Chapter 1
Introduction – Why Have Line Arrays?

Line Array History

Duran Audio, [13] define a line array as “a number of loudspeaker elements arranged in a column.”

Line arrays are not a new invention. A column of loudspeakers has one particularly interesting benefit, the way in which the output of the loudspeakers combines under certain circumstances to control the shape of sound along the vertical axis of the column.

Speech Intelligibility

Speech intelligibility is a measure of how comprehensible a sound system is in a given environment when used for the reproduction of the speech. As well as the irritation of not being able to understand announcements, there is a more serious health and safety issue when a sound reinforcement system is to be used to make emergency announcements. These must be clearly understood, with the issue being even more critical still when announcements are made at international airports where the listener could be listening to an announcement which is not in there first language.

A line array can help to improve speech intelligibility. The narrow vertical beam of sound that an array produces means that it can be installed in such a way as to direct sound where it is required, rather than at acoustically reflective surfaces, which leads to less reflected sound and therefore provides greater speech intelligibility.

Architectural Suitability

The fact that a line array is a long narrow enclosure that is vertically mounted means that it is easy to either conceal the array or to make it blend in with its architectural surroundings. This makes line arrays an unobtrusive solution to sound reinforcement for speech and background music in difficult acoustic environments, compared to other more bulky solutions such as constant directivity horns.

Control Technology

Modern control techniques using digital signal processing have lead to a resurgence of interest in line array technology. The fact that digital systems can exert control over audio signals that was never possible with older analogue techniques has lead to line arrays being implemented in applications from speech reproduction to full range touring systems for large live music events. Companies such as L Acoustics have developed systems that are so complex that they refuse to hire out the system unless an engineer that has been trained by them accompanies it.

Duran Audio, [13] “The SPL (sound pressure level) produced by a line array falls off much more gently than that of a conventional system; just 3dB per doubling of distance instead of 6dB … it only occurs in the nearfield. Beyond this point (where the listener enters the far field), the drop off mimics that of a conventional system.” This reduced rate of SPL drop off means that as Duran Audio, [13] say, “It is no longer necessary to deafen the front row in order to project to the back.”

As line arrays naturally produce a very directional sound beam, a useful benefit of the line array pointed out by Bauman, [20] is “The high degree of SPL (Sound Pressure Level) rejection obtained outside the coverage pattern of the system. Nominally as high as 20dB, this permits the installation of an … array behind or above microphones, with exceptionally high feed back immunity.”

Chapter 2

General Line Array Theory

For the purposes of speech reproduction, de Vries and van Beuningen, [2] state; "A public address array only needs a frequency response of 300 Hz to 5 KHz."

One way to consider a line array is as a non-ideal line source. A line source produces what is known as a cylindrical wavefront (figure 2.1).

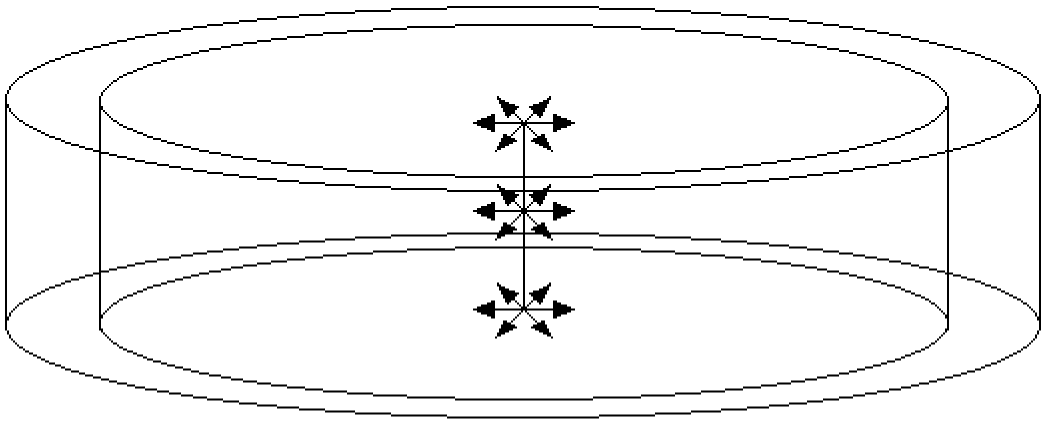

Figure 2.1 Cylindrical wave

Capel, [33] says of line sources, " A line source radiating sound through 360 degrees does so in the form of expanding concentric cylinders rather than spheres. Sound energy is concentrated within the horizontal plane with very little radiated vertically." The attenuation with distance is less than for a point source because the same amount of energy (for the purposes of comparison) is radiated through a smaller area.

The 3 main aspects of column design according to van der Werf et al, [1] are:

- "Designing for a vertical coverage angle that is independent of frequency.
- Designing for acceptable side lobes.
- Designing for maximum coverage area."

Speech Intelligibility

Speech intelligibility is a measure of how well a system for reproducing speech can be understood. It is measured in a number of ways including the Speech Transmission Index (STI), the Rapid Assessment of the Speech Transmission Index (RASTI) and the percentage articulation loss of consonants (%ALCONS).

It is possible for a customer to specify a sound system in terms of its speech intelligibility, and there are legal requirements for the speech intelligibility of a system if it is to be used to make emergency announcements.

The points in this section are made in order to emphasise the importance of speech intelligibility. A full assessment of the finished systems speech intelligibility is beyond the scope of this project and any research in the area of speech intelligibility will be used to guide the design of the system, particularly in the areas of beam steering and directivity control.

According to Mapp [31] p.1250, speech intelligibility is dependent on many characteristics including:

- “The sound system bandwidth and frequency response.
- The systems loudness and signal to noise ratio.
- The reverberation time (RT60) of the space.
- The volume, size and shape of the space.
- The distance from the loudspeaker to the listener.
- The direct to reverberant ratio.
- The talker’s annunciation and rate of delivery.
- The systems distortion (harmonic distortion and intermodulation distortion).
- The systems equalisation.
- Uniformity of coverage.
- The gender of the talker.
- The vocabulary and context of speech information.
- The talkers microphone technique”.

Mapp [31] also suggests a frequency range for speech reproduction of 100Hz to 8 KHz, differing from that quoted by Duran Audio in chapter 1. However, Mapp analyses the spectrum further in terms of speech intelligibility (table 2.1).

Table 2.1 Octave band contributions to speech intelligibility

Percentage	1	3	15	20	30	25	6
Band	125	250	500	1K	2K	4K	8K

Mapp, [31] also states that, “Vowels contain low frequencies with more energy, but the consonants that contain higher frequencies and correspondingly less energy contribute far more to speech intelligibility”.

Among the general desirable attributes for any sound system, van Beuningen and Start, [5] list:

- “An evenly distributed SPL.
- Spectral uniformity.
- The direct SPL should be sufficiently higher than the diffuse SPL and the ambient noise level.”

van Beuningen and Start, [5] suggest that line arrays can meet the last condition by, “projecting the sound … onto the relatively absorbing audience area while avoiding other surfaces.”

To gain optimal speech intelligibility, van Beuningen and Start, [5] state “There is a need to reduce the total amount of acoustical power and the level of discrete reflections (arriving after approximately 50 mS) without sacrificing direct SPL." As the discrete reflections will detract from speech intelligibility, a phenomenon known as the Haas effect, where reflections arriving up to around 30mS after the direct sound combine with it to increase intelligibility and those arriving later detract from it.

van der Werf et al, [1]also say, "Speech intelligibility depends on the direct to reverberant sound ratio. The second most important parameter is the signal to noise ratio."

van der Werf et al, [1] "If the acoustical properties of a hall are fixed, there are just a few possibilities for controlling speech intelligibility. The distance to a source can be reduced, or the number of sources can be reduced or the directivity factor has to increase."

When van der Werf et al, [1] say, "The only way to improve intelligibility is to enlarge Q without reducing the coverage area." The Q they are talking about is a measure of how directional a sound system is.

In international airports, speech intelligibility is critical as, "as visitors are 'often not addressed in their native language' which leads to a reduction in speech intelligibility." van der Werf et al, [1]

A column system was implemented at Schiphol airport in Amsterdam by Duran audio, van der Werf et al, [1] say, "The system is called 'the whispering voice system' because it is designed for the lowest possible signal to noise ratio without sacrificing intelligibility. In quiet moments, the signal level is 65 – 70 dBA and at rush hours the signal is just riding above the environmental noise and therefore the announcements are never annoying or unintelligible."

Directivity

The intensity of a sound source in a free field is defined by:

$$\text{Intensity} = \frac{\text{Power}}{\text{Area}} = I = \frac{W}{4\pi r^2}$$

The directivity factor Q is defined by Beranek [30] p.109 as, "The ratio of the intensity on a designated axis of a sound radiator at a stated distance r to the intensity that would be produced at the same position by a point source if it were radiating the same acoustic power as the radiator. Free space is assumed for the measurements. Usually the designated axis is taken as the axis of maximum radiation, in which case Q(f) always exceeds unity." Which can be expressed as:

$$Q = I_\theta / I_{mean}$$

Where I_θ is the sound intensity in a particular direction and I_{mean} is the sound intensity for a uniform source.

The directivity index of a sound source is:

DI = 10 log Q

Point Source
For a point source in a free field in a homogenous medium (the sound can travel at equal speed in all directions) a point source will produce a spherical wavefront (figure 2.2).

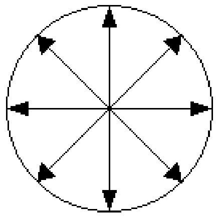

Figure 2.2 Point source

Q =1 so DI = 10 log 1 = 0

Hemispherical Source

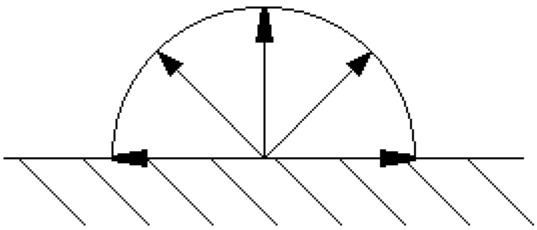

Figure 2.3 Hemispherical source

For a hemispherical source, the sound is radiating into half of the area of a source radiating into a free field, so Q is 2. This gives a figure for the directivity index of:

DI = 10 log Q = 10 log 2 = +3dB

Quadra spherical Source

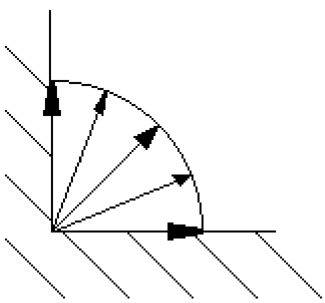

Figure 2.4 Quadra spherical source

If the area into which the sound is radiating is halved again, the value of Q increases to 4. This gives a directivity index of:

DI = 10 log Q = 10 log 4 = +6 dB

For a source in a corner

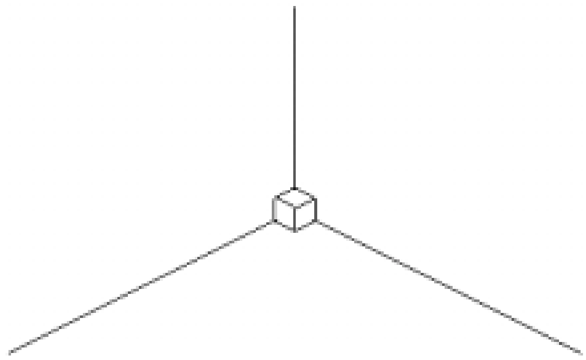

Figure 2.5 Source in a Corner

For a source in a corner, the area into which the sound is radiating is halved again when compared to a quadraspherical source, giving a Q value of 8. This produces a directivity index of:

DI = 10 log Q = 10 log 8 = +9 dB

Van der Wal et al, [6] state that the directivity of a line array "Is strongly related to the dimensions of the device in relation to its wavelength."

The 3dB drop in SPL per doubling of distance exhibited by a line array stops according to van Beuningen and Start, [5] when "In the far field of the source, the wave propagation is purely spherical, i.e. the inverse square law holds," which relates to a more usual 6dB drop in SPL per doubling of distance.

One of the reasons for using digital signal processing with a line array is to optimise its performance, van Beuningen and Start, [5] suggest that one of the ways in which this can be achieved by the DSP is to "shape the main lobe while reducing the level of all other lobes."

Although line arrays have radically different vertical characteristics from the individual drivers, van der Werf et al, [1] say, "The directional properties of a sound column in the horizontal plane are the same as the horizontal properties of the individual source."

Beaming

When the individual drivers in a line array combine their outputs, they do so in a destructive and constructive manner, the result being a narrow vertical beam of sound. Duran Audio, [13] state, "The effect of a line array on the vertical dispersion of a system is dramatic – even with modest array lengths, a beam width of only a few degrees is achieved."

One of the goals of the project was to use digital signal processing to maintain the vertical directivity of the line array, as in an uncontrolled array the directivity is strongly dependant on frequency. Duran Audio, [13] suggest that "Olson research showed that the directivity of the loudspeaker was controllable by varying the length of the array – but only providing the distance between the acoustical centres of adjacent drivers was smaller than the wavelength of the sound being produced." In other words for constant directivity, low frequency signals need a long array length and high frequency signals need a shorter array length.

van Beuningen and Start, [5] say "Grating lobes which are repetitions of the main lobe, originate from the fact that the array is too coarsely sampled (i.e. the distance between the elements is too large compared to the wavelength)." So when designing the array, it is important to get the driers as close together as possible to minimise undesirable secondary lobes.

Methods of Maintaining Constant Directivity Independent of Frequency

Power Tapering

Duran Audio, [13] "By using multi tap transformers, the outer hf (high frequency)drivers of the array could be turned down to simulate a shorter column, while lf (low frequency)drivers are run at full power to maximise the array length. On a larger scale, where multiple cabinets are involved, the feed to each cabinet is filtered to adjust the length of the array at any given frequency."

Capel, [33] explains power tapering, which "Consists of feeding the maximum power to the central units, with a gradually reducing amount proceeding outward, ending with minimum power at the ends."

The necessity to attenuate the signal at high power levels, using expensive multi-tap transformers, makes it an uneconomical method of maintaining constant directivity for this project.

Frequency Shading

When using frequency shading to maintain constant directivity, each channel of the array is low pass filtered with a different cut off frequency for each channel. This serves to change the effective length of the array depending upon the frequency and wavelength of the input signal. For low frequencies, more drivers are active, making the array behave in a similar fashion to a larger diameter single driver (at least along the vertical axis), thus narrowing the beam in the vertical direction. As the frequency increase and the wavelength decreases, the beam is prone to collapse as it becomes extremely narrow, so some of the drivers are turned off by the low pass filters, making the effective length of the array shorter, and the beam broader, i.e. the beam width is maintained.

Ballou, [31] expresses the idea from a different perspective, "We want the ratio between the effective length of the array and the wavelength of the sound to be invariant with respect to frequency".

van der Wal et al, [6] suggest, " An approach to array design which results in the construction of transducers with a frequency independent directivity pattern. Frequency independence is achieved by truncating the effective array aperture in a frequency dependent way"

One interesting aspect of frequency shading pointed out by de Vries and van Beuningen, [11] is that, "As more transducers are turned on for low frequency signals, the apparent source of the sound moves."

van Beuningen and Start, [5] state that, "To obtain a frequency independent shape of the main lobe, the effective array length should be made inversely proportional to the frequency."

The filtering to maintain directivity has to perform several functions, which van Beuningen and Start, [5] list as:

- "Compensate for the non uniform density of the sources.
- Correct for the varying number of sources as a function of frequency.
- Introduce a position dependant weighting (window) to reduce the level of the side lobes."

The first point is not relevant to the project as budgetary constraints mean that a logarithmically spaced array was not a viable option due to the limited number of drivers used in the project. The second point is important to consider as at low frequencies, all of the drivers will be active, meaning that there is significantly more output at low frequency than there is at high frequency when there could be only one driver in operation. The third case makes reference to the type of window applied to the digital filter (more of this in the chapters on software), which will affect the formation of secondary lobes as well as the shape of the main lobe.

The Gain Correction Filter

As stated in the previous section van der Wal et al, [6] point out that, "For low frequencies, more transducers will be active than for high frequencies." They then go on to say in the same paper that a, "weighting has to be applied to compensate for the large number of transducers that contribute at low frequencies compared to the smaller number of transducers that contribute to higher frequencies." This gain correction filter could be derived by measuring the un-weighted response of the array when the constant directivity filters have been implemented and developing an inverse filter from the experimentally measured response.

Beam Steering

The ability to steer the main lobe of the array is desirable according to Duran Audio, [13] as, "The conflict of form versus function often means that an installed PA product has to be positioned in a less than ideal location so the concept of a steerable array has instant appeal," because the sound can be positioned where it is required for maximum efficiency without having to excessively interfere with the aesthetics of the space.

It is possible to change the vertical angle of the main beam by applying a different delay to each of the drivers. Instead of the sound leaving all of the drivers at the same time, the delayed drivers will produce the sound at progressively more delayed times causing the wavefront to stop being parallel to the front of the array.

One side effect of large steering angles noted by van Beuningen and Start, [5] was, "A tendency for increased level of the side lobes and larger width of the main lobe as the absolute value of the steering angle increased."

Line Array Cabinet Design

All quotes in this section are from Dickason, [36].

Although enclosure design software was explored as part of the research; it was not used for two reasons. Firstly, the software required the Thiele Small parameters for the loudspeakers used in the box, which were not available from the manufacturer due to the low cost of the drivers used in the array. Designing with Thiele Small parameters is primarily about extending the bass response of the system, which was not a high priority for this project as the design was being set up for a narrower bandwidth speech only system rather than for full range music reproduction system.

When designing an enclosure it is important not to have any internal dimensions which are the same or integer multiples of each other as this causes reinforcement of resonant frequencies which is undesirable when an ideal enclosure has a flat frequency response. However Dickason, [36] states, "Box dimension ratios will be a secondary effect as long as the enclosure is appropriately damped with absorbent material." Which is to say that the energy peak at resonance is less critical if the cabinet is appropriately damped. It is still good design practice to avoid integer box dimension ratios.

Pipes develop standing waves that are dependant on the internal dimensions of the pipe. Dickason, [36] notes that "Long and narrow enclosures which can be prone to pipe resonance's, which can be broken up by using internal reflecting baffle panels." Which is especially relevant to a line array, as they are by definition long, narrow, enclosures.

Factors that can minimise enclosure vibration according to Dickason, [36] include "the choice of wall material …bracing techniques … and driver mounting techniques."

The simplest way to minimise enclosure vibration due to material selection is to adopt a "brute force technique," (Dickason, [36]) " which dictates the use of thick walled high density materials such as 1" mdf in conjunction with extensive bracing."

When a panel is braced the effect of the brace is to "divide the wall into two quasi independent panels, each having its own resonant frequency." (Dickason, [36])

Bracing the inside of an enclosure provides mechanical strength as well as stopping panels from resonating freely and causing peaks in the frequency response of the array. Dickason, [36] discusses the different types of brace (figure 2.6) "The horizontal brace can be used to break up the enclosure resonance around the girth of the box" and "the shelf brace which is a combination of the horizontal and cross braces. The shelf brace is basically a solid panel which is attached to three of four sides of the enclosure."

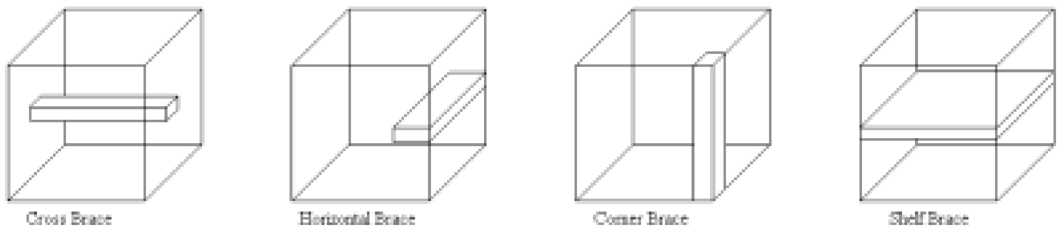

Figure 2.6 Bracing techniques

Although Dickason, [36] suggests "A ¼ inch bead of silicon placed on the driver mounting flange will provide an airtight seal as well as a degree of vibration damping." The decision was taken not to seal the drivers initially to allow easy access to the interior of the prototype as they could be sealed at a later date if vibration became an issue.

Chapter 3

Line Array Enclosure Design, Build and Preliminary Testing

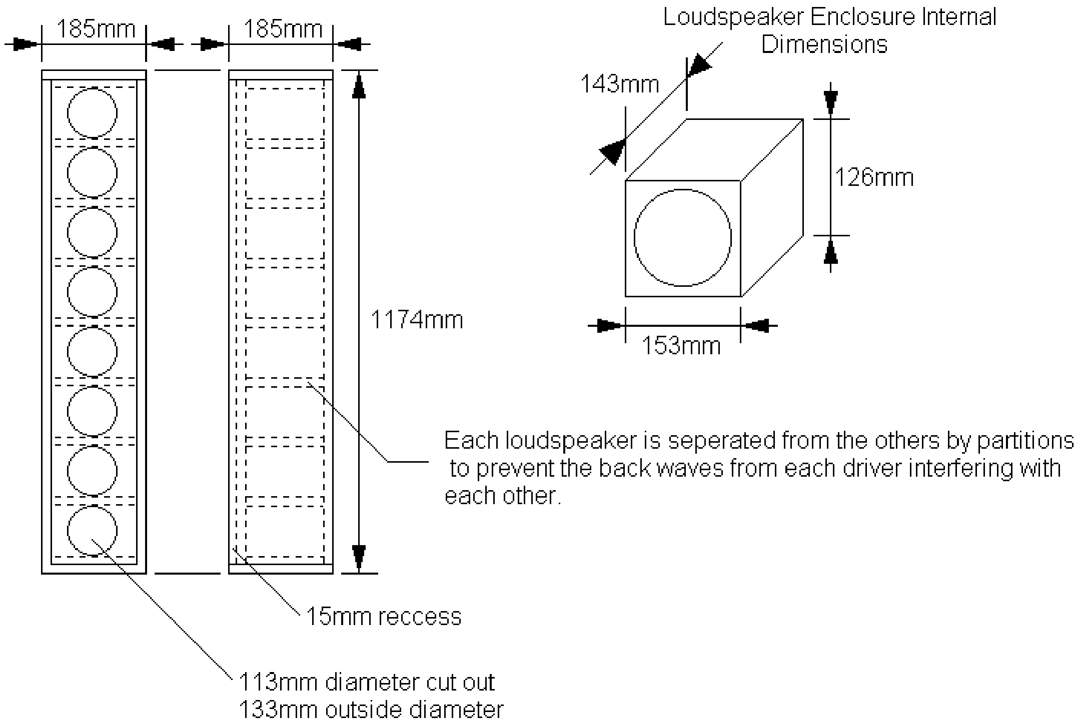

Figure 3.1 Enclosure dimensions

The enclosure was constructed from 18mm medium density fibreboard (mdf) as it had high density and uniform young's modulus (no grain considerations). As the enclosure was not intended for exterior use, the poor performance of the material under wet conditions was not significant. By using thicker sheet material, it was unnecessary to use additional bracing as the partitions between each loudspeaker compartment provided the necessary structural rigidity (figure ***).

By ensuring that none of the loudspeaker compartment internal dimensions were multiples of each other, the possibility of reinforcing resonant frequencies was minimised.

The panel layout was planned to minimise wastage prior to cutting. By planning the layout, the MDF requirement was minimised to a half sheet measuring 1200mm by 1200mm (figure 3.2).

