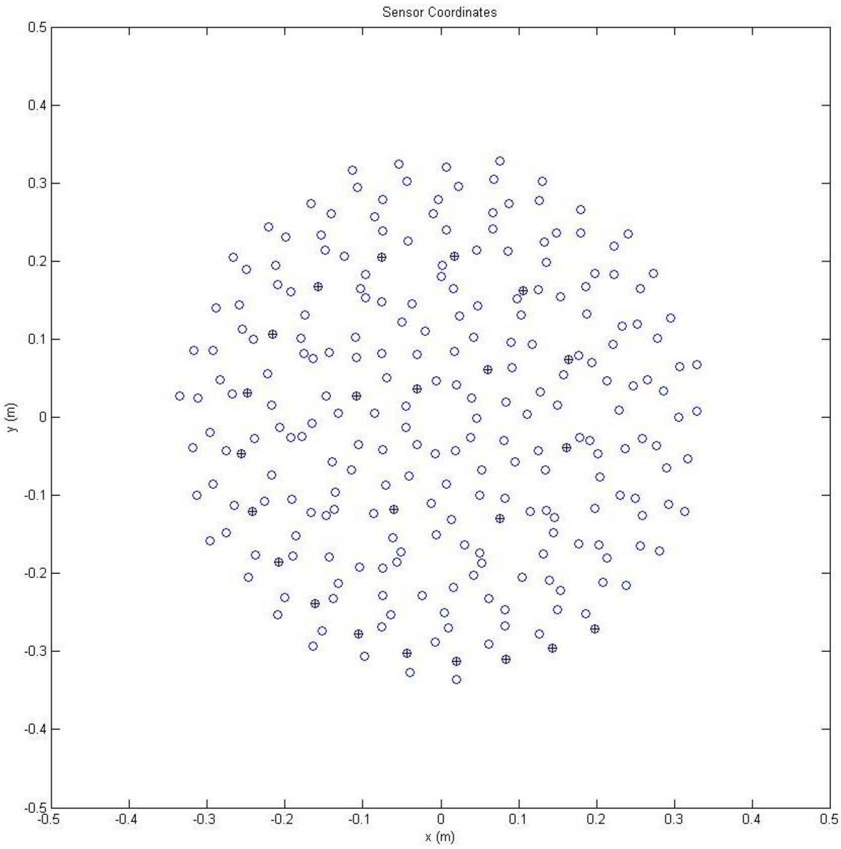


Figure 30 – Array design showing overall size and location of each microphone and the initial spiral that was used to plot the remaining microphone positions.

The darker spots show the initial spiral from which the rest of the microphone positions were derived. The positions are then determined by copying the spiral in a circular fashion. The overall size of the array is based on measurement of the Filton wind tunnel and is explained later. From these designs the co-ordinates are already available to create the main plate easily.

The following images are the point spread function for 1, 5, 10, 15 and 20 KHz frequencies showing an ideal response. Note that the sidelobes are low and spread far from the central peak and that the central peak itself is relatively thin – indicating high resolution at the frequency indicated.

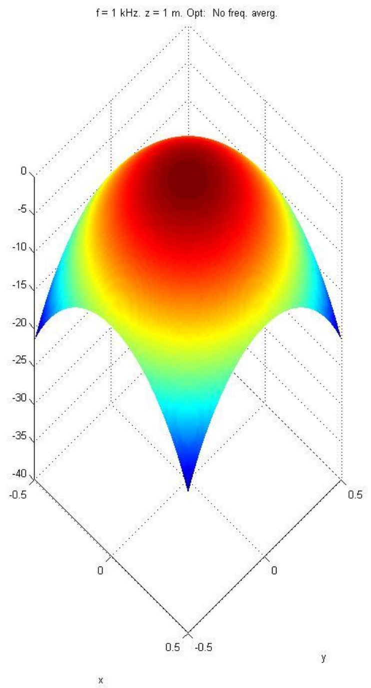


Figure 31 – Array point spread function for 253 microphones at 1 kHz frequency, view showing peaks relative levels.

The above image shows the mainlobe of the response, at low frequency, due to the small size of the array diameter, the response is expected to be low resolution. At higher frequencies the mainlobe should become thinner (it’s beamwidth decreasing) and sidelobes become present as can be seen in the following figures.

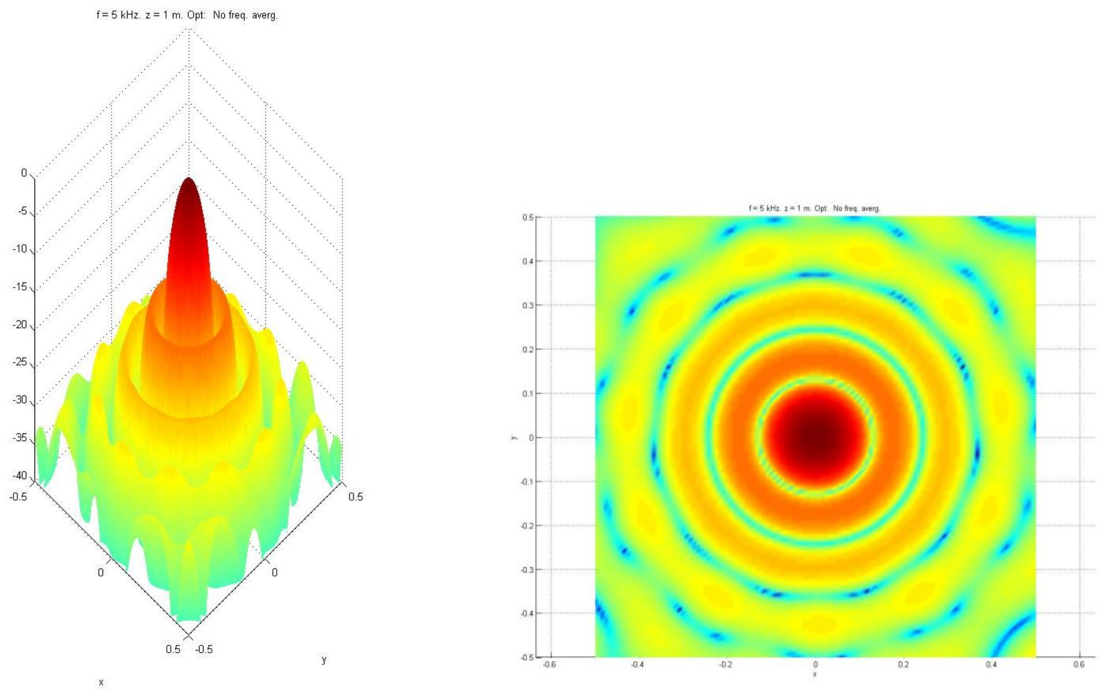


Figure 32 – Array point spread function for 253 microphones at 5 kHz frequency, views showing peaks relative levels and shape of array response.

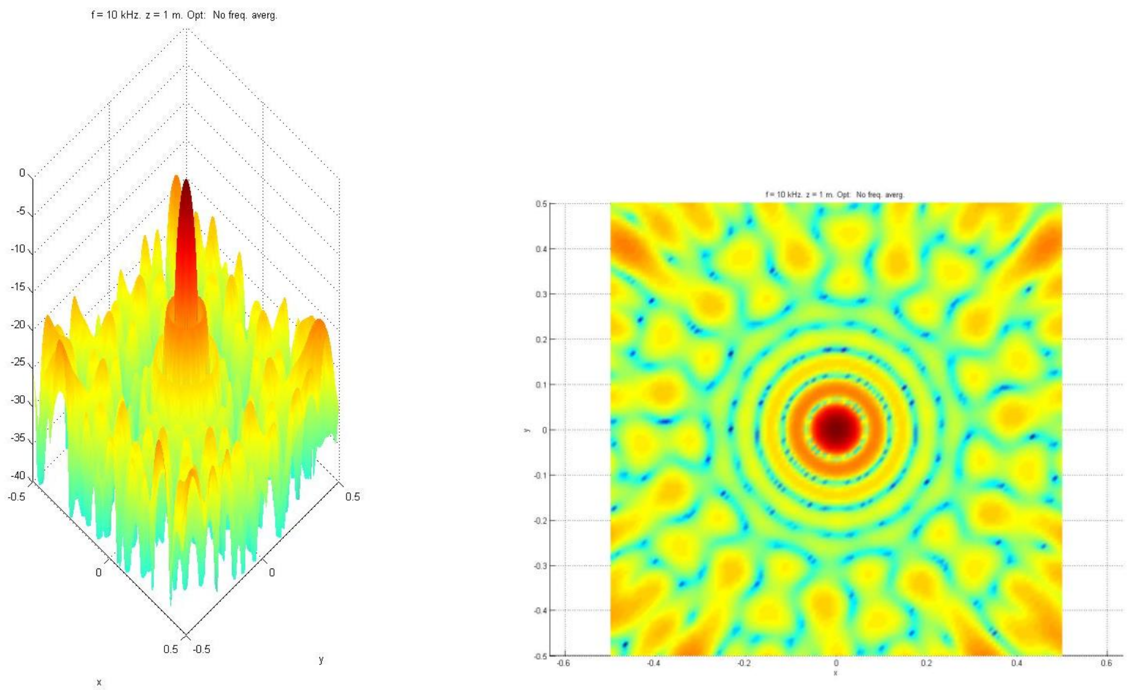


Figure 33 – Array point spread function for 253 microphones at 10 kHz frequency, views showing peaks relative levels and shape of array response.

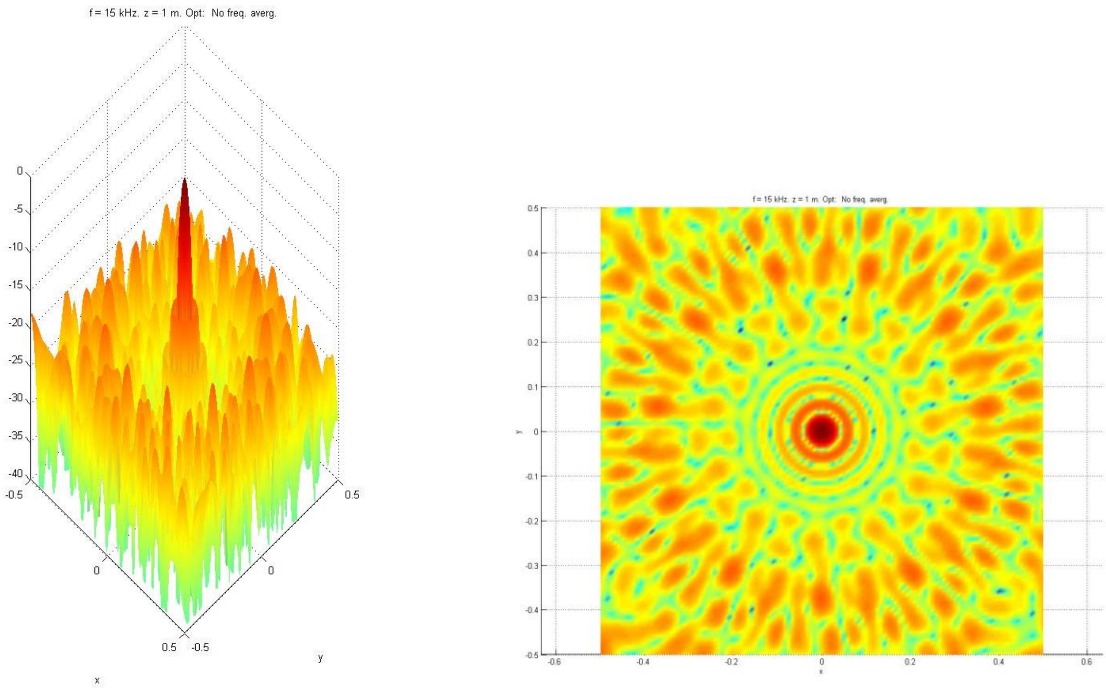


Figure 34 – Array point spread function for 253 microphones at 15 kHz frequency, views showing peaks relative levels and shape of array response.

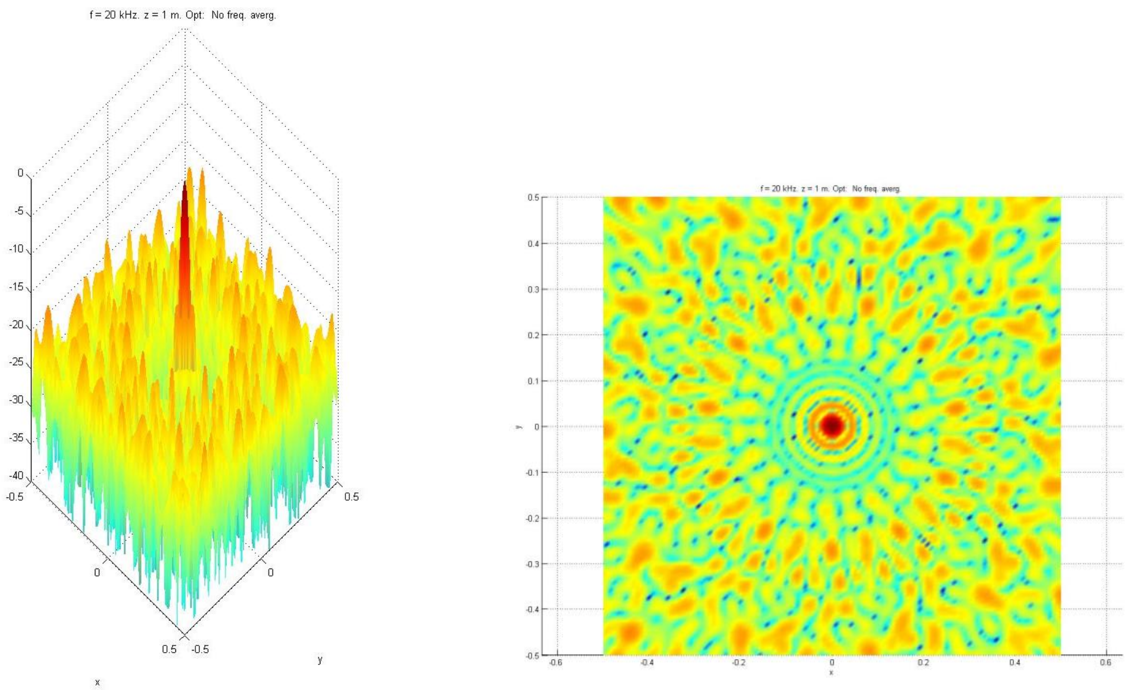


Figure 35 – Array point spread function for 253 microphones at 20 kHz frequency, views showing peaks relative levels and shape of array response.

The following figures show the co-array for the 253 microphone array set-up. It clearly shows the total covering over the area that the array extends with no gaps.

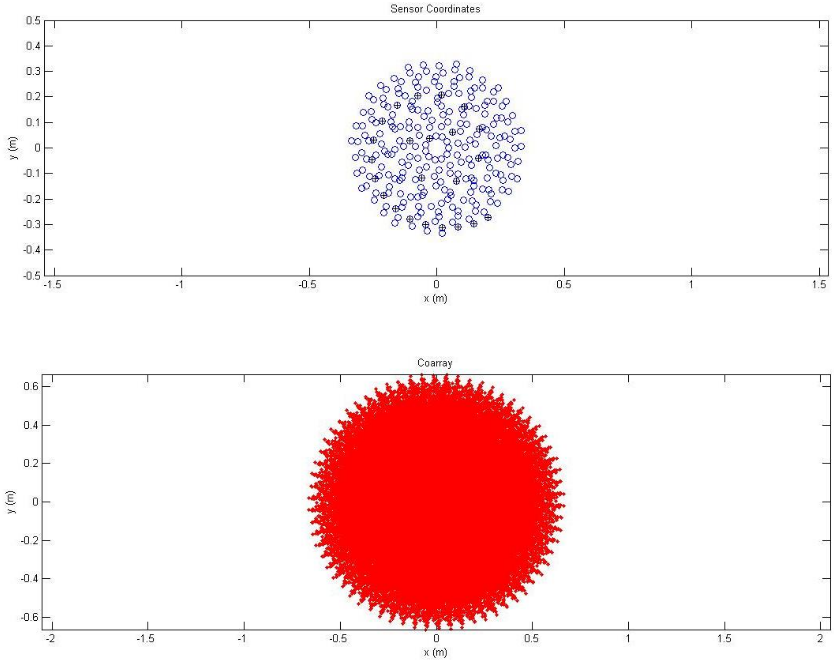


Figure 36 – Co-array for the 253 microphone array set-up.

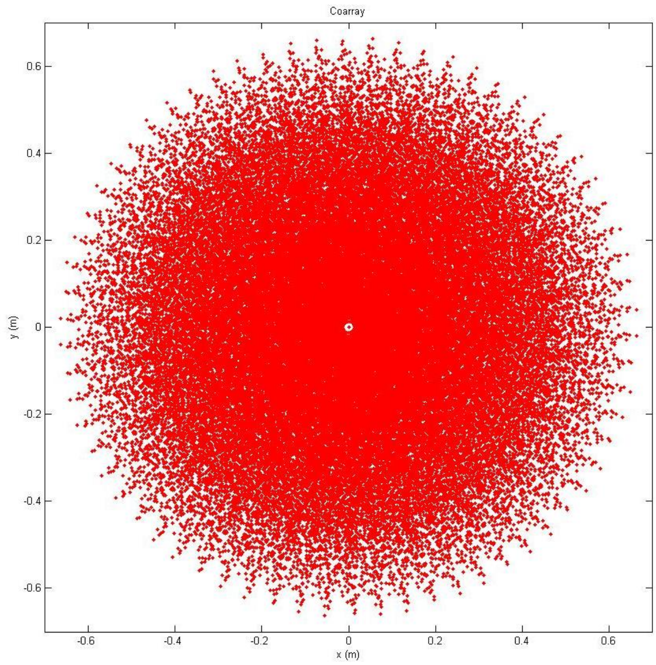


Figure 37 – Co-array for the 253 microphone array set-up.

The existing wind tunnel facility was not designed with acoustic testing in mind. Unsurprisingly there is not a conveniently large, clear unobstructed area close to the positioning of the wind tunnel model that allows a perfect array to be installed. The following figures shows the plan view of the Filton wind tunnel showing the location of the wind tunnel test section central circular rig and the associated plates, beams and rails that are found there. In constructing an array for this test section, all these components must be taken into account and the array must be designed with them in mind.

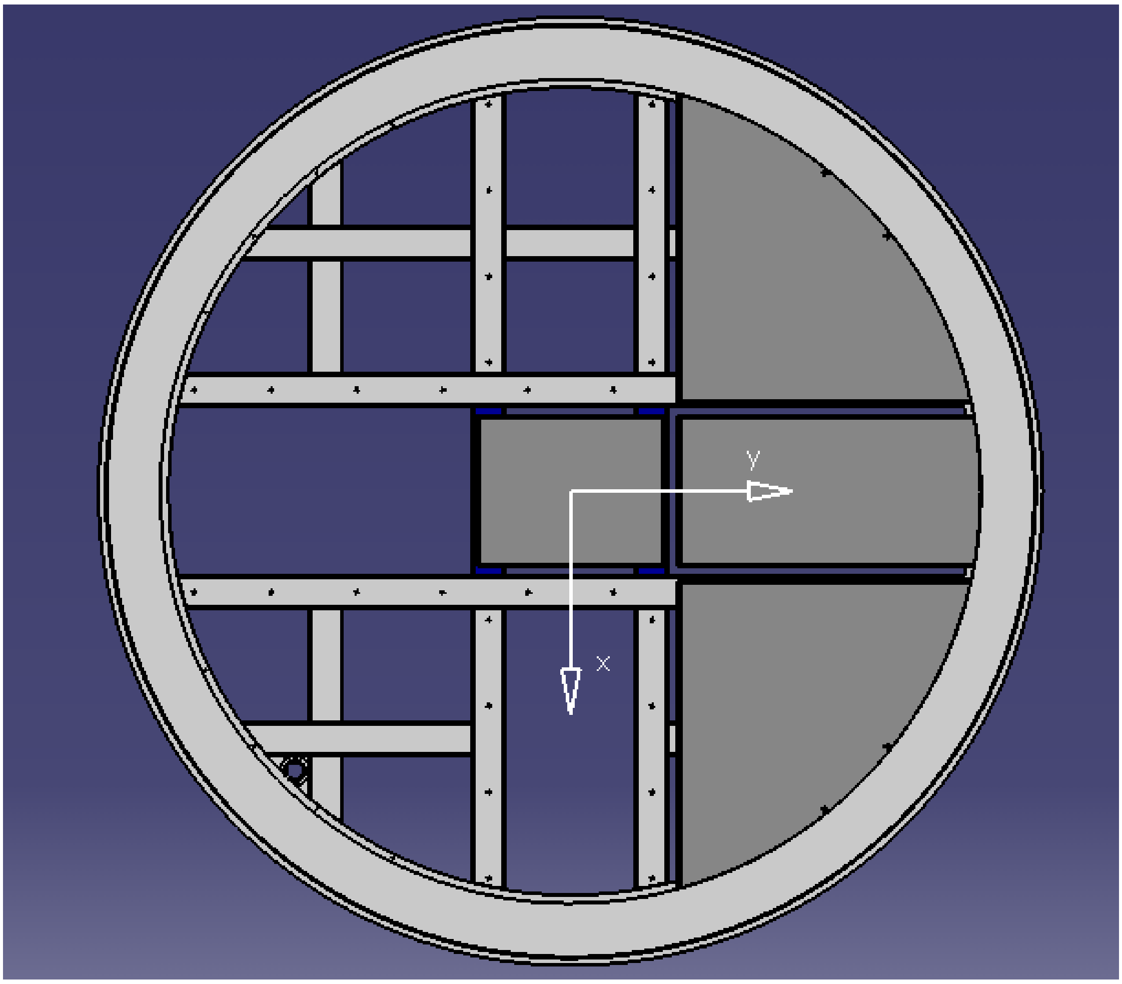


Figure 38 – Image of the Filton Wind Tunnel test section turntable.

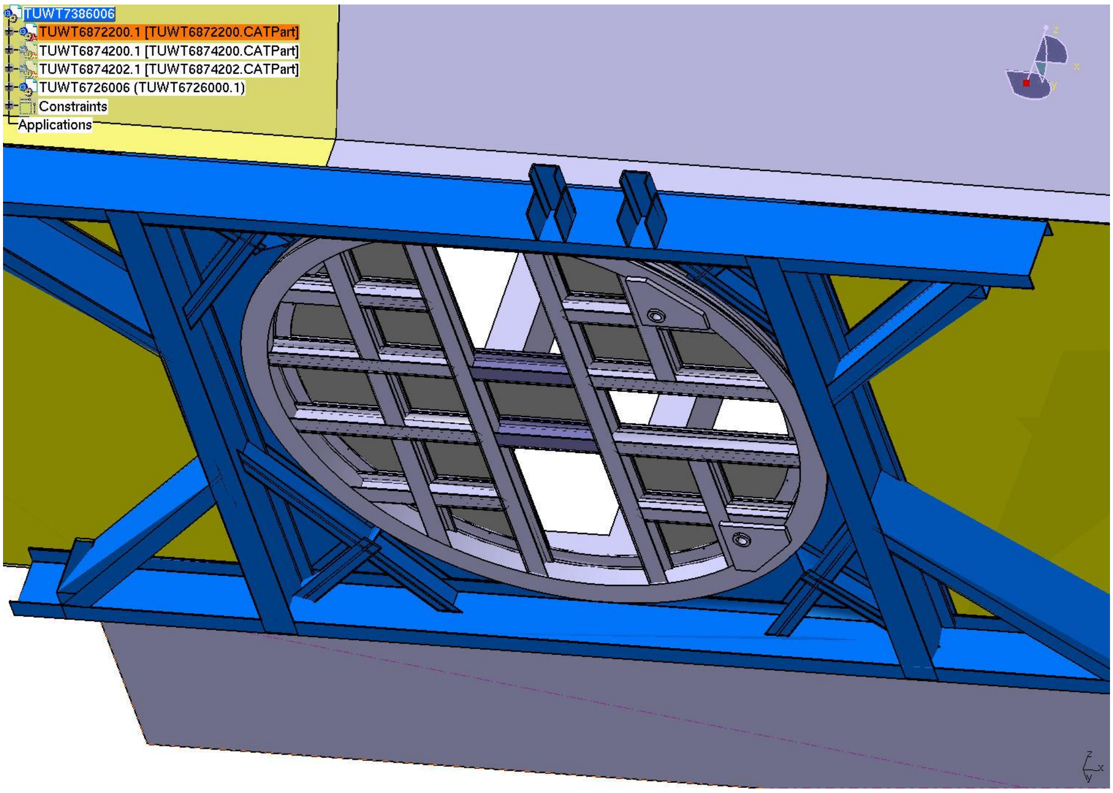


Figure 39 – View of the underside of the Filton wind tunnel test section.

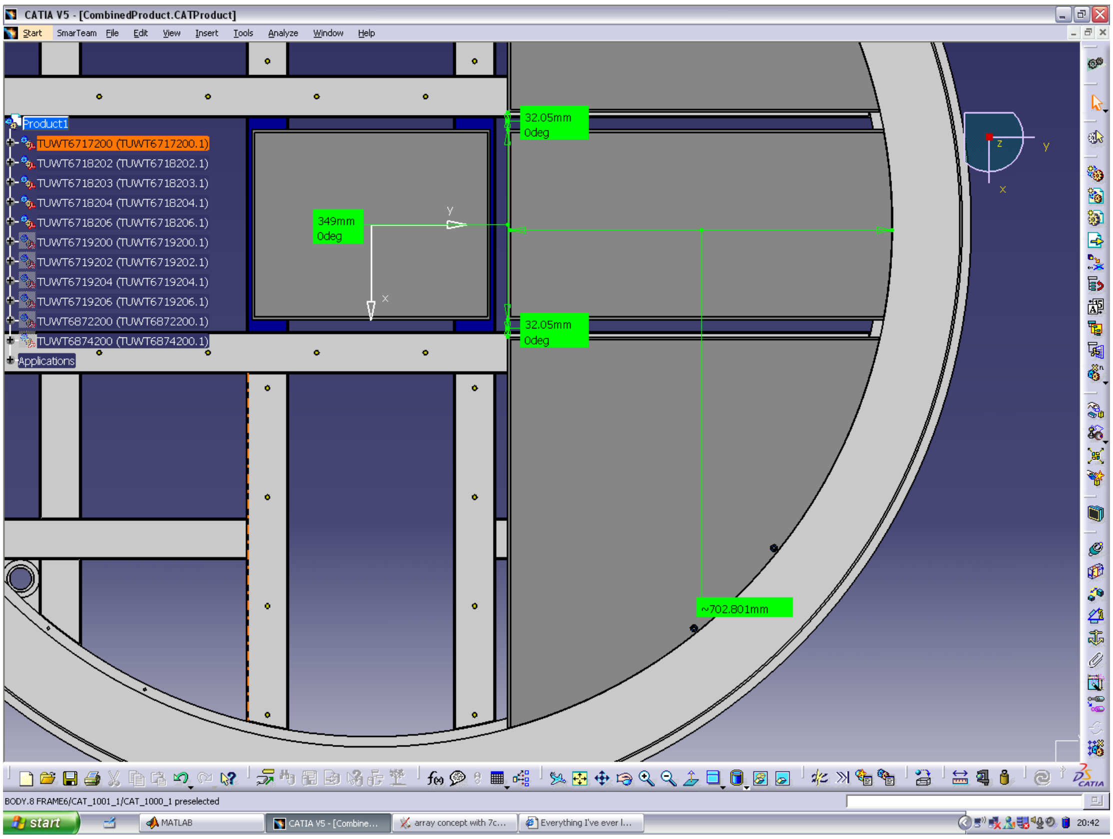


Figure 40 – Measurements of the rails between the plates in the floor of the Filton wind tunnel test section turntable.

As can be seen by the figures, the area of the test section floor is composed of a series of beams, rails and plates arranged in a circular fashion. Models are mounted over the centre of this circular area, with the areas of acoustic interest being over one side of the circular area. The wind in the figure above flows from the top to the bottom, whilst in the other figures wind flows from left to right.

With the measurements given in figure 40 the centre origin of the array is located at a dimension of 606.985 mm from the centreline of the tunnel, this distance would typically place the array under the wing of most models used in the wind tunnel. Whilst variations in model size and scale may mean that the array might not be directly beneath the wing of the model the origin of the array is located such that the whole wing can be seen. With smaller, scale models, the opportunity exists to view the entire length of the models, however, in general most models are mounted on a ground strut, so viewing the far side of any model is hampered and as a rule far side acoustic plots are not generally taken.

Previous designs in the University of Southampton 7 by 5 wind tunnel were installed in the wall [24] and (as with the 11 by 8) in the ceiling. Whilst a wall mounted array would be possible in the Filton wind tunnel, it would mean the array was looking at an acoustic image that was perpendicular to the plane of interest, which would not be useful for determining any noise source location. Thus the floor mounted array would be the most ideal location to place an acoustic array as it would be directly in line with the test model and would not interfere with the flow or other components of the wind tunnel. Additionally the plates as seen in Figures 38, 39 and 40 are easily (comparatively to other parts of the wind tunnel) removed and replaced by an acoustic array.

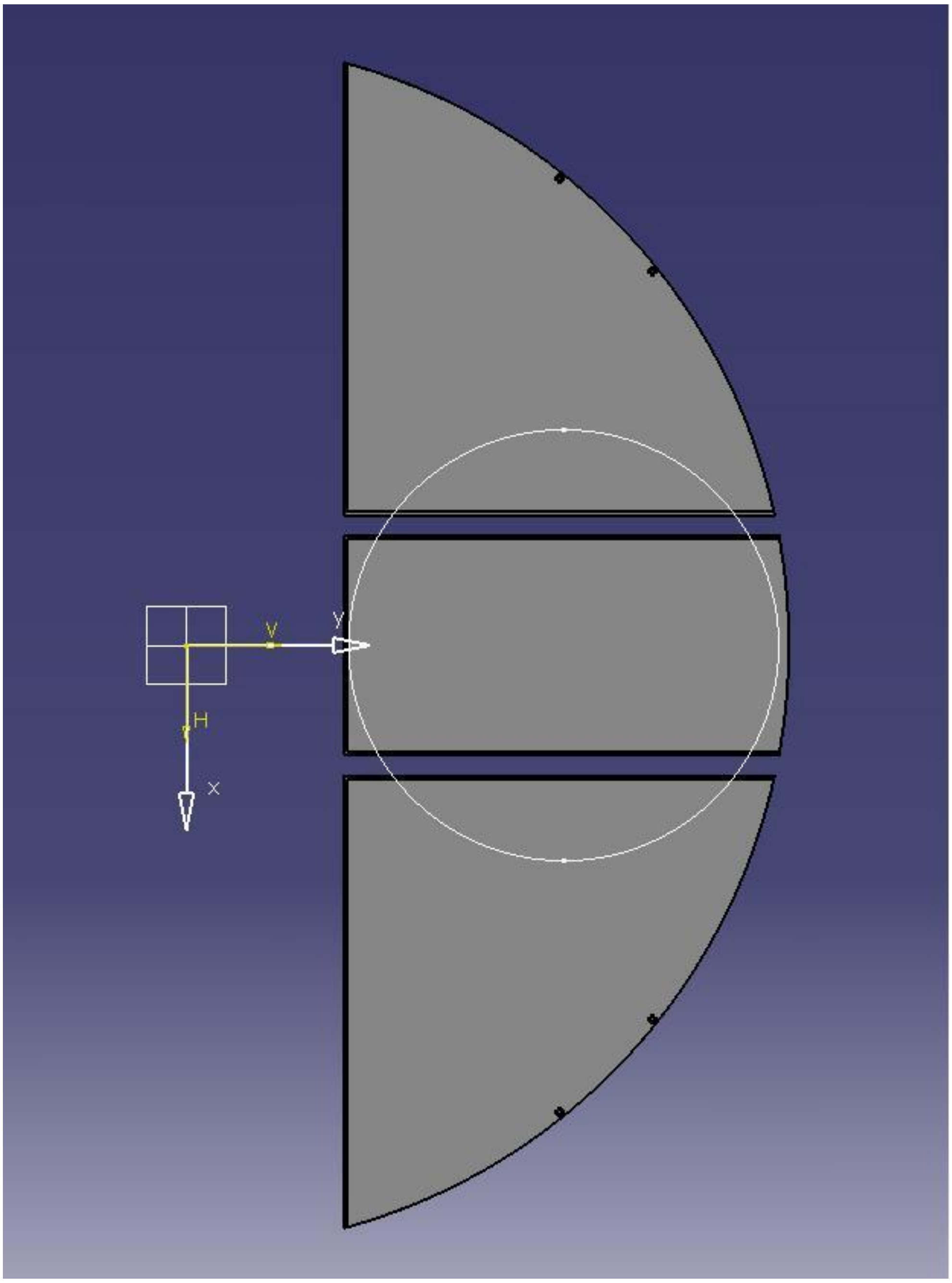


Figure 41 – Proposed location of the prototype array in the circle indicated across the 3 installation plates.

The design concept is to remove the 3 plates as seen in figures 38 and 40 and replace them with three plates that have the microphones installed (in the spiral configuration). With the measurements taken from the plates and interlaying rails the following microphone array is derived from the initial 253 microphone design (note that for the following diagrams the flow of the wind is now left to right not top to bottom as in figures 38 - 40) .

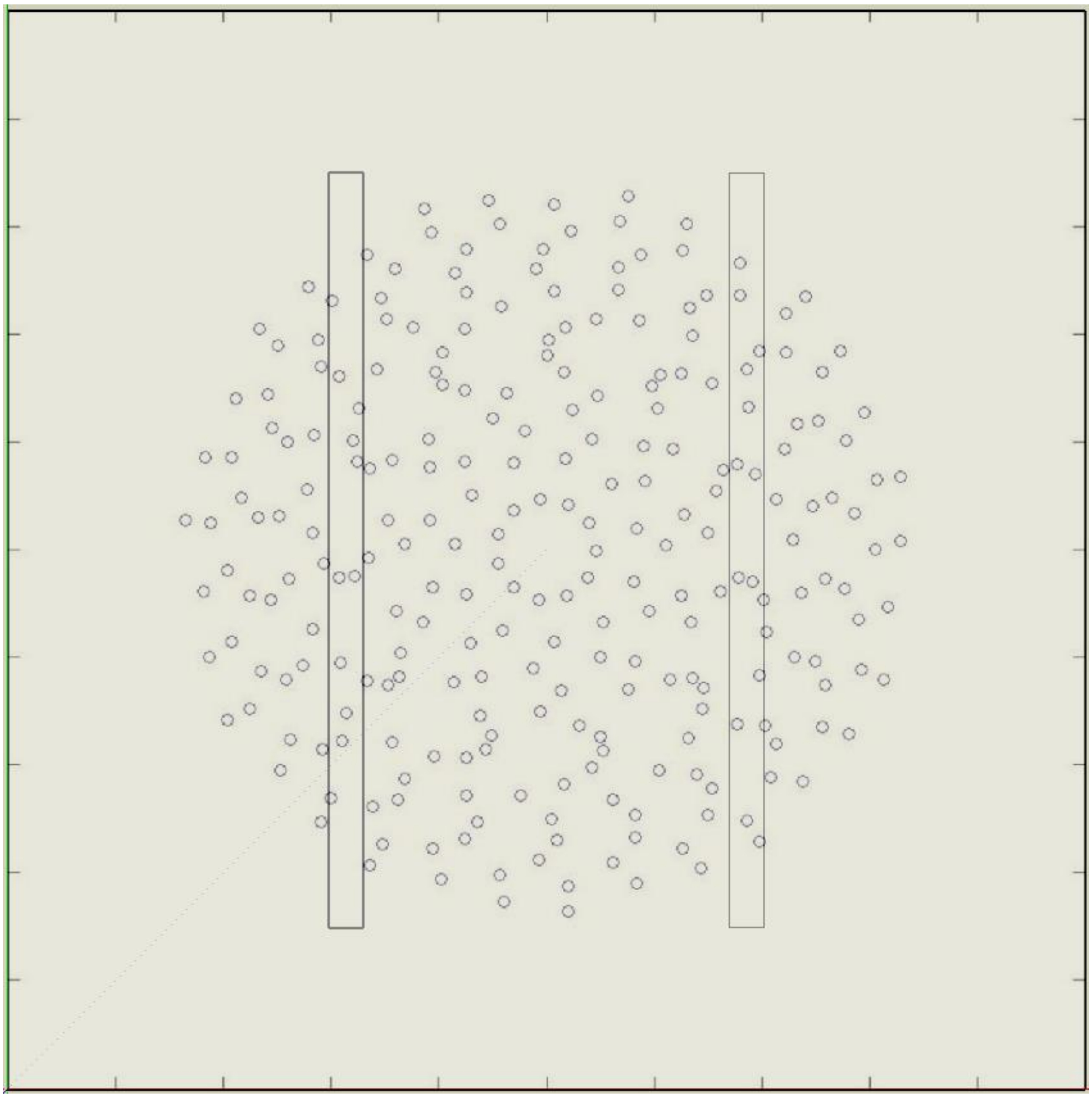


Figure 42 – Proposed location of the prototype array in the circle indicated, showing location of interfering beams.

The above diagram shows the location of the beams and rails that would interfere with the microphone array.

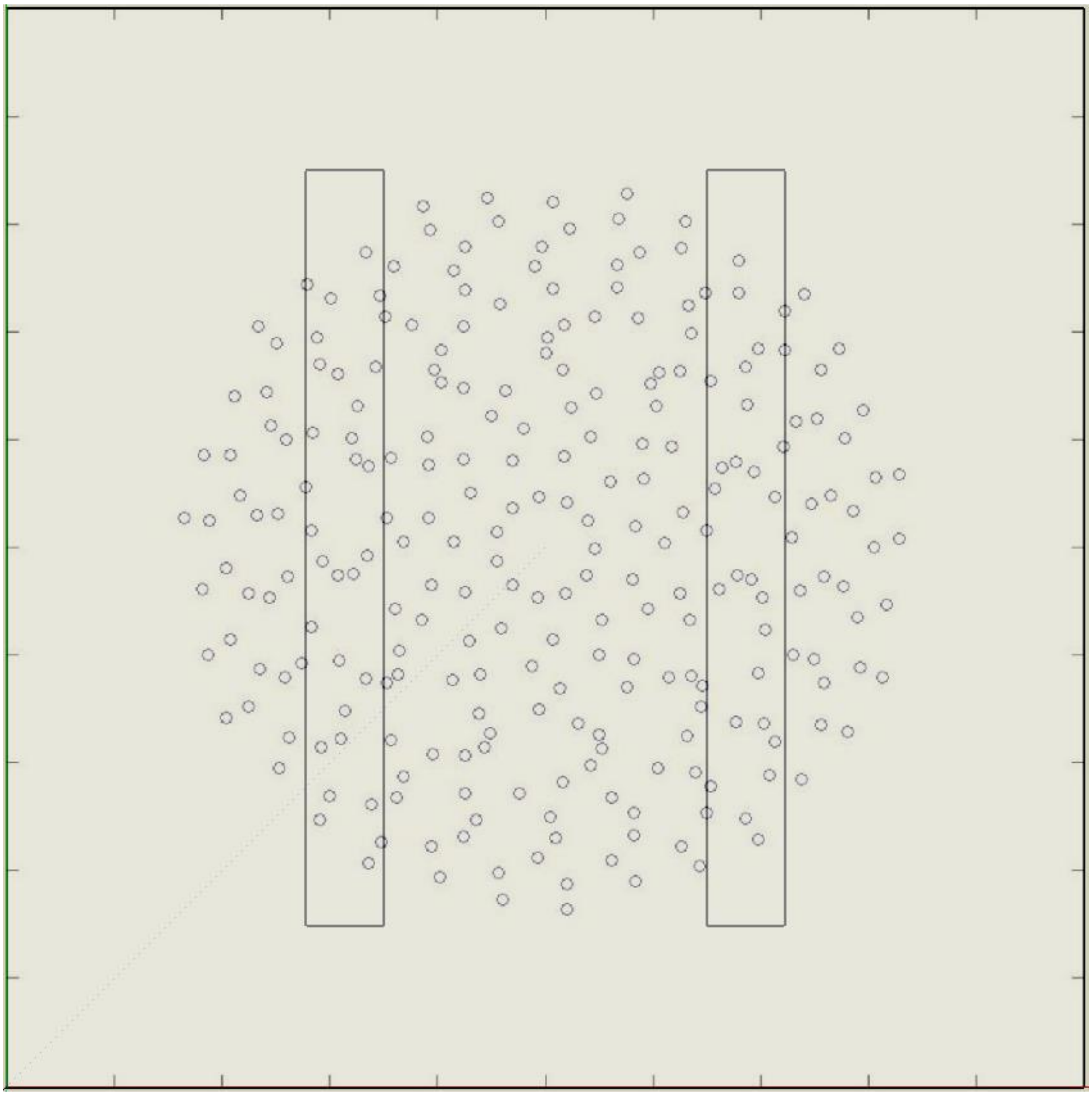


Figure 43 – Proposed location of the prototype array in the circle indicated, showing clearance required for access.

A 20 mm gap either side of the rails was added (so the total gap distance for one rail is 72 mm) this was due to the physical limitations such as access requirements and necessary clearance. The effective exclusion zone for the array is thus shown above in figure 43.

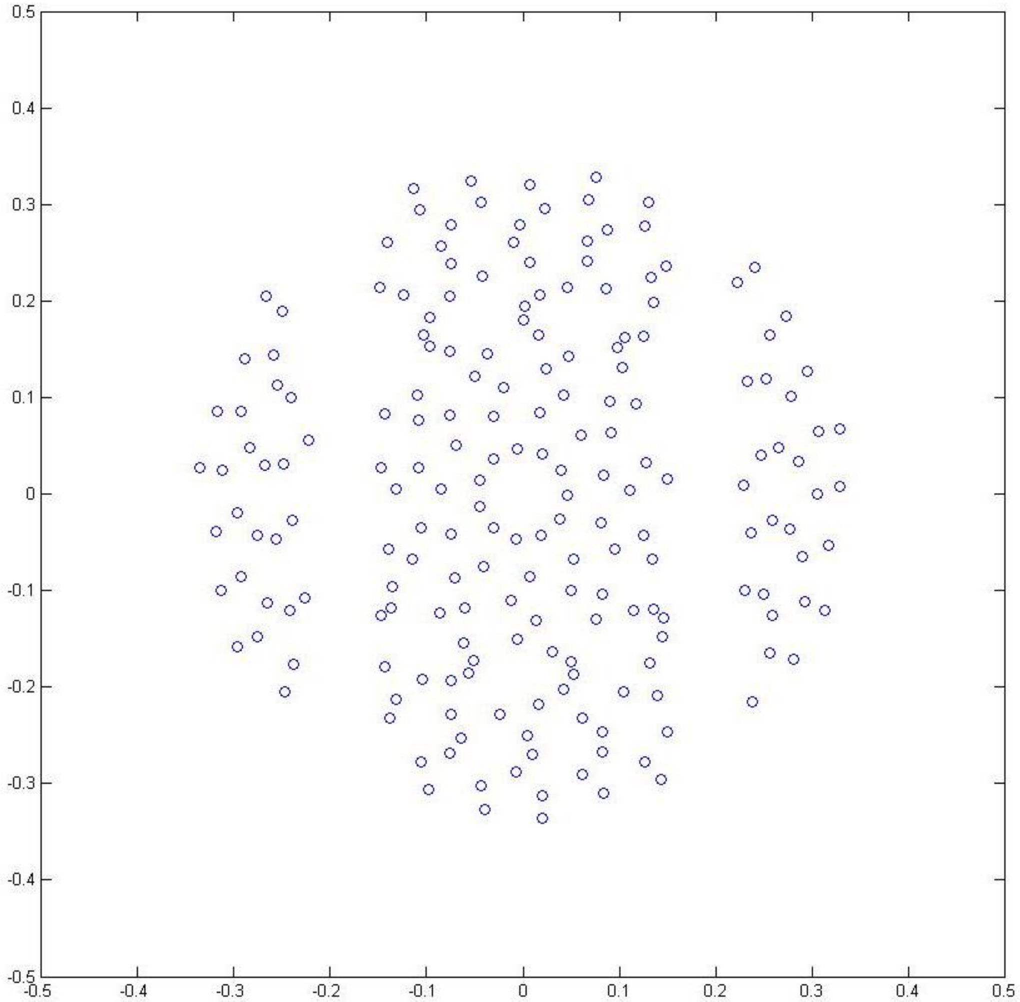


Figure 44 – Microphone layout taking into account the rails in between the plates.

The above diagram shows the microphone array with the microphones in the exclusion zone removed. The original design had 253 microphones and with the ‘lost’ microphones bringing the number down to 198 due to the rails in the floor of the wind tunnel.

The following images show the point spread function for the microphone layout as seen in figure 44.

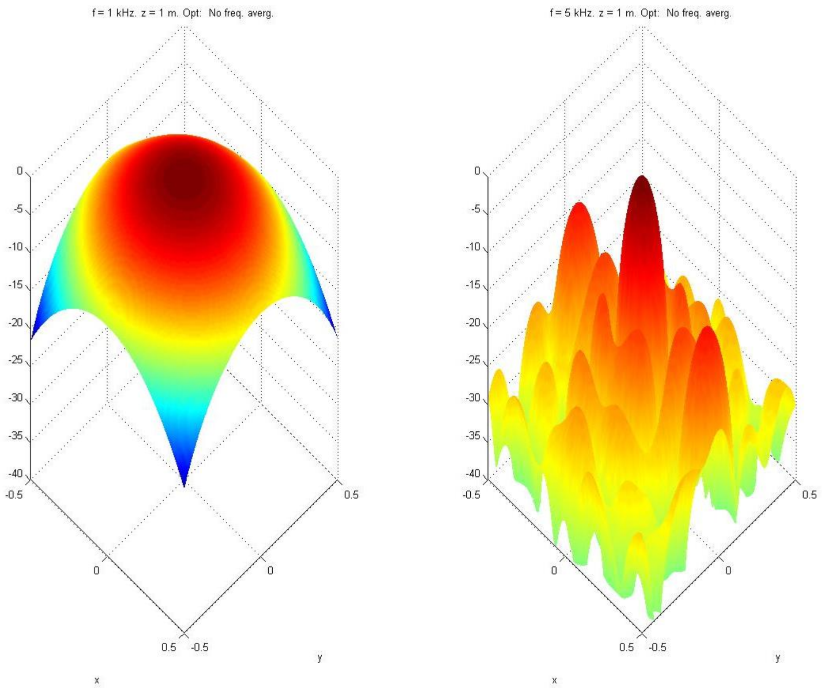


Figure 45 – Array point spread function for microphones at 1 kHz (left) frequency and 5 kHz (right) frequency.

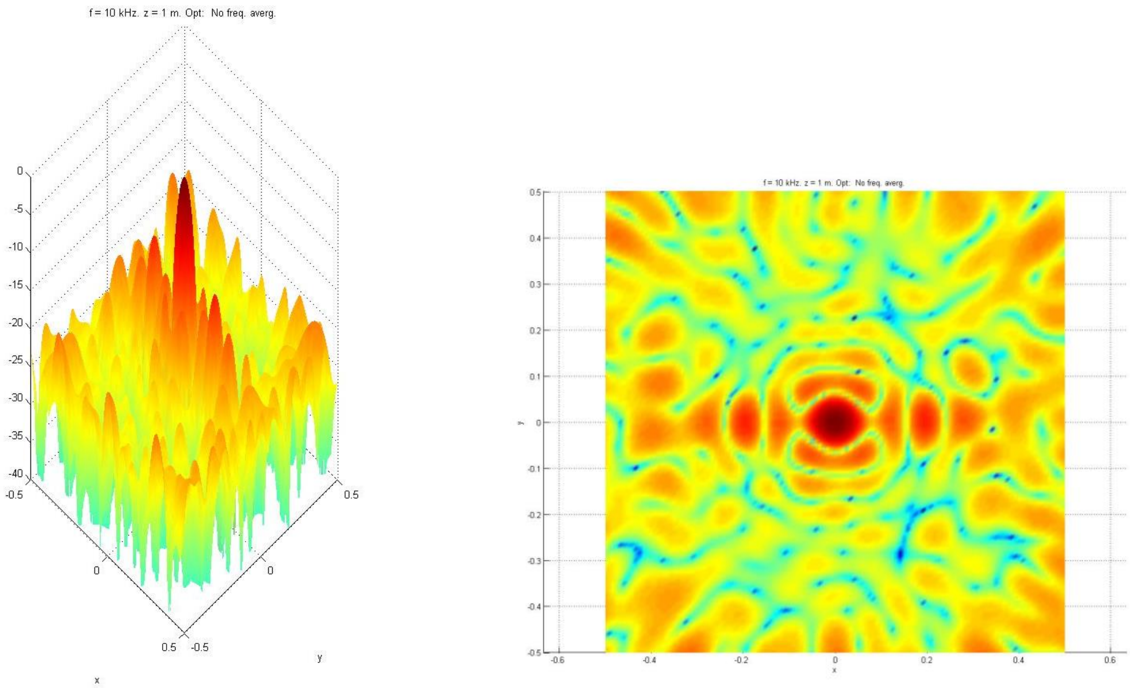


Figure 46 – Array point spread function for microphones at 10 kHz frequency, views showing peaks relative levels and shape of array response.

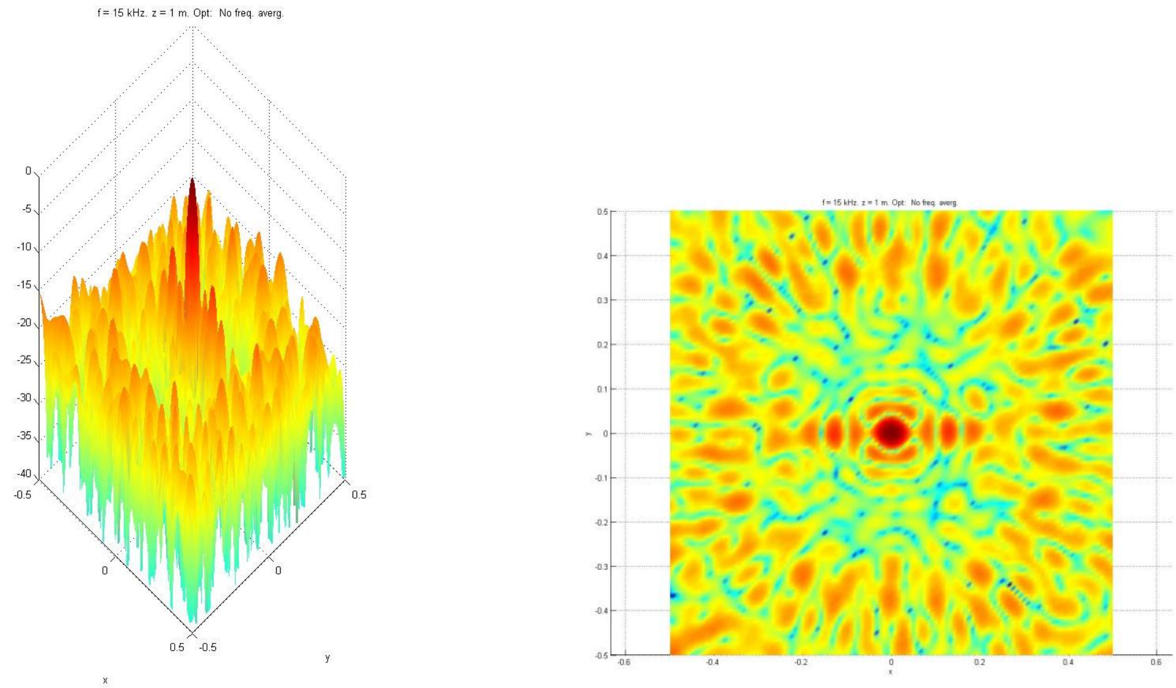


Figure 47 – Array point spread function for microphones at 15 kHz frequency, views showing peaks relative levels and shape of array response.

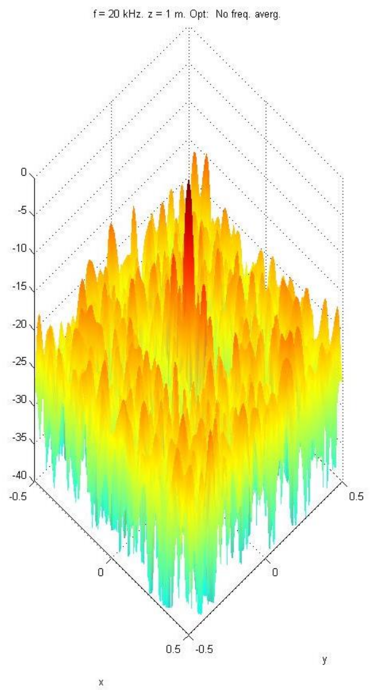


Figure 48 – Array point spread function for microphones at 20 kHz frequency.

From the results it can be seen that since the microphones in the place of the rails have removed a number of microphones and also unbalanced the array configuration the psf shows a non-circular result and sidelobes which are unacceptably close to the main peak.

A number of configurations were attempted to correct the imbalance in psf and return the ‘perfect’ response as seen in the 253 microphone configuration psf.

The following microphone arrangement is a design concept removing more microphones from the previous configuration in an attempt to rebalance the array and restore the point spread function response.

The total number of microphones left in this design is 127:

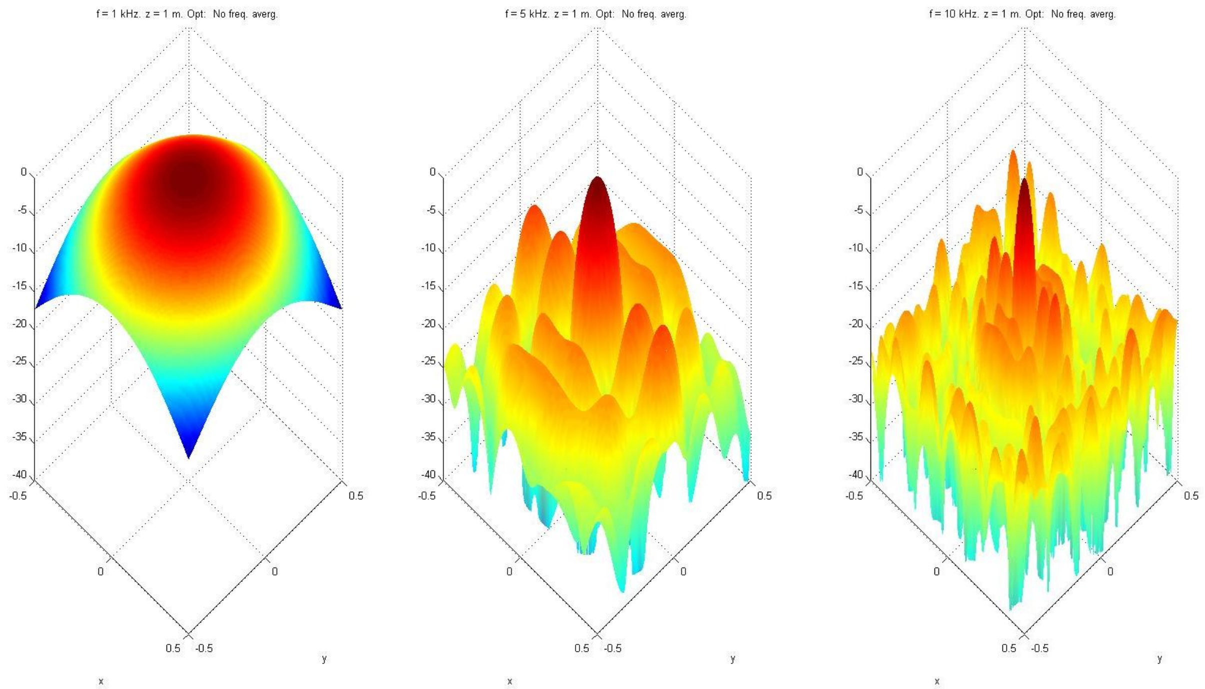


Figure 49 – Array point spread function for microphones at 1, 5 and 10 kHz frequency for a 127 microphone array.

