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Beamforming Using Scattering Conformal Microphone Arrays

By

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A thesis submitted to

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in partial fulfilment of

the requirements for the degree of

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Abstract

This thesis investigates the use of a scattering conformal array of sensors (with no mutual coupling). It is shown that the use of the scattering body and judicious use of absorption permit one to obtain an array that can operate, with relatively controlled side lobes, up to twice the commonly accepted maximum frequency.

The consideration of asymmetrical arrays leads to the development of a novel index that quantifies the amount of asymmetry in the main beam of an array.

A simple solution to correcting main beam asymmetry is also provided by the use of expressing a linear constraint in an innovative way. This reduces the number of constraints by two.

The simulation method advocated is validated by the extensive measurements of prototypes. This also illustrates clearly that scattering objects create phase non-linearities in a pressure field.

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List of Notations

a	- radius of a circular or spherical array (m)
c	- velocity of sound (assumed to be 340 m/s in air at 20°C)
$\mathbf{d}$	- transfer function between a source and the array
$\mathbf{d}_s$	- transfer function between the desired source and the array (look direction)
d_i	- transfer function at a particular frequency between a source and element i of the array
f_c	- cut-off frequency of a filter (-3dB frequency)
$f_n(\theta, \phi)$	- element directionality
f_{Schr}	- Schroeder frequency
g	- gain associated with constraint matrix $\mathbf{C}$
j_η	- Bessel function in spherical co-ordinates
k	- wave number (ω/c)
l_a	- length of x axis of an ellipse
l_b	- length of y axis of an ellipse
n_η	- Neumann function in spherical co-ordinates
p	- sound pressure
r	- radial distance (m)
s	- distance between elements in an array
s_i	- distance to the i^{th} element
u	- dimensionless circular array size factor (Eq. 3-20)
$\mathbf{w}_n$	- element weighting function (illumination)
$\mathbf{w}_{opt}$	- optimal weighting at one frequency
$\mathbf{A}^H$	- is the Hermitian of matrix $\mathbf{A}$ (complex conjugate transposition)
A	- total surface area

AI	- Asymmetry Index (defined in 6.2)
B_m	- spherical scattering amplitude (Eq. 3-23)
C	- constraint matrix
D	-directivity (defined in 2.1.2)
DI	- Directivity Index (defined in 2.1.4)
E{x}	- expected value of random variable x
FBR	- front to back ratio (defined in 2.1.6)
$F(\theta, \phi)$	- output response of an array
G	- gain of array
J_n	- ordinary Bessel function
L	- length of a linear array
K	- measurement system noise
N	- number of elements in array
$\bar{P}$	- acoustic power
P₀	- sound pressure amplitude
P_m	- Legendre function
Q	- directivity factor (defined in 2.1.3)
Q₀	- directivity of a source
Q_m	- directivity of a microphone
R	- distance between the source and the array
R_{rc}	- room constant
RT₆₀	- Reverberation Time (normalised to 60dB) (seconds)
S	- source amplitude
U	- Interfering noise amplitude
V	- volume (m ³)
W	- weighting matrix for array elements
X	- matrix of element responses
Y	- output signal from an array
α	- absorption coefficient
$\bar{\alpha}$	- average absorption coefficient

β_i	- Normalised time delay in a differential microphone (eq. 2-9)
ε	- eccentricity of an ellipse
δ	- spherical scattering phase (Eq. 3-24)
ϕ	- angle of rotation (azimuth) in spherical co-ordinates
γ_i	- value of linear constraint I
η_i	- System noise associated with element i of the array
κ	- solutions to maximisation of signal to noise ratio of an array
λ	- wave length (m)
μ	- small factor to whiten an array
v_i	- transfer function at a particular frequency between a interfering noise source and element i of the array
θ	- angle of elevation in spherical co-ordinates
ρ	- density of fluid
σ_n^2	- mean value of system noise
τ_i	- time delay to the i^{th} element
ω	- angular frequency
ψ	- dimensionless linear array size factor (Eq. 3-17)
Γ	- Correlation of inputs to an array
Γ_{vv}	- Correlation of interfering noise inputs to an array (noise matrix)

List of Abbreviations and Acronyms

ABS – Acrylonitrile Butadiene Styrene (thermoplastic resin used for moulded plastics)

A/D – Analogue to digital

ANSI – American National Standards Institute

DSP – Digital Signal Processor

ERP – Ear Reference Position

FIR – Finite Impulse Response

FFT – Fast Fourier Transform

GSM – Global System for Mobile communications

HATS – Head and Torso Simulator

IIR – Infinite Impulse Response

ITU-T –International Telecommunication Union – Telecom Standardization

MDF – Medium Density Fiberboard

MIPS – Million Instructions Per Second

MOS – Mean Opinion Score

MRP – Mouth Reference position

MVDR – Minimum-Variance Distortionless Response

SPL – Sound Pressure Level (usually in dB re 20 μ Pa)

WAV – Wave file standard

Chapter 1

Introduction

1.1 Thesis objectives

Microphone arrays have started making their appearance in commercial products such as highly directional microphones (Andrea)[1], hearing aid devices (Widrow)[2] and conference systems (Mitel)[3]. In most cases the products try to house the microphones minimally so that the free-field assumption holds. Exceptionally Mitel has chosen to use a scattering object to enhance the performance of a microphone array. This has also been reported by Stinson and Ryan [4], Anciant [5], Elko [6] and Myers [7]. In all of these an axi-symmetric array is used to permit uniform beam patterns to be steered in the plane of interest.

If one wishes to use microphone arrays in other situations (e.g. a typical telephone) then the industrial design precludes the use of an axi-symmetric array. The directional pattern of such an array would be significantly asymmetrical thus making their use impractical.

The high frequency limitation of the critical frequency (beyond which spatial aliasing occurs) means that to obtain an extended frequency range the number of transducers must be increased. In some cases some spatial aliasing is acceptable especially in confined areas (Ryan)[8] but this is rather exceptional. In commercial products this increase in cost

is generally unacceptable. The added complexity of the system can also create problems with a large number of inputs (Silverman)[9].

1.2 Problem statement

The thesis will primarily address the following two questions:

How to obtain reasonably consistent and symmetrical directivity patterns in three-dimensions from a conformal microphone array that is not necessarily axi-symmetric and embedded in a scattering structure. (Kummer defines a conformal array as one whose elements are flush mounted on a non-planar surface [10].)

How to extend the frequency range of microphone arrays embedded in a scattering structure beyond the spatial aliasing frequency while maintaining a reasonable directivity pattern.

The first question is similar to the one that Ryan [8] addresses in his thesis except that we are now dealing with arrays embedded in a scattering structure and the array, or the structure, or possibly both, are asymmetrical. These will produce asymmetrical directivity patterns using conventional techniques. As in Ryan's thesis we are looking at fixed beamforming.

The second question is rather novel in that we are trying to go beyond a generally accepted limitation of microphone arrays. In the free-field case there is no solution but in

exceptional cases one can accept these limitations (Ryan)[8]. In the embedded case the acoustical design of the scattering structure could provide solutions.

The major challenge is to finding a solution that solves both of these questions simultaneously.

1.3 Contributions to knowledge

This thesis contributes to the body of knowledge on beamforming and more specifically to microphone arrays. Though the specific application is speech acquisition, many of the results can be applied to other sensor arrays such as SONAR.

Two publications resulted from this work [11,12].

The fundamental contribution of this work is to show the benefits of exploiting the physical acoustics (scattering) of the housing of a microphone array to enhance the beamformer performance. Specifically:

1. Use of a scattering conformal array to overcome the spatial aliasing that is found in free-field arrays [11].
2. Developing a novel index quantifying the asymmetry of the main lobe in a beamformer [12].
3. Using a simple linear constraint within the calculation of the optimised beamformer weighting to correct an asymmetrical beam shape [12].

4. Validating a simulation technique by computing two significantly varied shapes and experimentally validating the results [11,12].
5. Synthesising beamforming theory and the effect of acoustical scattering. This also involves simulation of known equations to illustrate these concepts.

1.4 Thesis organisation

Chapter 1 describes the problem that this thesis addresses. It briefly explains the need for a constant beam width in both frequency and azimuth.

Chapter 2 provides background material on monaural speech capture in rooms. It starts with definitions, as many are not the same in the acoustics and DSP communities. A quick review of the basic acoustics of speech in a room is followed by a review of the traditional approaches to speech capture in rooms.

Chapter 3 is a synthesis of the basic mathematics of array processing followed by a brief discussion of the behaviour of free-field arrays. The scattering problem in its simplest form, about a sphere, is presented and it concludes by justifying the numerical approach to the calculation of the scattered sound field in the vicinity of a reasonably complex solid object.

Chapter 4 describes the simulation and measurement environment used. The major emphasis is on the measurement system as this is the part that requires the most detail for one to reproduce the results. The measurement system also highlights the need for careful design of a measurement system.

Chapter 5 introduces the first complex scattering conformal array studied. It is symmetrical and nicely illustrates how scattering object can help an array of sensors overcome the spatial aliasing that occurs in free-field arrays. The benefit of a good understanding of the physical phenomena is illustrated by the beneficial use of absorption to further improve the directivity of the scattering body. The chapter concludes by showing how significant signal processing benefits can be derived by the combination of acoustical design and digital signal processing.

Chapter 6 studies asymmetrical arrays. It starts by considering a free-field elliptical array on a reflecting plane. A novel index to quantify the asymmetry of the main beam is then introduced. An asymmetrical scattering body that is telephone like is then used to illustrate how one can, by the use of a simple linear constraint, obtain reasonably symmetrical beamshapes.

Conclusions are presented in Chapter 7.

Chapter 2

Background and Current Solutions

2.1 Definitions

The purpose of this section is to define terms as they will be used in this document. Certain terms such as beam width have several definitions so it is important to clearly define them. The primary reference used is ANSI S1.1 Definition of Acoustical terms [13]. (Other definitions will appear later and are rather specific so they will only be indicated by the use of bold italics.)

2.1.1 Spherical co-ordinates

Often spherical co-ordinates will be used for formulae or pattern descriptions. The hybrid spherical and Cartesian co-ordinate system adopted is that commonly used for antennae and is illustrated in figure 2.1.

Figure 2.1 Spherical Co-ordinate system

2.1.2 Directivity (D)

This term is used in many ways so it is important to define it clearly. In spherical co-ordinates: $p(\theta, \phi)$ is the sound pressure response for a plane wave in direction (θ, ϕ) [13]. Generally θ_0 and ϕ_0 are chosen to be the direction from which the peak power is received. In that case it is the directivity factor (Q). In electromagnetic antennas this does not include dissipative losses, as this is what gain is. Gain is often used interchangeably but this is not strictly correct.

