TAPERED LINE ARRAYS – A SOLUTION FOR THE SOUND SYSTEM IN THE OPERA THEATRE, SYDNEY OPERA HOUSE

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A six-channel loudspeaker system has been recently installed in the Opera Theatre at Sydney Opera House. The specification for the system imposed demanding requirements relating to frequency response, directional control and acoustic isolation between audience areas along with restricted locations and sizes for some loudspeakers. To meet these requirements, the authors designed a solution based around four types of custom-made tapered line arrays that provided high directional control at low and midrange frequencies. This paper discusses the design and implementation of the system.

0 INTRODUCTION

In late 1998, a six-channel loudspeaker system was installed in the Opera Theatre at Sydney Opera House in response to a demanding specification for the design, supply and installation of loudspeakers and related signal processing hardware. Comprising of a left/centre/right channel system for the stalls and another left/centre/right channel system for the dress circle, the system had to generate sufficient sound pressure level (SPL) and fidelity to handle a rock band or modern dance. It also had to be capable of providing subtle and non-evident amplification for enhancement of classical music and opera.

A number of line-array systems comprising stackable elements have recently become available for large-scale sound reinforcement. (Examples are [1] [2]) These arrays are able to provide excellent control of directivity down to the lower frequency limit of their drivers and are currently finding favour with the touring sound industry. Although these arrays were not readily available when this project was commenced, they would not have met all the Opera House's requirements and therefore the loudspeakers described in this paper are of relevance.

The Opera Theatre has the usual architectural elements of a proscenium arch lyric theatre. The maximum seating capacity is 1547, with 866 in the stalls, 466 in the dress circle and 198 in side boxes, some of which have restricted sightlines. Of the stalls seats, 6 rows, with approximately 270 seats, are located underneath the overhang of the dress circle. The stage opening is 11.5 m wide and 7.0 m high. Surrounding the stage is a 19 m high proscenium wall that is vertically pleated, with an offset of 250 mm between the pleats. Directly in front of the 800 mm high stage is the orchestra pit. The reverberation times were measured using the existing sound system and found to range between 1 and 1.4 seconds over the frequency range 40 Hz to 4 kHz. Fig. 1 shows a plan of the theatre, Fig. 2 shows a sectional view along its centre line, while Fig 3 shows a view looking toward the stage and indicates the relative positions of the loudspeaker systems as they were finally installed. Additional plans, specifications and photographs of the Opera Theatre are available from the Sydney Opera House [3].

1 SPECIFICATION AND REQUIREMENTS

The specification was produced internally by the Sydney Opera House, predominantly by the incumbent Manager of Sound and AV. In addition to the performance requirements described below, the specification required that; a) the brand of the system be internationally recognisable and acceptable for the comfort of visiting performers and producers and, b) tenderers demonstrate from technical perspectives other than using commercial modelling software that the system would deliver the required performance.



1.1 Frequency Response and Coverage Consistency

The frequency response and coverage parameters were to be derived from the measured impulse response (IR) of each sub-system and were to be met over 90% of the audience area. The required frequency response was 40 Hz to 12 kHz \pm 6 dB with 1/3 octave smoothing when measured with; i) direct field + 80 ms of IR data for 125 Hz to 12 kHz and, ii) with direct field + 1 second of IR data for 40 Hz to 125 Hz. The required consistency of coverage was stated as less than \pm 6 dB variation in the total SPL in the range 250 Hz to 5 kHz when measured with the direct field + 150 ms with the IR shaped to a pink-noise like spectrum using a 3dB/octave low-pass filter.

These two specifications are a large departure from typical specifications based around the total sound field in the audience measured with pink noise and a 1/3 octave analyser and they reflect an attempt to better describe our aural perceptions of frequency response.

1.2 Sound Pressure Level and Clarity Ratios

The required minimum total SPL over the audience was to be 106 dB lin Leq with a 12 dB crest factor using pink noise shaped to the spectrum of modern music. Fig 4 displays such a spectrum this was distilled from and averaged over a number of sources. In addition, for frequencies below 125 Hz the expected displacement-limited SPLs produced by the low frequency drivers at 30 m were to be stated for both the direct and reverberant fields.