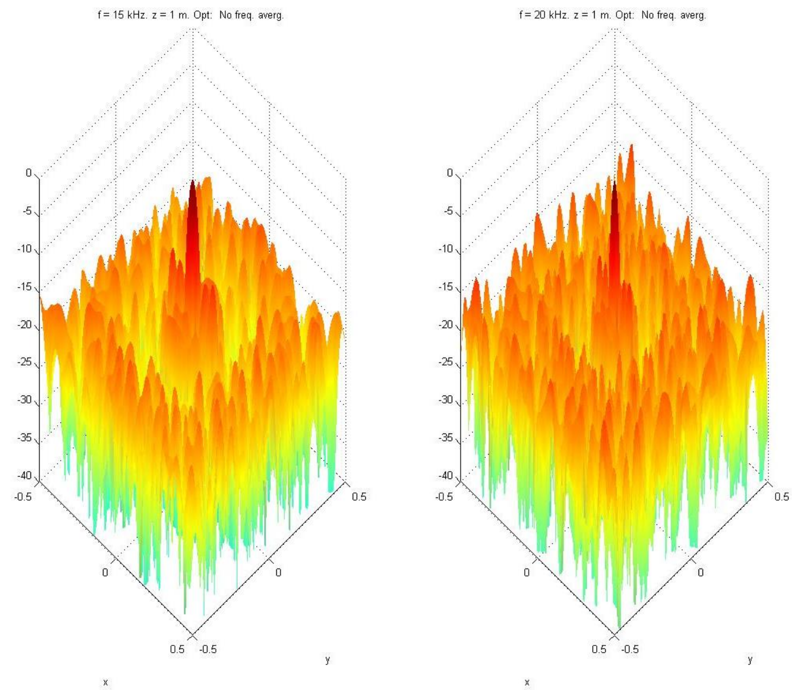


Figure 50 – Array point spread function for microphones at 15 kHz frequency and 20 kHz frequency for a 127 microphone array.

The above results still shows sidelobes very close to the main peak and the response is still somewhat unacceptable for accurate results. Looking to improve the response an array composed of just the centre microphones was tested. The following psf results were obtained.

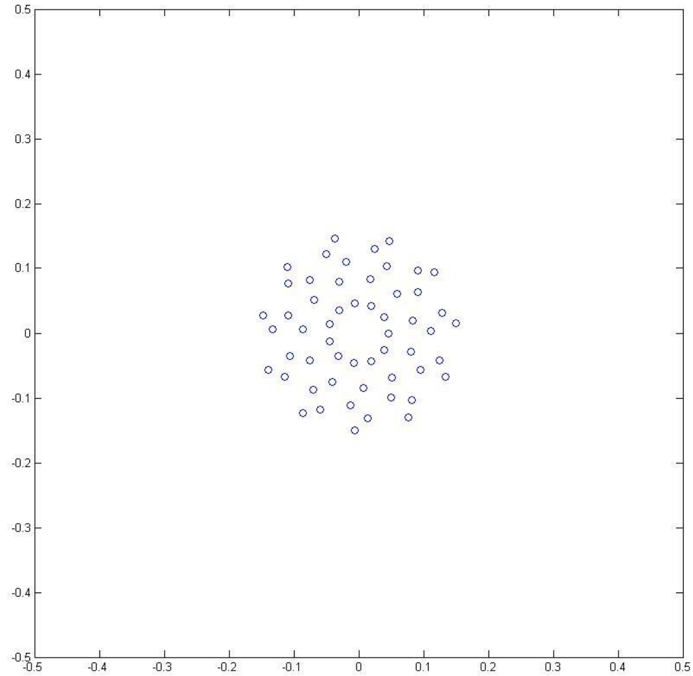


Figure 51 – Array configuration of the centre 55 microphones.

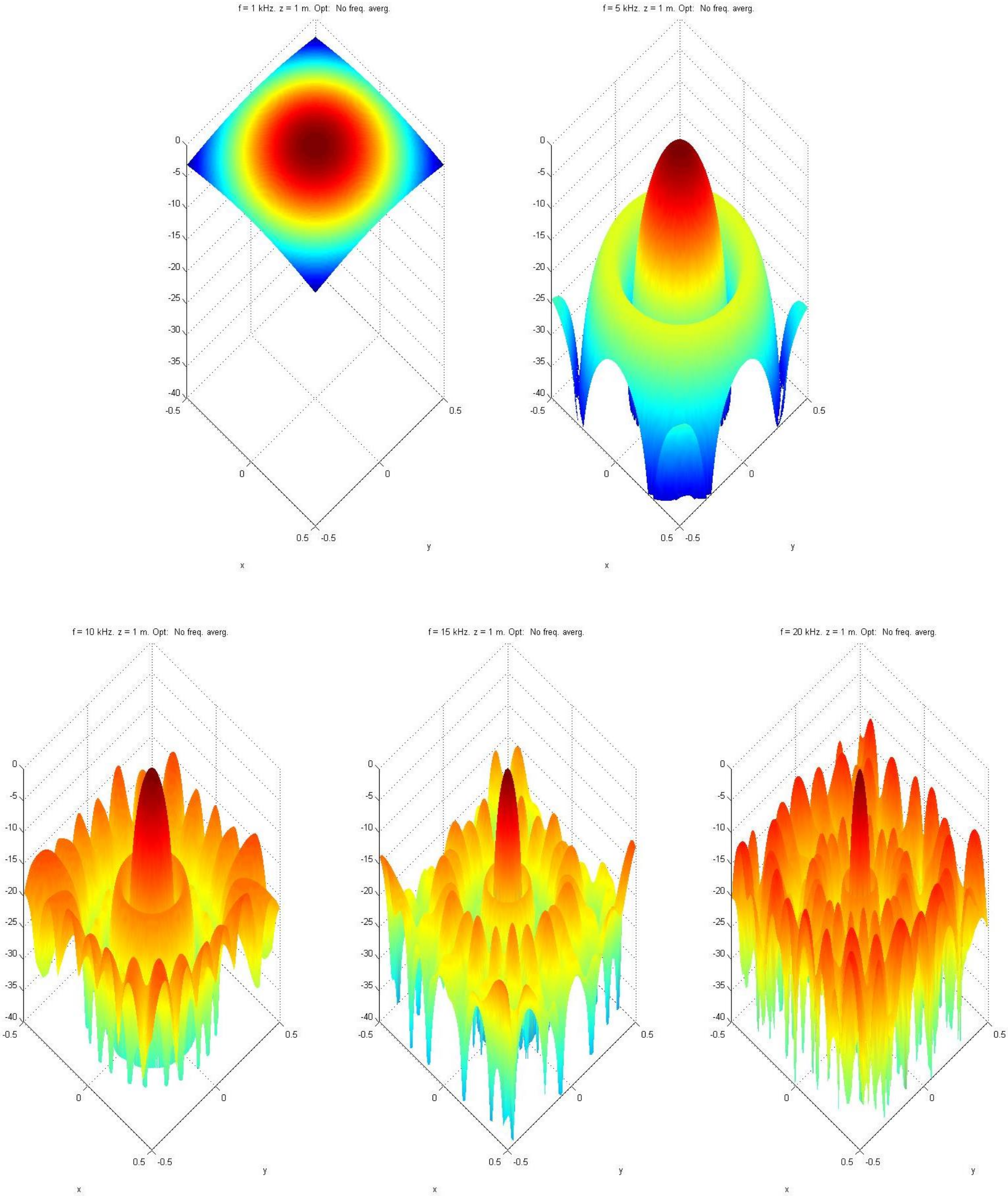


Figure 52 – Array point spread function for microphones at 1 kHz, 5 kHz, 10 kHz, 15 kHz and 20 kHz frequency for the centre 55 microphones.

As can be seen the response for the central 55 microphones is very good. Uninterrupted balanced spirals return excellent responses from the psf. To maximise on the potential of the uninterrupted centre plate it was decided to increase the maximum number of microphone in the centre, to this effect an additional 11 microphone placements were located in the centre circle. This has the effect of increasing the microphone count in the central plate and the total microphone count raises up to 264.

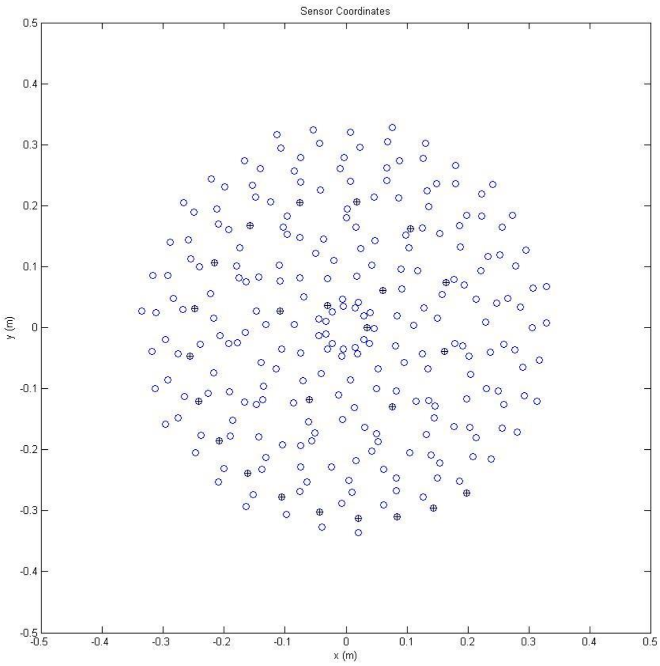


Figure 53 – Array of microphones with additional 11 central circle bringing the total count of sensors to 264, as can be seen the original spiral pattern is utilised.

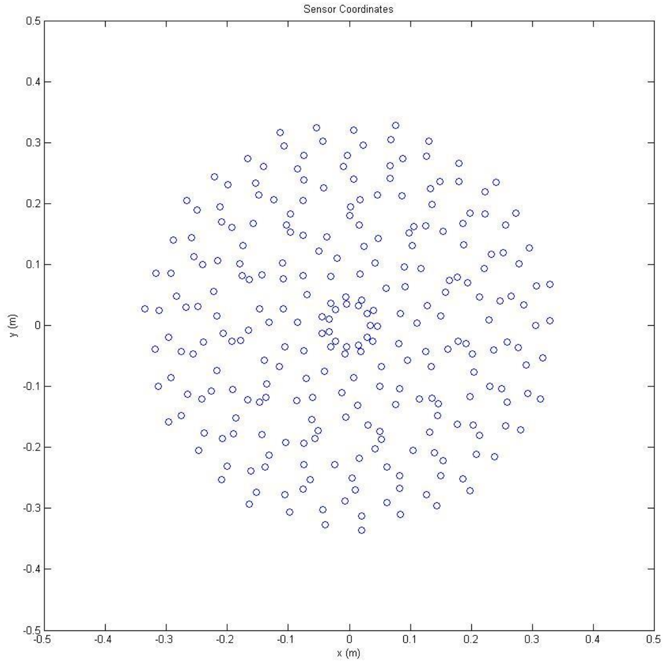


Figure 54 – Array of microphones with additional an 11 central circle bringing the total count of sensors to 264.

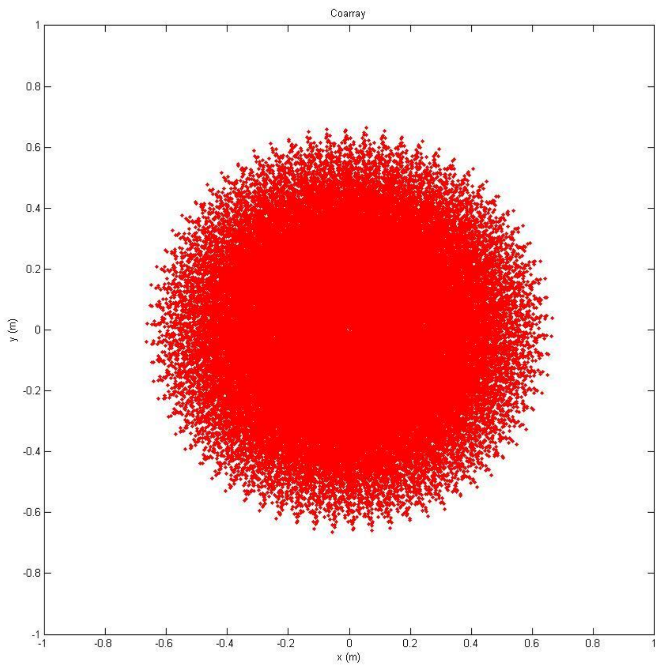


Figure 55 – Array of microphones of 264 sensors with associated coarray showing the coverage is if anything improved by additional sensors.

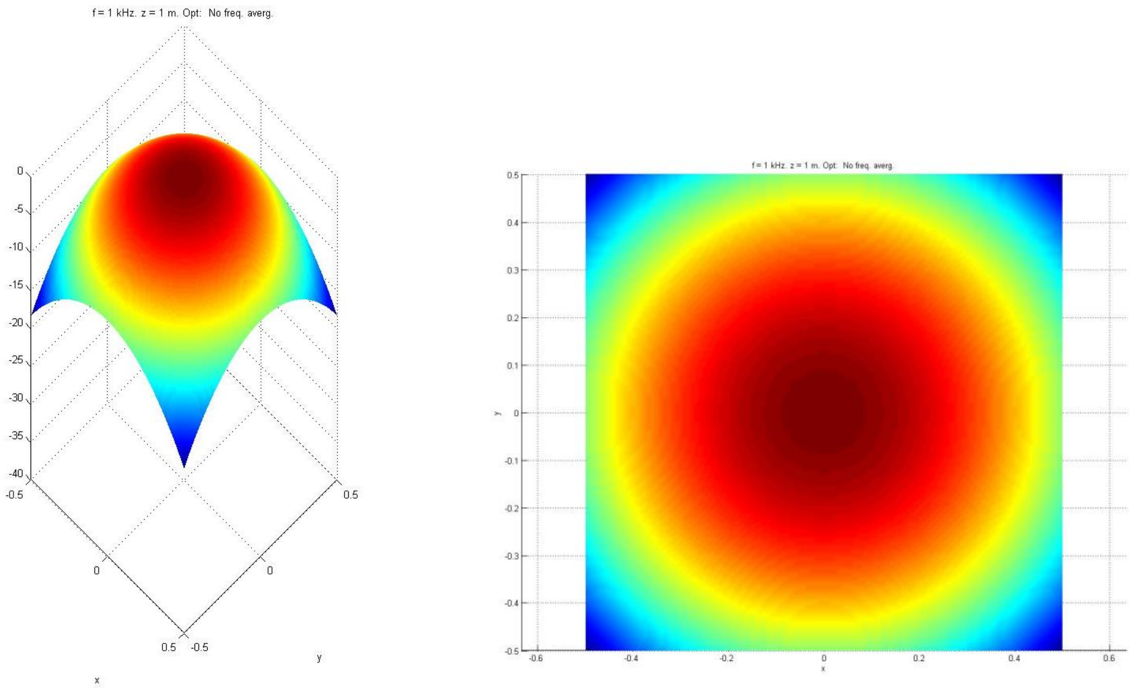


Figure 56 – Array point spread function for 264 microphones at 1 kHz frequency, views showing peaks relative levels and shape of array response.

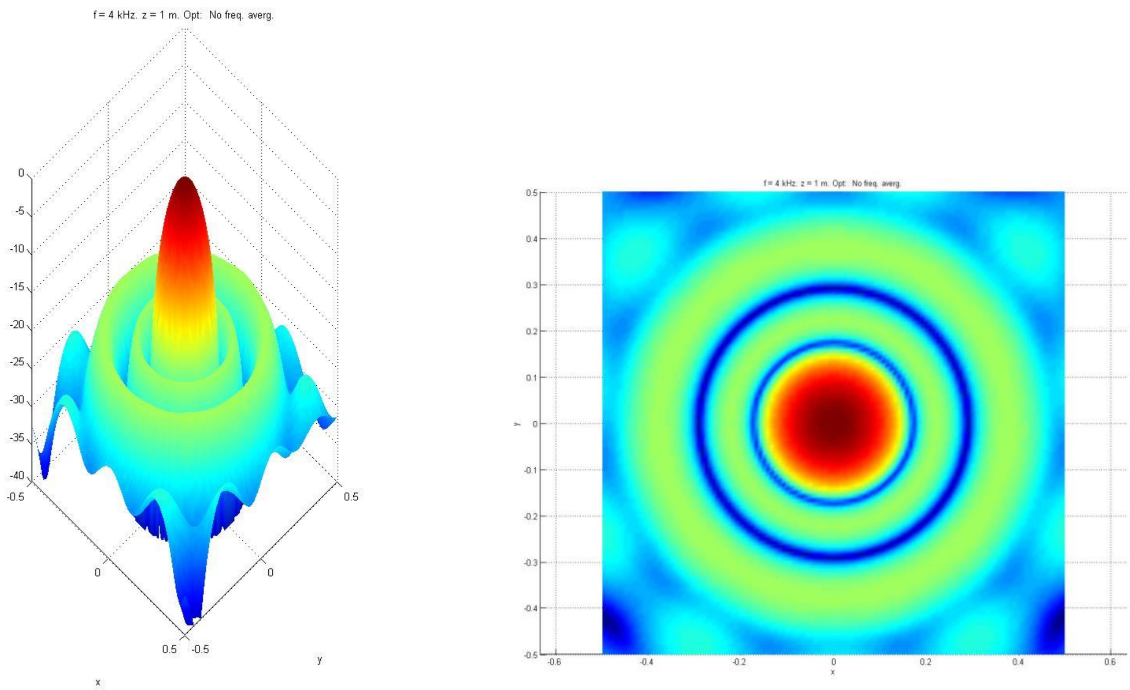


Figure 57 – Array point spread function for 264 microphones at 4 kHz frequency, views showing peaks relative levels and shape of array response.

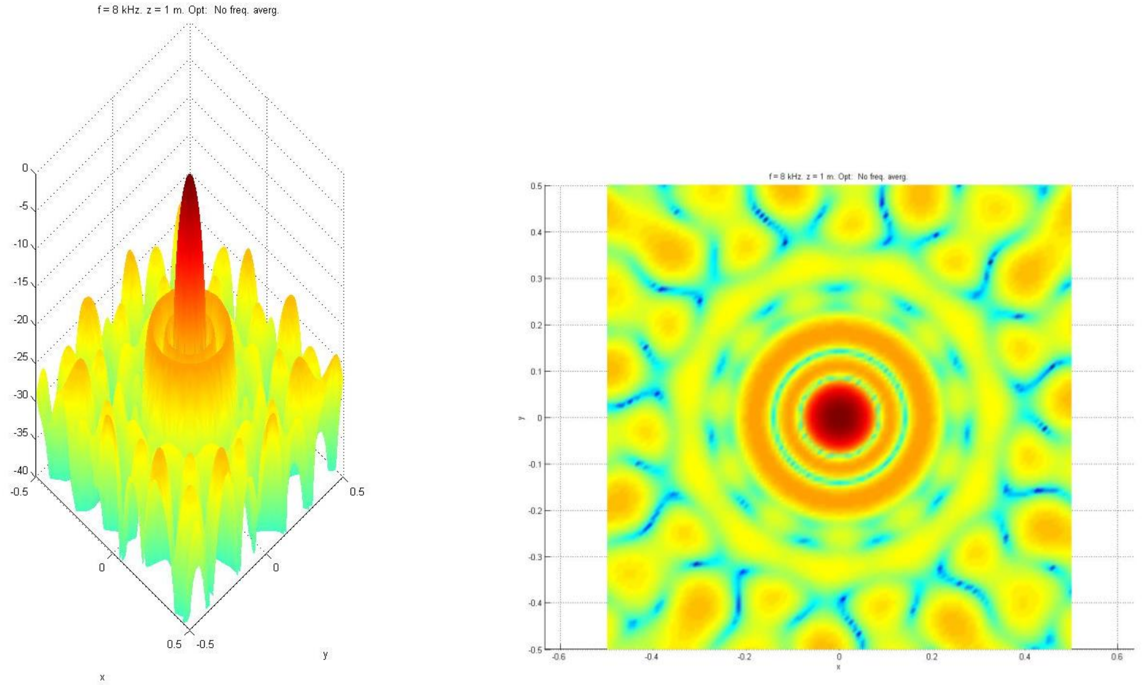


Figure 58 – Array point spread function for 264 microphones at 8 kHz frequency, views showing peaks relative levels and shape of array response.

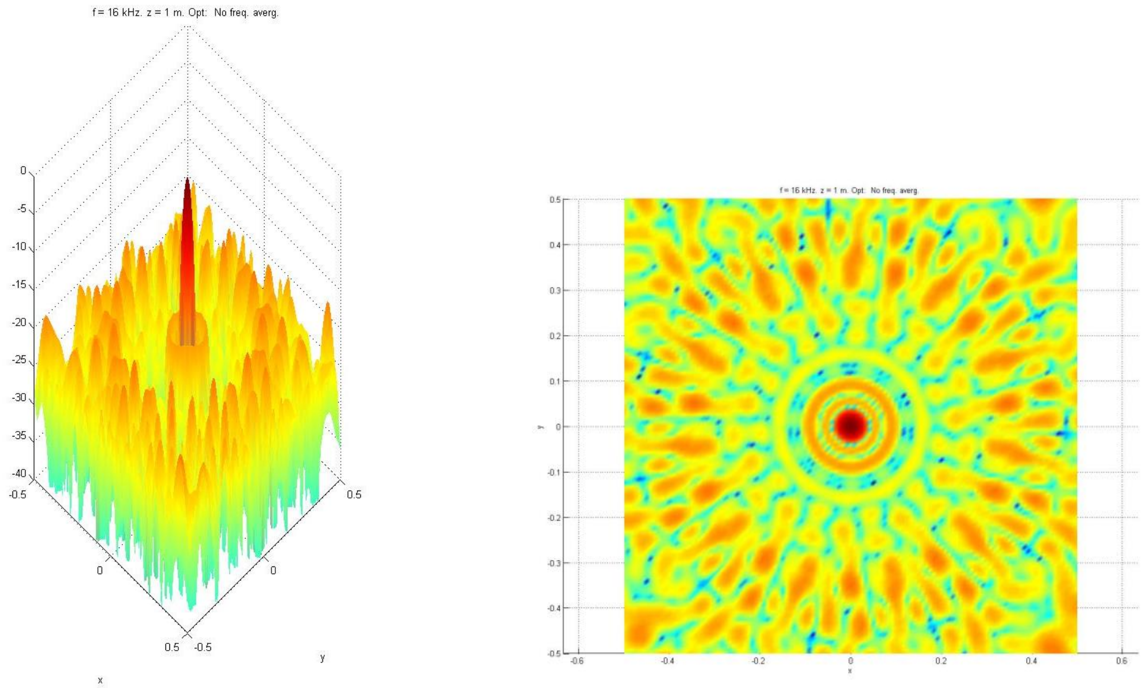


Figure 59 – Array point spread function for 264 microphones at 16 kHz frequency, views showing peaks relative levels and shape of array response.

A final set of 120 microphones (of which 112 will be used in the preliminary test) is selected and analysed via psf in the following pages. Although a number of microphones are lost, and the point spread function is not perfect, it is believed that a sufficient amount remains to perform the required data acquisition.

It is determined that the response from the following psf is sufficiently good to gather worthwhile acoustic data from the Filton wind tunnel test. A further analysis looking into even better array design could, and indeed should, be done as a continuation of this study.

The microphones which will be used are depicted below, although it is intended that all 264 holes will be drilled to enable up to 264 microphones to be installed, thereby extending the prototypes life allowing for future expansion and growth.

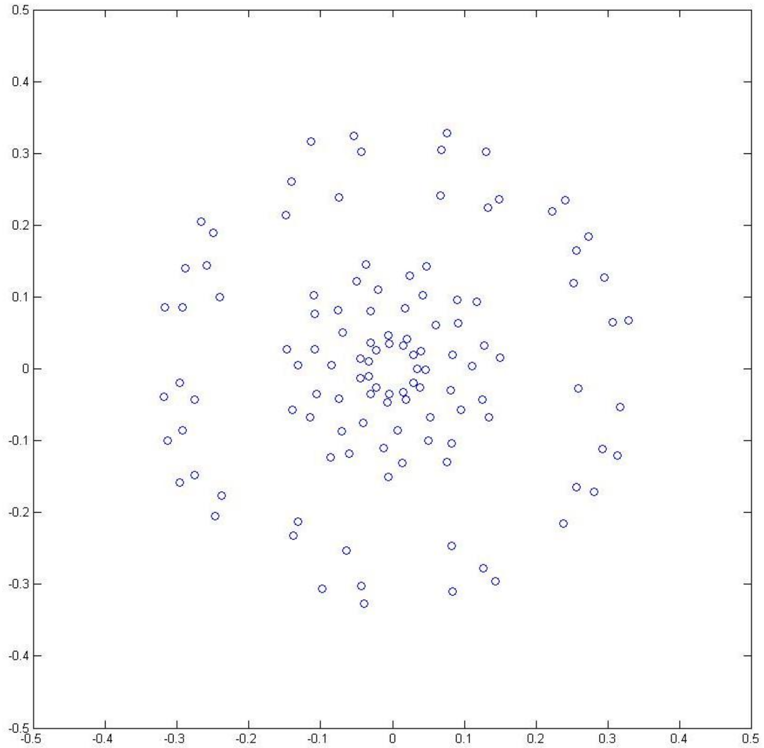


Figure 60 – Array design configuration for 119 microphones.

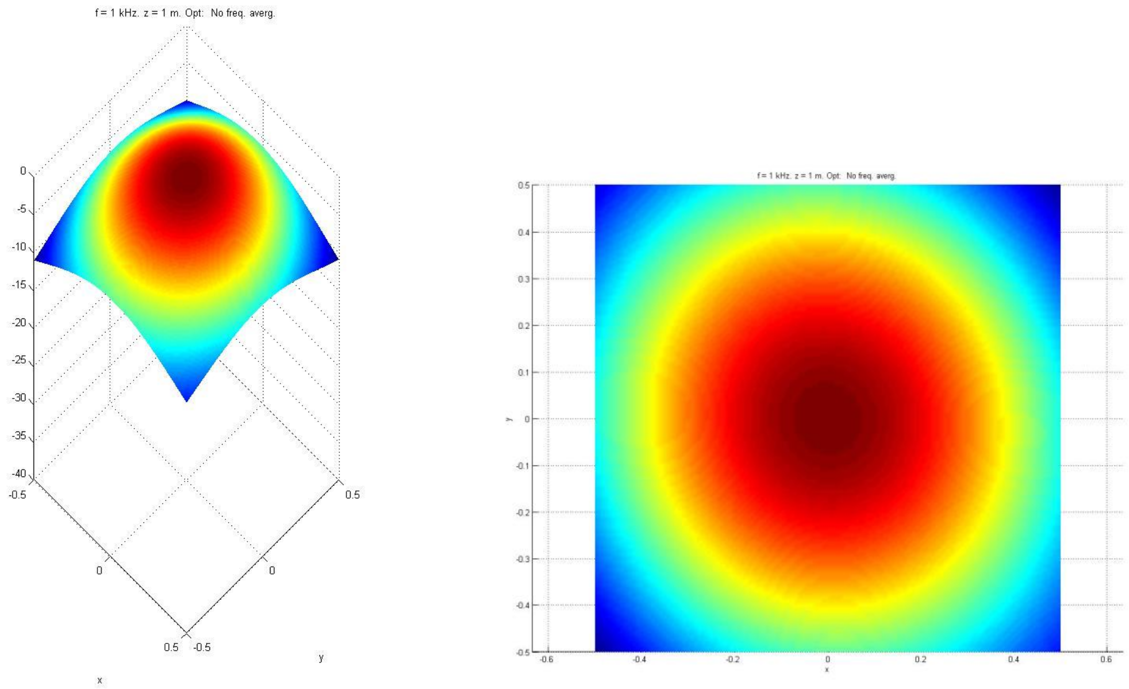


Figure 61 – Array point spread function for 119 microphones at 1 kHz frequency, views showing peaks relative levels and shape of array response.

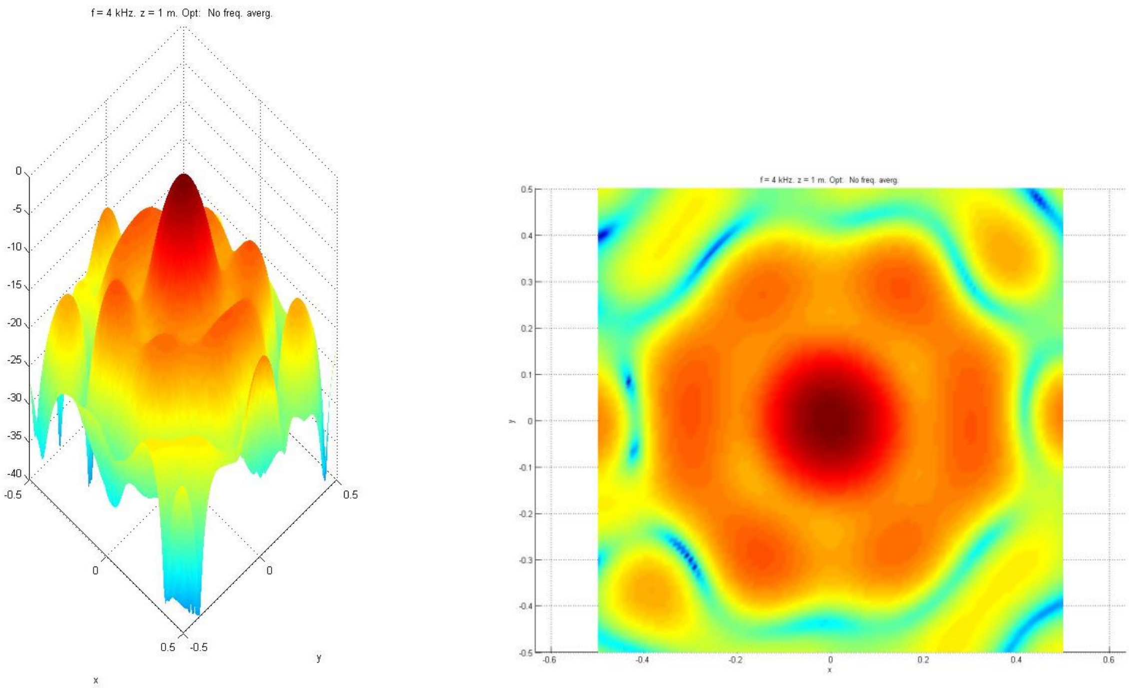


Figure 62 – Array point spread function for 119 microphones at 4 kHz frequency, views showing peaks relative levels and shape of array response.

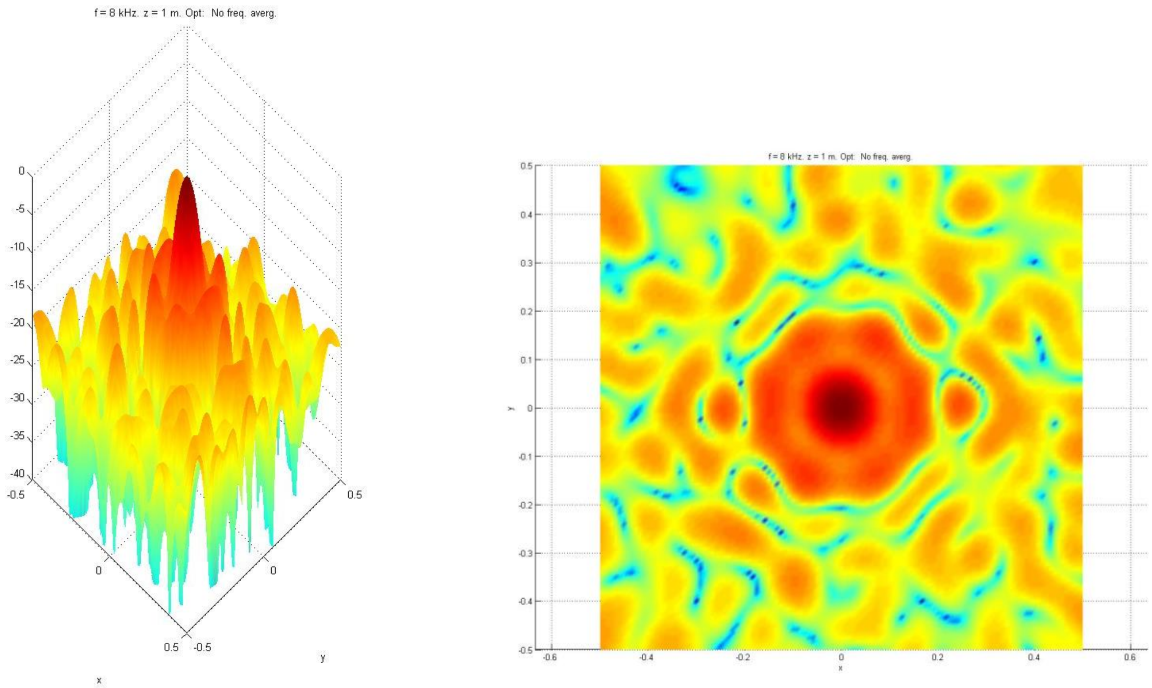


Figure 63 – Array point spread function for 119 microphones at 8 kHz frequency, views showing peaks relative levels and shape of array response.

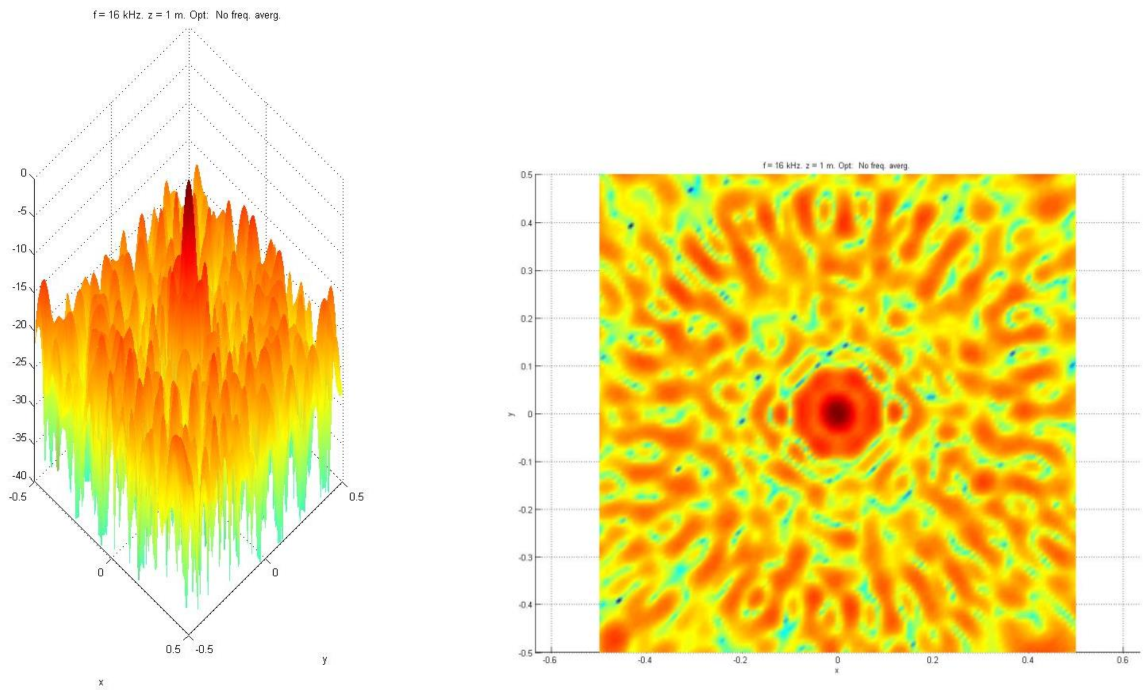


Figure 64 – Array point spread function for 119 microphones at 16 kHz frequency, views showing peaks relative levels and shape of array response.

The following images, figures 65 – 68, show the PSF of the 119 array compared to simulated perfect response. This selection of microphone placements was the array that was used in the November 2006 and July 2007 tests as described in section 4.

In a 'perfect' PSF response the sidelobes are low and spread far from the central peak and the central peak itself is relatively thin, indicating high resolution at the frequency indicated. As mentioned before at higher frequencies it is expected that the mainlobe should become thinner (beamwidth decreasing) with more sidelobes becoming present.

From the results it can be seen that the array configuration that was used is unbalanced as the psf shows a non-circular result and sidelobes which are relatively close to the main peak.

There were a number of microphone arrangements attempted to rebalance the array and restore the point spread function response to that of a 'perfect' response. The one used as seen previously was the closest achievable given the physical constraints of the wind tunnel.

The images show that the mainlobe of the response, at low frequency, is wide. This is likely to be due to the small size of the array diameter, as a consequence the response is expected to be low resolution. This is the first indication that the low frequency responses were going to be of a very low resolution, given the form and small size of the array, this was somewhat unavoidable and expected.

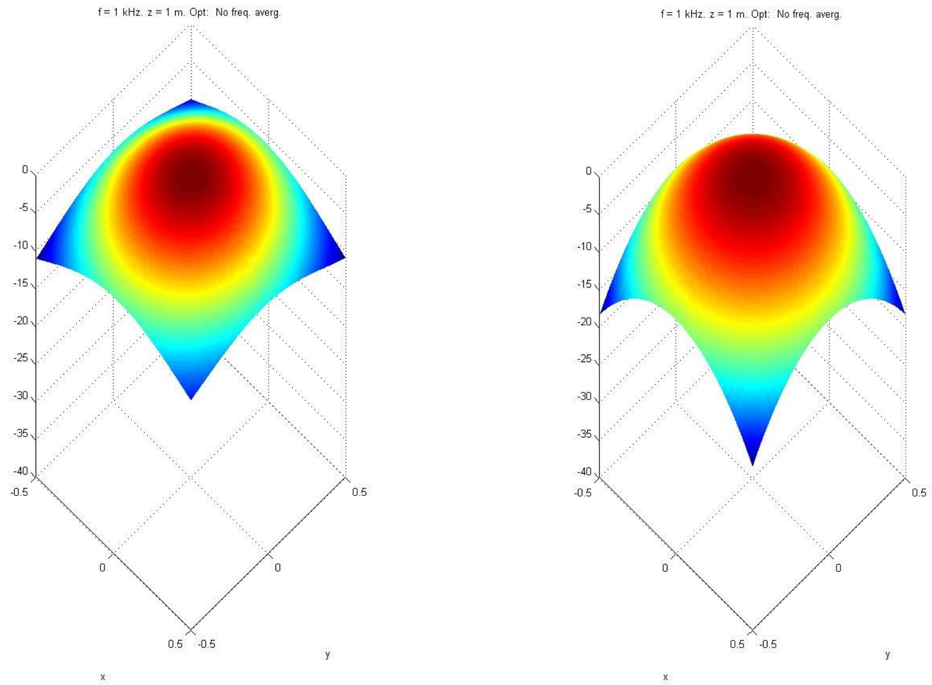


Figure 65 - Figure comparing PSF from the microphone array used in the test (left) to a 'perfect' PSF response (right) at a 1 KHz frequency.

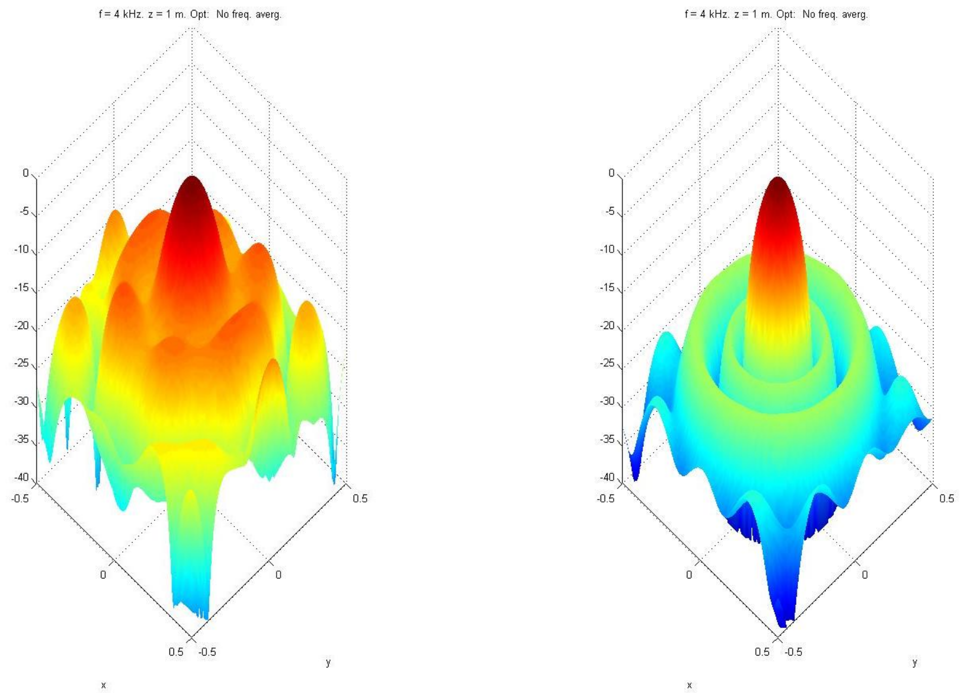


Figure 66 - Figure comparing PSF from the microphone array used in the test (left) to a 'perfect' PSF response (right) at a 4 KHz frequency.

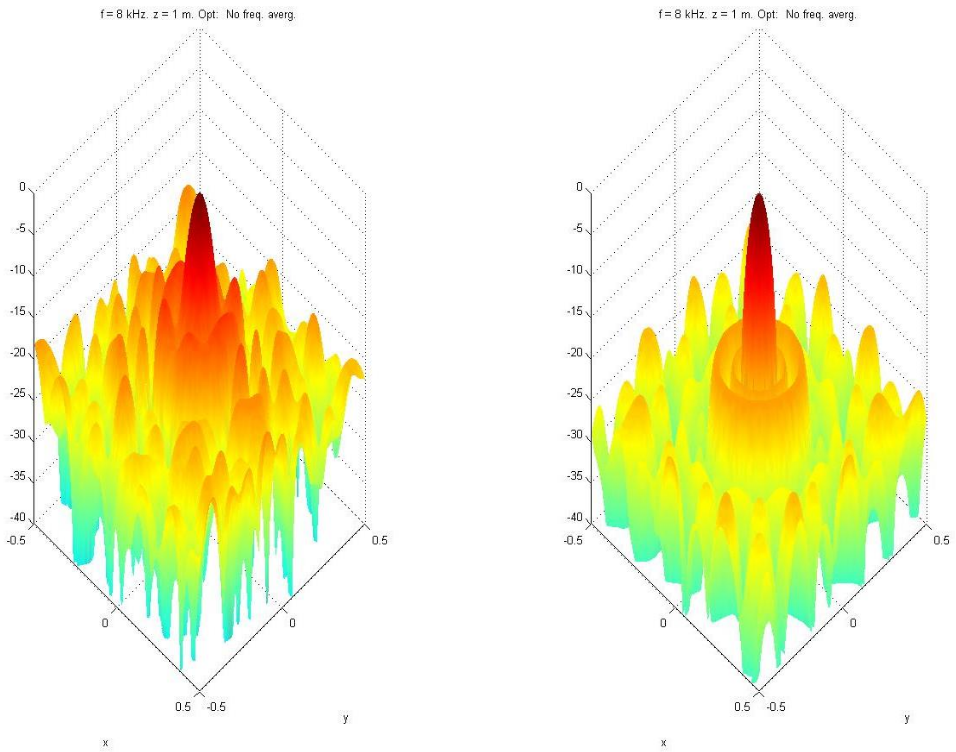


Figure 67 - Figure comparing PSF from the microphone array used in the test to a ‘perfect’ PSF response at a 8 KHz frequency.

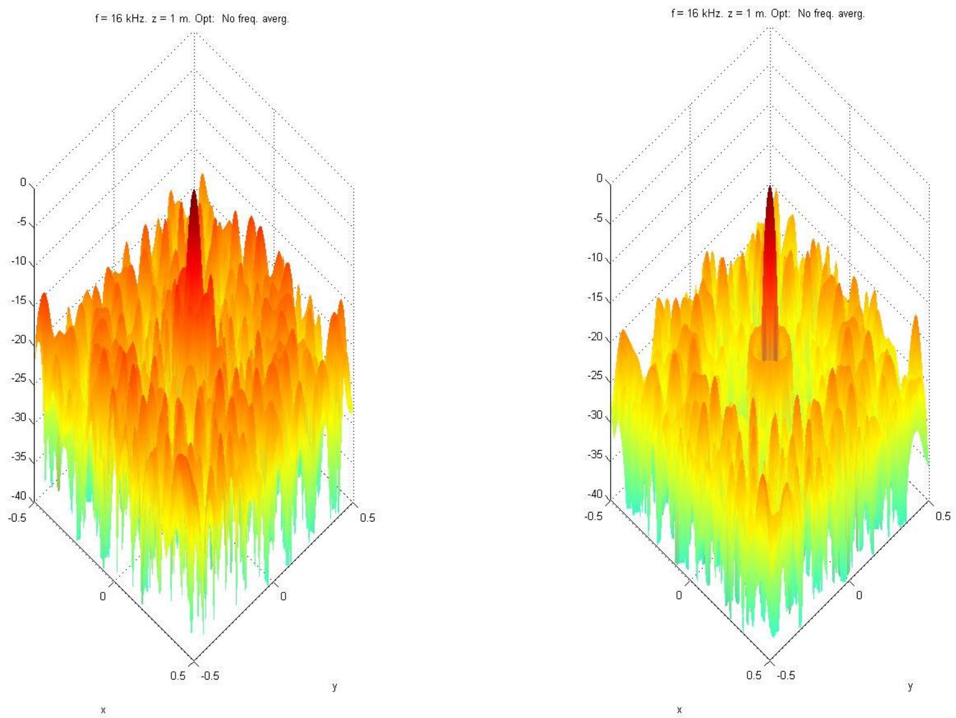


Figure 68 - Figure comparing PSF from the microphone array used in the test to a ‘perfect’ PSF response at a 20 KHz frequency.

3.4.2. Point Source Design

A useful technique in calibrating the microphone array is to employ a point source to generate a signal that can be easily determined and one that can be easily located by the microphone array. The design created in this study is based on a NACA symmetrical aerofoil (NACA – 0024), to provide an aerodynamic (and thus hopefully quiet) casing for a number of sources, with at least one in each direction. Sources would be located pointing in all four directions; all four directions are selected so that for future tests where arrays are mounted on ceilings or walls the same point source can be used. The design would also allow the point source to be used in any number of wind tunnels, which would prove useful if a mobile, modular array is constructed as the same point source could be used to calibrate the array.

The following diagrams show the design concept for the point source, with four holes set into a conical aerofoil shaped tube. An additional tube is attached to one side to allow cables and power to be fed through. The design is intended to produce a clear noise source with a minimum noise and interference from the placement of objects into the flow of the wind tunnel.

As mentioned before, current work includes the design and installation of a point source, the current design, based on the symmetrical aerofoil (NACA – 0024), is further modified and based off the McMasters-Henderson symmetrical airfoil co-ordinates, which were proven to be quieter in wind tunnel flow [25].

The following diagram show the design concept for the point source, a conical aerofoil shaped tube with holes cut for the tweeters. An additional aerofoil shaped tube is attached to one side to allow cables and power to be fed through to the tweeters from the power-amplifier and signal generator, both of which would be located outside the wind tunnel.

Not shown is the connection to the wind tunnel surface itself, this ‘foot’ would be different for each wind tunnel and dependant on which surface (ceiling, floor or wall) the point source was connected to at that specific time.

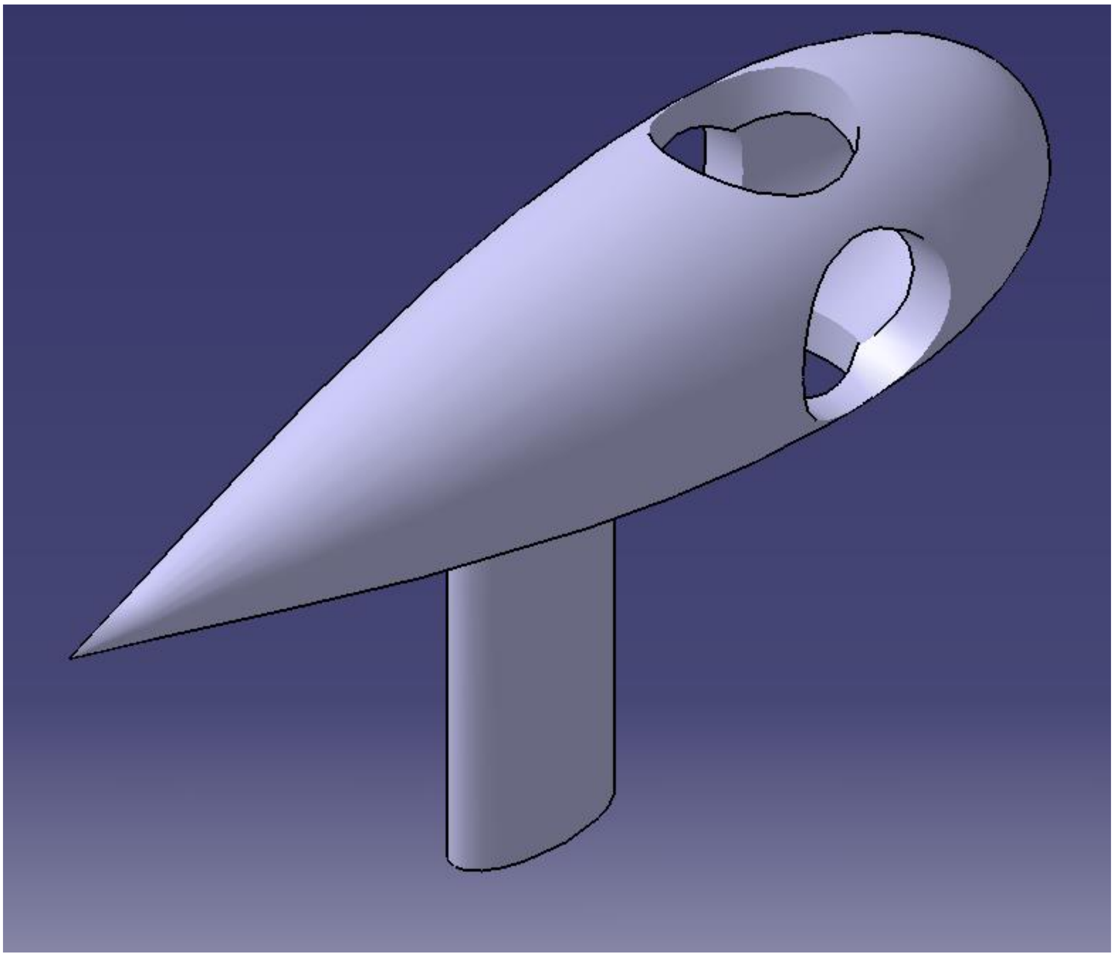


Figure 69 – Point source design concept.

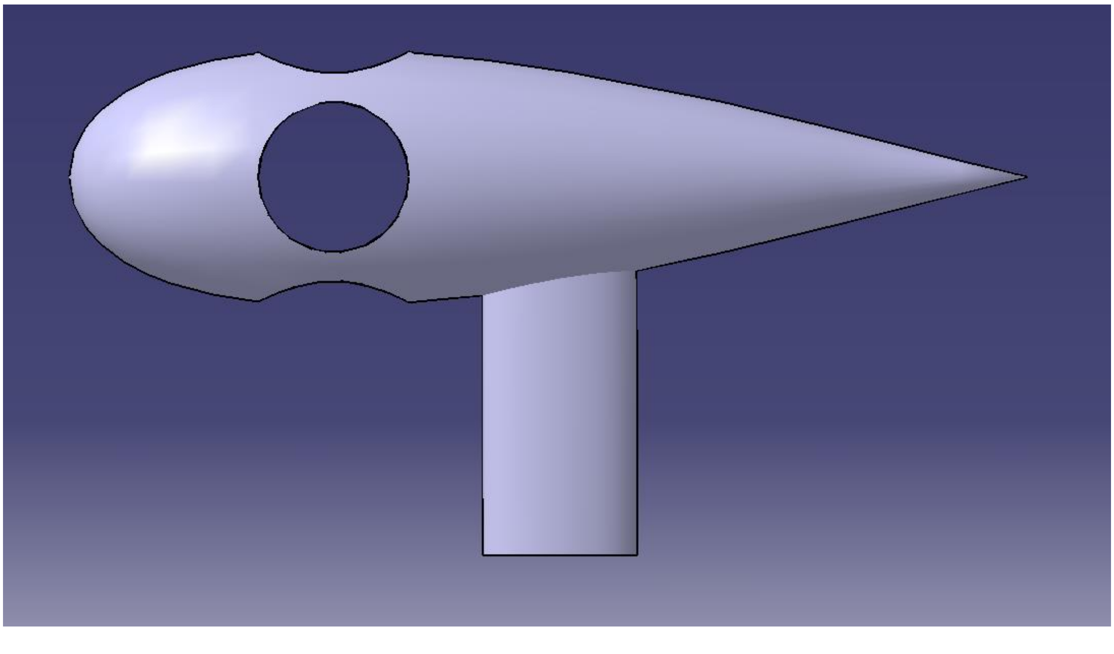


Figure 70 – Point source design concept – side view.

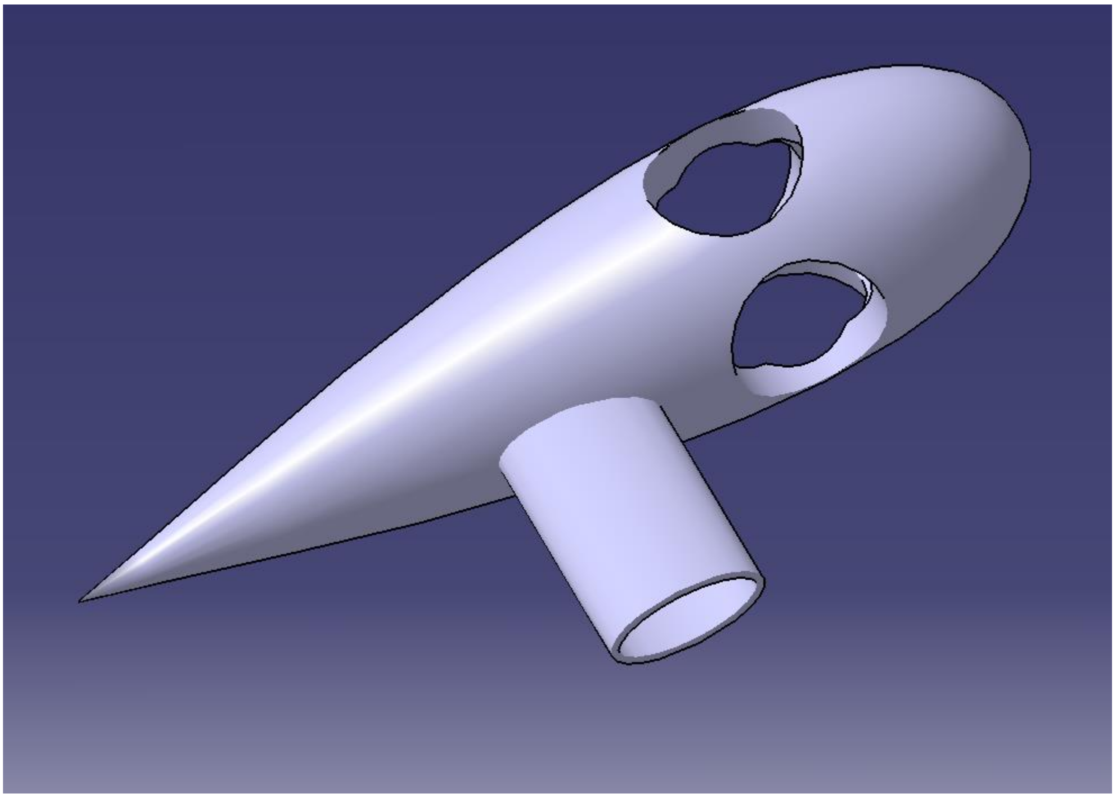


Figure 71 – Point source design concept – view showing access for cabling.