2.1.3 Directivity factor (Q)

The directivity factor is the ratio of the intensity on axis of a radiator at a stated distance r to the intensity that would be produced by a monopole radiating the same power at the same position. For a receiver it is the ratio of the power received in an isotropic field (diffuse) to that, which would be received by an onmi-directional receiver [14].

$$Q = \frac{4\pi |p(\phi_0, \theta_0)|^2}{\int_0^{2\pi} \int_0^\pi |p(\phi, \theta)|^2 \sin \phi d\phi d\theta} \quad \text{Eq. 2-1}$$

2.1.4 Directivity Index (DI)

The directivity index is used extensively for radiators and receivers of wave energy. The definition is simply: $10 \log_{10} (Q)$ where Q is the directivity factor. [13].

2.1.5 Beam width

The formal definition is: "*At a specified frequency, in a specified plane including the beam axis, the angle included between the two directions, one to the left and the other to the right of the axis, at which the angular deviation loss has a specified value. Unit, degree*" [10]. In this thesis the beam width will always be taken at the half magnitude points which means -3dB for power and -6dB for pressure.

2.1.6 Front to back ratio (FBR)

This is a figure of merit to quantify the ability of the array to discriminate between signals that arrive from the front plane (the hemisphere centred about the look direction of the array) versus those from the rear plane (the hemisphere centred about 180° from the look direction.). The ratio (in dB) for a directional receiver aimed at $\phi=0$ and $\theta=\pi/2$ is defined as [15]:

$$\text{FBR}(\omega) = 10 \log \left\{ \frac{\int_0^{2\pi} \int_0^{\pi/2} |F(\omega, \phi, \theta)|^2 \sin \phi \, d\phi \, d\theta}{\int_0^{2\pi} \int_{\pi/2}^{\pi} |F(\omega, \phi, \theta)|^2 \sin \phi \, d\phi \, d\theta} \right\} \quad \text{Eq. 2-2}$$

2.1.7 Illumination

Illumination is a term that comes from radio-frequency arrays and refers to the weighting of the elements.

2.1.8 Grating Lobes

A grating lobe is a lobe that has the same (or greater) amplitude as the main lobe in the direction of interest. This is also often called spatial aliasing.

2.1.9 Conformal Array

Kummer's definition is an array whose elements are flush mounted on a non-planar surface [10].

2.1.10 Scattering

The term scattering is applied to the phenomena of sound spreading out in all directions as it encounters an obstacle or inhomogeneity in an otherwise homogeneous medium. This term is meant to distinguish itself from diffraction, which is used for situations where the object is large enough, with respect to the wavelength, so that ray-acoustic approximations are valid [16,17].

2.1.11 Endfire

Endfire is a beam steered on the axis of a linear array (i.e. if the array is along the x axis the endfire beam is steered in the x direction).

2.1.12 Broadside

Broadside is a beam steered perpendicular to the axis of a linear array (i.e. if the array is along the x axis the broadside beam is steered in the y direction).

2.2 Monaural speech in real rooms

Monaural speech capture is used in nearly all real-time communication systems (e.g., telephone, radio communication, video-conferencing). Unfortunately, this removes much information that humans with binaural hearing use in reverberant and noisy environments. We therefore must have some understanding of the physics of this phenomenon to maximise the efficiency of such systems.

2.2.1 Monaural speech systems

We first consider the source directivity, as it is a very significant factor in determining the performance of the sound capture system. The directivity of the human voice has been studied in the past. The earliest measurements were those of Dunn and Farnworth (1939)[18] and these are still often used today. They are the basis of the directivity generally used in telephony for artificial mouths [19] and for Head And Torso Simulators (HATS) [20]. A recent study by Warnock and Chu [21] provides results that are in good agreement with the HATS implementation. In our work we will therefore use these results as well as an artificial voice conforming to ITU-T p.51 [19] as a measurement device. In simulations the voice is modelled as a monopole as it has uniform directivity in the front plane [18].

We now need to examine the subjective reaction to sound in a room and understand the subjective effects of reverberation and noise.

In a speech communication system signal to noise ratio is probably the most critical criteria. Speech intelligibility deteriorates rapidly when the signal to noise ratio is less than 10 dB [22]. To achieve highly intelligible speech it is generally agreed that the signal to noise ratio must be greater than 25 dB [23].

Reverberation has received little attention in general sound systems [24]. However, in monaural systems such as telephones it has a significant deleterious effect. Subjective studies have been conducted to understand the annoyance due to echoes in such systems [25]. The results are not surprising in that they indicate that the level and time delay of the echo have a reasonably log-linear relationship. The annoyance threshold is also fairly sharp.

In telephony there does not seem to be any study that has quantified the amount of room reverberation that is acceptable. However, the requirement for acoustic echo cancellers in speakerphone applications illustrates the importance of controlling reverberation. In ITU-T recommendations, the level of reverberation is recommended to be reduced to less than 30dB below the signal [26,27].

The frequency response of transmitted sound affects our perception of the quality of the sound. For a telephone to sound right we expect a band limited (300 - 3400 Hz) frequency response such as illustrated in Figure 2.2 [28]. Certain music has even certain microphones associated with it such as a Shure 508 is associated with a Blues harmonica [29]. Off-camera comments sound quality is due to the strange frequency response of the side lobes of a shot gun microphone [30]. In many cases, meeting these expectations

increases the perception of quality but interchanging these will illicit annoyance. For example, a large band width flat-frequency response system will not be acceptable for a telephone instrument. These perceptions of course can change over time - one simply has to listen to a 1960's $33\frac{1}{3}$ "hi-fi" recording to understand how perception of quality changes.

Figure 2 - Transmit Frequency Response
(See Clause 4.2.1.3.1)

Figure 2.2 Telephone transmit frequency response limits

It is conceivable that one will accept an improvement in sound quality (i.e. flatter frequency response or increase in signal to noise ratio). However, significant degradation is rarely tolerated unless there is another mitigating factor (e.g. a portable telephone has convenience that obviously overcomes the degradation in sound quality. (typical telephone quality is MOS of 4.5 or higher, G.S.M. with G.729 coding can only achieve 3.9 at best) [31,32].

2.2.2 Basics of room acoustics

In a situation where several persons are in the same room the expectation is that the sound quality will be the same from every person. Generally in situations where this is of importance such as courts, legislative bodies, international conferences, etc. each participant is provided with a microphone [33]. If one is to use less than one microphone per participant it is important to ensure that the signal to noise and frequency response from each participant remains as constant as possible.

The noise, reverberation and sound coloration are affected to a great extent by the acoustics of the room. The most common model of room acoustics is that of Sabine's theory of reverberant rooms [14,16]. The fundamental assumption is that the room consists of a diffuse field implying spatial uniformity. In such a room any sound source decays following a simple exponential decay and *reverberation time* can easily be calculated as:

$$RT_{60} = \frac{0.161V}{\sum \alpha A} \quad \text{Eq. 2-3}$$

where V = volume α = absorption coefficient and A = surface area

This simple formula does not take into account the absorption of air nor the non-uniform distribution of absorbing surfaces within the room. Norris-Eyring formula and those proposed by Embleton [34] consider such factors but these are beyond the scope of this thesis as Sabine's original formula gives a reasonable first approximation. The validity of the diffuse sound model is generally bound by a limit proposed by Schroeder

[35]. Below the **Schroeder frequency** one must consider the modal behaviour of the room.

$$f_{Schr} = c \sqrt{\frac{6}{A_s}} = \sqrt{\frac{c^3}{4 \ln(10)}} \sqrt{\frac{RT_{60}}{V}} \quad \text{Eq. 2-4}$$

In typical conference rooms this is less than 200Hz so we can ignore the modal behaviour of such rooms in the telephony frequency band (300-3400Hz).

2.2.3 Directivity effect of source and receiver

What is really of interest to us in the use of a microphone in a room is the effect of reverberation on the sound pick-up. Assuming a diffuse reverberation field one obtains the following [14,16]:

$$\bar{p}^2 = \rho c \bar{P} \left(\frac{Q_0}{4\pi r^2} + \frac{4}{R_{rc}} \right) \quad \text{Eq. 2-5}$$

where $\bar{P}$ = acoustic power of source; ρ = density of fluid;

c = velocity of sound; Q_0 = directivity factor of the source;

r = distance from source;

$$R_{rc} = \text{room constant} = \frac{\bar{\alpha}A}{1 - \bar{\alpha}}$$

where $\bar{\alpha}$ = average absorption; A = total surface area

Implicit in this formula is that the detector of the acoustic pressure is omni-directional.

If one is using a directional microphone pointed at the source then this effect is simply the product of the source and microphone directivity factor Q_m . The distance from the source at which the direct sound equals the reverberant sound is thus:

$$r = \sqrt{\frac{R_{rc} Q_0 Q_m}{16\pi}}. \quad \text{Eq. 2-6}$$

This is often called the **critical distance** or the distance factor.

In broadcasting, it is generally accepted that one would want the direct field to be 25dB louder than the reverberant field. Using this criterion the microphone must be at most the following distance away:

$$r = \frac{1}{17.8} \sqrt{\frac{R_{rc} Q_0 Q_m}{16\pi}} \quad \text{Eq. 2-7}$$

Using the Sabine room acoustic model the solution to capturing speech with the least reverberation, the highest signal to noise ratio and the least coloration is to place a microphone as close to the mouth as possible. To understand the scale it useful to consider a typical conference room 8m by 5m with a 2.4m high ceiling. Typically one would have drywall walls, carpeted flooring and a fissured mineral tile ceiling. To simplify consider only the effect at 500Hz (where typical RT_{60} values are quoted).

Table 2-1 Area and absorption of example room

Surface	Area (A)in m ²	Absorption coefficient (α)	αA
Floor	40.0	0.15	6.0
Walls	62.4	0.10	6.2
Ceiling	40.0	0.65	26.0
Total	142.4		38.2

$$\bar{\alpha} = \frac{38.2}{142.4} = 0.268 \quad 1 - \bar{\alpha} = 0.731 \quad R_{rc} = \frac{38.2}{0.731} = 52.3$$

Assuming an omni-directional source and receiver the spacing between these must be less than 57mm in order to have less than 25dB of reverberation. Using a microphone with a $Q_m=4$ makes the distance 114mm and with a $Q_m=9$ one gets a reasonable distance of 171mm.

This explains why in speech reinforcement systems the talker either comes close to a fixed microphone or has a wireless microphone placed close to the mouth.

However, placing a microphone that close may not be sufficient and in many instances a directional microphone may still be required. Typically most "vocal" microphones have a super-cardioid response (such as illustrated in figure 2.3 [36]) which maximises front to back ratio (FBR) [33].

Figure 2.3 Super- cardioid microphone polar response (Sennheiser MKH 40)

In some applications, it is not possible to place the microphone close to the talker, such as in filmmaking. In those cases, highly directional microphones are used to capture the

sound. The idea is that only the sound directly on axis of the microphone is captured and little of the ambient noise or reverberation. This requires the boom microphone operator to always point the microphone in the appropriate direction. When the desired signal is off axis it is strongly "coloured" by the large fluctuations in frequency response due to the significant side lobes of the "shot-gun" microphones used. The beam pattern for a commonly used high quality microphone (Neumann KMR82i) is illustrated in figure 2.4. Note that the beam width narrows quite considerably in the high frequencies [30].