To produce consistent early to late ratios, the directivity and radiated power of each system were to be as constant as possible over the full frequency range. Although clarity ratios and early decay times (EDT) were not specified, tenderers were required to state the anticipated worst-case C50, C80 and EDT results for each octave band.



Fig. 4 Spectrum of Modern Music distilled from and averaged over a number of sources.

1.3 Available Sizes and Locations for Loudspeakers.

For aesthetic reasons, the loudspeakers covering the dress circle were unable to be flown above the stalls near the dress circle and therefore had to be built into the proscenium wall, thereby placing them above the front of the stalls. Loudspeakers for the left and right stalls systems could not be flown at the left and right sides of the stage and had to be built into the articulated concrete structures either side of the stage. Subwoofers had to be built into the proscenium wall.

1.4 Acoustic Isolation Between Areas

The spectrum and level of sound spilling onto the stage sets the available acoustic gain with live microphones and are particularly important when omni-directional radio microphones are placed in an opera singer's wig. The specified equivalent acoustic distance [4] was 1 m and statements were required of the worst-case acoustic gain in each 1/3 octave band between 100 Hz to 10 kHz when an omni-directional microphone was used on the stage 1m from a talker. To ensure that the sound spilling onto stage was subjectively pleasant, its frequency response was to be 40 Hz to 8 kHz +6/-12 dB with direct field +200 ms (smoothed to 1/3 octave). Minimal sound was to also reach the orchestra pit located in front of the stage.

At all frequencies above 250 Hz, the levels of sound from the dress-circle loudspeakers spilling down to the stalls were to be at least 10 dB below the average levels of those speakers in the dress circle area. As the dress-circle seats were some 7.5 m further from the dress-circle loudspeakers than the stalls seats, the required directivity loss of those speakers at a downward angle of 55° (towards the rear of the stalls) was 12 dB above 250Hz.

2 DESIGN

The public tender was won by Jands Electronics Pty Ltd, now Jands Pty Ltd with a solution based around four types of custom, tapered-line-array loudspeakers utilising predominantly horn loaded components. Jands is the Australian distributor of JBL Professional products and elected to utilise JBL components for commercial reasons. In collaboration with Jands, Elecoustics, a firm of acoustical consultants, undertook the design and commissioning of the system.

Similar results would have been possible had other similar components been used. Where a model number has been detailed, this is to allow reference to published specifications.

2.1 Frequency Response and Directional Control

The delivery of a flat frequency response and consistent overall SPL to listeners both near and far from a loudspeaker is extremely important. This can be achieved with a system whose directivity is constant over the audio bandwidth and whose aiming and directivity compensates for different distance losses to each seat. This is therefore predominantly an issue in the vertical plane. In the horizontal plane the seating rows form an arc with relatively consistent distance losses. Control of directivity is not usually undertaken in the low and low-midrange frequencies in smaller systems, but is a feature of this system.

Although the tolerance window for the specified frequency response appeared to be large, the high resolution available with Fast Fourier Transform based measurements and the small time window for that frequency response placed strong demands on the frequency response of each sub-system's direct field. The authors' experience is that, even when excellent loudspeakers are installed in non-ideal locations, significant degradation in frequency response occurs due to phase cancellations with a) other loudspeakers carrying the same program material, and b) the inevitable image sources and diffraction caused by each loudspeaker's local radiating environment. It was therefore anticipated that the tolerance window would not be easy to meet.

The solution for the above requirements was to provide high directional control at all frequencies, both horizontally and vertically. As commercial horn devices were available for high frequencies, directional control had to be achieved for the low and midrange frequencies in both the horizontal and vertical planes through the selection of components (horizontal directivity) and design of tapered-line-arrays (vertical directivity).