The final point source design (depicted on the next page), modified and completed at the end of September 2006 is introduced with an additional 4 holes for further point sources. These extra holes would allow two sources to be 'pointed' at the array and can thus be used to determine the resolution in the array as two distinct sources should be detectable. Additionally this opens up the possibility of driving two sources in the same plane at different levels or frequencies (or both) allowing further calibration of the array to be done.

Further enhancements to the design include slots to allow the point source to be slotted into any number of wind tunnels with relative ease and a modular style reinforced strut to allow the point source to be located further away from any mounting surfaces.

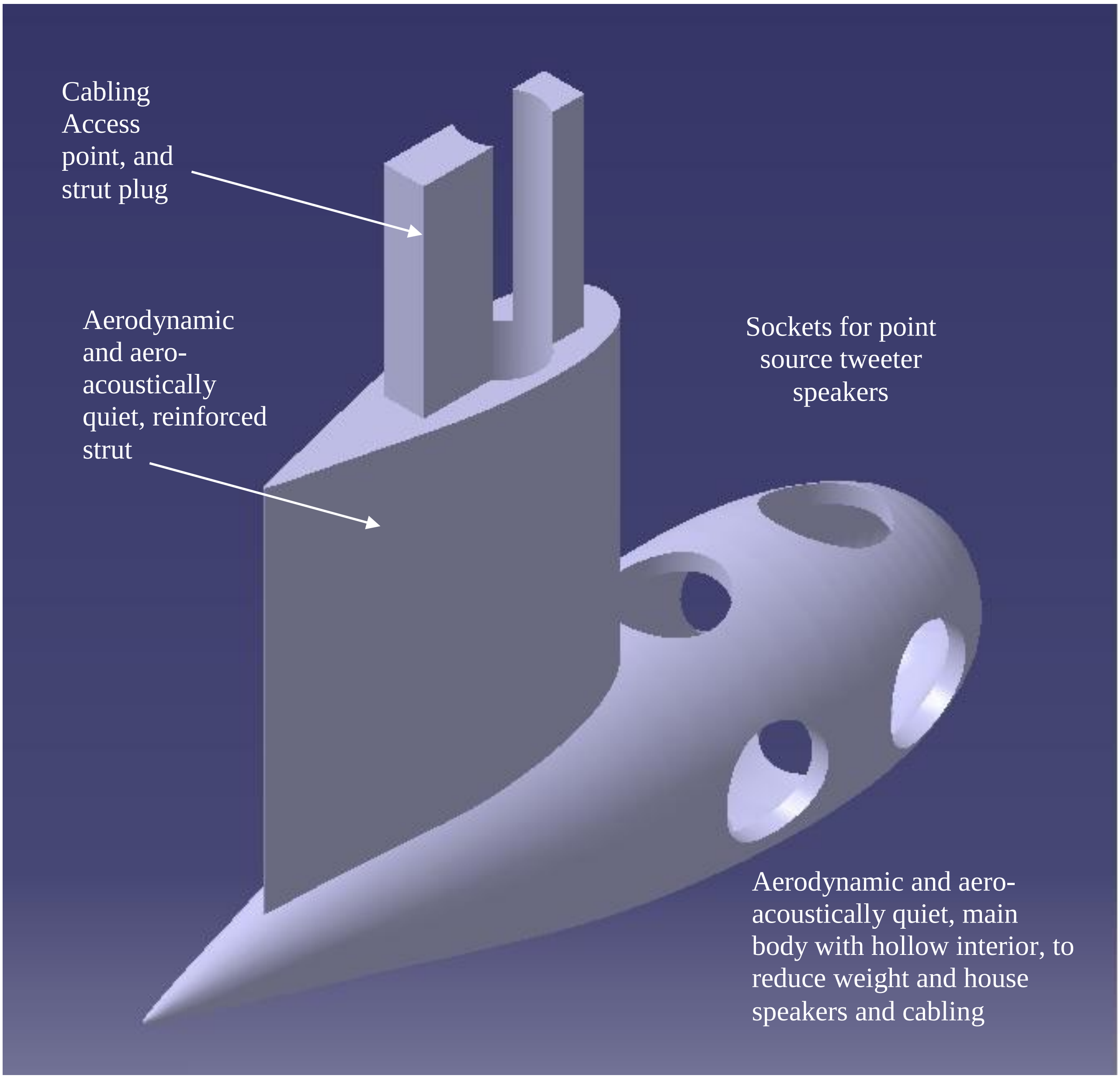


Figure 72 - Point Source design intended to produce a clear noise source with minimum noise and interference whilst in the flow of a wind tunnel.

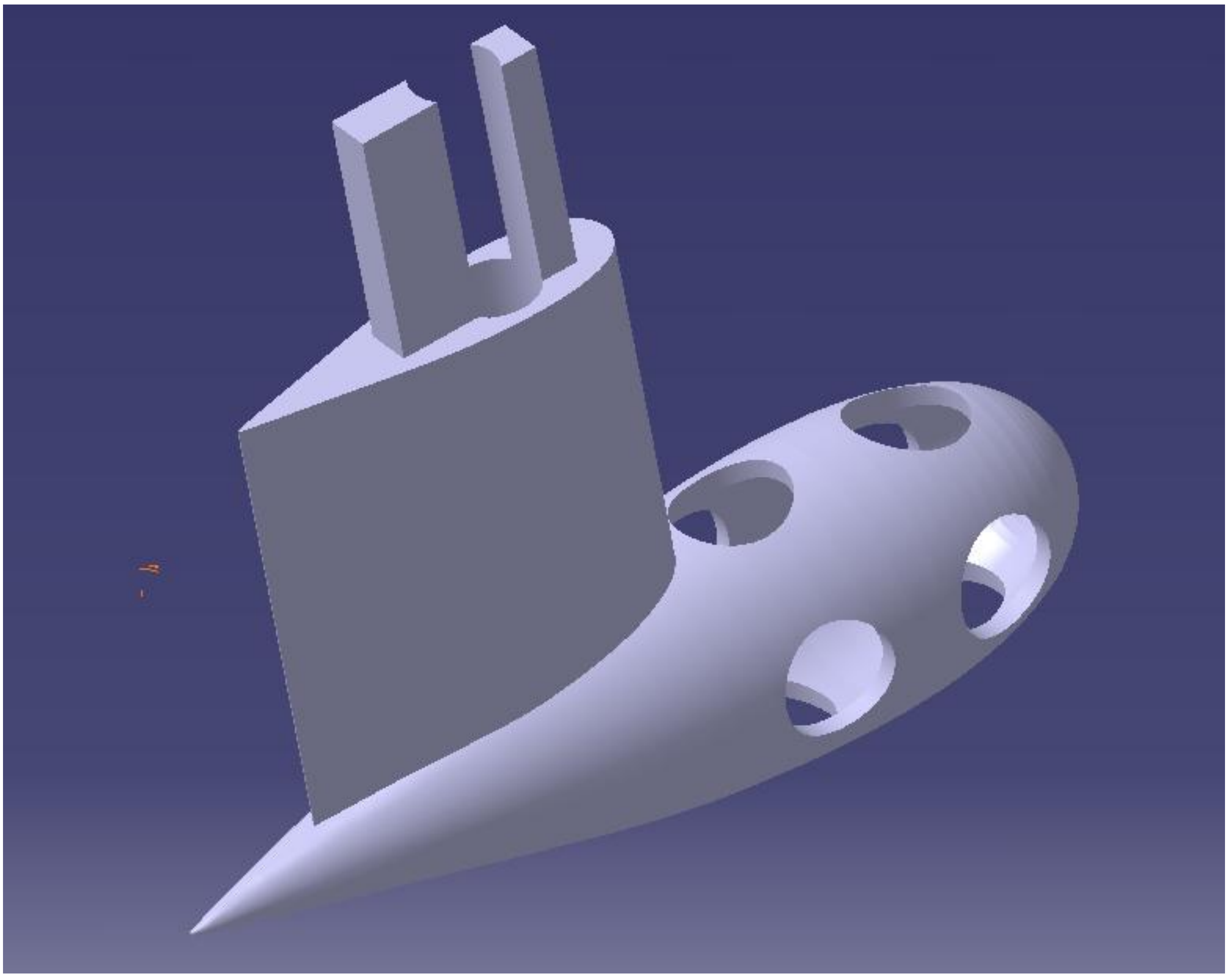


Figure 73 – Modified Point source design concept – view showing access for cabling.

4. Year 2 - Methodology and Testing

This section isolates the work completed in the second year and houses the reports given for each of the experiments and tests conducted during the period October 2006 to October 2007. During this time, three significant projects were undertaken each with a specific aim to advance the aero-acoustic field of experience and knowledge.

In November 2006, the prototype acoustic array was installed and used for the first time in the Airbus UK, Filton wind tunnel complex. A typical wind tunnel model was tested and acoustic plots were produced for a number of model and wind tunnel configurations and conditions.

In July 2007 the same array was used for a second time, with some modifications, to once again produce as set of acoustic plots. Several enhancements were also achieved at this time and are described in the following sections.

4.1. Aeroacoustic Testing

As planned, two complete, successful industrial trials of the aero-acoustic array and the University of Southampton's Beamforming code: 'SotonArray' [61], have been completed in the Filton Wind Tunnel after an acoustics feasibility test was conducted (see Section 3.3, [62]) and the wind tunnel was deemed suitable for such tests.

These aero-acoustics tests were performed on wind tunnel models, in both cases full aircraft models. Emphasis was made on the aero-acoustics of the wing and associated structure and as such, the tests were tailored for this.

As a continuing part of this study further acoustic and aeroacoustic test will need to be performed before a finalised product can be fully established and even then it can be expected that there will be constant upgrades. Given the nature of the way that industry works it is necessary to plan such events well in advance in order to be prepared for all future testing.

4.1.1. November 2006 and July 2007

In November 2006 the preliminary prototype array was tested in the Airbus Filton wind tunnel. A further test was completed in July 2007 using the same array, equipment and techniques, but with a software upgrade, introducing new algorithms (namely the CLEAN-SC coding, [56]) to improve the processing and plotting of the results.

As seen in the section 3.4 the finite width of the mainlobe and additional sidelobes can distort the output. Whilst ideally a larger array with more sensors would reduce this effect, realistically there will always be limitations to the physical size of the array and the number of sensors that can be employed. To this end deconvolution techniques can be employed to reduce the negative effects. Poor resolution at low frequencies and noise at high frequency are catered for using deconvolution techniques: DAMAS [73], CLEAN and CLEAN SC can be utilised in post-processing, the latter being implemented in this study after the November 2006 tests but applied retrospectively to the data acquired.

CLEAN-SC was developed by Pieter Sijtsma [56] as a variant of the DAMAS and a modified version of CLEAN [74] and implemented in the Matlab beamforming code by Benjamin A. Fenech [24, 70].

DAMAS (Deconvolution Approach for the Mapping of Acoustic Sources) [73] works on the principle assumption that there exists a number of independent noise sources equal to the number of grid points. This allows a linear system of equations to be solved which take into account the influence of all the sources at the different locations.

CLEAN [74] is an iterative process, starting with a standard beamforming plot which contains the sources of interest as well as spatial noise. A peak location is identified on the plot and a scaled PSF corresponding to the peak is subtracted and replaced by a clean beam (a beam without sidelobes). The next peak location is then identified and the process repeated.

Both CLEAN and DAMAS utilise theoretical ideal Point Spread Functions, which assumes that all the sources present are incoherent monopoles, which in aeroacoustics is often not the case. CLEAN-SC builds on CLEAN by not using ideal PDFs. CLEAN-SC removes the contribution of sources that are spatially coherent with a major source based on the fact that sources are spatially coherent with their corresponding sidelobes.

Like CLEAN, CLEAN-SC is an iterative approach extracting discrete sources from standard beamforming plots containing sources of interest as well as spatial noise. The iterative process starts by removing the peak source in the plot and the spatially correlated sidelobes from the cross-spectral matrix. The PSF is subtracted and the source is added to the new clean plot. The process is repeated until the energy in the cross-spectral matrix is similar to the energy of the last source removed. The result is a plot which shows the actual noise sources and not their sidelobes.

The following section describes the aeroacoustic array testing that was done in the Airbus Filton Wind Tunnel complex in November 2006 and July 2007. In both cases the testing was completed in three days; (between the 22nd - 24th November 2006 and the 10th – 12th July 2007). Both tests were utilised in investigating the aero acoustics aspect of an aircraft model. The floor mounted acoustic array of microphones, as described in section 3.4 was used for these acoustic measurements.

Figure 74 below shows the microphones holes layout as was used, the holes structure was such that the microphone would sit flush with the top surface and be supported and held by the plate itself.

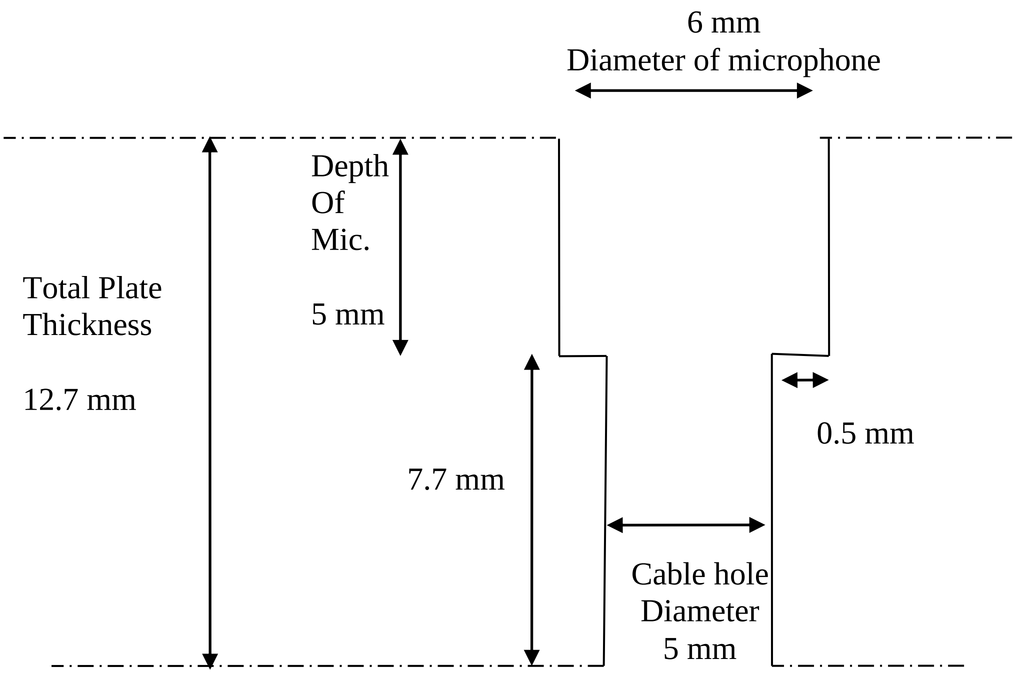


Figure 74 – Dimensions of microphone holes in the array.

4.1.2. Methodology

4.1.2.1. Set up Methodology

Installation of the array took a little under four days to complete; this was mostly due to the handling of large number of cables, the sheer number of individual connections and repairs of damage incurred during installation. Given the prototype nature of the installation, delays were to be expected and sufficient time was scheduled to allow installation and testing.

The diagram on the following page shows all the connectors and components utilised in the array assembly and use. There are three main sections involved in the installation

- The Array Itself - including the Microphones and Cloth Covering (to separate microphones from WT boundary layer) and the supporting structure
- Pre-Amplifiers – These are used to power the microphones
- Data acquisition and User interface

The array component consisted of a single plate dropped into the floor of the wind tunnel test section. A requirement for the placement of the array is the accuracy of it location. The location of the array with respect to the wind tunnel and model is required to millimetre accuracy to ensure accurate results in post-processing.

After ruling out building the array structure out of metal, as the microphones would then have to be isolated individually; the microphone array was drilled and cut out as one sheet of polycarbonate material, there were some initial concerns over reflectivity of the material at this point, however, using polycarbonate allowed the plate to be constructed as one piece and very accurately. Whilst the plate might move as whole, all the microphone positions will be fixed relative to each other with a tolerance of 0.01 mm.

Runners under the rail locations were installed to support the plate, so it was still not possible to use those areas for additional microphones.

Total Plate Thickness was 12.7 mm, giving a 5.7 mm clearance with the wind tunnel structure below, sufficient for a cable thickness of 3 mm to be passed through to the acquisition hardware located below the wind tunnel and by the control room.

The Pre-Amplifiers are used to power the microphones, two types were utilised in the test program as indicated in figure 75. The newer pre-amplifiers utilised a block connector, which proved very useful in installation as it allowed large numbers of connections to be made quickly and efficiently. The older pre-amplifiers require each and every individual microphone to be connected individually. This proved laborious and time consuming, but does have the advantage of allowing individual channels to be repaired as and when necessary. After installation, each and every microphone was tested using the embedded testing facility in the LabView Software (Measurement and Automation – MAX).

For the July test, the same set-up was used with minor modifications: Only the newer type pre-amplifiers were used, a purpose built pre-amplifier was utilised and whilst older types of pre-amplifiers require each and every individual microphone to be connected individually, (considered laborious and time consuming), the advantage of allowing individual channels to be repaired as and when necessary was lost. This did cause the second test to proceed with a number of broken channels.

Figure 76 shows the set-up used in the July test, with the change in cabling and hardware shown. Most of the hardware was located in the alcove between the wind tunnel control room and the test section, with a standard PC laptop used to remotely control the system. Raw data was stored in external hard drives and backed up onto a Networked hard-drive, from which the raw data was processed.

The acquisition unit, an NI-PXI 1042 Q chassis held fourteen PXI- 4472 cards, and a data transfer card (for connecting to the controller). The controllers used were an NI 8350 and in the July test an NI 8351. These controllers were used for interfacing with the acquisition unit using software written in the Labview 8 software and for processing the results using code written in Matlab (version R2006b) as described in section 3.2.1.

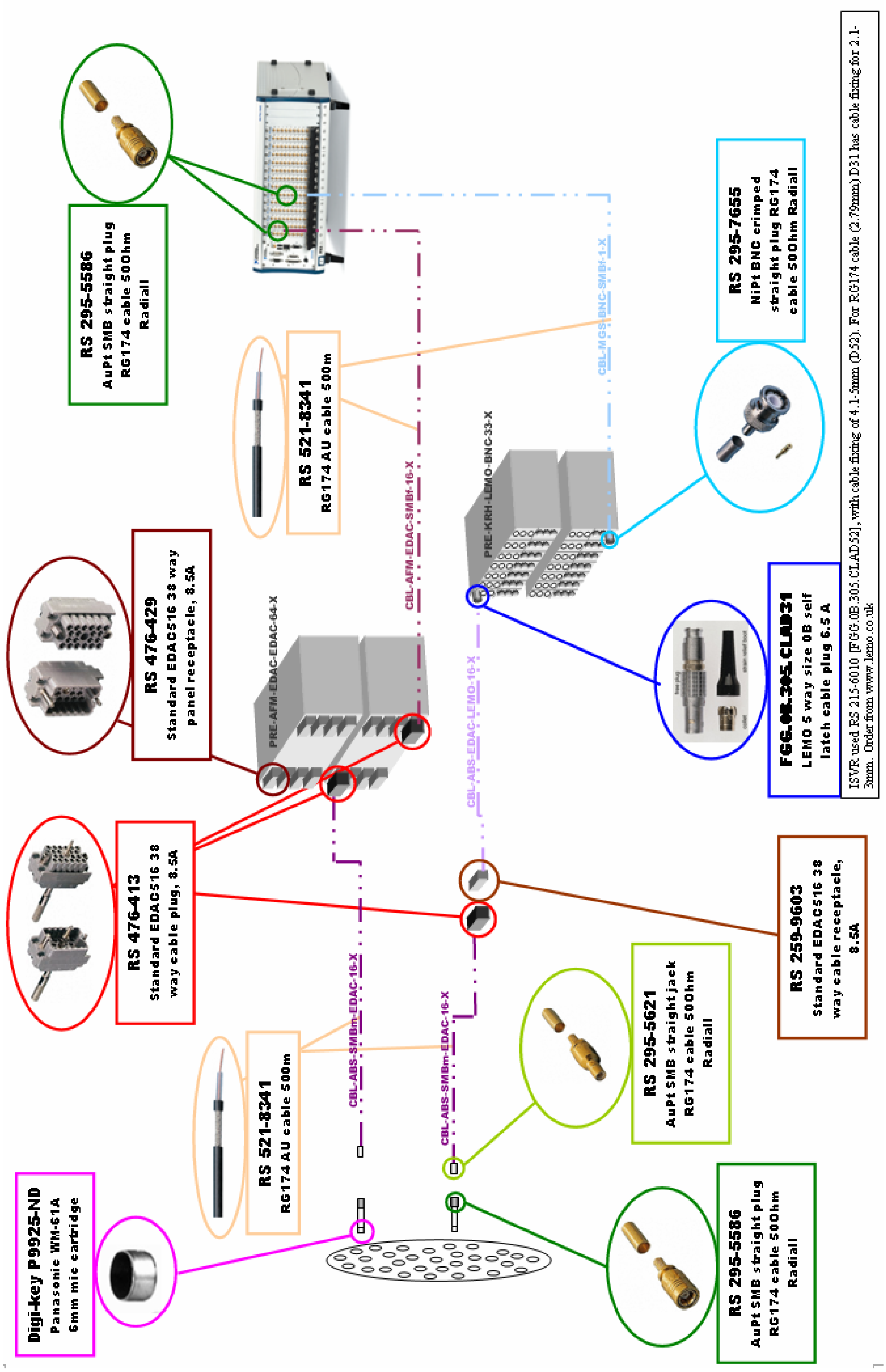


Figure 75 - Diagram showing the cabling and connectors used for the array and acquisition hardware. (Diagram courtesy of Benjamin Fenech)

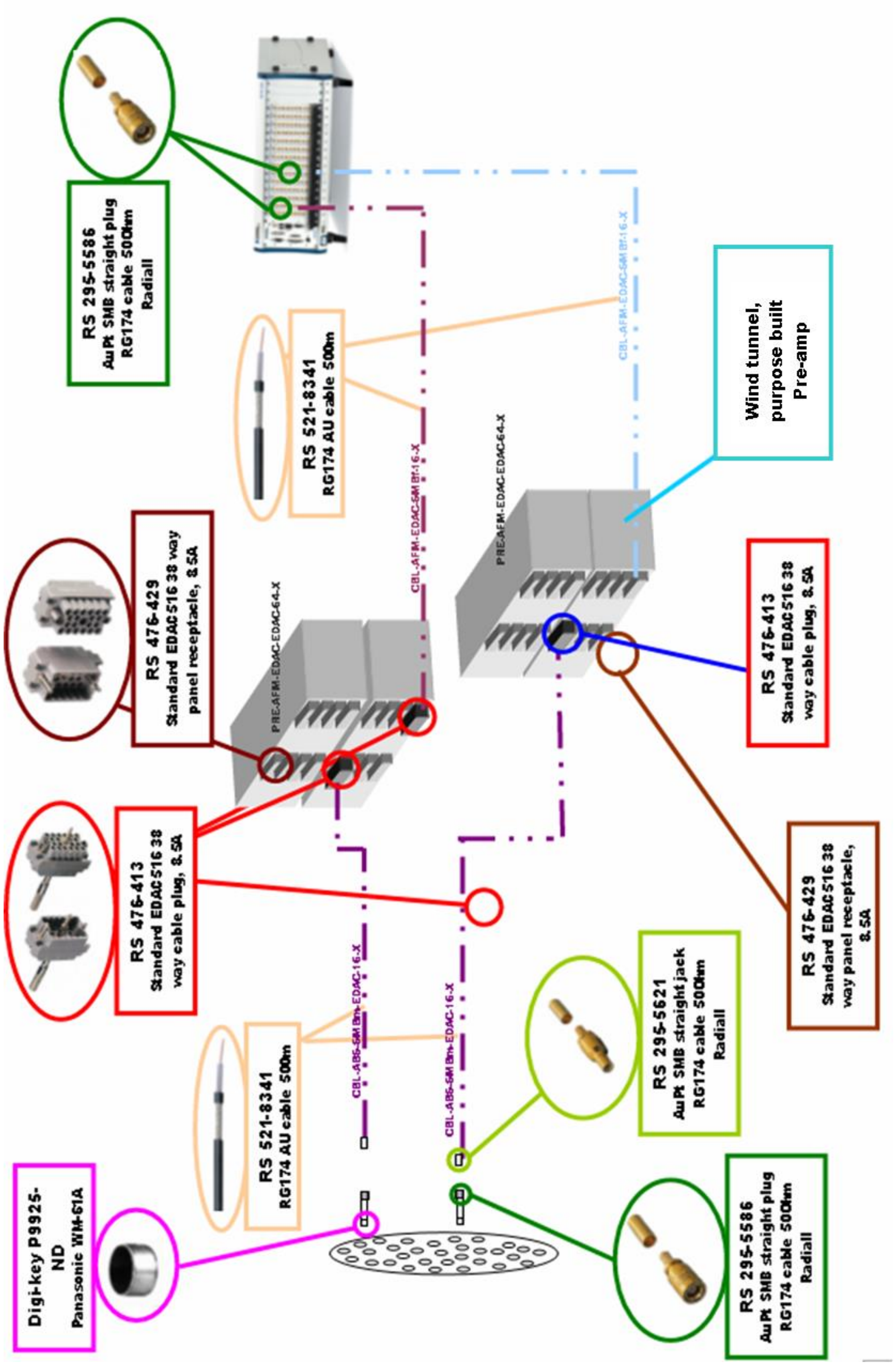


Figure 76 - Diagram showing the cabling, connectors and other acquisition hardware employed for the July 2007 Filton wind tunnel test.

4.1.2.2. Calibration Process

For any device utilising microphones where accurate levels are required a certain amount of calibration is required. Since low cost microphones were used for this test calibration is a necessity. Additionally the use of old and new pre-amplifiers to power the microphones meant that inherent errors were to be expected as the devices employed were sufficiently different. Two types of calibration were implemented:

- Pistonphone Calibration
- Reference Microphone Calibration

Pistonphone calibration involves utilising a pistonphone to generate a known tone at a known level at a known frequency. This test is very quick and easy to perform and has the added benefit of allowing each channel to be tested. During the installation of the array the pistonphone test was applied to every microphone. Allowing broken or damaged microphones to be located, eliminated and replaced before the testing.

In this case the pistonphone produces a tone set at 1kHz at 94 dB (1 Pa). The response is recorded in an excel file which is then kept for later use in the processing code.
The second type of calibration is known as Reference microphone calibration and involves comparing each and every microphone to a pre-calibrated Bruel & Kjaer (B&K) type 2615, ½" reference microphone (for the November 2006 test). Each channel is subjected to white noise at the same time as the pre-calibrated reference microphone is. Direct comparisons are made and the Coherence of the two microphone signals is used to confirm the quality of the microphones operation. This calibration technique eliminated the errors incurred from using two different pre-amps. From this form of calibration, phase and magnitude data is collected for each channel and used in the post-processing.

For the July 2007 test, a GRAS microphone was utilized as the reference microphone as the B&K microphone was limited to frequencies up to 24 kHz. The GRAS microphone allowed calibration up to 48 kHz, the upper limitation of the acquisition cards sampling rate after anti-aliasing. The reference calibration of the July test compares the Magnitude, Phase and Coherence between the GRAS reference microphone and one of the electret test microphones.

4.1.2.3. Testing Procedure

The testing procedure for data acquisition is relatively straight forward. For the test at the Filton site acquisition takes approximately 10 – 20 seconds with 112 channels, this is dependent on the system settings at the time of acquisition and meant that the model had to be stopped for a period for each acquisition. There are four settings to determine for acquisition: sample rate, block size, number of blocks and the buffer size. The first three affect the amount of data being acquired and to a degree the quality. Due to limitations in the acquisition software the sample rate was set to 48000 Hz. Due to post-processing computing limitations the block size and number of blocks were set to 8192 and 60 respectively.

Variables in Acquisition	Setting Used for the Nov 2006 Test
Sample Rate	48000 Hz
Block Size	8192
Number of Blocks	60

Figure 77 –Table showing variable values for acquisition in November 2006 test.

The front panel as seen in the following figures also allows any number of channels to be selected and the data file name to be specified. For the test the full 112 channels were selected and the data file name was set to match each wind tunnel run number. Once the wind tunnel is up to speed and the model fixed in the required configuration, acquisition is initiated. Checking the size of the acquired data confirms the acquisition (in the case of the November 2006 test, this was 430,080 KB per acquisition file run).

For the July 2007 test progress and modifications in the acquisition coding allowed more data to be acquired, allowing an increase in the number of blocks to be obtained. This did mean that additional storage space for the raw data was required, but more importantly this meant that the sampling frequency could be set to 96 kHz, allowing post-processing results up to 48 kHz (anti-aliased) .It was only the manufactured limitation in the acquisition cards that meant the sample rate was only set to 96000 Hz. The previous limitation of 48000 Hz was resolved through rewriting the controlling Labview code. The new sample rate limit was sufficient for the maximum 32 kHz requirement as stated for the test. New acquisition and post-processing computing limitations set the block size and number of blocks to 8192 and 200 respectively.

Variables in Acquisition	Setting Used for the July 2007 Test
Sample Rate	96000 Hz
Block Size	8192
Number of Blocks	200

Figure 78 - Table showing variable values for acquisition in July 2007 test.

With these settings the acquired data, stored in binary format using double precision, meant that a typical 17 second acquisition resulted in a file size of approximately 1.4 gigabytes. Checking the size of the acquired data confirms the acquisition (in the case of the July 2007 test this was 1.4 GB per acquisition file run).

4.1.3. Results & Discussion

Due to the nature of the aerospace industry, some of the details of the results are not publishable in the general public domain; as such generic plots from the two tests are used, with industrial data either obscured or not included. These plots reflect the variety of data that was acquired.

4.1.3.1. November 2006 Results

In total, over three days of testing and seven configurations 65 runs were made. In addition there were 70 test runs performed with point source. These test runs were used on the day to calibrate the plots to the distances involved and to confirm the processing and acquisition were operating as expected. They were also used to calibrate the plots in terms of area of view and the positioning of the model, at different angles of attack, within the processed sound source maps.

For each run sound map plots were produce for a number of frequencies (frequencies 3.1, 4, 5, 6, 8, 10, 13, 16 and 20 KHz were selected to give a wide range). In total 567 sound map plots of the runs were produced and a further 18 background plots also processed. Only a select few of those images are included in this report. The run plots were all overlaid on a photo of the model used in the wind tunnel allowing direct source location to be performed visually. For these plots only the area directly around the model was plotted, this saved time in processing. Additionally the plane selected for the processing to be set at, was the centreline of the model.

4.1.3.2. July 2007 Results

Over three days of testing and twenty configurations 136 acquisition runs were made in the second test. In addition there were test runs performed with point source as before in the November 2006 test.

In total 11 channel failures were experienced during the July test. This is accounted for in the processing code and thanks to the robustness of the code the loss proved negligible on the overall result.

For each run sound map plots were produce for a number of frequencies. Frequencies at 1/3 Octave bands (2, 2.5, 3.1, 4, 5, 6.3, 8, 10, 13, 16, 20, 25 and 32 KHz) were selected to give a wide range and to cover the frequencies of interest for the model being tested.

The run plots were all processed into Tecplot .dat format and Airframe geometry was produced for all the angles of attack to allow the aircraft geometry to be overlaid over the plots allowing direct source location to be performed visually. For these plots only the area directly around the half side of the aircraft was plotted (the side of the aircraft model of interest), this saved time in processing. Additionally the plane selected for the processing to be set at, was the centreline of the wing; which was altered, as required, with the changes in angle of attack.

The following section lists the limitation and possible improvements that exist for the current acoustic array.

4.1.4. Limitations

A list with regards to the limitations of the equipment and the processing (and thus the potential errors) is included in the following section, as well as where care should be taken in interpreting the results. Beginning with array itself, the main limitations is evident from the array design being interrupted by the bars in the wind tunnel floor. Future array designs should hopefully be a complete array, and thus give results with less interference from sidelobes. The array size is also limited by the physical constraints in the wind tunnel. Larger arrays should, in theory, produce better resolution results in the lower frequency domain. For the given physical parameters of the array as stated in section 3.4, results between 4 kHz and 35 kHz can be expected to be generally good, with the normal results below 3 kHz being effectively useless in determining sound sources.

Another limitation in the design is the number of channels was limited to 112 due to the number of acquisition cards available in a single chassis. It is generally agreed that the more microphones in an array the better the results [references: 2 and 4]. A higher number of channels and microphones should produce higher resolution result plots even at the lower frequencies.

In the July 2007 test, there were a further 11 channel losses during testing. The results shown are for effectively 101 channels, although improvements in the processing code and the overall reliability of the technique meant that the results were only slightly affected, with the lower frequency plots up to 5 kHz (rather than 3 or 4 kHz) being of lower resolution and quality.

For the November 2006 test the results initially were poor, however, after the apparent lack of coherent results that were generated it was determined that the difference between the new and old pre-amps was the problem. This was resolved by performing later calibration with a second microphone as mentioned before, and excellent results were produced as shown in figures 80 - 106.

Another issue with the microphone array was the limitations on the frequency range available with the current level of technology. Despite the inherent problems the array functioned as well as could be expected although as mentioned before the low frequency results were less useful. In the November 2006 test frequencies above 20 kHz were not made available as the sample rate for acquisition was set to 48 kHz (meaning the highest frequency available for post-processing was 24 kHz). This setting was chosen as there are limitations in the software and hardware at that time. In this way the set up used and in particular the settings for the acquisition can be the limiting factor; the settings for processing are determined by the settings used for acquisition.

During the setup of the array a number of problems were encountered which merit future consideration. With only 112 channels it took three and a half days to install; this time included, however, the calibration to check each microphone was still functional. Full channel calibration takes a long time and would be easier to do before microphones are permanently installed. Once calibration for each channel is completed microphones should not need to be calibrated for each subsequent test. Assuming the microphones are fixed into the array and the same pre-amps and cables are used for each test then calibration would only be required at regular long intervals as the calibration used for a previous test would be the same.

In particular regard to the cabling; the only limitation is the time required for assembly and the convenience of assembly. The vast number of channels and connections meant that the likelihood of a loose connection or broken link was driven high. Increasing the number of connections and individual cables increases the chances of a channel failure or error. In addition, the cabling can prove awkward if the need to repair (or replace) a channel occurs.

Care has to be taken to isolate each channel from any interference or signal noise, some interference was identified from the balance equipment below the wind tunnel test section. As the cables and the microphone were close by the balance equipment was turned off during the test. This has implications if a simultaneous sound mapping and force measurement (using the balance) is required. For the July 2007 test a further layer of insulation, polystyrene foam, was installed between the bottom of the array plate and the balance motors below the test section to eliminate any possible noise interference. These steps, to shield the microphones from this interference, were implemented as were necessary as simultaneous sound mapping and force measurement (using said balance equipment) was required. This proved effective in eliminating the noise from the results, as shown by comparing the power spectral density of the microphones from the November 2006 and July 2007 test, and should be considered as a permanent inclusion in the array set-up.

Also implemented in the Matlab code post-processing code is a channel checking feature which identifies any channels which may have produced an erroneous or unusual signal, this is at post-processing stage and as such after the acquisition has taken place. Therefore it would be wise (as was done) to run a microphone check directly after set up, this was done during the first test runs and whilst the model was being installed as post-processing and channel checking and repair is time consuming.

Regarding limitations on the plots produced it is vital that any users are made aware that only the same frequencies can be compared for levels. That is to say that the scale on the plots is only equivalent for same frequency plots on different runs. It should also be made clear that none of the plots show absolute levels and indeed the levels are only relative to each other (in the same frequency plots).

Regarding possible errors the temperatures were not precise on some of the runs, this was due to lost data from the wind tunnel, although it is likely that the accuracy of the temperature has little effect on the sound source map plots and an error of a few degrees on the temperature would most likely have no visible effect on the results.

The nature of the acoustic testing is such that visual results take time, full processing of a run can still take up to an hour or longer depending on the frequencies of interest and the number of frequencies required. In order to facilitate testing and making the appropriate modifications, it is necessary to be able to get results as soon as possible. In order to maximise the time preliminary results, using a subset of the sampled data to obtain quick plots, were achieved during wind tunnel model configuration changes. The preliminary results needed to be of a sufficiently good quality to be useful. Whilst this can be dependent on the model a level needs to be determined with all customers to set a level that is acceptable in order to continue. The quality of results is somewhat determined by the number of averages used to process the results and as such a direct link between processing time and quality of results can be drawn. From a possible total of 400 averages (acquisition was 200 blocks at a block size of 8192 giving a possible 400 processed at 4096) it was determined that a value of 50 averages was sufficient for reasonably quick, yet accurate and useful results.

On the sound source map plots a good general dynamic range is around 12-14 dB as this is sufficiently large enough to allow distinct major sources to be shown. A 10 dB level is the minimum for a good dynamic range as anything smaller would prove identifying individual sources from background noise and reflected noise sources difficult. Low frequency results proved to be less useful, this was expected given the small (relatively speaking) size of the array, however with the enhancements found using the CLEAN SC algorithm [56] in post processing the July 2007 results, these low frequency plots are improved and some become useful, especially in determining source location.

4.1.5. Typical Acoustic Source Location Maps

Typical acoustic source plots from a wing are shown in Figures 79 – 87. Figures 88 - 105 show acoustic source maps with the aircraft wing outlined over the plot. Note that for each source plot the colour scale is automatically adjusted to the maximum level in the plot, so that the colours for different plots are not directly comparable. The dynamic range is always around 12 dB – 14 dB as this is a reasonable setting to use to effectively pick out sources from above the background noise.

Included below are some examples of the data plots acquired on the test days for a comparison of data results. The low frequency results are clearly of a lower resolution, whilst higher frequency results are of a higher resolution. The plots have all been capped to a dynamic range (a different range for each frequency) allowing same frequency plots from different configurations to be compared for relative levels. These typical plots show how the resolution of the noise sources varies through the frequency range and show the different range levels assigned to each frequency.

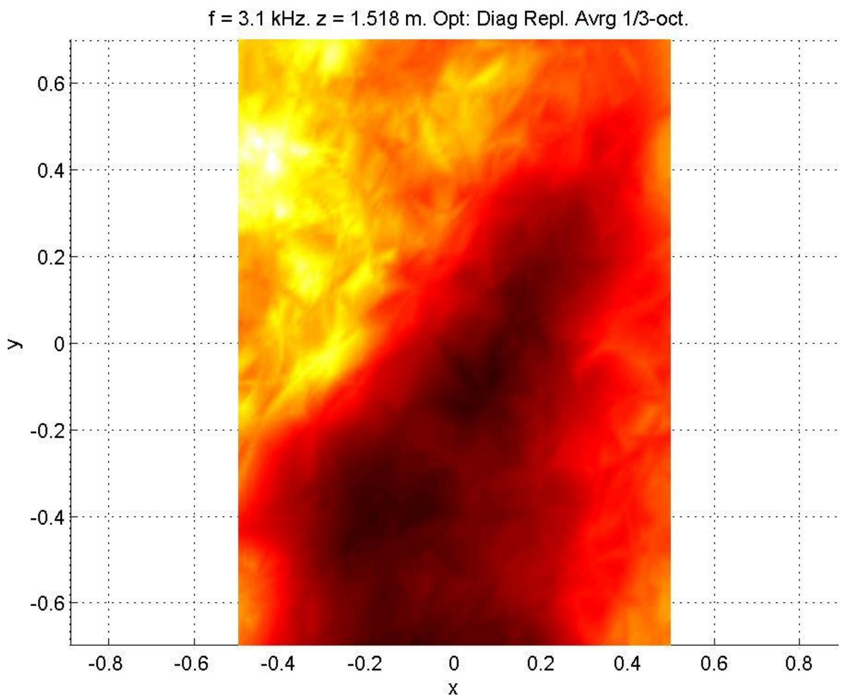


Figure 79 – Plot showing a typical result from the wind tunnel tests at 3.1 kHz.

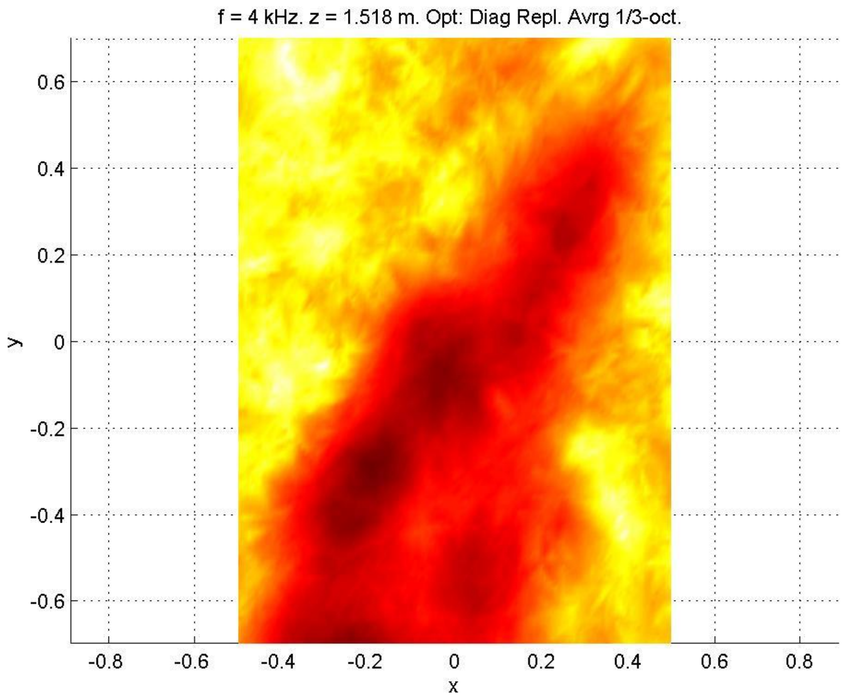


Figure 80 – Plot showing a typical result from the wind tunnel tests at 4 kHz.

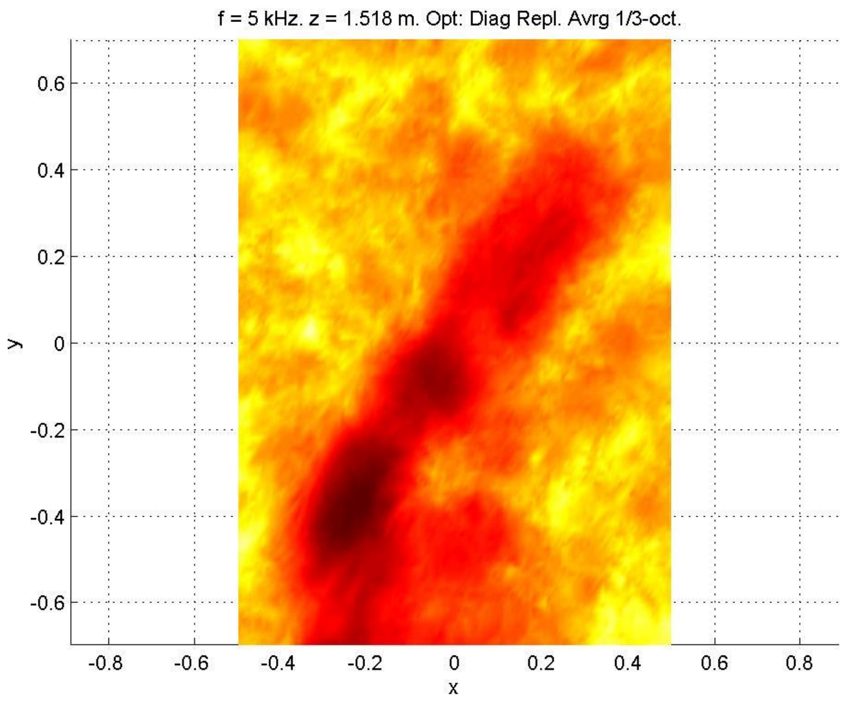


Figure 81 – Plot showing a typical result from the wind tunnel tests at 5 kHz.

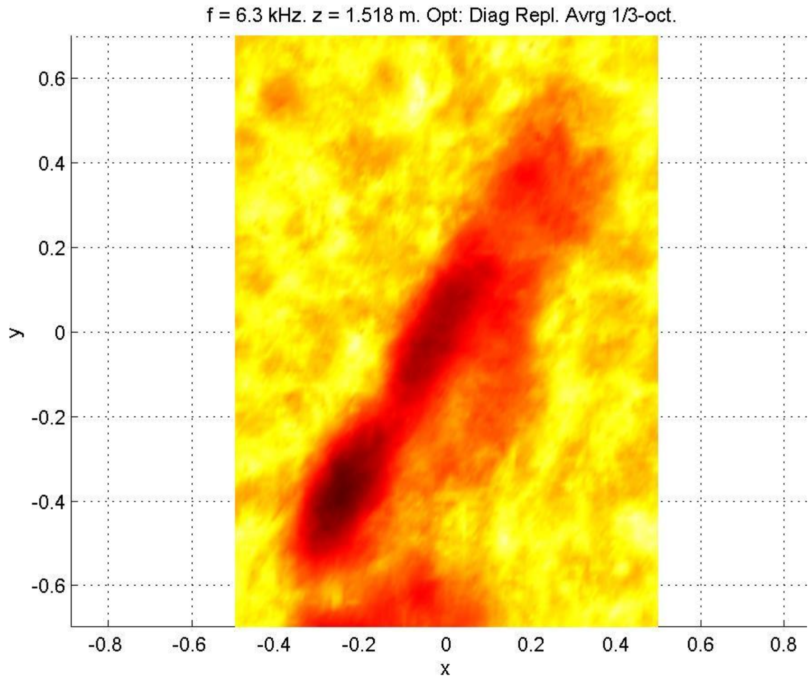


Figure 82 – Plot showing a typical result from the wind tunnel tests at 6.3 kHz.

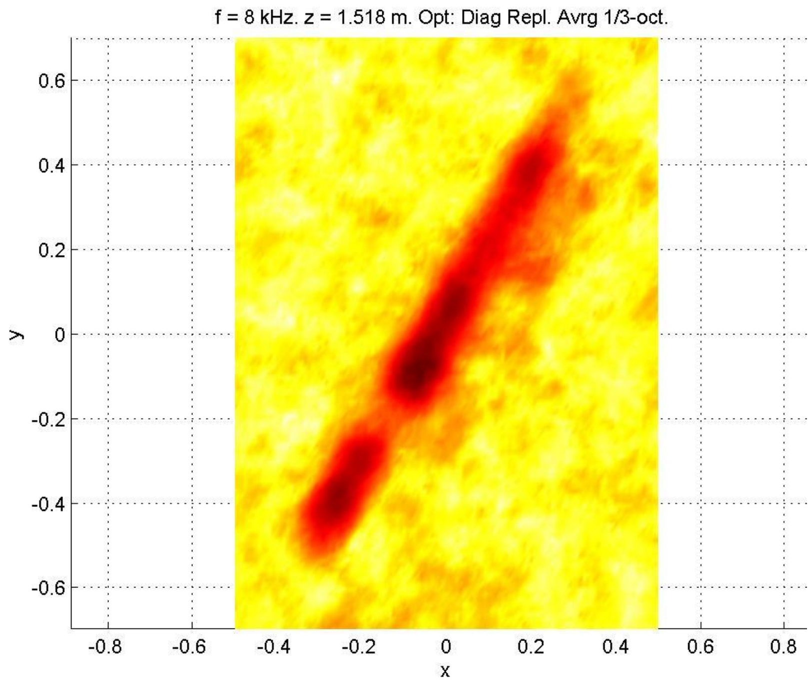


Figure 83 – Plot showing a typical result from the wind tunnel tests at 8 kHz.

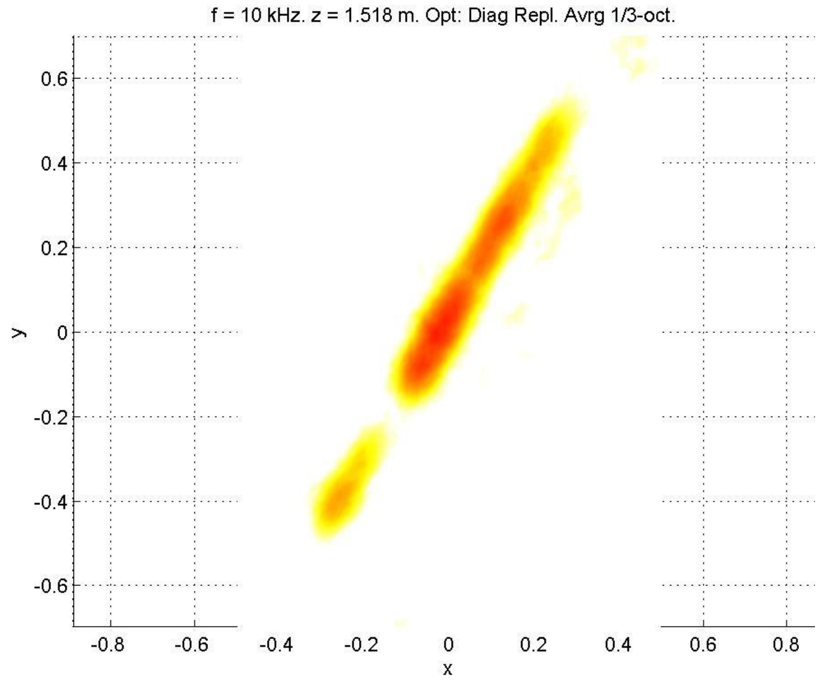


Figure 84 – Plot showing a typical result from the wind tunnel tests at 10 kHz.

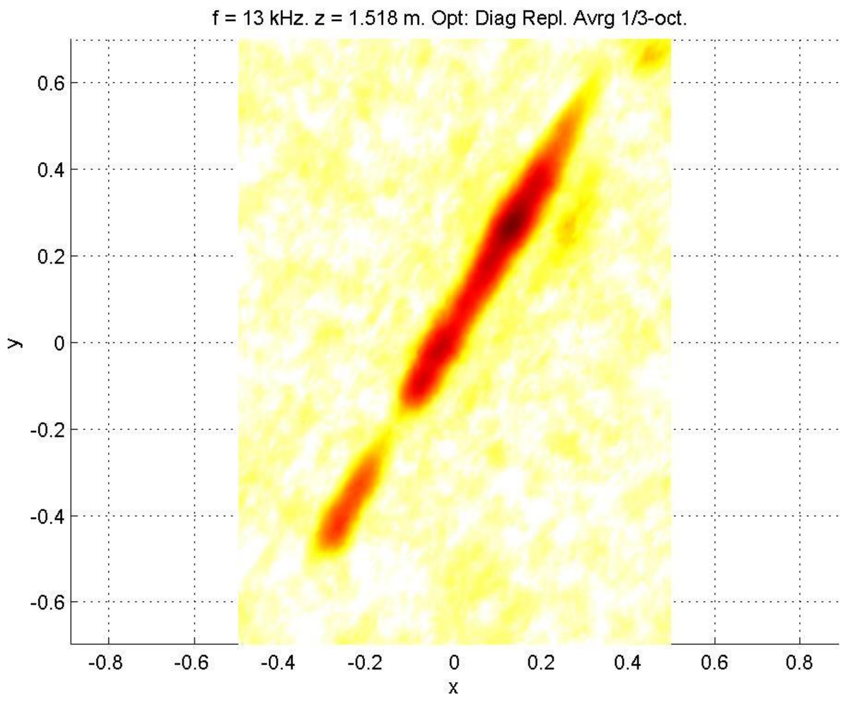


Figure 85 – Plot showing a typical result from the wind tunnel tests at 13 kHz.

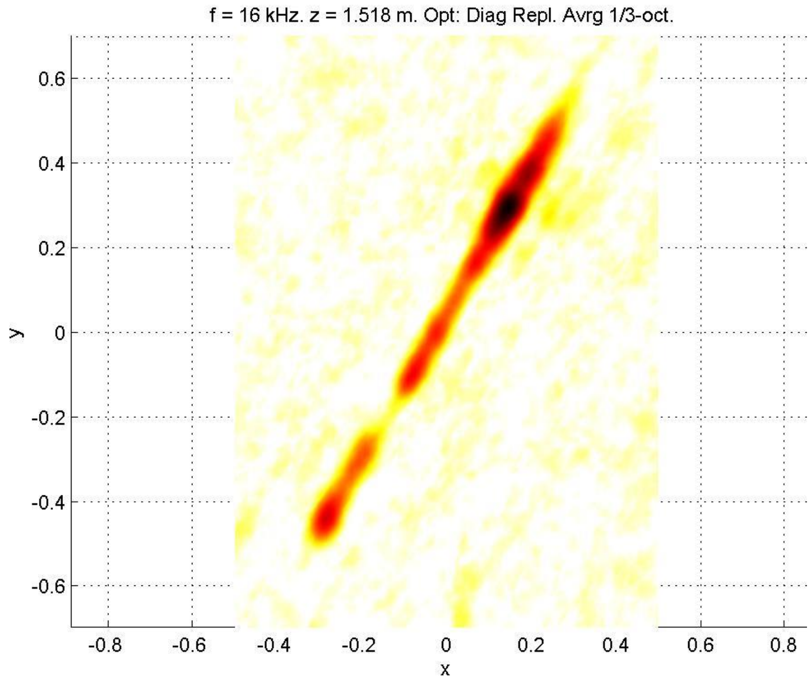


Figure 86 – Plot showing a typical result from the wind tunnel tests at 16 kHz.

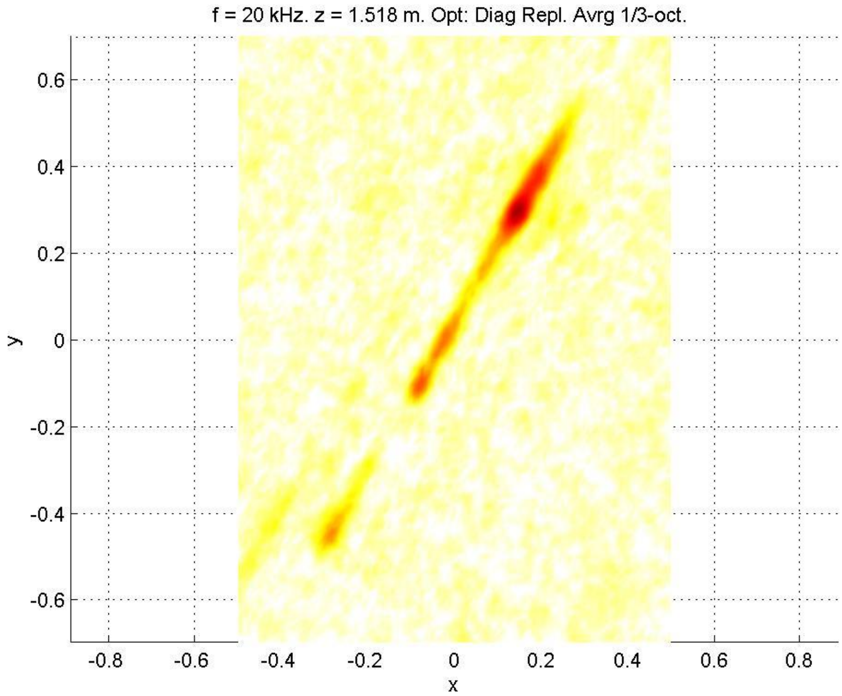


Figure 87 – Plot showing a typical result from the wind tunnel tests at 20 kHz.