Figure 2.4 Polar response of a "shot-gun" microphone (Neumann KMR 82i)

In many situations this is too directional so a “short shot-gun” microphone is often used in speech capture. A good example of this is the AKG C568 [37] and its polar response is illustrated in figure 2.5.

Figure 2.5 Polar response of a "short shotgun" microphone (AKG C568B)

2.3 Physical realisations of directional microphones

In this section realisations of directional microphones used for remote sound capture are examined.

2.3.1 Differential microphones

The differential microphone pair or array is the most common directional microphone. (This explanation is a summary of Elko's excellent chapter on differential microphones [15].) The fundamental assumption used is that we are dealing with plane wave propagation and omni-directional (0th order) elements. The element spacing must be $s < \lambda/4$ over the range of frequencies of interest to ensure that the on axis error is less than 1 dB ($ks \ll \pi$). This introduces a small time delay such that $\omega T \ll \pi$. A generalised “canonical” form for Nth order differential elements is:

$$F_n(\omega, \theta) \approx P_0 \omega^N \prod_{i=1}^N [\beta_i + (1 - \beta_i) \cos \theta] \quad \text{Eq. 2-8}$$

where ω - angular frequency; θ - angle of incidence;

P_0 - plane wave amplitude; N - order of the array

$$\beta_i = \frac{\tau_i}{\tau_i + s_i/c} \quad \text{Eq. 2-9}$$

τ_i - time delay between elements; c - speed of sound;

s_i - distance between elements

From this using a Taylor series expansion, normalising the terms, and assuming there is a filter to correct ω^n and we get:

$$F_{N_n}(\theta) \approx \prod_{i=1}^n [\beta_i + (1 - \beta_i) \cos \theta] \quad \text{Eq. 2-10}$$

The obvious conclusion of this is that we can easily get a wide variety of shapes simply by varying the time delay τ_i given an appropriate spacing s_i . This array gives uniform directivity with frequency. Obviously the major drawback to this type of array is that the microphones must all be closely match in phase (otherwise this affects τ_i) and the analysis is based on plane wave assumptions. There are four patterns that are commonly achieved with a first-order gradient microphone pair. They are summarised in table and their relative benefits are obvious.

Table 2-2 Typical first-order differential microphones

Microphone Type	β_i	DI(dB)	FBR(dB)	Beamwidth	Null(s)
Dipole	0	4.8	0	90°	90°, 270°
Cardioid	1/2	4.8	8.5	131°	180°
Hyper- cardioid	1/4	6.0	8.5	105°	109°, 251°
Super- cardioid	$\frac{1}{1 + \sqrt{3}}$	5.7	11.4	115°	125°, 235°

In the microphone placement discussion of section 2.2.3 the Q_m factor was used to describe the directivity factor of a microphone. For an array of sensors the directivity factor can be written as

$$Q = \frac{\mathbf{w}^H \mathbf{d} \mathbf{w}}{\mathbf{w}^H \mathbf{\Gamma} \mathbf{w}} \quad \text{Eq. 2-11}$$

where $\mathbf{w}^H \mathbf{d} \mathbf{w}$ is the pressure response of the array to a source in the “look direction”. $\mathbf{w}$ is the complex weight applied to the elements and $\mathbf{w}^H \mathbf{\Gamma} \mathbf{w}$ is the “noise response” of the array to the plane waves emanating uniformly from all angles. To maximise Q one can solve the above and get:

$$\mathbf{W}_{opt} = \mathbf{\Gamma}^{-1} \mathbf{d} \quad \text{Eq. 2-12}$$

This is the “difference” field and it is a reasonable assumption in diffusely reverberant rooms. It simply states that the probability of noise arriving from any direction is equal. The assumption here is that the desired source has a reasonable directivity and is not so far away from the microphone array as to be diffuse. The converse is true of “noise” sources.

For N closely spaced omni-directional microphones the maximum $Q = N^2$. This is in keeping with classical acoustics where the Q of the source is the square of monopoles. [14]. Elko shows that the maximum directivity of differential pairs is actually $(N + 1)^2$ (at least up to order 4) [15].

2.3.2 Interference tube microphones

Another class of microphones that are often used to achieve high directivity are the interference tube microphones also known as "shot gun" microphones. The polar response of two actual implementations of these is shown in figures 2.4 and 2.5. The basic idea here is to design a tube that will physically provide destructive interference to any sound arriving off axis and there is a pressure sensitive capsule at the closed end of the tube. The basic theory is that of a line array whose response is:

$$F(\theta) \approx \left| \frac{\sin\left(\frac{1}{2}kL \sin \theta\right)}{\frac{1}{2}kL \sin \theta} \right| = \left| \text{sinc}\left(\frac{1}{2}kL \sin \theta\right) \right| \quad \text{Eq. 2-13}$$

where $k = \frac{\omega}{c}$; $L = \text{length of line}$

The major challenge is to provide this interference over a wide frequency bandwidth. To achieve this, very elaborate mechanical systems are used [38].

The solution that has become more of interest recently is the use of an array of discrete microphones. Delay and sum line arrays provide similar performance to that of an interference tube microphone, are easier to construct and are more flexible. As alluded to above, there are superdirective arrays that can provide better directionality.

Chapter 3

Microphone arrays - Current approaches

Firstly a review of the mathematics of beamforming of discrete arrays will be presented as well as the optimisation techniques commonly used. This will be followed by a discussion of linear and circular free field arrays. The spherical scatterer is then presented and simulations illustrate important effects on arrays.

3.1 Mathematical expression of the problem

To understand the basic mathematical description of the problem and possible mathematical solutions, assume all the necessary information is available

This discussion assumes a monochromatic plane wave and, though the wave motion is time harmonic, these time dependencies are omitted from the notation for clarity. All the quantities considered are treated as complex quantities (amplitude and phase) to enable as general a discussion as possible. Obviously there are many special cases where simplifications lead to the use of real scalars. The notation is that scalars are normal font, vectors are lower case bold and matrices are upper case bold. The following is a summary of more detailed analysis found in texts [39,40,41]. Mankolosis [39] is used for the first part followed by Bitzer [40] and Herbordt [41].

Consider the general system diagram of figure 3.1

Figure 3.1 System level diagram of a digital beamformer

In it we find a desired source signal S . Waves emanate from it and travel to the array of sensors via different paths which modify the original signal by a transfer function d_i . There is an interfering noise source U (or sources) whose transfer function is designated by v_i . Its effect, as indicated here, is a contribution to the signal. The array of sensors is not perfect so there is some system noise that enters the system and this can vary from one sensor to the other. This is designated as K with transfer function η_i and is assumed to be white noise with random phase. The sum of these is weighted by a coefficient w_i and the sum of all these results in the output of the beamformer Y .

$$Y = \mathbf{W}^H \mathbf{X} \quad \text{Eq. 3-1}$$

where

$$X_i = Sd_i + Uv_i + K\eta_i \quad \text{Eq. 3-2}$$

If we assume all noise sources are mutually uncorrelated and that system noise is spatially uncorrelated (white noise) we can get a correlation (co-variance) matrix.

$$\mathbf{\Gamma}_x = E\left\{\mathbf{x}(n)\mathbf{x}^H(n)\right\} = M\sigma_s^2\mathbf{d}_s^H\mathbf{d}_s + \mathbf{\Gamma}_{vv} + \mathbf{\Gamma}_{\eta\eta} \quad \text{Eq. 3-3}$$

$$\mathbf{\Gamma}_{v+\eta} = \mathbf{\Gamma}_{vv} + \sigma_\eta^2\mathbf{I} \quad \text{Eq. 3-4}$$

To simplify we will assume that system noise is negligible ($\sigma_\eta^2 = 0$) to concentrate on the spatial performance of the systems. The object of optimal beam formers is to maximise the signal to noise (or reduction of the interference) ratio.

$$S/N \mathbf{\Gamma}_{out} = \frac{|\mathbf{w}^H \mathbf{s}(n)|^2}{E\left\{|\mathbf{w}^H \mathbf{x}_v(n)|^2\right\}} = \frac{N\sigma_s^2|\mathbf{w}^H \mathbf{d}_s|^2}{\mathbf{w}^H \mathbf{\Gamma}_{vv} \mathbf{w}} \quad \text{Eq. 3-5}$$

The array gain is then:

$$G = \frac{|\mathbf{w}^H \mathbf{d}_s|^2}{\mathbf{w}^H \mathbf{\Gamma}_{vv} \mathbf{w}} \quad \text{Eq. 3-6}$$

where $\mathbf{\Gamma}_{vv}$ is the noise correlation matrix and $\mathbf{d}_s$ is a vector of the transfer function of the desired signal to the various sensors.

$$\mathbf{\Gamma}_{ij} = \frac{\int_0^{2\pi} \int_0^\pi \{U^2(\theta, \phi) v_i(\theta, \phi) v_j(\theta, \phi)^H\} \sin\phi d\phi d\theta}{\sqrt{\left[\int_0^{2\pi} \int_0^\pi \{U^2(\theta, \phi) v_i(\theta, \phi)^H v_i(\theta, \phi)\} \sin\phi d\phi d\theta \right] \left[\int_0^{2\pi} \int_0^\pi \{U^2(\theta, \phi) v_j(\theta, \phi)^H v_j(\theta, \phi)\} \sin\phi d\phi d\theta \right]}} \quad \text{Eq. 3-7}$$

The underlying assumption is that the noise is spherically diffuse thus the element signal to noise ratio is constant. Maximising the signal to noise, as expressed in equation 3-5, the following sets of solutions are possible:

$$\mathbf{w}_{opt} = \kappa [\mathbf{\Gamma}_{vv}]^{-1} \mathbf{d}_s \quad \text{Eq. 3-8}$$

κ for three conditions is given in Mankolosis [39].