One challenge was to achieve the required coverage without the need for multiple loudspeakers. Current sound reinforcement practice often involves multiple full-range loudspeaker boxes (with typical frequency range 70 Hz to 15 kHz) being stacked and splayed in various vertical and horizontal configurations in an attempt to achieve the required coverage. While these arrangements can provide enhancement of low frequency directivity, the interdriver distances are usually greater than ¼ wavelength at midrange and high frequencies, which precludes wellcontrolled, constant directivity at these frequencies. Often, the directivity of the high frequency horns is sufficiently high at higher frequencies that their polar-patterns can be arranged for minimal overlap, but at lower frequencies there is substantial overlap and the directivity of one horn is substantially damaged by the addition of multiple horns. Ureda [5] shows good examples of this behaviour. Another technique with limited application to systems using multiple loudspeakers is to apply a signal delay of sufficient magnitude to one or more speakers to force the comb filters (due to interference) to be narrower than the ear's critical bandwidth without greatly shifting the apparent source of the sound [6].

2.2 Predictions of Array Performance

A prediction of the radiation pattern of each array was made using a computational model that included the contribution of the radiation patterns of the individual drivers. The model follows that of Meyer [7] Eq.1, [8] [9] with one important difference relating to the driver's off-axis phase response. The midrange horns used in two of the line-array types were components taken from commercial full-range products. Time restrictions did not allow laboratory-measurement of the off-axis amplitude and phase responses of these devices as individual components, so their radiation patterns were estimated from the published polar plots for the complete systems (which included crossovers). The exclusion of phase from the measurement data meant that our computation model became similar to the convolution model described by Ureda [10].

To optimise the radiation pattern of each array, a target radiation pattern versus frequency was set up for each array. A spreadsheet-optimisation routine based on least squared errors was then used to steer each array's polar characteristic towards the target by adjusting the inter-driver spacing and the time constants of the tapering filters.

Following completion of the array designs, the predicted radiation pattern for each array was input into the modelling software EASE v2.11 to allow refinement of the orientation angles of the array and the associated high frequency horns.

3 IMPLEMENTATION

3.1 Circle Centre Channel System

This system comprised of a low-midrange line-array, a high-midrange line-array and a mid-size high frequency horn with a nominal 90° H x 40° V pattern. The same 38mm exit, 100mm diameter voice coil, high frequency compression drivers were used for all six channels.

These two midrange arrays were constructed from the low-midrange/high-midrange horn assembly that is normally employed in the JBL HLA 4895 system. In the HLA system, the low-midrange horn (fed by a 355mm driver) is located directly below the high-midrange horn (fed by a 250 mm driver). Five such horn assemblies were rotated 90° and stacked vertically so that two line arrays of low-midrange and high-midrange horns were formed side-by-side. A total array height of 3.8 m was required to achieve the required constant directivity at low frequencies. At higher-midrange frequencies, the vertical radiation pattern of the HLA system is narrower than its horizontal, and therefore rotating the horn assembly by 90° brings an additional benefit of a further minimisation of the irradiation of the side walls of the theatre.

The whole array was recessed into the proscenium wall (through a large hole) with near flush mounting being achieved against its pleats.

Both arrays used three signal chains (X, Y and Z) each with constant-summation type tapering filters that were set up in the digital signal processor. Discussion of this filter type is given in [7] [8]. The outer two horns (1 and 5) in each array are driven by the X chain signal, horns 2 and 4 by the Y chain signal, and the inner horn (3) in each array by the Z chain signal. At low frequencies, all three chains deliver equal outputs, but as the frequency increases, the signal is progressively transferred from the X and Y chains to the Z chain. The single Z chain driver in each array is raised 6dB in level to match the pairs of X and Y chain drivers. Unlike tapering using low-pass filters, the electrical vector sum of the three chains is always constant and therefore overall equalisation of the array is not required, thus simplifying the design of the crossover filters.

The low-midrange array is crossed over to the high-midrange array at 320Hz. At frequencies above 600Hz, most of the signal is fed to the innermost high-midrange horn and the directivity of the array is controlled by the directivity of that horn rather than by the array. To minimise lobing at the crossover frequency of 1 kHz, the high frequency horn was located immediately beside the innermost horn of the high-midrange array. The vertical directivity of the line arrays was designed to blend with that of the high frequency horn, and the whole system was tilted to optimise its consistency of frequency response over the circle seating area. Both line arrays could have been made more directional, but at the expense of consistent frequency response at all seats.