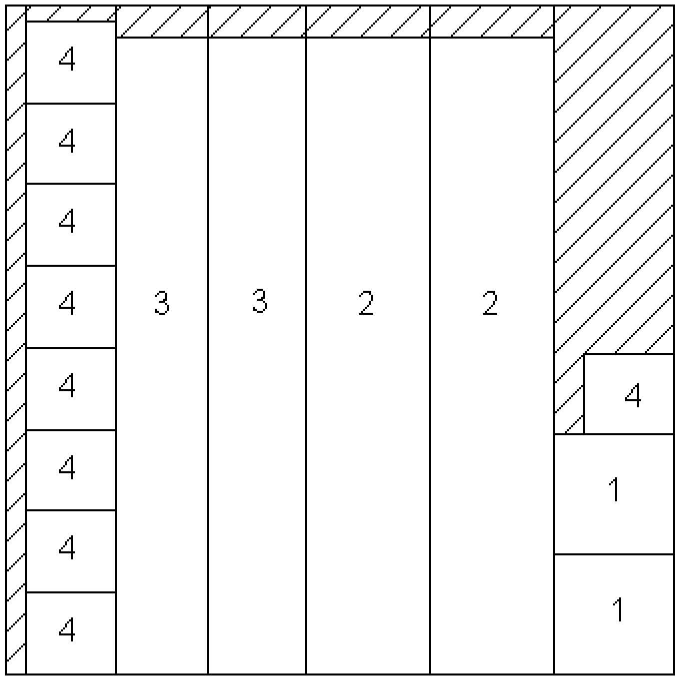

Figure 3.2 MDF sheet layout

Table 3.1 Panel description key

Part Number	Dimensions	Description
1	185x185mm	Top/ Bottom
2	185x1174mm	Sides
3	153x1174mm	Front/ Back
4	153x138mm	Internal Partitions

The mdf was cut using a suitable dust extraction system and an appropriate dust mask was worn at all times when working with the material due to the carcinogenic nature of the dust particles.

The loudspeakers were front mounted for ease of maintenance using 5mm tee nuts and bolts, as normal woodscrews would be prone to failure with repeated removal/ reinsertion.

All joints were glued with wood glue and then screwed together with 38mm number 8 woodscrews. All joints in the cabinet were sealed using silicon in order to make them airtight and to prevent the rear waves from the loudspeakers from interfering with the front waves and/ or the other loudspeakers.

The Thiele-Small parameters were unavailable for the budget loudspeakers used in the project, but were unnecessary as they are used in conjunction with enclosure design software for extending low frequency response. As the project was concerned mainly with speech reinforcement, achieving an extended bass response was not a high priority.

The loudspeaker compartments were lined with an acoustic wadding to absorb the energy emitted from the rear of the loudspeakers.

Each loudspeaker had a separate feed to allow maximum flexibility with the control of the array from the DSP.

The cables from the loudspeakers were rated at 3 Amp and were terminated into 4mm binding posts on two recessed dishes on the rear of the enclosure. Two dishes had to be used due to the small size of the individual loudspeaker compartments. Each dish required a cut out of 115x65mm. Six 4x12mm japanned screws secured each of the dishes.

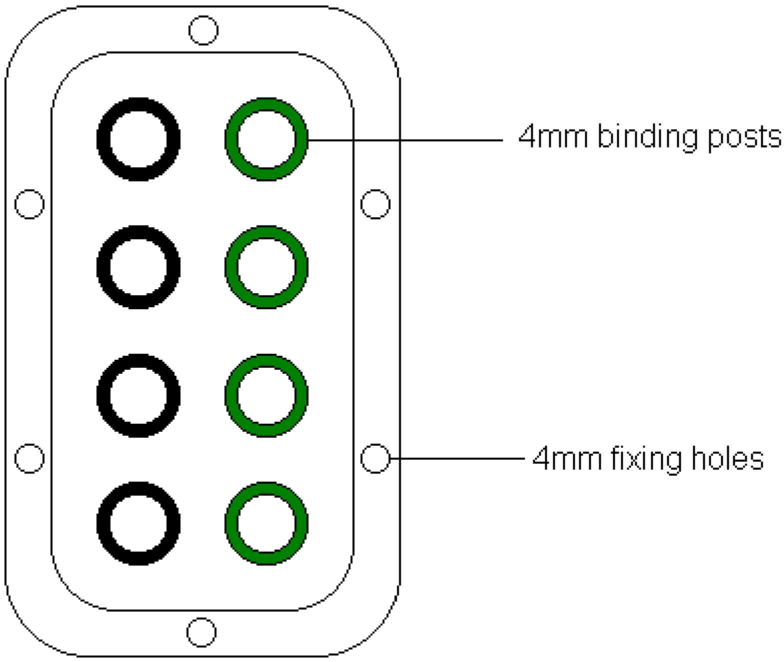

Figure 3.3 Recess dish detail

The enclosure had four large rubber feet fixed to the bottom in order to isolate it acoustically from the ground on which it stood.

Construction Details

1) All materials were collected and the design was finalised once the dimensions of the loudspeakers had been checked (large variation in dimensions was anticipated due to the budget nature of the drivers – which was not found to be the case).

2) The back of the enclosure was marked out with the positions of the internal partitions and the location of the cut outs for the recessed dishes for the electrical connections.

3) The internal partitions were notched in order to allow either one or two cables to pass through from the recessed dish to each of the loudspeaker compartments.

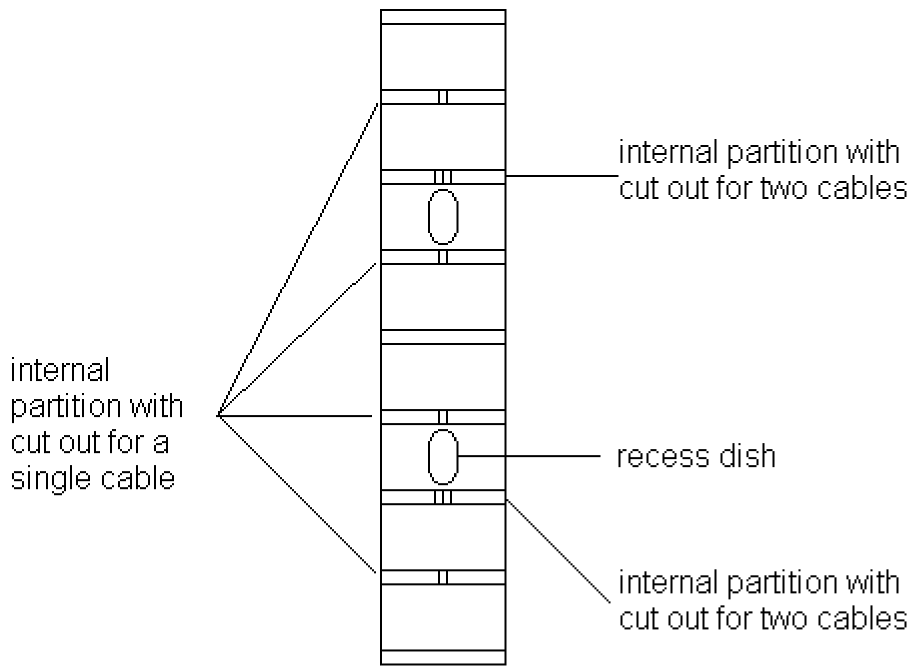

Figure 3.4 Internal isolating panel plan

The notch for a single cable was 5x3mm and the notch for a double cable was 10x3mm.

4) The back was pilot drilled and countersunk to take the screws for the partitions.

5) The partitions were attached to the back panel using glue and woodscrews.

6) The loudspeaker cables were run next as it was an awkward job that would have been exacerbated by the attachment of the sides. Excess cable was left at both ends to allow easy termination.

7) The edges of the back panel/ partitions were sanded back to ensure that the attachment of the side would achieve a flush fit.

8) One side was marked out, pilot drilled and countersunk, then glued and attached to the back panel/ partitions.

9) Before the second side was attached, the front panel was offered up to the enclosure and the position of the partitions was marked on it to ease the marking out of the cut outs on the front panel.

10) The back/ internal partitions were sanded back on the other side to achieve a flush fit when the second side was attached.

11) The second side was then attached in a similar fashion to the first.

12) The internal joints of the "back box" were then sealed with a flexible silicon sealant to ensure they were airtight, with particular attention to the areas where the cables run from one compartment to the next.

13) The top and bottom panels were pilot drilled, countersunk, glued and screwed into place.

14) The recessed dishes were drilled out to take the connectors, the connectors were attached and the solder tags attached to the appropriate speaker cables.

15) The recessed dishes were then attached to the enclosure.

16) The acoustically absorbent material (from Maplins electronics) was stapled into each internal compartment.

17) The front panel was marked out to take the loudspeakers and the holes cut using a jigsaw with a dust extraction system.

18) The speakers were offered up individually and the location of the mounting holes spotted through onto the panel.

19) The holes for the tee nuts were drilled and the tee nuts hammered home.

20) The front panel was then pilot drilled, counter sunk, glued and screwed onto the enclosure.

21) The front panel was sealed externally using silicon sealant.

21) The speakers were attached to their respective cables and then they were bolted into the enclosure.

Preliminary Enclosure Testing

The finished enclosure was tested using the following procedures:

Each speaker was tested individually with a digital audio source and power amplifier. The array was then tested as a whole with all loudspeakers connected to power amplifiers and the audio source.

The array was attached to four stereo amplifiers via a 12-meter cable run 0f 0.5mm^2 2-core cable.

Further testing on the array is detailed in chapter 6.

Chapter 4

DSP Theory

The chapter will be broken down into:

- A brief review of the Simulink development package.
- Delay Theory – A look at how delays are implemented in a digital environment.
- Filter theory – A general review of different types of filter including analogue, finite impulse response and infinite impulse response filters.

Simulink

To quote the Simulink help files, "Simulink is a software package for modelling, simulating and analysing dynamic systems (systems whose output changes over time)….For modelling, Simulink provides a graphical user interface (GUI) for building models as block diagrams …Simulink includes a comprehensive block library…Models are hierarchical so you can build models using a top down or a bottom up approach."

The main advantage of Simulink for the purposes of this project is that models can be built, altered and run very quickly, without having to de bug code, as the code has already been written as S-files that can be linked together in a graphical environment that is simple and intuitive to use.

Delay Theory

Delays are straightforward to implement in the digital domain. The continuous signal coming into the system is sampled at regular intervals and represented by a number at each point in time that represents the amplitude of the signal at that point in time. This list of numbers that represent the signal is stored in memory for the purposes of processing and eventual output. Many digital signal processors can access the data stored in memory using what is known as Modulo addressing which models the memory as a circle so that when the last address is reached the memory pointer wraps around to the first address automatically, the programmer does not need to worry about it.

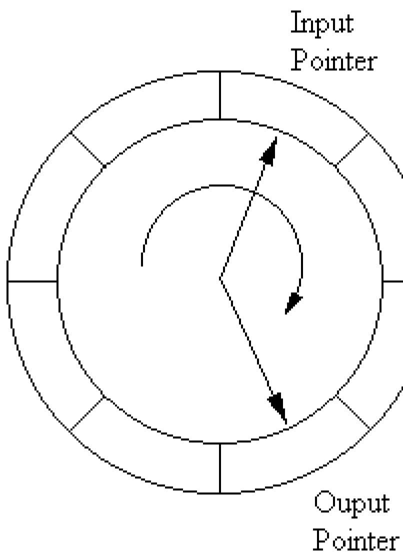

Figure 4.1 Circular buffer

The delay effect is achieved by using one of these circular buffers (figure 4.1) with two pointers, one to write sample data into the buffer and one to read sample data from the buffer. The length of the delay is dictated by the gap between the pointers and the maximum delay is dictated by the size of the circular buffer, as the two pointers can only get so far apart.

Filter Theory

Analogue Filters

The traditional approach to filtering a signal at its simplest was to construct a network of passive components. The filter shown in figure 4.2 was modelled using an electronic design package called EDWIN.

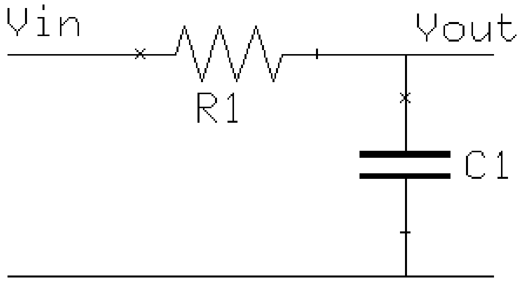

Figure 4.2 Simple analogue filter

The advantage of modelling the filter is that it’s frequency and phase response can be generated without having to take extensive experimental results. The frequency and phase response for the filter are shown in figure 4.3.

Smith [28] says of analogue and digital filters, “Analogue filters are cheap, fast and have a large dynamic range in terms of both amplitude and frequency. Digital filters in comparison, are vastly superior in the level of performance that can be achieved.”

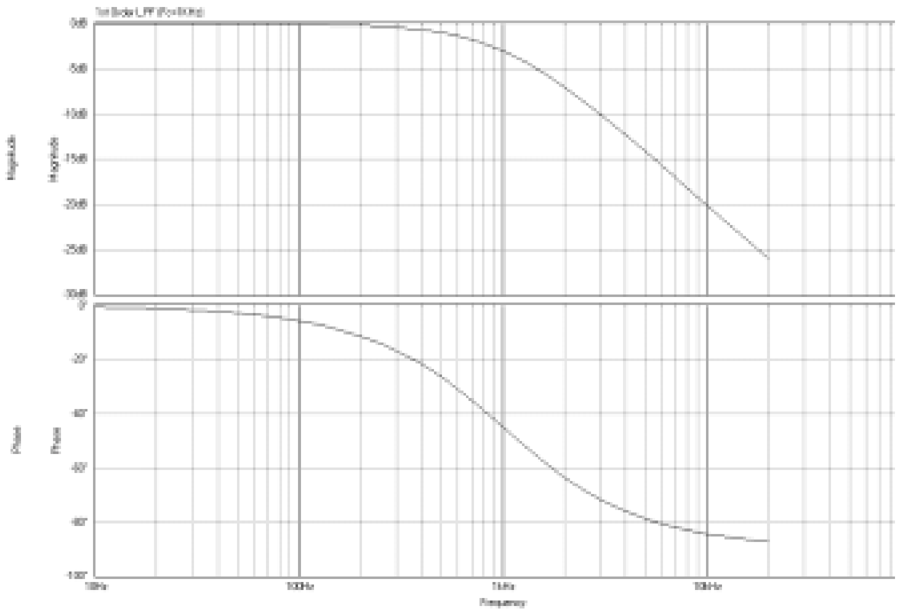

Figure 4.3 First order analogue filter frequency and phase response

As can be seen from the graphs, the filter allows low frequency signals to pass, whilst attenuating high frequency signals (a low pass filter), rolls off at a shallow rate and has a non-linear phase response. One of the other issues associated with analogue filters is that variations in component tolerance mean that the filter parameters are not consistently reproducible, that is they will vary with every filter that is constructed. Another problem is that of the filter order. The filter order dictates how quickly the frequency response will roll off, with an ideal filter rolling off vertically. The filter order can be increased by increasing the number of energy storage components in the circuit (in this case capacitors). The frequency and phase response of a passive second order filter is shown in figure 4.4.

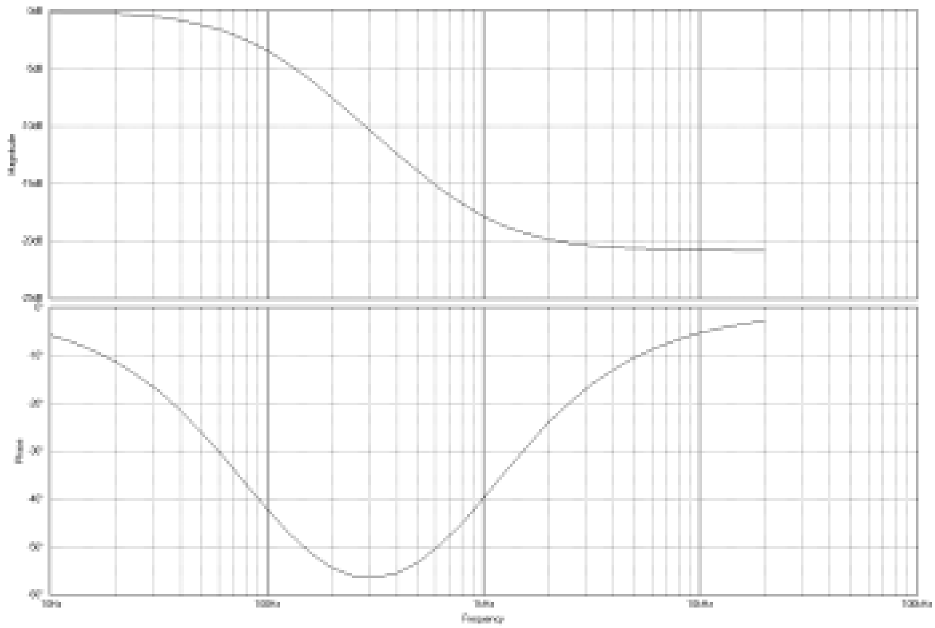

Figure 4.4 Second order frequency and phase response

As can be seen in figure 4.4, the frequency response rolls off at a steeper rate, but the trade off can be seen in terms of the increasingly non linear phase response. As more energy storage components are added, the frequency response roll off gets steeper as the phase response becomes increasingly non linear. The other down side to analogue filters is that as the filter order rises, the filters become increasingly unstable, with a tendency to break into oscillation with filters above fourth order being very unusual.

Digital Filters

While analogue filters operate on a signal, which is continuous in time, digital filters operate on signals, which are discrete in time. As can be seen in figure 4.5 the discrete time representation of a signal is achieved by measuring the instantaneous amplitude (or sampling) the continuous signal at regular intervals.

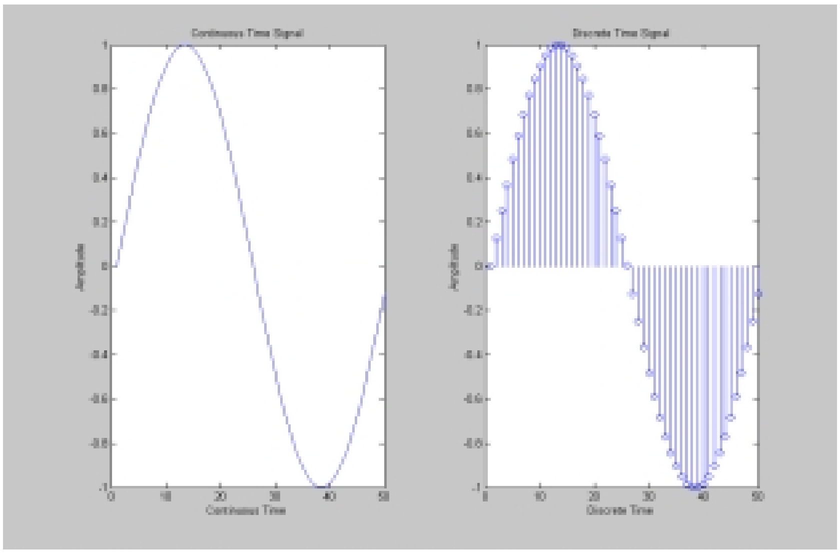

Figure 4.5 Discrete sampling of a continuous signal

One method of obtaining the frequency response of a system (or filter) is to apply a broadband noise signal (figure 4.6) to the input of the system and to measure the output of the system. The broadband noise signal is made up of random or pseudo random noise that covers the entire audio spectrum (20 Hz to 20 KHz).

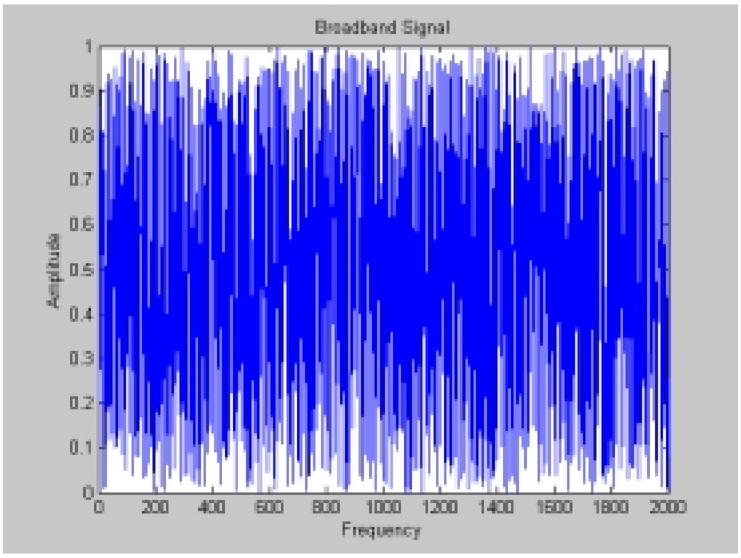

Figure 4.6 Broadband noise signal

The time domain representation of a suitable system test signal is called an impulse. An ideal impulse is shown in figure 4.7.

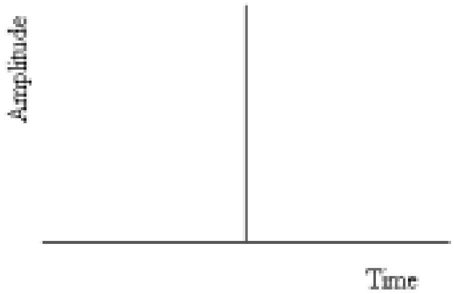

Figure 4.7 Ideal impulse

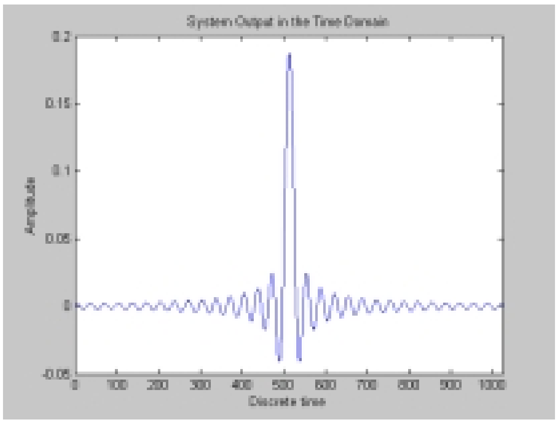

Figure 4.8 Impulse measurement of a filter

As can be seen from figure 4.8, the output response continues in both directions, infact it continues indefinitely in both directions. This is not useful in digital terms and so the output response must be shortened to a finite length. If the response is chopped off at the outer edges of the window, the frequency response develops undesirable artefacts due to the sudden transition at either edge of the response where the signal drops to zero (figure 4.10). The method used to improve this situation is to apply a window to the response (figure 4.11). If the window is applied to the impulse, the results can be seen in figure 4.12. As can be seen in figure 4.12, the edges of the impulse are smoothed out resulting in a frequency response with a much smoother pass band and stop band. In this case a Blackman window was used, but there are many different types of window that are categorised by the effect that they have on the signal.

So to summarise, a digital filter in the time domain can be represented by an impulse response, which is a list of filter coefficients, referenced to a discrete time index (like the graph in figure 4.10). By performing a Fourier transform on this impulse response, a more recognisable plot of frequency against amplitude in the frequency domain can be obtained. It is also a straightforward matter to perform an inverse Fourier transform on a frequency domain representation of a filter to obtain its impulse response (figure 4.9).

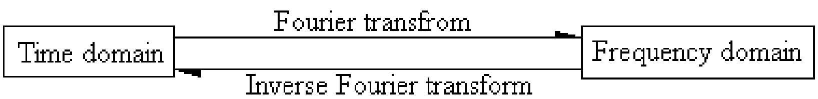

Figure 4.9 Relationship between the time and frequency domains

It follows then that if an existing filter can be modelled into a discrete time indexed list of numbers, then digital filters can be designed as a list of numbers or an impulse response. The next task then is to apply the filter to a signal.

