Design of Three Unusual Loudspeakers for the High Court of Australia*

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Three unusual loudspeaker systems providing specific radiation patterns were designed as part of the overall design of a sound reinforcement system for the High Court of Australia. The first loudspeaker was a distributed array of dipole loudspeakers, each housed in a trough; the second was an acoustically curved line array utilizing 34 drivers; and the third was a short tapered line array utilizing six drivers and a tweeter. Primary requirements for each loudspeaker were its architectural shape and finish, radiation pattern, and frequency response. The integration of these requirements necessitated a holistic approach to the design of each loudspeaker. The design process, the hardware implementation, and the performance of each loudspeaker are discussed.

0 INTRODUCTION

A sound reinforcement system has been recently designed and installed into courtroom 1 of the High Court of Australia, the most senior court in the country. As one of the major design constraints was that the loudspeakers disappear visually, they were required to have architectural elements of shape and finish similar to other structures in the courtroom.

This paper describes three of the system loudspeaker types used in the courtroom: a dipole mounted in a trough with reflector plates, an acoustically curved long line array, and a short electrically tapered line array. All loudspeakers employ 90-mm cone drivers. Each loudspeaker design integrated the requirements of specific and unusual radiation patterns, frequency response, and architectural aspects. An architect assisted in the architectural design of the loudspeakers, engaging with the author in an interactive process of refinement between electroacoustic and aesthetic aspects.

As only speech reinforcement was required, it was tempting to restrict the system bandwidth to 300 Hz to 5 kHz and thus simplify design and minimize loudspeaker enclosure size. But for high speech reproduction fidelity, a bandwidth of 80 Hz to 12 kHz is required [1], as long as the frequencies critical for intelligibility [2] are assigned prime importance in terms of coverage and early-to-late ratio. To ease the driver requirements and enclosure sizes, the response limits were relaxed so that the required system frequency responses were within ±3 dB from 100 Hz to 10 kHz (after equalization).

For two of the loudspeakers discussed in this paper, the desired radiation pattern reflected issues of coverage and acoustic gain. The maximum required equivalent acoustic distance (EAD) [2] was 2.5 m, with a gain margin of 6 dB and a talker-to-microphone distance of 0.7 m. (EAD is the apparent talker-to-listener distance with the sound system operating.) The maximum required sound pressure level at the edge of the coverage zone of each loudspeaker was 84 dB, corresponding to the peak level of a loud voice at the maximum EAD (based on an L_{eq} of 70 dB and a crest factor of 14 dB).

1 ROOM AND SYSTEM

The courtroom is 20 m wide, 21 m long, and 18 m high, with a volume of 7560 m^3. Its reverberation times range from 1.0 to 1.3 s over the range of 125 Hz to 4 kHz and are highest in the 2- and 4-kHz octave bands. Subjectively, the room sounds "live." More important than the reverberation times were late arriving specular reflections from the flat, high ceiling, as these could severely degrade speech intelligibility.

Microphones were provided for the seven judges who sit at a long beach and a single barrister who stands at a lectern. Sound reinforcement was required for five areas: the judges bench, the large barristers' area in front

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of the judges' bench, the tipstaves area behind the judges (tipstaves are judges' assistants), and the upper and lower public galleries.

As architectural requirements disallowed the use of a centralized hanging loudspeaker to provide coverage to the entire courtroom, a distributed system of loudspeakers with specific radiation patterns was employed to cover each area. The array of dipole loudspeakers covered the tipstaves, while existing bench-mounted loudspeakers were improved for the judges. Four short line array loudspeakers mounted on the front face of the justice bench covered the barrister area. The lower public gallery was served by loudspeakers mounted in the backs of the seats, along with ceiling loudspeakers mounted in existing (but nonideal) positions. Loudspeakers mounted on the balcony handrailing and the curved long line array covered the upper public gallery. Fig. 1 shows the layout of the courtroom, including the upper and lower galleries and the location of loudspeakers and microphones.

2 SYSTEM MODELING

Predictions of the radiation patterns of each loudspeaker were made using a computational model that included the contributions of the radiation patterns of individual drivers. The model follows that of Meyer [3, eq. (1)] and functions as follows:

Look-up tables held the driver's amplitude and phase responses at 15° polar increments and one-sixth-octave intervals normalized to the on-axis response. Using direction cosines, the angle and distance of a listener point relative to each driver's axis were calculated and quantized to 15°. At this point the total vector of each driver's acoustic output was computed from the sum of the following factors: amplitude loss due to distance, phase shift due to flight time, and electrical signal delay, and the complex frequency responses of both the driver (at the relevant 15° angle) and the electric filter feeding the driver. The total driver vectors were then summed to give the system response at this frequency and point, and the process was repeated at one-sixth-octave intervals and appropriate spatial points.

The chosen driver for all loudspeakers was the 90-mm Vifa S09FD-02-08 with a sensitivity (1 W/1 m) of 89 dB and a nominal thermal power rating of 5 W. Fig. 2 shows the measured on- and off-axis frequency response of the driver when mounted in a half-space environment.

3 DIPOLE ARRAY

3.1 Required Radiation Pattern

Each tipstaff sits behind his or her associated judge, approximately 0.7 m away from the front wall of the court. As the quantity and location of their book trolleys vary with each court hearing, the seating position of each tipstaff is not fixed. As the only available positions for tipstaff loudspeakers were on the front wall behind the tipstaves, those loudspeakers would face directly toward open judges' microphones, which were located approximately 3.5 m from the front wall. To provide adequate coverage of the tipstaves while maintaining the required acoustic gain margin, sound was required to be distributed left and right along the front wall with minimal forward radiation toward the microphones. This distribution pattern had some similarity to the figure-of-eight radiation pattern produced by a doublet source [4] and suggested that an array of dipole loudspeakers distributed along the front wall (with the driver axes parallel to the wall) might produce less forward radiation than an array of distributed monopole loudspeakers.

Over the 11.5-m-long tipstaves area, loudspeakers could only be located at intervals of 1.25 m, corresponding to architectural recesses on the front wall. The combination of loudspeaker spacing and required coverage area would place both justice microphones and tipstaff listeners in the near field of this loudspeaker array. In this near-field situation, destructive interference (resulting from large inequalities in the path lengths between sources and receivers) would cause strong variations in the frequency responses and level distribution with either dipole or monopole sources. In the far field of a distributed array of loudspeakers, where the distances from a receiver to each monopole are essentially equal, a monopole array sums coherently. In contrast, the far-field sum of a distributed array of dipole loudspeakers is zero. The near-field characteristics of both dipole and monopole arrays were therefore modeled (discussed in Section 3.3) to determine which array gave the highest attenuation at the microphones relative to the average level at the tipstaves.

3.2 Dipole Behavior

Fig. 3 shows the theoretical response of an ideal dipole with a spacing of 100 mm between the two drivers, considered as nondirectional sources. At angles away from the normal and at frequencies where the separation of the drivers is less than one-third of a wavelength, the dipole's frequency response rises at 6 dB per octave. Where the driver separation is greater than 0.45 wavelength, the dipole reaches "cutoff," and the response begins to exhibit comb filtering. In practice, narrowing the radiation pattern of each driver at higher frequencies would help ameliorate this comb filtering beyond cutoff.