4.1.6. Acoustic Source Location Comparison

The following four sets of results refer to four different configurations and are shown with an outline of the aircraft wing from which the sound source plots were derived.

These images show the plots in the context of the aero-acoustic test with reference to a real world model. To speed up processing only the area around the wing was processed, although the region over which the data was acquired extended to the forward fuselage, root of the opposite wing and the rear fuselage near the tail section.

Two of the configurations are considered quiet configurations and are compared in the following pages to a loud configuration and a configuration which gave distinct noise sources on certain points of the wing.

By comparing same frequency plots it can be seen how different configurations have vastly different noise levels as well as common noise sources that can be easily identified and if necessary treated.

The plots were capped to a fixed dynamic range (a different range for each frequency) allowing same frequency plots from different configurations to be compared for relative levels. The scale for each plot in this comparison emphasises that only same frequency plots can be compared. This is so as each of the plots is derived from median averaged values of a number of plots within a one-third octave band. This averaging is necessary as this reduces the effects of any sidelobes and negates any randomly falsely misleading illusory tonal sources.

The images on the subsequent pages (Figure 88 – 105) are arranged to compare different configurations to each other with same frequency plots on the same page. Figures 88 – 96 show a quiet configuration compared to a series of plots which show distinct noise sources, whilst Figures 97 - 105 show a loud configuration to a different quiet configuration.

The dimensions in the X and Y axis are in metres, and the darker colours (red and black) indicate the presence of sound.

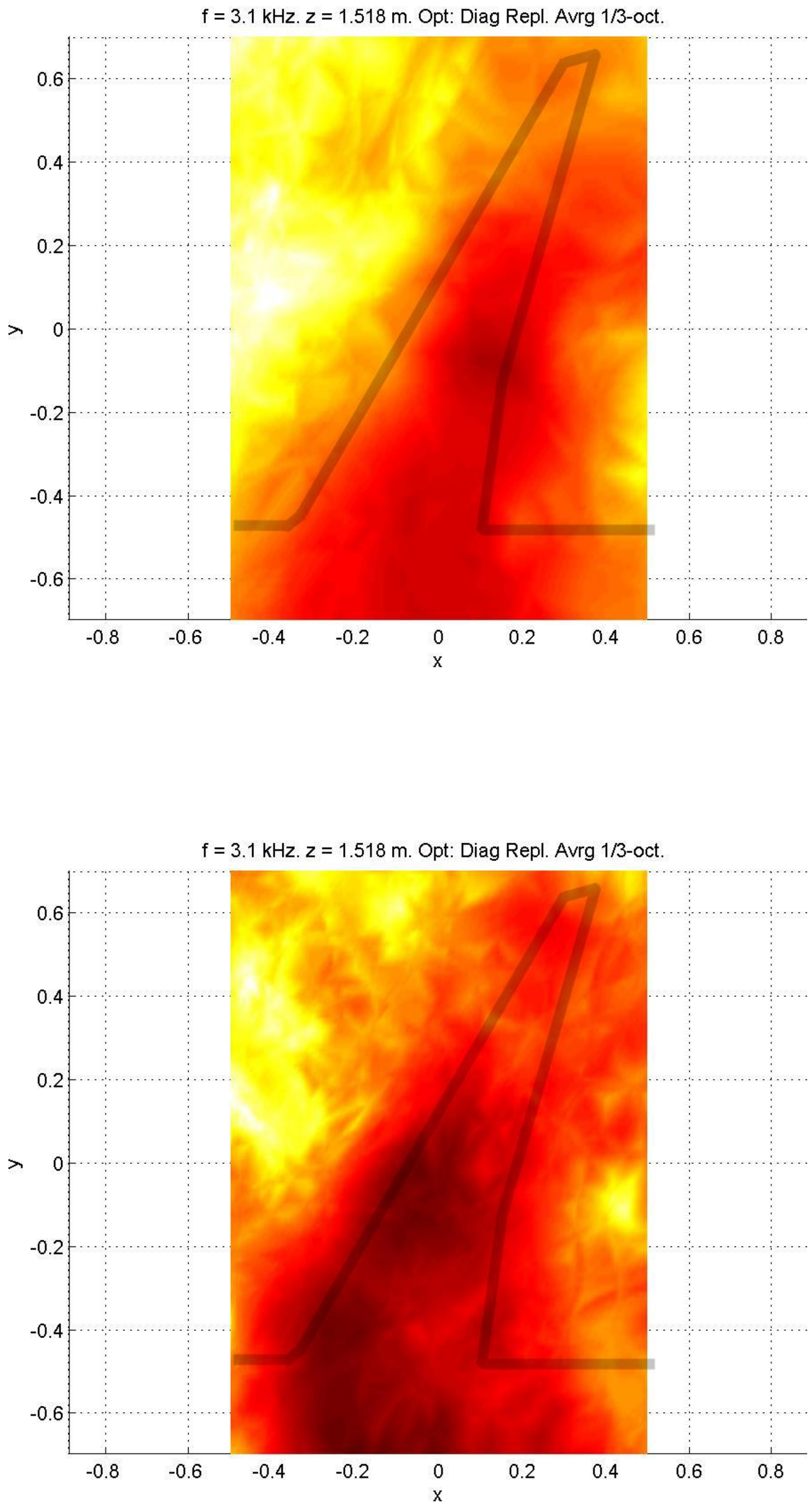


Figure 88 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 3.1 kHz.

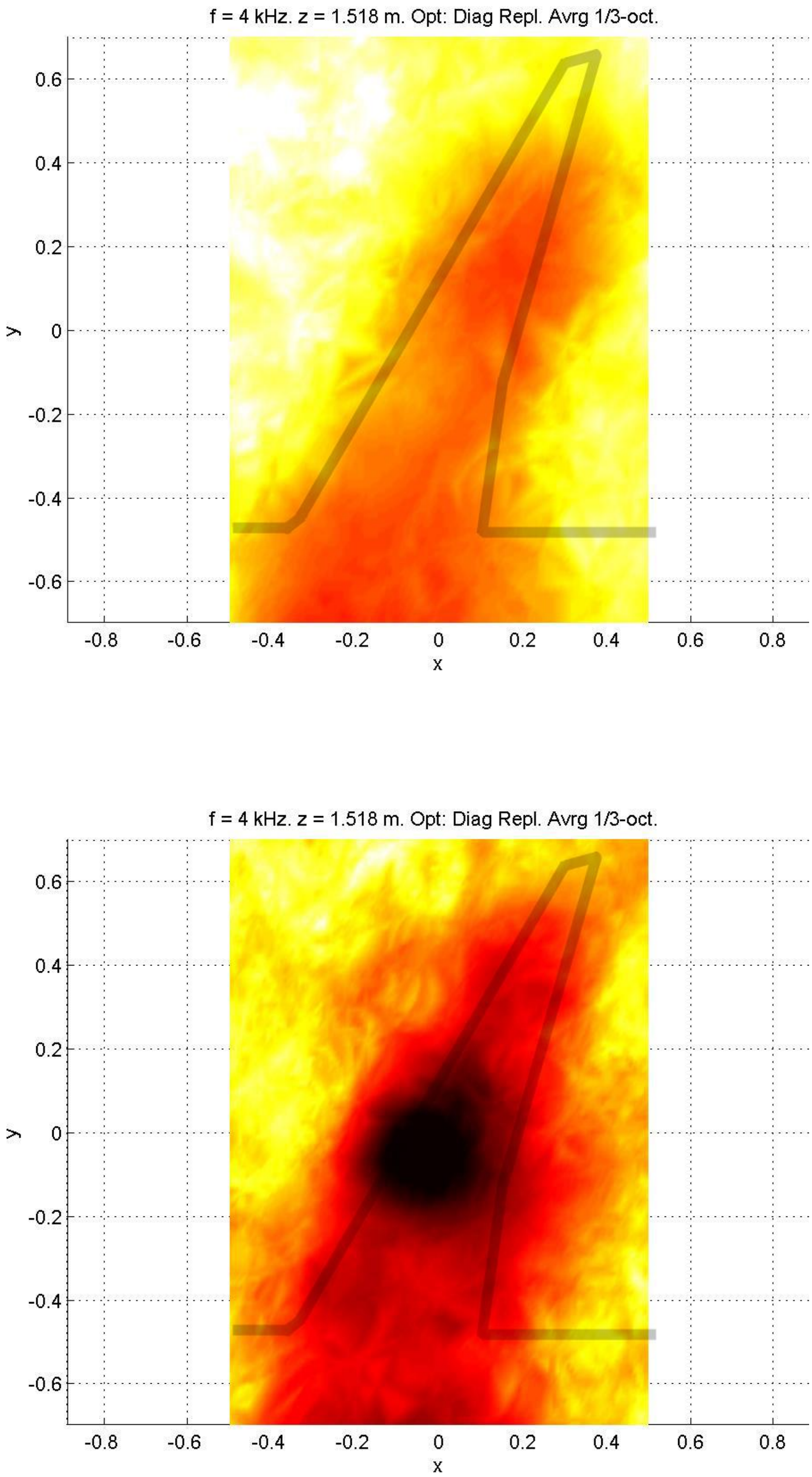
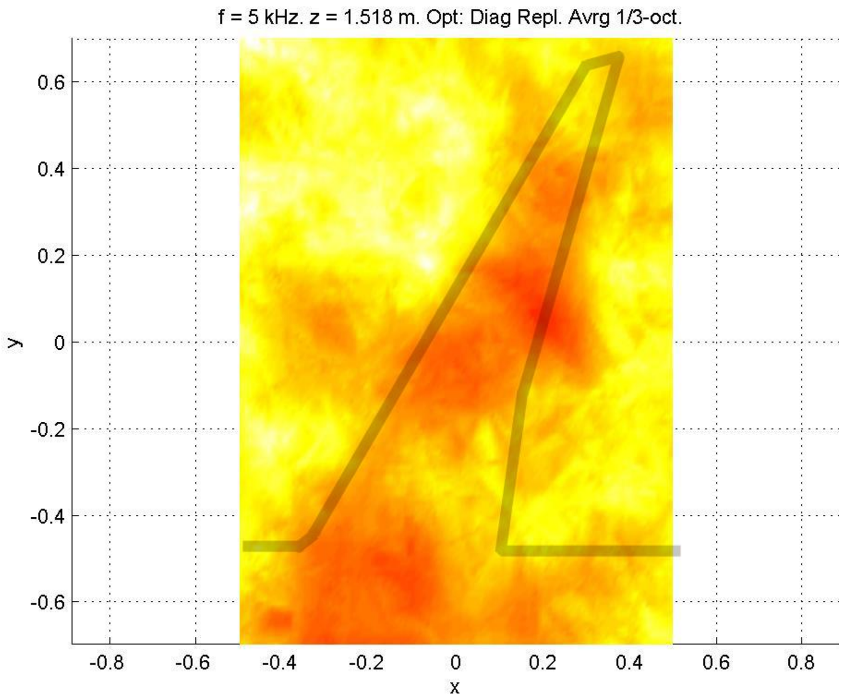


Figure 89 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 4 kHz.



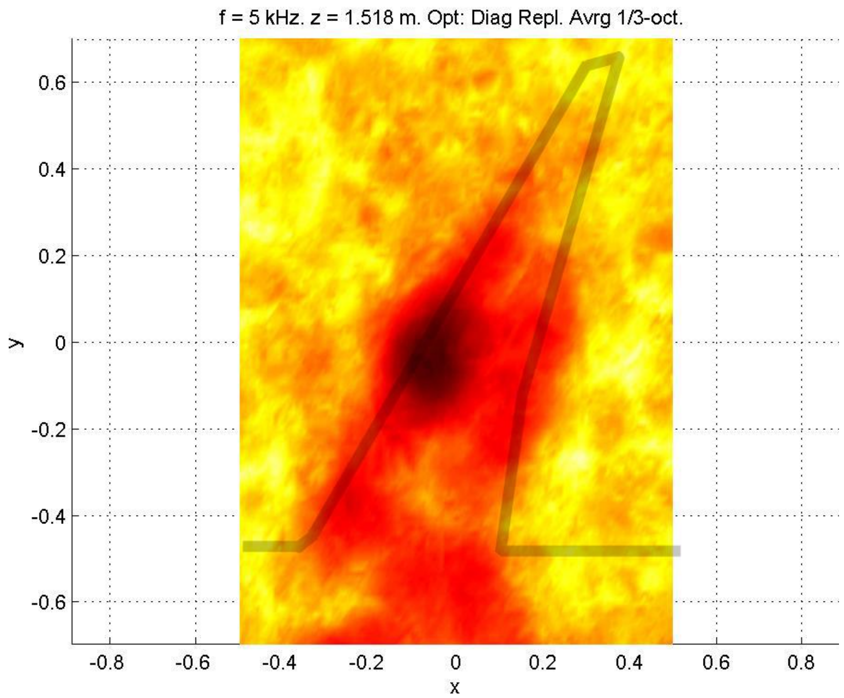


Figure 90 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 5 kHz.

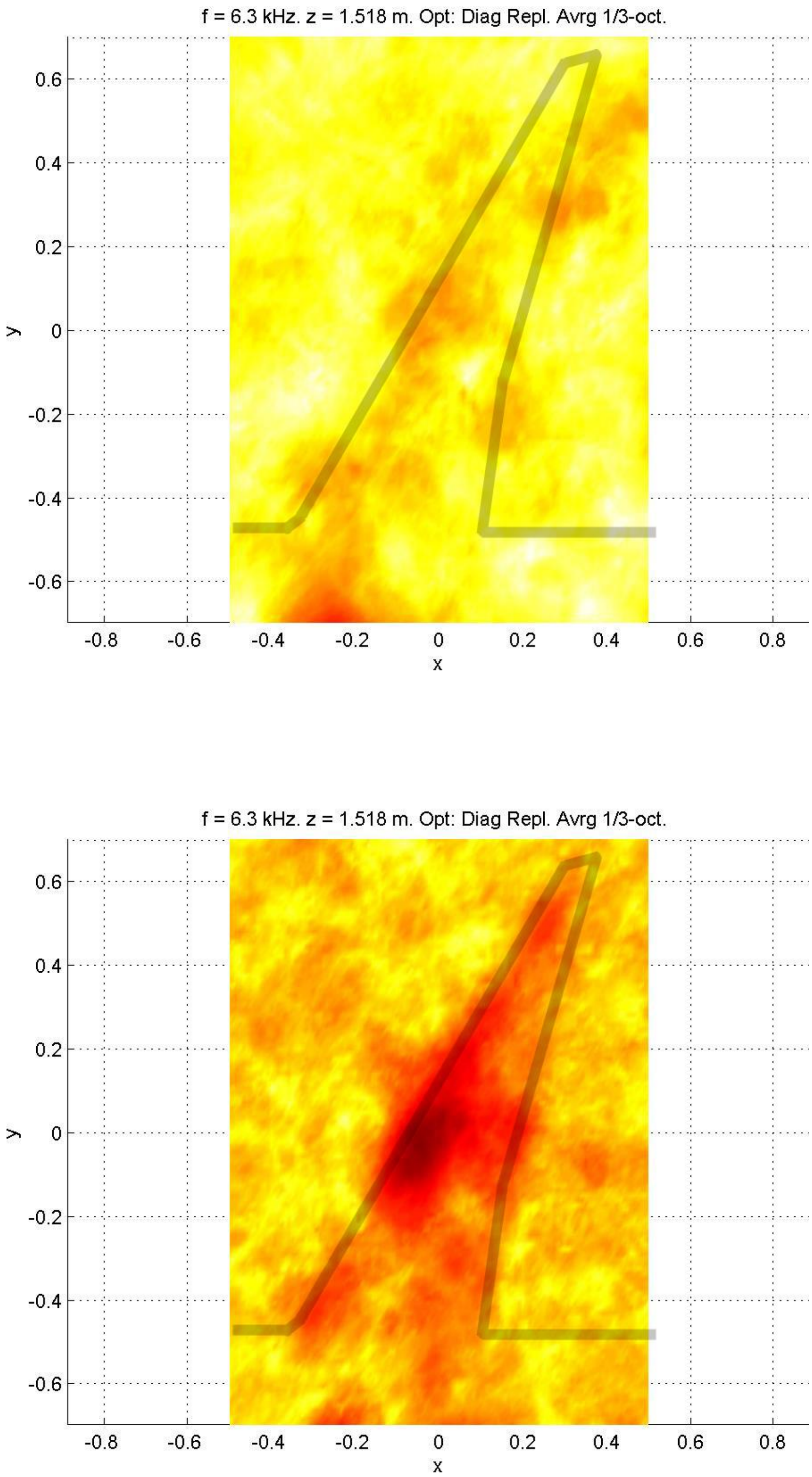


Figure 91 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 6.3 kHz.

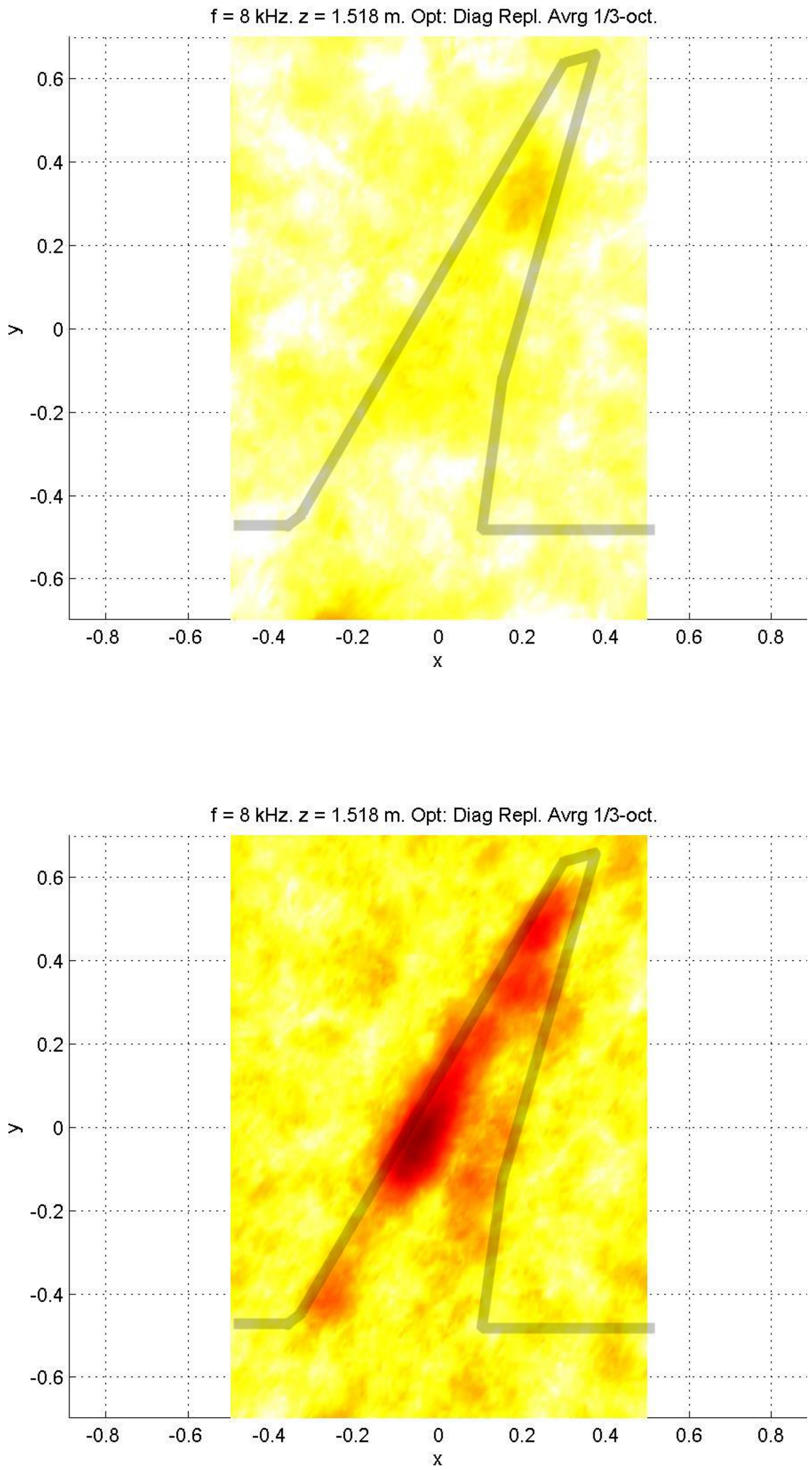


Figure 92 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 8 kHz.

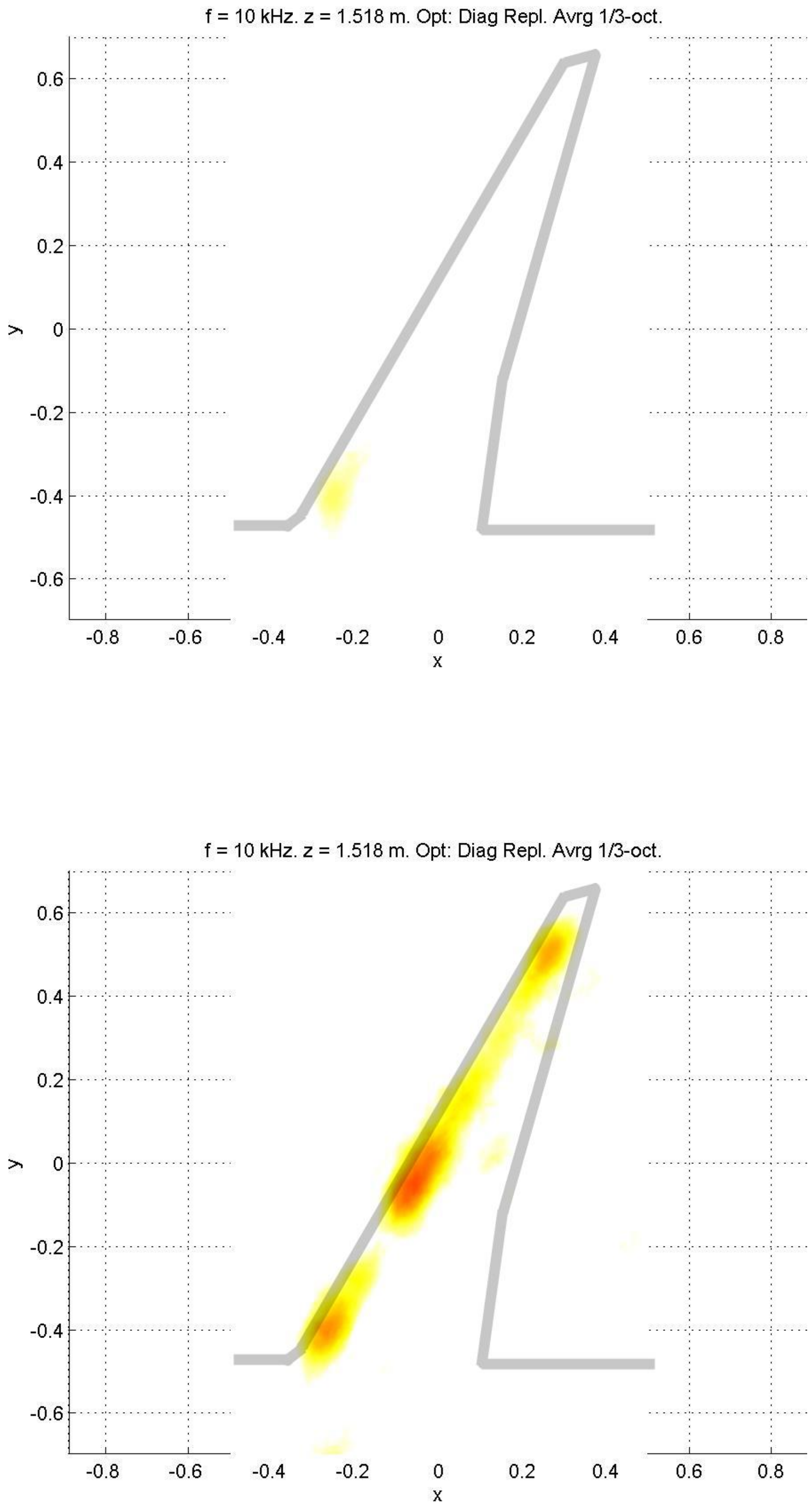


Figure 93 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 10 kHz.

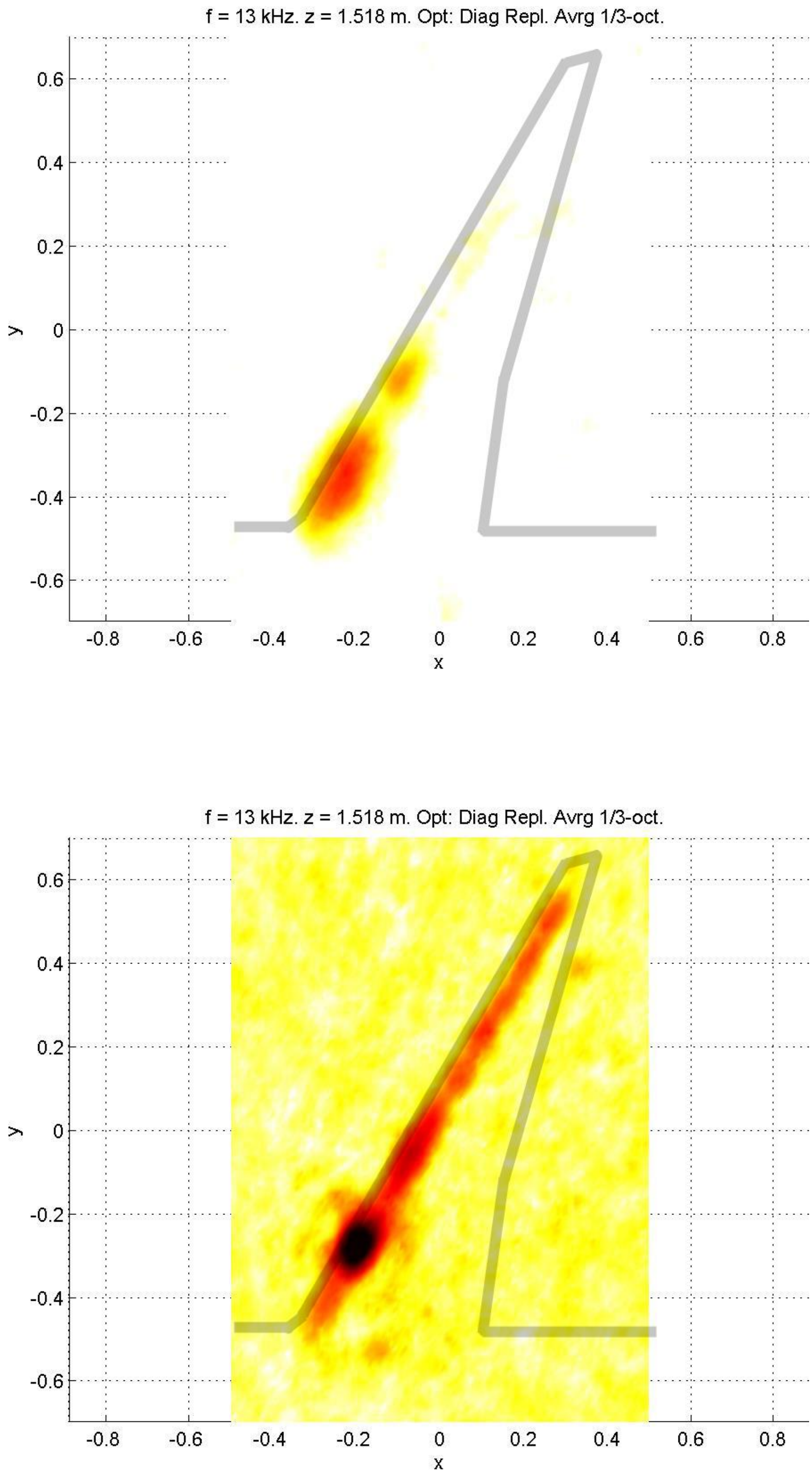


Figure 94 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 13 kHz.

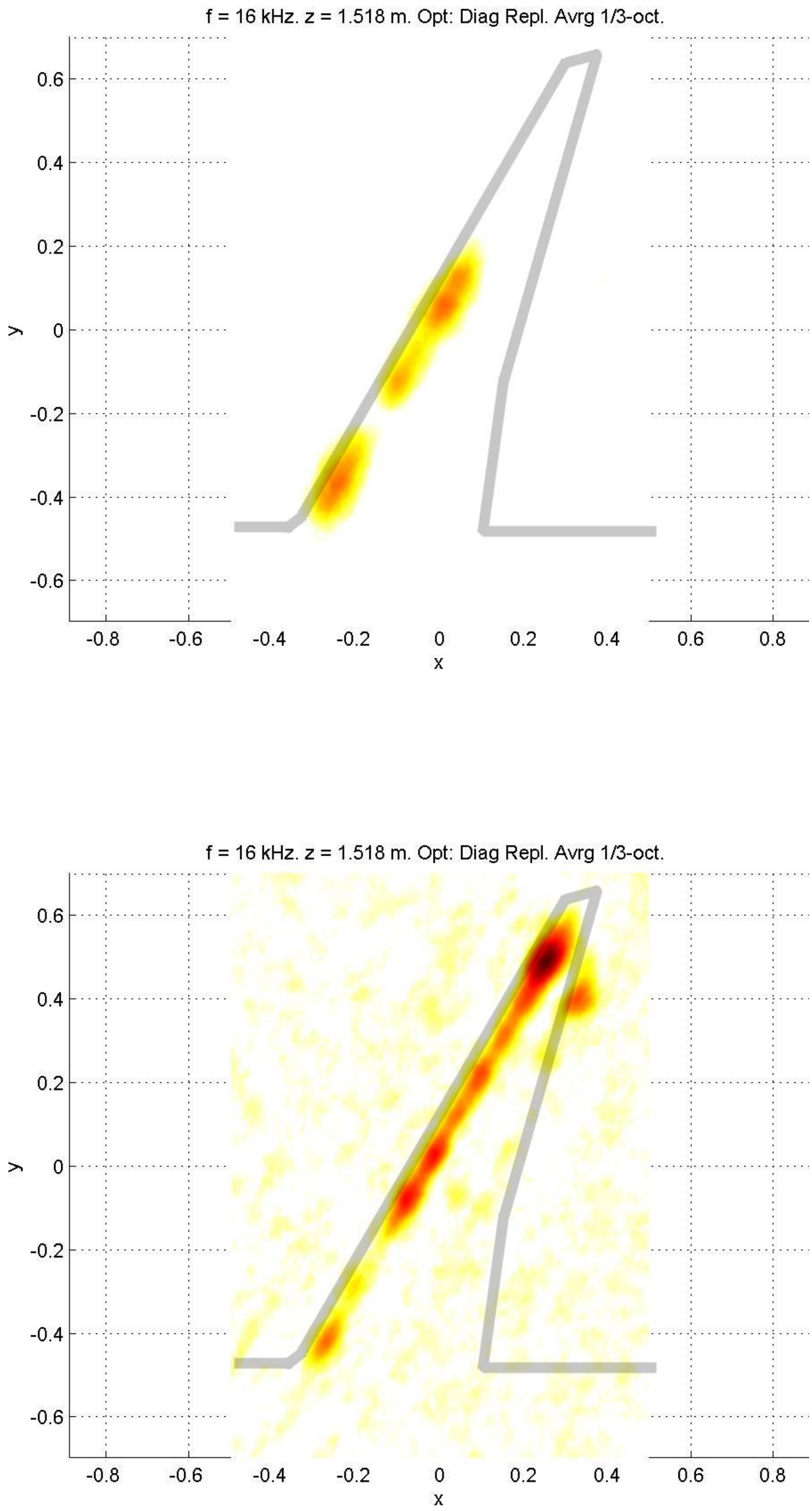
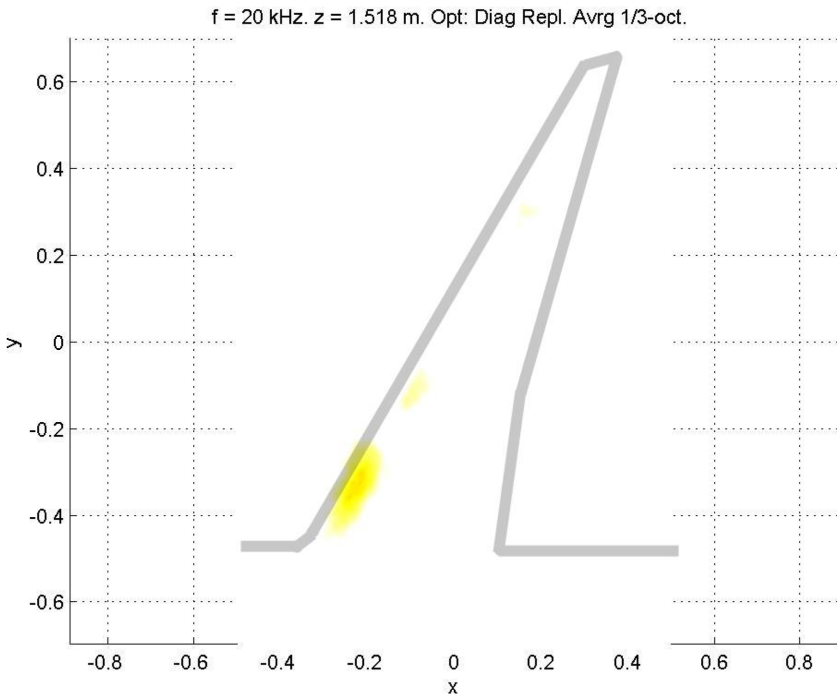


Figure 95 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 16 kHz.



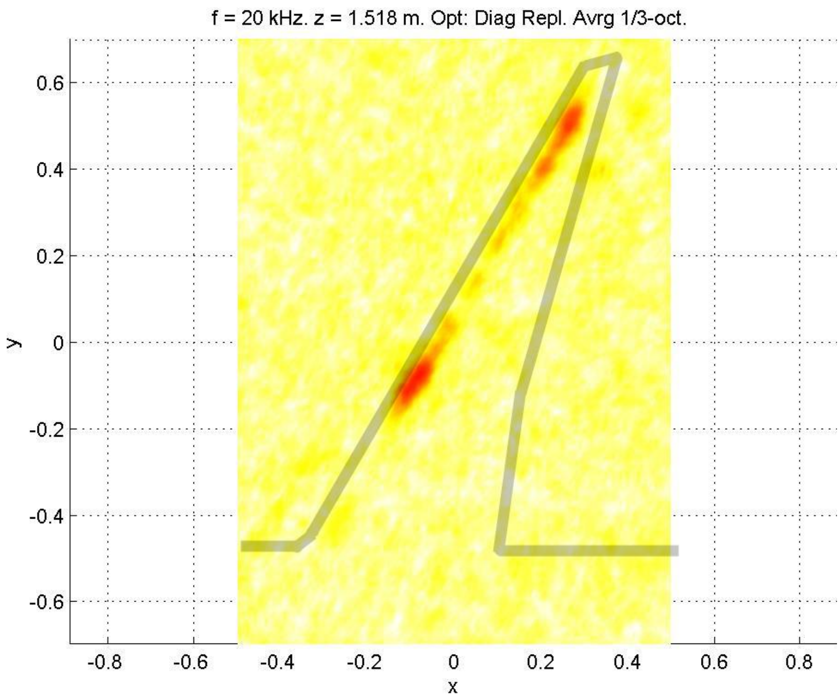


Figure 96 – Comparison between a quiet configuration (top) against a distinct noise source (bottom) at 20 kHz.

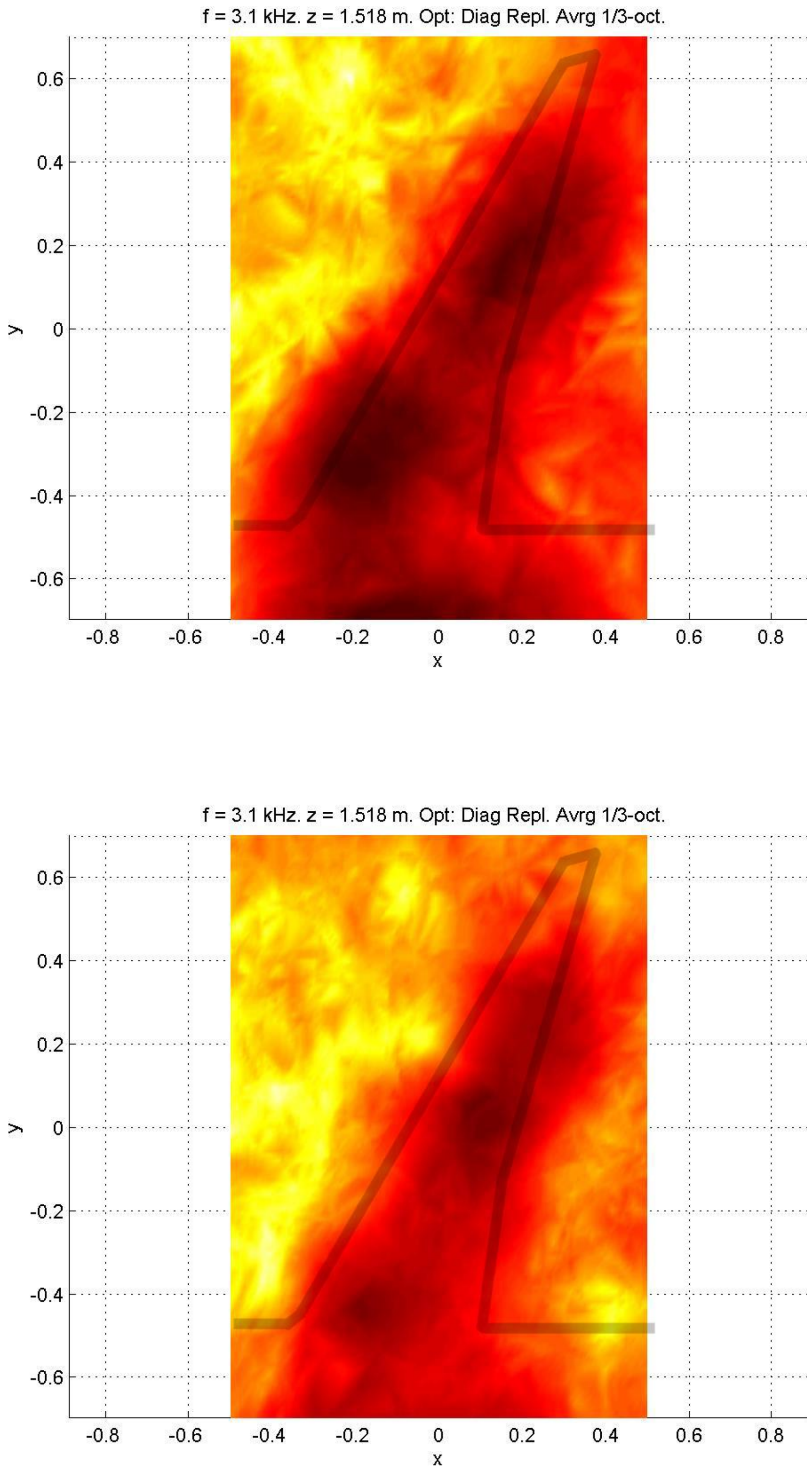


Figure 97 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 3.1 kHz.

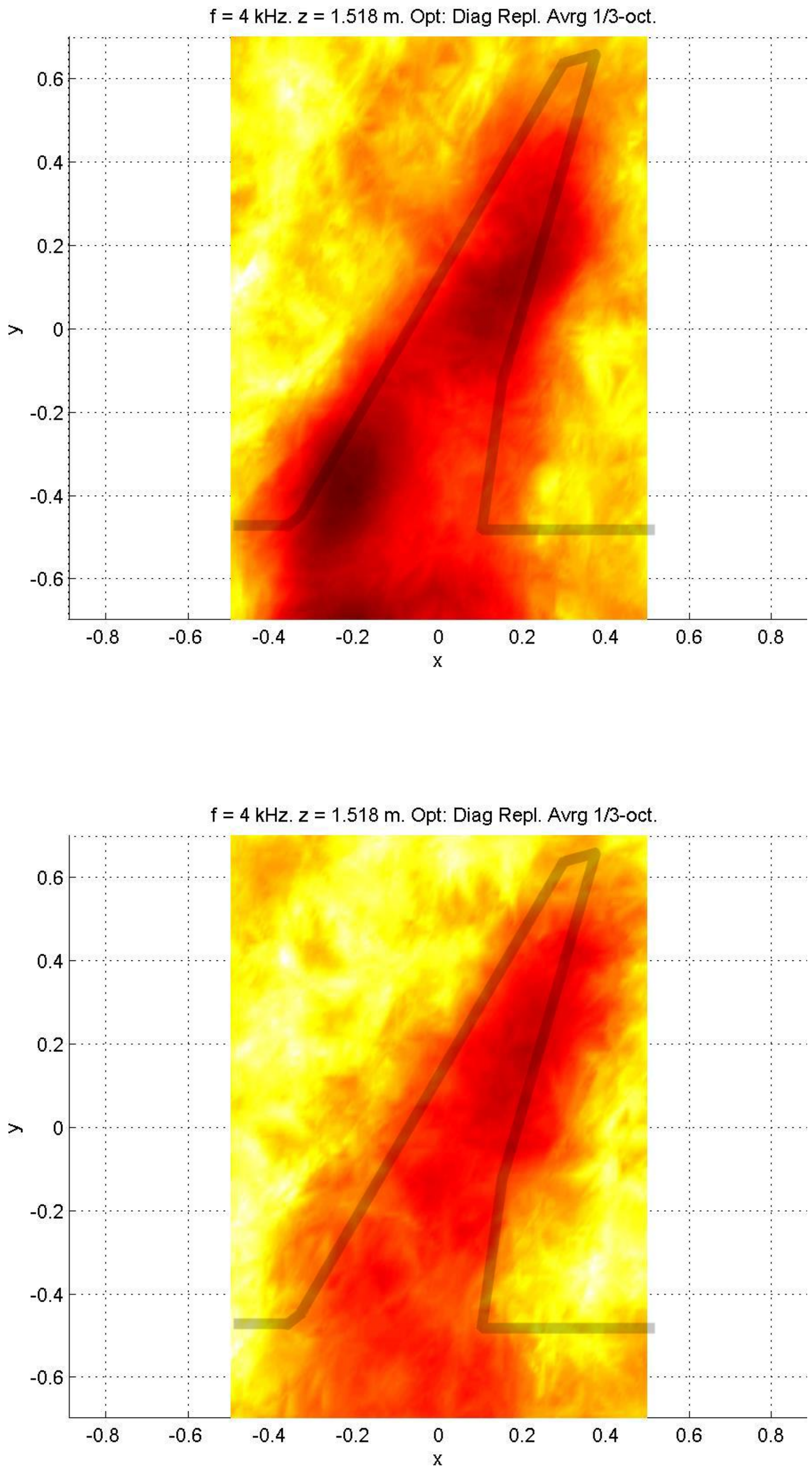


Figure 98 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 4 kHz.

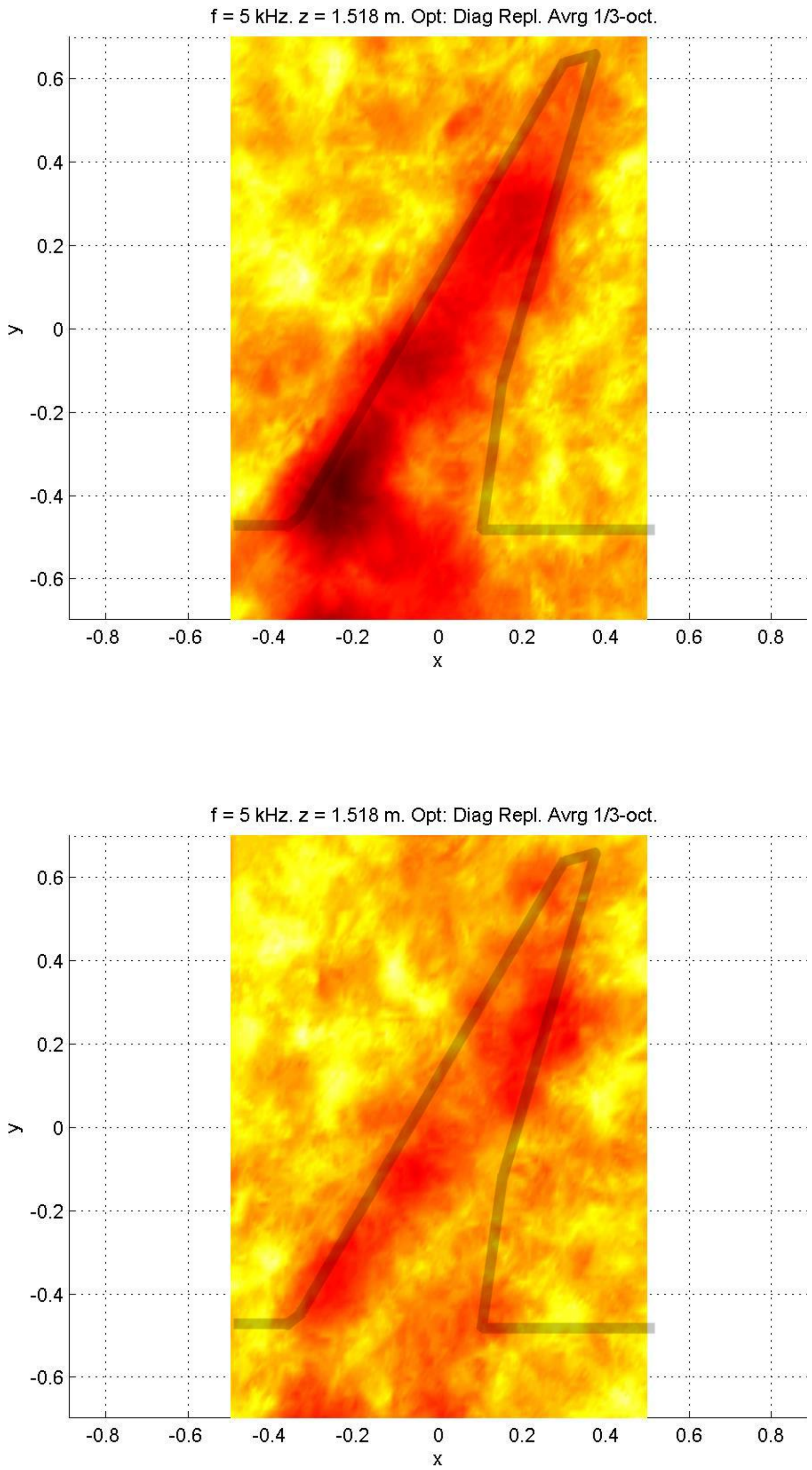
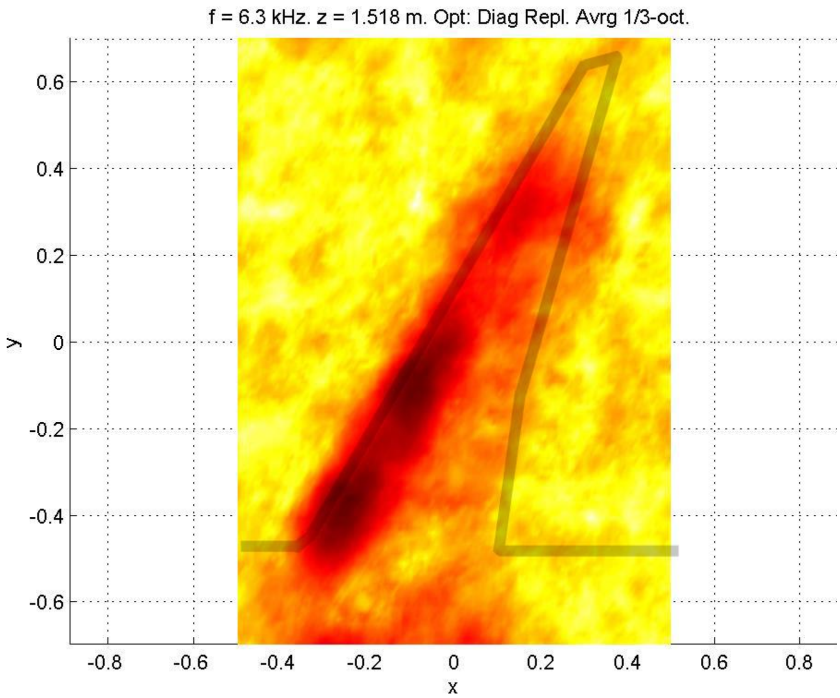


Figure 99 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 5 kHz.



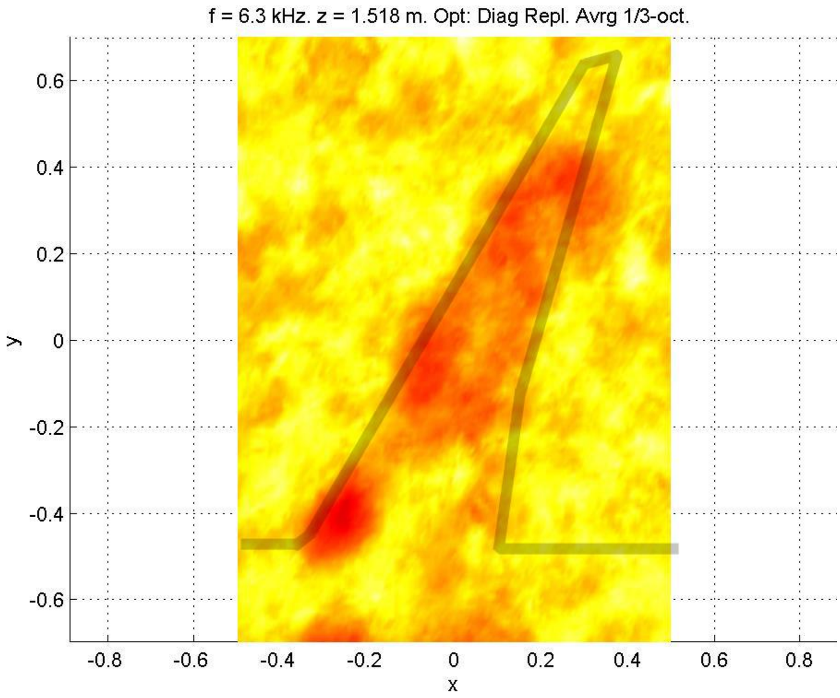
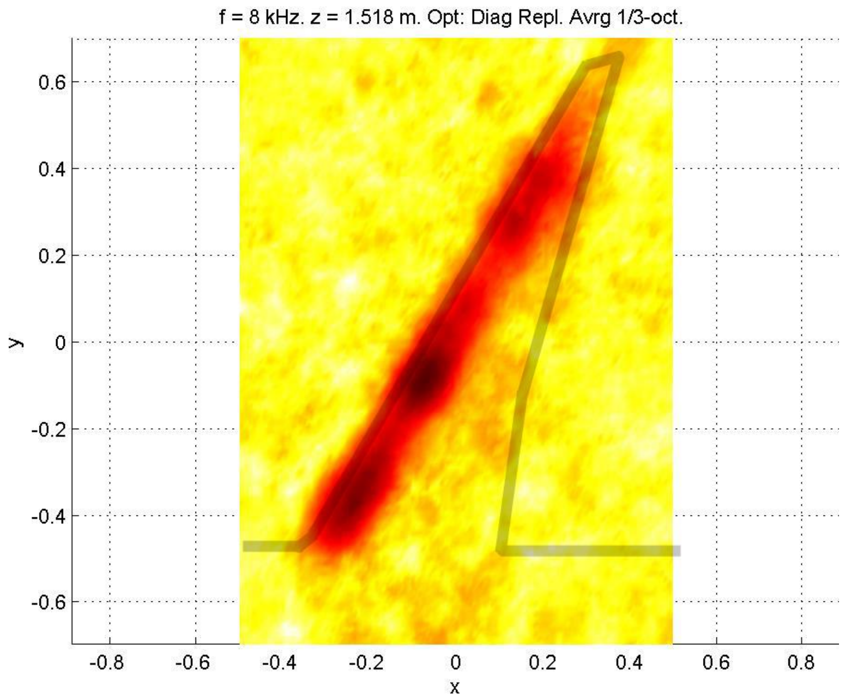


Figure 100 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 6.3 kHz.



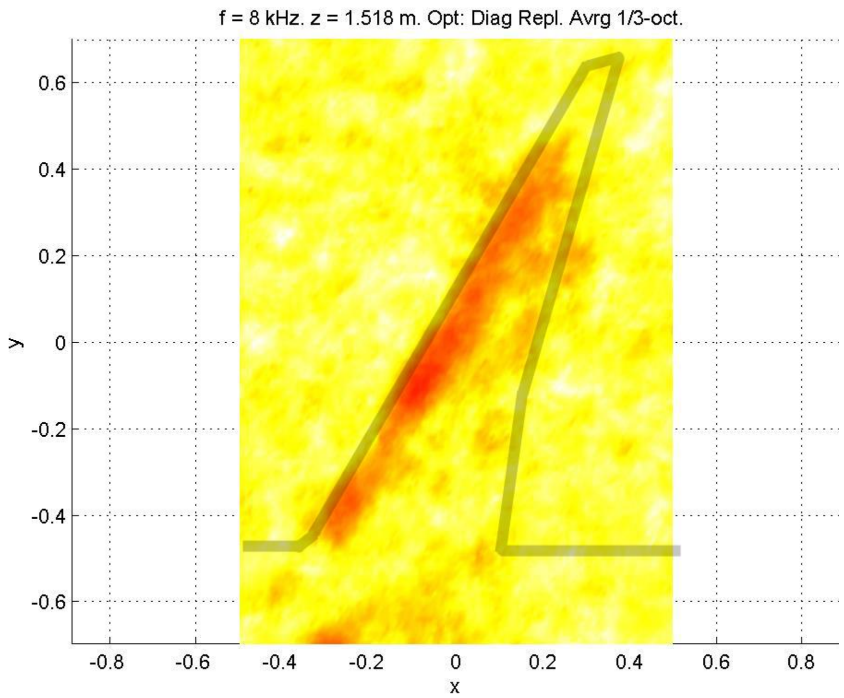
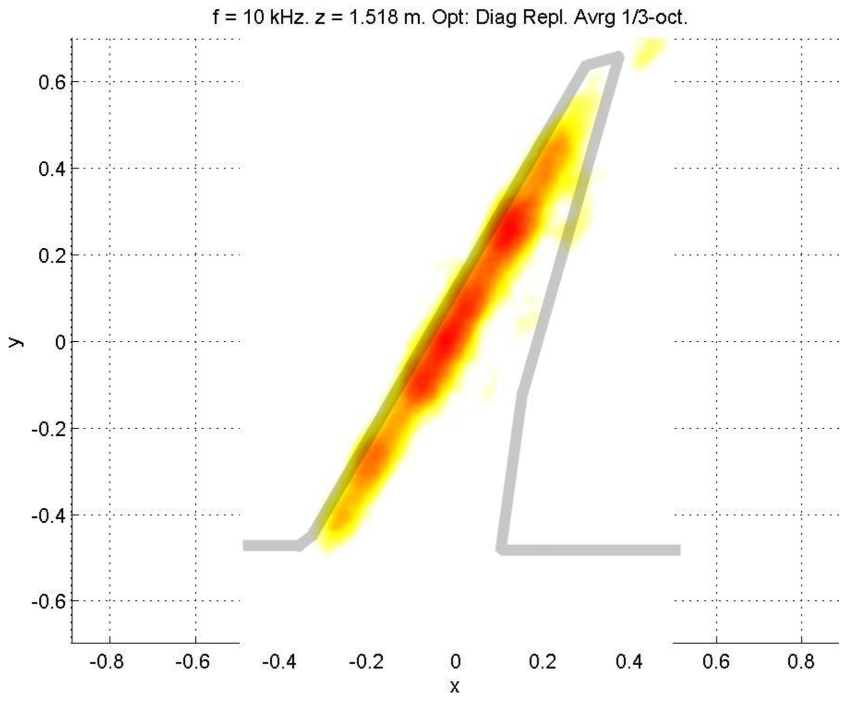


Figure 101 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 8 kHz.