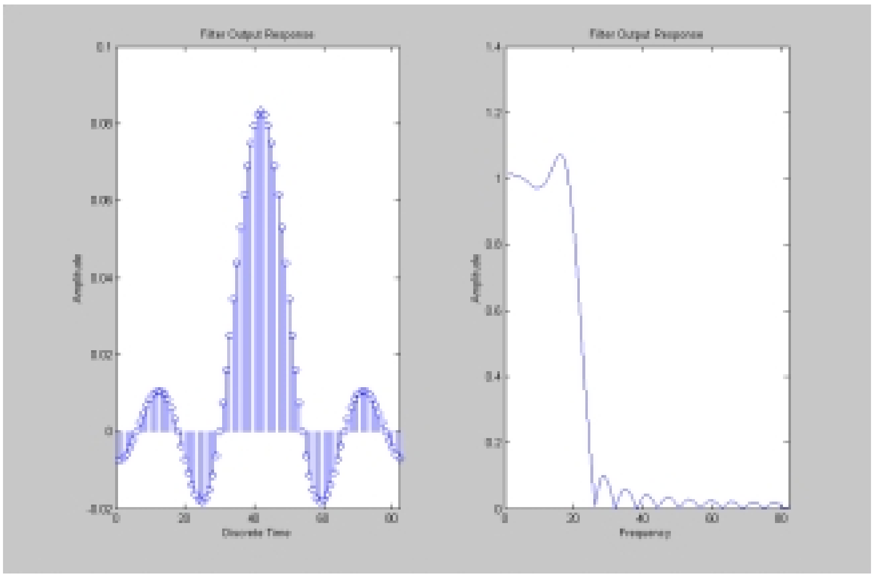

Figure 4.10 Unwindowed filter response

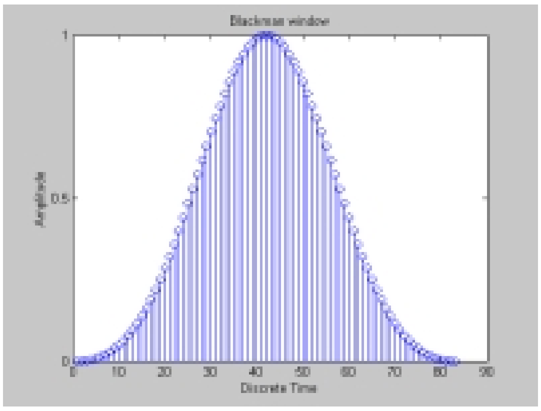

Figure 4.11 Blackman window

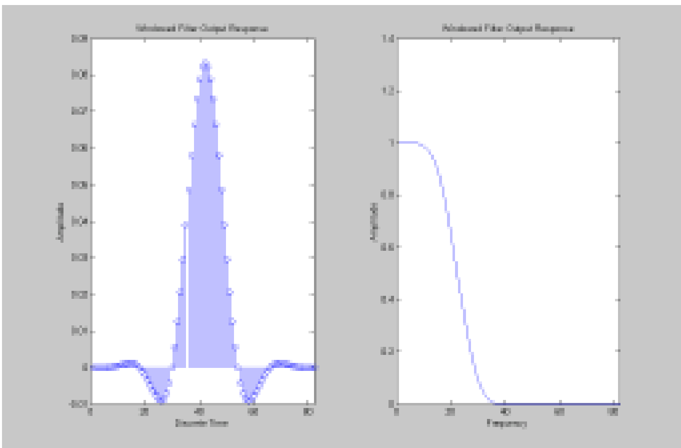

Figure 4.12 Windowed filter response

The filter can be applied in one of two ways, in the time domain or in the frequency domain.

Time Domain Filtering

Filtering in the time domain is carried out by a process known as convolution (figure 4.13).

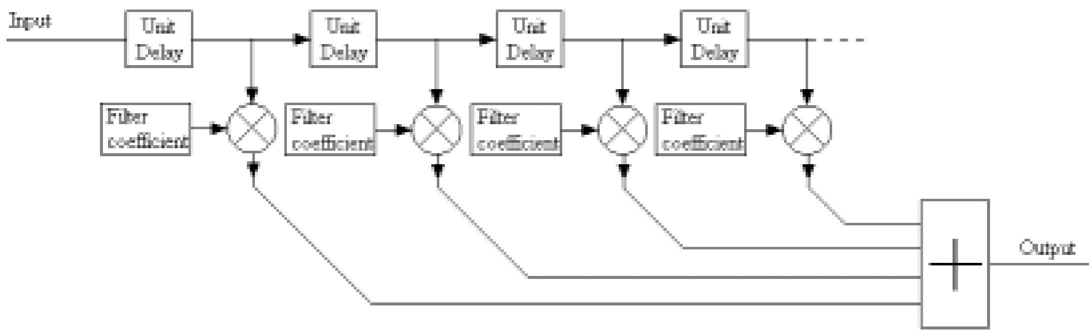

Figure 4.13 Convolution

The stream of input samples enters the filter (figure 4.13) and in the first pass, the first sample is multiplied by the first filter coefficient and the result sent to the output. In the second pass, the second sample is then multiplied by the first filter coefficient and the first sample is multiplied by the second filter coefficient, the two results being summed to produce the output. This process is then repeated, with the samples 'moving' along the chain of unit delays until the first sample is multiplied by the last filter coefficient and the first true filtered output sample is produced.

The number of unit delays in the chain dictates the order of the filter. It is a straightforward matter to obtain a digital filter with a much higher order and steeper roll off than it is possible to implement in the analogue domain.

Frequency Domain Filtering

In frequency domain filtering, an N length Fast Fourier Transform (FFT) is taken of the impulse response. The signal is divided into blocks N/2 long and the fft of each block is taken. The two frequency domain signals are then zero padded, then multiplied together point for point and the result is then converted back to the time domain using an inverse fft. The time domain blocks are then put back together in sequence using an overlap and add procedure. For filters that have an impulse response longer than N / 2, the impulse response must also be divided into N / 2 length sections and zero padded, before the initial FFT is taken.

Despite all of the conversion to and from the frequency domain, frequency domain convolutional filtering is often quicker to execute than time domain convolutional filtering due to the significantly reduced number of multiplications and additions that must be performed.

Phase Considerations

Although the use of finite impulse response filters can model any filter response, it is very processor intensive, requiring a large amount of computation to produce an output. The main advantage that puts FIR filters ahead of other types of filter is their phase linearity. This means that FIR filtered signals undergo no phase distortion, which is very important in line arrays to maintain the integrity of the wave front.

Infinite Impulse Response Filters

The general form of an IIR filter is given in figure 4.14

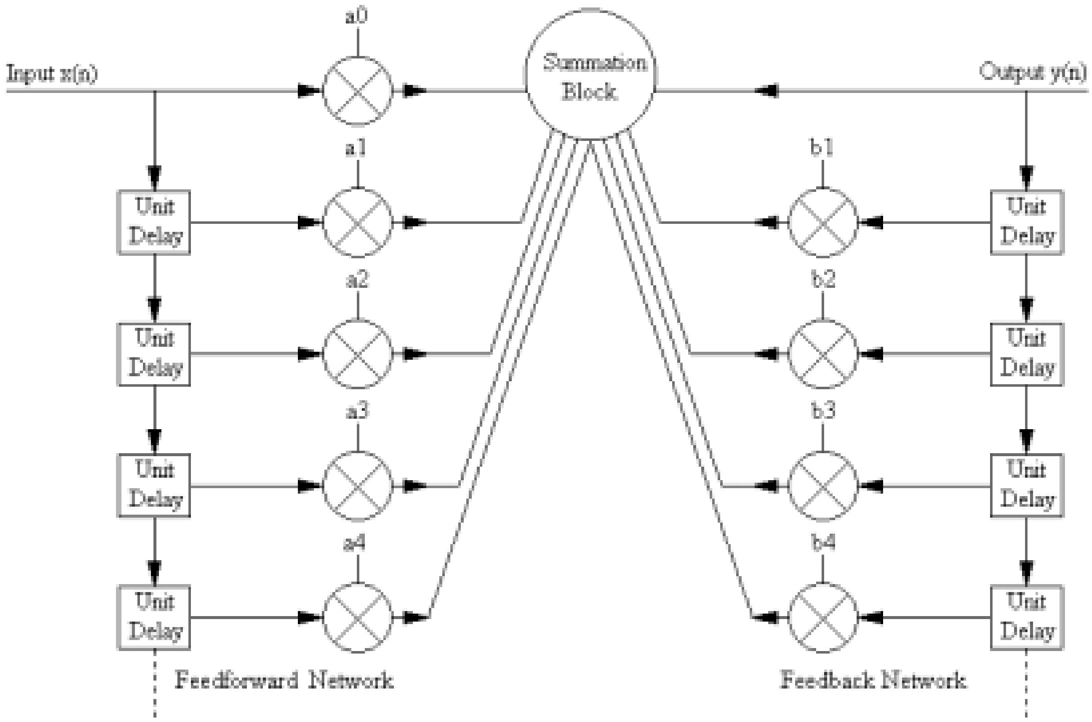

Figure 4.14 General form of an IIR filter

The equation for the output of this filter is given by:

Y(n) = x(n)a0 + x(n-1)a1 + x(n-2) + … - (y(n-1)b1 + y(n-2)b2….)

Instability

One of the main differences between IIR and FIR filters is that IIR filters use feedback, whilst FIR filters do not. The only problem with feedback is that if there is too much feedback then the filter becomes unstable.

Phase

The other problem with IIR filters is that they do not exhibit linear phase, which in a line array application would lead to distortion of the wavefront, making them unsuitable for this application.

Chapter 5

DSP Implementation

This chapter addresses two main areas, beam steering and beam width or dispersion control. Beam steering was implemented using delay and the beam width control was achieved using FIR filters.

Beam Steering

When the individual drivers in a line array couple together acoustically, the resultant waveform can be considered as a waveform that is parallel to the front of the array (figure 5.1).

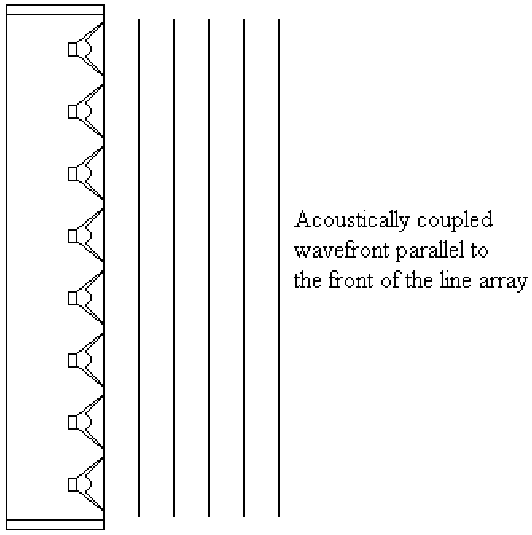

Figure 5.1 Line array coupled wavefront

By delaying the signal to each of the drivers by a specific amount, the wavefront can be steered so that it is no longer parallel to the front of the array.

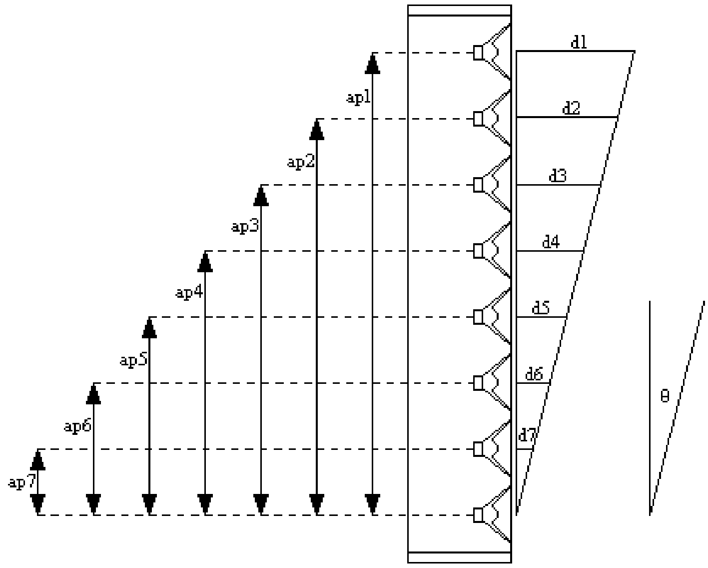

Figure 5.2 Steering diagram

If the angle formed between the front of the array and the wavefront is θ (figure 5.2), then simple trigonometry can be used to calculated the distances d1 to d8:

Tan θ = d1 / ap1 = d2 / ap2 = …d7 / ap7

Table 5.1 Delay calculations

distance	Value
D1	Ap1 tanθ
D2	Ap2 tanθ
D3	Ap3 tanθ
D4	Ap4 tanθ
D5	Ap5 tanθ
D6	Ap6 tanθ
D7	Ap7 tanθ

Apertures ap1 to ap7 can be calculated as the diameter of each driver was 0.13335m and the space between each of the drivers was 0.02m.

Table 5.2 Aperture calculations

Aperture	No of drivers	Length due to drivers (m)	No of gaps	Length due to the gaps (m)	Aperture Length (m)
Ap1	8	0.93345	7	0.14	1.07345
Ap2	7	0.8001	6	0.12	0.9201
Ap3	6	0.66675	5	0.10	0.76675
Ap4	5	0.5334	4	0.08	0.6134
Ap5	4	0.40005	3	0.06	0.46005
Ap6	3	0.2667	2	0.04	0.3067
Ap7	2	0.13335	1	0.02	0.15335

Using the information in tables 5.1 and 5.2 the length of the wavefront from the driver can be calculated for a given angle. This length can then be converted into a delay setting for each driver to steer the wavefront. One channel will remain undelayed and the rest of the channels will follow on from that channel. If the sample rate of the system is 44100 samples per second and the speed of sound is taken to be $344ms^{-1}$, then:

Time delay (s) = distance (m) * speed of sound (ms^{-1})

Delay length in samples = Time delay (s) * 44100

Example
If θ = 15 degrees, then the delays will be:

Table 5.3 Sample calculations

15 Degrees				
number of speakers	Centre to centre Dimension (m)	Distance out from cone (m)	Delay (s)	Delay (samples)
8	1.07345	0.294717081	0.000856736	37.782044410
7	0.92010	0.252614641	0.000734345	32.384609500
6	0.76675	0.210512201	0.000611954	26.987174580
5	0.61340	0.168409761	0.000489563	21.589739670
4	0.46005	0.126307320	0.000367172	16.192304750
3	0.30670	0.084204880	0.000244782	10.794869830
2	0.15335	0.042102440	0.000122391	5.397434916
1	0	0	0	0

These delays were then fed into a simulink model using a variable integer delay (figure 5.3).

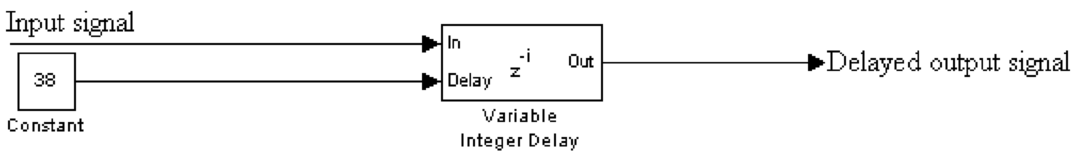

Figure 5.3 Integer delay

As the variable integer delay would only accept integer delays, the calculated delays were rounded off.

The example calculations for a 15 degree steering angle were entered into a simulink model (figure 5.4).

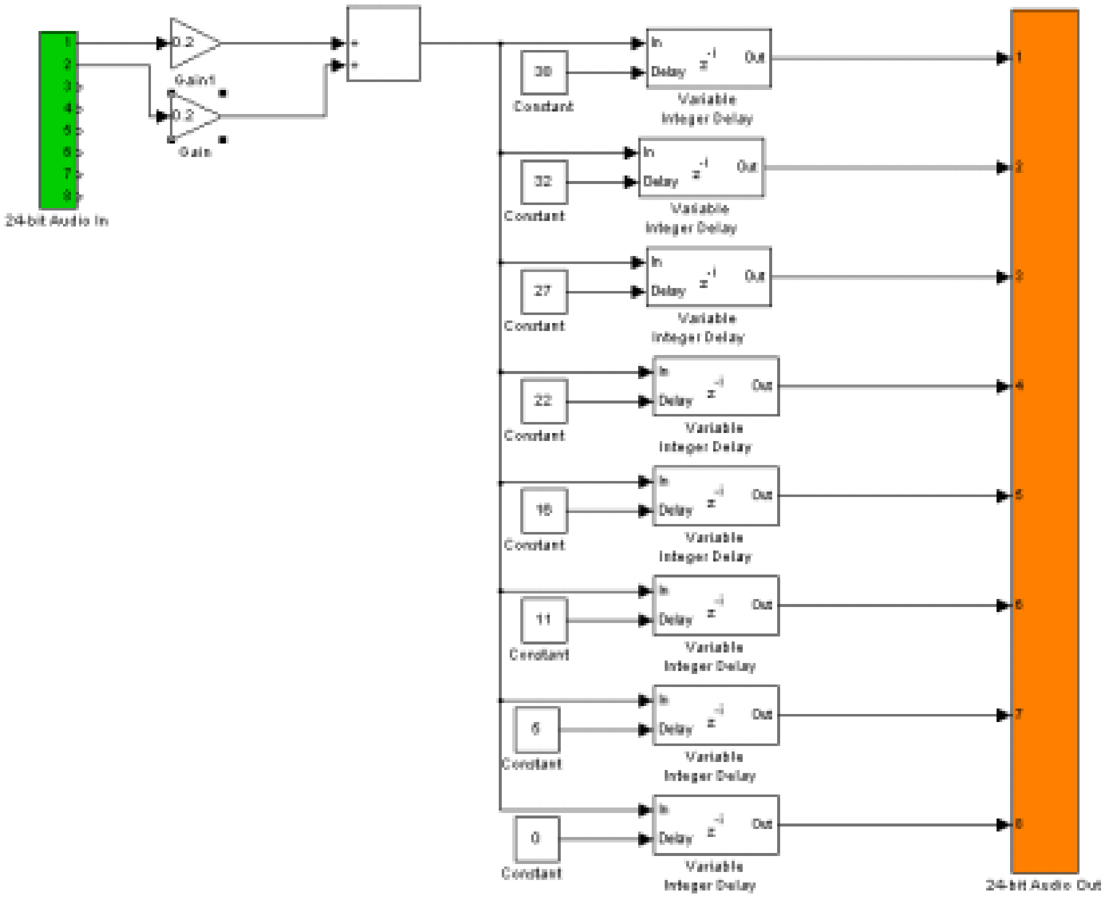

Figure 5.4 Simple steering model

This model was used to measure the output of the array using JBL's Smaart Pro software (see chapter 6) and was found to generate a main lobe, which was 15 degrees off axis.

Real Time Steering

Once the basic theory had been proven, the model was refined to allow steering in real time. Instead of calculating the sample delays using an excel spread sheet, the delays were calculated in real time using a slider in simulink as an input device to enter the steering angle between -45 and 45 degrees (figure 5.5).

The first constant provides unity gain for the angle slider.

The tan function in simulink requires an input in radians, so the second constant converts the degree output from the slider into a radian input for the tan function.

Radians = 2 pi * degrees/360 = 0.017453292 * degrees

There is a block in Simulink that performs this function called a degree to radian transformer.

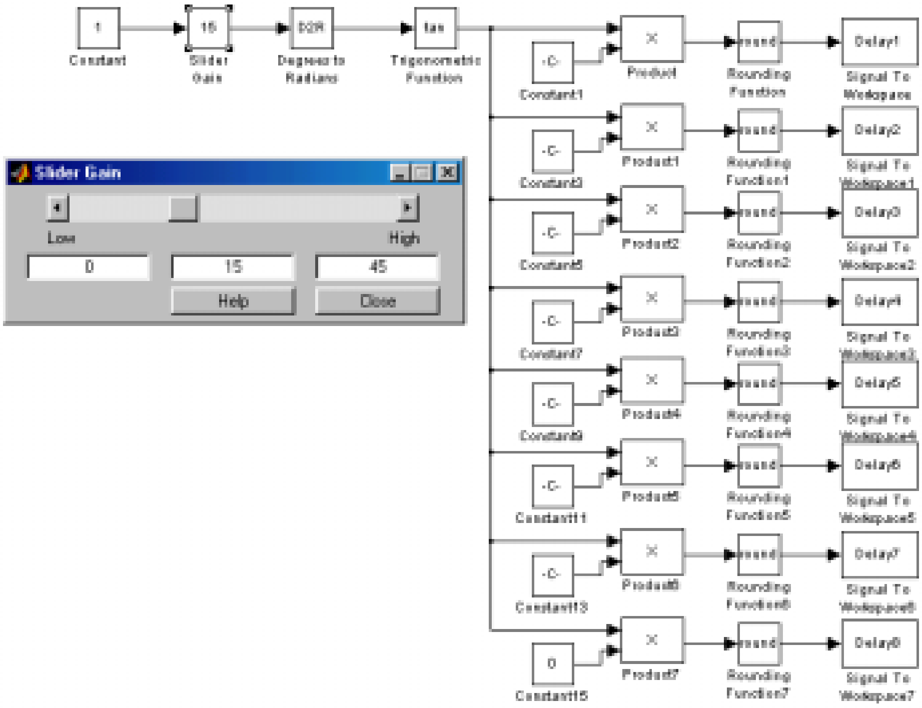

Figure 5.5 Simple steering model with angle slider

Constants 3 to 10account for the three multiplications that must be carried out when calculating each delay; the tangent of the angle must be multiplied by:

- The centre-to-centre length between speakers.
- 1/ the speed of sound (taken as 344ms^{-1}).
- The sample rate (44.1 KHz).

Table 5.4 Constant calculations

Constant	Centre to centre length (m)	1/c (ms)	$F_{sampling}$ (Hz)	Value
1	1.07350	0.002906976	44100	137.6137936
3	0.92010	0.002906976	44100	117.9546802
5	0.76650	0.002906976	44100	98.29556686
7	0.6134	0.002906976	44100	78.63645349
9	0.46005	0.002906976	44100	58.97734012
11	0.30670	0.002906976	44100	39.31822674
13	0.15335	0.002906976	44100	19.65911337
105	0	0.002906976	44100	0

For test purposes, the sample delays calculated by simulink were exported to the workspace so that they could be compared to the sample delays calculated using an excel spreadsheet. Three different angles were considered in excel; 5,10 and 15 degrees. The results showed that the simulink model performed as expected.

Negative angles

If the slider in figure 5.5 was set so that a negative angle could be entered into the system, a situation were the beam could be steered up or down, then a problem would arise where negative sample delays could be calculated, which would not produce the desired resultant beam angle.

One possible solution to this problem was to turn the array upside down so that up became down. A more elegant solution would be to route the delay channels to different drivers in software using an up or down selector. This solution was implemented in simulink by developing a routing subsystem for each channel.

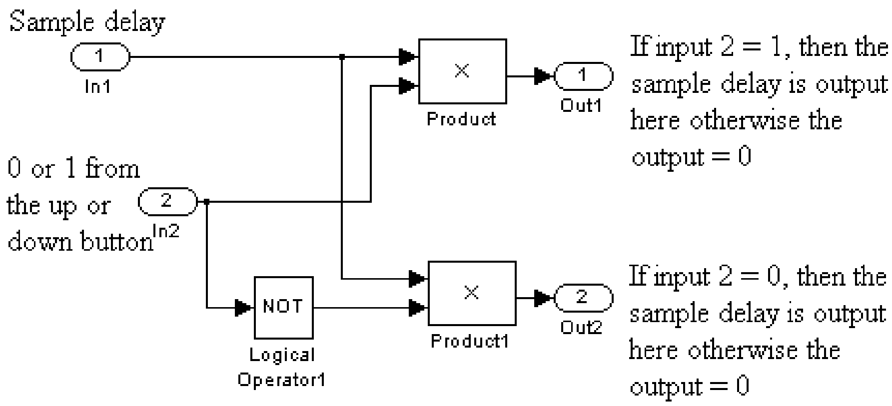

Figure 5.6 Delay routing subsystem in Simulink

It is important to realise that the output not sending the delay must be set to 0 to prevent erroneous sample delays adding up and distorting the wavefront. This subsystem was then applied to each channel of the model and the outputs routed accordingly (figure 5.7).