Equalization can flatten the frequency response up to cutoff, but the allowable boost is restricted by driver thermal and excursion limits. Spacing between the drivers is therefore a compromise between maximizing the cutoff frequency and minimizing low-frequency boost.

3.3 Modeling of Near-Field Sound Distribution

Using the method of Section 2, but with omnidirectional drivers, the near-field sound pressure distribution with frequency was modeled for monopole, standard dipole, and alternate-polarity dipole systems. All three systems consisted of eleven monopole or dipole sources, spaced at intervals of 1.25 m.

The system comprising eleven standard dipoles is characterized by a significant acoustic cancellation at low frequencies in the regions between adjacent dipoles, due to the opposite polarity of adjacent acoustic wavefronts. An alternating-polarity scheme (in which the polarity of a given dipole driver and its nearest neighbor in an adjacent dipole are identical) would reduce this cancellation, and was therefore investigated.

The goal of the modeling was to compare at each frequency the perceived average sound level in the tipstaves area with feedback-related levels reaching the microphones. In the tipstaves area the average level and frequency response determine the overall perceived loudness and tonal balance. In contrast, both the maximum and the average levels reaching the microphone area determine the acoustic gain margin of the sound system. While narrow-band equalization can reduce some peaks in the open-loop frequency response at a given microphone without greatly affecting the sound at the tipstaves, equalization cannot reduce the average broad-band loop gain significantly [1].

For the tipstaves, frequency responses were calculated at locations 0.7 m forward of the loudspeaker array at lateral intervals of 0.1 m over the 11.5-m length and then averaged. In the microphone area, responses were calculated at 0.1-m intervals over an 8-m length, at a distance 3.5 m forward of the array and the maximum and average levels found at each frequency. Responses

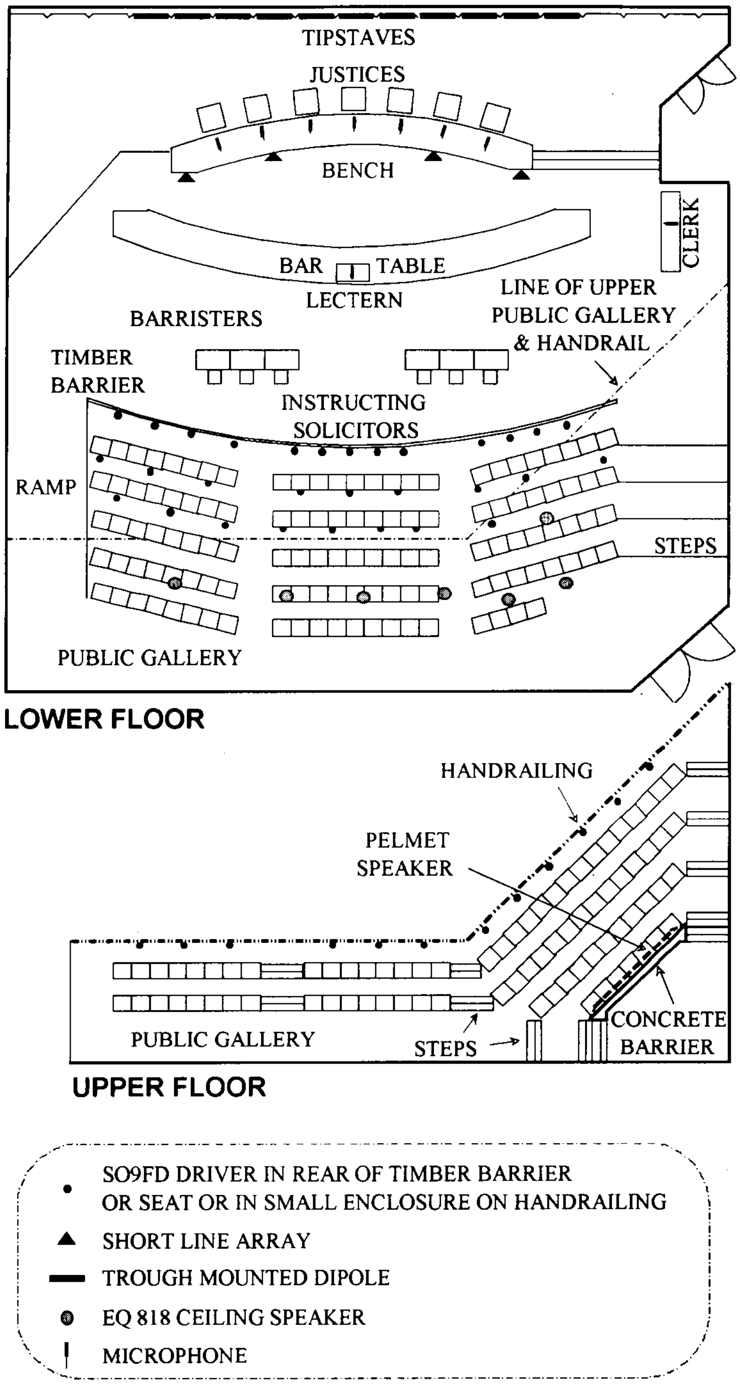

Fig. 1. Layout of room with locations of loudspeakers and microphones.

were *only* calculated up to 2 kHz, as this frequency was deemed to be the upper limit in which modeling would reflect practice. Above this frequency, diffraction effects would likely cause large differences between practical and predicted results.

3.4 Averaging over Tipstaves Area

The mathematical averaging of the responses over the tipstaves area is similar to the first step in equalizing the average acoustic frequency response to be as flat as possible [1] over that area. With any system whose average frequency response exhibits large dips due to phase cancellations, equalizing those dips to flatten the average response will produce large and undesirable boosts in the responses at some locations and should therefore be avoided. In addition, narrow dips in the response affect the perceived level and tonal balance significantly less than peaks.

As expected, with all of the systems modeled, there were tipstaff locations at which the response exhibited large dips due to phase cancellations, and other areas directly forward of a dipole where the overall response was insufficient to be usable, due to the doublet null at 90°. These areas of insufficient level were not expected to be problematic as modeling showed that they were narrower than 200 mm, and the tipstaves could therefore be expected to move slightly in order to receive useful sound levels.

The response dips and low levels caused the average response to be up to 3 dB lower at some frequencies than a perceptually based average response that would form a suitable basis for equalization. Simple mathematical averaging of the tipstaff responses therefore gives a false indication of the practical difference between the feedback-related levels at the microphones and the perceived levels at the tipstaves.

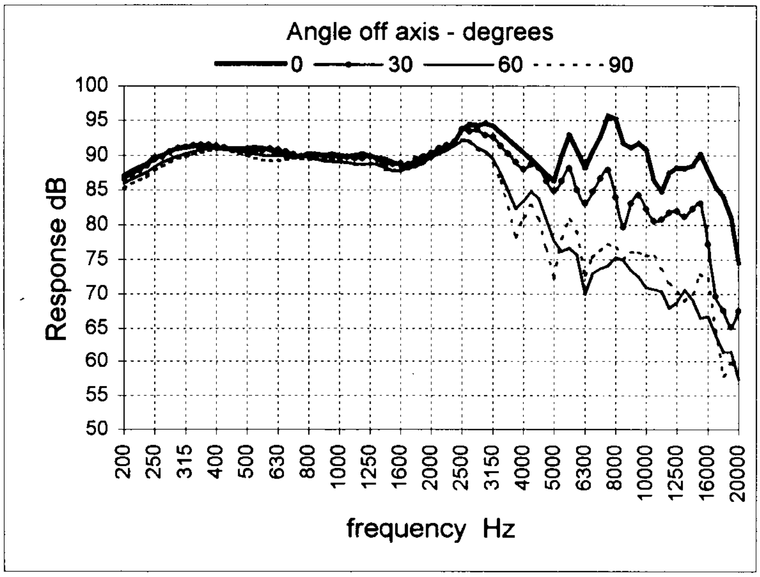

Fig. 2. Frequency responses on and off axis of Vifa S09FD driver at 1 W/1 m.