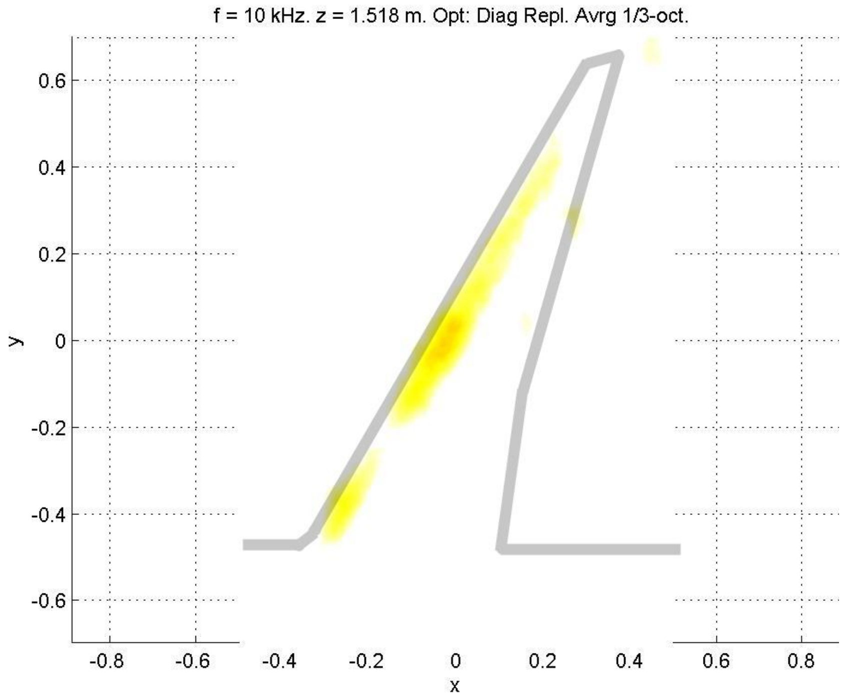
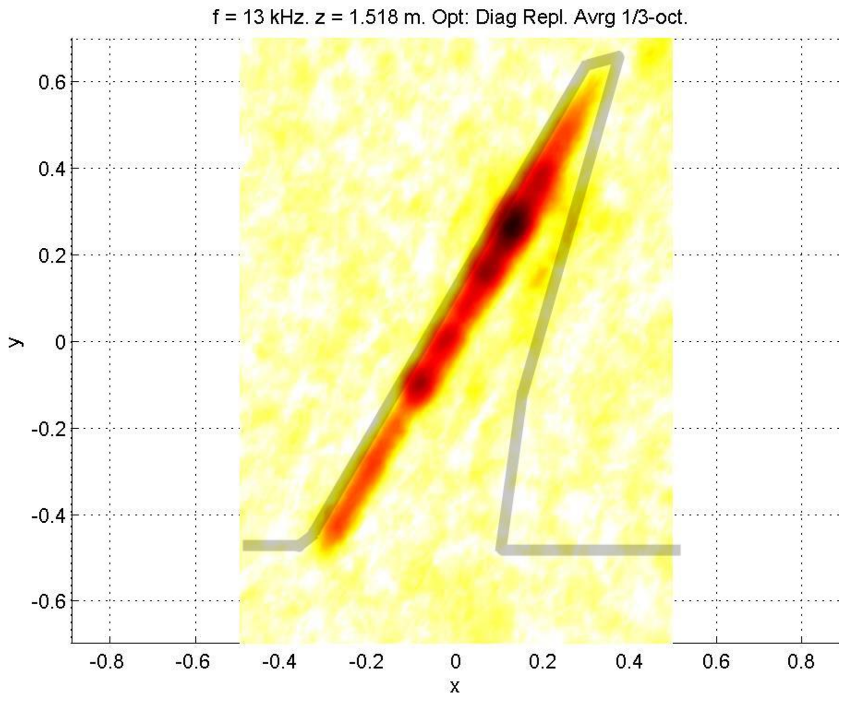


Figure 102 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 10 kHz.



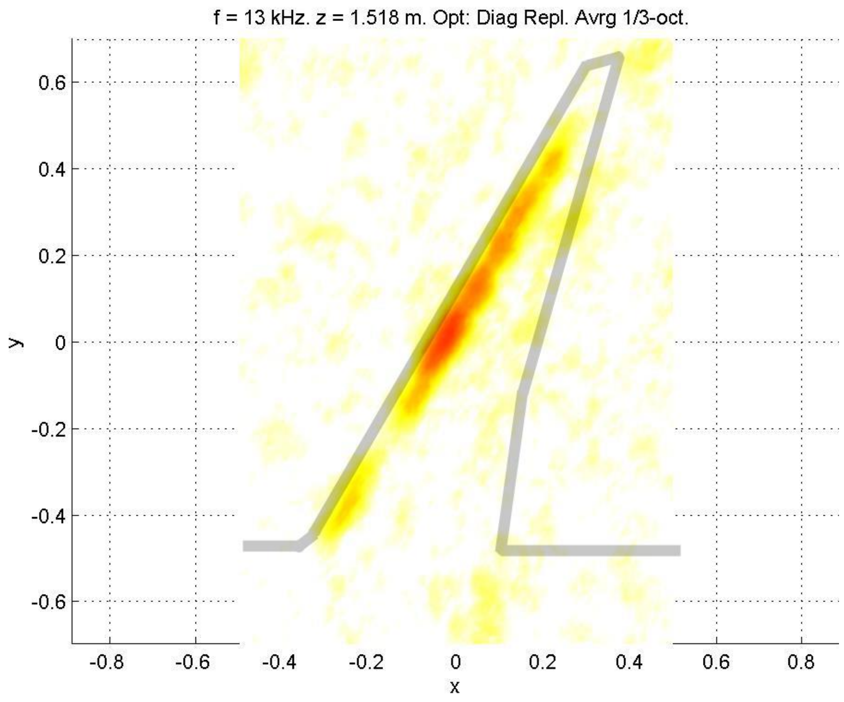


Figure 103 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 13 kHz.

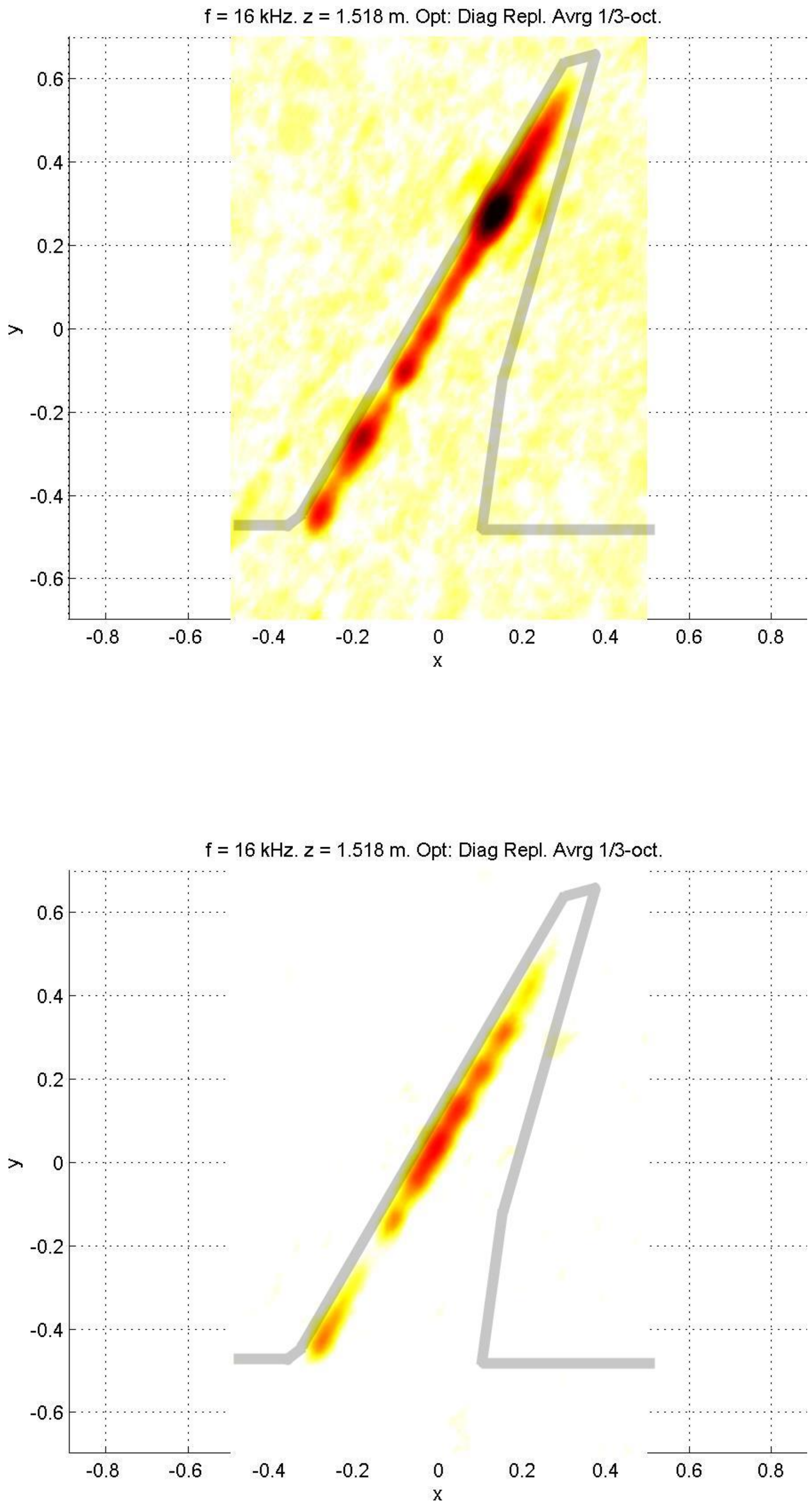
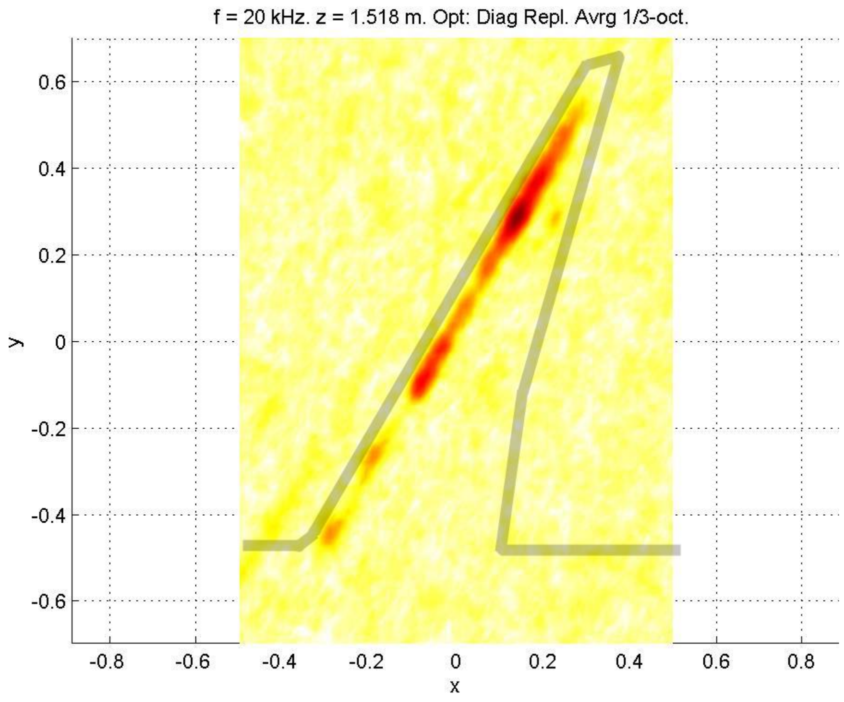


Figure 104 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 16 kHz.



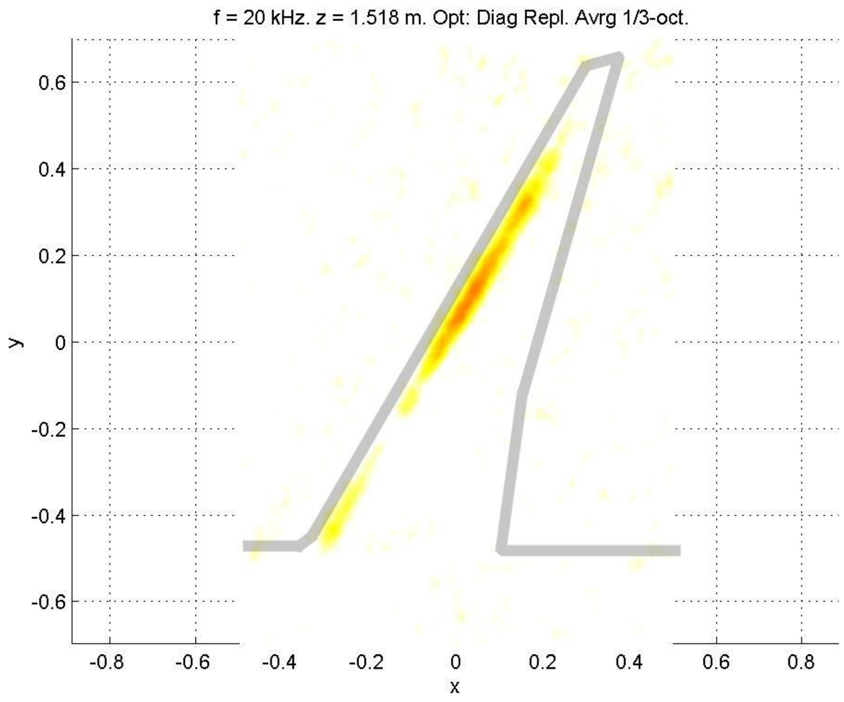


Figure 105 – Comparison between a loud configuration (top) and a quiet configuration (bottom) at 20 kHz.

4.1.7. Recommendations

In this section the known issues are listed and possible solutions and improvements are listed, additionally here the lists of the proposed modifications for the improvement of the acoustic source location system and the delivery of results are included.

Improvements to the user interface needs to be made as well as combining the array technology fully with the wind tunnel's own infrastructure. The latter would be beneficial as this would allow for secondary data - such as wind tunnel speed and temperature to be automatically associated with the appropriate acoustic data.

The sound source map plots should always be capped to a fixed dynamic range (a different range for each frequency) of at least 10 dB (12-14 dB recommended) allowing same frequency plots from different configurations to be compared for relative levels, and giving a clear image of sound sources.

Several simple solutions for some of the more physical problems such as the long wiring time and cable issues include wiring racks and fixing the wires in groups to the array underside. Binding large numbers of cables together is another obvious step for aiding in quick installation and de-rigging. Future installations may be of a more permanent basis and as such large bindings may prove inefficient with regards to maintenance and easy access. An intermediate step where a number of cables are bound together into one connector is potentially the best solution here. Such a solution was partly attempted with the new University of Southampton Pre-Amplifier box which had 16 cables wired to a single plug. Naturally multi cables to a single plug is only required if components are going to be moved on a regular basis as fixed connections are less likely to fail.

Limitations on the frequency due to the acquisition settings can be avoided by upgrading the hardware (specifically of the controller computer and acquisition cards). Additionally upgrading these components would lead on to allowing more channels to become a possibility as the current 112 channels is the limit of the hardware capability.

4.1.8. Further Development

There is scope for further improvements to the software used to drive the acquisition and processing of the acoustic data. Improving speed and the amount of data processed with a view to increase the number of microphones will be possible in future designs as hardware processing speeds increase.

Continuous improvements are being made to the processing code to provide better, clearer results and more accurate, higher resolution plots.

A number of experiments should be run to determine and confirm that a change in temperature in the post-processing of a few degrees has little effect on the results. An average temperature was used for some of the runs (as data from the day was lost). Temperature data was extrapolated from previous runs and conditions on the day. These values were determined to be accurate enough for the results.

Recent studies have also confirmed that the electret microphones used in the array do function up to 44 kHz [references 58 and 60]. Although it should be noted that to date there are no published results anywhere that show results above 35 kHz.

In the lower frequency region it would probably be necessary to increase the physical size of the array, as having a larger array would most likely improve the resolution of the low frequency results.

Using the CLEAN SC algorithm, [56] improved the July 2007 low frequency plots making some become useful, especially in determining source location. The CLEAN SC algorithm will be used retroactively to process the low frequency results from the November 2006 test, to further verify and study the improvements in sound source location at lower frequencies.

Another concept for array design would be to have a large array for low frequency measurements with a smaller section inside which would be used independently to measure the high frequency levels. For all these cases further study is required in determining the optimum microphone spacing and location for the best results and resolution at the various frequencies of interest.

If it becomes possible the suggestion exists to perform tests without the beams and rails of the working section circle. This would allow a perfect response to be attained, and comparisons could be made to the data already acquired. The possibility remains in installing a larger array in the ceiling of the Filton wind tunnel test section.

Nonetheless modifications will be made to improve the array to work around the wind tunnel components. If such a test is possible however, it would become possible to process results with and without the ‘lost microphones’ to simulate having the beams this would allow a determination of the effect the beams have on processed results.

A further possibility is a set-up, offset to a centrally arranged array, with the array centre located more under the swept back wing of the model. Although an off centred array would be much harder to achieve a balance point spread function for. A centred array at the moment is still best for overall distribution.

Further study into a variety of shaped arrays should also be considered, as well as extending into 3D arrays [64] to give a total acoustic picture of any wind tunnel model from all aspects.

Combining the acoustic data with other wind tunnel techniques such as flow visualisation, Particle Image Velocitometry (PIV) and infra-red thermal analysis is the most logical next step, with a view to develop an overall system to give a whole model description over a variety of techniques, each verifying and clarifying the other in a succession of plots and easily understood images.

4.1.9. Conclusions

A successful implementation of the beamforming technique and use of an acoustic array to determine noise sources on a wind tunnel model in the Filton wind tunnel has been carried out, twice, and major sources of noise identified from a model in flow conditions. This was achieved at various operating speeds, but most importantly at the standard operating speeds of 50 and 68 m/s (0.199 Mach), where concerns, of the overall background noise level affecting any source location, were made. [62]

The floor mounted acoustic array of microphones, as described in section 3.4, was used for these acoustic measurements. As can be seen in figure 106 the array was flush mounted in the test section under the model allowing simultaneous aerodynamics testing to be performed on the model.

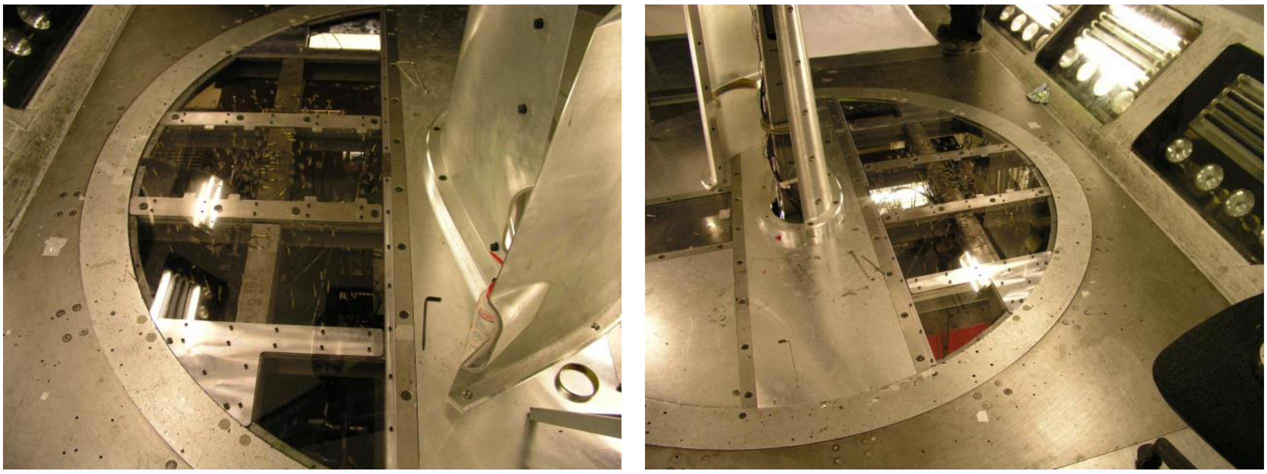


Figure 106 – Pictures of the floor mounted array in the wind tunnel for the November 2006 Filton test.

From the results it can be seen that the low frequency results are clearly of a lower resolution, whilst higher frequency results are of a higher resolution. Extremely low frequency results were poor but the implementation of the CLEAN SC code allows some of these low frequency results to provide some useful data.

The July 2007 test has led to the processing aspect of the data to be improved as the acoustic plots can now be automatically generated in Tecplot and improvements have been made to overall processing speed. This was achieved by using modified processing codes as well as upgraded hardware.

Some of the results return what appears to be a single source at the very centre of the array, sometimes with other faint sources encircling it, on closer inspection of the range it becomes apparent that this artificial source only appears at very low levels. This ‘artificial source’ relates to the point spread function of the array (with a central mainlobe and weaker sidelobes), it is effectively system noise and is easily identified as being at the very centre of the array on plots with extremely low levels and should not be confused with an actual source.

Further improvements are undoubtedly obtainable by using an increased channel count (more than 112 channels). The more channels in an array, the better the capability to detect noise sources which are otherwise buried in the background noise. However, this would require further upgrades in computing power, acquisition hardware and software improvements to handle the extra channels and the levels of data acquired.

As the array design was modified to fit around wind tunnel structure several problems became evident with results from the point spread function showing that there might be some improvements from installing a ‘perfect’ array (performing the tests without the beams and rails of the working section circle), although this would mean significant alterations to the wind tunnel would need to be made.

Other future possibilities to improve the array capabilities include acoustic lining of the test section opposite the microphone array; to avoid direct reflections from degrading the array processing and the design of a properly recessed array, which can give much better results; up to 20 dB lower levels at 5 kHz and 12 dB at 10 kHz, [13], however, the practicalities of installation and effect on overall wind tunnel performance must be fully examined before implementation of any of these concepts.

In summary, aeroacoustic measurements in the Filton wind tunnel can provide useful data for analysis of wind tunnel models, and there is scope for improving the equipment from the array design itself, such as recessing the microphones (section 4.2) to the hardware employed and the software used to allow more detailed, better resolution measurements across a wider range of frequencies to be carried out in the future.

4.2. Recess Testing

A Comparison of various recessing levels, differing microphone apertures and cloth types; for use in hard-walled, wind tunnel acoustic array measurements. A series of experiments were conducted in the University of Southampton's 3' by 2' Closed Section, Hard-walled, Wind Tunnel, to determine how different microphone set ups effect the acoustic data being received. Specifically a comparison between recessing the entire array of microphones versus recessing a single microphone was made as well as looking at various recessing depths, Cloths types and different microphone apertures.

In particular the study looks at varying levels of recessing, compared to flush mounted microphones as well as studying microphones behind two different types Kevlar cloths (and one silk one) as well as the effect of varying the shape and depth of the individual microphone hole. All the investigations were performed at varying wind velocities up to a maximum of 30 m/s. These tests conducted at different wind tunnel speeds were done with a sound point source consisting of a signal set to either white noise or different tonal frequencies

This section is based on a paper the author presented at the 2009 AIAA conference. Another paper validating recessing and screening microphones to positive effect [67] was published a year later, corroborating the findings in this study.

4.2.1. Recess Testing - Aims and Objectives

A series of Microphones are to be recessed to determine background noise effect and by varying the recessing levels, the best level for recessing for acoustic testing in a wind tunnel is to be determined. The tests in the 3 by 2 included the following points:

- Various styled cavity recessed microphone, Small section recessed microphones and flush microphones, compared to flush mounted microphone

- Testing of various materials used to separate microphones from boundary layer noise in wind tunnel.

- Testing of effectiveness of acoustic foam in recess gap in reducing background noise

4.2.1.1. Introduction

In current years the use of Microphone arrays in closed section Wind Tunnels for the purpose of sound source locating has been growing. Although today, measurements are still being made on a fairly limited basis in conventional wind tunnels by select organisations, it is however, increasingly important to acquire data as accurately and confidently as possible. With aeroacoustics becoming so important in everyday engineering and the need for more aeroacoustic testing growing it comes as no surprise that purpose-built facilities for acoustic testing have been constructed [13] but the current trend is to make efforts in installing microphone arrays into conventional wind tunnels, some of which may have not been modified acoustically at all.

Aeroacoustic testing in wind tunnels is becoming a necessary tool in the pursuit of quiet aircraft design. While open-jet anechoic wind tunnels can provide high-quality acoustic data, it is often difficult to exactly match aerodynamic testing conditions with those carried out in closed-section tunnels. Dedicated open-jet acoustic test campaigns also add to the time and cost in aircraft development programmes. It is, therefore, advantageous to be able to perform aeroacoustic measurements within hard-walled wind tunnels, although this is an acoustically challenging task. Microphone arrays can be used to provide source location and approximate integrated sound levels, although their performance is limited to the achievable signal-to-noise ratio.

Increasingly sophisticated array techniques are now available to improve array performance, from processing techniques that provide enhanced suppression of noise in the sidelobe regions of the array response [20, 23 and 56] to increasing the number of microphones and using improved hardware [21, 22 and 59]. In all these cases, however, the microphones still need to be mounted in an array and this array has to be positioned in the wind tunnel environment in the most effective way to minimize background noise effects.

The relatively high background noise level of these hard-walled facilities can be considered problematic. Originally microphones were mounted flush with either simple or no covering, as the methodology progressed; investigators began to typically cover the flush-mount, in order to separate the microphones from the boundary layer of the flow, or recessing the microphone array from the tunnel sidewall, entirely out and away of the flow with or without covering the sensors.

Several approaches have been investigated to reduce the problem, from approaches in introducing acoustic liner inside the wind tunnel; lining a significant part of the wind tunnel and including baffles and splitters to lower the overall background noise levels by attenuating noise propagating in the duct [12], and lining the wind tunnel test section to reduce unwanted reflections [14 and 24]. The Lockheed Martin low speed wind tunnel has implemented this by lining parts of their diffusers downstream of the test section. Other wind tunnels, such as IVK-Stuttgart and NASA-Ames 40x80 have also been modified by introducing liners and splitters [6]. These modifications can however, have a potential impact on overall wind tunnel aerodynamics performance and must be evaluated carefully before any of the solutions are considered.

With many different views on the best way to mount microphone arrays in closed-section wind tunnels [16, 18 and 25] all offering slightly different methods, it can be difficult to easily determine the best set-up for a successful acoustic test in a wind tunnel environment. The systematic examination, development and characteristics of these measurement methods has been lacking in the literature, so their effect on wind tunnel performance has not been well categorised. Recessing microphones would appear to be one solution to the wind tunnel background noise issue, which minimally affects wind tunnel aerodynamic performance. An extreme case of such an installation is the anechoic wind tunnel at Virginia Tech [68]. Previous studies have shown that a recessed array [21 and 15] can give much better results – up to 20 dB lower levels at 5 kHz and 12 dB at 10 kHz as shown in figure 107. This study hoped to recreate these results and further analyse the requirements necessary for such improvements in background noise levels.

In this section a systematic study of the effects of microphone mounting strategies on signal-to-noise ratio, for application in microphone array installations in hard-walled wind tunnels is presented. The effect of microphone recessing, aperture geometry, and cloth type are investigated. Section 4.2.1. describes background theory for the problem being investigated. Section 4.2.2. details the experimental methodology employed for the study. Section 4.2.3. presents results for the different microphone mounting arrangements, followed by concluding remarks in Section 4.2.4.

The recommendation at this time is to design a properly recessed array. Previous tests measurements in the University of Southampton’s 7 by 5 wind tunnel and in the Airbus UK Filton wind tunnel [24 and 62] already indicate a minor improvement by simply recessing individual microphones by a few millimetres.

Comparison with 7 by 5 tests show that noise is of the Order of 10dB lower with out of flow microphones, however, no recessing optimisation for the previous work was ever done.

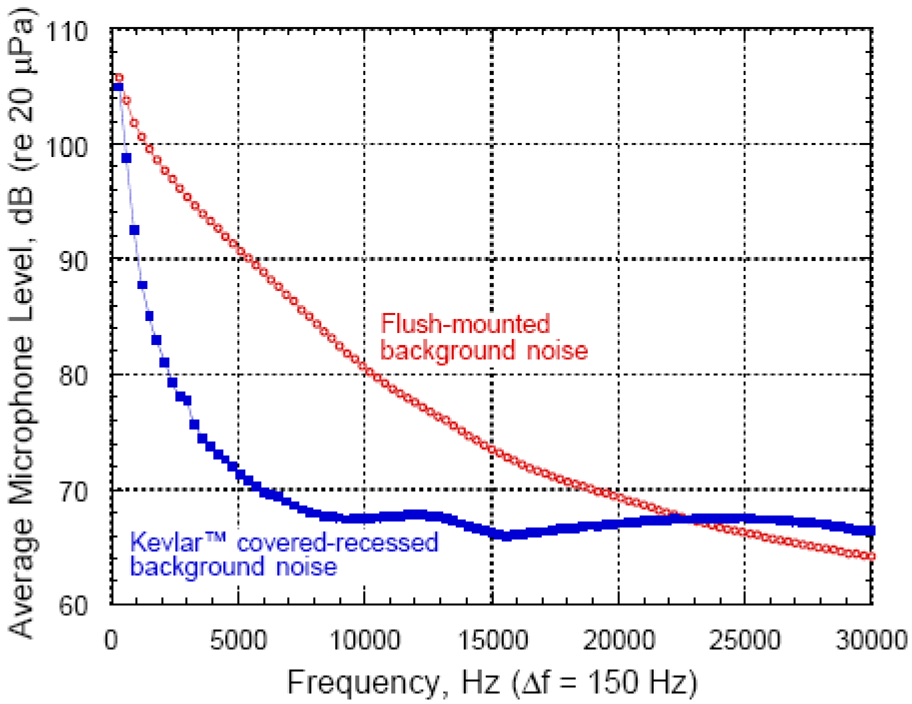


Figure 107 – Background noise levels recorded by a flush mounted microphone and one in a Kevlar covered recessed array. From ref. [13].

Another important feature is a tightly-stretched porous cloth (such as Kevlar) which is flush with the wind tunnel boundary, and behind which the array is installed. The cloth helps to separate the wind tunnel boundary layer from the microphone diaphragms, to reduce boundary layer noise [21 and 19].

Since there is no overall 'best' exact level given for recessing depth in any literature, a number of recessing depth were tested, as well as a number of different recessing techniques, in addition to various materials.

4.2.2. Theory

The current trend for using microphone arrays in hard-walled wind tunnel environments, is to mount the sensors behind a tightly stretched porous cloth, effectively shielding the microphones from the boundary layer and counter-acting the interaction of the microphone with the boundary layer. This interaction causes a substantial level of noise and the phenomenon is often referred to as microphone flow-induced ‘self-noise’.

4.2.2.1. Microphone Self-Noise

The definition of microphone ‘self-noise’ is the noise caused by this interaction and is mainly due to the Brownian movements of the air molecules bombarding the microphone diaphragm, creating an equivalent noise pressure. The small diaphragms utilized for microphone arrays behave as hard surfaces and the air molecules hitting them exchange their energy, producing sound pressure levels relative to the area and the sensitivity of the diaphragms; where sensitivity indicates how well the microphone converts acoustic pressure to output voltage. The self-noise level can be seen as the sound level that creates the same output voltage as the microphone does in the absence of any sound, and represents the lowest point of the microphone's dynamic range, particularly important should it be necessary to detect sounds that are quiet. With a view to improve and enhance this technology several methods have been introduced. It is believed that recessing microphones and covering materials will reduce so called microphone ‘self-noise’.

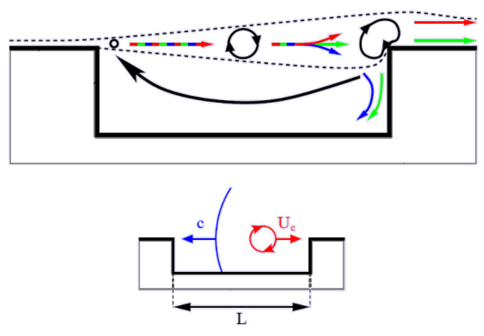
4.2.2.2. Frequency Range

In the case of aeroacoustics the frequency range of interest is dependent on the scale of the wind tunnel, and subsequently, the size of the model being tested. Generally frequencies below 1 kHz are not of interest in wind tunnel acoustics for aircraft models and components, the size of facilities and arrays limit the lower frequencies of interest to between 2 and 4 kHz. Current microphone (and acquisition) technology limits the upper frequency to 48 kHz for ‘Off the Shelf’ low cost microphone channels.

While frequency ranges in the ultrasonic region are of interest for model-scale tests, in this study an upper limit of 48kHz was considered. This is the resolvable limit of many current microphone array installations [58 and 60]. Consequently, in this study a frequency range between 2 and 48 kHz was looked at although it should be realized that future acoustic arrays are likely to be designed for higher frequency ranges.

4.2.2.3. Rossiter and Other Acoustic Modes

Rossiter modes are a type of aeroacoustic feedback loop found in rectangular cavity structures in line with flows passing over them (Fig. 108). These cavity oscillation mechanisms occur generally in high-speed flows producing tones and pressure oscillations induced by the flow over the cavity. A perturbation growing at the leading edge of the cavity and being amplified by instability before interacting with the trailing edge of the cavity produces a feedback into the cavity, which in theory could produce a noise signal interfering with the operation of the microphone. which may induce an acoustic feedback loop and the generation of cavity tones.



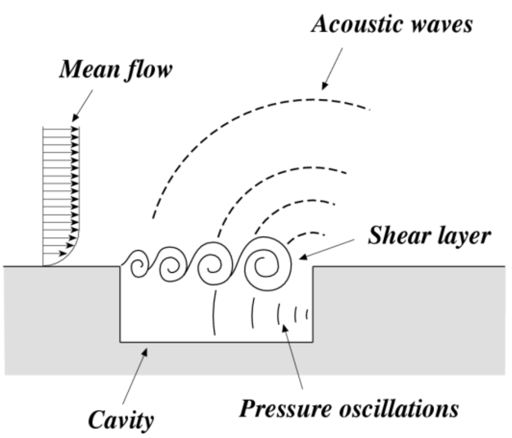


Figure 108 -Diagram showing Rossiter Acoustic Model Figure from Reference [12] (left image) and Cavity acoustic feedback mechanism schematic (right image)

Although some of the microphone apertures employed in this study are not rectangular and some of the aperture geometry can be considered to be quite shallow (where the ratio of the aperture's length to depth is approximately 5 or above) there remains a possibility of Rossiter Modes being present.

The following formulae can be used to determine the frequency at which a Rossiter mode for a give aperture (cavity) length and depth can be expected. Equation 2 is used in determining the frequencies at which the Rossiter mode can be expected.

Rossiter Mode theory:

$$St = \frac{fL}{U} = \frac{n}{\frac{u}{u_c} + M} \qquad (1)$$

Rossiter's Model (1964):

$$f_n = \frac{U_\infty}{L} \frac{n-\gamma}{M_\infty + \frac{1}{k}}$$
$$\gamma = \gamma(L/D) \qquad k = 0.57 \qquad (2)$$

Flow Speed (m/s)	45 Degree	60 Degree	Recessed (by d)
10	~426	~540	~852
20	~852	~1081	~1704
30	~1704	~1622	~2556

Figure 109 - Table showing Rossiter Mode Frequencies (Hz)

Flow Speed(m/s)	Blade Pass Frequency(Hz)
10	~ 6
20	~ 11
30	~ 24

Figure 110 – Table of Blade Pass Frequencies for the Southampton 3 by 2 wind tunnel.

For this paper a comparison of the various apertures in the test rig will be made (see section 4.2.2), it is possible that Rossiter modes will be present. In order to identify any Rossiter effects, Figure 109 lists the theoretical possible Rossiter modes made for each of the different aperture types tested.

Another potential acoustic source in the wind tunnel is the fan. Often in the low frequency end of the acoustic range of interest the signal is dominated by a series of strong tones. Such tones are a common feature of fans, and occur at a frequency called the blade pass frequency (BPF), which is a function of fan revolutions per minute (RPM) and the number of blades, with corresponding harmonics. Figure 110 lists the fundamental frequencies corresponding to all the wind tunnel speeds at which tests were performed. In the case of this analysis, it is likely that it would only be the harmonics, if any, of the BPF that would be observed in the results.

4.2.3. Methodology

The following section describes how the test rig was designed and assembled.

The tests were performed in the University of Southampton's 0.9m by 0.6m (3' by 2') wind tunnel facility. For this test four individual microphone apertures were used, as shown in Figures 108a, and 112 to 115. Each was selected as it had either been used in previous wind tunnel tests by various facilities or in the case of the 60 degree countersunk, as a comparison to determine the effect of geometrical variation. The four aperture types were:

- Flush mounted, microphone set flush with surface;
- 45 degree countersunk aperture, microphone set inside a 45° circular hole;
- 60 degree countersunk aperture, microphone set inside a 60° circular hole, it should be noted that the '45 and 60 degree' refer to the angle of the drill bit used to form the countersunk holes, and finally;
- recessed microphones, where microphones are set to a depth (normalized by the diameter of the microphone) inside its individual aperture.

Figure 111 (also figures 112 to 115) show the side cut view of the four types of microphone aperture; recess depths are normalized to microphone diameter, *d*.

The recessing test has other considerations, which include the covering required for the recessed microphones, to keep the flow away from the microphones and the recess gap. Previous studies performed by NASA and Audi [13 and 18], suggest Kevlar as the optimum material for covering microphones. Kevlar materials selected for use in these tests were a 50 gm/m^2 and a 95 gm/m^2 type material. Installation testing of 30 gm/m^2 material showed it to be too weak and thin to be useful.

For this test the microphones to be tested will be:

- Cavity recessed microphone
- Small section recessed

For this test four individual microphone apertures were designed. Each formed differently as described in the testing procedure section on the following pages. The recessing test has other considerations, which include the covering required for the recessed microphones, to keep the flow away from the microphones and the recess gap.

The studies performed by NASA and Audi, [13 and 18] suggest Kevlar as the optimum material and as such a source of appropriate Kevlar material is required. Thus, Kevlar materials used in the tests were a 50 gm/M^2 and a 95 gm/M^2 type material. Initial research revealed three types of Kevlar available for dispatch in cloth form from online sources: [65 and 66] 30 gm/M^2, 50 gm/M^2 and 95 gm/M^2. After several attempts at mounting the 30 gm/M^2 Kevlar it was determined that it would be too weak and thin to provide effective an adequate barrier against the flow in the wind tunnel.

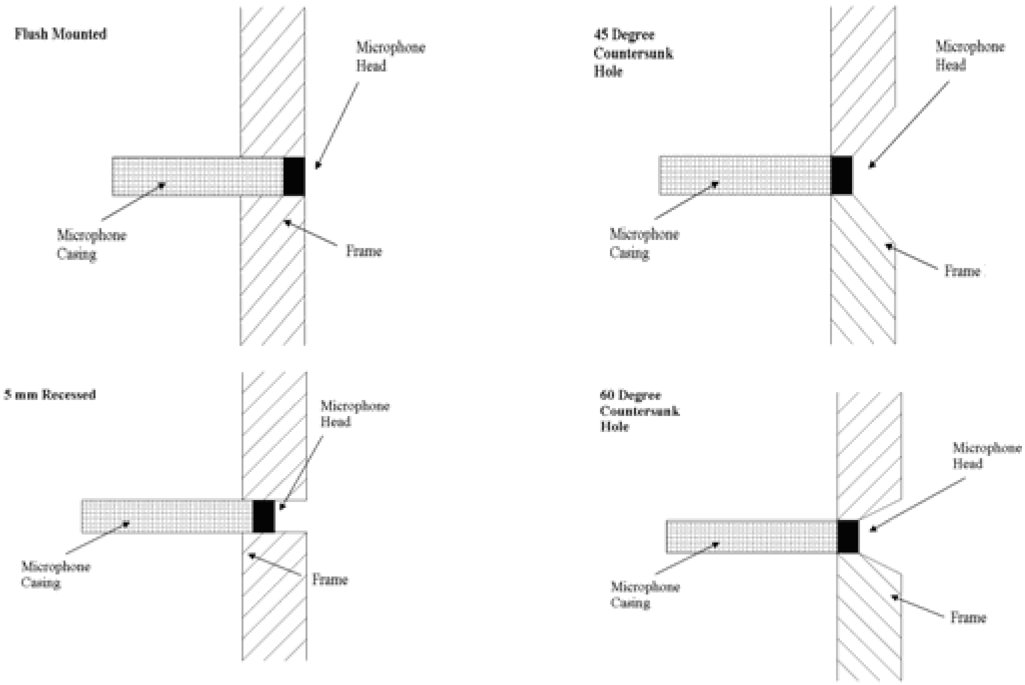


Figure 111 -Diagram showing microphone aperture geometries.

The Kevlar cloths at 50 gm/M^2 and 95 gm/M^2 were mounted successfully and were considered to be the most useful given the forces involved in the wind tunnel. An initial source of 300 gm/M^2 Kevlar from the University of Southampton’s Transport Lab was ascertained to be too thick for effective acoustic tests.

4.2.3.1. Set-Up and Calibration Process

A wooden box was constructed, using 20 mm thick MDF, to house the central recessed plate and when appropriate acoustic foam inside the recess gap. It was necessary to construct a modular structure that could attach to the side of the 3 by 2 wind tunnel. A frame to support and connect the box to the wind tunnel was then constructed. The frame had to be constructed in such a way as to allow a Kevlar cloth to be stretched over it, and modular so the 5 different sets of test assembly could be interchanged easily. Two distinct microphone sections were identified and constructed:

- The first section would be a free plate containing the microphone array that could be recessed behind the Kevlar material at different recessed levels; this would allow a small section to be recessed.

- The second section would be flush mounted, and without Kevlar, Near this set was also included a control microphone, this second set had individual cavity recessed microphones.

The central recessed box sits behind the Kevlar cloth as indicated in Figure 112. The entire structure replaced a plate in the test section wall of the 3 by 2 wind tunnel.

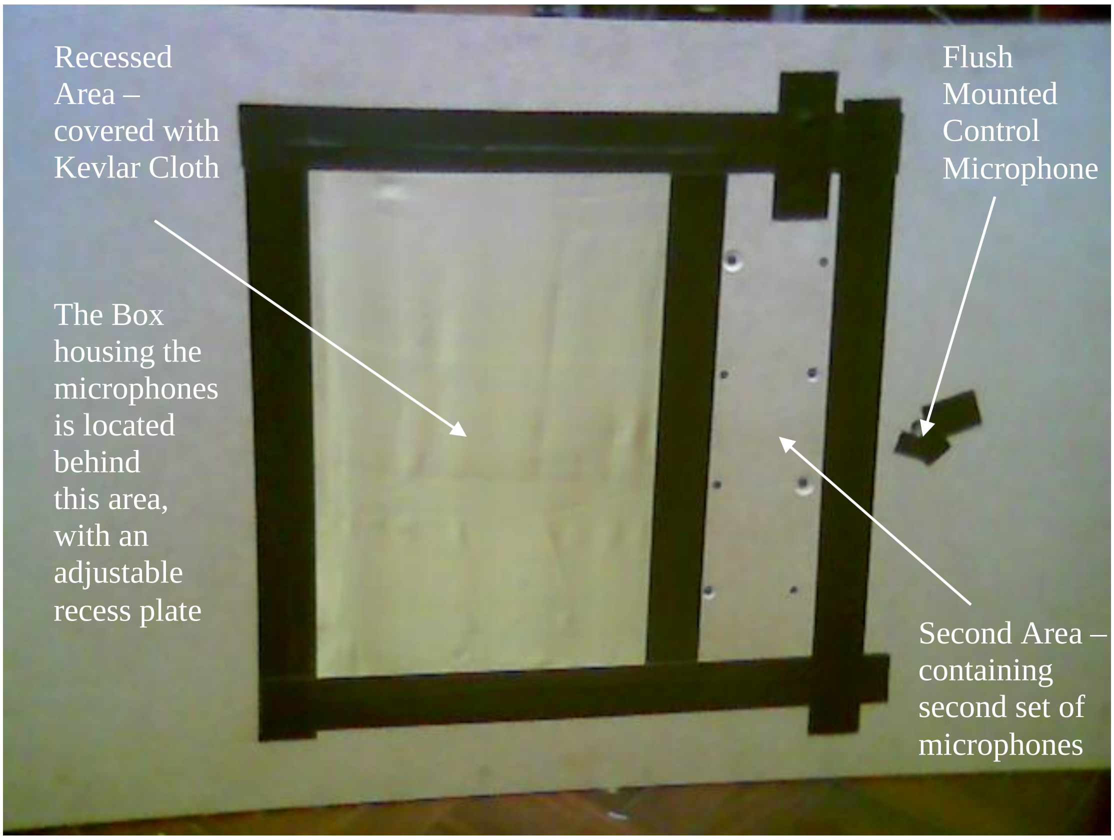


Figure 112 – Descriptive diagram of test rig, identifying microphone sections.

Other concerns include the need to determine fixing mechanism for the material cover. Initial design concepts looked at the possibility of using the same technology that is employed in holding embroidery material or that employed in holding drum skins. A simple wood frame was finally constructed and double sided tape was used to secure the material as this proved more than sufficient in previous tests; although it was important to have a suitably large surface area to attach the frame to the material. The box positioned behind the Kevlar was pushed and locked into place and ensured the Kevlar was as taut as possible. Black tape was employed simply to fill the crack joins in the construction and to smooth off edges in order to avoid creating any additional noise sources.

4.2.3.2. Testing Procedure

In order to analyse the maximum number of configuration in the limited time given the following test order was formed. A large number of microphone mounting configurations were carried out. Tests would look at four different recessing gaps from 0 to 20 mm (0 to 4d) and at different recessing gaps up to 10d (10 times the diameter of the microphone) using two Kevlar weights and silk cloth to cover the microphones. Tests were carried out over a speed range of 0 to 30 m/s, equivalent to a Reynolds number per meter of 2.06×10^6 at the highest speed. The test order can be found in the test matrix in figures 113 and 114 on the following page.

In addition microphones were also separated from the flow using three distinct cloths; two types of Kevlar material were utilized on the recessed plate section. Silk Taffeta cloth was also used as a third material as this was a material used in several University of Southampton acoustic wind tunnel measurements. The Silk material was used in a final set of runs to cover the flush mounted set of microphones. Five sets of test rig were used depending on which Kevlar, Silk and Foam would be in use, for convenience of rigging and to minimise the time and level of configuration changes the following order was used.

1. Kevlar Type 1 – 50 gm/M^2 Kevlar cloth

2. Kevlar Type 2 – 95 gm/M^2 Kevlar cloth

3. Kevlar 1 and Foam – 50 gm/M^2 Kevlar covering recessed plate and 1 inch thick acoustic foam in recess gap

4. Kevlar 2 and Foam – 95 gm/M^2 Kevlar covering recessed plate and 1 inch thick acoustic foam in recess gap

5. Silk Material – Taffeta Silk cloth covering second area set of microphones and the control microphone

A possible 6th set with the silk material and foam was not tested due to the time limitations and constraints. Additionally the silk material was only tested on the non-recessed plate microphones as this was only required to compare against previous tests done in wind tunnels where the same material had been used on flush mounted microphones.

Kevlar / Silk and Foam	Recess (mm)	WT Speed (m/s)	Noise Source
Kevlar Type 1	0,5,10,20	0,10,20,30	Tones and white noise
Kevlar Type 2	0,5,10,20	0,10,20,30	Tones and white noise
Kevlar 1 and Foam	0,5,10,20	0,10,20,30	Tones and white noise
Kevlar 2 and Foam	0,5,10,20	0,10,20,30	Tones and white noise
Silk Material	0	0,10,20,30	Tones and white noise

Figure 113 – Table of the test matrix identifying all the combinations tested in the experiment.

Cover Material	Foam	Recess (d)
Kevlar Type 1	No	0,1,2,4
Kevlar Type 2	No	0,1,2,4,10
Kevlar 1	Yes	0,1,2,4
Kevlar 2	Yes	0,1,2,4
Silk Material	No	0

Figure 114 -Table of the simplified test matrix showing cover material, foam and recess in d.

Selection of Tonal Frequencies used to test microphones
2.5 kHz
4 kHz
8 kHz
16 kHz
25 kHz
31.5 kHz
44 kHz
47 kHz

Figure 115 -Table listing the tonal frequencies used for the test signal.

The signal for testing the microphones, an acoustic source, comprising a speaker, was driven at a number of frequencies to provide a strong signal for the microphones (see figure 115), this selection was made to cover some low frequency tones as well as analyse the higher frequency tones. The lowest frequency was determined by the limitation of the tweeter speaker and the high end tone was set by the 48 kHz limit set by the anti-aliased sample frequency of the acquisition hardware as described on the following pages. Initially, for each configuration, a background noise measurement to account for general wind tunnel noise was also made. The results presented here show the 25 kHz tone case.

The tests were carried out in the University of Southampton 3' by 2' wind tunnel closed test section. The tunnel is rated up to 30 m/s and operates at room temperature.

A set of 16 electret microphones, Panasonic WM-60A / WM-61A were used. These microphones are widely used for aeroacoustic testing in wind tunnels and engine test facilities, and are shown to be effective up to 48 kHz [58, 60]. Furthermore, these microphones have to be coupled with custom-build preamplifiers, which provide a constant voltage to power the microphones and amplify the weak microphone signals to line levels; these preamplifiers form part of the signal path from the microphones and as such have a need to also be validated when used for the higher end frequencies [24, 62 and 9]. This calibration has been performed in a previous project at Southampton.

The electret microphones were installed in the specially designed rig as seen in figure 112 and in addition a 17th microphone, a Type 2615 Bruel & Kjaer (B&K) Microphone was installed flush with the tunnel wall to provide a control microphone for the experiment, its location can also be seen in figure 112.

Regarding microphone positioning, it was unfortunate that due to physical constraints it was necessary that the microphones were placed in the wake of each other. This was taken into account by repeating the type the microphone aperture at least once in different positions, in different wakes to negate and minimise any possible error.

Due to small size of the wind tunnel test section the microphone holes were still positioned fairly close to one another, with only a 140 mm gap between some of the microphone aperture edges. This is representative of the dense packing found in most microphone array installations. In a densely packed microphone array the likely situation is that microphones will be packed even closer together, so in that respect the test reflects conditions the microphones are likely to be under in a real microphone array. Nonetheless efforts were made to substantiate the results by repeating the different microphone apertures at different points in the test rig.

Figure 117 and the related table in figure 118 on the next page show how the different microphone holes were positioned during construction so that no two of the same hole was directly in line with the same type, either upstream or downstream.

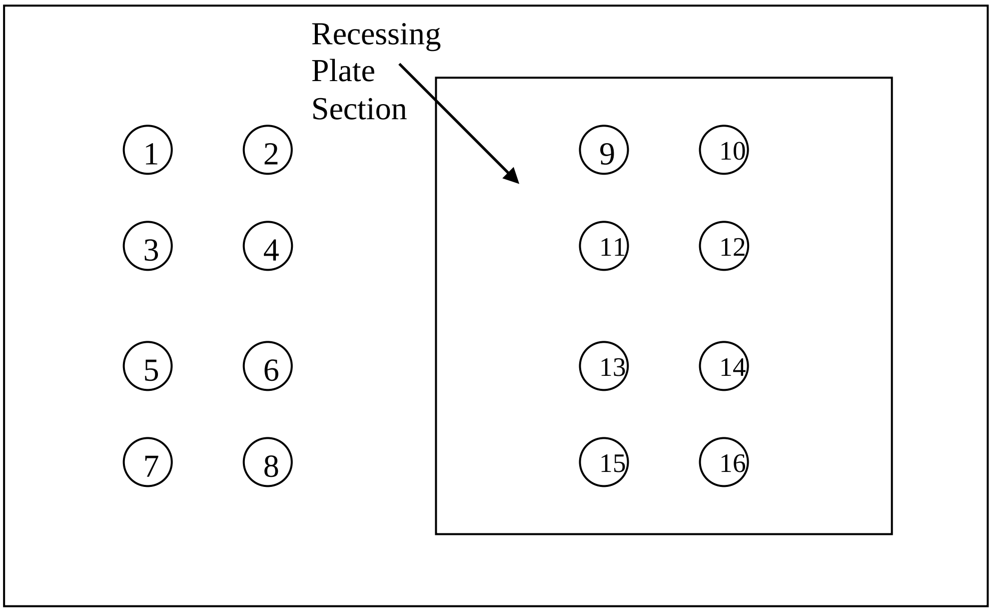


Figure 116 - Diagram showing microphone positioning to account for wake effects, viewed from outside the wind tunnel.

Microphone Number	Type of Microphone Hole
1	Flush mounted
2	45 degree countersink hole
3	60 degree countersink hole
4	5mm Recessed
5	45 degree countersink hole 2
6	Flush mounted 2
7	5 mm Recessed 2
8	60 degree countersink hole 2
9	Flush mounted in Recessed plate
10	45 degree countersink hole in Recessed plate
11	60 degree countersink hole in Recessed plate
12	5mm Recessed in Recessed plate
13	45 degree countersink hole 2 in Recessed plate
14	Flush mounted 2 in Recessed plate
15	5 mm Recessed 2 in Recessed plate
16	60 degree countersink hole 2 in Recessed plate

Figure 117 -Table listing each microphone hole location.

The four microphone aperture types as described in figure 117 are explained below. Each was selected as it had either been used in previous wind tunnel tests by various facilities or in the case of the 60 degree countersunk, as a comparison aperture to determine the effect apertures have on the results.

- **Flush mounted –** Microphone set flush with surface
- **45 degree countersunk aperture –** Microphone set inside a 45° circular hole
- **60 degree countersunk aperture –** Microphone set inside a 60° circular hole
- **5 mm Recessed –** Microphone set 5 mm inside its individual aperture

Figures 118 to 121 below shows the side cut view of the 4 types of microphone aperture, the microphone diameter was approximately 5 – 6 mm in diameter (with variances in each microphone).

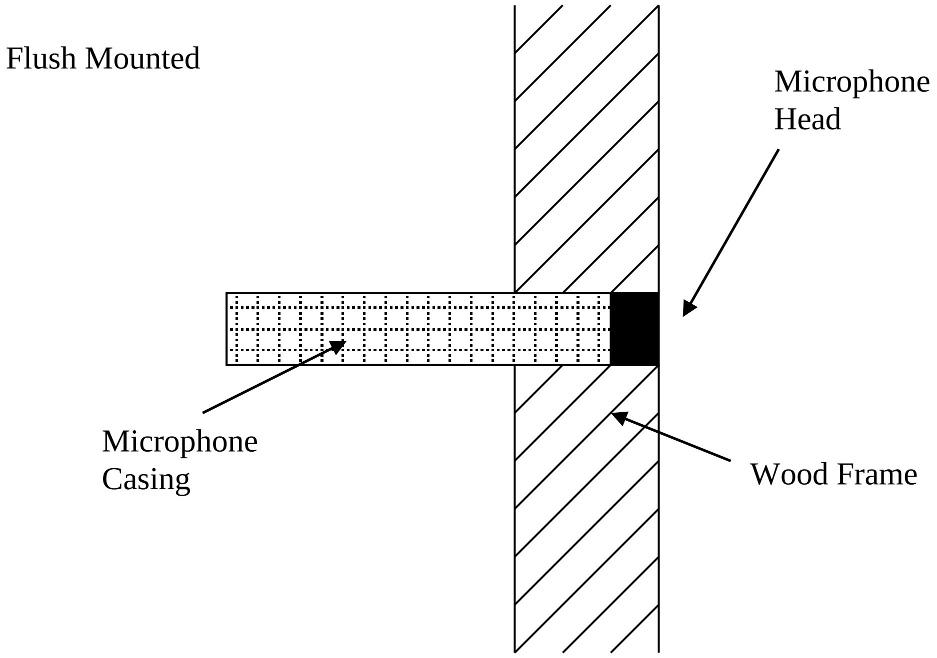


Figure 118 – Diagram of the Flush mounted microphone aperture.