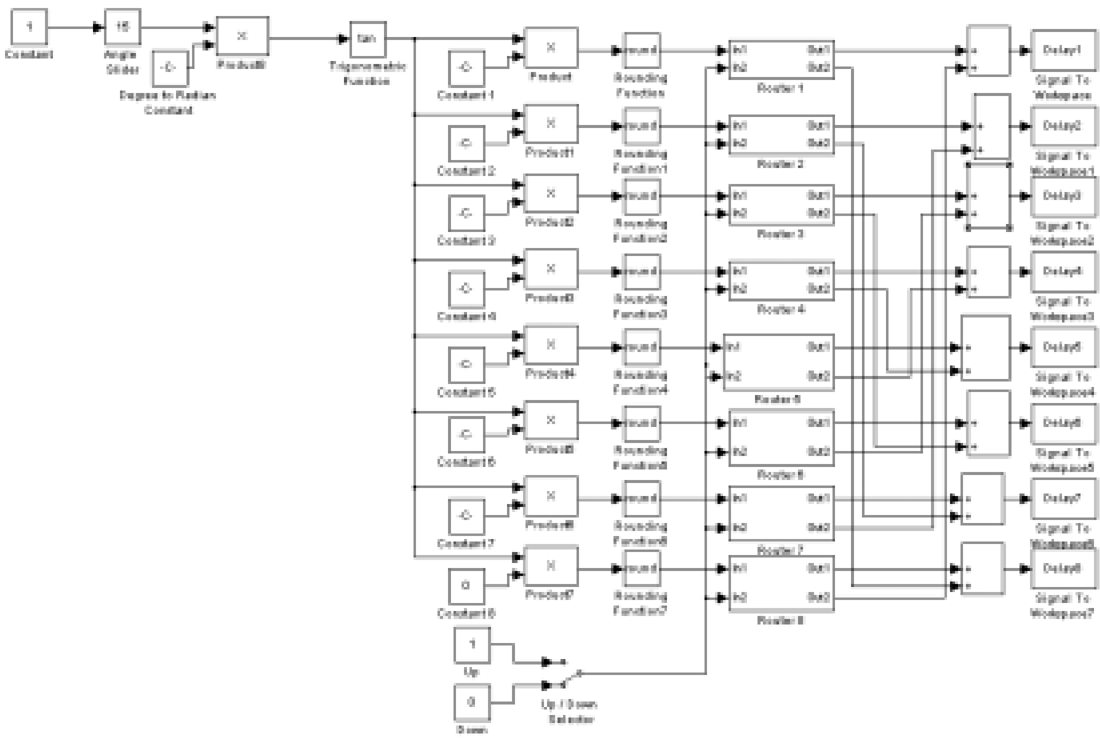

Figure 5.7 Steering model incorporating router subsystem

With the steering angle set to 15 degrees as in figure 5.7, the outputs to each delay are listed in table 5.5.

Table 5.5 Router subsystem test results

Delay	Button = 1	Button = 0
1	37	0
2	32	5
3	26	11
4	21	16
5	16	21
6	11	26
7	5	32
8	0	37

As can be seen the router effectively turns the array upside down depending on what the up / down selector switch is set to.

This router design failed to work when implemented with the steering delays as the second in port of the subsystem would not clock at the same speed as the rest of the system because the port defaulted to getting its clock speed from the preceding blocks and in the case of port 2, there was no clock in the preceding blocks as they consisted of constants and a manual switch.

A different and more elegant solution to the problem was adopted (figure 5.8). The angle slider was set to work from –45 to degrees to 45 degrees. The modulus of this angle was then fed into the degree to radian converter stage so that a positive set of delays would always be calculated. The angle was also fed to a relational operator block, which compared the angle to a reference constant of zero and gave an output as detailed in table 5.6.

Table 5.6 Relational operator output

Angle	Relational Operator Output
Positive	1
Zero degrees	1
Negative	0

In other words, the routers flip where the delay is being sent to when the angle is negative. The zero degree case was included for completeness, but is irrelevant as when the angle is zero, the delays would all be zero to keep the wavefront parallel to the front of the array.

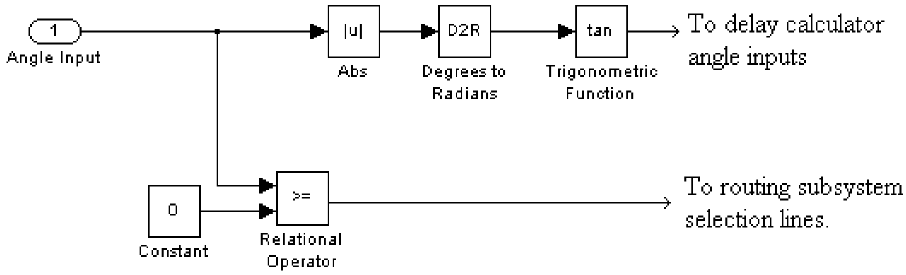

Figure 5.8 Routing selector

Multiple Beams

In certain difficult acoustic environments it may be desirable to direct sound to multiple audience locations whilst keeping sound away from acoustically reflective items in between (figure 5.9).

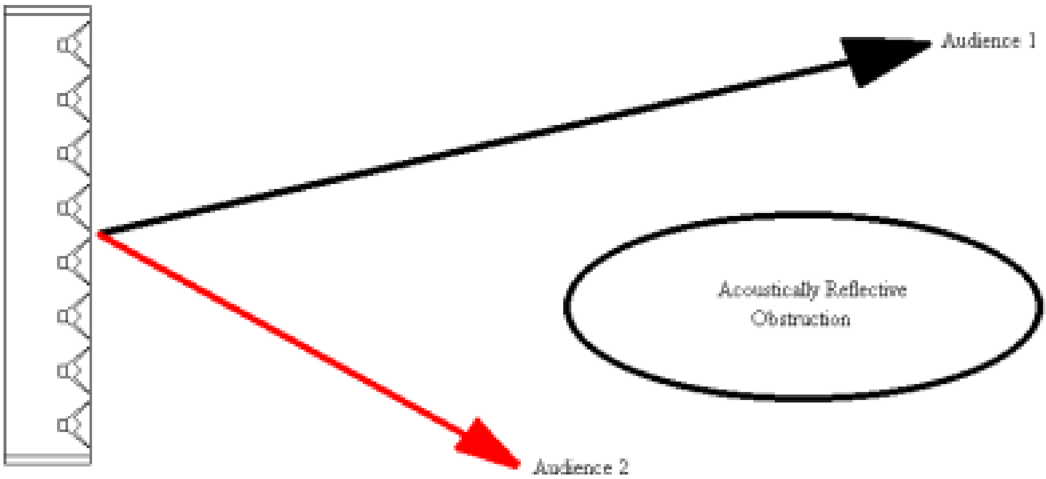

Figure 5.9 Dual beam

As the 2 beams will have different delays applied to each driver, the multiple beam steering can be achieved by repetition of the software developed for steering a single beam.

The first step was to incorporate the 8 channel steering delay in figure 5.10 into a subsystem.

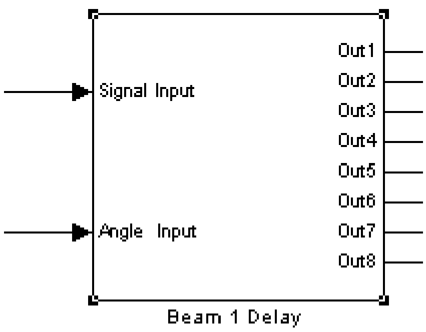

Figure 5.10 Simulink beam steering subsystem

With the delay subsystem is opened, the inside is shown in figure 5.11.

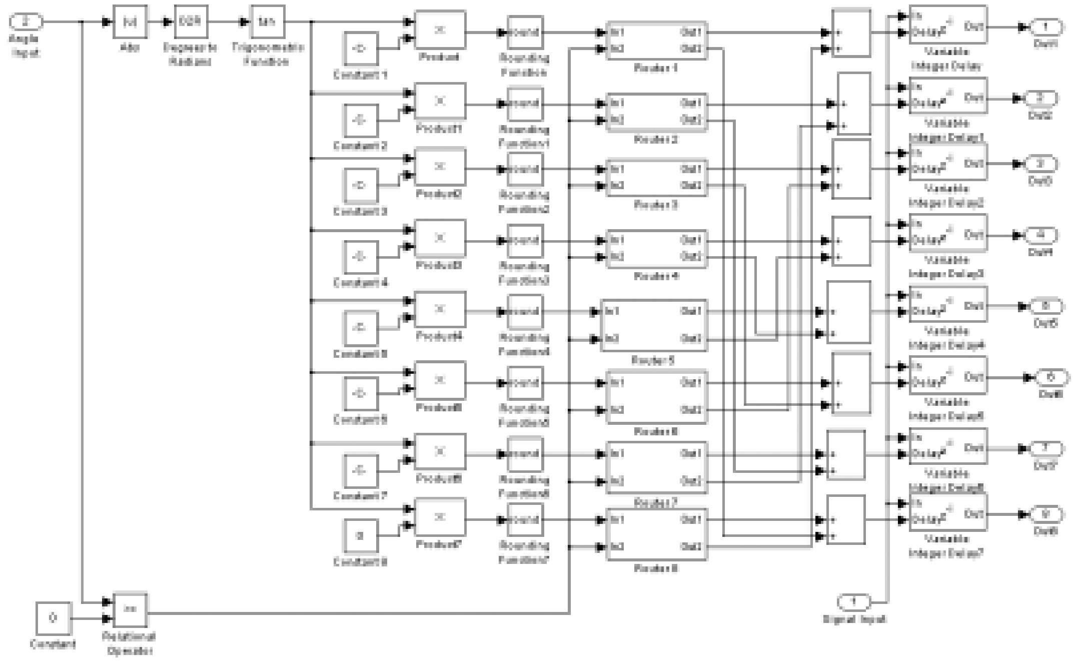

Figure 5.11 Simulink beam steering subsystem contents

For reasons of clarity and ease of operation, the integer delay was incorporated into the subsystem and the steering control was moved outside the subsystem so that it appeared at the highest level in simulink.

The block was then copied and the outputs of the two blocks were then summed before going to the soundcard output (figure 5.12).

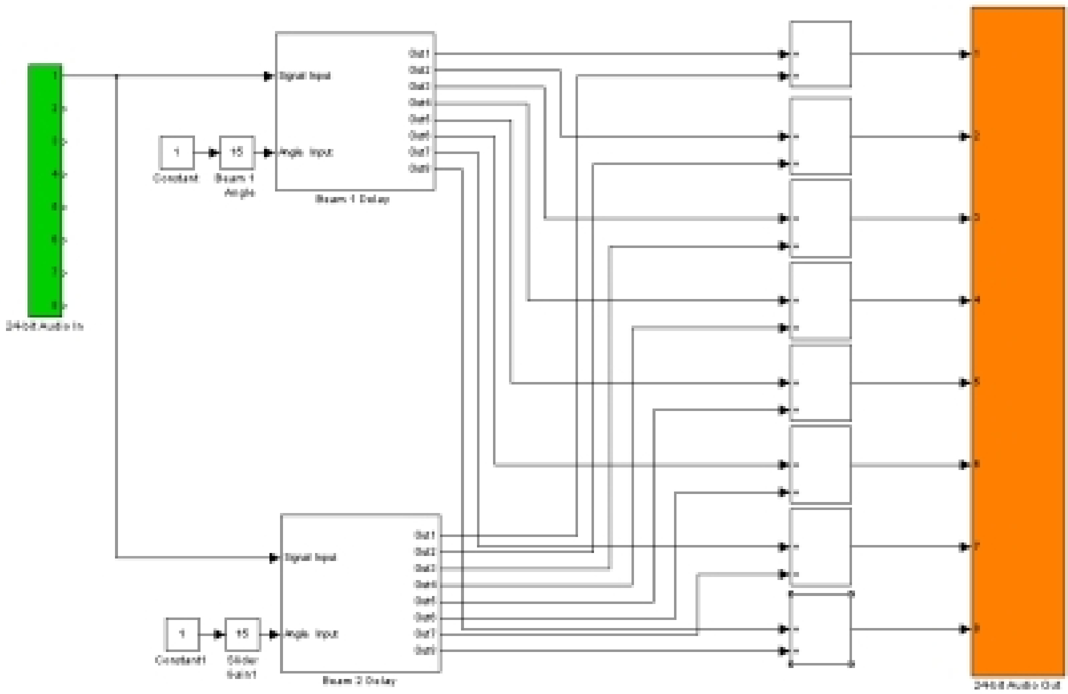

Figure 5.12 Dual beam steering model in Simulink

Beam Width Control

The nature of a line array is for the beam to narrow in the vertical plane due to the output of the drivers in the array coupling together to behave as a larger single driver. The only problem with this is that the beam width is frequency dependent, for higher frequencies, the beam becomes narrower and at lower frequencies, the beam approaches an omni directional response. As the directivity of the beam for a given wavelength is dependent on the diameter of the driver or group of drivers it would be ideal if the length of the array could be changed depending on the wavelength of the input signal. For a signal with a long wavelength, the array should be as long as possible to get a narrow beam and for a signal with a short wavelength, the array should be as short as possible to maintain this narrow beam width.

The longest the array can be is when all 8 speakers are active and the shortest the array can be is when only one driver is active. The way in which the effective length of the array can be changed according the wavelength or frequency is to filter each channel so that it is only active for certain frequencies. The first step in this procedure is to look at the 8 possible array lengths that are available. The 8 effective apertures are calculated from:

Effective aperture = (n*d) + (n-1)*g

Where n = the number of active speakers, d = the diameter of the loudspeaker (0.13335m) and g = the gap between the loudspeakers (0.02m).

Table 5.7 Effective aperture calculations

Number of active speakers	Aperture (m)
8	1.20680
7	1.05345
6	0.90010
5	0.74675
4	0.59340
3	0.44005
2	0.28670
1	0.13335

Taking the largest aperture of 1.2068m, the lowest frequency can be calculated that the array can control the directivity. Frequencies below this will have an uncontrolled omni directional response if the drivers are capable of reproducing them. The frequency response of an individual driver was measured and the results shown in chapter 6.

Taking the effective aperture of 1.20680m, this will be equal to the wavelength of the reproduced sound for an omni directional response.

$C = f * \lambda$ where $c = 344ms^{-1}$ (the speed of sound in air at 20 degrees Celsius) and λ = 1.20680m giving a lowest operating frequency of 415.1392 Hz. This is actually of little practical use because low frequency signals tend to have an omni directional response anyway when the wavelength of the signal is equal to the diameter. The calculation is based on the case shown in figure 5.13 where the effective aperture is equal to the wavelength.

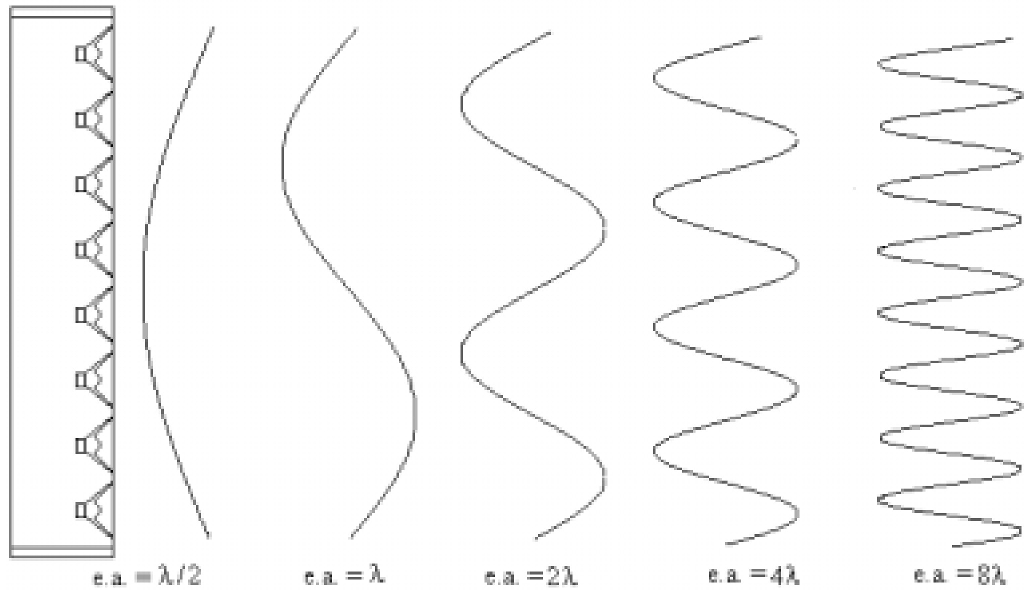

Figure 5.13 Wavelength relationship to array length

As can be seen from the above diagram if a narrower beam is required, then the wavelength must be reduced for a given effective aperture. Cut off frequencies for each effective aperture were calculated for two of the cases shown in figure 5.13.

Table 5.8 Cut off frequency calculations (1)

Number of speakers	Effective Aperture = 2* λ	
	Cut off Frequency	Cut off Freq (normalised to 22.05KHz)
8	570.1028	0.025855000
7	653.0922	0.029618695
6	764.3595	0.034664831
5	921.3257	0.041783480
4	1159.420	0.052581419
3	1563.459	0.070905156
2	2399.721	0.108830883
1	5159.355	0.233984357

Table 5.9 Cut off frequency calculations (2)

Number of speakers	Effective Aperture = 4* λ	
	Cut off Frequency	Cut off Freq (normalised to 22.05KHz)
8	1140.206	0.051710000
7	1306.184	0.059237390
6	1528.719	0.069329661
5	1842.651	0.083566961
4	2318.841	0.105162838
3	3126.917	0.141810313
2	4799.442	0.217661765
1	10318.71	0.467968715

The theory of how the effective aperture changes with respect to frequency is shown in table 5.10 for the low pass filter cut off frequencies calculated for the case when the effective aperture of the array is equal to twice the wavelength (table 5.8). As the roll off of the filters will not be ideal (vertical) then there will be some blurring of the region between active drivers and inactive drivers for a given test frequency.

Table 5.10 Loudspeaker activity

Filter Cut off Frequency (Hz)	Test Frequencies (Hz)		
	500	1000	5000
570.1028	Active	Inactive	Inactive
653.0922	Active	Inactive	Inactive
764.3595	Active	Inactive	Inactive
921.3257	Active	Inactive	Inactive
1159.420	Active	Active	Inactive
1563.459	Active	Active	Inactive
2399.721	Active	Active	Inactive
5159.355	Active	Active	Active

As can be seen, at 500 Hz, the entire array is active to control the directivity, at 1 KHz, roughly half of the array is active to maintain directivity and at 5 KHz, only one of the drivers is active.

The normalised cut off frequencies were fed into a simulink model that was attached to the line array, the only difference being that one of the channels was left as a through channel as the smallest effective aperture was one driver. The through channel still included an FIR filter as there will always be a delay associated with any digital filter and each channel must be subjected to the same delay in order to maintain wavefront integrity.

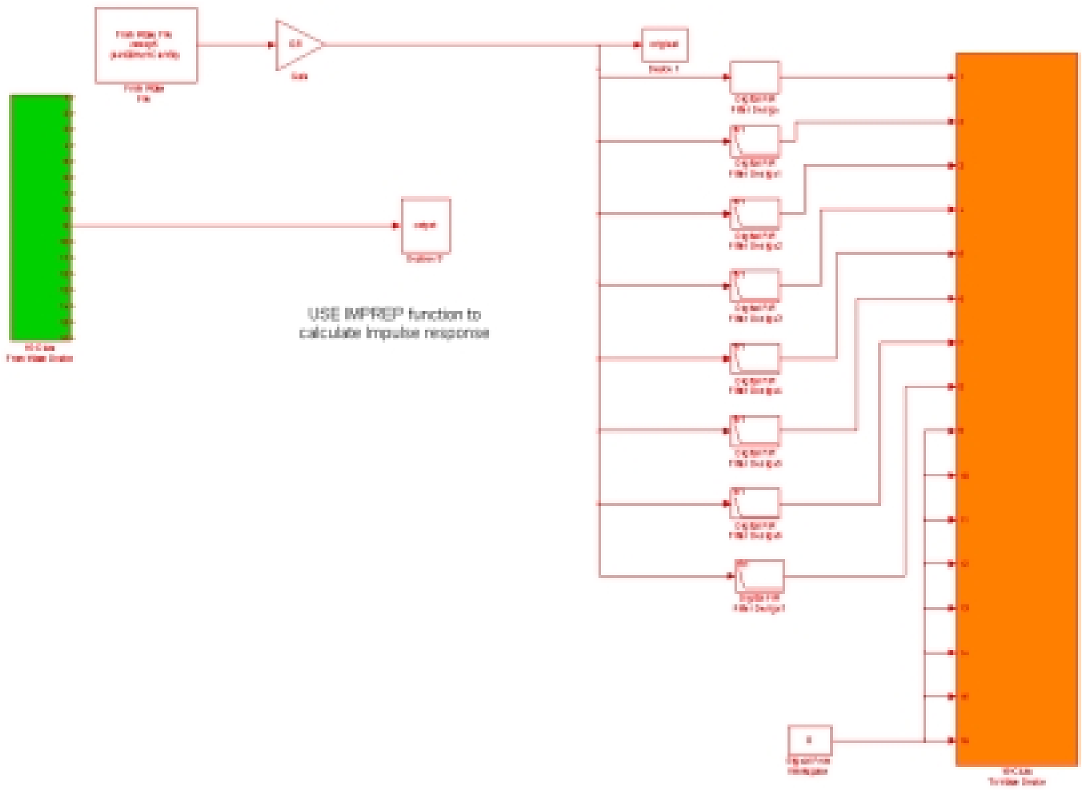

Figure 5.14 Dispersion control model

Of the 16 outputs on the sound card, only 8 channels were used to drive the line array and so the other channels were fed with zeros to enable the software to run without error. The test signal sent to each channel was a mono 16 bit wav file which was an 8 second sine wave sweep from 20 Hz to 20 KHz. The sweep was also exported to MATLABS workspace. The output of the line array was measured using a Neutrik 3382 omni directional test microphone, which was phantom powered using a Yamaha O2R mixer and then fed back into channel 9 of the soundcards input via a Tascam Digital InterFace (TDIF). This was then also exported to the workspace so that it could be compared to the original test signal. The 2 signals were used alongside a Matlab command called IMPREP to calculate the frequency response of the filtered array. Full details of the testing and results are shown in chapter 6. What became apparent from the testing and by listening to audio through the filtered array was that there was far too much bass being produced by the system. This is reasonable when considering that 7 of the 8 channels were being low pass filtered.

One solution to this problem was to develop a filter that was the inverse of the system response to apply to the signal prior to the filter bank that controls the effective aperture of the array.

Developing an inverse filter

The system was set up as for measuring the unfiltered array, except that the filters were included in the Simulink programme.

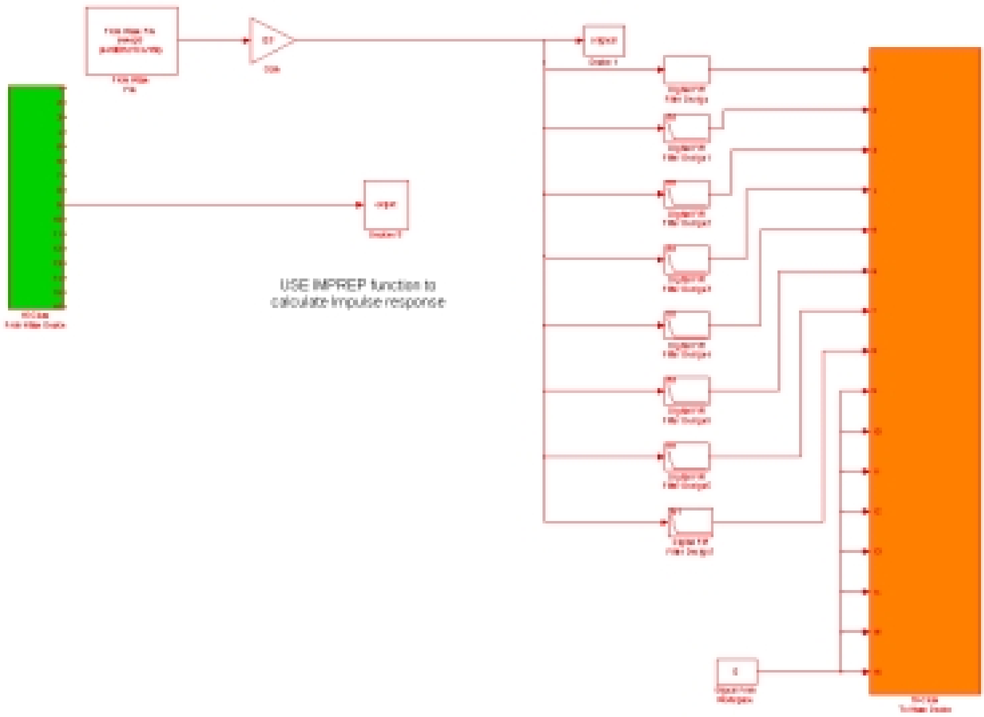

Figure 5.15 Simulink model for measuring the impulse response of the filtered array

The following normalised cut off frequencies were used:

Table 5.11 Normalised cut off frequencies

Driver Number	Cut Off Frequency (Hz)	Normalised Cut Off Frequency
1	22500	'Through'
2	1184.66899	0.026863242
3	771.8501703	0.017502271
4	572.3905724	0.012979378
5	454.8494983	0.010314048
6	377.3584906	0.008556882
7	322.4276908	0.007311286
8	281.4569536	0.006382244

The through filter was included so that no phase distortion of the wave front would occur due to a different propagation delay between the through channel and the filtered channels.