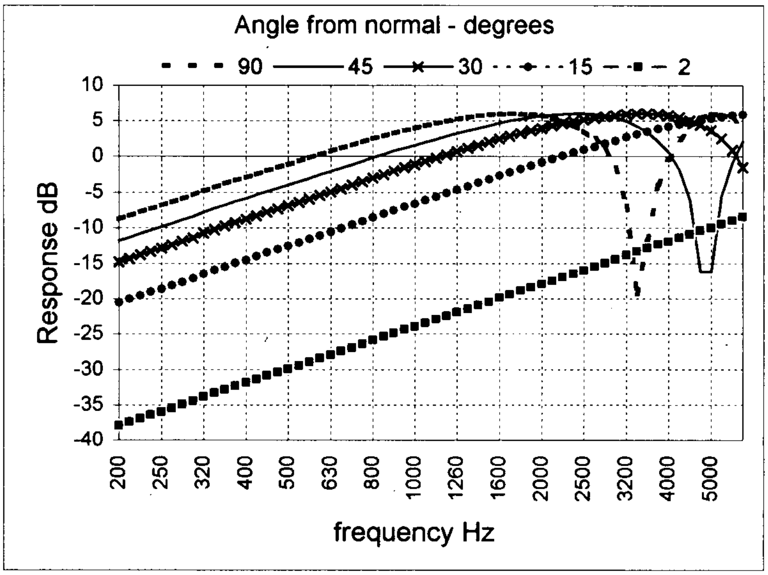

Fig. 3. Frequency response of ideal dipole with 100-mm spacing between drivers.

To form this perceptually based average response, low responses were removed at specific frequencies from the averaging process. If at a given frequency the level at any position was 12 dB below the maximum level present in the entire 11.5-m length, that position was removed from the group at that frequency.

3.5 Results

Figs. 4 and 5 show the relative differences at each frequency between the levels in the tipstaves and microphone areas produced by the three systems. Fig. 4 shows the differences between the tipstaves' adjusted average level and the maximum microphone level, whereas Fig. 5 shows the differences between the tipstaves' adjusted average and the true microhone average.

For a given adjusted-average level at each frequency at the tipstaves, the standard dipole system offers the overall least spill into microphones, followed by the alternate-polarity dipole system. The monopole system has the poorest overall attenuation with distance, reflecting its far-field trend. Although the differences between the systems are in general small, they should be viewed in the light of improvements to the EAD of the total sound system. A reduction of 3 dB in the level reaching the microphones would improve the system EAD by 30%. In the context of the High Court environment in which judges speak softly and a long way from their microphones, improvements in the EAD of this order are worthwhile.

Fig. 6 shows the adjusted-average frequency response of the three systems with the 0-dB level corresponding to the level that a single monopole driver would produce at 1 m. In this simulation, all drivers are connected in parallel, resulting in the dipole systems consuming twice

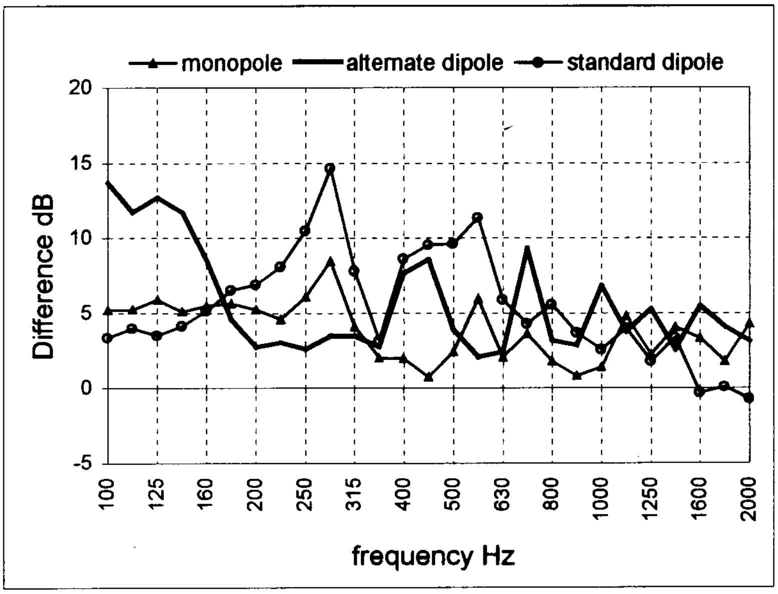

Fig. 4. Difference at each frequency between adjusted average level in tipstaves area and maximum level in microphone area.

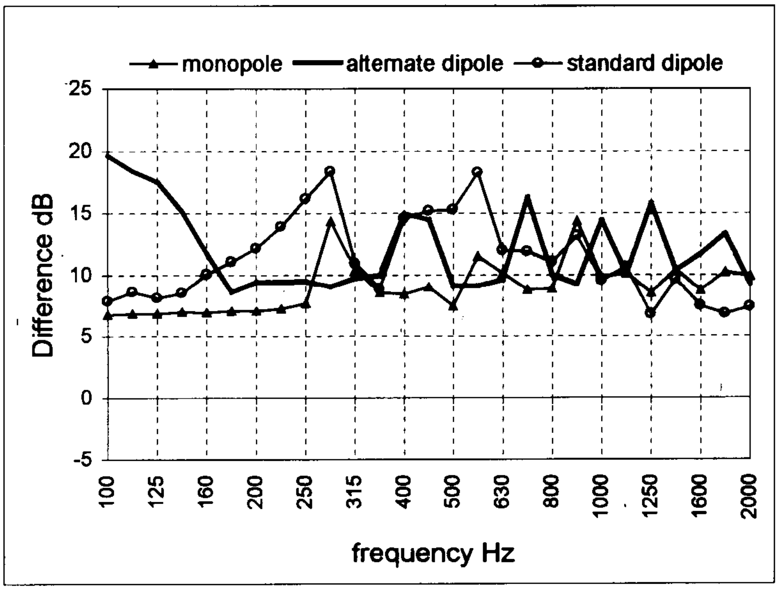

Fig. 5. Difference at each frequency between adjusted average level in tipstaves area and true average level in microphone area.

the electric power as the monopole system. As the response of the monopole system is relatively flat, only modest equalization is required to flatten its average response. In contrast, the standard dipole shows the combined effects of the low-frequency cancellation between adjacent dipoles and the classical dipole response of Fig. 3. A boost of 25–30 dB would be required to flatten its response below 200 Hz. The alternate dipole response is substantially flatter than the response of the standard dipole, requiring only 10–15 dB of low-frequency equalization boost. The ability of the 90-mm drivers to accept a 25-dB low-frequency boost without overload and the difficulty in producing that boost meant that the standard dipole system was discarded, despite its lowest overall spill into the microphone area.