3.1.1 Performance of the Circle Centre Array

To confirm that our mathematical model was reasonably accurate, JBL Professional made acoustical measurements of the electrically-tapered HLA array at a large outdoor site using ground plane techniques. Unfortunately, for various reasons, the inner low-midrange and high-midrange horns (Z chains) were not raised 6dB in level for the measurements and the spacings between the array elements were not identical to our design.

Fig. 5 shows the measured polar response of the array, while Fig. 6 shows our prediction of its polar pattern in that test configuration. As the measured and predicted responses were reasonably similar, confidence in the mathematical model was gained. Fig. 7 shows the predicted polar responses for the final system with a 6dB boost to the inner horn and the correct inter-element spacings.

Comparison of the responses in Fig. 7 with the published horizontal polar responses of a single HLA element in Fig. 8, shows that the as-installed array has substantially improved directivity below 500 Hz. (Note the published horizontal polars of the HLA become the vertical polars for this array orientation). Fig. 9 shows some published polar responses of the mid-size 90° H x 40° V (JBL-2352) high frequency horn for comparison with the array polars of Fig 7.

3.2 Circle Left and Right Channels

To ensure that most listeners in the dress circle perceived a noticeable image shift between the left, centre and right channels required a sufficiently large horizontal separation between those systems, which in turn placed the left and right systems at the sides of the proscenium wall. As the height of that wall reduced rapidly at its sides and the Opera House management required flush mounting of the left and right speakers for visual acceptability, it was not possible to replicate the large centre-channel array for these speakers. The compromise was to reduce the height of these systems and use a single HLA low-midrange/highmidrange assembly and the same mid size 90° H x 40° V high frequency horn as the centre channel. Naturally, this system could not provide the high directivity of the centre channel array in the mid range and resulted in greater spill to the stalls, a slightly poorer frequency response in the circle and a lower early to late ratio. This compromise between aesthetics and acoustic performance was deemed appropriate, as most soloists would be panned to the centre channel.



Fig. 5 Measured vertical polar response of array of 5*HLA horns (5 dB per division)







Fig. 7 Predicted vertical polar response of array of 5*HLA horns in as-installed configuration (5 dB per division)



Fig. 8 Published vertical polar response of single HLA low + mid-low horn element (5 dB per division)



Fig. 9 Published vertical polar response of JBL 2352

3.3 Stalls Left And Right Channel Systems

The only spaces for the stalls left and right channel loudspeakers that did not require large holes to be made in structural concrete, were 1.5 m high x 500 mm wide slots located on the outer areas beside the stage and above the stalls in an unused loge (a type of private seating box). This space had been provided for the original line arrays (comprising six JBL 2105 125 mm cone drivers), which were installed in 1973, being part of Benson's tapered array designs for the Opera House [11].

Again, the delivery of a constant frequency response over the stalls area with minimal irradiation of the stage and ceiling suggested the use of a small line array of midrange horns whose vertical radiation pattern matched that of an associated high frequency horn. But this location for the system meant that destructive interference of the system's direct field with acoustic reflections and diffractions from the three nearby reflecting surfaces of the loge (side wall, floor and ceiling) could cause significant damage to the frequency response over much of the listening area. These nearby surfaces would also be in the near field of the array.

An important advantage of an array of midrange horns over a similarly sized array of direct radiators is that the horn array will have higher directivity in its near field (so long as the horn directivity is higher than the direct radiator directivity). This higher near-field directivity will produce a corresponding reduction in the strength of image sources forming the reflections and diffractions, thereby reducing degradation of the frequency response in the stalls.