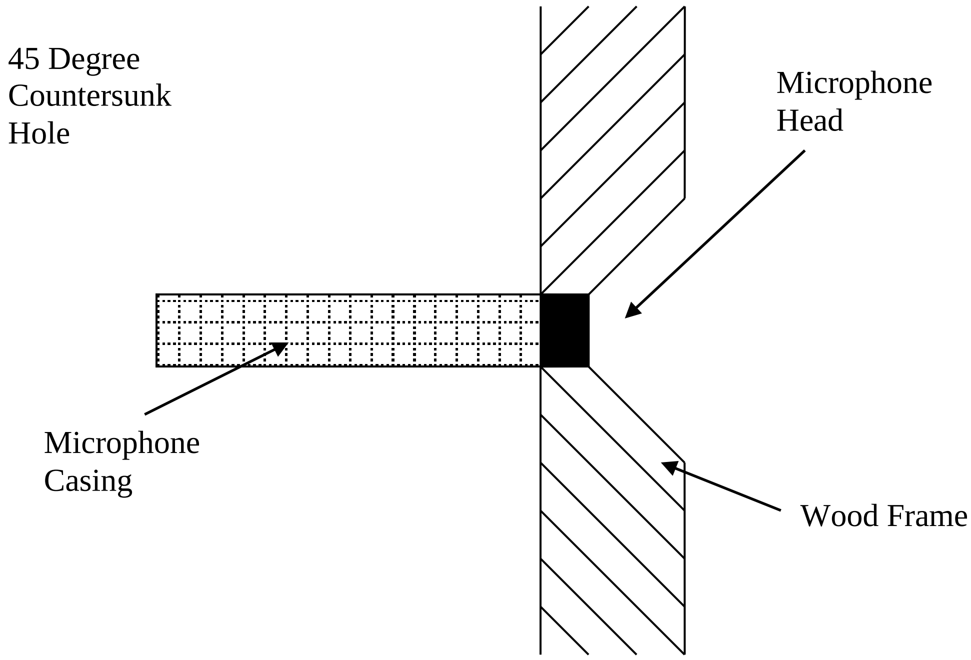


Figure 119 – Diagram of the 45 degree countersunk microphone aperture.

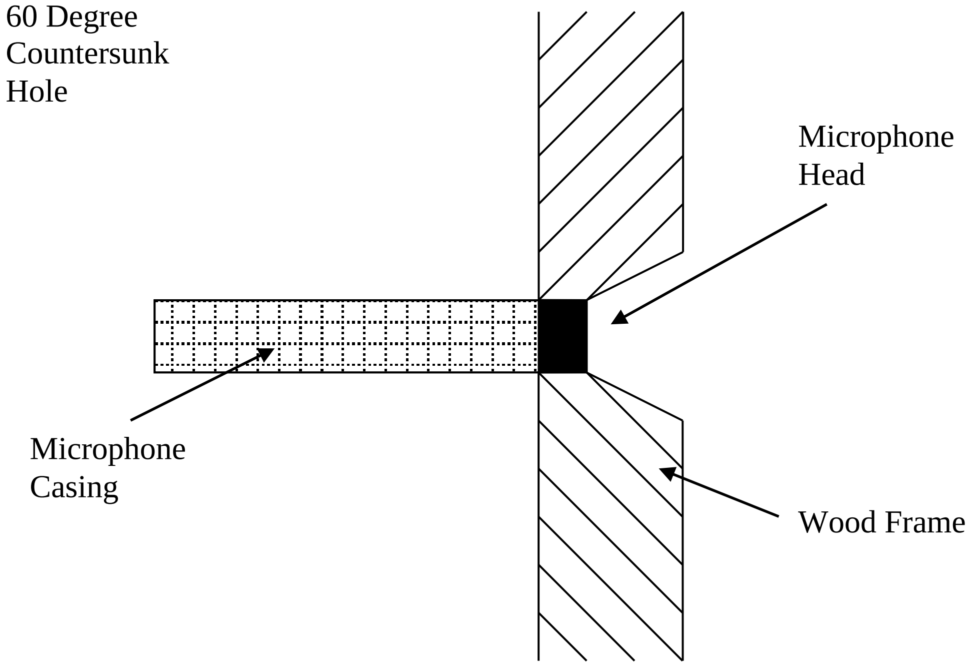


Figure 120 – Diagram of the 60 degree countersunk microphone aperture.

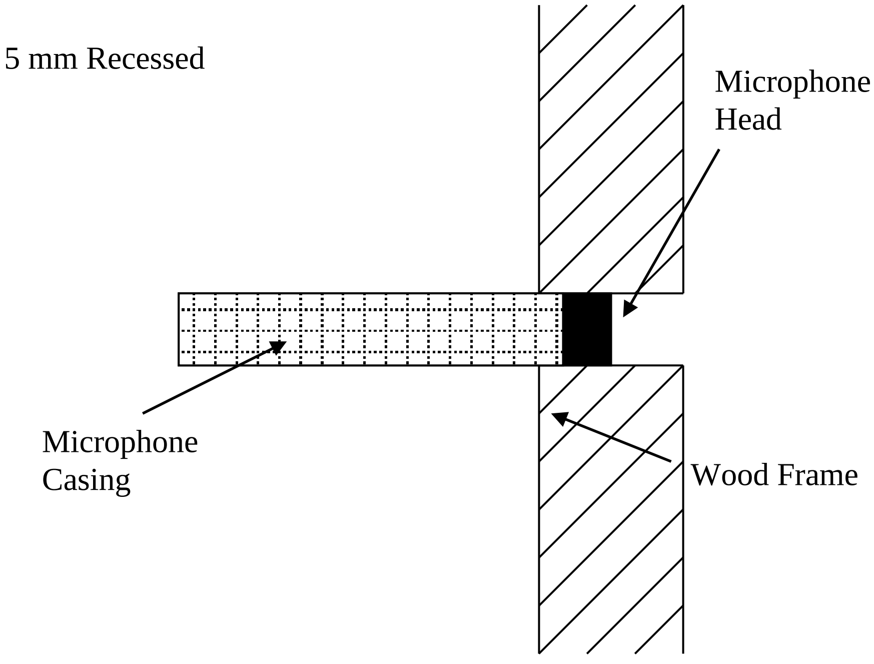


Figure 121 – Diagram of the 5mm Recessed mounted microphone aperture.

4.2.3.3. Equipment

The control microphone was a B&K Microphone Type 2615, (S/N – 175426) with a ½” Microphone Head 4163, (S/N – 580478) and powered by a B&K Amp Type 2609, (S/N – 1084149). This control microphone was mounted flush with the wind tunnel wall close to the flush mounted microphones as seen in figure 112 and in the same way as seen in figure 118.

All the electrect microphones were also calibrated using a pistonphone. The pistonphone produces a tone set at 1 kHz at 94 dB (1 Pa) to which every microphone is tested against. Additional calibration involving the use of a reference microphone (in this case a GRAS) was also done to calibrate each individual electret.

As well as the Kevlar material used to cover the recessed plate, a silk material was used for the final set of tests in covering the microphones. This silk material was a Caress Eton taffeta lining, 100% anti-static polyester and it was supplied by John Lewis UK.

For this experiment a sound source was employed in the form of a simple tweeter speaker (dome type) with a rated range of 2 kHz to 80 kHz. The tweeter was assumed to be a monopole point source and was driven by a signal generator: Programmable Function Generator HM 8130 (Hammeg serial #130015P04692), ISVR – SV6154 and B&K Signal Generator, Sine-Random Generator Type 1024 (S/N 207988), and powered by a SoundLab power amplifier.

Data acquisition was performed using Labview 8 based software, used to drive the National Instruments (NI) acquisition chassis. The acquisition unit, an NI-PXI 1042 Q chassis held a PXI-4462 card, and a data transfer card (for connecting to the controller). The controller used was an NI 8351. The controller was used for interfacing with the acquisition unit using software written by Benjamin Fenech, of the University of Southampton in the aforementioned Labview 8 software and for processing the results using code written in Matlab (version R2006b). The data acquired using the NI software was saved in binary format and processed against calibration values that were also acquired using the NI software and hardware. The processing was done using Matlab and consisted of a simple FFT (Fast Fourier Transform) of the data and then plotting the power spectral density against frequency. The Data Acquisition Unit, with the acquisition software was operated with a sampling frequency of 96 kHz (allowing anti-aliased to 48 kHz) and using a block size of 16384 and 200 blocks, yielding 3276800 averages acquired over an approximately 30 second period. Matlab was then used to produce the power spectral density spectra presented here.

4.2.3.4. Errors and Uncertainties

In this section we describe possible sources of errors and inaccuracies, and discuss their relevance to the results presented here. Variations due to the FFT processing, of up to 1 dB, are expected. This is deemed acceptable for the purposes of this study, as improvements of 2 dB and above are clearly visible.

Due to the slight variance in location of the microphone to the signal source, slight discrepancies might be expected, however since signals were acquired simultaneously, using the same acquisition cards and software, any discrepancies are minimised. Minor inconsistencies with the control microphone values for same setting runs were observed, which were down to experimental variation. For this reason comparison between plots must be made carefully and the control microphone results used as a datum. Comparison between microphone aperture types are made in all the figures and the plots used for comparison only show the first of each type (both recessed and non-recessed), repeated apertures were installed to provide a check for any possible random errors, or to identify systematic errors, such as wake effects. From the analysis of the entire results matrix show no significant variation. It should be noted, however, that a microphone failure partway into the test did mean that the second recessed microphone on the recessed plate did not have a backup and so the assumption is made that no errors developed during the following tests.

4.2.4. Results and Discussion

4.2.4.1. Baseline Results

Figure. 122 shows the comparison of the flush-mounted electrets microphone with the tunnel motor turned off (0 m/s) and running at the test speed of 30m/s with a 25kHz source. The spike at 10kHz and hump at 19-23kHz is observed to be independent of tunnel speed, and hence associated with the wind tunnel motor system, and not deemed to be aeroacoustic in nature. The levels recorded in this test serve as an equipment check and an indicator of what can be expected for the other results. All the subsequent plots are at the 30 m/s speed setting.

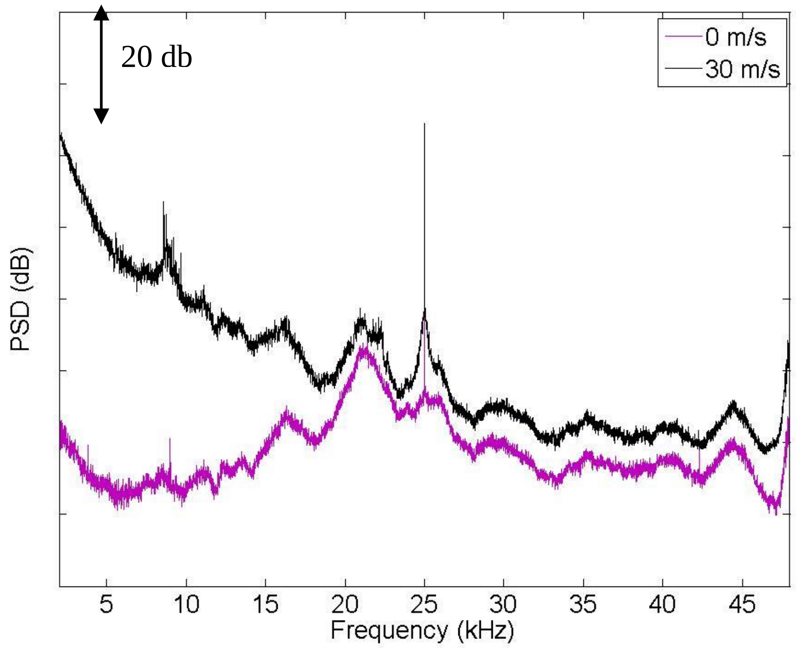


Figure 122 - Comparison of Microphone noise levels at wind tunnel speeds of 0 m/s and 30 m/s.

4.2.4.2. Aperture Results Plots

Figure 123 shows a comparison of the 45° and 60° countersunk apertures with the flush-mounted and 5mm recessed single microphones. It can be seen that the differing apertures do have a measurable effect on the signal and noise received by the microphones. Specifically in this case the 45-degree aperture is shown to be significantly noisier than any of the other aperture types. This is also apparent across the speed range (not shown). Such variation is not seen in the recessed array results shown in Figure 124; the recessing brings all the apertures signals in line with each other. This indicates that perhaps the 45° aperture has distinctive unsteady flow characteristics that increase the noise level.

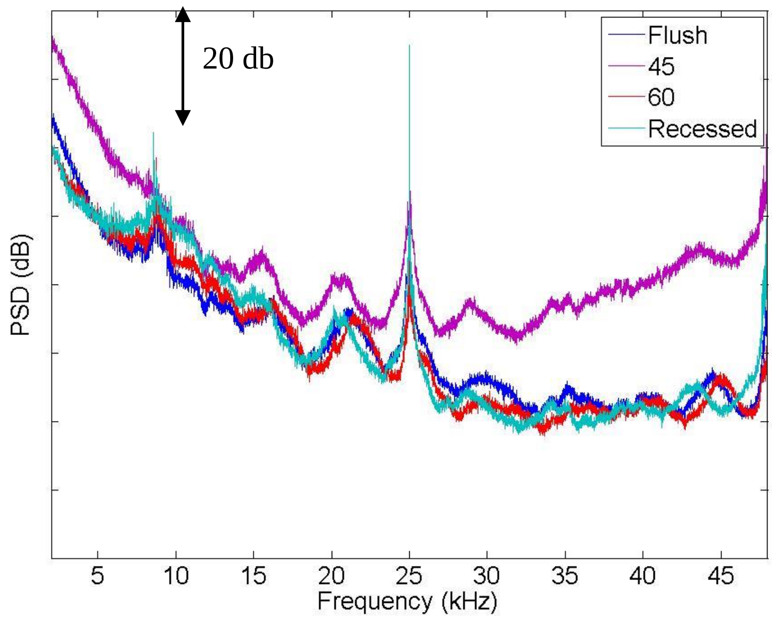


Figure 123 - Aperture comparison of noise and signal.

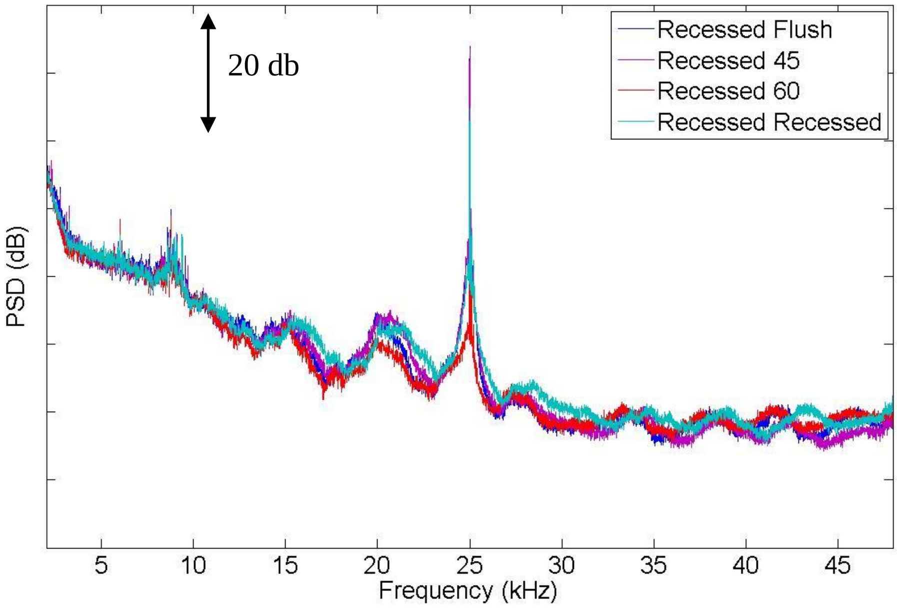


Figure 124 - Aperture comparison of noise and signal, recessed plate at 2d.

4.2.4.3. Recessing Results Plots

Here we present results that show the effect of recessing. Figure 125 shows the PSD observed as the recess level is increased. For the 10d recess, noise reduction of between 10 dB and 20 dB can be observed at frequencies below 20 kHz and 5 dB, to 10 dB above 20 kHz, whereas the 4d recess has a noise reduction of up to a maximum of 3 dB. Smaller recess levels of d and 2d showed only slight improvement in noise, although the trend indicates improvements of 1 dB to 2 dB over the frequency range.

Comparisons between different apertures, at the different recesses, are shown in Figure 126. 15 dB to 20 dB noise reductions can be observed for the 10d recessed 45-degree aperture noise level. These improvements are again seen across all the aperture types and at the various wind tunnel speeds.

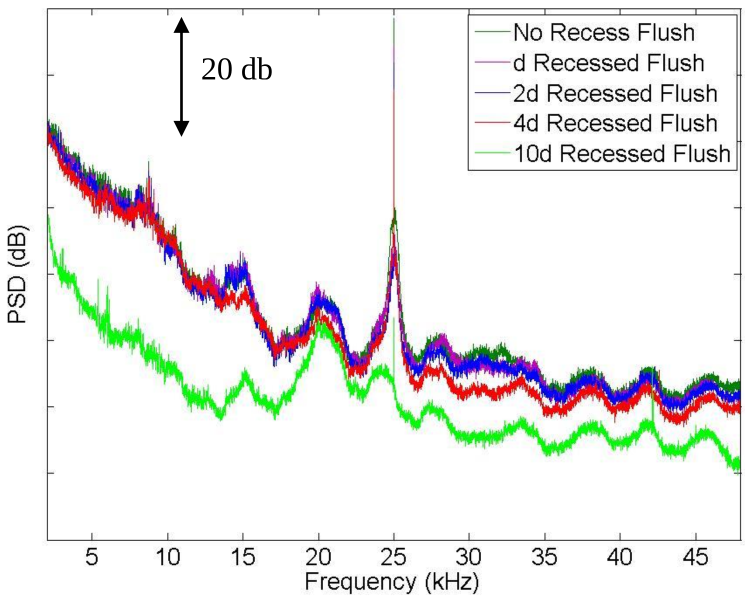


Figure 125 - Comparison of noise at various recessing levels between no recessing to a 10d recess.

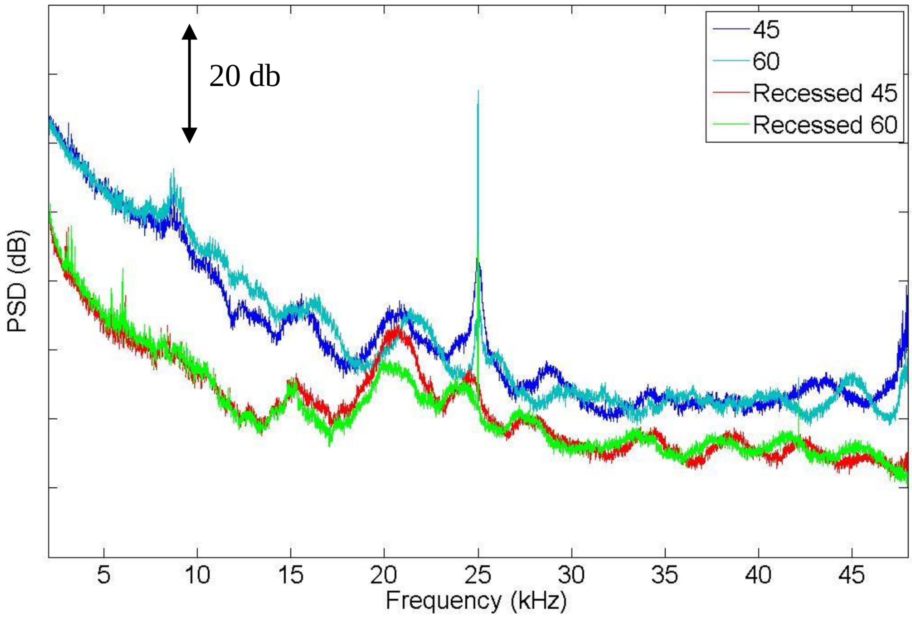


Figure 126 - Comparison of noise of the 45-degree aperture between zero recessed and 10d recessed.

4.2.4.4. Material Results Plots

Comparisons of the effect of different covering materials show significant differences. Introduction of the foam in the recess has a dramatic effect on the signal to noise ratio, noise was considerably decreased and whilst some signal was lost, the signal strength remained relatively high. Figure 128 shows the comparison of the two aperture types on the recessed array behind Kevlar type 1 (50 gm/m^2), with and without acoustic foam lining, typical noise reductions of 5 dB to 10 dB were observed. Significant differences between the two types of Kevlar were observed, as shown in Figure 127 at various recess depths. Figure 129 shows that application of the foam to both Kevlar type 1 and Kevlar type 2 (95 gm/m^2) brings both noise levels recorded to the same value. This is an indication that the foam has more effect than the type of Kevlar used and it can be interpreted that foam lining is effectively the primary factor in reducing background noise levels.

The effect of silk covering on the microphones, especially at lower frequencies, shows a clear improvement over background noise at frequencies below 20 kHz (Fig. 130). However, compared with the Type 2 Kevlar over the Silk, the silk noise levels are higher. This would seem to indicate that the Kevlar is superior to the silk for acoustic testing. Further research in to comparing Kevlar to the Silk will be required to validate these results.

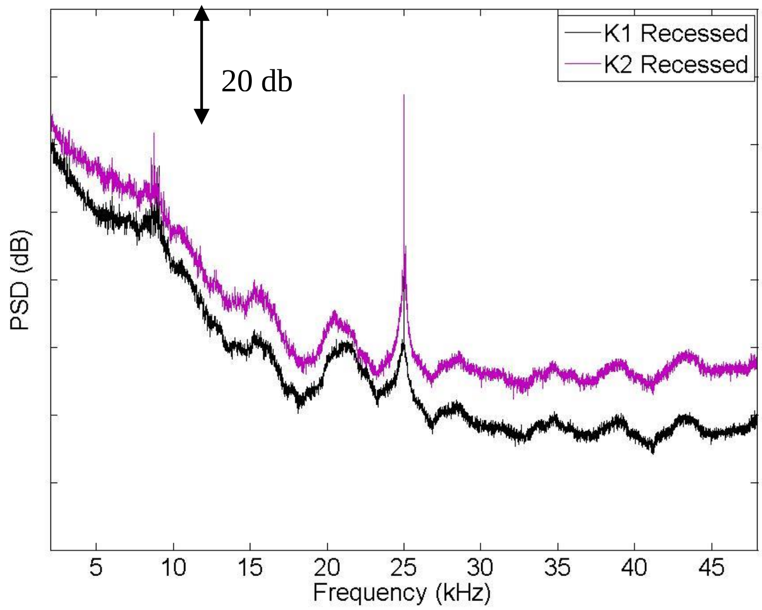


Figure 127 - Material comparison of noise and signal, Kevlar type 1 versus Kevlar type 2.

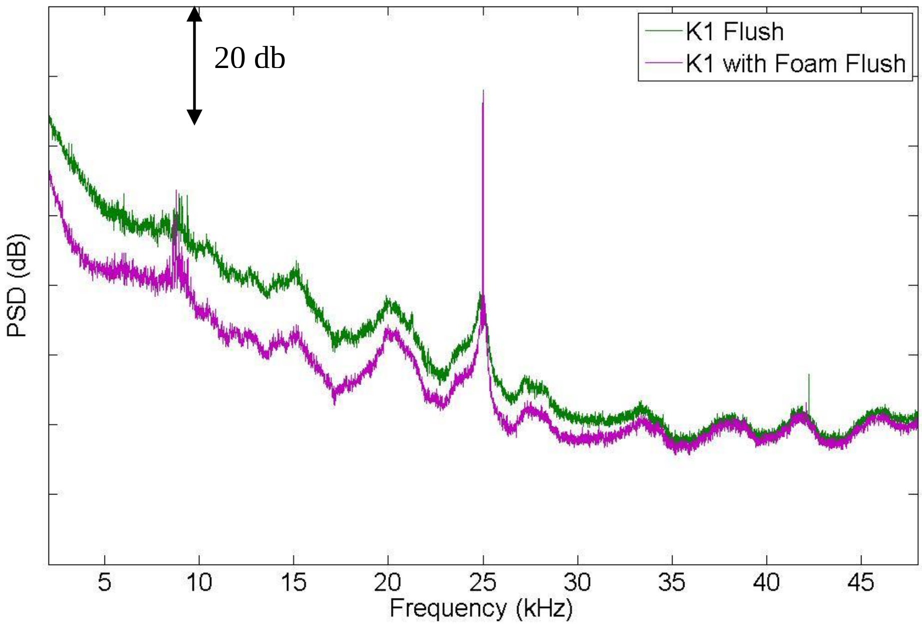


Figure 128 - Material comparison of noise and signal, Kevlar type 1 versus foam on the flush mounted microphone.

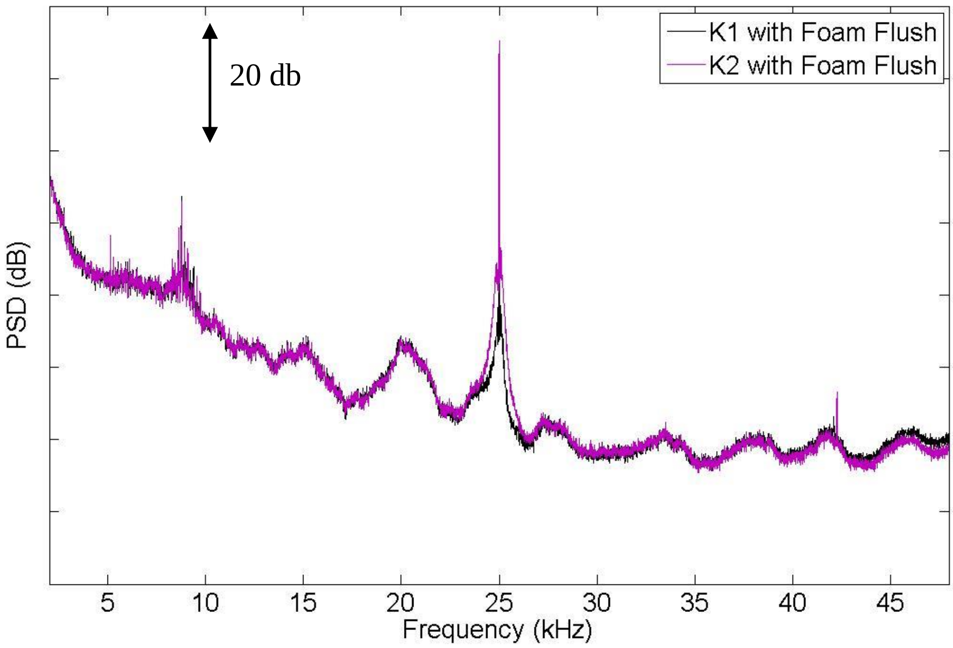


Figure 129 - Material comparison of noise and signal, Kevlar types 1 and 2 both with foam lining.

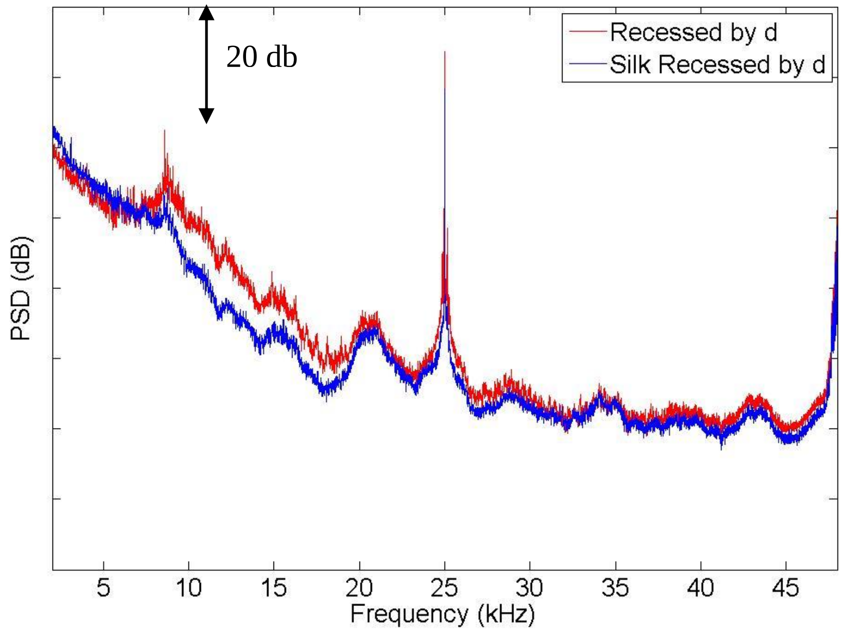


Figure 130 - Comparison of noise and signal between silk covered microphones and uncovered microphones.

4.2.5. Conclusions

This paper has presented results of a systematic study of the effects of microphone aperture geometry, recessing and material coverings, as can be applied to microphone array installations in closed-section, hard-walled wind tunnels.

Different levels of recessing were tested and it was determined that a recess of depth *d* provided a measurable reduction in background noise level, which was similar to those with *2d* or *4d* recess depth. Increasing the recess depth to *10d* resulted in significantly greater noise reduction, from 10 to 20dB across the frequency range.

Various microphone apertures were also tested and from the results it can be seen that the 45-degree aperture seemed to fair the worst as it consistently gave the highest signal and the highest noise. Although not on all plots a significant number of results from the 45-degree aperture mounted on the flush section of the test rig produced high levels of noise registering noise levels up to 20 dB or more relative to the other signals, this occurred in most runs at 30 m/s and it would appear some form of mode is being activated at this point.

It was shown quite effectively the effect that boundary layer interaction has on noise levels as none of the recessed (behind the Kevlar or cloth) results showed the levels of noise indicated with the unshielded microphones and although the levels shows that recessing has only a relatively small effect, shielding the microphones from the flow is definitely advantageous in reducing noise and can be achieved with a minimal loss to the signal. In addition an analysis into the effect of different materials were quantified, with the Kevlar proving to be more effective than silk. There is a significant difference between the different Kevlar used; the lighter material being better possibly due to the weave of the second Kevlar being insufficient to shield the microphones effectively or perhaps the thickness of the second Kevlar causing some form of acoustic resonance. In either case, it is shown that the foam (in the recess gap) appears to reduce noise, eliminate the poor effect of the second type of Kevlar and substantially improve the signal to noise ratio, far more than was expected.

In summary, the preliminary analysis of the results leads us to the following conclusions:

- Recessing the microphones had a small, but nonetheless positive effect on the signal to noise ratio. It is shown that all the types of recessing have generally lower noise, than the non-recessed microphones. Levels up to 10 db lower were recorded, whilst the peak levels remain the same indicating no loss of the tonal signal. Results gained from d depth are equivalent to 2d or 4d recessing. Recessing at 10d can cause noise levels to fall up to 20 dB at frequencies below 20 kHz.
- Introducing a covering had a limited, but measurable effect, reducing noise in the order of 2dB. The silk material reduces noise by 2 dB to 5 dB across entire frequency range depending on the aperture. The Kevlar material had the same general effect as the silk with variances between type 1 and type 2 Kevlar showing that the type 2 Kevlar produced poorer signal to noise results.
- The addition of foam to the recessed gap had a significant effect on the signal to noise ratio, in the order of 5 dB to 10 dB improvements on the noise baseline.
- The 45-degree type aperture acted to increase background noise, although higher signal levels were also observed.
- An analysis of the results showed no definitive evidence of Rossiter Modes in the 45°, 60° and rectangular apertures, although other acoustic features were observed. Further study into the 45° aperture is warranted.
- A combination of recessing, optimised cloth type and acoustic foam lining can create significant, O(20dB), reductions in background noise level.

A significant new result from this work is the positive effect that acoustic foam lining has on the recessed array, further improving the SNR. This study demonstrates that detailed design and analysis of microphone array installation effects can result in performance improvements that would otherwise require significant increases in channel count, or significant physical modifications to a closed-section wind tunnel system. While aerodynamic closed-section wind tunnels are now able to measure aeroacoustic noise sources on current generation airliners, the ability to evaluate next-generation quiet designs is limited by the achievable SNR. The results from this study provide a way of improving this SNR in a practicable fashion to extend the closed-section wind tunnel aeroacoustics measurement capability in the near-term.

4.2.6. Limitations

Due to the small nature of the wind tunnel test section the microphones were arranged such that the microphones were potentially under the influence of microphone apertures further upstream. Careful analysis allows any wake effects to be identified and the test rig was arranged such that the apertures repeated as necessary to account for this.

The working pressure difference in the wind tunnel was unexpectedly high and as a consequence the framework for the test rig was an under engineered structure, which meant that at the highest speeds of 30 m/s the low pressure in the wind tunnel caused deformities in the wood frame to develop; this in turn caused bowing of the Kevlar material into the test section. Although the material remained taut throughout (the pressure difference keeping the material very tight) the bowing did mean that the recessing levels were somewhat altered. The majority of the bowing occurred towards the end of the 30 m/s test runs, and only during the recess tests, with the first Kevlar cloth. Increasing the recess also increased the bowing. As a consequence the high recessing tests at 30 m/s, for the first Kevlar test all have an inherent error. The bowing of the Kevlar had a consequence of forming a cavity which was not uniform as a result measurements. Recess depths, varying up to 10 mm for the centre of the test section formed a domed structure, which would have inevitably housed reverberations and reverberation in a domed structure mean that the results acquired with the first Kevlar have low confidence. Results for the flush microphones where the Kevlar was securely attached mean that the results for these remain unaffected.

Other visible features in the plots which affect results include the presence of the Blade Pass frequency of the wind tunnel fan. These peaks are evident in the lower noise plots towards the lower end of the frequency spectrum.

Additional errors in measurements are possible, although stringent testing procedures were followed and each experiment was conducted identically to every other experiment. Careful calibration and repeating of each microphone set should account for errors in experimental procedure. Concern over the tweeter overloading meant that the levels were kept down so that the white noise level is just above the level of the wind tunnel at 30 m/s, this limits the effectiveness of these results, rendering many of them as insufficiently unambiguous to give a clear result, instead trends are taken and, patterns over sets of results are measured rather than individual plots.

Form the plots it can be seen that there are minor inconsistencies with the control microphone values for same setting runs. This is to be expected as no two wind tunnel runs are likely to be identical. For this reason comparison between plots must be made carefully and the control microphone results used as a datum.

4.2.7. Recommendations

The results show trends at 0 - 20 m/s wind tunnel speeds. Extrapolation of these trends to higher wind tunnel speeds would prove useful given the nature of the wind tunnel tests in industry, where speeds of 60 and 70 m/s are more common in the wind tunnels. There is scope for repeating these tests in larger wind tunnels.

Further analysis of the results would be beneficial, given the amount of raw data acquired during the tests (well over 100 GB worth) and more than 700 runs, patterns could be determined and more trends in the results could be uncovered.

Additionally analysis of the repeated data would allow any potential errors to be screened further, and drive the accuracy of the results higher.

4.2.8. Further Development

Further complex analytical techniques can be employed, to uncover trends in the data; both raw and derived. Curve fitting, of a suitably high and robust nature can be used to clear up data and identify underlying tendencies of the different test rigs.

Continued study of the data is required, and repeating the experiments, with modified and reinforced test rig would be advantageous in validating the results further and testing successfully at speeds in excess of 30 m/s would expand the usefulness of the results as most wind tunnel tests of interest are at wind tunnel velocities exceeding 50 m/s. To this end repeating the test in alternative wind tunnels should be explored as a possibility as this would validate the results gained for an alternative environment and would allow possible testing at much higher speeds, the conclusions drawn from those results could be applied directly to current acoustic wind tunnel test programs and would prove beneficial in the immediate future.

5. Year 3 - Acoustic Directivity Wind Tunnel Test

This section is a description of the decisions, with justifications, made for the test in 2009 using an Acoustic Array installed in ceiling of the Filton Wind Tunnel (FLSWT) in order to determine acoustic array limitations for detecting noise sources in terms of the range and directivity of the sound source.

5.1. Introduction

An inverted model, in conjunction with a number of noise sources was used to test the boundaries of the capabilities of the acoustic array, with a view to quantify the maximum range of the array whilst identifying the errors associated with such extreme measurements. Furthermore the Directivity of sources are not easily determined (if at all) and reflections not accounted for or easily identified in the highly-reverberant sound field of the closed section wind tunnel.

This study aims to find ways of improving the microphone array for a given number of sensors. The primary concern is to find the range at which the array is a viable tool for source location and to determine the error in sources at the extremes of the range of the array.

It is hypothesised that improvements of the plots given by beamforming can be achieved by identifying patterns in different directivity images by direct comparison and by examining distortions it will be possible to determine and apply corrections for the distorted images of the sources.

The motivation for this study is that Aeroacoustic noise from aircraft is still a major issue and measurements in wind tunnels are required for research and development, and to validate computational results. Whilst Open-jet Wind Tunnels are a better measuring environment acoustically, Closed-section wind tunnels offer high confidence in the aerodynamic characteristics of the testing conditions.

Prior to testing the acoustic array will be calibrated as the array would have been calibrated in previous acoustic tests. This body of work is intended to further enhance the reliability and accuracy of acoustic source measurement in hard-walled wind tunnel environments.

5.1.1. Objectives

The objectives for this study include:

- – Finding the range at which the acoustic array is viable, quantify the limit of the array's effective arc
- – Comparisons of sources in range and at edges of ranges
- – Effect of sources at or near the arc limit is secondary study

- Looking at noise sources that point away from array
- Determine the dependence of the array's location, geometry and size
    - The direction effects of noise sources are to be studied
- Determine robustness of data given directed noise sources
    - By how much is the sound source locations altered by the directivity?
    - Signal strength altered by the directivity – possible correction factors to be determined?
    - highly-reverberant sound field expected in wind tunnel environment
- Effect of interference
    - Measure any signal from sources that are fully and partially shielded from array line-of-sight
    - Noise sources behind structures
        - Determine if they can be detected at all
- Reflection effects
    - Identify any reflection effects and the identify the characteristics of the reflected sources
    - Reflections not accounted for / easily identified

Determine the error in sources at the extremes of the range of the array

Physical appearance and location of the sources

Direction effects of noise sources to be studied

5.2. Test Facility and Equipment

The strategy was to use the existing microphone array, used in the previous aero-acoustic tests, normally installed in the floor of the test section but this time installed in the ceiling. This eliminates most of the lead-time and cost associated with the design and construction of a new array.

All the tests will be performed in the Filton Low Speed Wind Tunnel (FLSWT) test section, at a variety of speeds with a maximum speed of Mach 0.2. FLSWT Dimensions of tunnel: Width: 3.6576 m, Height: 3.0480 m and Cross Sectional Area: 10.4051 m^2

The acoustic array is to be installed in the ceiling turntable, with the acquisition equipment stored in the above balance room. All components will be controlled remotely from a terminal in the control room. Figure 131 contains schematics of the wind tunnel turntable with the plates that form the semi-circular acoustic array as shown.

As the array was installed in ceiling it was capable of being rotated through a 180 degree arc. This allowed the array to be positioned in three distinct positions (figure 132). The primary location was the array situated directly above the starboard wing. The two additional locations were at 90 degree increments fore and aft of the primary location.

Model 505 is a full span, unpowered low speed representative model. It has been used several times before in the FLSWT. Model 505 will be installed inverted, as in the previous test the model was used in, see figures 133 to 135. With the model inverted this eliminates any possible interference from the strut. Images of the ceiling array are shown in figure 137.

The following figures also show the support strut and faring that will be used.

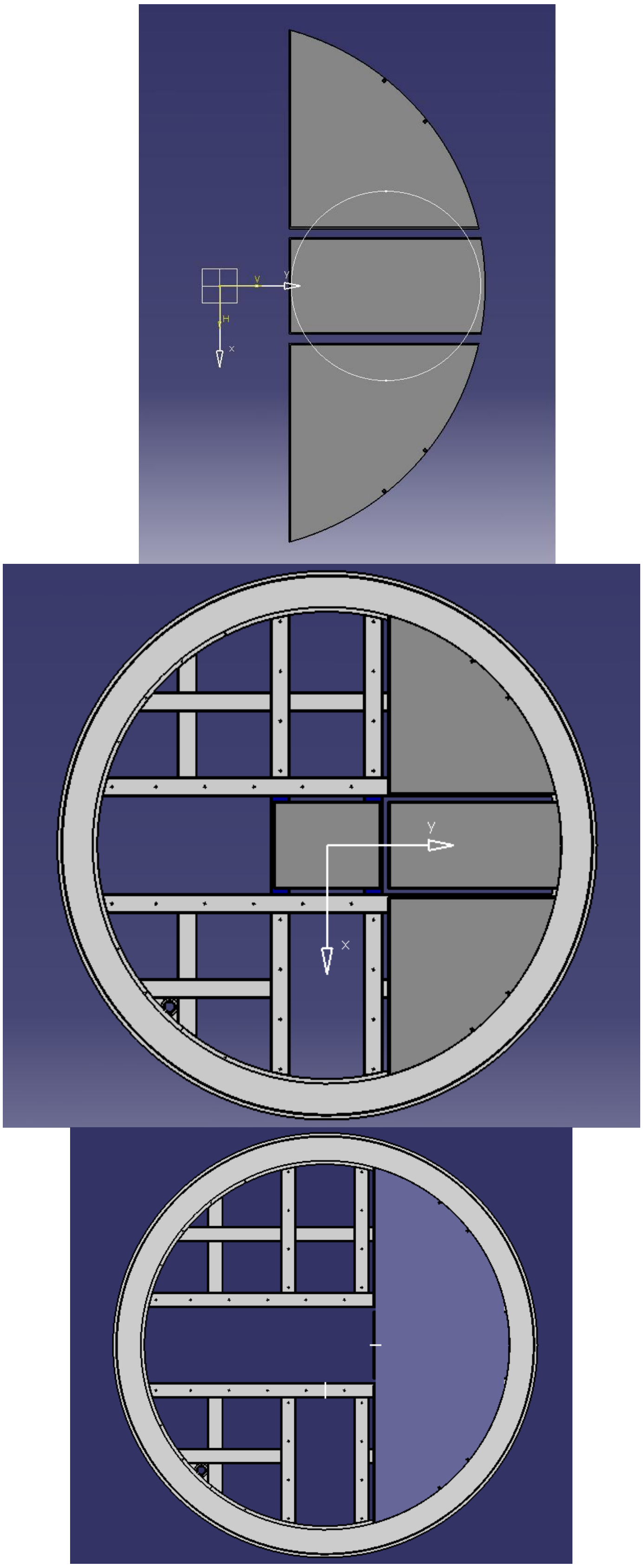


Figure 131 - Image of the Filton Wind Tunnel test section turntable showing the three plates being replaced by the semi-circular microphone array.

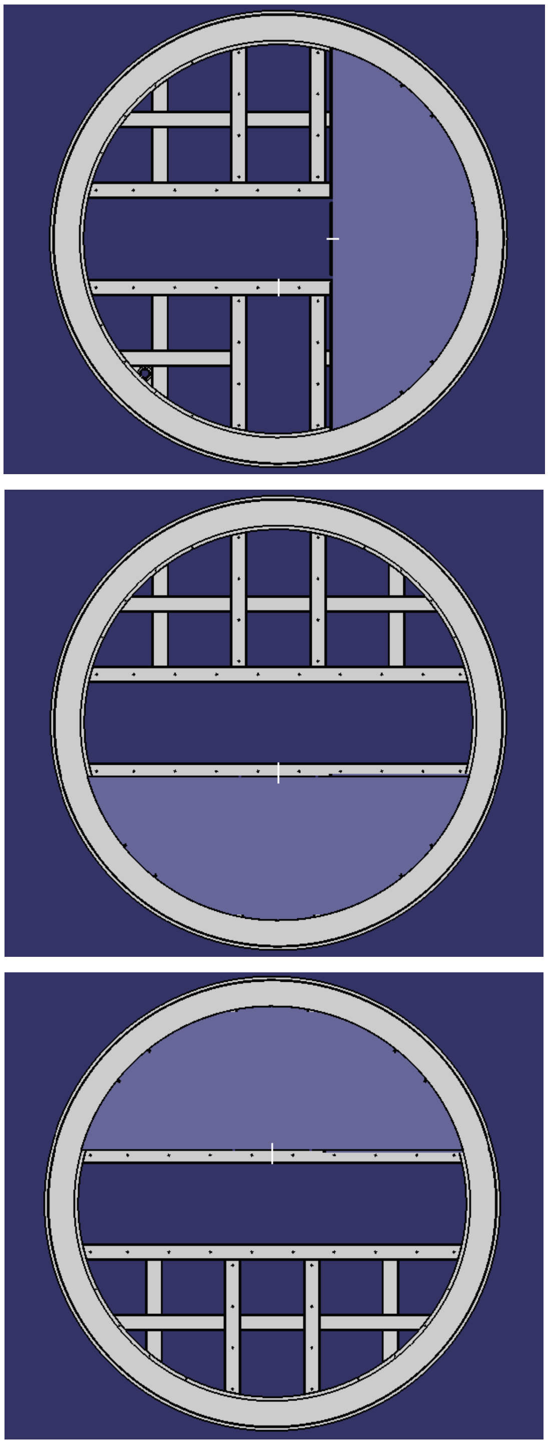


Figure 132 – Location of the array in the circle indicating the 3 array locations in the wind tunnel ceiling: centred, 90 degrees aft and 90 degrees fore.

Model 505 is to be installed inverted on a single mono strut assembly, with appropriate fairing and central strut. The whole model and strut assembly will be located in 3 positions on the under-floor balance.

- Test Position 1 – The strut and model will be located in the centre of the balance and turntable as in figure 133.
- Test Position 2 – The strut and model will be located upstream of the centre point as in figure 134.
- Test Position 3 – The strut and model will be located downstream of the centre point as in figure 135.

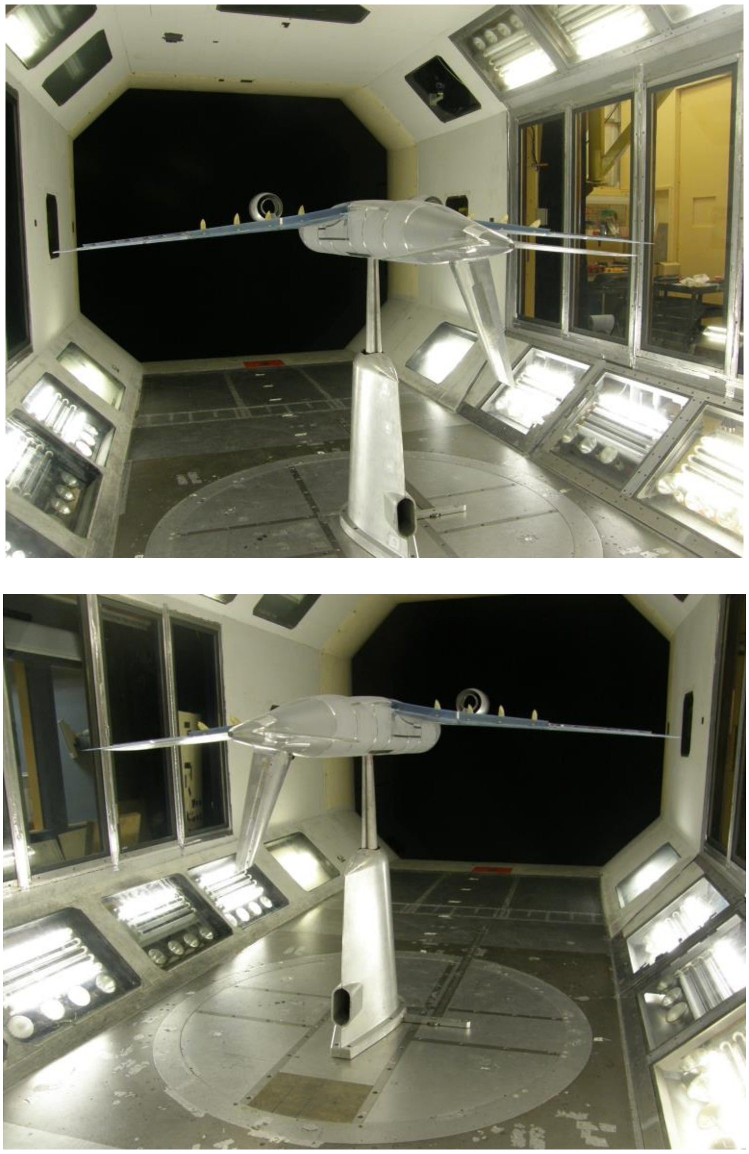


Figure 133 – Model in strut position 1. Images showing Model 505 mounted inverted in the test section.

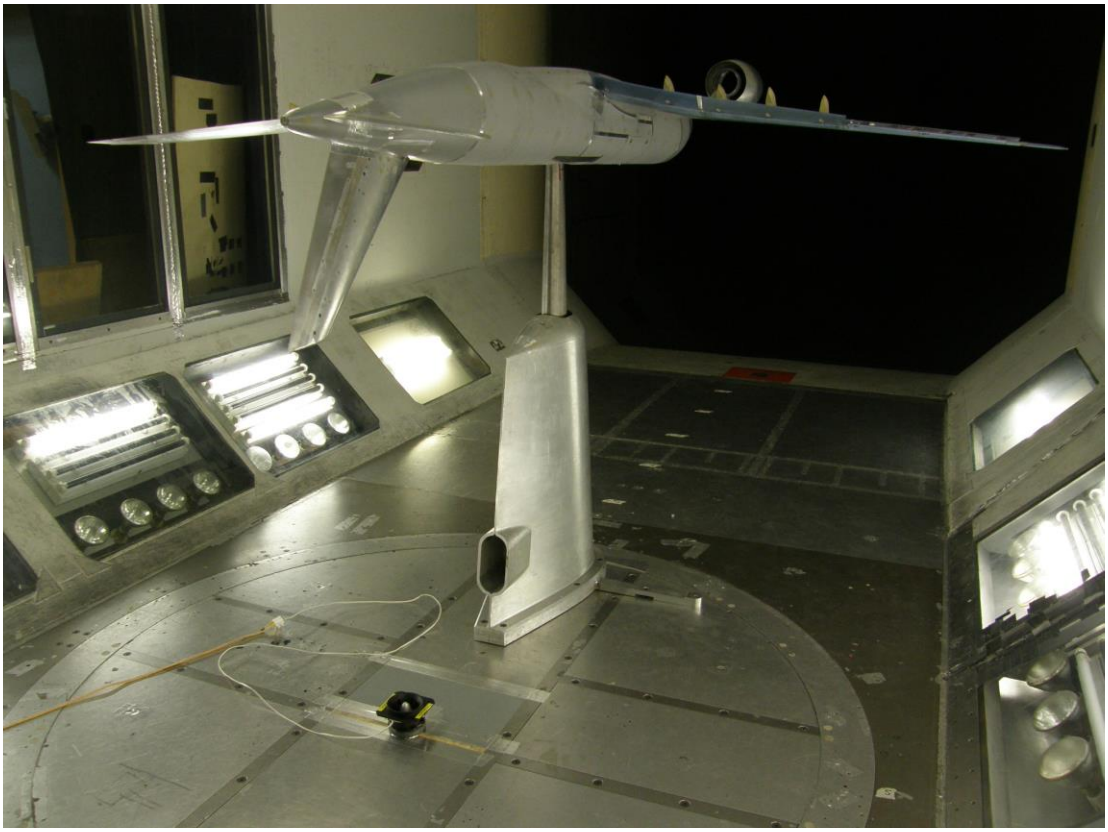


Figure 134 – Model in strut position 2.

Figure 135 – Model in strut position 3.

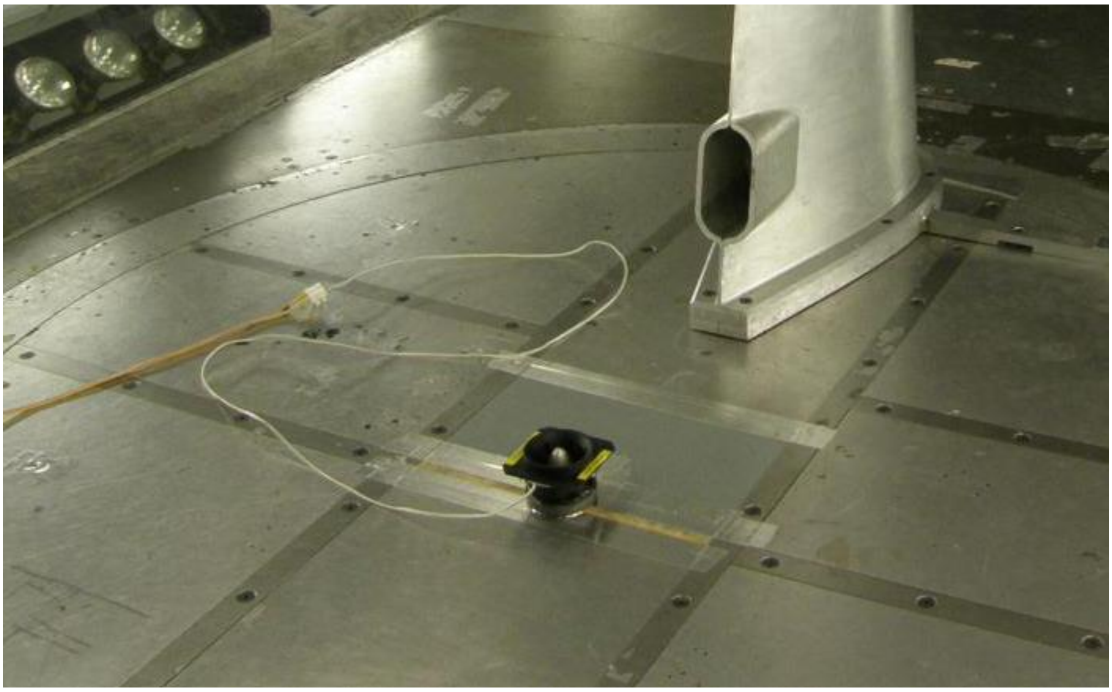


Figure 136 – Noise Source used for testing acoustic array.

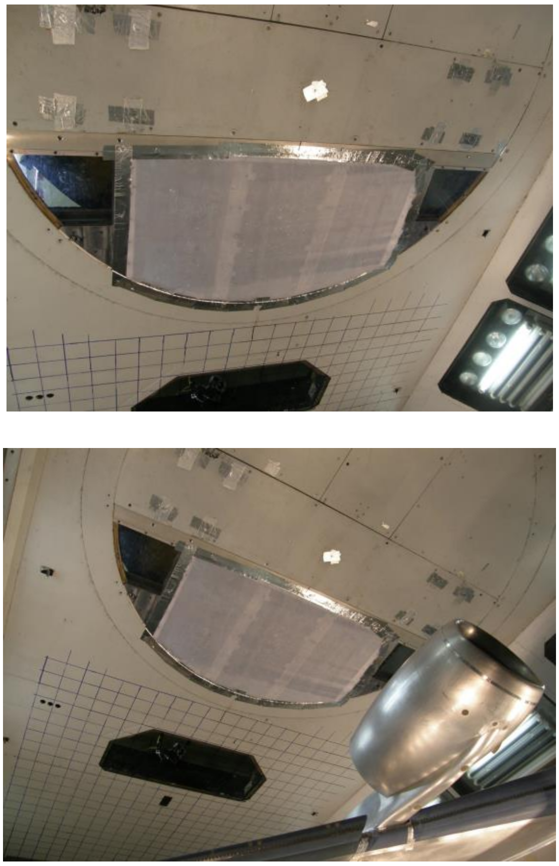


Figure 137 - Images showing acoustic array mounted in the ceiling of the Filton wind tunnel test section.