3.6 Dipole and System Implementation

As high sound pressure levels were not required from these loudspeakers, equalization boosts of 10 dB at 150 Hz could be applied to these drivers without exceeding their thermal or excursion limits. The drivers in each dipole shared a closed common rear chamber with a volume of 1.5 L, but as they were connected with opposite polarity, the normal reduction in total system compliance [5] did not occur. The motional impedance of the pair was similar to that of two drivers in free air connected in parallel.

The preferred structure of each dipole was to use two drivers mounted in the ends of a 100-mm-long brass tube, fixed to the front wall. Aesthetics, however, dictated that the dipoles were to be recessed into existing 120-mm-high cavities located at seated ear height above bookcases on the front wall. The dipoles were therefore housed within a trough of rectangular cross section, and reflector plates (curved in the horizontal plane) were used to direct the sound out of the trough. The structure, shown in Fig. 7, was made from 1.5-mm-thick steel and painted with thick resonance damping paint.

Compared to the ideal arrangement with tube-mounted drivers, the trough and reflector plates caused substantial disturbance to the dipole radiation pattern and frequency response. As the generation of a mathematical model describing this system was too complex for the scope of the project, a trial-and-error process was used to optimize the shape of the reflectors, balancing adequate frequency response at each angle with smooth and suitable changes in directivity. Measurements of the frequency response of candidate reflector shapes were made at 15° increments in the horizontal plane using a MLSSA fast Fourier transform (FFT) analyzer, and the most suitable shape was chosen. The spatial frequency response of the trough-mounted dipole without reflector plates is shown in Fig. 8, and Fig. 9 shows the responses with the final reflector shape.

The outer dipole systems were located within 3 m of acoustically hard reflecting surfaces. To prevent specular reflections at higher frequencies from causing disconcering image shifts, a 3-kHz, 6-dB per octave low-pass filter (inductor) was wired in series with the outer driver in these two systems. For correct dipole operation, the outputs of both drivers must be identical. Thus the inner driver in these outer systems was fed through a resistor

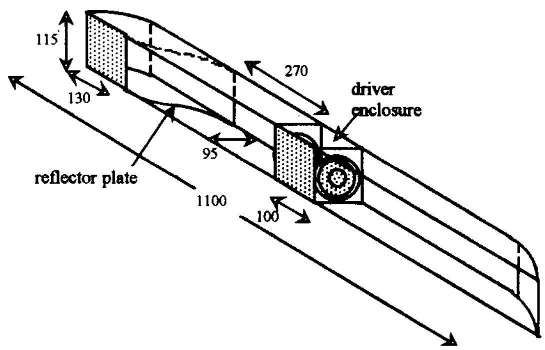

Fig. 7. Structure of dipole in trough. For clarity, reflector plate at one end is not shown.

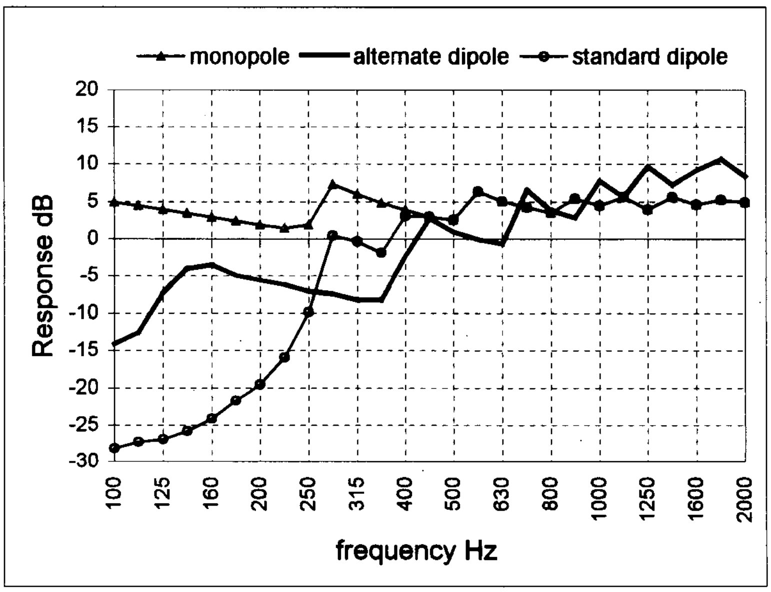

Fig. 6. Adjusted average frequency response of three systems in the tipstaves area.

equal to the inductor resistance. Modeling indicated that the phase shift caused by the filter did not significantly upset the dipole behavior below its cutoff frequency.

4 CURVED LONG LINE ARRAY

4.1 Requirements and Investigations

In the upper gallery area the primary source of coverage was 11 S09FD drivers mounted in equally spaced individual brass enclosures, which were fixed to the front handrailing of the gallery, facing the listeners. As each loudspeaker was within 1.2 m of its nearest listener, the inevitable tradeoff between distance loss of sound level and consistency of coverage allowed the coverage of only two seating rows. On one side of the gallery, a third row of eleven seats and a fourth row of seven seats required coverage from additional loudspeakers.

The only architecturally feasible location for additional loudspeakers was on a low, free-standing, 3.9-m-long concrete wall located immediately behind the fourth row of seats. Drivers would be housed in a pelmet fixed along the length of the wall and located 0.35 m away (above and behind) the heads of the fourth-row listeners. In the third row, listeners would be 1.2 m away, with two seats lying beyond each end of the wall.

The design process commenced with modeling the spatial frequency responses of a system using a driver mounted behind each fourth-row seat. Fig. 10 gives the predicted responses along one-half of the third row of seats. Response variations that would cause poor tonal quality and a significant reduction in intelligibility are evident. The predicted 8-dB variation in the direct-field levels between rows 3 and 4 was also unacceptable.

Increasing the number of drivers on the wall and decreasing the interdriver spacing offered two potential advantages. The first was that the system would begin to function as a true line source up to a certain frequency, producing a loss of 3 dB per doubling of distance, which

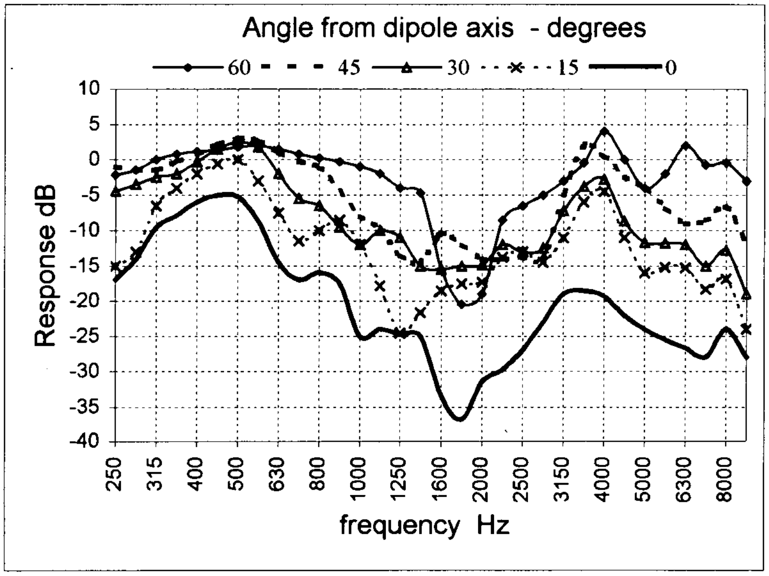

Fig. 8. Frequency responses of dipole in trough without reflector plates.