The selected midrange horn has a nominal pattern of 60° H x 50° V and was coupled to a 250mm cone transducer mounted in an aluminium rear enclosure. This combination was selected because of it's suitable dimensions and performance which allowed the formation of a vertical array of three such horns within the physical constraints of the existing opening in the concrete structure. A small format, 90° H x 40° V, high frequency horn coupled to a 38mm exit, compression driver, was chosen for its small size and pattern control. The horns were mounted on a space-frame with the mouths of the midrange horns protruding forward of the concrete face, the neck of the horn passing through the slot in the 150mm thick concrete wall and with the horn throats and drivers being mounted in the void behind the concrete.

By adjusting the inter-horn spacings and the electrical tapering filter, the radiation pattern of the midrange-array was set-up to be similar to that of the high frequency horn. Tapering consisted of attenuating the outer two drivers in the array by 6 dB and feeding them via a passive first-order low-pass filter (inductor with impedance compensation). This "rolling off" caused the overall power response to fall as the frequency increased, necessitating auxiliary equalisation before the active crossover to flatten the frequency response beyond the crossover region.

The restricted physical width of the system dictated locating the high frequency horn directly above the inner midrange horn rather than beside it. To minimise the width of the frequency response nulls in the crossover region between the high frequency and midrange horns, an eighth order crossover was employed between those devices.

The selected midrange horn had a restricted low frequency response, rolling off at 250 Hz, resulting in a spectral gap of approximately one octave between the upper response limit of the subwoofer and the lower response limit of the stalls left/right systems. To overcome this response loss, the left and right stalls signals between 130 Hz and 250 Hz were bandpass filtered and fed to the stalls centre system, which was able to reproduce them. Psycho-acoustically, with full range program material, the image shift below 250 Hz from the left and right systems to the centre system was not particularly noticeable, and this compromise was therefore considered acceptable.

Figs. 10 and 11 respectively show the measured and predicted vertical polar responses of the midrange-array. To illustrate the improvement in directionality provided by the array, Fig. 12 shows the polars of a single midrange horn. Unfortunately, limited project time precluded refinement of the array design after the measurements were made.



Fig. 10 Measured vertical polar response of 3*midrange horns (5 dB per division)



Fig. 11 Predicted vertical polar response of 3*midrange horns (5 dB per division)

3.4 Stalls Centre Channel System

The only location for this loudspeaker was in a recess immediately under the proscenium arch in the area holding the lighting bar, and after much discussion with the Opera House lighting department, a small space between the central two lights was made available. As this location allowed limited dimensions for the components, a midrange array system based on horns could not be implemented.

The low frequency system consisted of three 355 mm direct radiators arranged in a line array that was aimed downward at 42°. To fit the three drivers into the available height, it was necessary to articulate the driver baffles by \pm 20°. Utilising second-order low-pass tapering filters, the array at 550 Hz, produced attenuations of 8 dB directly below it (towards the orchestra pit), and 14 dB at 90° from the baffle in a downstage direction.



Fig. 12 Published vertical polar response of single 3215 midrange horn (5 dB per division)

Two high frequency horns with nominal coverage 80° H x 50° V were located beside this array. As the vertical coverage of a single horn above 4 kHz was insufficient to cover the entire stalls area, the second horn was

introduced at 4 kHz via a high-pass filter. To ensure the smoothest spatial transition between these horns, their radiation patterns were orientated (using EASE) for best power summation and a delay of 5ms was applied to the second horn to narrow the spacing of the inevitable phase cancellations in the response in areas where their patterns overlapped [5].

3.5 Subwoofer System

Central to the design of the six-channel system was the use of a common subwoofer located above the stage on the proscenium wall. The signal fed to this subwoofer was the sum of the six channels (with appropriate gain reductions) and was crossed over at 130 Hz to the six channels.

The subwoofer is a 5m long line-array comprising six JBL 2242H 460 mm drivers in separate enclosures. Its tapering filters are also constant-summation types with the three signal chains (X - drivers 1 and 6, Y - drivers 2and 5 and Z – drivers 3 and 4) implemented in the digital signal processor. Through appropriate choice of tapering filter time constants and the asymmetrical spacing of the woofers, the array is made to radiate minimally towards the ceiling whilst compensating for the different distance-losses in the direct field to all seats more than 2 m forward of the orchestra pit. Fig. 13 shows the predicted polar responses of the array, with the asymmetrical behaviour being clearly evident. Again, limited time precluded measurement confirmation of the array's directivity.