The system captured the impulse response of the filtered line array and this response was used to generate an inverse filter, again using MATLAB and the imprep command.

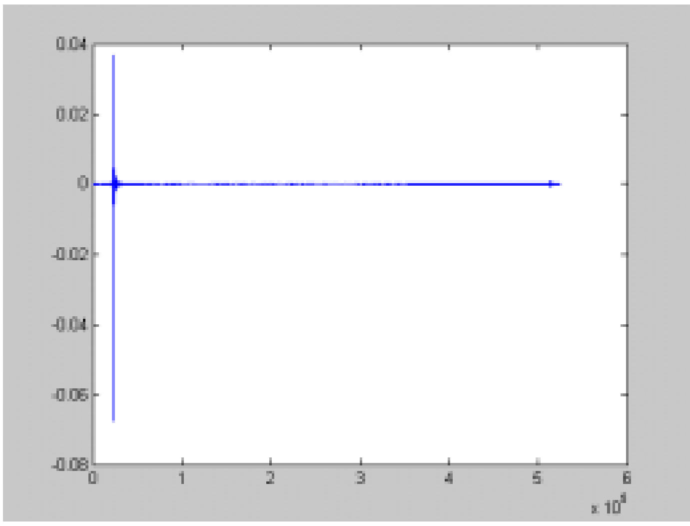

Figure 5.16 Linear filtered array impulse response

The impulse response was truncated to remove noise that occurred during measurement. Using impulse = impulse (15000:31500);

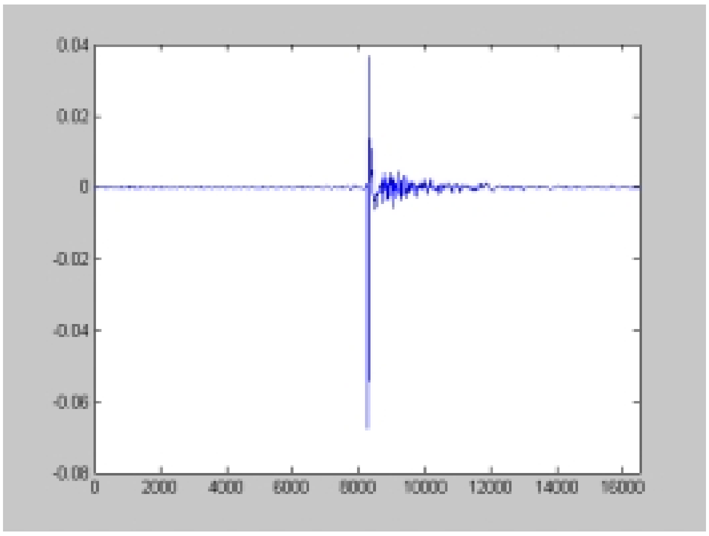

Figure 5.17 Truncated impulse response

The response was then zero padded to ensure that the impulse could hold the filter.

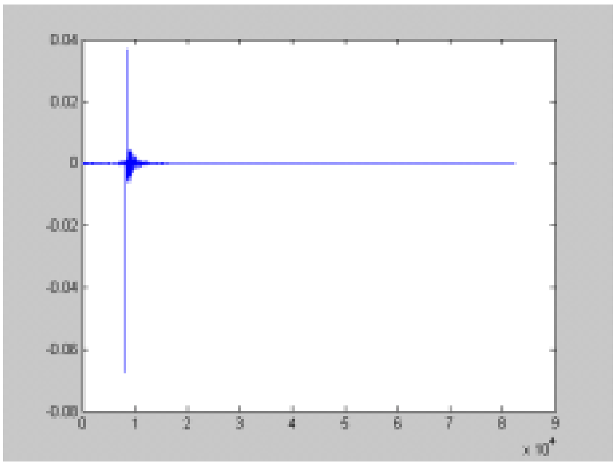

Figure 5.18 The zero padded impulse

The impulse was then centred so that it could be windowed. The impulse was then multiplied point for point with a hamming window of the same length (16501).

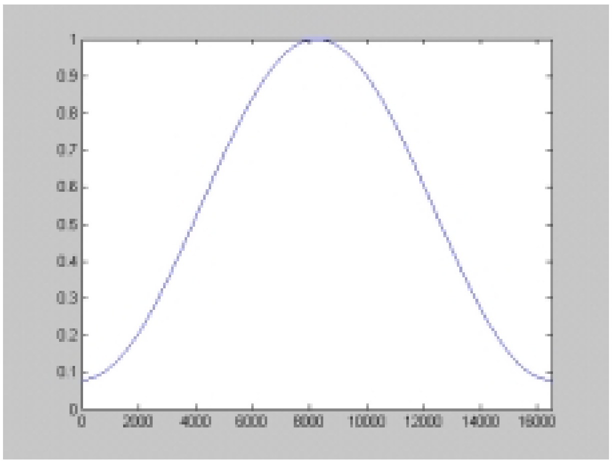

Figure 5.19 Hamming window

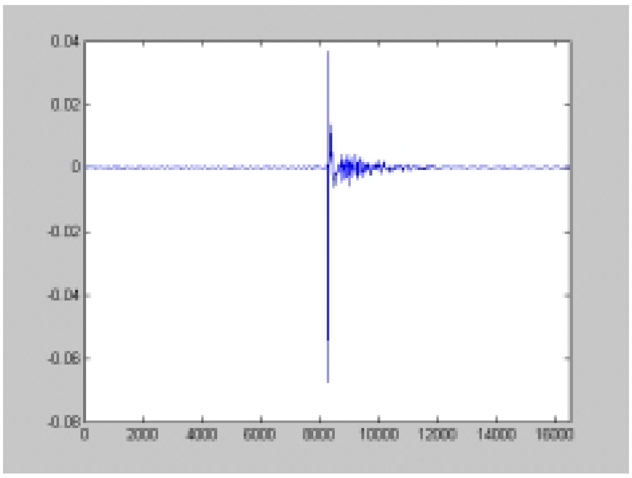

Figure 5.20 The windowed, truncated, zero padded linear array impulse

The inverse filter was then calculated using:

Invfilter = imprep (imppad, pulsepad);

This generated the following impulse response:

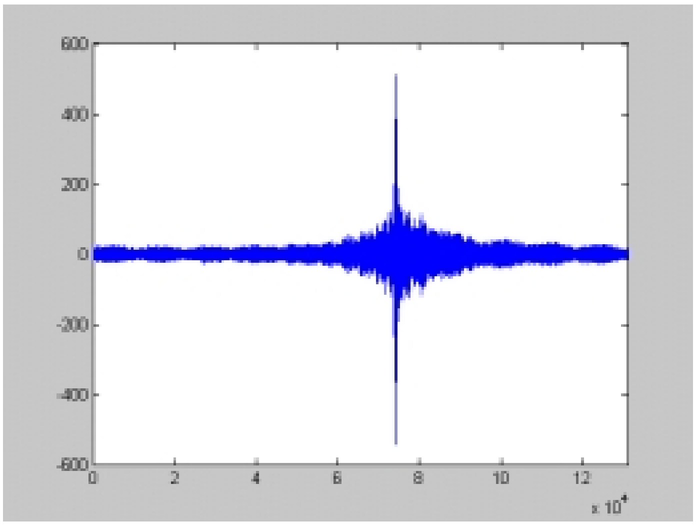

Figure 5.21 The linear array inverse filter impulse response

In the frequency domain, this can be shown as:

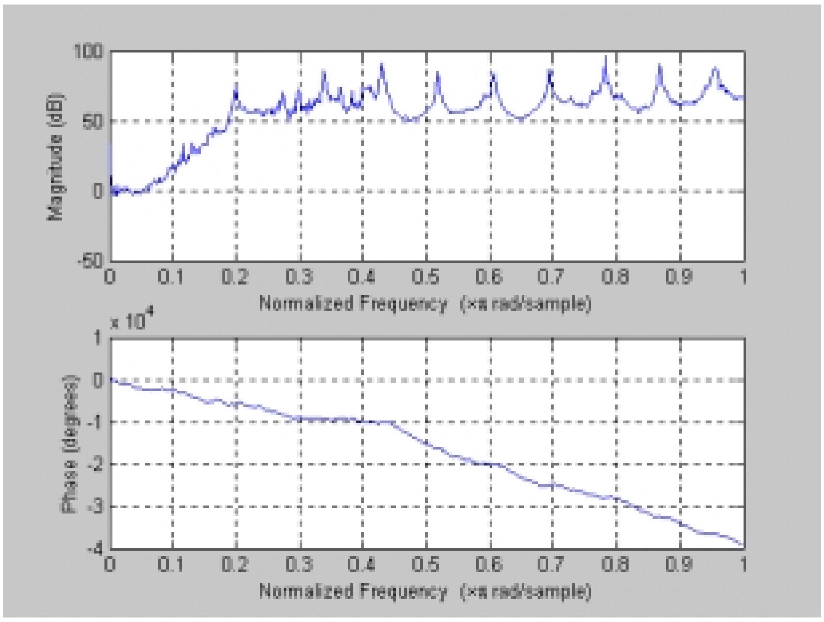

Figure 5.22 Linear array inverse filter frequency response

By including both the arrays filtered frequency response and it's inverse filter frequency response on the same diagram, it can be seen that the one is the mirror image of the other.

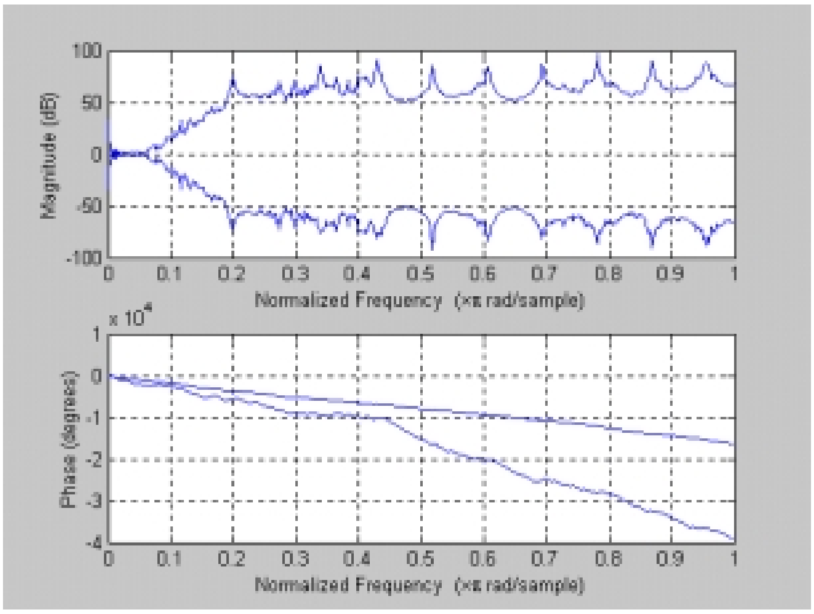

Figure 5.23 Comparison between the linear array frequency response and the inverse filters frequency response

In practical terms, the inverse filter would have to apply so much gain that even a small input signal would send the system into clipping, so an approximation of the inverse filter will have to be developed to correct for excessive bass boost. Another problem with the inverse filter developed from measuring the low pass filtered array was that the inverse filter was too long to implement in real time at 131072 samples long.

An alternative approach was adopted for the dispersion control filters. The need for an inverse filter arose out of the bass heavy frequency response due to the FIR filters used for dispersion control from Simulink attenuating the stop band by a large amount. By using a program called SPTools within the Matlab environment, a more flexible approach to filter design could be adopted. Using Sptools allowed the development of filters with less stop band attenuation than those from the simulink browser.

The edge of the pass band and the order of the filter were specified and the edge of the stop band was adjusted to give a stop band attenuation of 30 dB. The filter order was set at 30 for all filters in order to maintain the integrity of the wavefront. The cut off frequency for each filter was calculated for each loudspeaker dependant on its position in the array and the required directivity as before.

The finished filters were then exported from Sptools to the Matlab workspace, where they were converted to a list of filter coefficients for use in simulink using the following command:

$F_n = filt_n$. tf.num where n is the number of the filter being implemented (from 0 to 7), tf stands for the transfer function and num stands for the numerator. The workspace containing the filter coefficients was then saved for each set of filters.

In each simulink model that was to use the filters, the standard Simulink filters were replaced with direct form 2 transpose filters that are FIR filters where the list of filter coefficients can be specified as a matrix in the workspace (figure 5.24).

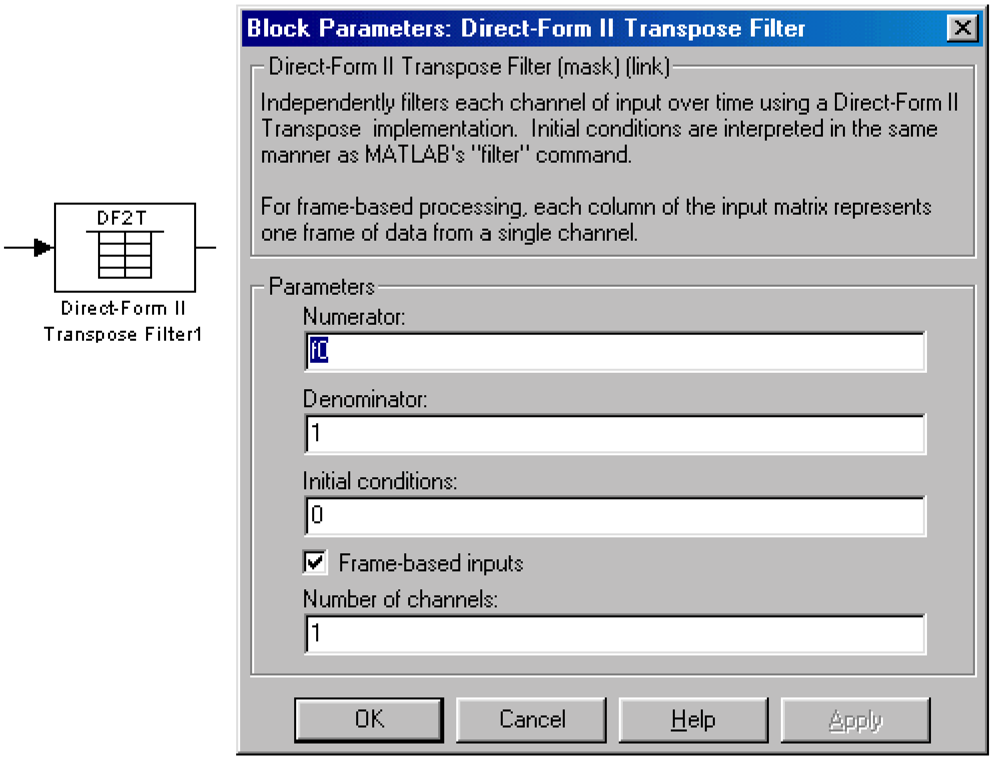

Figure 5.24 Direct form 2 transpose filter

One major advantage of this system that soon became apparent was that it was extremely quick and simple to change the filter parameters of a given Simulink model by simply loading another workspace whose filters shared the same names as the preceding workspace. This allowed different designs of filter to be quickly developed and then implemented.

Chapter 6

Testing and Results

The test methods that were used for assessing the array during the project underwent modifications over the project that are worth noting. Initial frequency responses were measured using a Neutrik 3382 measurement microphone. The technical specification for this microphone was obtained after the microphone had been used and was found to be ± 5 dB from 20 Hz to 20 KHz which was of insufficient quality for the measurements being taken. An Earthworks test microphone with an essentially flat response between 20 Hz and 20 KHz was used to replace the Neutrik mic.

The initial frequency responses were obtained using a frequency sweep in conjunction with Simulink and Matlab which was a moderately time consuming procedure. When the realisation was made that a minimum of twenty eight frequency response were needed for each assessment of the arrays on and off axis response, it became apparent that speeding up the procedure would be vital. Using Smaart Pro by JBL performed the same task as the Simulink / Matlab set up, but was found to be slightly quicker. The programme works in a similar way by feeding back a reference signal from its output to its input which it then uses in conjunction with the signal measured by the test microphone to produce a transfer function of the system under test (figure 6.1).

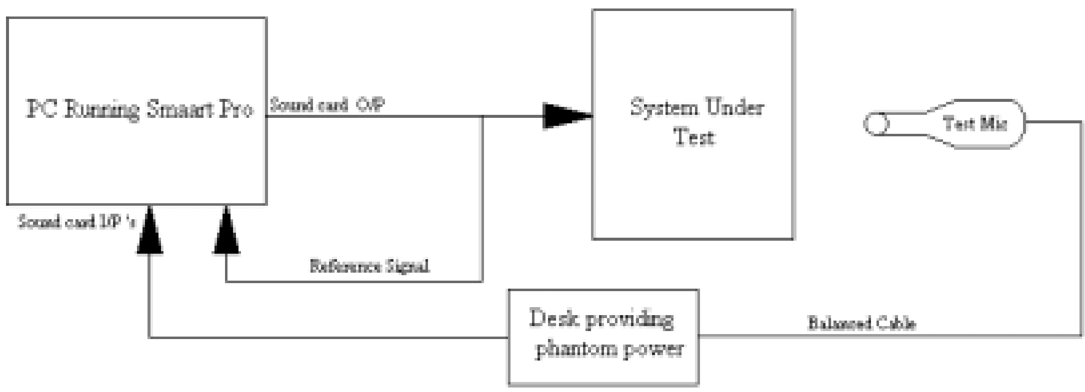

Figure 6.1 Smaart Pro initial set up

This set up was adequate for measuring analogue systems, but it was realised that as well as the soundcard of the pc running Smaart Pro, the analogue to digital converters and digital to analogue converters in the system under test (the simulink model and line array) would also effect the signal passing through and these would not be cancelled out by the reference signal. By modifying the set up so that the reference signal also passed through the system under test, but only through the ADC / DAC and not through any of the delays or filters in the model, the effects of the converters could be cancelled out (figure 6.2).

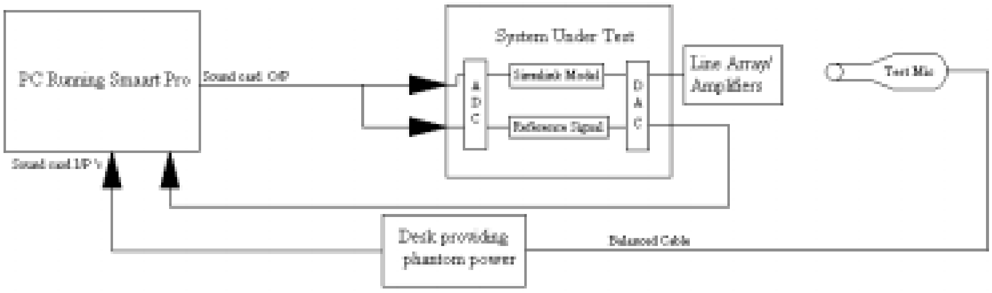

Figure 6.2 Smaart Pro final set up

Before any graphs could be produced from the system, the delay between the test signal and the reference signal had to be measured so that the two signals could be synchronised in order to produce an accurate transfer function. With the system set up as shown in figure 6.2, the delay measurement is made up of two factors. The first factor is the delay due to the propagation of the wavefront from the line array to the measurement microphone, which could be calculated from the speed of sound and the distance between the two transducers. The second contributory factor to the delay was due to the signal propagating through the delays and filters in the simulink model

Tests Performed

The single driver

The single drivers frequency response was measured using the Neutrik microphone and Matlab / Simulink. This measurement was important for two reasons. The response of one driver was the same as the horizontal response of the array. The second reason that the response was important was that it provides information about what frequencies the driver and therefore the frequencies that the array is capable of reproducing (figure 6.3).

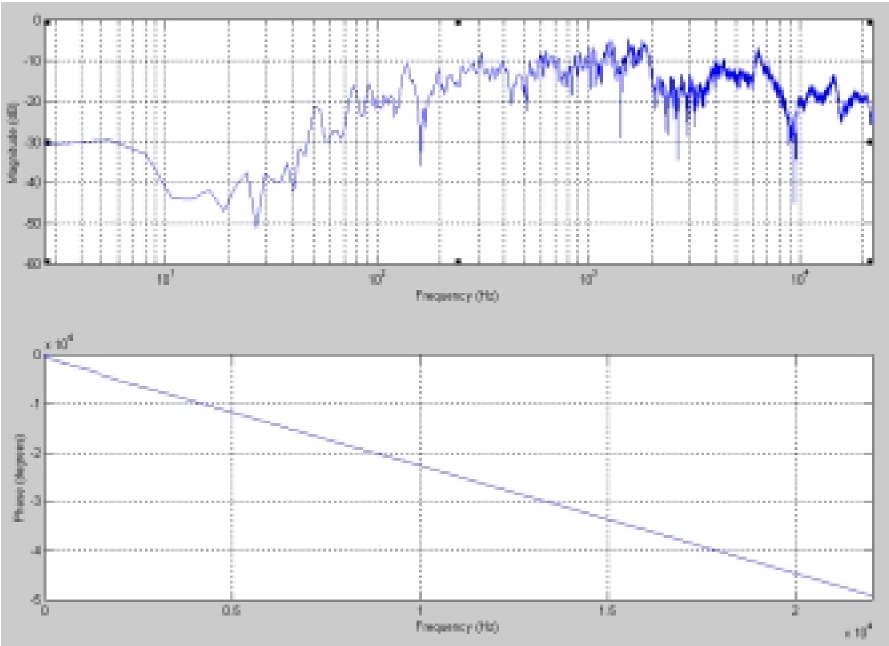

Figure 6.3 Single driver frequency and phase response

As can be seen, the response is far from flat, with a rapid roll off below 100 Hz. This is not too critical for speech, but the trough from 2KHz to 4 KHz is more of a concern as these are the frequencies that relate directly to speech and speech intelligibility. The graph was compared to one produced by Duran audio for one of the T2212 transducers used in one of their commercial line arrays (figure 6.4).

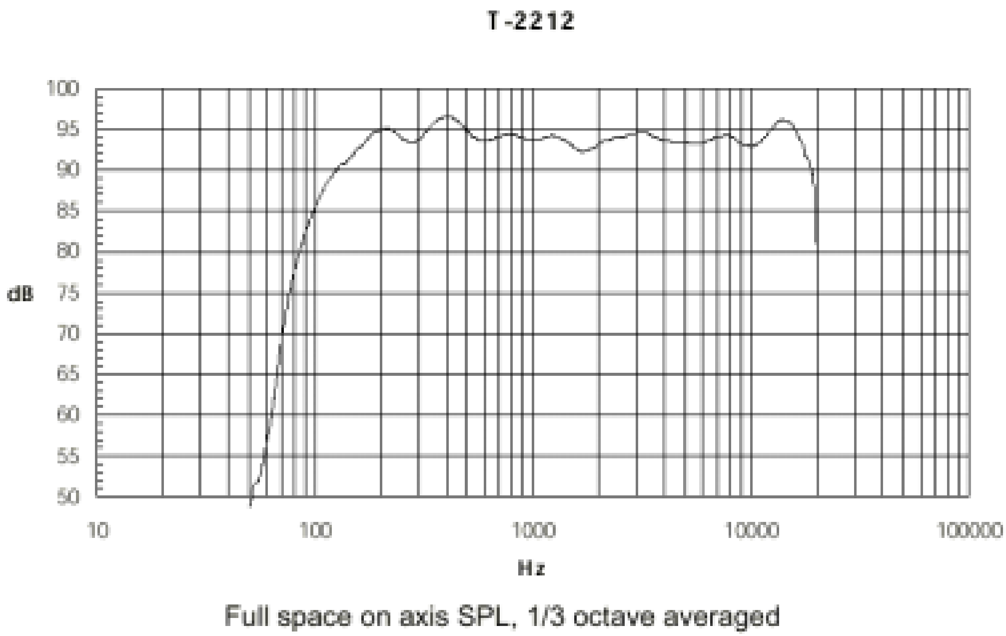

Figure 6.4 T2212 frequency response (Duran Audio [11])

Direct comparisons can not be made between the two as the Duran audio is averaged over 1/3 octave, which serves to smooth the response in a favourable way, but it is interesting to note that the commercial driver rolls off at a similar frequency to the prototype driver.