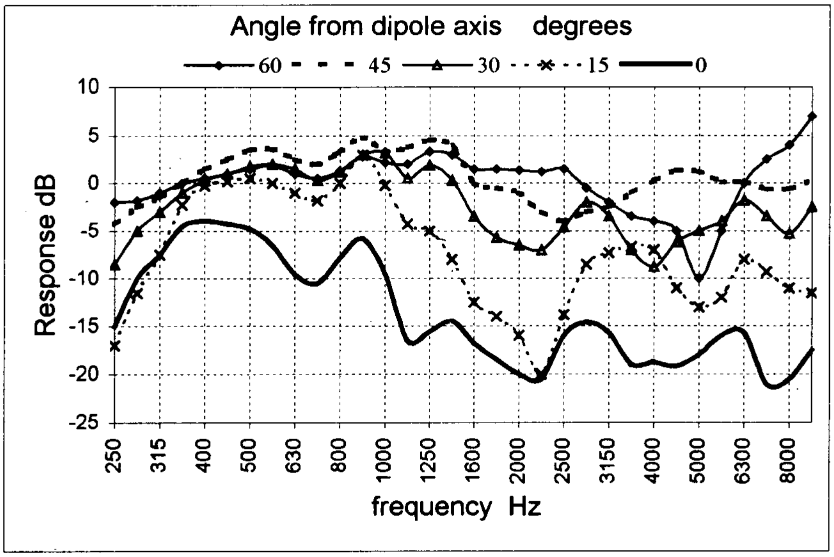

Fig. 9. Frequency responses of dipole in trough with reflector plates.

would improve the consistency of coverage over the two rows. The second advantage was that the frequency at which the consistency of response broke down would be raised. The predicted response along the third row of an array of 20 omnidirectional drivers is shown in Fig. 11. Improvements were evident, but the response was insufficiently smooth up to 4 kHz, and hence inadequate for good speech intelligibility and tonal naturalness.

4.2 Final Array Implementation

It was proposed to decrease the interdriver spacing to its minimum and to fill the entire length of the wall with a straight, horizontal array of 34 S09FD drivers. The drivers would be tilted downward at an angle of 45°, thus ensuring that the two rows of listeners would be located in the drivers' vertical radiation beamwidth. The available enclosure volume of 1 L per driver would give adequate low-frequency response.

The predicted response along the third row of the 34-driver array (now with driver directionality included) is given in Fig. 12. It shows a consistent and smooth frequency response up to 3 kHz, above which the response is inconsistent and ragged. However, the horizontal beamwidth of the line array was too narrow to give adequate coverage of the outer two seats at each end of the third row.

A curved line array was found to widen the horizontal coverage significantly. With an array curvature equivalent to the 170-mm offset between the inner and outer drivers (the maximum allowed by the architect), the coverage of the outer seats was improved but was still insufficient. By progressively delaying the signal fed to the outer four drivers at each end of the curved array, the "acoustic" curvature was increased, resulting in improved coverage at the outer seats between 630 Hz and 3 kHz. Cascaded sections of passive, second-order, all-pass filters with maximally flat delay were used, with a

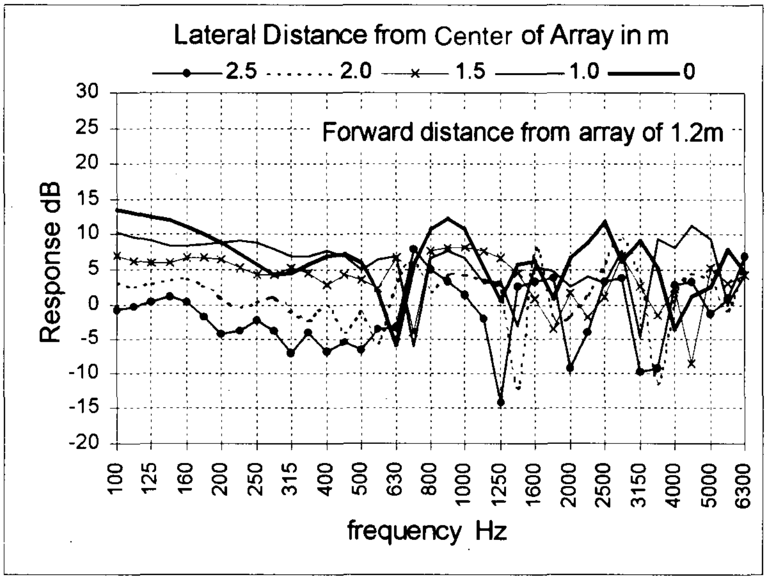

Fig. 10. Response across one-half of third seating row with array of 7 drivers on wall.

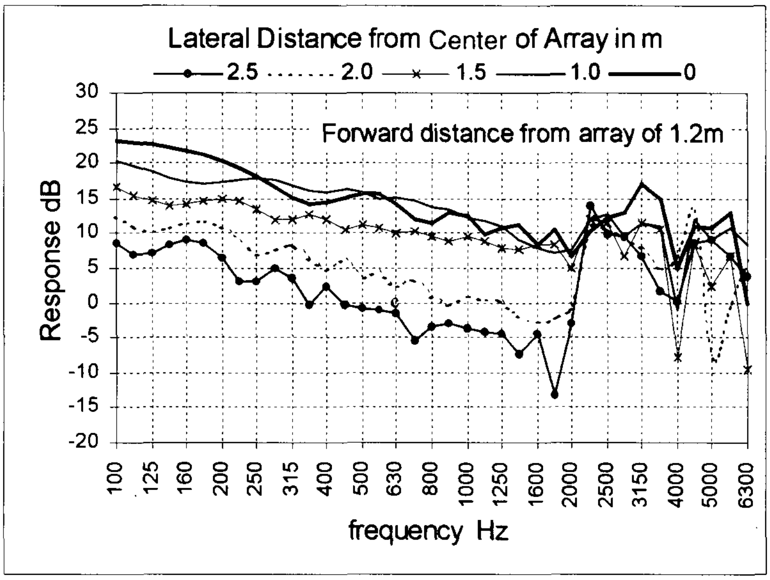

Fig. 11. Response along one-half of third seating row with array of 20 drivers on wall.

total electrical delay of 0.35 ms being applied to the outermost drivers. The combination of electrical delay and physical offset gave these drivers an apparent offset of 290 mm relative to the innermost drivers. The predicted response across one-half of rows 3 and 4 of the final arrangement (known as pelmet loudspeakers) is shown in Figs. 13 and 14. The responses fall at 3.3 dB per octave up to 2.5 kHz, at which the consistent array behavior breaks down.

One disadvantage of the line array was that its impulse response was dispersive at all listeners [6], showing a "wake" behind it. However, the benefits of consistent coverage for intelligibility far outweighed degradation due to waveform "smearing."

4.3 System Implementation

The need for equalization to restore tonal balance is apparent in the responses shown in Figs. 13 and 14. As the building structure prevented the installation of a separate loudspeaker cable for the pelmet loudspeakers, this system was connected in parallel with the aforementioned railing loudspeakers. It was therefore proposed to set up the active equalizer serving the upper gallery for optimum response in the seats covered by the railing loudspeakers (with responses similar to those shown in Fig. 2) and to use an additional passive filter to equalize the pelmet system.