When a subwoofer system is mounted against a wall, the phase cancellation from the image source behind the wall will substantially damage the overall power response of the system [12]. Ideally, the drivers would be located flush with the proscenium wall, but in the Opera Theatre it was not possible to recess the subwoofer enclosures into the wall due to obstructions by structural steel members. The system was therefore designed with the maximum distance between a driver and the wall being 0.3 m, as this distance would move the cancellation to 285 Hz (out of the subwoofer band) and produce an in-band loss of 2.2 dB at 125 Hz. To allow one side of each enclosure to locate flush with the high side of the pleat in the wall together with the other side being flush with the centre channel low-midrange array, a small and shallow enclosure was required. In turn, this necessitated a sixth-order low-frequency alignment to produce the required lower -3 dB response at 40Hz. The combination of the low vented-box resonant frequency and shallow enclosure necessitated a few iterations in the design of the enclosure and vent to achieve a



Fig. 13 Predicted vertical polar response of subwoofer array (5 dB per division)

suitably high value of vented-box loss Q_L without excessive port compression [13].

Another benefit of the tapered line array is that at lower frequencies where vented box systems become limited due to driver-displacement, all drivers in the array are delivering almost equal power. At the higher frequencies the inner drivers have a higher load share, but are no longer excursion limited. Calculations of the maximum direct and reverberant sound pressure levels for both thermal and excursion limiting were made at each frequency using a combination of equations given in [14] and [15], the frequency responses of the tapering filters and the power ratings of the amplifiers.

With the stalls centre and circle left and right systems, there are large path length differences between the subwoofer and those systems to each listener. With any crossover system, those path differences would cause a pattern of peaks and troughs in the crossover region of the direct-field frequency response that is unique at each listener. Noting that the total direct field responses could not be optimised at all seats, and that with ideal loudspeakers, a true Butterworth crossover system of any order delivers constant acoustic power with frequency, fourth order Butterworth crossovers (with low and high-pass filters at 120 Hz) were used with these systems to produce flat power responses in the crossover regions. As the subwoofer array and the low-midrange array of the circle centre speaker were located side by side, in-phase summation of their direct fields in their crossover region occurred at all seats. To overcome the 3dB peak that occurs with in-phase summation and even order Butterworth systems, the -3dB frequency of the low-midrange array in the circle centre system was raised to 160 Hz so that its crossover system was similar to a fourth-order Linkwitz Riley system.

4 INSTALLATION AND COMMISSIONING

4.1 Installation

Significant mechanical design was required to allow the speakers to fit the cavities whilst providing adjustment for the fine-tuning of the speaker orientations.

As the Opera Theatre operates approximately 51 weeks a year, there was insufficient downtime to allow uninterrupted installation of the proscenium-mounted systems. To allow installation to progress overnight whilst rehearsals and performances continued each day and evening, access was obtained using a swinging stage hung from electric chain hoists. To cover the large holes in the proscenium wall during construction, and also the arrays after installation, a 10 m x 6 m grille of acoustically-transparent black fabric was constructed in advance and hung directly in front of the construction area for the circle centre and subwoofer arrays, such that it followed the pleats of the proscenium wall.

The aiming points of all speakers were set up using an inclinometer, protractor and laser pointer to the points derived from the EASE modelling.

4.2 Commissioning

The commissioning process included the following tasks; i) checking the frequency response and acoustic polarity of each driver in each sub-system, ii) confirming that all the speaker cable loss vs. frequency characteristics were as expected, iii) inserting the responses of the active tapering filters into the system processors and designing the active crossover filter parameters, iv) fine tuning the speakers' aiming angles and v) equalising each loudspeaker channel. All frequency responses and polarities were measured with the MLSSA analyser.