One solution is to set the look direction to be unity gain, called the *minimum-variance distortionless response (MVDR)* beamformer. That is,

$$\text{Min}_{\mathbf{w}} \mathbf{w}^H \mathbf{\Gamma}_{vv} \mathbf{w} \quad \text{subject to} \quad \mathbf{w}^H \mathbf{d}_s = 1 \quad \text{Eq. 3-9}$$

which yields the optimum weights:

$$\mathbf{w}_{opt} = \frac{\mathbf{\Gamma}_{vv}^{-1} \mathbf{d}_s}{\mathbf{d}_s^H \mathbf{\Gamma}_{vv}^{-1} \mathbf{d}_s} \quad \text{Eq. 3-10}$$

The noise matrix $\mathbf{\Gamma}_{vv}$ influences the shape of the beam pattern. If the noise field is defined as originating only from the back, it will be vastly different than a spherically diffuse field or cylindrically diffuse field. The principal trade-off is always between low frequency directivity and white noise gain. In this thesis we only consider the case of a spherically diffuse noise field ($U(\theta, \phi) = U$). To calculate it one integrates over a sphere, plane waves emanating from all possible directions. Thus:

$$\Gamma_{ij} = \int_0^{2\pi} \int_0^\pi \{v(\theta, \phi)_i v(\theta, \phi)_j^H\} \sin \phi d\phi d\theta \quad \text{Eq. 3-11}$$

If we assume a signal with no interference and only system noise then $\Gamma_{v+\eta}=\mathbf{I}$. (i.e. $\Gamma_{vv}=0$ and $\sigma_\eta=1$ in equation 3-4) This provides an estimate the discrimination of the array against system noise. The *White Noise Gain* of the array is defined as

$$\mathbf{WNG} = \frac{|\mathbf{w}^H \mathbf{d}_s|^2}{\mathbf{w}^H \mathbf{w}} \quad \text{Eq. 3-12}$$

Thus, if $\mathbf{WNG} > 1$ the array gives less noise than a single sensor. A delay and sum beam former is one where the WNG is optimised by making $\Gamma_{vv}=\mathbf{I}$ (this is effectively the same as maximising the gain (G) by making $\mathbf{w}^H \mathbf{w}=\mathbf{I}$). Thus, $\mathbf{w} = \frac{1}{N} \mathbf{d}_s$ and $\mathbf{WNG} = N$. In a superdirectional (MVDR) array, to offset the loss of WNG, we can add some white noise by a factor μ on the diagonal [8,41] .

$$\mathbf{w} = \frac{(\Gamma_{vv} + \mu \mathbf{I})^{-1} \mathbf{d}_s}{\mathbf{d}_s^H (\Gamma_{vv} + \mu \mathbf{I})^{-1} \mathbf{d}_s}. \quad \text{Eq. 3-13}$$

In the low frequencies Γ_{vv} is not well conditioned, as the wavelength is much larger than the inter-element spacing. Since the array samples a small part of the wavelength only small variations in phase and amplitude occur. A small positive μ makes its inversion easier and it trades off low frequency directivity for improved WNG. Recalling equation 3-4 this more closely models a real system and in this case μ models the system noise σ_η^2 .

A set of linear constraints can be added to the optimisation problem of equation 3-8 as in Herbordt [41]. Such constraints have been used to impose a null in a given direction or a constant beam width [41,42,43,44]. This type of constraint will be used in chapters 5 and 6 to provide constant beam width and to provide symmetrical beams.

Consider the following set of i ($i=\{1,2,\dots,N\}$) linear constraints can be applied

$$\mathbf{w}^H \mathbf{d}_i = \gamma_i \quad \text{Eq. 3-14}$$

where $\mathbf{d}_i$ are the transfer function between the source of interest and the array element.

In this case the optimisation problem under constraint can be written:

$$\text{Min}_{\mathbf{w}} \mathbf{w}^H \mathbf{\Gamma}_{vv} \mathbf{w} \quad \text{subject to} \quad \mathbf{C}^H \mathbf{w} = \mathbf{g} \quad \text{Eq. 3-15}$$

where $\mathbf{C}$ is a rectangular matrix defined by:

$$\mathbf{C} = [\mathbf{d}_1 \quad \mathbf{d}_2 \quad \dots \quad \mathbf{d}_N] \quad \text{Eq. 3-16}$$

and $\mathbf{g}$ is a vector defined by:

$$\mathbf{g} = \begin{bmatrix} 1 \\ \gamma_2 \\ \gamma_3 \\ \vdots \\ \gamma_N \end{bmatrix} \quad \text{Eq. 3-17}$$

The optimal weight vector w_{opt} under these conditions is given by:

$$w_{opt} = \mathbf{\Gamma}_{vv}^{-1} \mathbf{C} [\mathbf{C}^H \mathbf{\Gamma}_{vv}^{-1} \mathbf{C}]^{-1} \mathbf{g} \quad \text{Eq. 3-18}$$

If one now lets $\mathbf{C}=\mathbf{d}_s$ and $\mathbf{g}=1$ then one can easily get the MVDR of equation 3-9 [41]. These additional constraints adversely affect white noise gain and this limitation must be borne in mind in actual implementations with physically limited array elements and digital signal processing.

3.2 Linear free field array

The most commonly studied free field array is the uniform linear array. It is assumed that there are monochromatic plane waves that pass across the array and the array has no effect on the impinging wave front. Thus the phase relationships between elements is a function of the element spacing. This simplification holds up well in many cases, especially with electromagnetic antennae mounted on tall masts. It is worth looking a bit at these to understand some of the limitations of discrete arrays. To further simplify the discussion we will assume even illumination; that is, all the coefficients are the same magnitude. The pattern from such an array then becomes the well known [45,39]:

$$F(\phi) = \frac{\sin(N\pi\psi)}{\sin(\pi\psi)} \quad \text{where} \quad \psi = \frac{s(\sin\phi - \sin\phi_0)}{\lambda} \quad \text{Eq. 3-19}$$

$\phi_0 = 0$ at broadside to the line array and $\pi/2$ at end fire.

To get a narrower beam the length of the array and the number of elements must be increased for a given wavelength. Two other observations are pertinent. Firstly the spacing between elements must be less than half the wavelength to avoid grating lobes at all scan angles. Mathematically the spacing for the first grating lobe is expressed as:

$$s = \frac{\lambda}{\sin\phi_0 - \sin\phi_g} \quad \text{where} \quad \phi_g = \text{grating lobe angle} \quad \text{Eq. 3-20}$$

This is illustrated in figure 3.2.

Figure 3.2 Effect of inter-element spacing for a end-fire uniform linear array

Secondly the beam pattern changes quite dramatically from broadside to endfire. This is illustrated in figure 3.3.

Figure 3.3 End fire (left) and Broad side (right) linear arrays for $s = \lambda/4$

For linear arrays [45,39]:

1. There is an annular ambiguity about the axis of the array in broadside due to the fact that it is a linear array in free field and only completely disappears at end fire.
2. The beam pattern narrows with increasing frequency.
3. The beam width is much broader at endfire than at broadside.
4. There results an ambiguity in the pattern when the spacing is less than the grating lobe criterion. The spacing between elements must be less than $\lambda/2$ if one wants to cover all angles from broadside to end fire and can be relaxed to λ if only broadside steering is required.

These limitations have been dealt with to a great extent by use of variable element spacing and variable weightings of the elements [45]. However, they do remain fundamental limitations of linear arrays and serve to illustrate, in a simplified form, the problems that this thesis proposes to address.

3.3 Circular free field array

One of the stated objects of this thesis is to obtain symmetrical beam shapes in azimuth. Linear arrays make this difficult but a circular array has the basic geometry to provide this solution.

Circular arrays in free space have been extensively studied and the well-known results are presented [46,47]. The response of a free field circular array of N elements can be written as:

$$F(\theta, \phi) = \sum_{n=0}^{N-1} w_n f_n(\theta, \phi) e^{jka \sin(\theta) \cos(\phi - n\Delta\phi)} \quad \text{Eq. 3-21}$$

where a = radius of the array;

$f_n(\theta, \phi) = f(\theta, \phi - n\Delta\phi)$ = element directionality pattern; $k = \frac{\omega}{c}$;

w_n = element weighting function (illumination)

Assuming omni-directional elements such that the total pattern becomes [48]:

$$F(u, \theta) = N \left[J_0(u) + 2J_N(u) \cos(N(\theta - \pi/2)) + 2J_{2N}(u) \cos(2N(\theta - \pi/2)) + \dots \right]^2 \quad \text{Eq. 3-22}$$

where $u = ka \sin(\phi)$, a = radius of the array, $k = 2\pi/\lambda$ = wave number, N = number of elements and for uniform illumination. In order to draw general conclusions it is best to express the results in term of the dimensionless ka factor, which is similar to the ψ used in linear arrays (see eq. 3-17). The first term dominates and thus, in the plane of $\theta = \pi/2$:

$$F(u, \pi/2) \approx [J_0(ka \sin \phi)]^2 \quad \text{Eq. 3-23}$$

The two major limitations of the circular array now become apparent. Firstly the sidelobes are quite high as the first maximum takes on a value of 0.4026 which corresponds to a level of -7.9dB . The second limitation is bandwidth related. If the argument of the Bessel function changes too much the pattern is lost. Davies [47] argues that the criterion should be less than $\pi/8$ change. This leads to a bandwidth of $\Delta f \sim f_0 \lambda / 8a$. (Where f_0 is the design frequency of the array.)

Any array of sensors with uniform illumination (or weighting) results in a narrowing of beamwidth with frequency. As noted above, for a circular array it follows that of an ordinary Bessel function. As $\sin\phi$ can only take on the interval $(-1,1)$ the beam pattern varies from $J_0(0)$ to $J_0(\pm ka)$. The plot of the absolute value of $J_0(x)$ illustrates the beam pattern (Figure 3.4). For values of $ka < 2.4$ there will be only one large main lobe. As ka increases, sidelobes will appear and the main lobe will occupy less of the ka -axis, which corresponds to the beam width.

At low frequencies (small ka) the 0th order Bessel function dominates. However as the frequency increases side lobes arise and are augmented by the higher order Bessel functions. At a certain frequency grating lobes will arise as in the linear array: figure 3.5. Also, as in a linear array, the directivity scales with the size of the array (radius a).

The variation of beamwidth with look direction is illustrated in figure 3.6. The array comprises of six elements as did the linear array of figure 3.3 and as expected the variation is less severe.

Figure 3.4 0th order ordinary Bessel function (J_0) in dB

$$ka=\pi/4$$

$$ka=3\pi/4$$

$$ka=\pi/2$$

$$ka=\pi$$

Figure 3.5 Effect of inter-element spacing (grating lobes) for a uniform circular array

Figure 3.6 Beam shape variation for circular free-field array

3.4 Circular arrays about a hard sphere

Mailloux [46] argues that a way to address the shortcomings of a circular array is to use an array of directional elements, which can be achieved using a conformal approach. If one now places an object that significantly affects the plane waves and places the array elements on the surface of the object one has a conformal array. These are reasonably common on cylindrical objects such as missiles and torpedoes. The array is generally circular and usually comprises of several rings. This has been the object of quite interest in the RADAR and SONAR communities. The simplifying assumption is that one makes is that the cylinder is infinite as analytical solutions exist for this case [49,50,51,52].

Meyer treats a problem that is of great interest mounting a circular array on a hard sphere. He assumes plane wave propagation as the distance from the source to the array is larger than Balanis' [53] criterion of $R > 2L^2/\lambda$. This is a much larger distance than Ryan's criteria of $R > (L^2/2\lambda) - \lambda/8$ for an error of less than 1dB[54]. Ryan considered the spherical spreading from a monopole and thus obtains a more accurate result while Balanis considers only a plane wave.

Meyer [7] uses a solution found in Bowman [55] which is similar to that in Morse [17]. Considering the pressure field from a plane wave impinging upon the sphere from various directions, the total pressure at a point on the sphere indicates the directionality. Naturally, the solution scales with the size of the object and the frequency. Kinsler and Frey [56] illustrate this effect and it is reproduced as figure 3.7, no significant

directionality occurs at frequencies below approximately $ka < 1$ where $k = 2\pi f/c$ (f = frequency, c = speed of sound) and a is the radius of the sphere.