With the railing loudspeaker system equalized and both the railing and the pelmet systems operational, the frequency response over rows 3 and 4 was measured using 50 ms of impulse response data, smoothed to one-sixth octave and averaged [1]. In the averaging process the broad-band level difference between rows due to relative distance loss was removed, allowing the average to be based only on tonal variations. Fig. 15 shows the average responses in the third and fourth rows (prior to distance loss adjustment), along with the overall average. A slope of 3.5 dB per octave is evident in the overall shape.

A passive network was designed to have a response equal to the inverse of the general shape of the average response between 200 Hz and 5 kHz. Because a passive equalizer cannot produce boost directly without lowering its input impedance, boost must be accomplished by bypassing attenuation resistors. As the average level at 5 kHz from the line array in rows 3 and 4 had fallen to that of the railing loudspeakers in rows 1 and 2, attenuation could only be bypassed up to 5 kHz. To produce the desired response of the passive filter, it was necessary to equalize both the array's inductive and its motional impedance. In addition, to present a flat frequency response at the equalizer input with the 2-Ω source impedance of the 100-V distribution system, the input impedance of the total array system was also equalized flat. This flat input impedance also allowed adjustment of the overall array level without changing its response, by the insertion of resistance pads. Figs. 16 and 17 show the array schematic and the circuit diagram of the equalizers and all-pass filters.

The aural results were very satisfactory, with the coverage, voice fidelity, and sound level being highly consistent over the bulk of the two rows. The aural transition between the coverage areas of the railing and pelmet loudspeakers was smooth and not particularly noticeable. While the coverage and voice fidelity at the outer seats of the third row were not of the standard of the inner seats, the speeach was tonally satisfactory and the intelligibility was high.

5 SHORT TAPERED LINE ARRAY

5.1 Requirements

This loudspeaker system has the largest coverage area of all systems in the High Court and also provides source localization cues in the upper and lower public galleries for the justice and barrister microphones. The only available location for the loudspeakers was on the front of the judges bench, with the top of the loudspeakers positioned 1.1 m above the floor. The width and depth of the louspeakers needed to be as small as possible.

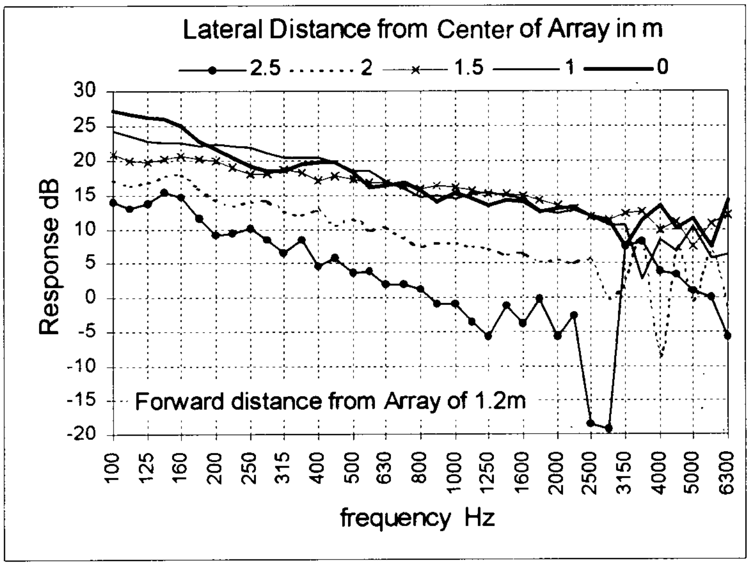

Fig. 12. Response along one-half of third seating row with flat array of 34 drivers on wall.

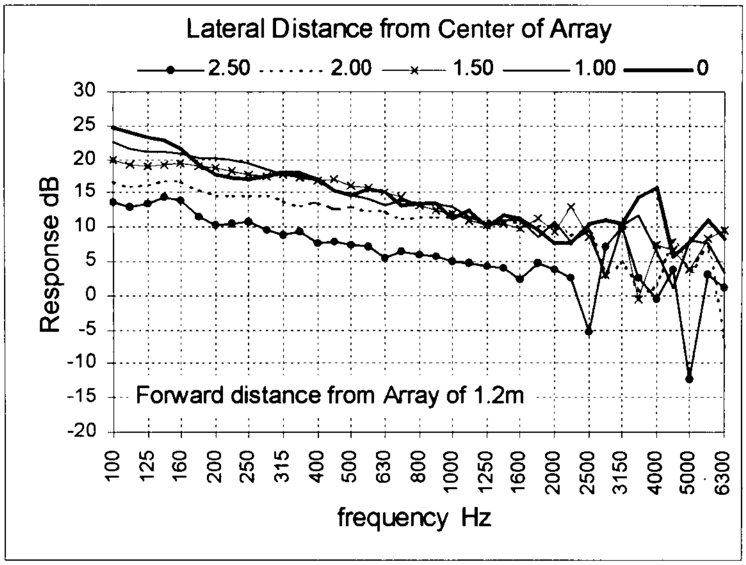

Fig. 13. Response along one-half of third seating row with acoustically curved array of 34 drivers on wall.

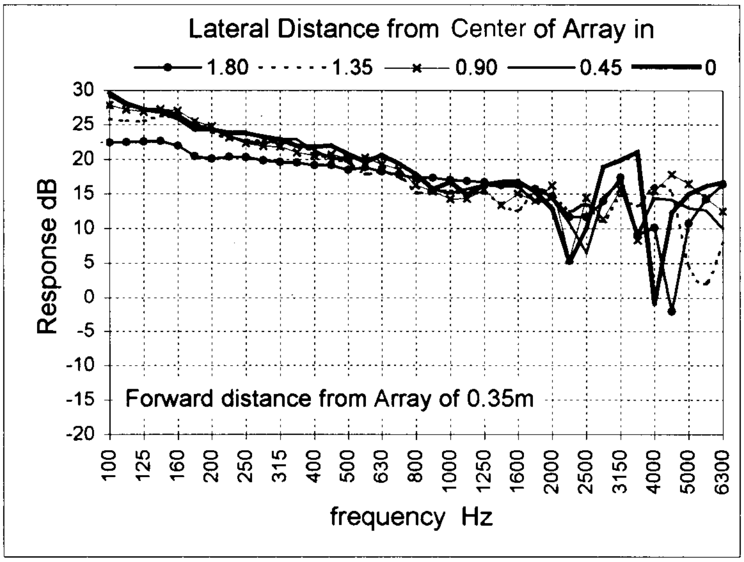

Fig. 14. Response along one-half of fourth seating row with acoustically curved array of 34 drivers.