The Unfiltered Array

The frequency responses were obtained using a sine wave sweep in conjunction with Simulink, a Pentium 3 800 PC, using a 16 channel Mixtereme Soundcard and a Yamaha O2R desk (to provide the mic pre amp, phantom power and analogue to digital converter).

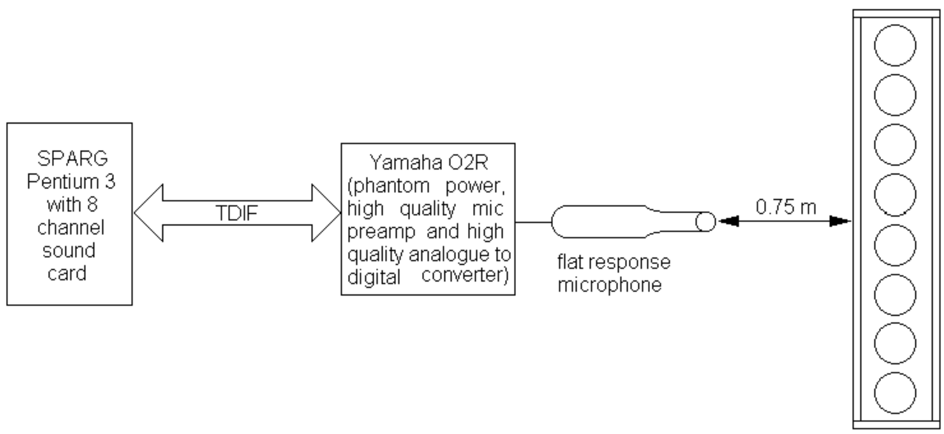

Figure 6.5 Block diagram of the system used to measure the output of the line array

Within Simulink, a swept sine wave of 8 seconds duration was used from 20 Hz to 20KHz. The original sine wave sweep was exported to the MATLAB workspace so that it could be compared to the measured output of the line array. The measured output from the line array was also exported to the MATLAB workspace. The imprep command was used to obtain the impulse response of the line array, which was then plotted using the freqz command.

The gain on the O2R was adjusted to give maximum signal amplitude without clipping.

The experiment was repeated for a single driver, two drivers, three drivers and all the way up to 8 drivers (figure 6.6).

As the number of drivers was increased, the bass response was seen to improve. The response still dropped away at around 100 Hz due to the diameter of the drivers.

As well as applying a sine wave sweep to the array and measuring the response, the array was set up to reproduce audio and it was listened to whilst laid on its side suspended 0.5 m above the floor. What the graphs do not clearly show is quite how directional the array is along the vertical axis. A strong hot spot was observed on axis where the sound was considerably louder than at a short distance to either side.

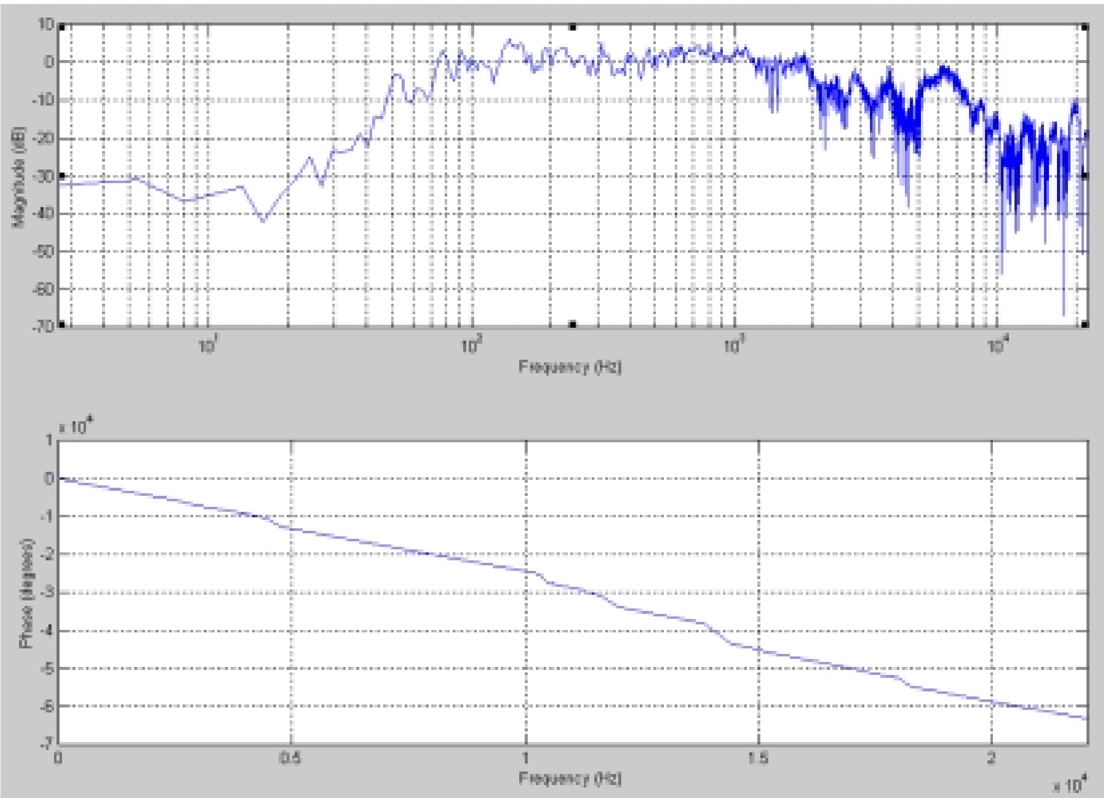

Figure 6.6 On axis frequency response of all 8 unfiltered drivers

As can be seen the array has a much smoother response from 100 Hz to 2 KHz with a reduced trough from 2 KHz to 4 KHz than the individual driver.

The Basic Steering Model

A Simulink model was developed as shown in figure 6.7, to steer the beam by 15 degrees. The through channel for the reference channel can be seen passing from input 2 to output 9.

The model was measured using Smaart Pro, but first a set of readings from an unsteered model was taken to use a benchmark.

The floor in front of the array was marked out as shown in figure *** and the microphone was positioned at each grid location pointing at the centre of the array. The graphs are labelled as 1m @ 15 degrees etc. As can be seen from the grid, each set of readings consists of 28 separate measurements, which is a large amount of data to consider. Smaart pro allows the combination of up to four graphs on one set of axes for the purposes of comparison. Figure *** shows the unsteered array measured at all positive angles at 2m. As expected, the amplitude of the response drops away as the measurements move away from the on axis hot spot. Figure *** shows the same measurement for the array with the model in figure *** implemented. The graph clearly shows that the 15-degree off axis line is at a higher level than the on axis line, indicating that the main lobe has been steered 15 degrees off axis.

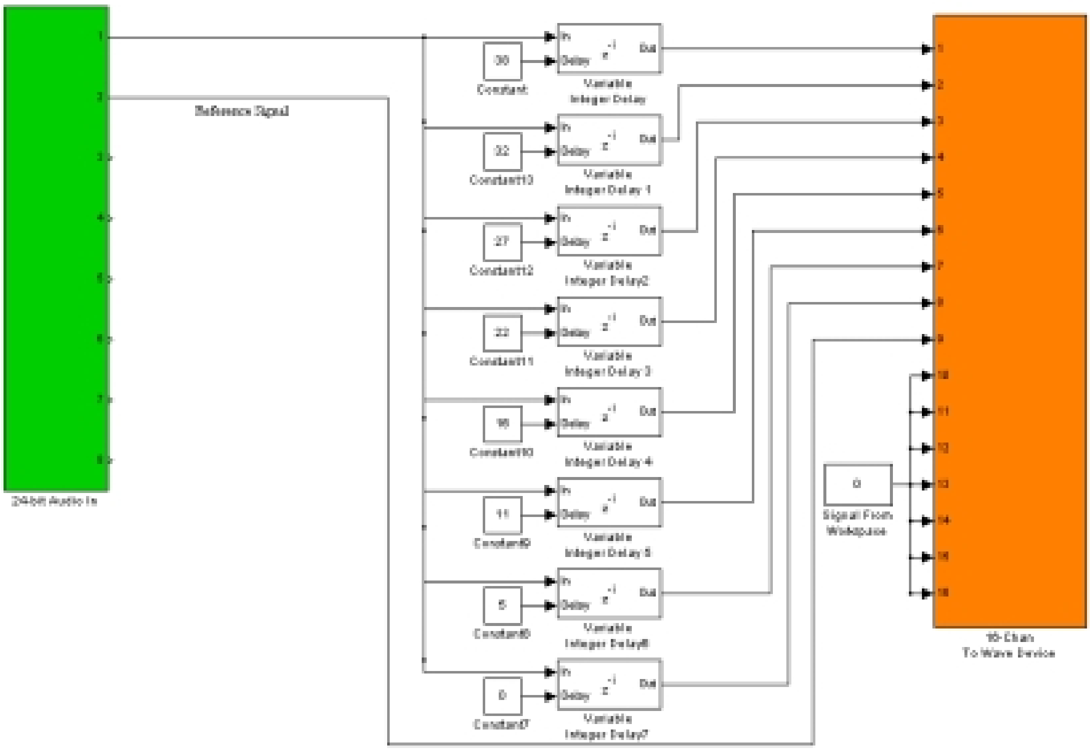

Figure 6.7 15 degree beam steering model

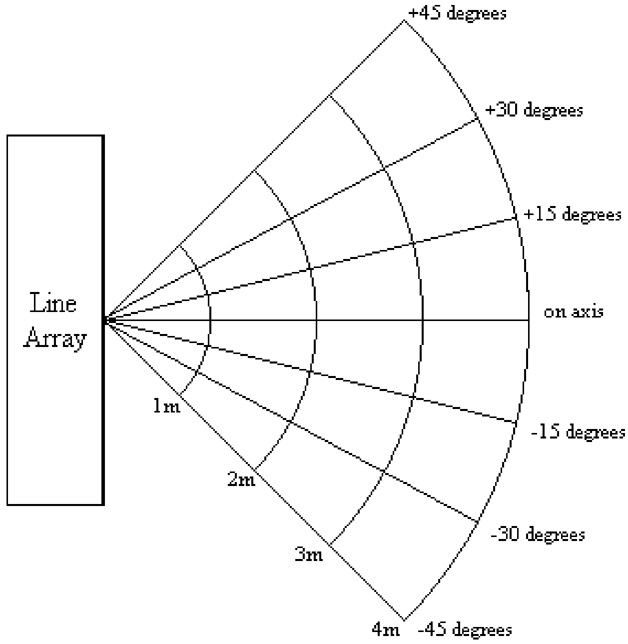

Figure 6.8 Line array microphone measuring positions

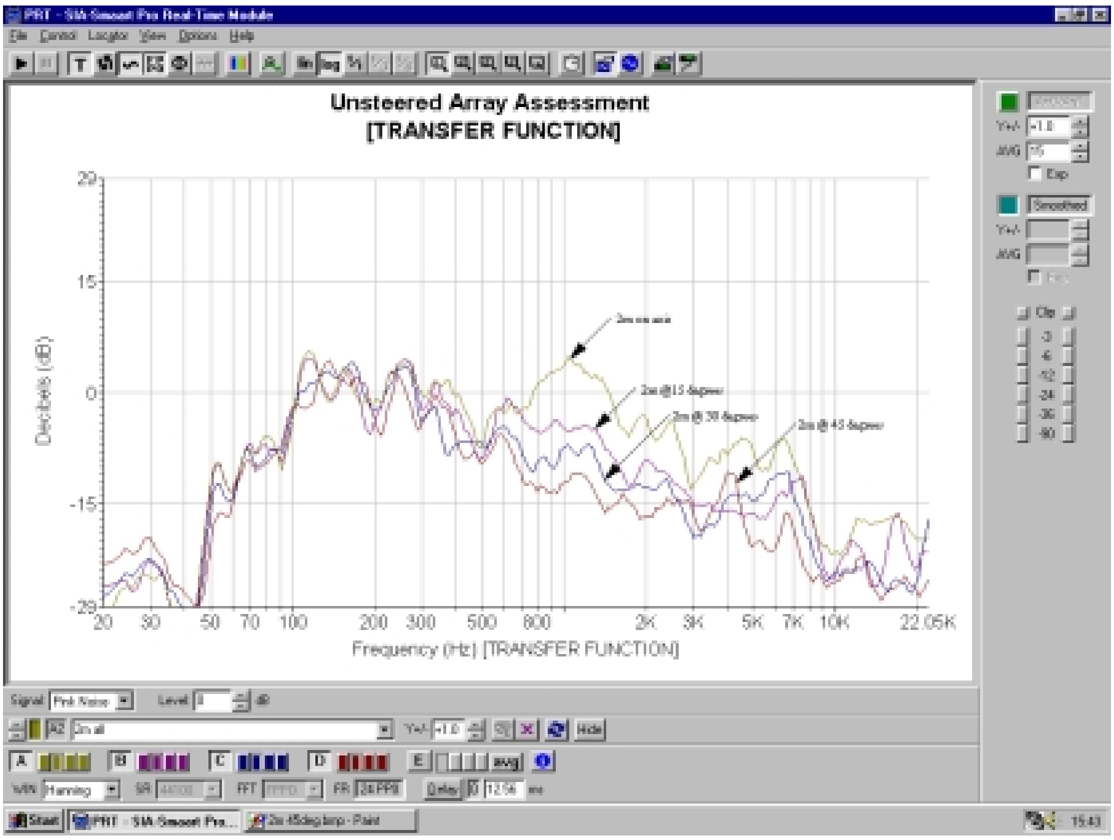

Figure 6.9 Unsteered array frequency response

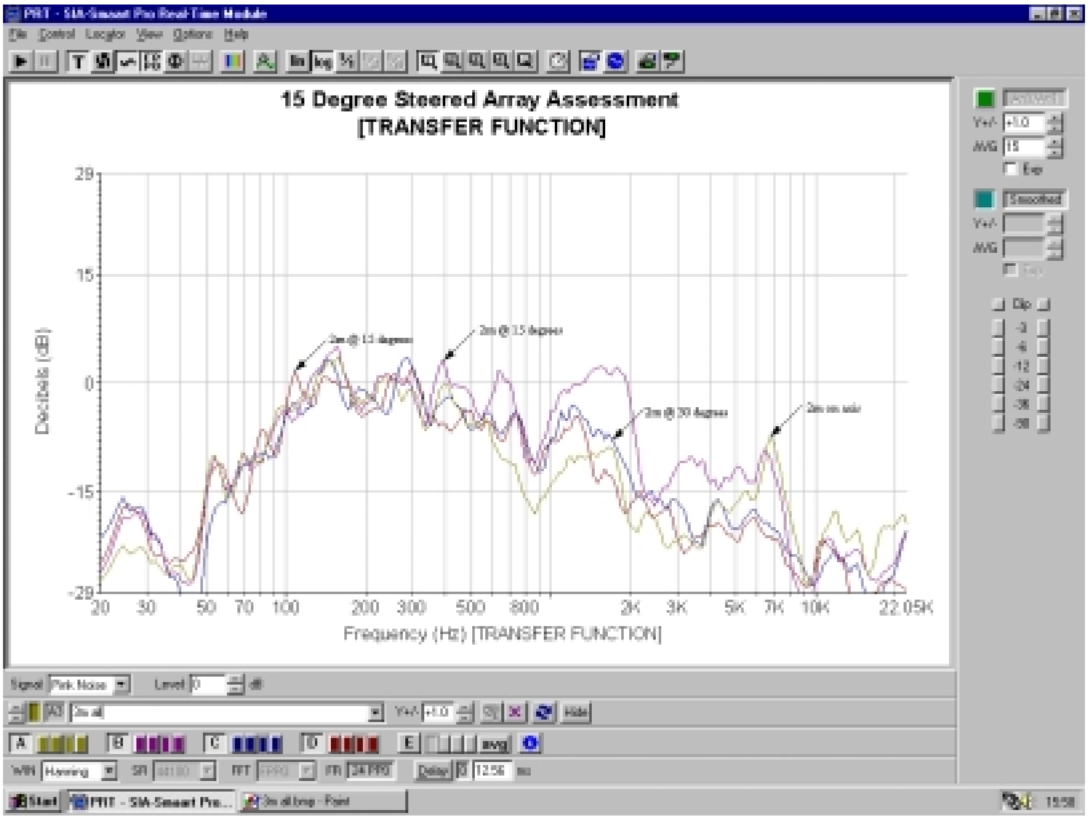

Figure 6.10 Steered array frequency response

The Dual Beam Steering Model

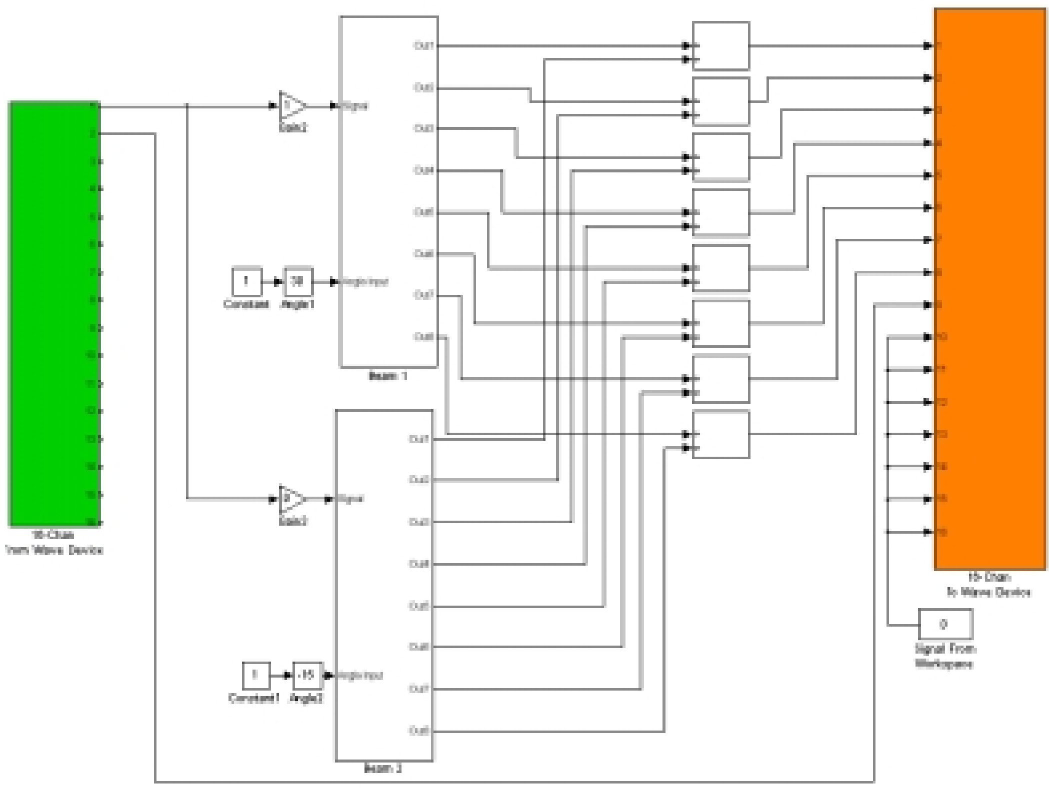

Figure 6.11 Dual beam steering model

The model in figure 6.11 was implemented in Simulink and measured in the same way as the simple steering model. The graphs produced were inconclusive, but a subjective listening test with audio produced the expected hot spots at 30 degrees and –15 degrees off axis. The audio test was repeated to a final year live performance student specialising in audio systems and a Phd student specialising in surround sound and both agreed that distinctive hot spots could be heard. They were not told the angles that the beams had been set to, but located them with a reasonable degree of accuracy.

Filtering and Steering Assessment

Two basic layouts of filters were tested with and without any steering delays in order to assess which gave the best control over dispersion. The first layout was called a symmetrical array (figure 6.12) where the centre two drivers were the through channels and F1>F2>F3.

The second array layout was called a linear array and each transducer was set up with its own cut off frequency, with the through filter positioned at one end of the array and the cut off frequencies getting progressively lower towards the other end of the array.

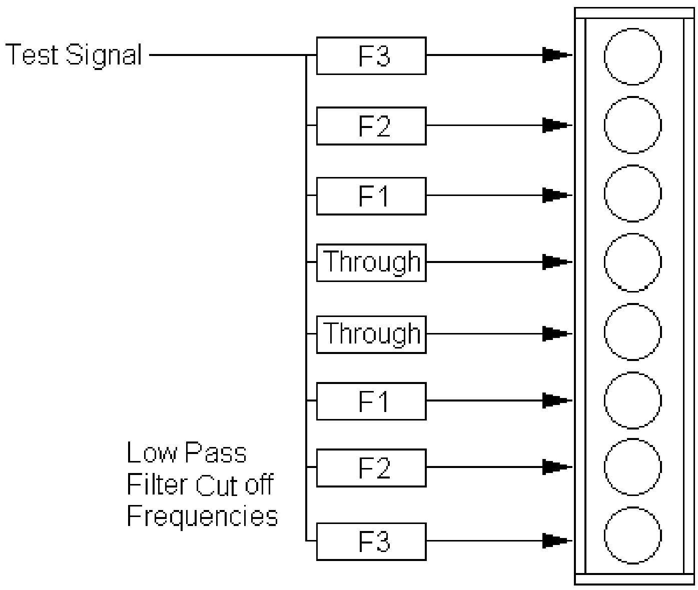

Figure 6.12 Symmetrical filter array

The two array layouts were set up with two sets of cut off frequencies, based on the relationship between the effective aperture of the array and the desired level of directivity. The filters using these cut off frequencies were developed in Sptools and saved as two different workspaces.

Table 6.1 Workspace cut off frequencies

Set 1	Set 2
570	1140
653	1306
764	1528
921	1842
1159	2318
1563	3126
2399	4799
20000	20000

The Symmetrical Array

Test A

The array was set up with a bank of symmetrical filters (figure 6.12) and a set of cut off frequencies from set 1 (table 6.1) so that F1 = 2399 Hz, F2 = 764 Hz and F3 = 570 Hz.

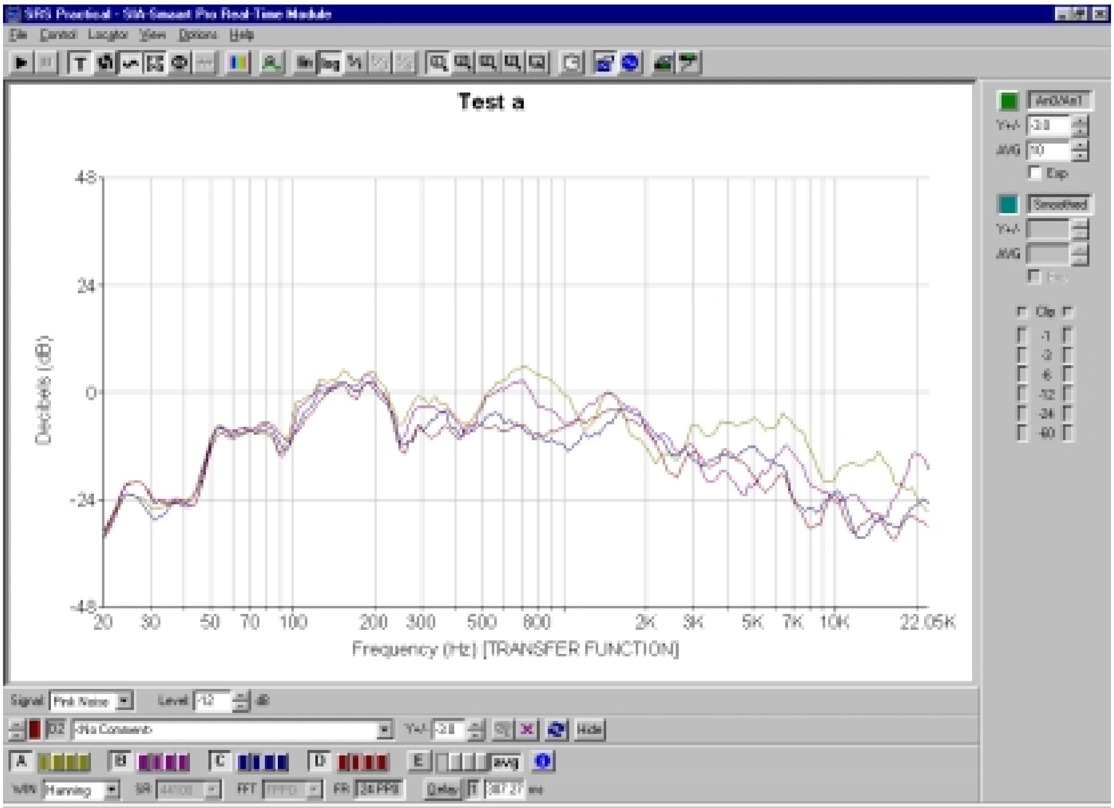

Figure 6.13 1m all negative angles (test a)

The gold trace represents the on axis response @ 1m, the purple trace the response @ 1m @ -15 degrees, the blue trace the response @ 1m @ -30 degrees and the red trace the response @ 1m @ -45 degrees. This colour scheme was adopted for all of the graphs produced in Smaart Pro.