The most physically challenging task was a final minor re-alignment of the two stalls centre-channel horns as a result of measurement and listening tests. This occurred after all high-level access equipment had been removed from site. Due to their physical location under the proscenium arch and above the orchestra pit, one of the authors utilised climbing ropes and abseil (rappelling) equipment to gain access.

Crossovers were set up using pseudo-anechoic measurements of the magnitude and phase of the frequency response on axis and at $\pm 10^{\circ}$ vertically for each driver group being crossed over. The crossover types, orders, frequencies, and signal delays were then chosen for best acoustic summation in these three locations. Equalisation was undertaken for each channel using the average of frequency response measurements (direct field plus 50 ms) made at a minimum of 15 seats in each area, followed by listening. Initially, at those frequencies at which a good correlation between the subjective frequency response and the measured average was present, the applied equalisation was the inverse of the average. Refinements and additions to the equalisation were then made by listening to a wide variety of program material whilst moving around the appropriate area.

4.3 The Frequency Response and Radiation Pattern of the Subwoofer Array.

To determine the true frequency response of a large low-frequency array, a pseudo-anechoic environment must be simulated using a ground plane measurement either outdoors or by windowing the impulse response obtained in an extremely large room. As limited project time precluded this type of measurement, the authors attempted to measure the response of the subwoofer array *in situ*, and as expected, difficulties were encountered.

When determining the frequency response from an impulse response that is windowed to duration T, the frequency resolution of the response is 1/T. To achieve a resolution of 1/6 octave at 50 Hz of the direct field only, the impulse response should be approximately 140 ms long [16]. As no theatre venue in the world provides this amount of reflection-reflection-free impulse response, there is an unavoidable trade-off between low frequency resolution and corruption of the loudspeaker's response by reflections. Using larger amounts of time data partially fills in troughs in the measured response caused by major reflecting the overall power output of the speaker.

Only the downward part of subwoofer array's asymmetrical polar pattern could be measured *in situ* and therefore frequency response measurements were made at 10° intervals at seats located on the array's vertical axis at

microphone heights of 1.2 m and 2.2 m above the floor. However, assessing the array's radiation pattern from the frequency responses proved difficult. The responses at both heights showed differences of up to 6 dB at some frequencies, mainly in the form of peaks and troughs. Even larger variations in frequency response resulted when the length of the time window applied to each impulse response was varied.

A time window of 250 ms was ultimately chosen for the impulse response data, and the resulting frequency responses at both microphone heights were averaged for each angle and smoothed over a 1/3 octave bandwidth. The average response over all the angles was also computed. Fig. 14 presents the height-averaged response at each angle and the overall average response, which is offset by -10dB for clarity. The response at 30° is missing as this location was underneath the circle overhang and its local acoustic environment significantly corrupted the response.



vertical angle off axis

Fig. 14 Measured frequency responses of subwoofer array on horizontal axis at various vertical angles. Average response is offset by -10dB.

Some noteworthy points emerge from Fig. 14. As the array's polar response was intended to compensate for the array-to-seat distance loss, the response at each angle should be similar. The levels at 60° , 70° and 80° are mostly lower than the overall average level, showing greater distance compensation than required. The response at 0° shows a markedly reduced level below 60 Hz, which is not evident in the responses at -10° or -20° . At the 0° position, the rear wall was 2.5 m away, close enough for a reflection to cause a primary null at 34 Hz and also affecting frequencies above that. The average frequency response in Fig 14 provides some indication of the overall radiated power, and as that response is relatively flat, strong degradations due to image sources close to the subwoofer array appear to have been minimised.

Both measurements and listening showed that the overall level of bass sound in the front half of the stalls (between 50° and 80° below axis) was 2 to 3 dB lower than the levels in the dress circle (between 20° and on-axis). The array's tapering filters were adjusted to deliver more energy downward to the stalls and aural improvements resulted. The polar responses of Fig. 13 include this change.

The variability in the frequency response measurements with microphone height and impulse response length suggests that there is a need for improved measurement techniques in auditoria at low frequencies to improve the correlation between measured and subjective performance.