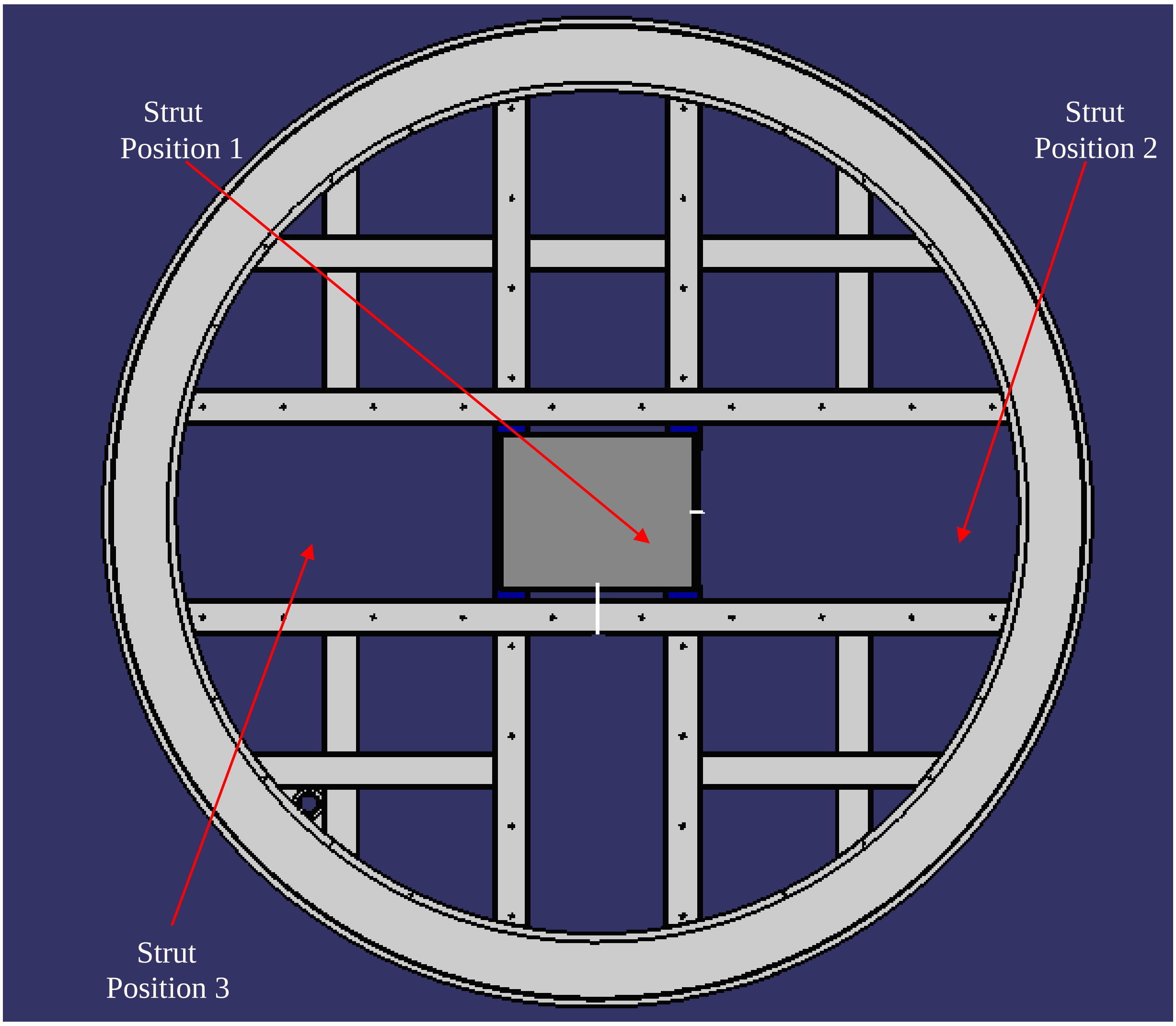


Figure 138 - Locations of support strut, with respect to the turntable for all the test positions.

Modified central struts fairings may be required to support the model in positions 2 and 3. Additional plates for the test section floor to fill the gaps when the model and strut are in positions 2 and 3 will also be required.

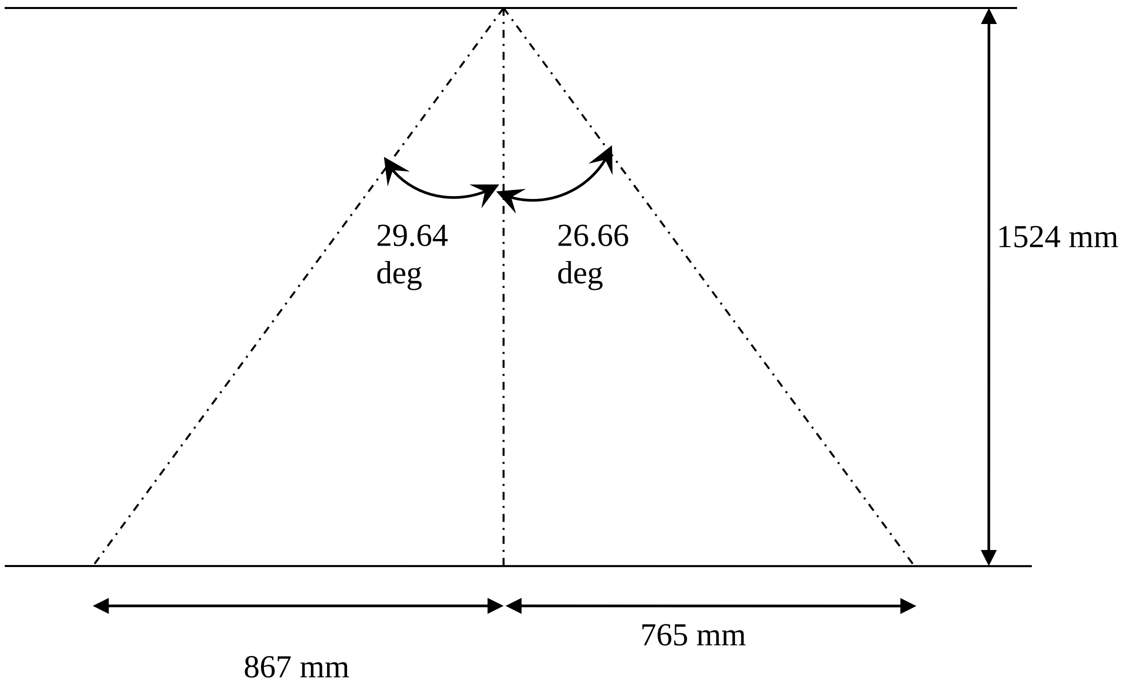


Figure 139 - Diagram showing location of model centre point in the three locations compared to array centre point (array position centred).

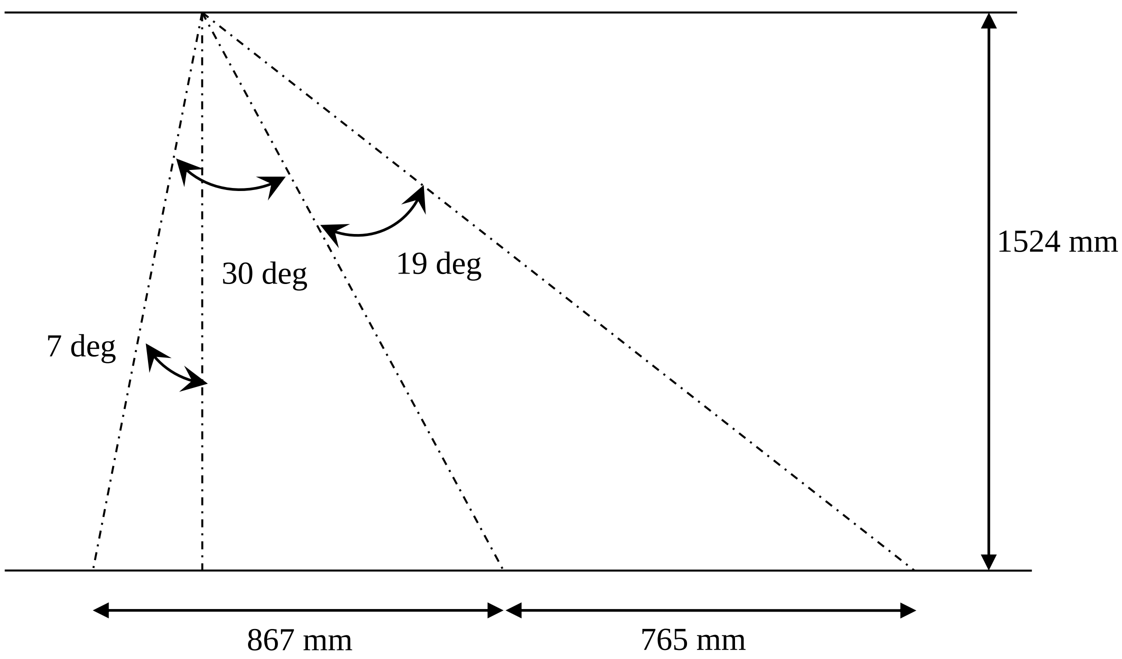


Figure 140 - Diagram showing location of model centre point in the three locations compared to array centre point (array position forward).

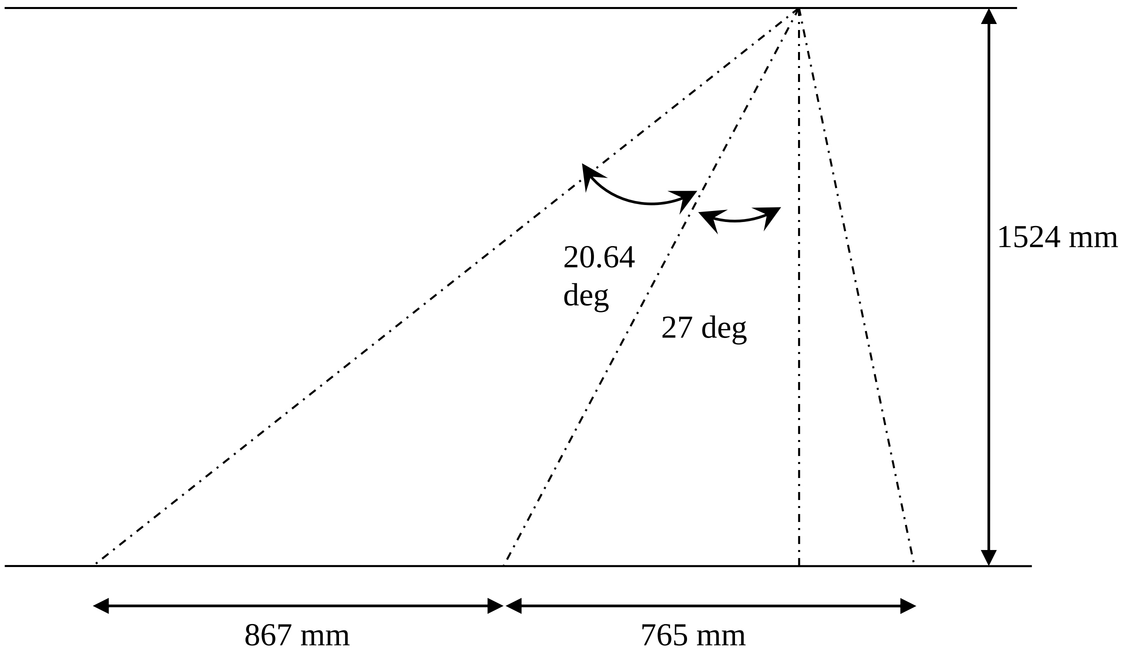


Figure 141 - Diagram showing location of model centre point in the three locations compared to array centre point (array position aft).

The additional measurements with the array in the second and third positions will also be taken to give an overall aircraft view whilst the primary array location is designed for more localised wing noise measurements.

Focus will be made on the test ranges specified above, with the entire model analysed for sound sources from all three array positions and in all the model positions, giving a total of nine model positions relative to the array.

5.3. Measurements

For this test samples of 17 - 20 seconds were acquired using 111 channels, acquiring simultaneously, at a sample rate of 96 kHz.. This enabled post-processing up to frequencies of 48 kHz. FFT processing was done with a block size of 4096, and 400 averages with no overlap, using a Hanning window and a narrowband frequency of about 23Hz. The autospectra are discarded from the Cross Spectral Matrix to suppress the background noise of the tunnel (Diagonal Removal technique). A typical acquisition (approximately 17 – 20 seconds) results in a raw data file size of 1.4 gigabytes. Wind tunnel data, (Alpha, temperature, Barometric pressure and time stamp) were taken at each data point.

Alpha sweeps to maximise the noise generation of the aircraft were performed, with pauses at selected Alphas (as indicated on the test matrix), the preliminary Alpha selection included taking measurements at -4 degrees, 0 degrees, 4 degrees. 8 degrees and 12 degrees. This was performed at the top speed of 68 m/s, Speed sweeps to determine effect of wind tunnel flow on the sources at the extremes of the array's cone of acquisition were also performed. Speeds of 40 m/s, 50 m/s, 60 m/s and 68 m/s were used.

Acoustic Beamforming plots were produced showing source location. In order to quantify the distortion of the same source from different directions, comparisons of the source area for a given frequency, alpha and speed settings can be made from the different test positions. This direct comparison of a sound source across several plots should allow a qualitative determination of the effect of acoustic measurement from different distances and directions. By overlaying the images of the recorded sources and comparing source sizes and levels a quantative determination of any distortion can then also be made. Integration over source areas will give source levels (comparable at the same frequency).

5.3.1. Data Validation Process

As before for any device utilising microphones where accurate levels are required a certain amount of calibration is required. For this test calibration is a necessity as large numbers of sensors are used. Additionally the use of multiple pre-amplifiers to power the microphones meant that inherent errors were to be expected as the devices employed were sufficiently different. Reference Microphone Calibration eliminates any inconsistencies by comparing each and every microphone to a pre-calibrated reference microphone. Each channel and the pre-calibrated reference microphone are subjected to white noise simultaneously. With the reference microphone placed as close to the test microphone during this test direct comparisons can be made and the Coherence of the two microphone signals is used to confirm the quality of the microphones operation. From this form of calibration, phase and magnitude data is collected for each channel and used in the post-processing.

Reference microphone calibration and compares each electret microphone to an instrumentation-grade microphone in this case a GRAS Type 46BE microphones was simultaneously subjected to white noise with the Electrets. The transfer function is recorded and the coherence between the microphones is used as integrity check. From this calibration, phase and magnitude data can be used in the post-processing in the ‘SotonArray’ (the name used for the set-up in Airbus) software.

The coherence between the test microphone and the reference microphone is an important indicator showing that the test channels are functioning correctly and within the required parameters.

5.3.2. Validation Strategy

This calibration procedure allows the use of the electret microphones with a high level of confidence. It also eliminates any discrepancies between channel pre-amps, cabling, connectors and microphones. This calibration technique eliminated the errors incurred from using separate pre-amps and allows all the channels to be used together effectively with common values set by the reference microphone.

The calibration of all the channels also acts as a check to ensure all the channels are functioning, any broken or damaged microphones can be located, repaired or replaced before the test.

A number of numerical tests were made on the microphone layout to determine the effectiveness of the design selected. The first is the co-array, a vector spacing view of the array. The Co-array of the microphone layout shows the vectors covered from the microphones layout. It allows easy determination of appropriate microphone layout as it can show any gaps or holes in the array's ability to receive data. Although not depicted in this report the co-array of the design was seen to leave no gaps.

As mentioned earlier in this thesis, the Point Spread Function (PSF) is another test that can be made to microphone array designs to determine their suitability. The point spread function is a mathematical analysis of a microphone array layout which can be used to determine how good the mainlobe response will be compared to the sidelobe. The point spread function also shows the distribution of the sidelobes with respect to the mainlobe. The PSF shows levels at certain frequencies from the microphones layout. Again not depicted in this report the PSF was determined to be sufficient for data acquisitions.

The mainlobe of the response for this array was found to be, at low frequency, quite wide. This is likely to be due to the small size of the array diameter, as a consequence the response is expected to be low resolution; given the form and small size of the array, this was somewhat unavoidable and expected.

5.3.3. Limitations

One of the aims of this test is to determine the limit of the array's capabilities. As such the area limit of the array will be discussed in later sections. Due to the physical limitations of the array, as mentioned in section 3, low frequency analysis was not possible with the array used. As a consequence frequency values below 4 kHz are not analysed in this study and a focus is made on higher frequency plots of 16 kHz and 20 kHz as these frequencies are amongst the most relevant for wind tunnel tests.

Equivalently Hardware and Software limitations, specifically the acquisition cards made the upper limit of the Frequency range to be that of 48 kHz, although testing beyond 30 kHz was not deemed necessary for this test. Limitations still exist in the processing of the results, the desire to have near real time results available during a test campaign is a continuous target and this capability for instant results is being actively developed within the system for future use, but the technical limitations of the hardware involved remains a large factor and real time processing is unlikely to become available until the advent of much faster processors and hardware engines, as of 2009 – 2010.

5.3.4. Errors and Uncertainties

As with any system there are a number of weaknesses which could produce uncertainties in the results. The location of the array requires millimetre accuracy in order to ensure accurate results in post–processing. As such care was taken during array installation ensuring accuracy in the range of 0.01 m.

System noise is inevitable, but is accounted for through calibration and careful processing, this could affect the accuracy of the levels. Processing errors are unlikely; these errors are mitigated by using repeat runs and processing the same configuration over a number of frequencies. Care has to be taken to isolate each channel from any interference or signal noise.

5.4. Results

In total, over the days of testing and a number of configurations the following statistics were recorded.

- 12 Day test – 10 days of data acquisition
- Over 130 runs made
- Over 1.9 Terra Bytes of data
- Over 1400 acquisitions
    - For each acquisition plots can be produced to compare all sources from a number of directions

For each run two beamforming plots (conventional beamforming and CLEAN-SC) were produced for each 1/3 octave centre-band frequency, 2, 2.5, 3.1, 4, 5, 6.3, 8, 10, 13, 16, 20, 25 and 32 KHz. Plots were generated for a scan area encompassing the whole model, and then cropped to half-model for display purposes. This is to ensure that the integrated levels are not contaminated from noise sources on the half of the model shadowed by the strut. The scan plane is 1.524m from the tunnel ceiling at zero incidence, and rotates at the aircraft pivot point with angle of attack. The beamforming plots were all saved in Tecplot .dat format without fixed levels. The airframe geometry was produced for all the angles of attack to allow the aircraft geometry to be overlaid over the plots allowing direct source location to be performed visually, and saved as .dat files.

These results (figures 142 to 144) show the landing configuration of the wind tunnel model located at the three different model points with the array location in the primary location centred over the starboard side (looking at the underside of the port wing). In these plots the array centre is located at the [0,0] co-ordinates.

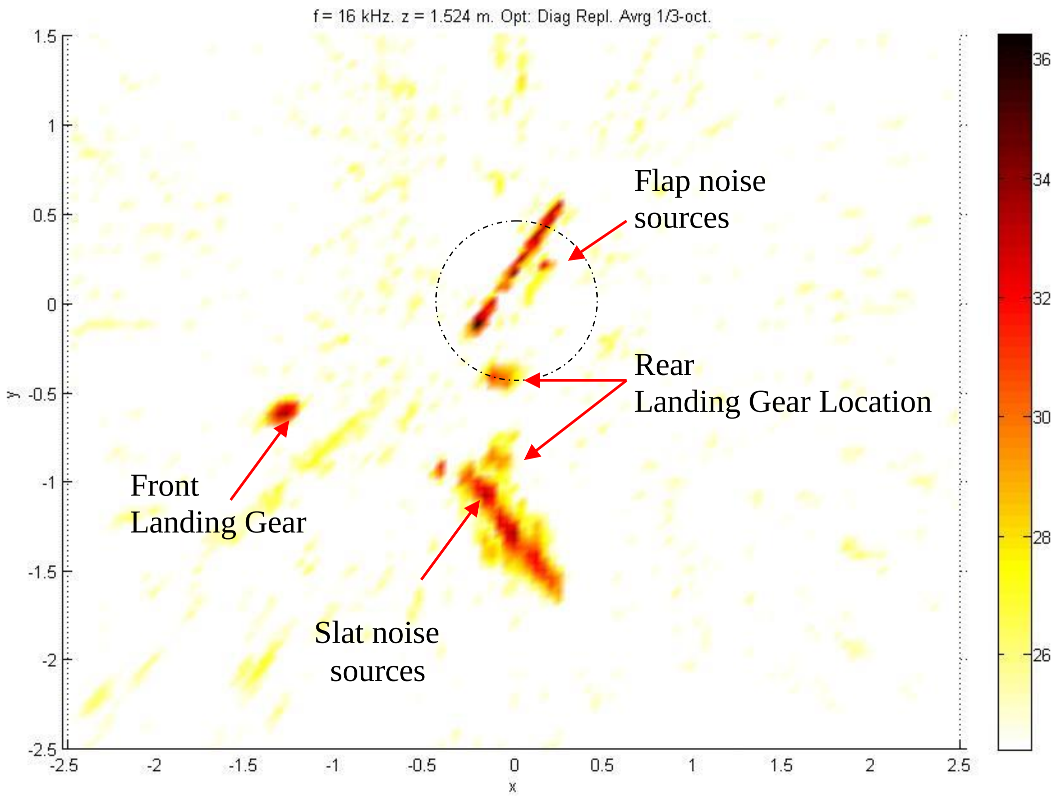


Figure 142 - Beamforming plot of aircraft model located in centre point from array, at the primary location over the starboard side (looking at the underside of the port wing), at 16 kHz.

The figure above shows the location of the array relative to the model as well as highlighting the features that are easily identified for the landing configuration.

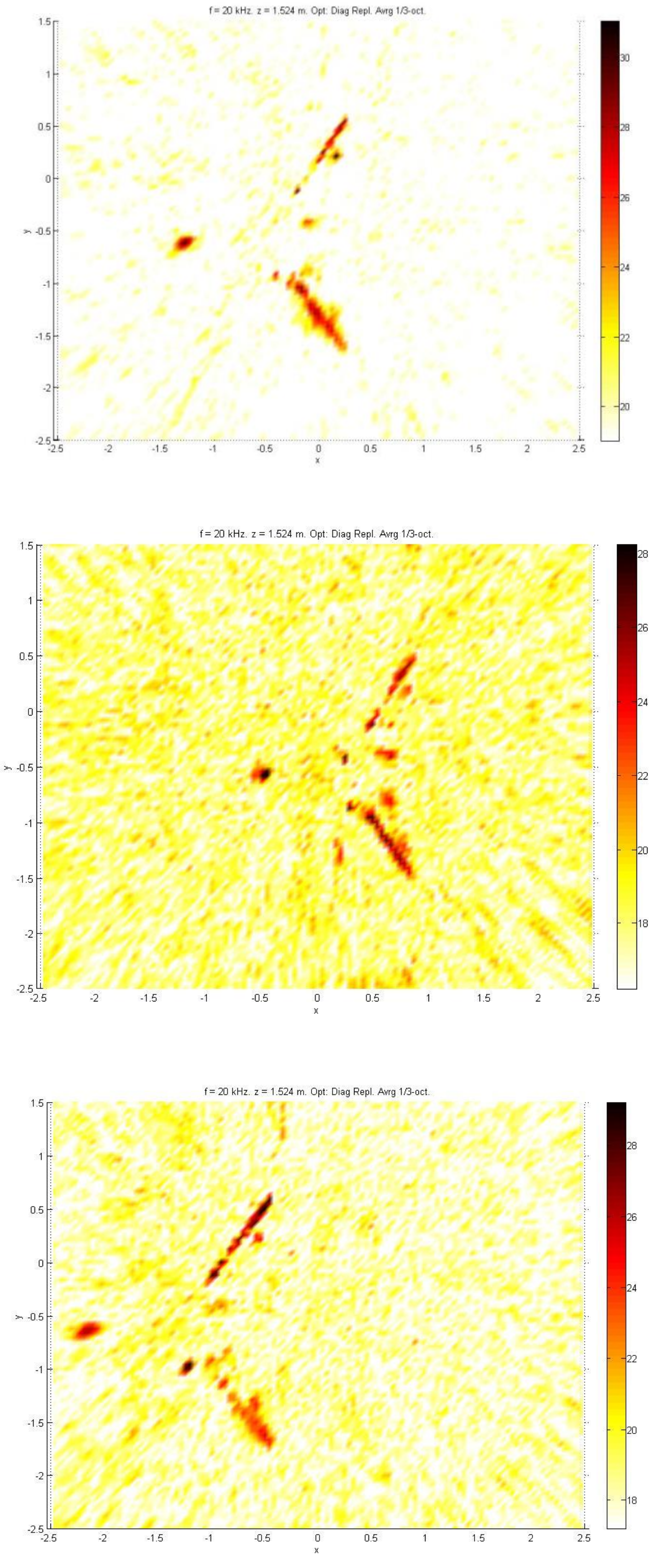


Figure 143 - Beamforming plot of aircraft model in landing configuration, at three positions, from array at primary location over the starboard side (under the port wing located at the top of the images) at 20 kHz.

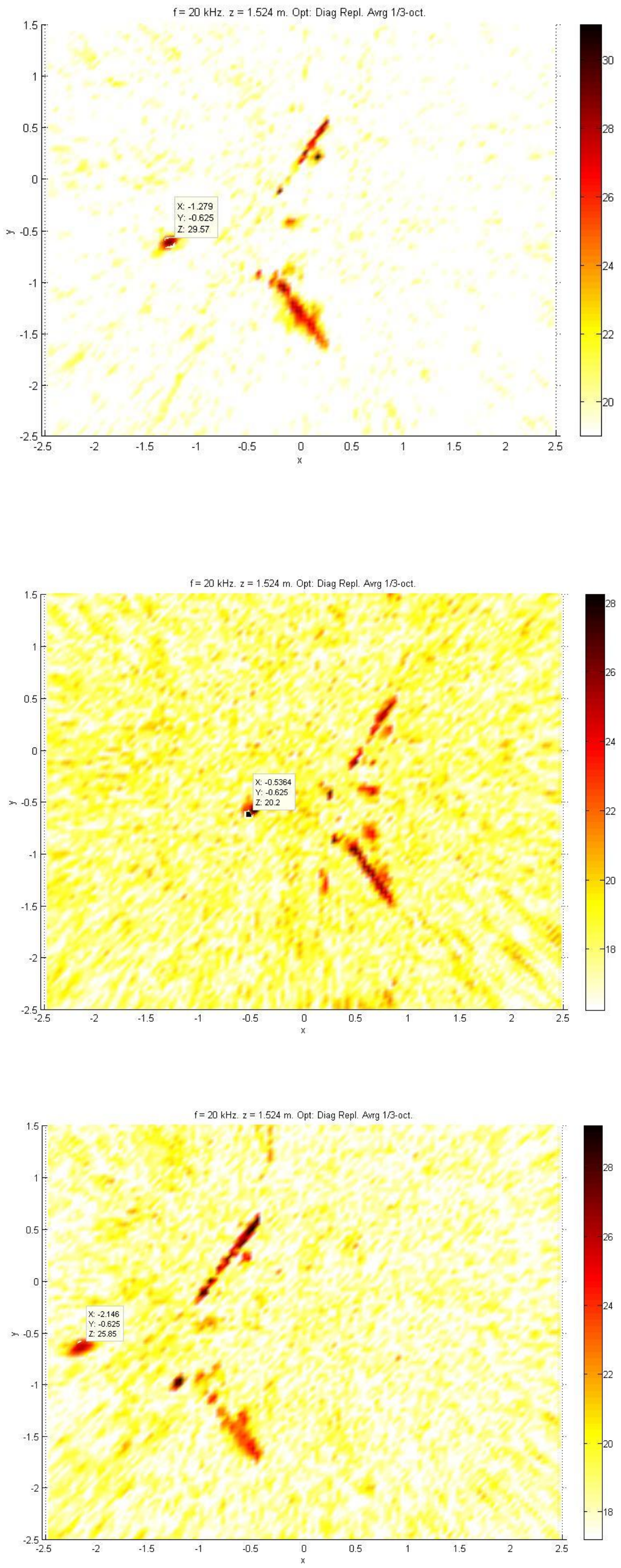


Figure 144 - The comparative plots showing the same source from various directions, levels are only compared with like frequency plots.

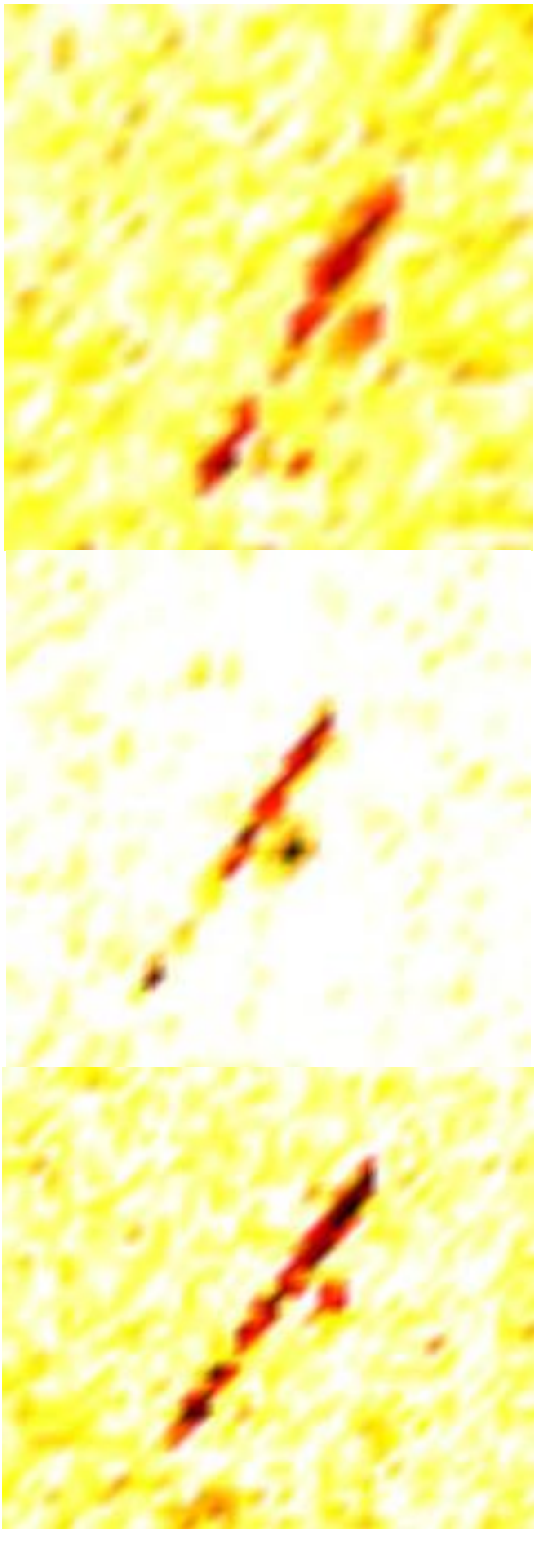


Figure 145 - Flap and Slat focus at -29 degrees (top), 0 degrees (middle) and 26.66 degrees (bottom).

5.4.1. Primary Noise Sources

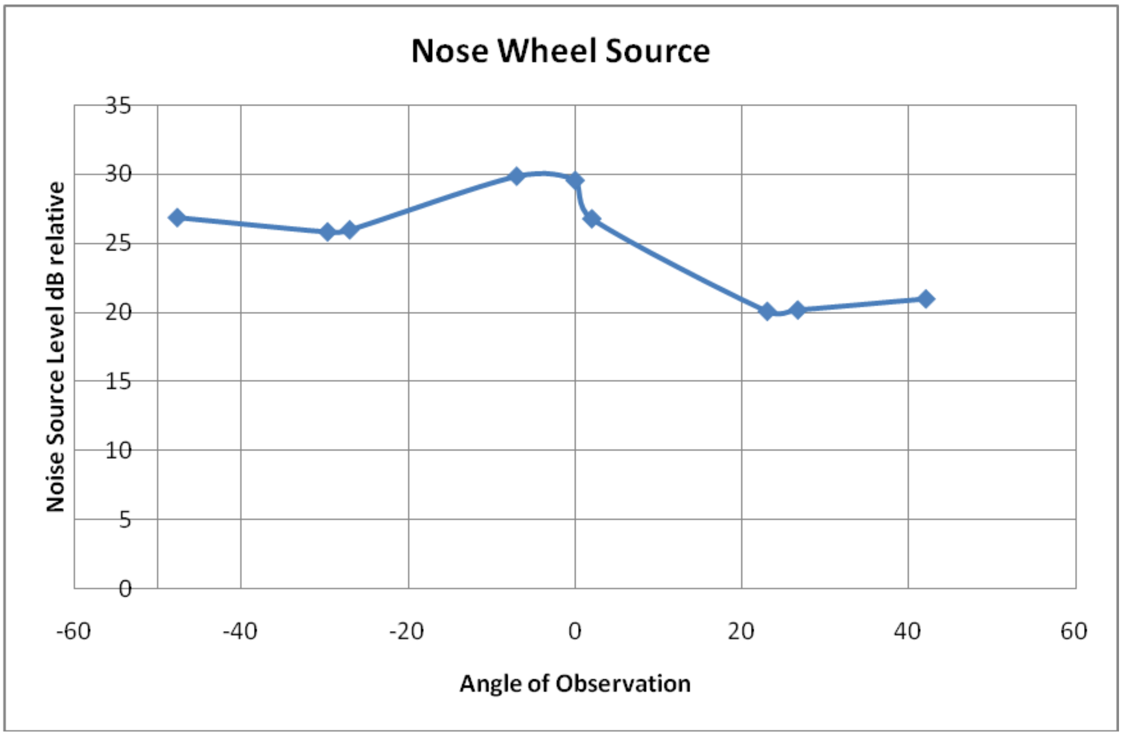


Figure 146 – Nose Wheel Source Directivity analysis, showing noise levels of the Nose wheel source recorded at various angles of observation (degrees).

From these plots the front landing gear source is seen to vary considerably in level, depending on the angle of observation. A 20 db drop in noise level can be seen between viewing the nose wheel from 0 degrees to 20 degrees.

The results for the rear landing gears also varied considerably. When the rear landing gears are viewed from 'behind' the source levels drop considerably and in some cases are not picked up at all on the plots. (Comparing the third panel to the other two in Figures 143 and 144).

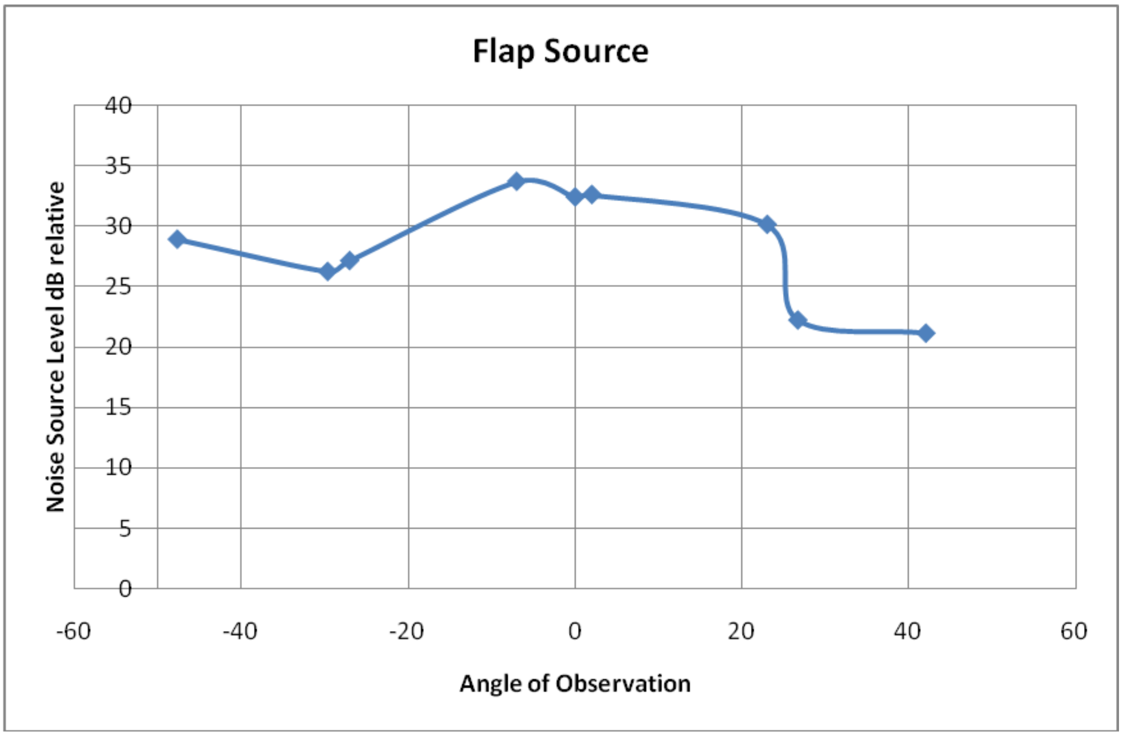


Figure 147 – Flap Source Directivity analysis, showing noise levels of the Flap source recorded at various angles of observation (degrees).

Flap noise was recorded to be at higher levels when viewed from ‘below’, whilst the noise level drops when viewed from ‘behind’ and drops considerably when the source is viewed from the ‘front’.

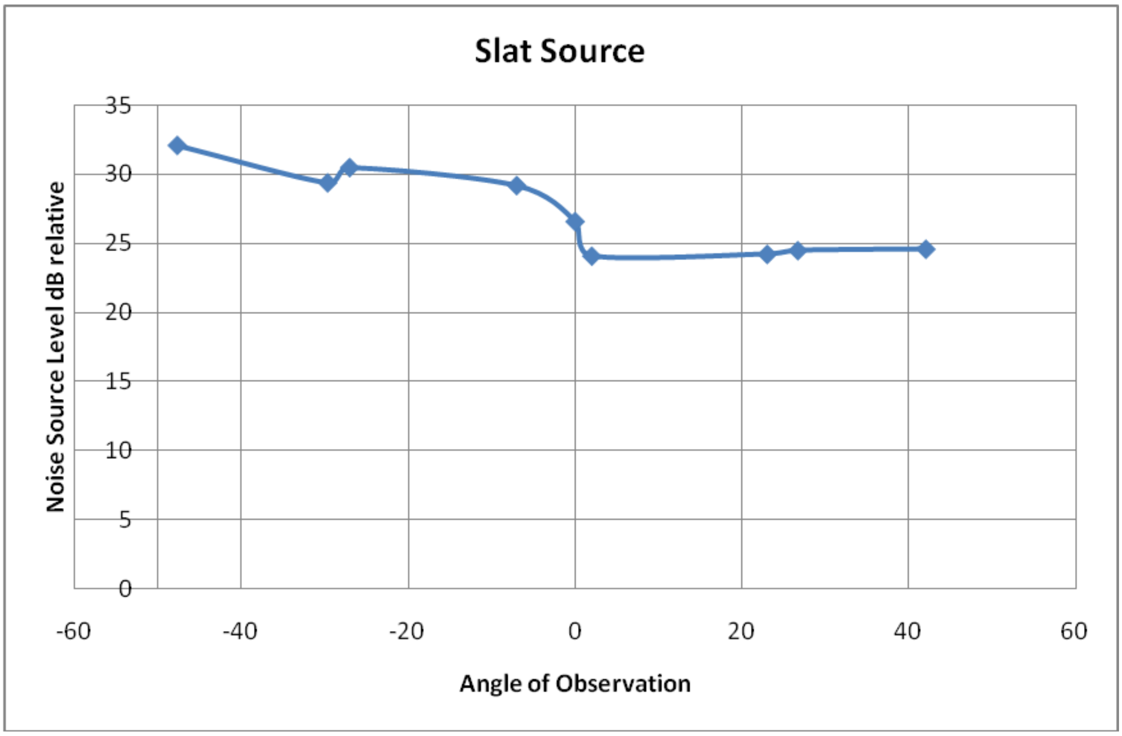


Figure 148 – Slat Source Directivity analysis, showing noise levels of the Slat source recorded at various angles of observation (degrees).

It is also noted that the slat noise sources are emphasised in different locations depending on which view is taken. In the examples above slat noise appears to be more prevalent when viewed from behind.

5.4.2. Discussion

The low frequency results are clearly of a lower resolution, whilst higher frequency results are of a higher resolution. The plots have all been capped to a dynamic range (a different range for each frequency) allowing same frequency plots from different configurations to be compared for relative levels. These typical plots show how the resolution of the noise sources varies through the frequency range and show the different range levels assigned to each frequency.

In general the plots have a dynamic range of between 10-14 dB. All plots are clipped at 15dB below the peak for that plot. For plots where there are no significant sources, a phantom source may become apparent at the centre of the array.

As expected low frequency results suffer from poor resolution, particularly below 4kHz. This is somewhat improved by CLEAN-SC, since the sources are replaced by an artificial main lobe which is frequency dependent [[4] Sijtsma, P., CLEAN based on Spatial Source Coherence, AIAA 2007-3436].

On most of the plots noise from the strut/aircraft junction is evident. In order to remove such effects, a common technique is to subtract the background CSM. This refinement was performed, and whilst it reduces this effect, it also affects other sources of interest noticeably. Therefore, the delivered results have not had the background CSM subtracted.

Sources identified using beamforming have finite size, which is frequency dependent. This means that both levels in plots and integrated levels should not be compared between frequencies. Levels in plots can be used for relative comparisons at a given frequency. Integrated levels should be used for more accurate level conclusions, however they cannot be treated as absolute levels, because of factors such as source position, array size and location and source directivity effects. Most current research on microphone array processing is focused on this issue.

Different frequency plots show that the same configurations show significantly different levels in the source strength. From this it can be deduced that it is important to only compare same frequency plots with each other as the level ranges are dependent on the frequency of interest.

Sources located at over 1 m from the array centre (taken from a line directly below the array centre) show distortion compared to the same sources identified in plots taken with the array centre closer to the source, with the array diameter measuring 0.7m, distortions were noted in several results.

These distortions follow the pattern of an increasing ratio between the major and minor axis. From these results it is hoped that a correction algorithm can be determined.

5.5. Directivity Conclusions

This acoustic test was conducted in the FLSWT between 23rd March and April 4th 2009 where a beamforming array was installed in ceiling in three locations and comparisons of sources in range and at edges of ranges were made. The effect of a known source at or near the array arc limit was studied and the effective range of the array was determined to be approximately within 1.5 times the diameter of the array from the array centre.

A number of plots revealed distortion in the extreme edge of the array, where circular sources were expected, and distorted images were generated in the results. Additionally it was proven that the source strength from wind tunnel models is dependent on directionality and the same source viewed from different angles produced up to a 10 db difference.

Reflections and false sources were also identified within the plots and the emphasis was reinforced that care must be taken in distinguishing and categorizing so called "noise sources" in acoustic plots.

Correction of the distorted sources at the edges of the array scan capability should be achievable by determining the major and minor axis of the oval shaped sources and applying an algorithm directly to the beamforming code to give correct source locations and strength levels. This should form part of the continued work for continued research or a doctorate to be undertaken at Southampton University in the future.

6. Conclusions

This final section summarises the conclusions for the three sections of this thesis. The implementation of an acoustic array in the Filton (Section 3.3 to 4.1), the Recessing study (Section 4.2) and the Directivity study (Section 5).

6.1 Initial Conclusions

The use of arrays is well documented and it is clearly a technology that is being used more and more widely and has greatly increasing demand. Research in the area is relatively new, but continuously increasing with a wide variety of avenues being explored in recent years as well as currently.

This study aimed to implement a working acoustic array for measuring noise sources in a non-acoustic closed section working wind tunnel. In summary, the first twelve months of this study have yielded a set of specifications for an initial array design after a successful acoustic survey of the Filton wind tunnel. With respect to the array design there were two primary concerns; that of the microphone layout and the physical design, with future thoughts on the possibility of implementing a recessed array with Kevlar covering to improve the array performance.

Through simulations of array configurations, techniques such as Co-arrays which showed the vector covering from microphones layout and the Point Spread Function (psf) which showed the response levels at certain frequencies from microphones layout, calculations were performed and different frequency point spread functions responses were compared. A number of microphones set ups were designed and tested and a wide range of frequencies used. In this report frequency values of 1 kHz up to 40 kHz were tested. The lower limit set by the size of the array and the higher limits set by the limitation of the array hardware and software.

The array design was modified to fit around wind tunnel structure and problems with results from the Point spread function were subsequently found. Several redesigns and tests were instigated and despite a number of microphones being ‘lost’ a sufficient amount remained to perform the required data acquisition. The maximum number of microphones of the original design having had 264 microphones the ‘lost’ microphones brought the number down to 209 due to beams and rails and finally down to 112 for the final design. The co-array and point spread function techniques allowed easy determination of appropriate microphone layout, and the co-ordinates were already made available to create the main plate ahead of the November 2006 preliminary test.

A Microphone layout was finally established using an Underbrink Spiral design, and utilising initially only 112 channels due to a limitation in the number of pre-amplification and data acquisition channels and as was expected the array was installed in the Filton wind tunnel by the middle of 2007; where a second test was subsequently performed successfully and with enhancements to the hardware and software, which improved on the results achieved in the November 2006 preliminary test.

Furthermore the test results achieved in the two tests in 2006 and 2007 have proved that the use of electret microphones in microphone arrays is sufficient for beamforming and consequently source location. Microphone failures in the tests also showed that the beamforming code employed is fairly robust as the results returned were still accurate and useful, and only a loss in resolution of some low frequency results was identified.

Microphone array technology continues to be a critical part of the overall instrumentation suite for experimental aero-acoustics. This project involved building on existing technology; augmenting and modifying said technology to fit various wind tunnel facilities. After successful implementation of the microphone array, the current endeavour is to increase the user friendliness of the microphone array technology, as well as extending it beyond existing limitations. Studies performed in various wind tunnels will eventually determine the most appropriate construction and technology integration techniques and combine the acoustic array technology with other current and future wind tunnel techniques.

A study into acoustics, aero-acoustics and computational methods to gather and present the data received in a clear and coherent way was performed as the initial phase of this thesis with an emphasis on producing data that is exclusively relevant to industrial needs. Industry and the advanced technologies implemented in a working wind tunnel form the basis for any future work.

Microphone arrays can provide qualitative acoustic measurements in wind tunnels and whilst current beamforming technology is now relatively mature; limitations on accuracy and confidence remain. Beamforming and the associated acoustic experimental practices allow routine measurement of aero-acoustics, giving only qualitative results and with severe limitations in processing data. Assumptions are made, such as all the noise sources are uncorrelated monopole noise sources and moreover source strength predictions are meaningless, and due to necessary averaging processes, only comparative levels can be accomplished. Current research programmes are in place to try to overcome these limitations, with ongoing studies trying to develop quantitative measurements from acoustic arrays installed in closed, hard-walled, wind tunnels.

Near-real time processing should also be attainable once data processing, management of said data and the workflow of the data is fully realised. With acquisition of data taking less than 20 seconds and quick results techniques, developed for the July 2007 tests, which allowed results to be displayed in 10 minutes already achieved, it is conceivable that even faster processors and future hardware will allow full results to be displayed even faster after acquisition. Whilst full results still take a considerable amount of time, with the advent of advanced processing and data handling mechanisms and techniques, results should be available much faster in the future. If these techniques become available to this system, they should be applied as appropriate.

Several successful implementations of the beamforming technique and use of an acoustic array to determine noise sources on a wind tunnel model in the Filton wind tunnel has been carried out and major sources of noise identified from a model in flow conditions. This was achieved at various operating speeds, but most importantly at the standard operating speed of 68 m/s (0.199 Mach), where concerns, of the overall background noise level affecting any source location, were made.

In summary, aeroacoustic measurements in the Filton wind tunnel can provide useful data, and there is scope for improving the equipment from the array design itself to the hardware employed and the software used to allow more detailed, better resolution measurements across a wider range of frequencies to be carried out in the future.

6.1.1. Further Software Development

There is scope for further improvements to the software used to drive the acquisition and processing of the acoustic data. Main objectives would be to improve speed and the amount of data processed, with a view to have a system capable of quickly processing a lot of data from an ever increasingly large number of microphones. To this end, the application of the CLEAN-SC algorithm [56], in the post processing has already been achieved. Further comparison of the old beamforming code to the newer beamforming code and the CLEAN-SC algorithm must be done to quantify the improvements that are made.

Additional modifications to the beamforming processing code, that have already been implemented include the splitting of the initial processing of the initial raw data into smaller chunks, which the Matlab software can deal with at a faster rate than all the raw data at once. Also the format of the raw data read in from the acquisition Labview software has always been in binary format. The Matlab code was originally written to only accept Matlab own input files. Modifications to the code have meant that the binary format is now the norm for the Matlab input and as such the need for any form of file conversion before processing has been eliminated. This has sped the processing speed considerably. Further code improvements are always possible.

The ability to plot data in virtually any format was realised by the July 2007 test, where Tecplot output was required. Additionally preliminary ‘quick’ results, processed during wind tunnel tests need to be improved; allowing changes to any wind tunnel models to be made, with a high level of confidence and in situ, during the test programme itself.

One aspect of the system that can be improved is the user interface. Work has started on a graphical user interface (GUI) for the data processing code that is implemented in Matlab. The GUI will allow any user to preset all the necessary variables for a series of tests and process entire batches, producing easily labelled and viewed pictures stored in separate folders for easy reading. The GUI should allow any user of moderate knowledge to operate the Matlab code without having any previous Matlab experience. Initial designs are for a single page user interface with all the necessary inputs on one page, however given the number of inputs it might be easier to have a multi-page user interface with a set of instructions or a tabbed series of pages with individual pages for user inputs, test inputs, overall inputs and output selection.

A GUI would seem to be essential to permit the system to be used by any number of people and to reduce the technical know-how prerequisite of the system. This would be the first step in any potential commercialisation of the system to broader use.

The final step in any software is to achieve a full automation of the system and have it totally integrated with the internal Filton wind Tunnels system. Any automated systems must not be so integrated as to be too difficult to maintain and upgrade. The system must allow constant upgrading of equipment and software in order to improve processing time and accuracy in the future.

6.1.2. Further Hardware Development

A fundamental question exists in the use of Electrets microphone versus other types. Specifically the use of these cheaper microphones is considered insufficiently accurate for the purpose of advanced beamforming acoustic acquisition.

As an example, in the case of NASA performing acoustic tests, the use of more expensive B&K microphones for a 600 channel array is known; even further these costly microphones are taken and stripped down to the diaphragm.. Unfortunately, high channel cost, from using such microphones remains one of the limiting factors in the construction of arrays containing hundreds of channels which could be used to implement advanced beamforming algorithms. However, the study here has confirmed that the electret microphones used in the array do function up to 44 kHz. Other studies [58 and 60] have taken a look into the use of electret microphones and have also validated their use in microphone arrays. The results of the investigations revealed that microphone frequency response from these electrets covers a range of 250 Hz - 40 kHz. The performance is also evaluated and results up to 20 kHz have been achieved by various members at the University of Southampton, whilst tests at Filton using these microphones have returned results up to 32 kHz, see section 4.1.

With regards to frequency range, any study into increasing frequency range must remain useful, that is to say within a frequency range that is in the audible range after conversion from a model to real size, for studies into audible acoustics, whilst other ranges of frequencies might prove useful in determining flow conditions and irregularities in the aerodynamics of a model.

Further development of the point source can be made, especially in to a variety of source types. The assumption is that the sound source is a monopole, but it is known that this is not necessarily the case in real wind tunnel models; for example slats on aircraft wings tend to be dipole in nature. Creating a device to simulate a known dipole at a known level could lead to a benchmark against which the beamforming software and microphone array configuration could be tested against.

Any new array design would include the two primary concerns of the microphone layout and the physical design, (i.e. whether flush or recessed with a covering). The microphone layout in designing a much larger Microphone array could use the Underbrink Spiral design, as before or a possible alternative [28 and 54]

Creating a much larger array with a large number of channels would introduce issues with wiring and the physical space required for all the necessary hardware, but a larger number of microphones would allow multiple configurations to be realised. Array within arrays could be designed, with a few microphones being used for one application and a larger number used for others. With a large selection of microphones to choose from the appropriate array can be selected for different tests. A smaller spread of microphones may be more appropriate for high frequency tests and a selection spanning a larger diameter would be more appropriate for lower frequency tests. Selecting microphones in line with regions of interest would also prove beneficial, as line of sight is a requirement for successful acoustic source location. This is not to say that the array would be limited to only one of these two versions, the possibility exists to select arrays from microphones which may be offset from the centre or to select two (or more) fewer microphone arrays from the field of available arrays. Many variations exist, centrally arranged arrays, or offset ones with their centres located under areas of interest such as swept back wings, tail sections or forward landing gear. Co-ordinates for any new array design could be made available to create the main plate, and any new array design would need to be validated using the Co-array and Point Spread Function as before, see section 3.4.

The initial Array Design was modified to fit around the wind tunnel structure and as such caused problems with results from the Point spread function. Any newer array would need to also fit within the wind tunnel structure and as any new array design is likely to be recessed under taut material, most likely Kevlar as it is proven strong enough to avoid flapping in 80 m/s winds yet porous and thin enough to not interfere in sound reception. Design concerns would include an appropriately strong frame for the material and an appropriate recess level for the microphone array. Finally, a fixing mechanism for the material cover would be required. In order to finalise any design for another Filton Array, the new array would require testing and debugging time.

In the lower frequency region it would probably be necessary to increase the physical size of the array, as having a larger array would most likely improve the resolution of the low frequency results. Another concept for array design would be to have a large array for low frequency measurements with a smaller section inside which would be used independently to measure the high frequency levels. For all these cases further study is required in determining the optimum microphone spacing and location for the best results and resolution at the various frequencies of interest. The practicalities of installation and effect on overall wind tunnel performance must be fully examined before implementation of any of these concepts.

6.1.3. Limitations

A list with regards to the limitations of the equipment and the processing (and thus the potential errors is included in the following section, as well as where care should be taken in interpreting the results.

Beginning with array itself, the main limitations is evident from the array design being interrupted by the bars in the wind tunnel floor. Future array designs should hopefully be a complete array, and thus give results with less interference from sidelobes. The array size is also limited by the physical constraints in the wind tunnel. Larger arrays should, in theory, produce better resolution results in the lower frequency domain.

Another limitation in the design is the number of channels was limited to 112 due to the number of acquisition cards available in a single chassis. It is generally agreed that the more microphones in an array the better the results [2 and 4]. A higher number of channels and microphones should produce a higher resolution result plot even at the lower frequencies.

Another issue with the microphone array is the limitations on the frequency range available with the current level of technology. Despite the inherent problems the array functioned as well as could be expected although low frequency results were less useful, with results below 3 kHz being effectively useless in determining sound sources.

During the setup of the array a number of problems were encountered which merit future consideration. With only 112 channels it took three and a half days to install; this time included, however, the pistonphone calibration to check each microphone was still functional. Full channel calibration takes a long time and would be easier to do before microphones are permanently installed. Once calibration for each channel is completed microphones should not need to be calibrated for each subsequent test. Assuming the microphones are fixed into the array and the same pre-amps and cables are used for each test then calibration would only be required at regular long intervals as the calibration used for a previous test would be the same.