The input to the model was high pass filtered at 570 Hz as the array could not control the dispersion of frequencies below this, so everything below 500 Hz is of little interest as it was due to the fact that the high pass filters exhibited a roll off that allowed frequencies below 570 Hz to pass and they would all be allowed though the network of low pass filters to all eight channels, accounting for the relatively high level.

Figure 6.13 shows clear off axis attenuation between 500 Hz and 1 KHz and again between 3 KHz and 9 KHz, with the region in between getting slightly unusual results again probably due to the test environment. What is interesting to note is that the off axis attenuation is reasonably constant, between approximately 10 dB and 20 dB over the frequency range of 500 Hz to 8KHz when the room effects around 1KHz are taken into account.

Although 28 graphs were produced to fully test this arrangement of filters, only two are shown as they convey the effectiveness of the dispersion control over a given frequency range.

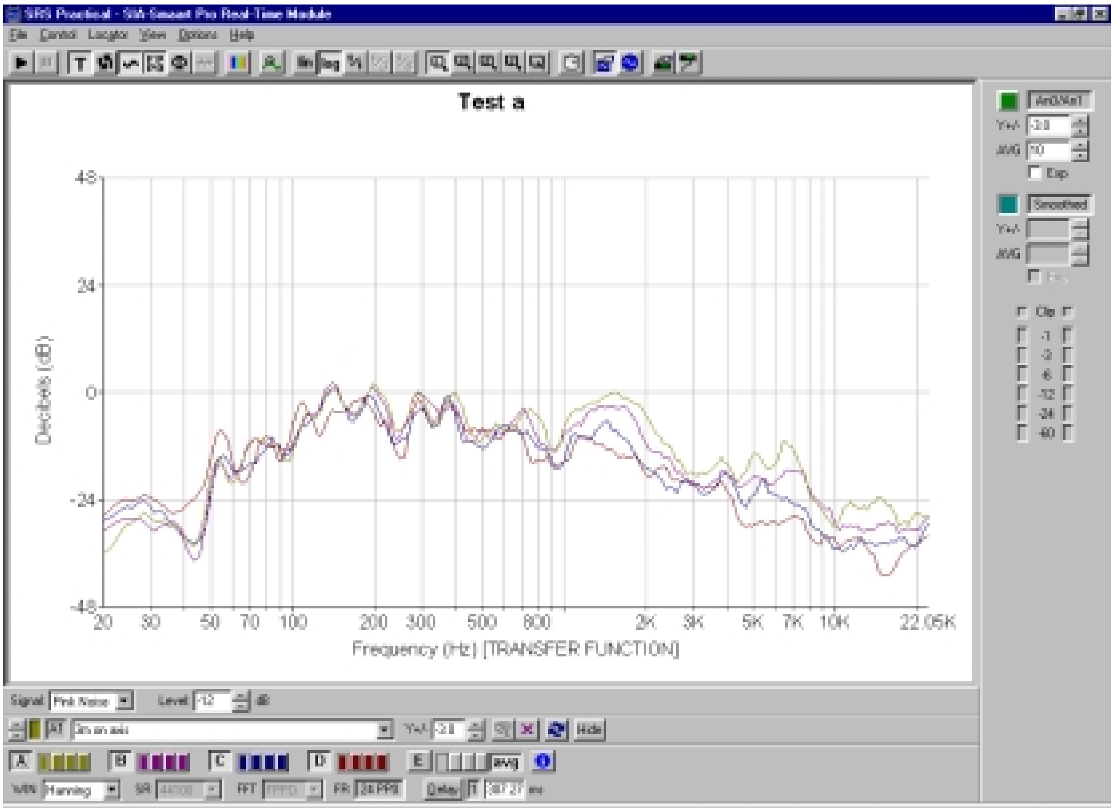

Figure 6.14 3m all positive angles

Figure 6.14 also shows that the array can control the dispersion reasonably successfully above 500 Hz, but not below this frequency as the array was not long enough.

Test B

Test B used the same set of symmetrical filters as test a, but with cut off frequencies from set 2, such that F1 = 4799 Hz, F2 = 1528 Hz and F3 = 1140 Hz. The same high pass filter with a cut off of 570 Hz (and an order of 50 taps) was applied to the input.

Figures 6.15 and 6.16 show that the frequency response dropped away off axis but to a lesser extent than in test a. This could be due to the coarseness of the measurements not picking up increased side lobes that occur with higher directivity.

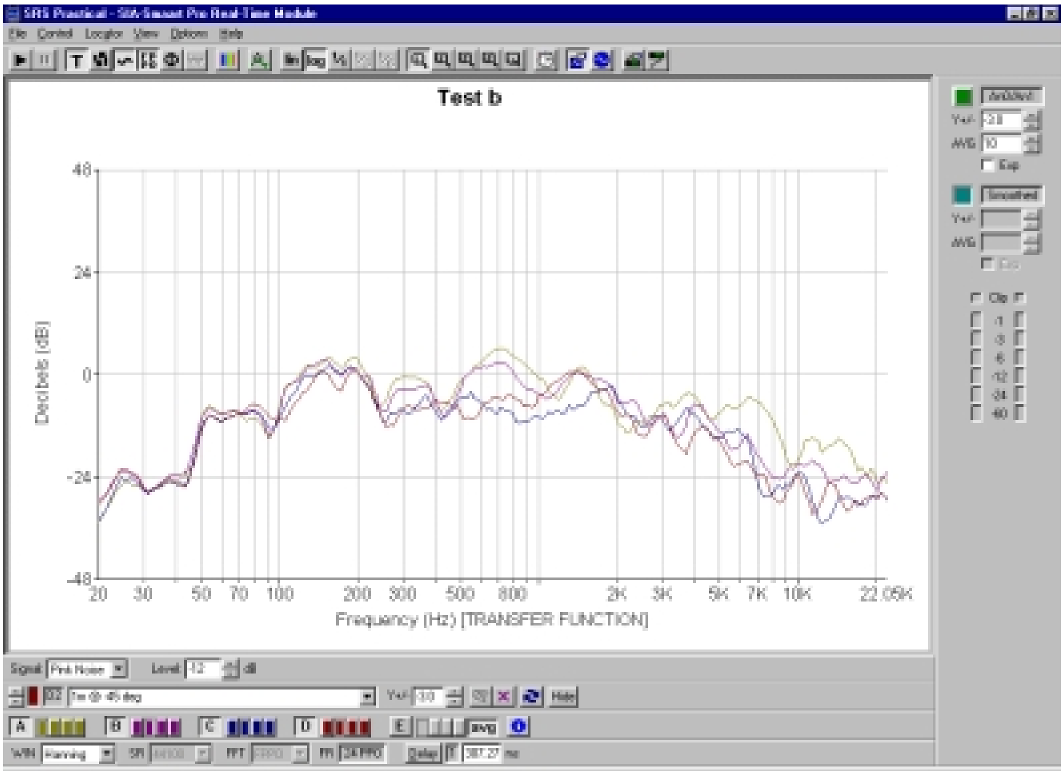

Figure 6.15 1m all negative angles (test b)

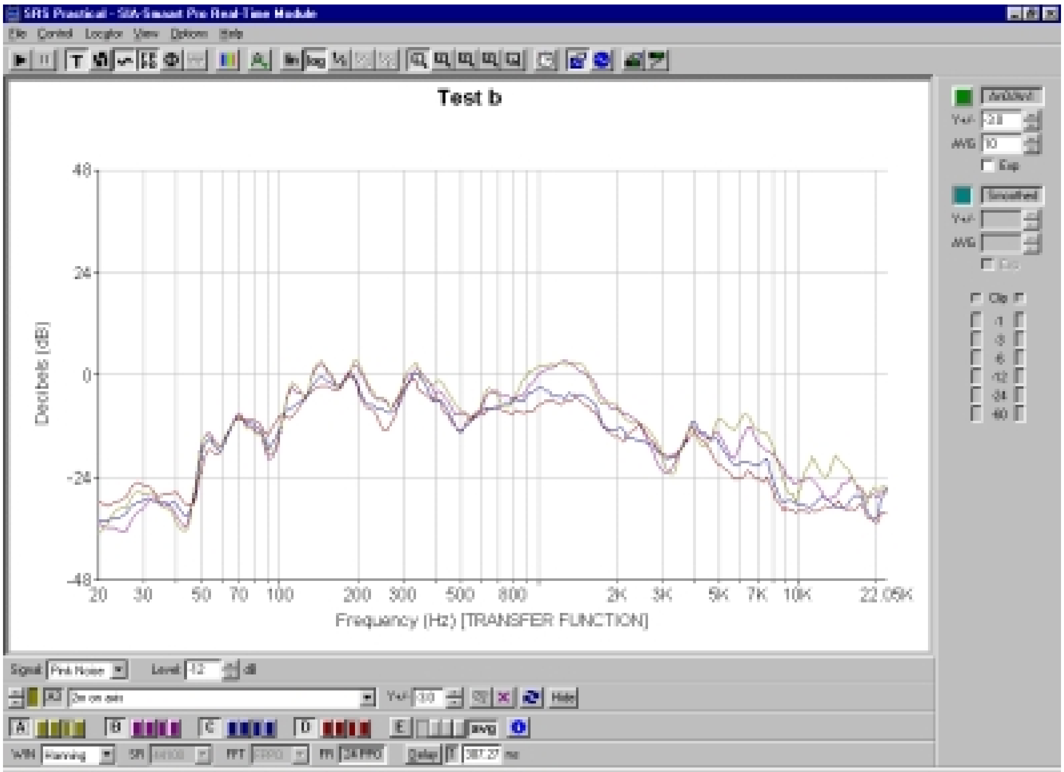

Figure 6.16 2m all positive angles

Test C

Figure 6.17 shows the simulink model used in test C. The array was set up with a bank of symmetrical filters (figure 6.12) and a set of cut off frequencies from set 1 (table 6.1) so that F1 = 2399 Hz, F2 = 764 Hz and F3 = 570 Hz.

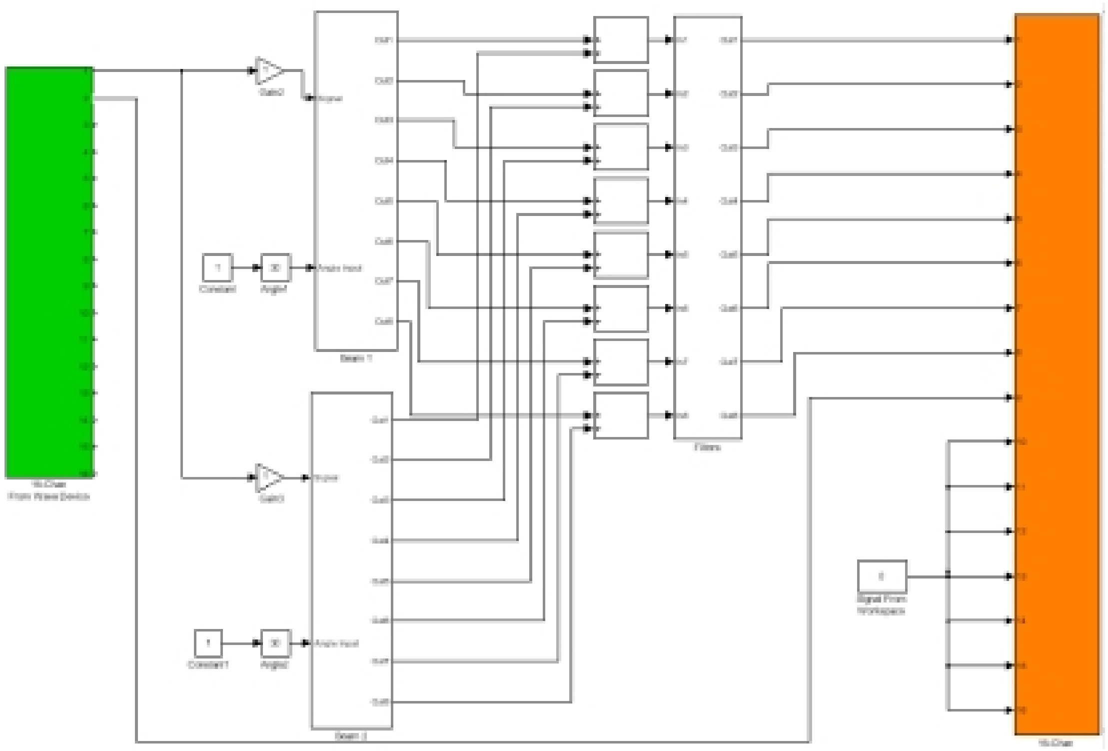

Figure 6.17 Dual beam steering model with symmetrical filter array

Angle 1 was set to 30 degrees and angle 2 was set to –15 degrees. Figure 6.18 shows that the dispersion control off axis is again clearly evident above 500 Hz, but the beam steering cannot be seen. Again the system was tested with an audio signal and the hot spots, one at +30 degrees off axis and the other at –15 degrees off axis were clearly discernable.

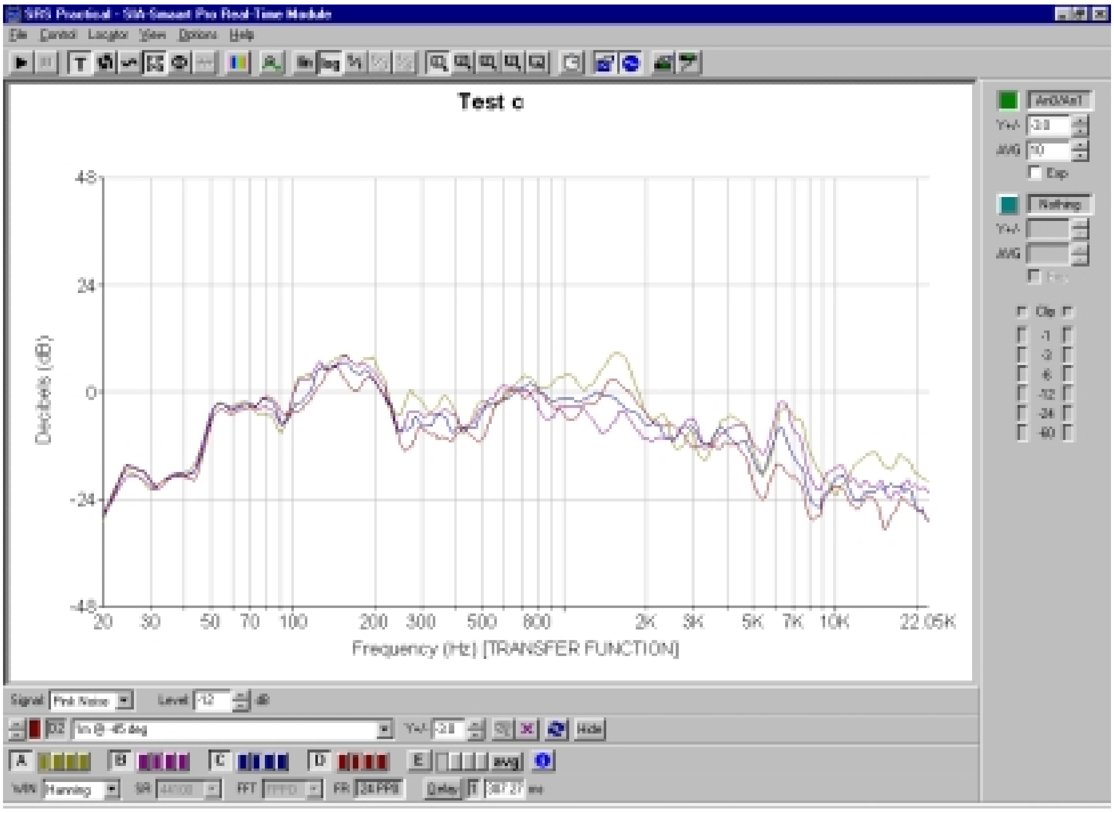

Figure 6.18 1m all negative angles (test c)

Test D

Test D was essentially the same as test C, except that the workspace was changed such that F1 = 4799 Hz, F2 = 1528 Hz and F3 = 1140 Hz. The two angles were left the same to allow more direct comparisons to be drawn.

As with test B, the cut off frequencies designed to give a more directional response actually produce less off axis attenuation, again possibly due to the production of more side lobes that were not detected due to the coarse resolution of the measurements taken.

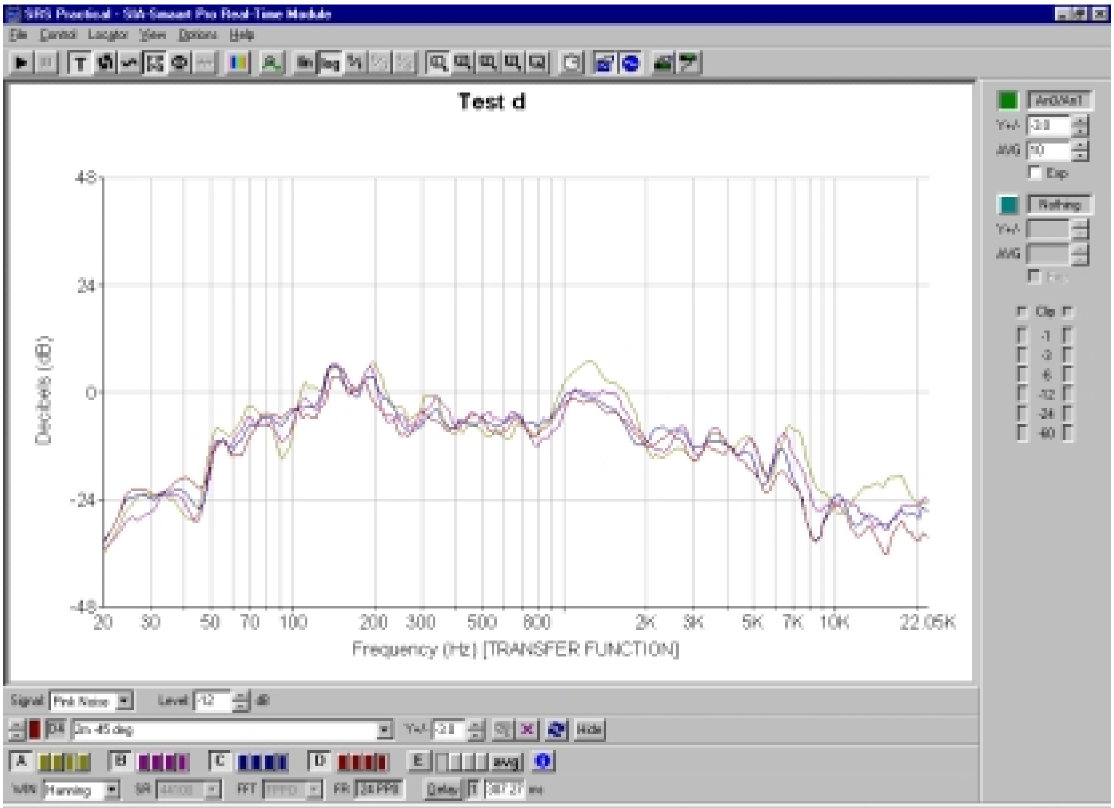

Figure 6.19 2m all negative angles

The Linear Array

In order to test the linear array, the arrangement of the measuring conditions was first altered. Figure 6.20 shows the on axis response being in line with the through channel at one end of the array instead of being aimed at the centre of the array as it was for the symmetrical array.

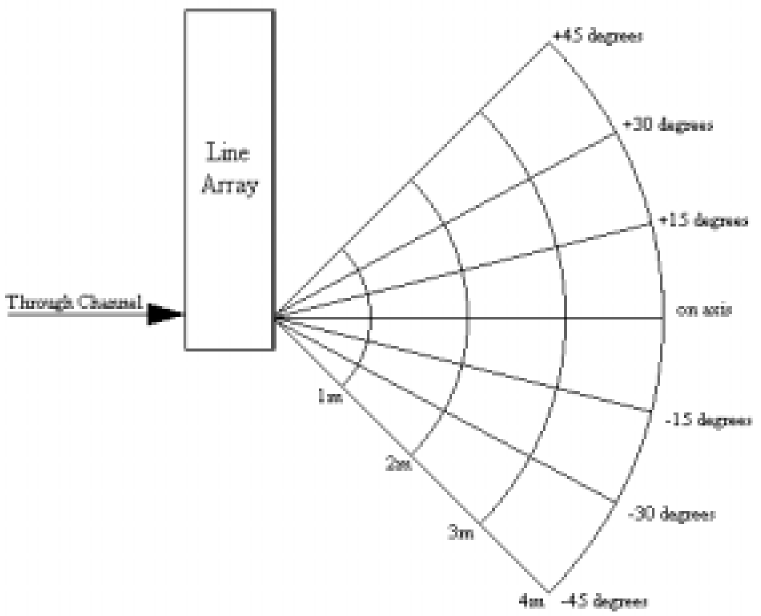

Figure 6.20 Linear microphone measuring positions

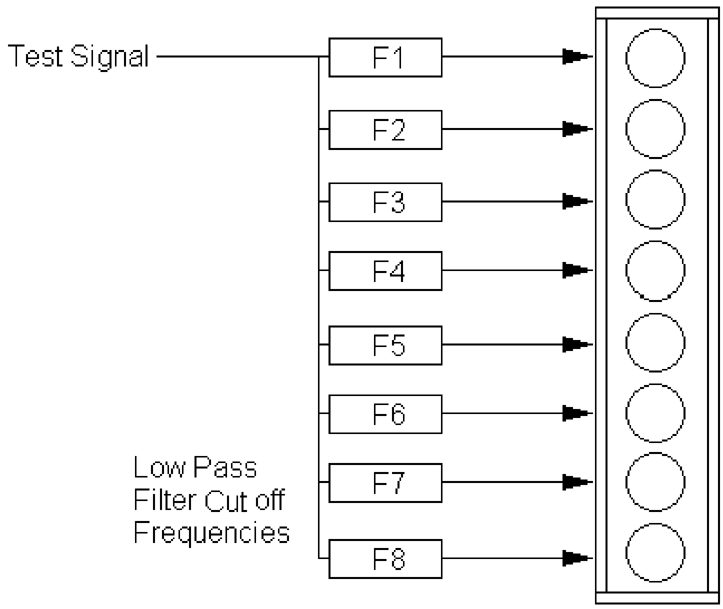

Figure 6.21 Linear filter array

Test E

The array was set up as shown in figures 6.20 and 6.21 with dispersion control filters only and no steering delays. The workspace was loaded into Matlab such that F1= 570 Hz, F2 = 653 Hz, F3 = 764 Hz, F4 = 921 Hz, F5 = 1159 Hz, F6 = 1563 Hz, F7 = 2399 Hz and F8 = 20 KHz. The input signal was also high pass filtered before the low pass filters by a 50 tap FIR HPF whose cut off frequency was set at 570 Hz.