Figure 14.8.1 Sound pressure on the surface of a sphere of radius a_h ensonified by a plane wave incident from the right. The angle θ is measured from the direction of approach of the wave.

Figure 3.7 Diffraction about a hard sphere (Kinsler & Frey figure 14.8.1)

Figure 3.8 Sphere and co-ordinate system used by Meyer $a=85\text{mm}$

Meyer [7] provides results for a sphere of 0.085 m radius, which is approximately the size of a human head (illustrated in figure 3.8). Because it is possible to derive an analytical formula to calculate the pressure on a sphere, he is able to use the phase mode method typically applied in circular arrays. Only the less restrictive pattern is achieved in

his measurements (figure 3.9). The results are explained by the severe degradation in white noise gain of the second pattern.

(a) Pattern A

Figure 3.9 Directivity patterns obtained by Meyer

3.5 Numerical simulation of the analytical solution of an ensonified hard sphere

As it has already been studied and it lends itself to study we will consider the object in some detail. Let us start by considering a sphere at the origin of the co-ordinate system as illustrated in figure 3.8. It is subjected to a monochromatic plane wave. As we are interested in microphone arrays we use the linear acoustic equations.

The sphere is assumed to be a solid that is perfectly reflecting and has no internal acoustic path. The resulting sound pressure is thus the sum of the incident wave and the scattered wave. We are interested in the acoustic pressure at the boundary of the sphere which can be written as [17]:

$$p_a = P_0 e^{-i\alpha} \left(\frac{1}{ka} \right)^2 \sum_{m=0}^{\infty} \frac{2m+1}{B_m} P_m(\cos\theta) e^{-i(\delta_m - \pi/2)} \quad \text{Eq. 3-24}$$

where the amplitude and phase of the scattering from a sphere are:

$$B_m = \sqrt{\frac{[m n_{m-1}(ka) - \{(m+1) n_{m+1}(ka)\}]^2 + [(m+1) j_{m+1}(ka) - \{m j_{m-1}(ka)\}]^2}{(2m+1)^2}} \quad \text{Eq. 3-25}$$

$$\delta_m = \tan^{-1} \left(\frac{(m+1) j_{m+1}(ka) - \{m j_{m-1}(ka)\}}{m n_{m-1}(ka) - \{(m+1) n_{m+1}(ka)\}} \right) \quad \text{Eq. 3-26}$$

where $P_m(x)$ is the Legendre function for spherical co-ordinates, $j_m(x)$ is the Bessel function in spherical co-ordinates and $n_m(x)$ is the Neumann's function in spherical co-ordinates. (These can be found in tables VI and VII of [17] or App. A4 [56]). To obtain

an accuracy of better than 1dB in amplitude it has been shown that only the first $ka+10$ terms are required in the sum of equation 3-22 [5].

It is useful to study the resulting pressure and phase variations along the equator (where one would place sensors) of the sphere. Firstly, consider the pressure. Note that for a sphere much smaller than the incident wavelength (small ka) no significant effect occurs, as expected. As the radius of the sphere approaches $ka=1$ significant pressure fluctuations start as illustrated in figure 3.10 and 3.7. Once ka reaches about 6, the object is large enough with respect to wavelength, that the diffraction approximations become valid. The pressure starts exhibiting the fluctuations normally associated with the Airy function as described in standard texts [16].

Figure 3.10 Pressure variation on an ensonified sphere - simulation results

Consider now the phase. Recall that in a free-field array the phase is linear with ka as is evident by inspecting equation 3.19. Figure 3.11 illustrates the unwrapped phase of different points on the equator of an ensonified sphere referenced to the point of first contact of the incident plane wave. It has a linear behaviour for the larger ka values but there is evident non-linear behaviour in the smaller ka range.

Figure 3.11 Unwrapped phase for an ensonified sphere

Figure 3.12 is a partial view of figure 3.11 for ka up to π overlaid with the phase for a free-field array of the same size. The non-linearity is very evident as is the apparent increase in size of the array behind the scattering object (time delay is proportional to the slope of the phase which corresponds to distance in free-field).

Figure 3.12 Unwrapped phase for an ensouffled sphere (solid lines) versus free-field (dashed lines) $ka=0$ to π

It is therefore not surprising that using free field coefficients on a solid spherical baffle the results suffer from aliasing problems. Using the delay-and-sum weighting of a free-field array (of figure 3.5) with the pressure calculated by equation 3-22 one gets the results of figure 3.13.

Figure 3.13 Spherical baffled array using free-field coefficients

Figure 3.14 Spherical baffled array using coefficients accounting for scattering (eq. 3-22)

Using the delay and sum beamformer ($\Gamma_{vv}=\mathbf{I}$) and correcting for the calculated pressure field (eq. 3-22) for a sphere the same radius as in figure 3.5 ($a=0.05\text{m}$), one obtains the beam patterns of figure 3-14. The shape of the beams is obviously better than those of figure 3-13 as the main lobe is in the correct direction and there is no aliasing. In the lower frequency (lower ka) area the beam is wider. Aliasing that occurs just after $ka=\pi/2$ in the free-field condition now appears at $ka=3\pi/4$. The array therefore appears to be about twice as large as it was in free field. This result has been previously reported [4].

3.6 Conical arrays

These have been studied in radar and sonar arrays. Often one does not take into account the effect of the cone and it is assumed to be negligible in the interest of obtaining a reasonable formula [52].

When the cone is much larger than the wavelength then diffraction approximations can be made and these are described in detail in [49] for electromagnetic waves.

3.7 Conclusion

In this chapter the basic mathematics for discrete arrays without mutual coupling has been summarised from various authors. Free-field array behaviour for the simple case of uniform illumination has served to illustrate the shortcomings common to free-field arrays most notably the spatial aliasing and the variation in beamwidth in azimuth.

The results reported by Meyer [7] are presented and some of the benefits of a conformal array on a scattering structure are highlighted.

Finally a simulation of the analytical solution to the scattering by a sphere is computed. This provides some insight into the benefits that a scatterer can bring to the problem of beamforming. The obvious benefit is that of the increased pressure variation from one sensor to another about the sphere increases the signal to noise ratio at the sensors. The other phenomenon that is often overlooked is that in the low frequency regime (small ka) the phase becomes non-linear. This is of special importance as in that range of ka the pressure variations are less important. This provides a means to perform beamforming which is unavailable in the free-field case.

Unfortunately, infinite cylinders and spheres are not practical shapes for most microphone arrays. To use any other shape the calculation of the acoustic field on the scattering body becomes impossible analytically so one must resort to approximate methods.

The finite difference methods, with the advent of modern digital computation, are cost effective and efficient. The Boundary Element Method is the approach that will be used for this study as explained in the following chapters as it has been successfully used in the past by others [4,5].

Chapter 4

Simulation & Measurement Environment

This chapter describes the simulation environments used (IDEAS and MATLAB) as well as the measurement system and physical environment to perform validations on real arrays.

4.1 Boundary element method

This method is now a reasonably mature method and several acoustical codes have been commercialised. Vibro-Acoustic (Rayon) [57] integrated in IDEAS is used as it is available at Mitel and has been successfully used to model telephone devices [58]. Familiarity with this research made it much easier to use it as there a level of confidence and familiarity.

The primary assumption is that the shape of interest is on an infinite reflecting plane. This simplifies the modelling. This can be justified by arguing that the devices of interest are audio-conferencing units that sit on large tables. The narrowest dimension would be in the order of 1.2 metre.

To calculate the diffuse field and the sources of interest some reasonable spacing had to be used. A spacing of 10° in both the azimuths and elevation is used. As we are simulating a free field (anechoic) the pressure variations are very smooth from point to point permitting us to interpolate if more points are required. The field of interest is

modelled as a hemisphere centred on the object with a radius of a metre. With a resolution of 10° this represents 325 sources.

Figure 4.1 Grid of the 325 sources at 1m from array

The diffuse field is simulated by setting the sources at 10° intervals but on a hemisphere with a radius of 10m thus ensuring free field conditions. Using Ryan's criteria of $R > (L^2/2\lambda) - \lambda/8$ for an error of less than 1dB [54] and rewriting it and solving for frequency we get: $f < c / \left(-4R + 2\sqrt{4R^2 + L^2} \right)$. Thus for an array of 0.1m diameter (L) at a distance(R) of 10 m any frequency below 680kHz can be considered to be a plane wave with less than 1dB of error.

The sources are simulated as acoustic monopoles, as this is a reasonable approximation of the human voice especially in a semi-anechoic environment [18,19,20,21].

Measurements are carried out in a semi-anechoic room so this should provide us with reasonable results

The scattering object is assumed to be a perfectly rigid body. This simplifies the boundary conditions and is a reasonable approximation to the modelled ABS plastic box typically used to house telecommunication devices.

The density of the mesh of the scattering object determines the maximum frequency at which the calculations are valid. The denser the mesh the higher the maximum frequency.

The other peculiar attention that is required is that the mesh be fitted to the object such that the microphone position of interest corresponds to a node of the mesh. Manual input is required to ensure that the mesh generated by the built-in mesh algorithm provides us with the required nodes. The “mapped” semi-automatic mesh generator is used. Keeping track of the nodes of interest is also a tedious detail that must be taken care of.

4.2 MATLAB environment

The greater majority of the MATLAB code has been written used in the 6.5 version [59]. The signal processing toolbox is also sometimes used.

The programs have been developed to be as flexible as possible but also with the principles of reproducible research in mind. Thus every MATLAB figure has an associated M-file that generally only acts as a script calling the functions that have been written.

Three-dimensional visualisation uses many of the improvements in version 6.5 but most of the other functions used are reasonably core routines to MATLAB and have been in MATLAB for quite sometime.

The results from the numerical simulation are exported from IDEAS as ASCII files and imported that way into MATLAB. The nice graphics are the primary data exported from MATLAB. However, the filter coefficients necessary to implement the beamformers are exported in delimited ASCII format for maximal compatibility. Some audio simulation done within MATLAB's is exported as WAV files.

4.3 LabVIEW Environment

To implement the filters and to measure the prototypes LabVIEW is used extensively. The programs are written in version 6.1 [60], with use of some of the elements of LabVIEW Sound and Vibration toolbox [61] and the Signal Processing suite [62].

Test signals are all imported from WAV files, as this is a ubiquitous format for audio in the personal computer world and generally in the acoustic research world.

4.4 Measurement system hardware description

There are several systems available to do data acquisition of multiple microphones. Three systems are considered on the basis of their availability. The best available solution was to use the NI4472 8ch system [63] with a custom built Burr-Brown INA131 [64] preamplifier. A two-channel system was used for preliminary validations.