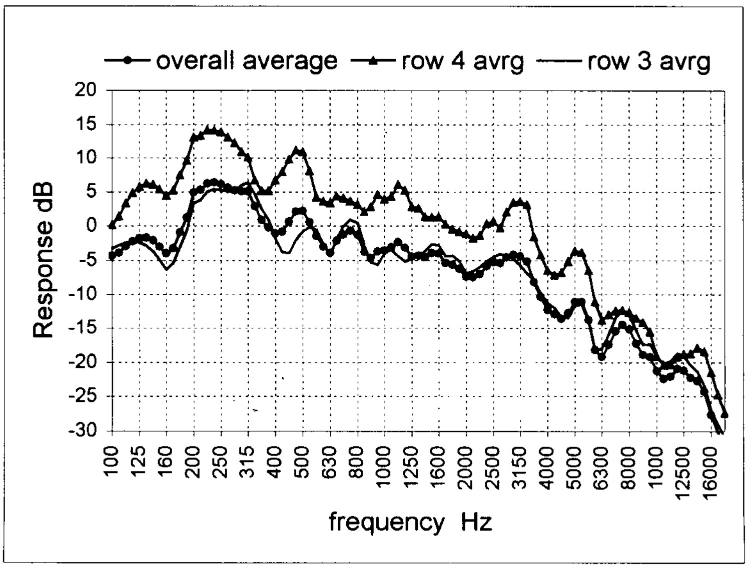

Fig. 15. Average of measured frequency responses of pelmet loudspeaker.

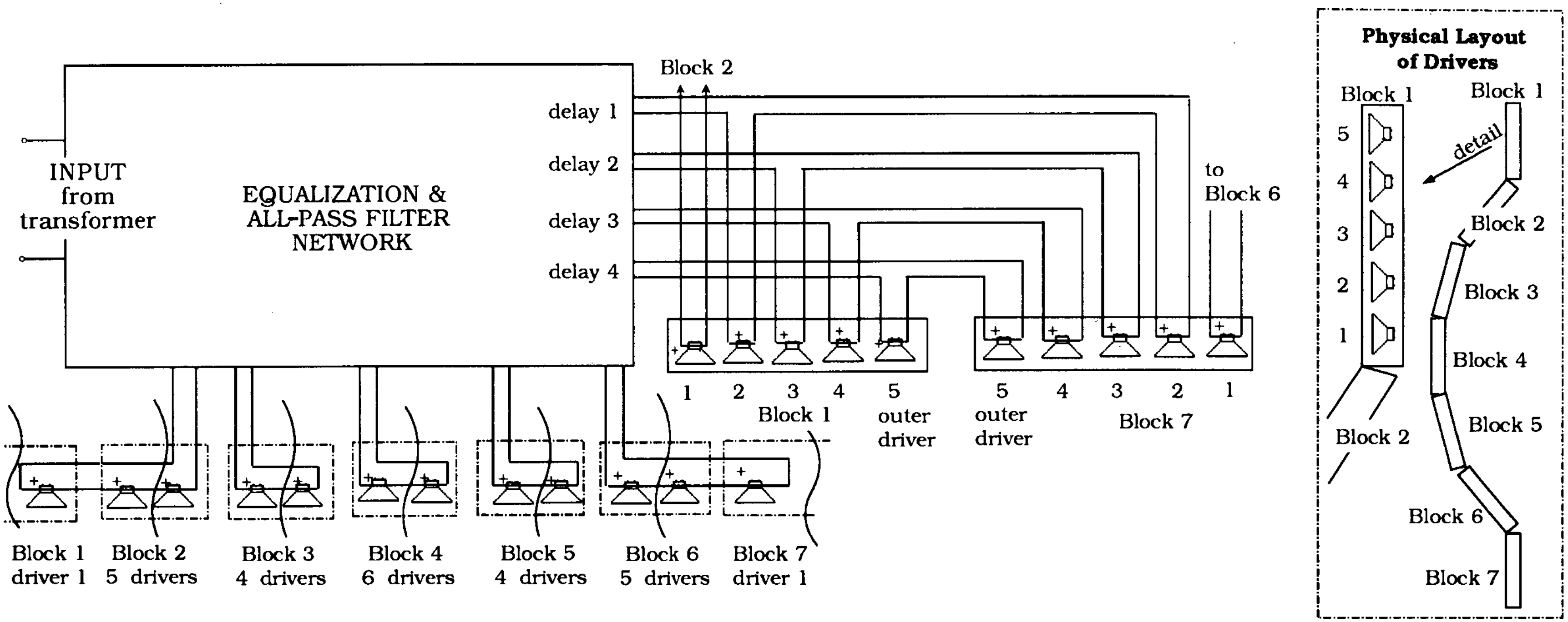

Fig. 16. Schematic diagram of pelmet array.

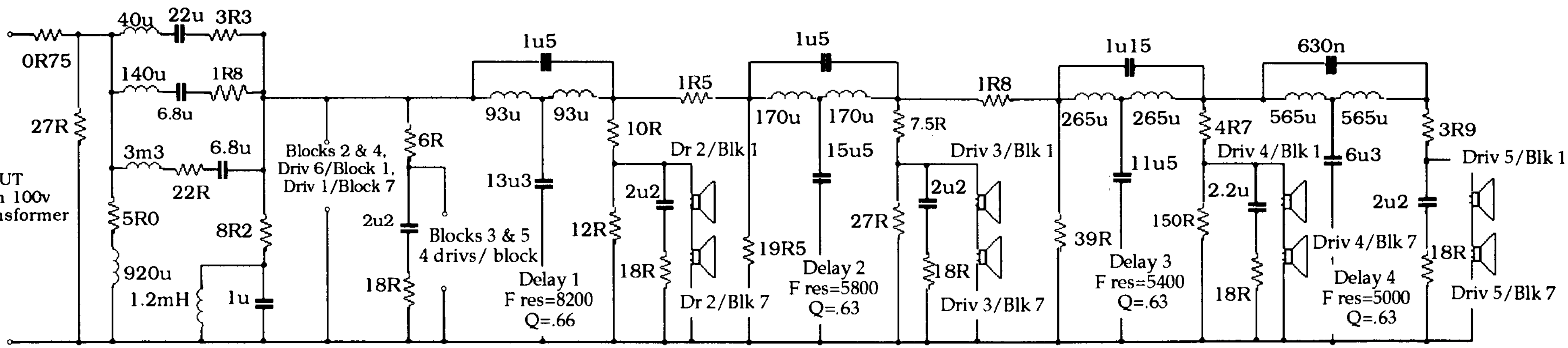

Fig. 17. Circuit diagram of equalizer and all-pass filters for pelmet loudspeaker.

The vertical radiation pattern of the loudspeakers was to be as constant with frequency as possible and as narrow as possible for a number of reasons. To produce the required acoustic gain, the amount of sound diffracting over the top edge of the bench and reaching a justice microphone (located 500 mm above and 300 mm behind each loudspeaker) needed to be minimized. Intelligibility and voice fidelity standards required that 1) the early decay times of the room not be increased by upwardly directed sound, 2) the level of specular, late-arriving reflections off the high flat ceiling be minimized, and 3) the early-to-late ratio be maximum and constant with frequency. A vertical -6-dB beamwidth of 20° would ensure that the barrister and lower gallery areas were within the radiation pattern of the loudspeakers.

5.2 Implementation

The requirements were fulfilled by a short electrically tapered line-array system using six S09FD drivers and passive tapering filters. Frequencies above 2.5 kHz (where the directional response of the array was ragged) were handled by a Vifa H25TG-35-06 dome tweeter coupled to a short horn, which provided reasonable directivity at high frequencies.