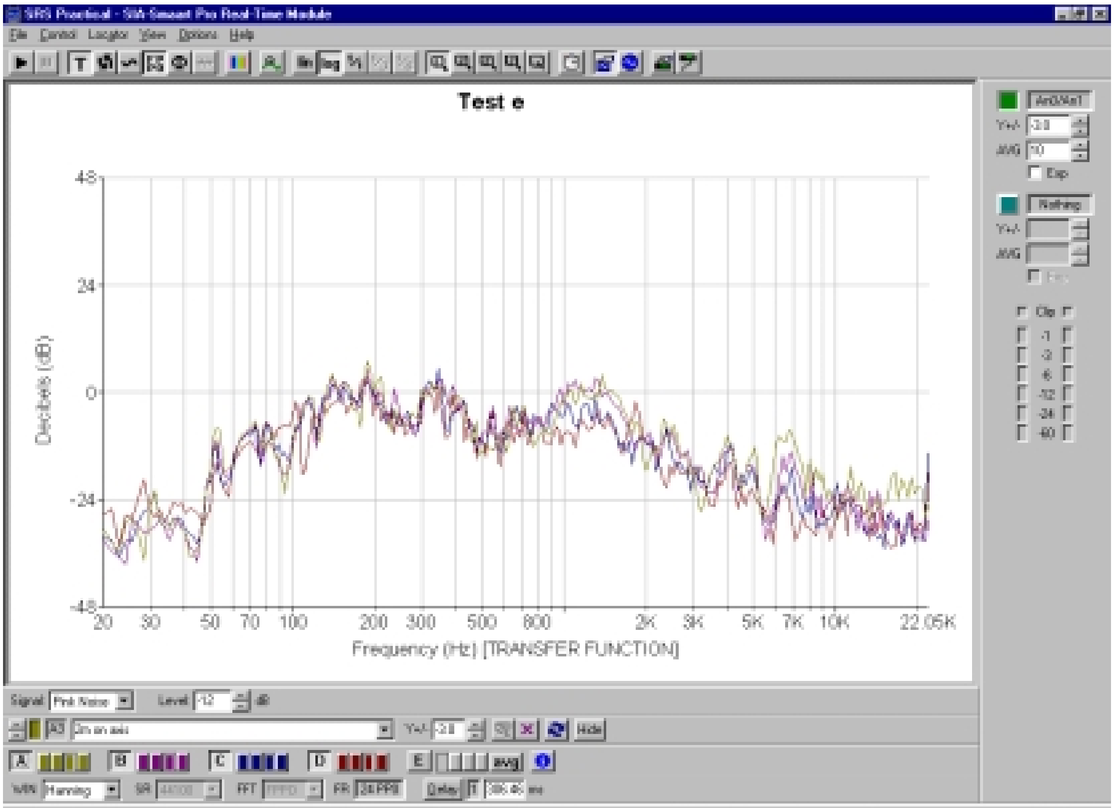

Figure 6.22 2m all positive angles

Figure 6.22 shows the response falling away off axis, but by a smaller amount than was observed in test a where the same filter cut off frequencies were implemented in a linear array. There is no observable dispersion control below 700 Hz as the array was too short to control such frequencies.

Test F

Test F, like test E was only assessing the performance of the dispersion control filters and no steering delays were in used. Test F used a Matlab workspace such that F1= 1140 Hz, F2 = 1306 Hz, F3 = 1528 Hz, F4 =1842 Hz, F5 = 2318 Hz, F6 = 3126 Hz, F7 = 4799 Hz and F8 = 20 KHz. The input signal was also high pass filtered before the low pass filters by a 50 tap FIR HPF whose cut off frequency was set at 570 Hz.

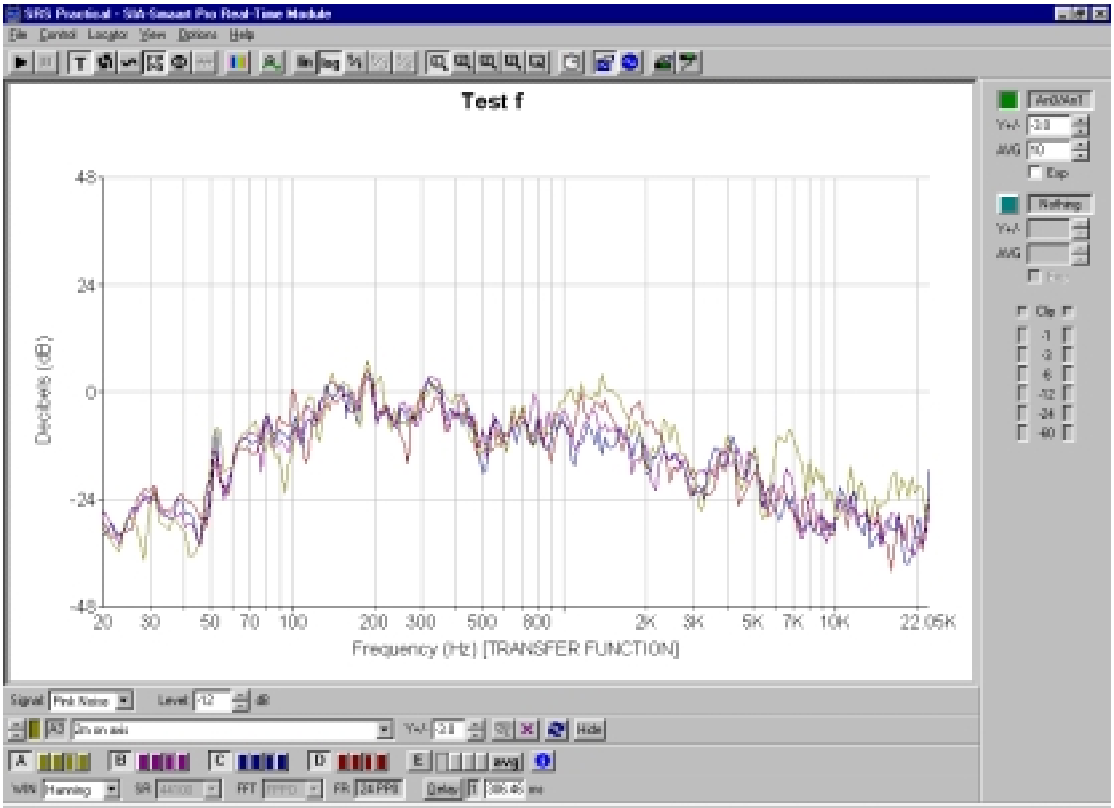

Figure 6.23 2m all positive angles

Figure 6.23 shows only a small amount of off axis attenuation above 500 Hz and no dispersion control below this frequency.

Test G

Test G bore similarities to test E in that it used the same set of dispersion control filters. Test G was also similar to test C in that it also implemented the dual beam steering model in Simulink using the same two angles of +30 degrees and –15 degrees.

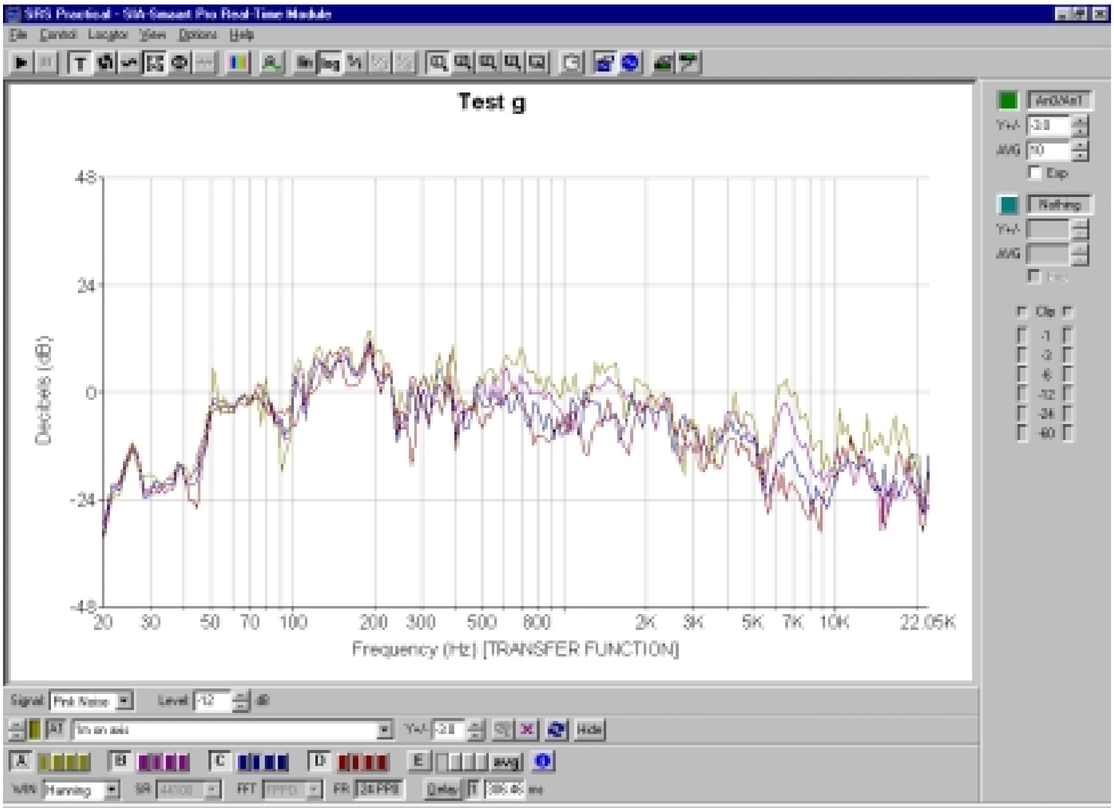

Figure 6.24 1m all negative angles (test g)

Figure 6.24 shows reasonable off axis attenuation above 500 Hz, with little dispersion control below that frequency. Although no steering effects are discernable from figure 6.24, an audio source used with the steering model produced audible hot spots at angles of +30 degrees and –15 degrees.

Test H

Test H was the same as test G except that the Matlab workspace was changed such that F1= 1140 Hz, F2 = 1306 Hz, F3 = 1528 Hz, F4 =1842 Hz, F5 = 2318 Hz, F6 = 3126 Hz, F7 = 4799 Hz and F8 = 20 KHz. The input signal was also high pass filtered before the low pass filters by a 50 tap FIR HPF whose cut off frequency was set at 570 Hz.

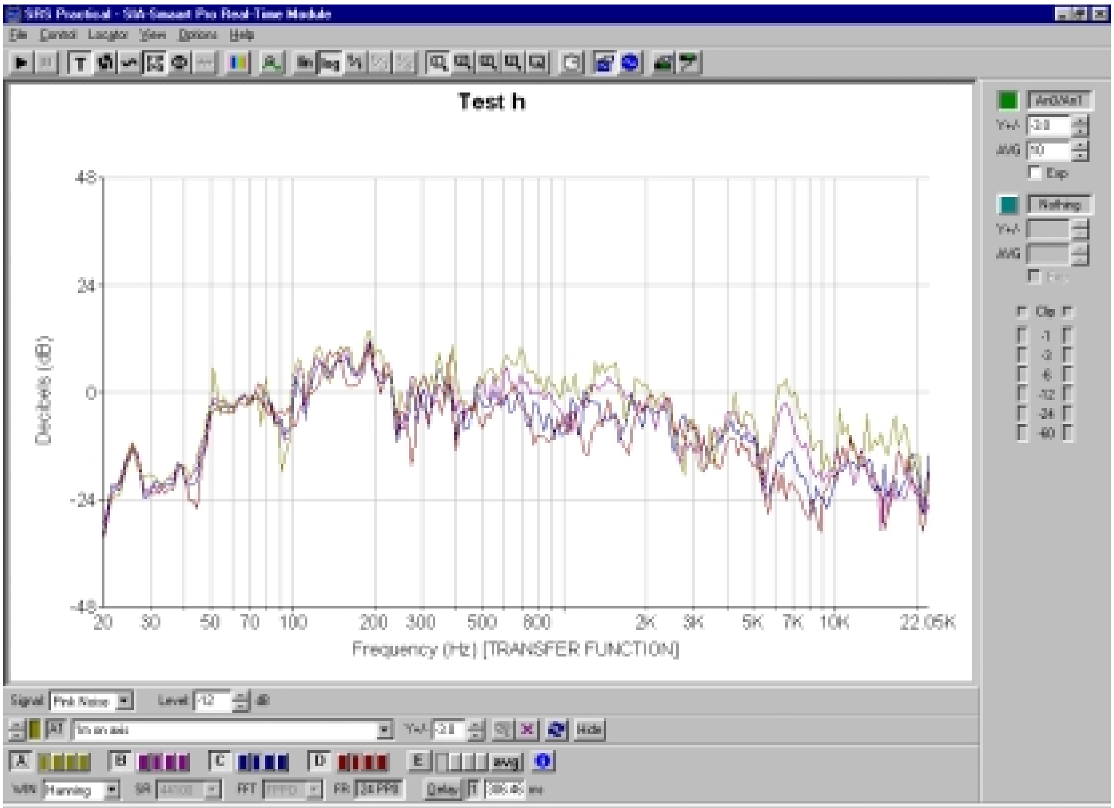

Figure 6.25 1m all negative angles

Figure 6.25 shows the off axis rejection above 500 Hz with little dispersion control below this frequency. Figure 6.26 also shows that the -15 degree off axis line has a larger amplitude than the on axis response between 1 KHz and 2 KHz, which could be due to the beam steering in the Simulink model. When the model was run with an audio sound source, the hot spots due to beam steering were clearly audible at =30 degrees and –15 degrees.

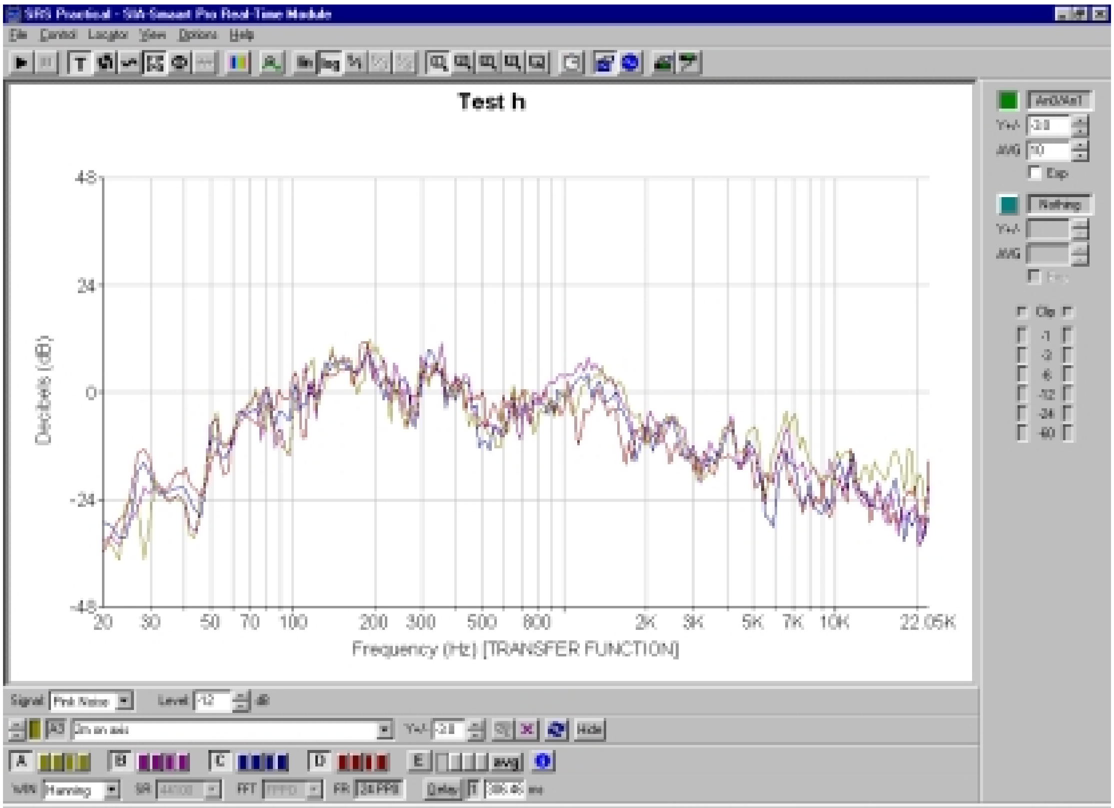

Figure 6.26 2m all positive angles

Chapter 7

Conclusions and Recommendations

The conclusion will be broken down into salient points on a chapter by chapter basis followed by a more in depth assessment of the project in terms of its educational value, methods in which it could be improved and a brief look at it's implementation on the Motorola DSP 56000.

Chapter 1
Chapter one addressed the need for line arrays, their use in difficult acoustic situations, their architectural suitability and their importance for speech intelligibility.

Chapter 2
Chapter two detailed the extensive research that was carried out into line array theory, control concepts and enclosure design. Research was carried out using a number of sources that included:

- University of Derby library.
- Inter library loans.
- The Internet.
- Lecture notes from the music technology and audio systems design course.
- E-mailing Duran audio – a company specialising in dsp controlled line arrays.
- A senior lecturer specialising in DSP.
- A Phd student with extensive knowledge of dsp, audio, Matlab and simulink.
- A lecturer with extensive experience in loudspeaker design, construction and testing.

Chapter 3
Chapter three detailed the design and construction of a prototype line array. The line array was constructed within budget and all of the health and safety implications of working with power tools and materials that were potentially hazardous to health were met. By carefully planning the layout, the sheet material requirements of the project were halved from the initial estimate, with an associated reduction in cost.

Chapter 4
Chapter four included a comparison of analogue, FIR and IIR digital filters including their relative merits and shortcomings. The chapter then went on to look at Simulink, delay theory, digital filter theory (including windowing, impulse responses, the Fourier transform, time domain filtering, frequency domain filtering and the advantages and disadvantages of each).

Chapter 5
Chapter five looked at the software implementation involved with the project, by breaking the software down into two parts.

The first part of the chapter looked at beam steering, with a simple delay model implemented in Simulink. A more complex model was then developed to allow real time steering of the beam.

The development of a dual beam steering model made innovative use of Simulink to convert positive and negative angles entered into the system using a gain slider to generate the integer sample delays necessary for beam steering, by using the modulus of the angle to generate a positive integer delay and the sign of the angle to route the delay to the appropriate loudspeaker.

The second part of the chapter looked at methods of controlling dispersion using low pass filters with differing cut off frequencies. The problem with this method of dispersion control was that the system developed an excessively large bass response. Two solutions were attempted, the first being the development of an inverse filter to correct the magnitude response, which was too long to implement in real time. The second was to custom design the dispersion control filters using a software package called SPTools, that allowed the design of low pass filters which all have the same order but with less stop band attenuation.

Chapter 6

Chapter six consisted of the tests performed on the array and the results obtained. As well as detailing the testing and results, the chapter also looked at how the test methods were refined over the course of the project.

The frequency response of a single driver was measured for two reasons. The first reason was that the horizontal frequency response of the array is identical to the frequency response of a single driver. The second reason was that the frequency response of a single driver provided information pertaining to the frequencies that the array would be capable of reproducing. The measurements taken were representative of a small diameter budget loudspeaker, but compared reasonably well to a frequency response published by Duran audio for one of their professional transducers of a similar diameter.

The frequency response of the unfiltered array was measured using Simulink, Matlab and a Neutrik measurement microphone. This measurement demonstrated the natural behaviour of an uncontrolled line array, with the dispersion approaching an omni-directional pattern at low frequency, with the dispersion pattern narrowing as frequency increased. These observations reinforced the need for low frequency dispersion control. The measurement was repeated later on in the project once the test methods had been refined with a higher specification microphone and professional audio measurement software that allowed the process to be carried out more quickly. The test was repeated so that it could be used as a benchmark against further developments to the control software.

The basic steering model developed in Simulink was measured and compared to the Smaart pro graph obtained for the unfiltered array. The delay theory was borne out with a 15 degree off axis main lobe clearly visible and at a higher level than the other traces.

The majority of testing in chapter 6 involved the assessment of the filtering and steering techniques. Two types of filter array were examined, a linear filter array and a symmetrical array. Two sets of filter cut off frequencies were tried in each type of filter array. The test details are summarised in the table below.

Table 7.1 Summary of filtering and steering assessments

Test	Array type	Set of cut off frequencies	Off axis attenuation above 500 Hz	Was the steering model implemented?
A	Symmetrical	Set 1	Good	No
B	Symmetrical	Set 2	Reasonable	No
C	Symmetrical	Set 1	Good	Yes
D	Symmetrical	Set 2	Reasonable	Yes
E	Linear	Set 1	Reasonable	No
F	Linear	Set 2	Poor	No
G	Linear	Set 1	Reasonable	Yes
H	Linear	Set 2	Reasonable	Yes

From these tests, the following conclusions can be drawn:

- For both the linear and symmetrical arrays the filter cut off frequencies in set 1 provided better off axis attenuation using the measurement system detailed in the chapter. Finer resolution measurements could indicate higher directivity with the set 2 cut off frequencies but with more prominent side lobes.

- Both linear and symmetrical arrays can be readily implemented; with the height of the installation dictating which filter array is used. If the height of the listeners' ear is near the centre of the enclosure, then the symmetrical array could be implemented, but if the height of the listeners' ear is near the bottom of the array (that is, the array is higher up) then the linear line array could be implemented.

- The Smaart Pro measurements of the dual beam steering model were inconclusive, possibly due to the pink noise test signal that was used being both complex and random and making excessive demands on small budget loudspeakers. The other possibility for the inconclusive results was that the two different sets of delayed signals were interfering with each other as each loudspeaker attempted to reproduce the two signals. When tested subjectively with an audio input, the dual beam model performed well with two different listeners being able to detect the 'hotspots' produced by the two main beams of the model. One listener was a PhD student specialising in surround sound audio, with extensive experience of critical listening and the other was a final year live performance technology student specialising in audio systems. Neither listener was informed of the location of the beams before listening. The success of the subjective test in conjunction with the success of exporting the integer delays to the Matlab workspace to test the steering system and the successful results of the simple steering model suggest that the dual beam steering model works well and it is the method of measuring it that needed to be changed, possibly by using a sine wave sweep.

- The dispersion control filters did control the low frequency response of the array, but only down to 500 Hz, which was the lowest frequency that an array of 1.2 m in length could control.

Project assessment

The project was successful on several levels.

From an educational point of view, the project provided the following benefits:

- Advances were made in terms of research techniques.
- Experience was gained in enclosure design and construction methods.
- Further experience of Matlab, Simulink, SPTools and Smaart Pro was obtained.
- An in depth understanding of line array theory was developed.
- DSP theory was advanced and reinforced.
- Theory involving digital filter design was extended and reinforced.
- Project planning and management skills were developed.
- The ability to plan and carry out a project on time and within budget was achieved.

DSP Implementation

The line array control system was not implemented on the Motorola DSP 56000's as originally planned for the following reasons:

- Delays in the delivery of the loudspeakers due to university bureaucracy caused a re-assessment of the second half of the project Gantt chart.
- The specification of the DSP 56000 is such that it would limit the systems sampling frequency and the length of filters that could be implemented.
- The software had essentially already been written in the Real Time Audio DSP course, where the delay was implemented using a circular buffer and modulo addressing and the FIR filter coefficients were developed in Matlab and exported to an .asm file, which was then imported into the DSP 56000. The coefficients were then used to filter the incoming signal using the multiply and accumulate (mac) command and the repeat (rep) command which repeats the next line a specified number of times in a more efficient manner than a for next loop.

Recommendations

The project could be improved by:

- Building a longer array with more loudspeakers to control the dispersion of lower frequencies.
- Using higher specification transducers would lead to an improved frequency response.

- Logarithmically spaced transducers would improve the dispersion control for a given number of transducers, with an associated increase in the complexity of the control theory.
- The assessment of the dual steering model may have given more positive results if a sine wave sweep test signal was used instead of pink noise.
- Redesigning the test methods for assessing the arrays performance, as the system used involved 28 separate mic positions and measurements for each model making the array development a very lengthy process producing an large amount of data to assess.
- The measurement sampling resolution of every 15 degrees needed to be reduced as line arrays can regularly produce beam widths of just a few degrees, and much of the detail could easily be missed.
- A quieter measurement room would improve the accuracy of the measurements taken. The room used for measuring the array was subject to noise from pc cooling fans, building services and the industrial drilling machine in the next room.

In a final summing up, the project has demonstrated that the coupling of long standing line array theory with contemporary digital control systems can produce very narrow constant directivity wave patterns which are extremely useful in difficult acoustic environments for optimising speech intelligibility.

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