4.4.1 National Instruments NI-4551

This two-channel 16-bit system is very easy to use with LabVIEW, as it is a NI product [65]. The accuracy is reliable and the system can easily be calibrated. The major drawback is that it is only a two-channel system and to perform multi-channel measurements one has to always keep one channel as the reference. Performing such measurements takes a long time and is subject to operator error. There is also significant post processing making it quite time consuming. The results for the first validation were performed using this system.

4.4.2 National Instruments NI-4472

This eight channel input only system is very easy to use with LabVIEW as it is a NI product [63]. It has 24bit A/D converters and the preamplifiers are such that it gives a very low noise performance and excellent phase match ($<0.1^\circ$). The background noise level of the system is low enough that one could connect the output of a biased electret microphone directly without amplification. Unfortunately, even with a reasonable environment, it is difficult to avoid ground loops (60Hz ground paths are unequal) and this tends to be louder than the desired signal.

Figure 4.2 Noise Floor of system

The solution that was to provide a preamplifier instead of a simple buffer that would remove this common mode noise. By use of precision instrumentation amplifiers one can get very high common mode rejection ($>100\text{dB}$) with accurate gain ($40\text{dB} \pm 0.05\text{dB}$) and reasonably low noise ($10\text{nV}/\text{Hz}$). (Burr-Brown INA131 [64] was used.) The microphone capsule also has an inherent noise floor and the use of long cables also adds noise. If one then calibrates the microphone one can get an equivalent dB SPL noise level for these various conditions; figure 4.2 illustrates this (800 line FFT – Hanning window). The noise floor of the amplifier is about 20dB above that of the measurement card alone. As we are supplying 40dB of gain this means a net gain of 20dB of signal to noise. The

noise of the amplifier has some influence on that of the microphone, as it is at worse case only 4dB quieter. However, when we connect the long wire the noise picked up by the wire and microphone is 20dB louder than the amplifier thus meaning that there is no material contribution due to the amplifier noise. The maximum level that the system can measure is about 114dB SPL so even in this worse case we have a signal to noise ratio on the order of 100dB for this maximum signal (80dB for a 94dBSPL (0dBPa) signal).

The other major concern is cross talk, especially when we use long cables. Figure 4.3 illustrates the performance we can expect. With very short cables the cross talk is quite low on the order of 110dB. The long cables can degrade this significantly to about 70dB. This however will be quite acceptable for microphone arrays as the level difference are on the order of 10dB so 70dB of cross talk will be well below the limit that would concern us.

Of note here is that at 114dB SPL (from a calibrator) we get close to the maximum input that can be handled by the NI4472 with the 40dB gain provided by the preamplifier. In the event that this is insufficient dynamic range one can replace this amplifier by a pin compatible part with different gains.

Figure 4.3 Crosstalk of short cable

Figure 4.4 Magnitude and Phase of Microphones

4.5 Acoustical measurement environment

The measurements were carried out in a semi-anechoic chamber 3m x 4m x 2.2 m high. To ensure reasonable reflection from the floor it was covered with 38mm thick medium density fibreboard (MDF) laminated with a 3mm thick high-pressure laminate.

The measurements were carried out every 15°. The sound source was generated by the NI4551, connected to a power amplifier and then to a Brüel & Kjær 4227 [66] artificial mouth with a 1/2" pressure microphone (B & K 4134 [67]) at the MRP permitting equalisation of the source. This microphone signal is amplified with a B & K 2639 [68] preamplifier and 2610 amplifier [69] before being sent to the NI 4472 A/D card.

The artificial mouth was placed at 1 m from the prototype at azimuth 0°. The prototype was rotated about 360° taking measurements at every 15° (except for elevation 90°). The artificial mouth elevation was varied from 0° to 90° in 15° increments. Thus, a set of measurements required 192 individual measurement points (8 X 24 + 1 at apex).

4.6 Test signal and analysis

The test signal starts with a short 1KHz tone burst followed by a 4096-point chirp that is repeated 10 times. A short (10 second) speech sample (Hamlet Act II Scene 5) follows. Finally the 10 chirps are repeated. The sampling frequency is 20,480 Hz.

The measurements are sampled simultaneously at the six microphones. Also measured is a signal from the microphone at the MRP position in front of the artificial voice. Finally the electrical signal sent to the artificial voice amplifier is recorded.

The measurements are performed under control of a LabVIEW programme. (see Annex 1) This programme plays out the test signal WAV file to the artificial mouth and records the eight input channels after having filtered them with a digital 8th order high pass Bessel filter ($f_c = 60$ Hz) to remove the low frequency noise present in the anechoic chamber. (This noise is due to the building mechanical system and is primarily at 30 Hz.) The program records the signal in two files, one is the 10 chirps and the other is the speech sample only.

The free-field frequency response of the microphones is calculated by applying a 4096 point FFT (with no window as a chirp is a periodic signal) and calculating the ratio of each microphone to the reference microphone. This transfer function gives us the complex function between a source at 1 m and each microphone with a 5Hz bandwidth. Unfortunately the 6 microphone signals are to some extent corrupted by noise which is aperiodic. This can result in the characteristic rectangular window problems that appear as a periodic frequency response. To avoid measuring the data again the data was re-analysed using a sliding window with periodic averaging. This reduces noise effects and provides results that correspond to swept sine and long averaged noise measurements.

The programme is listed in the appendix both in the LabVIEW and MATLAB implementations.

The speech samples are useful to process speech with the beamformers that are designed. This permits one to perform auralisations of the beamformer without the need for the physical model.

4.7 Microphone Calibration

The microphones that were used in the model are typical of those used in telephones [70]. They are significantly less expensive than measurement microphones and from an engineering perspective provide a realistic evaluation of the performance that one could expect in a cost effective microphone array. The manufacturer grades these typically in grades of $\pm 3\text{dB}$. For measurements a higher accuracy is desirable. Six microphones were selected and their phase was measured in an anechoic room with a B&K 4227 mouth [66] at 2.5m to obtain a reasonably planar wave. One microphone was kept constant as the reference and the others were placed just beside it. Using a 2-channel acquisition system [65] the magnitude and phase were measured. The results are illustrated in figure 4.4. The microphones were selected because of their reasonably good phase match. The magnitude variations are independent of frequency so they are easily corrected by scaling the data with a resulting match of less than 0.5dB over the frequency range of interest. The phase in the low frequency is within 2° and diverge up to about 6° at 3300Hz. While significantly better phase matching is possible with instrumentation microphones this is the best that could be obtained from a batch of 100 inexpensive electret microphones.

4.8 Validation of BEM models

Two validations were carried out, one quite early in the project and one much later. As discussed previously the eight-channel system provided much improved performance. Detailed results are illustrated in the following chapters.

4.9 Real time emulation environment

Using the same combination of data acquisition cards (NI 4472 and NI4451) and the amplifier described above a real-time emulation system was designed. Using a PC (personal computer) with a 700MHz Celeron processor we can obtain real time operation with an 8000Hz sampling rate and using buffers of at least 250 samples. (This results in about 32ms delay.) In order to use this the optimal weights calculated for each frequency have to be used to design an FIR filter. This is done using a least squares method in MatLab and good results are obtained with 60 tap FIR filters.

The program is in LabVIEW (see App. 1) so one must refer to the wiring diagram to completely understand it. In the initial part the two cards are initialised and then we read in the FIR coefficients. A button on the front panel controls the main While loop. The while loop starts by reading in a block of data. The data is passed to a bank of IIR filters (we use them as FIR with only forward coefficients). The output is then summed. To do this we take the last microphone input and then use this as the initial value for the shift register in the FOR loop that sums together the shift register and all the inputs -1 of the maximum. The output is now the beamformer output. There is a gain multiplication just

before the data is output to the channel 0 of the output card. Once the While loop terminates we clear the two cards.

4.10 Conclusion

The simulation environment and the measurements systems and set-up have been described in sufficient detail that the results presented can be reproduced.

Chapter 5

Inter-element Spacing of Scattering Conformal Arrays

This chapter explores the consequences of embedding a microphone array on a scattering object. A truncated cone shape is used. The consequence of the acoustical scattering is that the spatial aliasing that plagues free-field arrays is overcome. A wide-band beamformer is proposed that goes well beyond the $\lambda/2$ inter-element spacing and also provides significant signal processing savings.

5.1 Introduction to the problem

The problem in its most general form can be illustrated as in figure 5.1. In this problem sensors are placed in an arbitrary pattern about an arbitrary solid body that can have varying surface impedance. There is no coupling between sensors. The goal is to obtain from this array uniform beam widths in all directions over a wide range of frequencies. This is very challenging and only some compromise solution is realistic.

Figure 5.1 Generalised shape for a microphone array

The presence of the surface in the problem is opaque to the wave field and of significance to the performance of the array. It is important to note that it is assumed that there is no acoustical transmission through the object. Generally this is achieved, practically, when the density of the object is much greater than that of air.

Several researchers have looked at similar problems in electro-magnetics and acoustics but they have assumed a diffracting object (the object is large with respect to the wavelength), which does not hold for this problem [7, 46, 48, 49, 50, 51, 52].

5.2 Array about a solid truncated cone

The object studied comes from a Mitel product but a similar shape can also be found in electromagnetic antennae [46,48]. The choice of a truncated cone was initially motivated by the desire to increase the sound pressure at the transducers to enhance the signal to noise ratio. Its geometry is illustrated in figure 5.1.

Figure 5.2 Symmetrical Truncated Cone shape

The base is 10cm in diameter and there are six microphones spaced at 60° placed as close as possible to the base. The microphones used are standard, 1cm diameter, omnidirectional electret microphones and they are wired using small gauge wires that exit the base to prevent any significant acoustical effect. As illustrated the device is 6cm high and the top is slightly domed with a radius of 50cm. The diameter at the top is 17cm. It is machined in ABS and the wall thickness is nominally 3mm.

5.3 Validation of simulation of a truncated cone

A model of the truncated cone prototype is validated [11]. Figure 5.3 illustrates the boundary element model mesh used to calculate the pressure at each microphone due to the sources at the 325 positions set out in section 4.1. The results were taken for the first sector (0° , 15° , 30° , 45°) and rotated by 60° to complete the 360° required. The measurements were carried out with a two-channel data acquisition system so this data reduction was very important.

Figure 5.3 Boundary Element mesh of truncated cone object

The agreement between the measured and simulated results is reasonable. In figure 5.4 it is obvious that the pressure variation due to the scattering is very well modelled. The source is at 15° of elevation but there are some discrepancies at 2000Hz between 120° and 150° . Figure 5.5 shows the sound pressure at different angles of azimuth normalised to that of the source at 0° versus frequency. The agreement is very good except at 120°

and 150° which explains the discrepancies evident in figure 5.4. The sound pressure measurements are very sensitive to positioning errors as well as any reflections that may occur. It is important to note that the pressure varies quite importantly with frequency. This is due to the scattering of the wave upon the solid object. In a free field array the attenuation from one element to another has very little frequency dependence, as the absorption of sound in air is small, less than 0.1 nepers per meter (0.869dB/m) [16]. Naturally, this effect would also affect sound propagating about a solid object but given that the effects we are observing are on the order of 20dB it is quite reasonable to ignore such small effects.