In particular regard to the cabling; the only limitation is the time required for assembly and the convenience of assembly. The vast number of channels and connections meant that the likelihood of a loose connection or broken link was driven high. Increasing the number of connections and individual cables increases the chances of a channel failure or error. Appendix II contains some image of the cabling in situ.

Care has to be taken to isolate each channel from any interference or signal noise, some interference was identified from the balance equipment below the wind tunnel test section. As the cables and the microphone were close to the balance equipment, this was turned off during the testing. This has implications if a simultaneous sound mapping and force measurement (using the balance) is required.

Although implemented in the Matlab code post-processing code is a channel checking feature which identifies any channels which may have produced an erroneous or unusual signal, this is at post-processing stage and as such after the acquisition has taken place. Therefore it would be wise (as was done) to run a microphone check directly after set up, this was done during the first test runs and whilst the model was being installed as post-processing and channel checking and repair is time consuming.

Regarding limitations on the plots produced it is vital that any users are made aware that only the same frequencies can be compared for levels. That is to say that the scale on the plots is only equivalent for same frequency plots on different runs. It should also be made clear that none of the plots show absolute levels and indeed the levels are only relative to each other (in the same frequency plots).

Validation of the data is an ongoing process and constant improvements are being made. Data delivery is determined at the time by whatever the needs are. The July 2007 test showed that the data could be processed in any number of required formats and future integration with the Wind tunnels systems should allow database interaction and finally combining with other wind tunnel systems Cross-comparison between flow visualisation, PIV, and other techniques will become an aim for the future development of the system and this study.

The existing wind tunnel facility was not designed with acoustic testing in mind. Nevertheless the test results shows that noise sources from inflow models can be determined and visualised effectively and successfully. The following is a summary of the main possible actions that aim to improve the performance of the tunnel in terms of aeroacoustic measurement capability.

- Identification and mitigation of aeroacoustic sources and minimize effects – de-correlate 2D vortex shedding flow
- Identification of the 1 – 2.5 kHz noise, investigating motor as likely source
- Sound proofing/dampening of the motor
- Investigating splitter designs to attenuate background noise, and predict performance impact on tunnel operations
- Optimising microphone recessing and optimising microphone coverings in accordance with the conclusions from section 6.2
- Possibility of lining solutions for the test section (especially directly opposite the microphone array)
- Identifying and eliminating local sound sources, like small protrusions and discontinuities – this is potentially a test by test solution
- Isolate frequencies that are particularly problematic – map them to avoid in future testing

6.2. Conclusions for Recessing (Section 4.2)

A requirement for this MPhil is that a published paper needs to have been written, to this end section 4.2 forms the basis of such a paper, which was presented at the AIAA Conference in January 2009, looking at the effect of physical structures of microphone mounted in arrays to the effectiveness of data received over a variety of frequencies.

At the University of Southampton an attempt was made to determine the effect of how recessing affects the effectiveness of microphone used in a hard-walled wind tunnel at various frequencies and wind tunnel speeds. As part of the continued effort to improve the microphone arrays ability to acquire data in the wind tunnel environment; further study was done on recessing design and the workings therein of recessing an acoustic array.

Evidence suggests that recessing an array can have up to a 20 dB improvement. Recessing a single microphone at the acoustic tests at the Filton site also showed some improvements. It was considered a necessity to perform a recessing test to fully determine first-hand the advantages and the levels of improvement which could be achieved in the wind tunnel environment.

A recessing test where a series of microphones were recessed to determine the effect on the background noise was performed. The test, performed in the University of Southampton's (3 by 2) wind tunnel to compare: 'In flow' microphones against 'Cavity recessed' and 'Small section recessed' microphones was done and furthermore compared to flush mounted microphones. Different Kevlar materials were also tested in this set of experiments (as well as a silk covering), along with four distinct microphone apertures. The investigation was performed at varying wind velocities with a sound point source consisting of a signal set to either white noise or different tonal frequencies

A series of microphones were recessed to determine background noise effect on data acquisition, with additional test to be done to determine the best level for recessing;

with a view to quantify how much of a recess should be allowed for a given test environment and compare the results to flush mounted microphone in the same environment. 50 and 95 gm/M^2 type Kevlar material was used for these tests.

Different levels of recessing were tested and it was determined that a 5 mm recess (*d*) proved the most consistently reliable in reducing background noise, when compared to 10 or 20 mm (2*d* or 4*d*). Various microphone apertures were also tested and from the results it can be seen that the 45 degree aperture seemed to fare the worst. From this it can be speculated that the geometry of the aperture might prove more important than first realised. In addition an analysis into the effect of different materials were qualified, and from the results it was clear that the Kevlar material proved as good if not better than silk material and two types of Kevlar material were equal in reducing the effect of background noise.

In summary, from the results there appeared to be benefits gained from recessing, the following points were determined:

- A 5 mm recess (*d*) proved the most consistently reliable in reducing background noise, when compared to 10 or 20 mm (2*d* or 4*d*).

- The Kevlar material proved as good if not better than silk material and the two types of Kevlar material were equally good in reducing the effect of background noise.

- It can also be seen that the 45 degree aperture, one of the various microphone apertures that were tested, seemed to perform badly in comparison to others.

From this it can be speculated that the geometry of the aperture might prove more important than first realised and continued study of the data is required, and repeating the experiments, with modifications would prove useful. Supplementary study will be required to further validate these results, as well as complementary research to determine the specific values for microphone configurations from which the best methods for acoustic acquisition at a given frequency can be ascertained for use in the future. Paper [67] subsequently corroborated these results.

6.3. Conclusions for Directivity (Section 5)

A directivity acoustic test was conducted in the FLSWT between 23rd March and April 4th 2009 where a beamforming array was installed in ceiling in three locations and comparisons of sources in range and at edges of ranges were made. The effect of a known source at or near the array arc limit was studied and the effective range of the array was determined to be approximately within 1.5 times the diameter of the array from the array centre.

A number of plots revealed distortion in the extreme edge of the array, where circular sources were expected, and distorted images were generated in the results.

Additionally it was proven that the source strength from wind tunnel models is dependent on directionality and the same source viewed from different angles produced differences in excess of 10 db.

Reflections and false sources were also identified within the plots and the emphasis was reinforced that care must be taken in distinguishing and categorizing so called “noise sources” in acoustic plots.

6.3.1. Recommended Future Work

Correction of the distorted sources at the edges of the array scan capability should be achievable by determining the major and minor axis of the oval shaped sources and applying an algorithm directly to the beamforming code to give correct source locations and strength levels.

Further work could be done to find the range at which different layout arrays are viable for source location and to compare the error in sources at the extremes of the range of those array in order to improve the aeroacoustic measurement technologies in wind tunnels for the future.

7. References

[1] Patricio A. Ravetta, Ricardo A. Burdisso and Wing F. Ng, Wind Tunnel Aeroacoustic Measurements of a 26%-scale 777 Main Landing Gear Model, In AIAA volume 2004-2885, 2004

[2] Alan S. Hersh, Paul T. Soderman, and Richard E. Hayden. Investigation of acoustic effects of leading edge serrations on aerofoils. (Journal of Aircraft), 11(4): 197 - 202, April 1974.

[3] Marinus M. Boone, Niels Kinneging and Teun van den Dool, Two-Dimensional noise source imaging with a T-shaped microphone cross array, Journal Acoustical Society of America 108, December 2000

[4] National Instruments Newsletter, Boeing Measure Noise Emissions with LabVIEW and PXI, Second Quarter 2006

[5] Stefan Oerlemans and Peter Sijtsma, Determination of Absolute Levels from Phased Array Measurements using spatial Source coherence. AIAA 2002-2464

[6] Lars Koop, Klaus Ehrenfried and Stefan Krober. Investigation of the Systematic phase mismatch in microphone-array analysis. AIAA 2005-2962.

[7] Thomas F. Brooks and William M. Humphreys Jr., Flap-edge aeroacoustic measurements and predictions, Journal of Sound and Vibration 261 (2003) 31-74, 2000 - 2002

[8] Thomas F. Brooks and William M. Humphreys Jr., Effect of Directional Array Size on the measurement of airframe noise components. In AIAA, Vol. 99-1958, 1999.

[9] William M. Humphreys Jr., Carl H. Gerhold, Allan J. Zuckerwar, Gregory C. Herring, and Scott M. Bartram. Performance analysis of a cost-effective electret condenser microphone directional array. In AIAA, volume 2003-3195, 2003.

[10] William M. Humphreys Jr., Thomas F. Brooks, William W. Hunter Jr., and Kristine R. Meadows. Design and use of microphone directional arrays for aeroacoustic measurements. In AIAA, volume 98-0471, January 1998.

[11] W. Dobrzynski, L. C. Chow, P. Guion and D.Shiells. A European Study On Landing Gear Airframe noise sources. In AIAA, volume 2000-1971, 2000.

[12] Edward G. Duell, Mark Perry, and John Callister. Influence of an acoustic liner on wind tunnel background noise levels. In AIAA, volume 97-1680-CP, pages 710-718, 1997.

[13] Gerhard Wickern and Norbert Lindener. The Audi Aeroacoustic Wind Tunnel: Final Design and First Operational Experience. SAE 2000 World Congress Detroit, Michigan. SAE Technical Paper Series. 2000-01-0868, March 6-9 2000.

[14] M.G Smith, B. Fenech, L. C. Chow, N. Molin, W. Debrzynski, and C. Seror. Control of noise sources on aircraft landing gear bogies. AAIA, 2006-2626, 2006.

[15] Nathan J. Burnside, Stephen M. Jaeger, Bryan R. Reinero, W.C. Horne and Paul T. Soderman. Array design and performance for a large scale airframe noise study. In AIAA, volume 2002-2576, 2002.

[16] Nobuhiro Yamazaki, Kiyoshi Nagakura, Mitsuru Ikeda and Akio Sagawa. Methods to Measure Acoustic Sources in a Wind Tunnel Test Section. AIAA 2005-3003.

[17] P. O. A. L. Davies. The New 7 1/2 ft. and 15 x 12 ft. Low Speed Wind Tunnel at the University of Southampton. Technical Report 202, A.A.S.U., December 1961.

[18] Paul T. Soderman, Fassi Kafyeke Nathan J. Burnside, Reuben Chandrasekharan, Stephen M. Jaeger, Josèe Boudreau. Airframe Noise study of a CRJ- 700 Aircraft Model in the NASA Ames 7 by 10 Foot Wind Tunnel No. 1. AIAA 2002- 2406

[19] D. F. Long. Acoustic Source Location in Wind Tunnel Tests via Subspace Beamforming. 2004.

[20] William M. Humphreys, Jr., Qamar A. Shams, Sharon S. Graves, Bradley S. Sealey, Scott M. Bartram, and Toby Comeaux, Application of MEMS Microphone Array Technology to Airframe Noise Measurements, AIAA-2005-3004, May 2005

[21] S. M. Jaeger, W. C. Horne, and C. S. Allen. Effect of surface treatment on array microphone self-noise. In AIAA volume 2000-1937, 2002.

[22] R. A. Gramann and J. W. Mocio. Aeroacoustic Measurements in Wind Tunnels using Adaptive Beamforming Methods. Journal of the Acoustical Society of America, 97:3694-3701, June 1995

[23] Marianne Mosher, Michael E. Watts, Michael Barnes, and Jorge Bardina. Microphone Array Processing System (MAPPS): Phased array system for acoustic measurements in a Wind Tunnel. In (SAE AIAA), volume 1999-01-5576, October 1999.

[24] B.A. Fenech, Aeroacoustic Measurements – Transfer Thesis. SES, University of Southampton, 2006.

[25] T. J. Mueller, editor. Christopher S. Allen, William K. Blake, Robert P. Dougherty, Denis Lynch, Paul T. Soderman, and James R. Underbrink. Aeroacoustic Measurements. Springer, 2002.

[26] Paul T. Soderman and Christopher S. Allen. (Aeroacoustic Measurements), Chapter 1. Microphone Measurements In and Out of Airstream. Springer, 2002.

[27] Robert P. Dogherty. (Aeroacoustic Measurements), chapter 2. Beamforming in Acoustic Testing. Spinger, 2002.

[28] James R. Underbrink. (Aeroacoustic Measurements), Chapter 3. Aeroacoustic Phased Array Testing in Low Speed Wind Tunnels. Springer, 2002.

[29] W.L Willshire Jr., F. Farassat, Progress Toward Quieter Aircraft, NASA Langley Research, Vehicle System Technologies "The Science of Flight", Office of Aerospace Technology,

[30] Stefan Oerlemans and Beatriz Mendez Lopez. Acoustic Array Measurements on a full scale Wind Turbine. AIAA 2005-2963.

[31] Thomas F. Brooks and William M. Humphreys, Jr. Three Dimensional Application of DAMAS Methodology for Aeroacoustic Noise Source Definition. AIAA 2005-2960. May 2005

[32] Eugene Weinstein, Kenneth Steele, Anant Agarwal , and James Glass, LOUD: A 1020-Node Modular Microphone Array and Beamformer for Intelligent Computing Spaces, MIT/LCS Technical Memo MIT-LCS-TM-642. April 2004.

[33] Allan D. Pierce. Acoustics - An Introduction to Its Physical Principles and Applications. Acoustical Society of America, 1994.

[34] David P. Arnold, Toshikazu Nishida, Louis N. Cattafesta III, and Mark Sheplak. MEMS-based acoustic array technology. In AIAA, volume 2002-0253. University of Florida, January 2002.

[35] Dom H. Johnson and Dan E. Dudgeon. Array signal processing: concept and techniques. P T R Prentice Hall, 1993.

[36] Georges Elias. Source localization with a two-dimensional focused array: Optimal signal processing for cross-shaped array, volume 2, page 1175, 1995. Compilation and indexing terms, Copyright 2004 Elsevier Engineering Information, Inc.

[37] I. A. McCowan. Robust Speech Recognition using Microphone Arrays, PhD Thesis, Queensland University of Technology, Australia, 2001

[38] Ivan Tashev, Gain Self-Calibration Procedure for Microphone Arrays, Microsoft Research

[39] Jim Ross, PMAT - Phased Microphone Array Technology, Quiet Aircraft Technology Planning Workshop, NASA Ames Research Center, April 10-11, 2000

[40] Joshua M. Sachar, Harvey F. Silverman and William R. Patterson, Microphone Position and Gain Calibration for a Large-Aperture Microphone Array, IEEE Transactions on Speech and Audio Processing, volume 13, no. 1, January 2005

[41] K .R. Holland and PA Nelson. Sound Source Characterisation: The Focussed Beamformer vs the Inverse Method. In (Tenth International Congress on Sound and Vibration), July 2003.

[42] P .A. Nelson and SH Yoon. Estimation of Acoustic Source Strength by Inverse Methods. Part i, Conditioning of the Inverse problem. (Journal of Sound and Vibration), 233(4):643-668, 2000.

[43] Robert P. Dougherty. Extensions of DAMAS and Benefits and Limitations of Deconvolution in Beamforming. AIAA 2005-2961, May 2005

[44] Thomas F. Brooks and William M. Humphreys jr., A Deconvolution Approach for the Mapping Acoustic Sources (DAMAS) Determined from Phased Microphone Arrays. In AIAA, volume 2004-2954, May 2004.

[45] Y. Kim and P. A. Nelson, Spatial resolution limits for the reconstruction of acoustic source strength by inverse methods, Journal of Sound and Vibration 265 (2003) 583-608, 2001 - 2002

[46] Stanley T. Birchfield and Amarnag Subramanya, Microphone Array Position Calibration by Basis-Point Classical Multidimensional Scaling, IEEE Transactions on Speech and Audio Processing, 2004

[47] Prof. J. G. McWhirter, Developments in Sensor Array Signal Processing, Asiolmar conference Presentation Notes, QuinetiQ Ltd, Malvern, 2003

[48] C. H. K. Williamson. Vortex dynamics in the cylinder wake. (Annual Review of Fluid Mechanics), 28:477-539, 1996.

[49] J. f. Piet and G. Elias. Airframe Noise Source Localization using a Microphone Array. In AIAA, volume 97-1643-CP, 1997.

[50] Sazzadur Chowhury, M. Ahmadi, G. A. Jullien, and W. C. Miller. A Surface Mountable MEMS Beamforming Microphone Array and Associated MEMS Socket Structure. Symposium on Microelectronics Research & Development in Canada, June 2001.

[51] Pieter Sijtsma. Experimental Techniques for Identification and Characterisation of Noise Sources. National Aerospace Laboratory NLR, 2004.

[52] Peter Brandstatt and Helmut V. Fuchs and Manfred Roller. Novel silencers and absorbers for wind tunnels and acoustic test cells. Noise Control Engineering Journal, 50(2):41-49, 2002.

[53] Marianne Mosher, Michael E. Watts, Srba Jovic, and Stephen M. Jaeger. Calibration of microphone arrays for phased processing. In AIAA, volume 97-7678-CP, pages 693-701, 1997.

[54] Keith Holland and Phillip Nelson. Array Processing in Semi-Reverberant Spaces. In 5th International Workshop on Microphone Array Systems, Erlangen, Germany 2003.

[55] S. Oerlemans and P. Sijtsma, Acoustic array measurements of a 1:10.6 scaled Airbus A340 model, National Aerospace Laboratory NLR, AIAA paper 2004-2924, 10-12 May 2004 and August 2004.

[56] Sijtsma, P., CLEAN based on Spatial Source Coherence, AIAA 2007-3436. 2007

[57] Joe C. Chen, Kung Yao, Ralph E. Hudson, Acoustic Source Localization and Beamforming: Theory and Practice, Received 17 February 2002 and in revised form 21 September 2002

[58] William M. Humphreys, Jr., Carl H. Gerhold, Allan J. Zuckerwar, Gregory C. Herring, Scott M. Bartram, Performance Analysis of a Cost-Effective Electret Condenser Microphone Directional Array, AIAA 2003-3195, 9th AIAA/CEAS Aeroacoustics Conference & Exhibit 12-14 May 2003

[59] S.M. Jaeger, W.C. Horne, and C.S. Allen. Effect of surface treatment on array microphone self-noise. AAIA, 2000-1937, 2000.

[60] Benjamin A. Fenech and Kenji Takeda Frequency Response Analysis of Microphone Preamplifiers in the Audible and Ultrasonic Regime, Southampton, UK, University of Southampton, 19pp., School of Engineering Sciences, Aerospace Engineering AFM Technical Reports, AFM 07/02, University of Southampton, AFM Tech Report, January 2007

[61] Fenech, Benjamin A. and Takeda, K (2007), SotonArray: Southampton University wind tunnel microphone array system guide. Southampton, UK, University of Southampton, 35pp, School of Engineering Sciences, Aerospace Engineering AFM Technical Reports, AFM 07/03.

[62] Alexander Carballo-Crespo, Kenji Takeda, Benjamin Fenech and Koen Boorsma, Filton Wind Tunnel Acoustic Testing Feasibility Study For Airbus UK, University of Southampton, School of Engineering Sciences, August 2006

[63] N.J. Burnside, S.M. Jaeger, B.R. Reinero, W.C. Horne and P.T. Soderman. Array design and performance for a large scale airframe noise study. AIAA, 2002-2576, 2002.

[64] J. Hileman, B. Thurow, M. Samimy, Development and evaluation of a 3-D microphone array to locate individual acoustic sources in a high-speed jet, Journal of Sound and Vibration 276 (2004) 649–669

[65] Bidirectional Woven Kevlar from Aircraft Spruce http://www.aircraftspruce.com/catalog/cmpages/bikevlar.php

[66] Hi-Tech Materials http://www.freeflightsupplies.co.uk/hitech.htm

[67] Vincent Fleury, Laurent Coste, Renaud Davy, Andre Mignosi, Jean-Marc Prosper, Charles Cariou, Optimization of Microphone Array Wall-Mountings in Closed-Section Wind Tunnels, AIAA paper 2010-3738, 2010

[68] M. C. Remillieux,1 E. D. Crede, H. E. Camargo, R. A. Burdisso,W. J. Devenport, M. Rasnick, P. Van Seeters, and A. Chou, “Calibration and demonstration of the new Virgina Tech Anechoic Wind Tunnel”, AIAA 2008-2911, 2008

[69] F. Faly and J. Walker, "Advanced Applications in Acoustics, Noise and Vibration, chapter 3."Source identification and location by P.A. Nelson, pages pp.100-153. Spon Press, 2004.

[70] Benjamin A. Fenech, Accurate Aeroacoustic Measurements in Closed-Section Hard-Walled Wind Tunnels. SES, University of Southampton, 2009.

[71] C. Brauer S. Guidati and S. Wagner. The reflection canceller – phased array measurements in a reverberating environment. AIAA, 2002-2462, 2002.

[72] P. Sijtsma and H. Holthusen. Corrections for mirror sources in phased array processing techniques. AIAA, 2003-3196, 2003

[73] Thomas F. Brooks and William M. Humphreys, Jr. A deconvolution approach for the mapping of acoustic sources (DAMAS) determined from phased microphone arrays. AIAA, 2004-2954, 2004

[74] J. A. Högbom. Aperture synthesis with a non-regular distribution of interferometer baselines. Astronomy and Astrophysics Supplement, 15:417-426, 1974

Publications

Papers Published by the Author:

Alexander Carballo-Crespo and Kenji Takeda. An Investigation of Microphone Array Installation Effects. AIAA, 2009-883, 2009

8. Appendix

8.1. Appendix I: Project Milestones

The following section summarises the progress over the first twenty-four months, with a description of the academic progress and a separate account of the practical experience gained.

Summary of Results of Works Completed in Year 1 and 2

Academic Progress

- Year 1 - 7 Modules (3 Management, 4 Technical) – 4 in Semester 1, 3 in Semester 2.
    - Various Labs (Advanced Measurement Techniques)
    - Short Course in Aeroacoustics
- Year 2 – 3 Modules (2 Management and 1 Technical)
- Generic Skills Modules – 4 weeks worth, not including coursework

- One month plan

- Mini-Literature Review

- Presentation for University Website
- Presentation for Skills day
- Presentations for Viva and end of year reports

- Gantt Chart for Years 1, 2 and proposed 3 and 4 completed

- Abstract for postgraduate day at Chilworth Manor, 2006 and 2007
- Poster for postgraduate day at Chilworth Manor, 2006 and 2007

8.2. Appendix II: Research Progress

- Wind tunnel time (as Observer) - Landing Gear Experiments
    - Familiarising usage of the Array

- Microphone Calibration for Koen experiment
    - Familiarising with Array components

- Wind Tunnel Time – 7 by 5, Koen
    - Installation and Dismantling of Array for 7 by 5

- Microphone Calibration for Andrew Wells experiment
    - Familiarisation with Labview program and code

- Wind Tunnel Time – 11 by 8, Andrew Wells Experiment
    - New Array Set-up for 11 by 8

- Post Processing for Andrew Wells
    - Familiarising with Matlab processing code
    - Several weeks of post processing data – over 1000 data sets

- Airbus Visits and tests
    - Acoustic Feasibility test
    - November 2006 and July 2007 acoustic array tests
    - Study into point source

- Wind Tunnel Time – 11 by 8, Andrew Wells Experiment
    - New Array Set-up for 11 by 8

- Additional Wind Tunnel Time – 7 by 5,
    - Continuation of use of Array for 7 by 5

- Matlab Graphical User Interface

8.3. Appendix III: Time-scales

The majority of the first year tasks from the following pages were achieved in the first 12 months. Although the time scales for the individual tasks were not adhered to exactly. The Airbus acoustic survey and subsequent array designs were delayed due to restricted access to the Filton wind tunnel at times. This is to be expected from an industrial and commercial point of view and such delays are unfortunately unavoidable. Conversely most of the second year tasks were completed in the time frame as it was required by the industrial timetable. The delays in the acoustic survey led to subsequent delays in the development of the microphone array. Nonetheless preliminary designs were made and materials required for such a construct were sourced.

Updating the software and developing the software code was achieved by performing additional experiments using the 7 by 5 and 11 by 8 wind tunnel arrays. Indeed the construction of a second 7 by 5 wind tunnel microphone array was not initially part of the plan, the second array was constructed by the University of Southampton to allow further acoustic test to be performed and the opportunity was taken to perform additional software development during these tests, as well as developing construction techniques for the array itself. Subsequent arrays being built using various different materials and structural forms. The construction of the three arrays at the University of Southampton has led to knowledge being acquired in how arrays should be constructed and the methods best employed in constructing and maintaining them. The operation of the array designed for use at Filton has led to further array modification and plans to be developed.

The literature survey is a continuous task as there are constantly new techniques and experiments being run around the world; this is especially true of the aero-acoustic measurement field of research. Although the number of facilities is limited, the research is ongoing. The technical and management modules which form part of this research were completed on time and on schedule.

The following pages show the updated Gantt charts showing the progress as of the twelve and twenty-four month point as well as a proposed thirty-six month plan.

Time-scale for the first twelve months

- **Literature Survey.** **October 2005 – December 2005**
    - A look at current wind tunnel measurement techniques and technologies.
- **7 x 5 Wind tunnel tests.** **December 2005 - Onwards**
    - Using the current array in the Southampton wind tunnel.
- **Technical and Management Module Exams** **January 2006**
    - Required Technical and Management module exams.
- **Microphone array equipment development.** **February 2006**
    - Modifying the current microphone array technology.
- **Microphone array software development.** **February 2006**
    - Developing the interface for the microphone array.
- **Airbus preliminary acoustic survey and tests.** **March 2006**
    - Analysing the Filton Airbus wind tunnel for initial design specification and limitations. Perform test in Filton tunnel as required.
- **Microphone array designs for Airbus wind tunnel.** **July 2006**
    - Designing the working array that will be installed and used in the Filton Airbus UK site.

Time-scale for the second twelve months

- **Continued Literature Survey.** **October 2006 – December 2007**

    - A look at current wind tunnel measurement techniques and technologies.

- **Airbus first acoustic tests.** **November 2006**

    - Performed acoustic tests in Filton wind tunnel as required and analysing the Filton Airbus wind tunnel acoustic results.

- **University of Southampton tunnel tests.** **December 2006 - Onwards**

    - Continued use of the current array in the Southampton wind tunnels for further technological development.

- **Technical and Management Module Exams** **January 2007**

    - Final Technical and Management modules and exams.

- **Microphone array software and hardware development.** **February 2007**

    - Modifying the current microphone array technology and developing the interface for the microphone array.

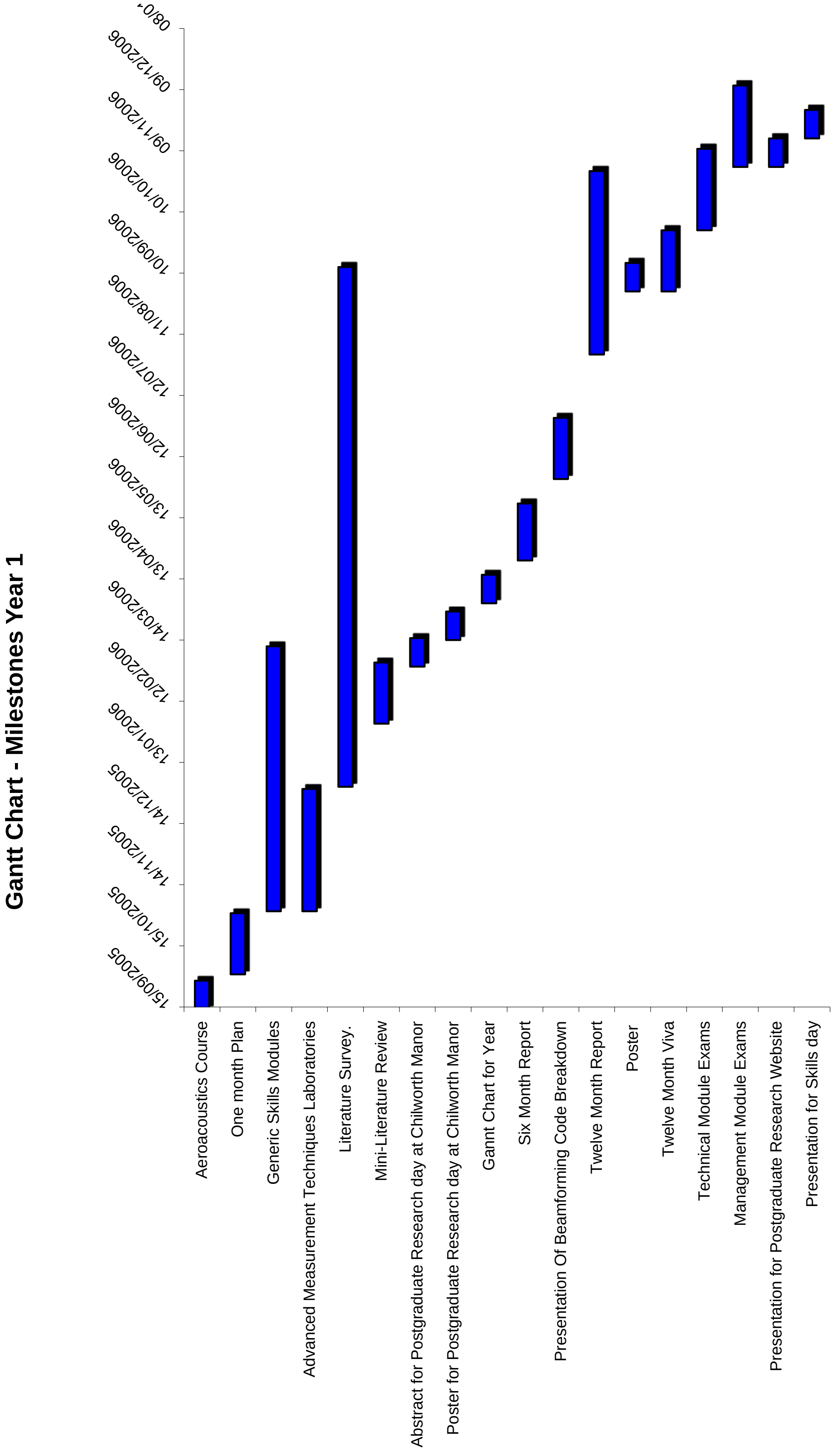
- **Modified array design for Airbus wind tunnel.** **July 2007**

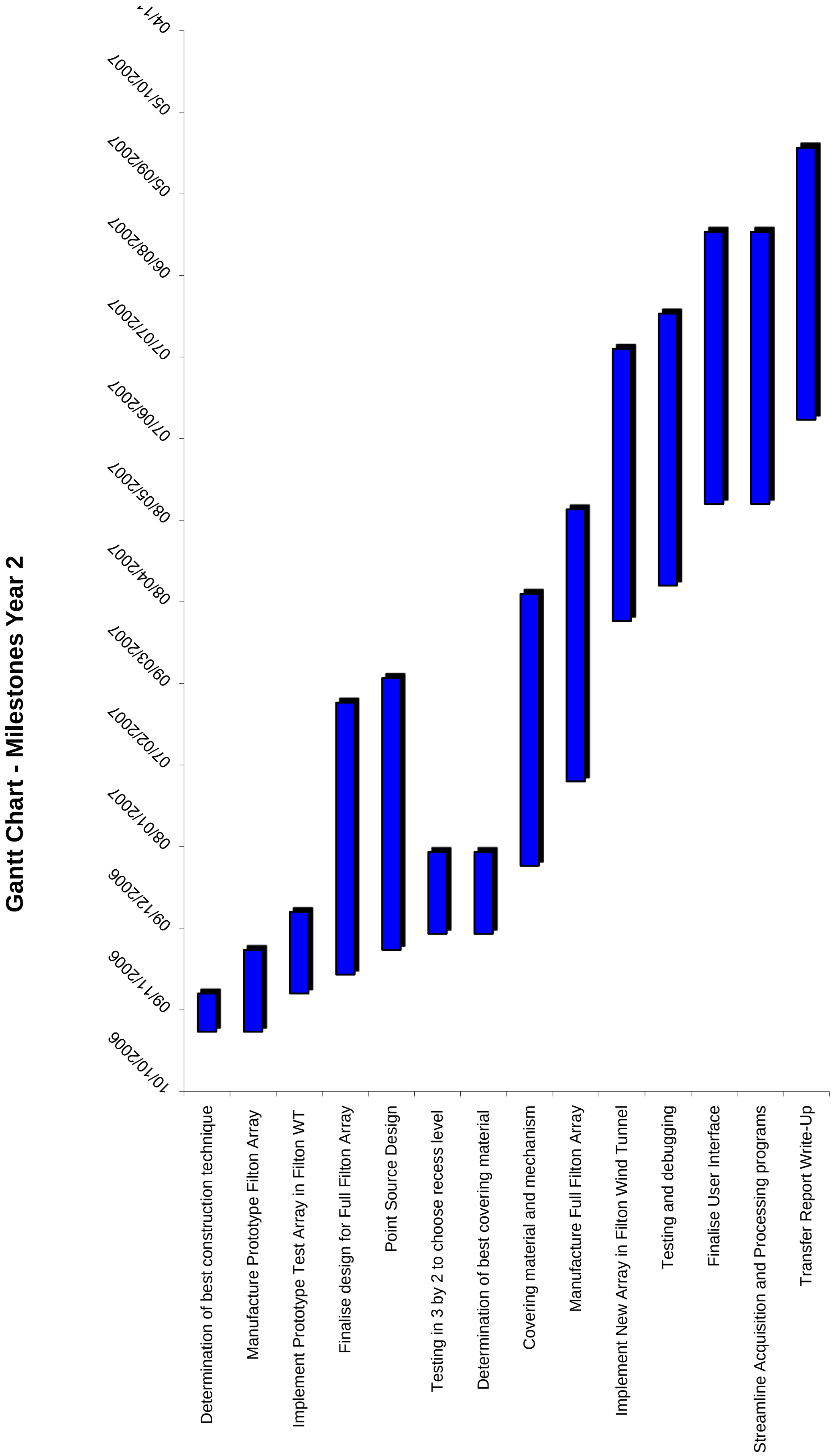
    - Second use of the microphone array in the Filton wind tunnel at the Airbus UK site, with modified processing techniques and extended test program.

- **3 by 2 wind tunnel microphone recessing technique test.** **August 2007**

8.4. Appendix IV: Gantt Charts

The milestone Gantt charts for the three years.




Gantt Chart - Milestones Year 2
Determination of best construction technique
Manufacture Prototype Filton Array
Implement Prototype Test Array in Filton WT
Finalise design for Full Filton Array
Point Source Design
Testing in 3 by 2 to choose recess level
Determination of best covering material
Covering material and mechanism
Manufacture Full Filton Array
Implement New Array in Filton Wind Tunnel
Testing and debugging
Finalise User Interface
Streamline Acquisition and Processing programs
Transfer Report Write-Up
10/10/2006
09/11/2006
09/12/2006
08/01/2007
07/02/2007
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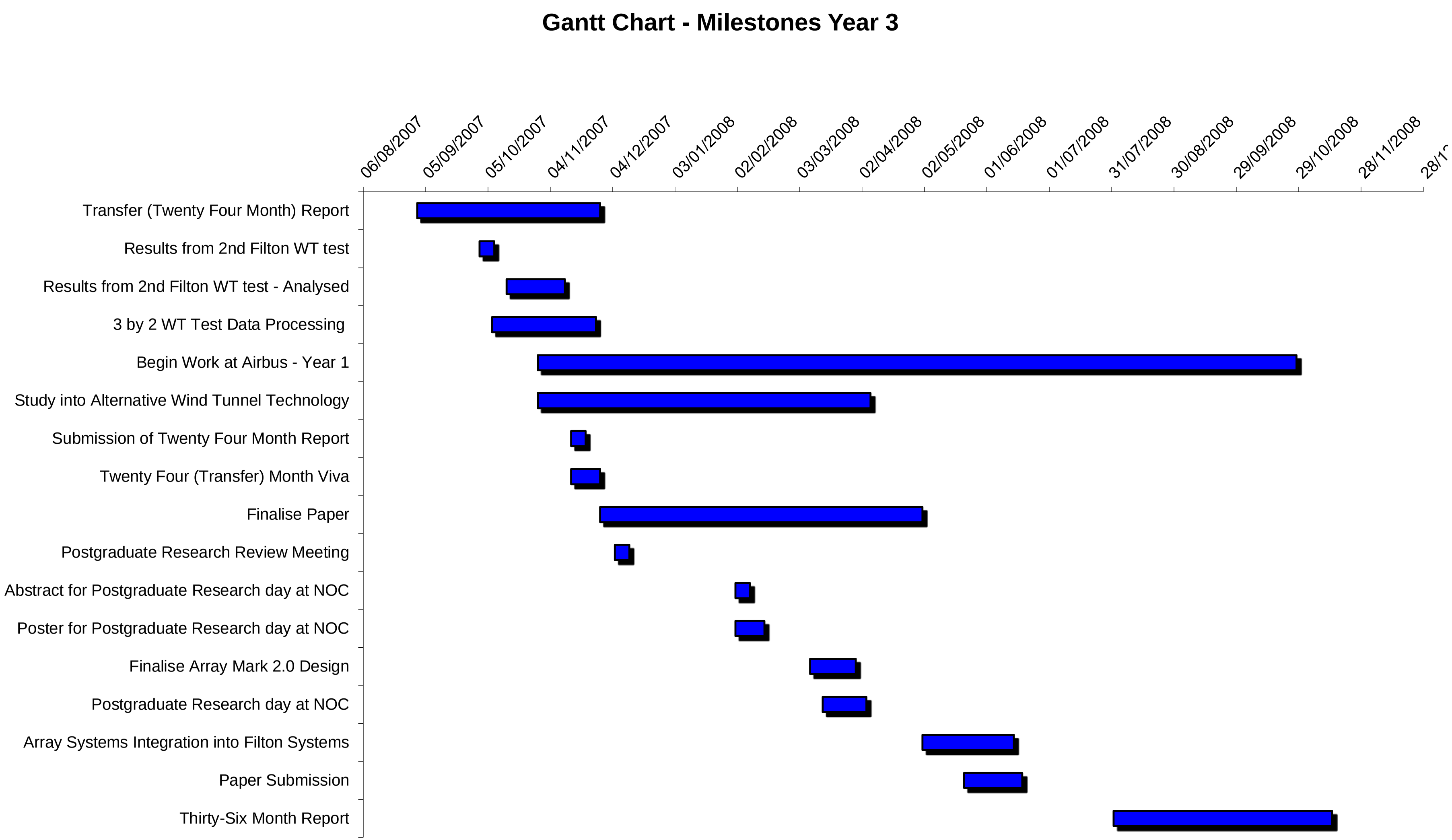
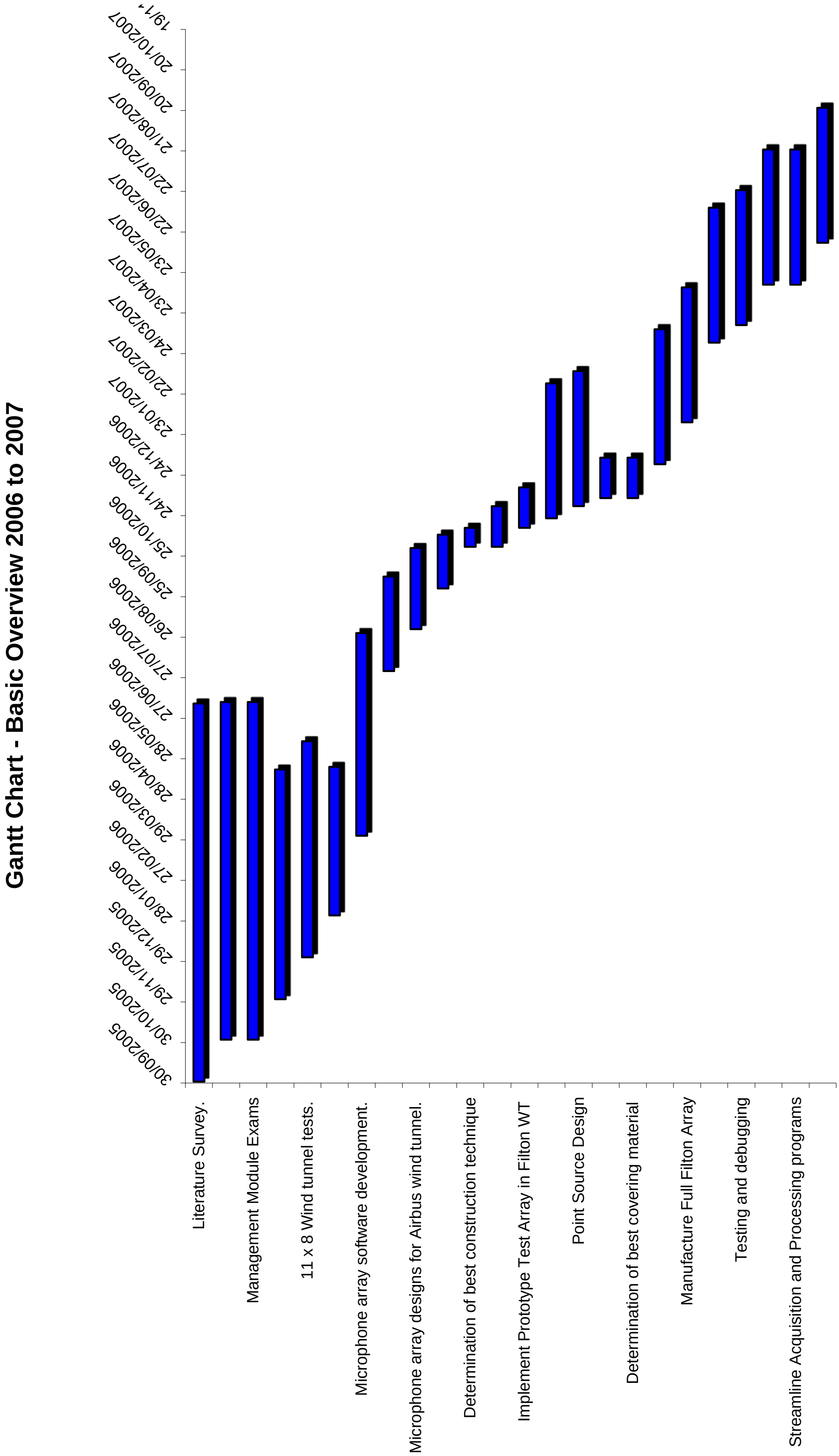
05/09/2007
05/10/2007

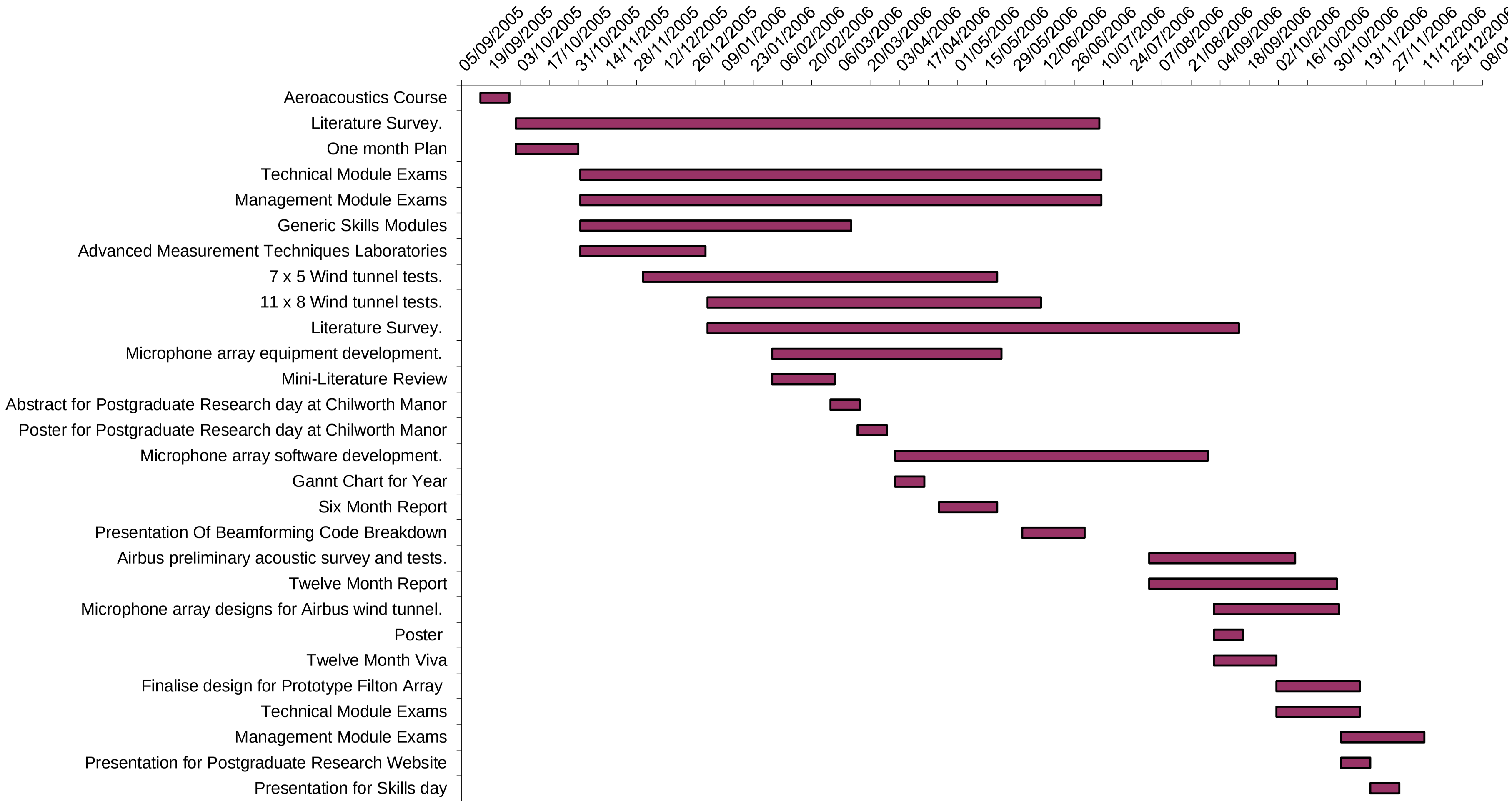

Gantt Chart - Milestones Year 3
06/08/2007
05/09/2007
05/10/2007
04/11/2007
04/12/2007
03/01/2008
02/02/2008
03/03/2008
02/04/2008
02/05/2008
01/06/2008
01/07/2008
31/07/2008
30/08/2008
29/09/2008
29/10/2008
28/11/2008
Transfer (Twenty Four Month) Report
Results from 2nd Filton WT test
Results from 2nd Filton WT test - Analysed
3 by 2 WT Test Data Processing
Begin Work at Airbus - Year 1
Study into Alternative Wind Tunnel Technology
Submission of Twenty Four Month Report
Twenty Four (Transfer) Month Viva
Finalise Paper
Postgraduate Research Review Meeting
Abstract for Postgraduate Research day at NOC
Poster for Postgraduate Research day at NOC
Finalise Array Mark 2.0 Design
Postgraduate Research day at NOC
Array Systems Integration into Filton Systems
Paper Submission
Thirty-Six Month Report

Additional Gantt Charts

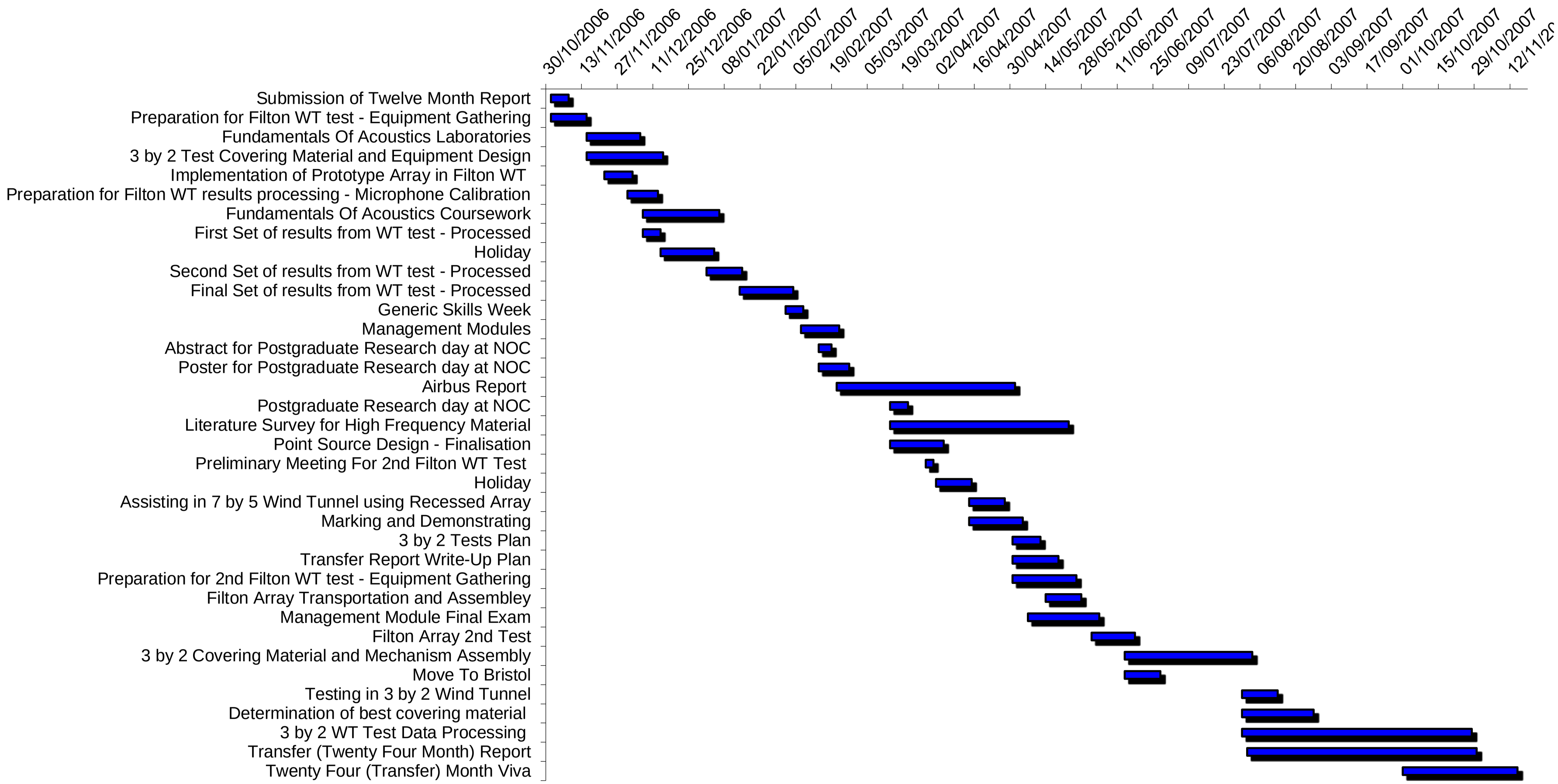
Overview Gantt charts up to the end of the third year.

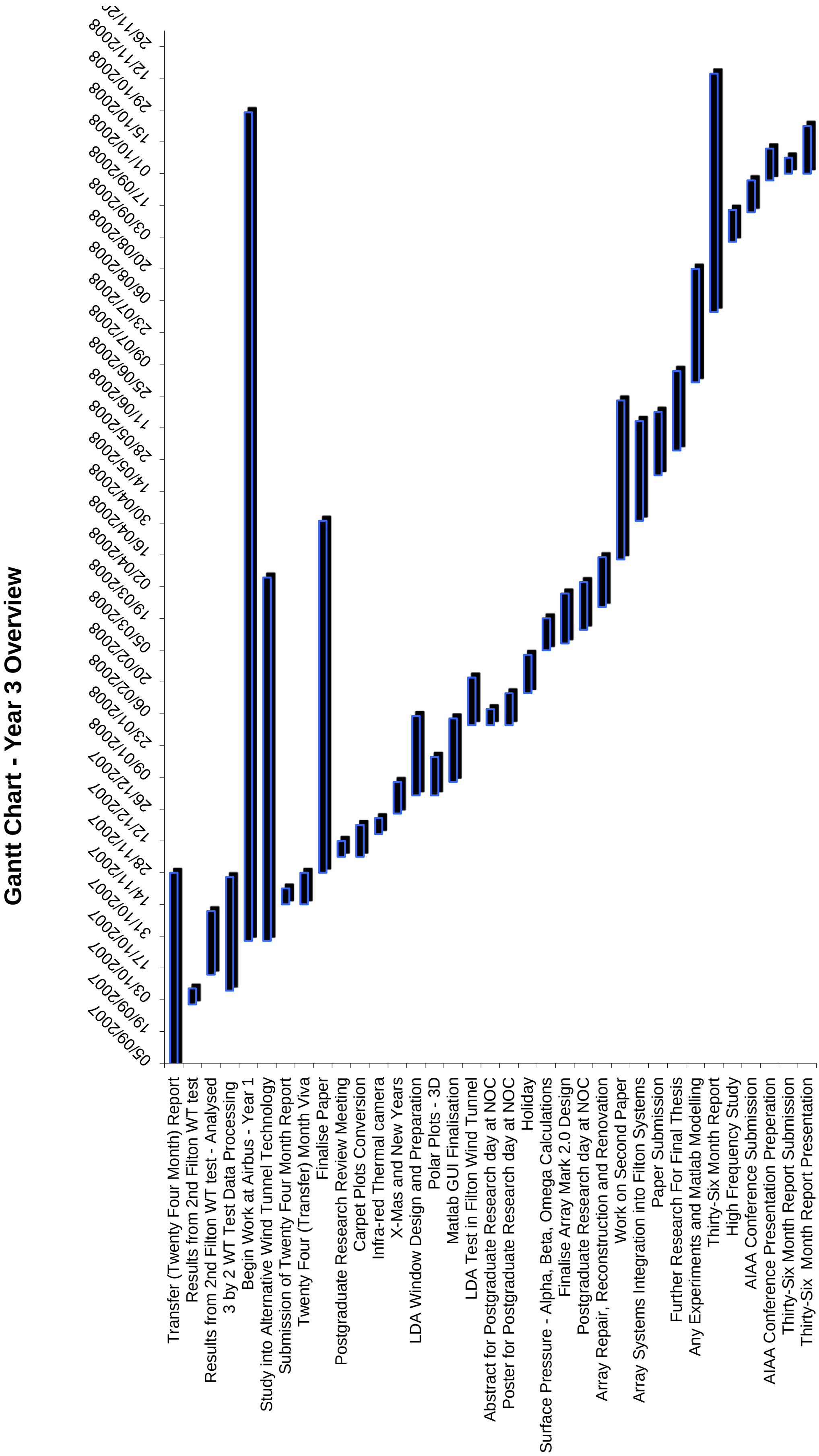


Gantt Chart - Year 1 Overview



Gantt Chart - Year 2 Overview




Gantt Chart - Year 3 Overview
Transfer (Twenty Four Month) Report
Results from 2nd Filton WT test
Results from 2nd Filton WT test - Analysed
3 by 2 WT Test Data Processing
Begin Work at Airbus - Year 1
Study into Alternative Wind Tunnel Technology
Submission of Twenty Four Month Report
Twenty Four (Transfer) Month Viva
Finalise Paper
Postgraduate Research Review Meeting
Carpet Plots Conversion
Infra-red Thermal camera
X-Mas and New Years
LDA Window Design and Preparation
Polar Plots - 3D
Matlab GUI Finalisation
LDA Test in Filton Wind Tunnel
Abstract for Postgraduate Research day at NOC
Poster for Postgraduate Research day at NOC
Holiday
Surface Pressure - Alpha, Beta, Omega Calculations
Finalise Array Mark 2.0 Design
Postgraduate Research day at NOC
Array Repair, Reconstruction and Renovation
Work on Second Paper
Array Systems Integration into Filton Systems
Paper Submission
Further Research For Final Thesis
Any Experiments and Matlab Modelling
Thirty-Six Month Report
High Frequency Study
AIAA Conference Submission
AIAA Conference Presentation Preperation
Thirty-Six Month Report Submission
Thirty-Six Month Report Presentation
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