The passive electrical tapering filters produced three signal chains: low, mid, and high [1]. At low frequencies all three chains delivered equal outputs, but as the frequency increased, the signal was progressively transferred from the low and mid chains to the high chain. As the vector sum of the three chains was always constant, overall equalization of the array was not required, simplifying the design of the crossover filters.

Nonstandard inter-driver spacings precluded the use of standard array arrangements such as [7]. Modeling was therefore used to choose the time constants of the tapering filters and the positions of the S09FD drivers and tweeter to optimize the total radiation pattern. As a passive filter was to be used for the crossover between the array and the tweeter (disallowing the use of very high slope filters, see [8]), a minimum distance between the high-chain drivers and the tweeter was vital to minimize nulls in the off-axis response. The final arrangement is shown in Fig. 18. Figs. 19 and 20 show the predicted frequency responses of the woofer array at a number of points located on a 5-m radius above and below the central point of the array. The asymmetrical driver arrangement produced substantial directivity above the array up to 2 kHz, at the expense of directivity below the array. Radiation below the array axis was partially absorbed by the carpeted floor, was also interfered with by the barrister tables and chairs, and did not create a significant problem. The measured response of the array was within 2 dB of the predicted response at all angles.

5.3 System Implementation

The dual requirements for minimum external size and sufficient internal volume for the driver low-frequency response meant that a timber enclosure was unsuitable for the array. An enclosure of 1.5-mm sheet steel was constructed and faced with marble matching the bench face, with two steel sheets separated by a layer of car underbody paint used for the driver baffle. Drivers were fed via a wiring loom from the passive tapering and crossover filters located under the judges bench, external to the loudspeaker enclosure. With an internal volume of 1 L per driver the system f_{ct} was 270 Hz, with Q_{tc} = 1.0 [5], necessitating equalization to achieve the desired low-frequency response [9]. As all drivers handled the low frequencies, sufficient displacement-limited SPL was available from the array to produce the acoustic power of a raised voice at 125 Hz. The crossover between the array and the tweeter was an offset third-order type at 2.5 kHz. Equalization of the motional and inductive impedances of each driver pair in the array and the overall input impedance of the tapering filters was necessary to allow correct operation of the tapering and crossover filters.

Modeling of the EAD and coverage of the barrister area were done in octave bands using in-house software [1], based on the calculated radiation pattern of the array and tweeter system. Four of these systems were necessary to cover the total area occupied by barristers during a large court hearing, with the outer loudspeakers angled 15° outward. Architectural considerations constrained the lateral locations of the loudspeakers on the bench. Inevitable phase cancellations wider than one-sixth octave were present at the lower frequencies across the area, but these were more than offset by the benefits of improved coverage at higher frequencies.

6 CONCLUSION

Three unusual loudspeakers have been designed for the High Court of Australia. The form of each loudspeaker embodies strong architectural requirements and the need for a specific frequency response and radiation pattern to suit its local environment. A holistic process was used to design the loudspeakers, with the design work ranging from concept formulation and the modeling of radiation patterns to the laboratory and on-site passive-filter equal-

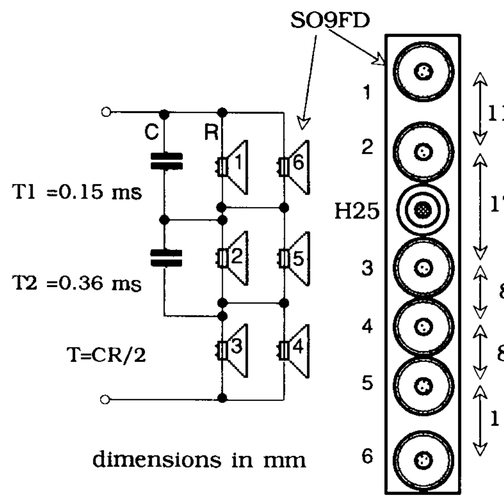

Fig. 18. Arrangement of drivers and tapering filter for short line array.

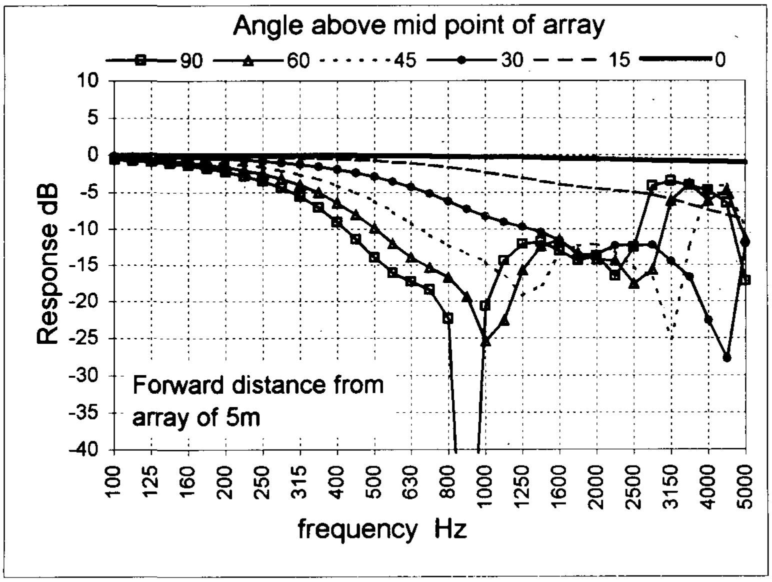

Fig. 19. Calculated frequency response above short line array.

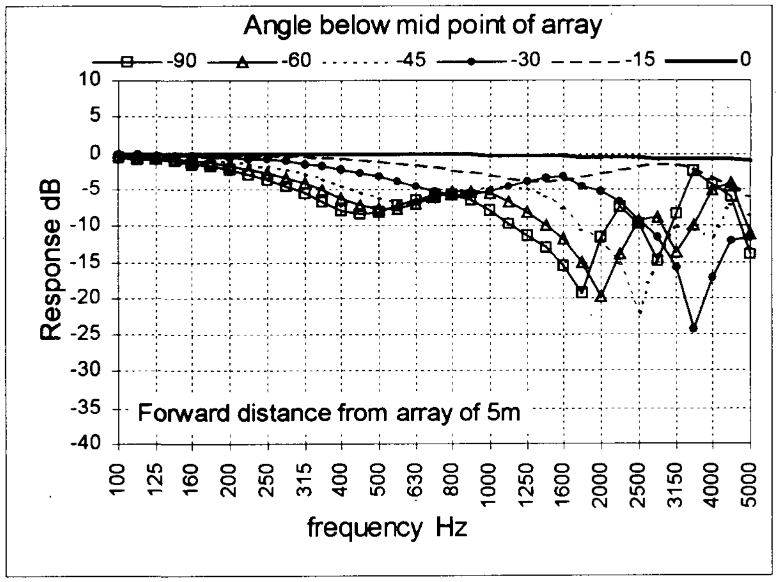

Fig. 20. Calculated frequency response below short line array.

ization of response and impedance. The final performance of each loudspeaker system has fully satisfied both the design goals and the user expectations.

7 ACKNOWLEDGMENT

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Mr. Leembruggen is a member of the AES and the Australian Acoustical Society and, until recently, lectured on loudspeaker and sound system design at Sydney University. He has a strong interest in directional loudspeakers and the application of beam-steered arrays to solve acoustic and architecture problems. Among his personal interests are restoring old houses and playing saxophone and clarinet in a rhythm and blues band.