The accuracy of the phase is illustrated in figure 5.6. The measurements follow the simulation reasonably closely. Phase errors are difficult to avoid when 6 independent points are to be compared but they are measured individually. The measurements validate the model as the phase exhibits the non-linear behaviour at the same frequencies as in the simulation results.

These measurements confirm that the boundary element simulation accurately reflects the acoustical behaviour of a scatterer in the shape of a truncated cone on a reflecting plane.

Figure 5.4 Polar plots of microphone response at base of truncated cone; measurements versus simulation (solid line)

Figure 5.5 Normalised frequency response for microphone positions: measurements versus simulations

Figure 5.6 Unwrapped phase normalised to mic. 1 for various microphones: measured versus simulation

5.4 Consequences of a scattering object

As in the case of the sphere the truncated cone is expected to provide benefit to the conformal array by providing an increase in "apparent" size.

The increase in "apparent" size can be explained by the scattering effect of a conformal array about a solid object of a reasonable size. Figure 5.7 clearly illustrates the advantage provided in pressure variation of the truncated cone (microphones 1 to 6) compared to the pressure variations one would get in a free-field array of the same diameter (broken lines) with a monopole at 1 m. It is also interesting to compare the effect of a truncated cone to that of a sphere. Figure 5.8 illustrates the pressure variation at various positions on the cone versus ka (where a is the radius of the microphone array at base). As in the case of the sphere (figure 3.7) there is no significant effect for $ka < 0.4$ where the object is much smaller than the wavelength. Starting at about $ka = 1$, the truncated cone provides significantly more pressure gain at large ka as it reaches close to 3.5 whereas the sphere only gets to 2 (the expected pressure doubling). Again, as in the case of the sphere the diffraction effects do not manifest themselves clearly until ka reaches about 6.

As in the case of the sphere the non-linear phase behaviour is also a very important effect of a scattering object. In figure 5.6 it is obvious that the phase remains fairly linear for high frequencies. In the lower frequency area the non-linearity is more evident. In contrast to the sphere, the phase does make fairly abrupt changes in the shadow area of the object. Of course in a free field array the phase always remains linear as the delay between elements remains constant due to the homogeneity of the medium.

Figure 5.7 pressure response at the microphones

Figure 5.8 Sound pressure on cone ensounded by a point source at 1m

As in the case of the sphere there is a significant reduction of the spatial aliasing. To evaluate this, consider the delay and sum beam shapes obtained with the truncated cone conformal array illustrated in figure 5.9 and compare them to figure 3.14. Similar behaviour as was observed with the sphere in section 3.5 is evident. The spatial aliasing is no longer evident and the array behaves as if it was larger. The cone's benefit over the sphere is evident in the smaller side-lobes.

Figure 5.9 Delay and sum for a conformal array at the base of a truncated cone

As explained in section 3.1, more directionality can be obtained by simply applying the MVDR method with some nominal regularisation (white noise gain factor, μ of 0.01). The choice of this regularisation corresponds to a system noise of 1% or 40dB signal to

noise ratio, which is applied at all frequencies. In an actual implementation this varies with frequencies and generally the SNR is smaller at lower frequencies. These choices are primarily for illustrative purposes.

As illustrated in figure 3.10, the beamwidth narrows as the frequency increases and sidelobes become more important. The effect of the obstacle is that the beam width at lower frequencies is narrower than in the free field but at high frequencies the narrowing of the beam can become problematic in sound capture. With a very narrow beamwidth any misalignment between the beam and the target source will result in significant coloration due to the sidelobe pattern. The approach retained is to apply linear constraints.

Figure 5.10 Truncated cone array MVDR ($\mu=0.01$)

Three constraints applied are: the MVDR $\mathbf{w}^H \mathbf{d}_s = 1$, and $\mathbf{w}^H \mathbf{d}_{s+30^\circ} = 0.707$, $\mathbf{w}^H \mathbf{d}_{s-30^\circ} = 0.707$. The resulting beam pattern illustrates the uniformity that can be achieved with this type of constraint. There is of course reductions of the directionality (DI) as the main lobe is wider than in the case of the MVDR without constraints. Figure 5.11 illustrates the results obtained.

Figure 5.11 Truncated cone array with linear constraints at $\pm 30^\circ$

This method also ensures that the main beam width is similar regardless of the azimuth chosen. As previously illustrated in section 3.3 a circular array can have variations in beamwidth as the beam goes from being on axis of a sensor to being in between two

sensors. This problem remains in conformal arrays but can be solved using the same constraints as noted above that are used to ensure a reasonably uniform beam width over a large frequency band as illustrated in Figure 5.12 .

Figure 5.12 Beam shape variation before and after linear constraint of -3dB

In a system where one assumes that the bearing direction is known the most important design criteria would still remain directivity. However, for uniform sound capture over a wide frequency range a uniform main lobe over frequency is more desirable to avoid coloration of the signal when the bearing direction and the source are not perfectly aligned. As in any engineering problem the answer is a compromise.

5.5 Improving the diffraction to extend the frequency range

Stinson and Ryan noticed that the obstacle increases the "apparent" size of the array, as there is an increase in the wave travel time from one microphone to the other compared to a free-field situation [4]. They also considered the effects of air-coupled surface waves due to a reactive surface impedance, but this type of surface is not considered here [71] as these effects are narrow band and more suited to low frequencies.

Consider the pressure magnitude at the six microphone locations when the source is located directly in front of microphone 1 (in figure 5.7). As expected, the object has very little effect in the low frequencies. However, from about 1000Hz a significant shadow effect arises. From about 3000Hz the difference from the microphone directly facing the signal to that at the other side of the object is on the order of 10dB. This implies a fairly strong directionality. Figure 5.4 illustrates the simulated and measured results [11]. The directional pattern has a significant front to back rejection and are reasonably similar to those obtained for the constrained beamformer in figure 5.11.

To enhance this high frequency directionality a 6mm thick layer of felt was applied as illustrated in figure 5.13. The resulting microphone responses (figure 5.14) show a reasonably constant beamwidth in the high frequencies, which also correspond reasonably closely to those of figure 5.11. The directionality is about that of a super-cardioid microphone [36] that is often used for sound capture.

Figure 5.13 Location of absorptive treatment on truncated cone

Figure 5.14 Improvement in directionality of microphone response due to the surface absorptive treatment on a truncated cone

Combining this with the constrained constant beam width method described above, one will get reasonably uniform beam width over the wide-band speech range of telephony (300-7000Hz), illustrated in figure 5.15. This effectively overcomes the spatial aliasing restrictions described in chapter 3.

The problem is how to implement a smooth transition between the constrained superdirective method using all six microphones in the lower frequencies and the use of only one or two microphones in the higher frequencies. In ITU-T G.722 a 24 tap QMF filter is used to separate the bandwidth in two [72]. It would seem logical to follow this partition and to use this frequency band partition for the spatial filtering. Thus for the lower band, (0-4000Hz) a six microphone beamformer would be implemented while for the higher frequencies a simple microphone selection scheme or a two microphone array would be used. From a signal processing resource perspective this is very attractive as it requires very little processing of the high frequency signals (8000-16000Hz). To give an idea of the order of the savings, assume the ideal case where only one microphone is switched. The high frequency band would require only one read/write operation in the subbanded implementation.

300 Hz

1 kHz

2.5 kHz

3 kHz

4 kHz

5 kHz

6 kHz

7 kHz

Figure 5.15 Proposed wide band array response

The alternate solution (without sub-banding) is to add twice as many microphones (to avoid grating lobes) and to use a beamformer using these twelve microphones. The computational load for the high frequency band would be four times more (twice the number of microphones and twice the sampling rate) than that of the lower frequency beamformer assuming that similar length filters could be used. A realistic length of these types of filters is a 40 tap FIR filter [8]. Even with very optimised code assuming Analog Devices DSP this results in a minimum of 960 operations per low frequency band sample [73].

Table 5-1 MIPS use for proposed scattering wideband array versus conventional array

Method	Low band (MIPS)	High band (MIPS)	Total (MIPS)
Conventional	15.4	30.8	46.2
Proposed	7.7	0.02+18.4	26.2

As illustrated by the results of table 5-1 a significant computational load can be avoided. Even with current digital signal processors a savings of 20 MIPS is very substantial. The actual MIPS usage will vary depending on the processor used and the code efficiency. However, there will remain a significant DSP load that can be saved.

5.6 Conclusions

The validation of the numerical model with a physical prototype provides confidence in the method and shows that it can be used for the design of conformal arrays.

The scattering effects of a truncated cone on a reflecting plane are more significant than those encountered on a sphere in free space but the same general characteristics are present: significant high frequency effects at ka above 6 and phase non-linearities in the lower range of ka .

The use of simple linear constraints has been used to provide a reasonably uniform beam pattern in both frequency and azimuth.

The exploitation of physical acoustic phenomena of scattering and sound absorption has been used to extend the frequency range of a conformal microphone array two times beyond the generally reported $\lambda/2$ inter-element spacing criteria. Significant benefit to the signal-processing load can be realised by combining the linear constraint for the lower frequencies and the physical acoustics for the higher frequencies. This effectively answers the second major question of this thesis.

Chapter 6

Proposed Symmetrical Beam Shapes for Asymmetrical Conformal Arrays

As one would expect, an asymmetrically shaped array, yields asymmetrical beam shapes. To start the discussion a free-field elliptical array that produces asymmetrical beams is considered. This leads to the need to develop an indicator of the symmetry of a beam pattern: the Asymmetry Index is proposed. An asymmetrical scattering object with an elliptical conformal array is simulated and measured. Solutions for obtaining significantly more symmetrical beam shapes are presented. The results are shown to apply not only to the simulated data but also to a real implementation.

6.1 Elliptical free field array

The response of a free field elliptical array of N elements can be written as:

$$F(\theta, \phi) = \sum_{n=0}^{N-1} w_n f_n(\theta, \phi) e^{jk \sqrt{l_a^2 / (\cos^2 \phi_n + \varepsilon^2 \sin^2 \phi_n)} \sin(\theta) \cos(\phi - \phi_n)} \quad \text{Eq. 6-1}$$

where l_a = axis in the x direction and

$\varepsilon = l_a / l_b$: the eccentricity of the ellipse;

$f_n(\theta, \phi) = f(\theta, \phi - \phi_n)$ = element directionality pattern; $k = \frac{\omega}{c}$;

w_n = element weighting function

If we now assume omni-directional elements such that $f_n(\theta, \phi) = 1$ and assume that $\theta = \pi/2$ (in the plane of the array) we get

$$F(\phi) = \sum_{n=0}^{N-1} w_n e^{jk\sqrt{l_a^2/(\cos^2 \phi_n + \varepsilon^2 \sin^2 \phi_n)} \cos(\phi - \phi_n)} \quad \text{Eq. 6-2}$$

$$e^{jk\sqrt{l_a^2/(\cos^2 \phi_n + \varepsilon^2 \sin^2 \phi_n)} \cos(\phi - \phi_n)} = J_0(k\sqrt{l_a^2/(\cos^2 \phi_n + \varepsilon^2 \sin^2 \phi_n)}) + \sum_{m=1}^{\infty} 2j^m J_m(k\sqrt{l_a^2/(\cos^2 \phi_n + \varepsilon^2 \sin^2 \phi_n)}) \cos(m(\phi - \phi_n)) \quad \text{Eq. 6-3}$$

This is now becoming fairly involved mathematically and a simple interpretation is not really possible. A numerical simulation will be used to illustrate the types of beam patterns that one can expect from an elliptical array. To illustrate an array of six microphones with angular spacing of 45° , 90° , 135° , 225° , 270° , 315° is chosen. The ellipse has a major axis (l_b) of 75mm and a minor axis (l_a) of 20mm. The pattern obtained for a MVDR beamformer is quite asymmetrical as illustrated in figure 6.1.