5 AURAL PERFORMANCE AND CONCLUSION

The Opera House has informally indicated to the authors that the system has met with excellent acceptance by the operators and hirers of the Theatre. Each of the six channel loudspeakers is sufficiently well balanced that during commissioning, the authors were able to hear changes to the equalisation of only 0.5dB in most third-octave bands over 90% of the Theatre. Even at the very back of the theatre, the sound from the circle centre channel system is extremely dry and intimate with an excellent tonal balance. Although unintentional, the left and right circle (non line array) systems provide a reference for the circle-centre line array system and allow an A/B listening comparison. The sound of the left/right circle systems is very pleasant but lacks the intimacy and clarity of the circle-centre system.

The six-channel loudspeaker system in the Opera Theatre has met the requirements for frequency response, maximum sound pressure levels, directional control, and feedback margin whilst satisfying the restrictions of size and location that were applied to four of the loudspeaker channels. The design approach was to primarily use custom-made tapered and steered line-array loudspeakers comprising horns and direct radiators. The vertical radiation pattern of each array was designed to either match that of the associated high frequency horn or to compensate for distance loss of sound level. Strong attention was also paid to the minimisation of the strength of each loudspeaker's local image and diffraction sources and the crossover types and turnover frequencies.

REFERENCES

- [1] C. Heil, J. Urban, "Sound Fields Radiated by Multiple Sound Source Arrays", Presented at the 92nd Convention of the Audio Engineering Society, March 1992 Preprint No 3269
- [2] JBL White paper "Achieving Optimum Line Array Performance Through Predictive Analysis, Unique Acoustic Elements and a New Loudspeaker System".
- [3] "Sydney Opera House Web Site", <<u>http://www.soh.nsw.gov.au</u>>, September 2002
- [4] D. Davis & C. Davis, *Sound System Engineering,* (Sams, Indianapolis, 1987). pp 238-242, pp 262-268
- [5] M. S. Ureda, "Directivity Responses of Horn Arrays" Presented at the 98th Convention of the Audio Engineering Society, February 1995 Preprint 3963
- [6] A. Mochimaru, "An Advanced Concept for Loudspeaker Array Design in Sound Reinforcement Systems" Presented at the 91st Convention of the Audio Engineering Society, October 1991 Preprint 3139
- [7] D.G. Meyer, "Digital Control of Array Directivity", Presented at the 74th Convention of the Audio Engineering Society. Preprint 2014 Convention Oct 83.
- [8] G. Leembruggen, D. Connor, "Design and Commissioning of Sound Reinforcement Systems for the Australian Parliament House A Holistic Approach" *J. Audio Eng. Soc.*, vol. 44, No10, 1996 October
- [9] G. Leembruggen, "The Design of Three Unusual Loudspeakers for the High Court of Australia", *J. Audio Eng. Soc.*, vol. 48 No5 2000 May
- [10] M. S. Ureda, "The Convolution Method for Horn Array Directivity Prediction" Presented at the 96th Convention of the Audio Engineering Society, February 1994 Preprint 3790
- [11] J.E. Benson, D.F.Craig, Australian Patent 58981/78
- [12] G. Adams, "Time Dependence of Loudspeaker Power Output in Small Rooms" *J. Audio Eng. Soc.*, vol. 37, No 4, 1989 April

[13] A. Salvatti, A. Devantier, D. Button, "Maximizing Performance from Loudspeaker Ports", *J. Audio Eng. Soc.*, vol. 50, No1/2 2002 January/February

[14] R. H. Small, "Vented-Box Loudspeaker Systems Part II: Large-Signal Analysis" *J. Audio Eng. Soc.*, vol. 21, No 6, 1973 July/August*

- [15] M. Engebretson, "Low Frequency sound Reproduction" J. Audio Eng. Soc., vol. 32, No 5 1984 May
- [16] F. Seidel, H. Staffeldt, "Frequency and Angular Resolution for Measuring, Presenting, and Predicting Loudspeaker Polar Data", *J. Audio Eng. Soc.*, vol. 44, No7/8 1996 July/August