Figure 6.1 Elliptical free-field array (MVDR $\mu=0.01$)

6.2 Asymmetry Index

It is always convenient to use a single number descriptor to evaluate a certain desired or undesired property. Directivity Index is one such index. If one computes the DI of a highly asymmetrical beam pattern it could numerically have the same value as that of a perfectly symmetrical beam. The DI does not give us any information as to the shape of the main beam.

Generally in the beamforming literature, the symmetry of the main beam is not really considered. In some cases, one will look in two perpendicular planes (e.g. E and H planes

for electromagnetic antennae) but the basic assumption is perfect symmetry. There are therefore no measures of symmetry.

Two methods of measuring the symmetry of a beam seem reasonable. The basic assumption is that the beam symmetry is only of significance in a plane at a specific angle of elevation (θ) or azimuth (ϕ). In this specific case the array is on a table and an elevation of 20° above the table (reflecting plane) for the source is a reasonable choice based on typical ergonomic considerations [27].

One measure of symmetry is to simply choose a beam width and determine the difference in decibels at a specific angle away from the desired look direction (e.g. $\pm 30^\circ$). A symmetrical beam will yield a measure of 0dB and a highly asymmetrical beam will either positive or negative. The drawback of this measure is that there may be some local anomaly (e.g. a sharp null) such that the measure may be quite large although the beam may not necessarily be significantly asymmetrical.

The preferred measure is to integrate the power within the beam width on either side of the look direction and take $10 \log$ the ratio of these powers. Again a symmetrical beam will yield 0 dB but asymmetrical beams may be either positive or negative. To provide some meaning to the sign the numerator is defined as being clockwise from the desired direction so that a positive asymmetry means there is more energy in the clockwise direction [12]. Asymmetry Index is defined as:

$$AI = \frac{\sum_{i=1}^N 20 \log_{10} (F_{\theta+\theta_i} / F_{\theta-\theta_i})}{N} \quad \text{Eq. 6-4}$$

Figure 6.2 illustrates the asymmetry of one of the beams that was studied.

Figure 6.2 Asymmetry Index example

These measures provide us with objective functions, which can be used to evaluate the algorithms used to make a beam symmetrical.

The numerical scaling of the index is such that expected values are in the range of -10 to 10.

6.3 Asymmetrical shape studied

Generally devices that would house microphone arrays are not symmetrical. A business telephone is a very likely candidate. To study the effects of such a shape a very simplified and stylised shape was developed. For convenience it houses an array of six microphones. The simple shape allowed it to be easily modelled numerically (figure 6.3) and as an actual prototype using ABS plates as shown in figure 6.4.

Figure 6.3 Boundary Element mesh of asymmetrical object studied

Figure 6.4 Asymmetrical object studied

The simulation results were compared to measurements using the eight-channel data acquisition system.

The unwrapped phase follow those obtained from the simulation. One of the worst cases is illustrated in figure 6.5.

Figure 6.5 Unwrapped phase at the six microphone for a source at a declination of 60° and 330° of azimuth

The magnitude is also quite accurately modelled. In the simulation the source is modelled as a 1Pa sound pressure. The results therefore are in dBPa. In the measurements the pressure is measured relative to the reference position at 1m and thus in dB loss from the 1m position. If we assume that the source is 1Pa then the measurements should

correspond to the simulation results. Figures 6.6-6.10 illustrate the results for a source at a declination of 60° .

Given the agreement of measurements to the simulation is reasonable to simply use simulated results to explore arrays embedded on scattering objects.

Figure 6.6 Magnitude Simulation Vs Measurements $\theta=0^\circ$ $\phi=30^\circ$

Figure 6.7 Magnitude Simulation Vs Measurements $\theta=60^\circ$ $\phi=30^\circ$

Figure 6.8 Magnitude Simulation Vs Measurements $\theta=90^\circ$ $\phi=30^\circ$

Figure 6.9 Magnitude Simulation Vs Measurements $\theta=120^\circ$ $\phi=30^\circ$

Figure 6.10 Magnitude Simulation Vs Measurements $\theta=180^\circ$ $\phi=30^\circ$

Again, it is instructive to look at the phase linearity of the various microphones with respect to one of the microphones on the face when the source is placed directly in front and centre of the object. As in the case of the sphere and the truncated cone, in the higher frequencies (above 3000Hz) the phase exhibits a linear behaviour and in the lower frequencies it is significantly non-linear as illustrated in figure 6.11. In this case there is a significant jump at about 2500Hz for the microphones at the back. The measured data fits reasonably well to the simulation.

Figure 6.11 Phase variation re: reference position (mic1)

6.4 Beam patterns from an asymmetrical conformal array

One of the stated goals of this thesis is to obtain consistent beam patterns over the whole range of azimuth. The importance of this in speech acquisition has been argued in section 2.2. While it is obvious that using a free-field elliptical array significant asymmetry can occur, it must also be quantified for a conformal array.

Given significant asymmetry of the pressure variations at the chosen microphone locations, the only way to obtain reasonable beam patterns is to calculate the optimum weights using this information using an optimisation method such as MVDR. Simply applying this method to the array one obtains (not surprisingly) beam patterns that are quite asymmetrical and of varying beam width. (Figure 6.12) The asymmetry of these is more obvious in figure 6.14 with a plot of AI and DI.

Figure 6.13 illustrates beam patterns obtained using the measured transfer functions and applying the weights computed from the simulation data. Compare these to those of the simulated data in figure 6.12. There is an obvious coarseness in the angular resolution as the measurements were carried out only every 15° and the simulation is every 10° . Bearing this in mind, the three dimensional features are very similar thus validating, once again, the accuracy of the simulation to the measurements.

Figure 6.12 Uncorrected asymmetrical beam patterns

Figure 6.13 Uncorrected beam patterns - measured data

Figure 6.14 Asymmetrical beams and Symmetry Index vs. DI

6.5 Linear constraints to correct asymmetry

Several quadratic and linear constraints were considered and tried. The choice of constraints in this method is notable by its simplicity, as they are not frequency dependant. As usual we choose to make out look direction $\mathbf{d}_s$ to be unity gain. Two additional constraints are imposed to ensure symmetry and are chosen to be the difference vectors between two directions symmetrical about the look direction:

$$\mathbf{w}^H (\mathbf{d}_{s-\Delta\theta_l} - \mathbf{d}_{s+\Delta\theta_l}) = 0 \quad \text{Eq. 6-5}$$

To ensure a symmetrical beam they are set to be zero. In the specific example illustrated two pairs of difference vectors were chosen to be $\pm 30^\circ$ and $\pm 40^\circ$. Figure 6.15 illustrates the corrected beam patterns compared to the original beams of figure 6.12. The main lobe is now much more symmetrical and aimed in the proper direction. Asymmetrical sidelobes persist and in some cases have become more important. There is some widening of the main lobe. Again the measurement data is presented in figure 6.16 to illustrate the good agreement between the simulation and measurements. The degradation of the agreement between the figures can be attributed to the loss of WNG. Figure 6.17 shows this as the D.I. decreases somewhat and the A.I. is reduced and varies very little.

Figures 6.18, 6.19, and 6.20 show the beam pattern in the plane of interest for 60° , 90° , and 120° . The shaded area is used for the S.I. calculations. The correction of the symmetry is now more obvious as are the artefacts of increased beam width and increased side lobes.

Figure 6.15 Corrected beam patterns

Figure 6.16 Corrected beam patterns - measured data

Figure 6.17 Asymmetry Index and DI for beams before and after correction

Figure 6.18 beam pattern correction at 60 degrees

Figure 6.19 beam pattern correction at 90 degrees

Figure 6.20 beam pattern correction at 120 degrees

6.6 Conclusion

The problem of asymmetrical arrays can now be quantified easily by the use of the Asymmetry Index. This provides one with a quick tool to quantify the asymmetry and to evaluate corrections.

Correcting the asymmetry proves to be reasonably easy by the use of linear constraints that operate over the full bandwidth of interest. One can therefore correct asymmetry while still using other constraints to obtain frequency dependant characteristics independently. Consequently, the first question of the thesis is well addressed. As in any constraint problem care must be taken that the degradation in directivity and WNG is a reasonable compromise for the improvement in symmetry.

Results from the Boundary Element simulation were again shown to be in very close agreement to measurements thus providing very conclusive validation of this method of design for conformal microphone arrays.

Chapter 7

Conclusions and Future Work

The fundamental contribution of this work has been to show the benefits of exploiting the physical acoustics (scattering) of the housing of a microphone array to enhance the beamformer performance. Specifically:

1. Use of a scattering conformal array to overcome the spatial aliasing that is found in free field arrays. [11]
2. Developing a novel index to describe the asymmetry of the main lobe in a beamformer [12]
3. Using a simple linear constraint to correct an asymmetrical beamformer [12]
4. Validating a simulation technique by computing two significantly varied shapes and experimentally validating the results [11,12]
5. Providing a synthesis of beamforming theory and its interaction with scatterers.

This involved simulation of known equations to illustrate these concepts.

It has been clearly shown that a scattering body not only provides significant pressure variations but that in the scattering frequencies the phase exhibits non-linear behaviour. This is accounted for in the simulations that were performed so that good agreement between the simulations and actual measurements were possible.

There have been two publications [11,12] and a conference abstract has been submitted to explore in greater detail the phase non-linearities created by the scattering object.

In the future it would be interesting to study this phenomena further by considering a simple spherical scatterer. Once one has developed analytical solutions their validity can then be studied on more complex shapes. As this requires a good understanding of acoustics, signal processing and math it would require a doctoral thesis.

From an applied physics perspective (engineering) this thesis has shown that it is valid to work in the simulation domain to design microphone arrays on scattering objects. This is important as it permits the investigation of many shapes without the costly prototyping stage. The rudimentary explanation of the acoustical phenomena of the scatterer also helps one to design a real product.

Appendix A

LabVIEW Programmes

read Chrip and make rpt type file.vi

2cards multich filter & sum read filfile.vi

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