Design and Commissioning of Sound Reinforcement Systems for the Australian Parliament—A Holistic Approach*

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The Australian Parliamentary Chambers present difficult circumstances for the reinforcement of speech, including an unusually wide dynamic range, large orator-to-microphone distances, high levels of uproar noise, and architectural and acoustical constraints. These circumstances were a challenge to designing a sound system that would provide high intelligibility with faithful spectral reproduction, wide angles of coverage, and a high acoustic gain and sound pressure level capability. Recognizing the interdependence of all factors, a holistic process was used to design and specify hardware. The solution included tapered line array loudspeakers and the use of original software. Commissioning the system involved numerical optimization of the gain structure and a novel technique that uses measurements of acoustic loop gain to predict equalizations that optimize the acoustic gain margin.

0 INTRODUCTION

The requirement of the new sound systems was to provide highly intelligible, natural sounding speech to each politician in the House of Representatives and the Senate Chambers under a wide range of operating conditions. The design process reflected a holistic consideration of the systems' interdependence of components, architecture, and performance requirements. The new systems replaced existing systems which gave poor intelligibility and tonal quality and had an early onset of acoustic feedback and distortion that prevented an orator's voice from being heard above the uproar noise of the politicians. Systems for the public galleries were to be installed in the future.

Measurements showed that the existing loudspeakers had a poor frequency response which varied spatially. Substantial energy was radiated by the loudspeakers toward both the chamber ceilings and the microphones, causing detrimental reflections and a reduction in acoustic gain margin. Measurements of the systems' speech intelligibility via RASTI were made using the DRA Laboratories MLSSA v 6.1 [1] and the Bruel and Kjaer 3361 analyzers in a quiet, empty chamber, ensuring that no compression or clipping was present. While results ranged from good to fair, with some bordering on poor, the subjective intelligibility was significantly worse than measured. This is possibly because RASTI is only a two-octave band screening device. A full-band STI analysis may have reflected subjective results.

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1 ENVIRONMENTAL CONSTRAINTS

1.1 Acoustic Environment

The Parliamentary Chambers are rectangular prisms approximately 23 m long, 23.5 m wide, and 16 m high. Open public galleries are located on each side of the chambers, 3 m above the floor.

Using Schroeder decay plots, measurements were made of the octave band reverberation times over the decay range of 5 dB to 15 dB in the empty House of Representatives [1]. The averages were around 1.1 s below 500 Hz, 1.3 s for 1 kHz, 1.5 s for 2 kHz, and 1.3 s for 4 kHz. While these early decays are adequate for speech in rooms of this size [2], [3], the higher RT values at higher frequencies imparted a coloration to the perceived sound. But the reverberation times did not give the whole picture. Surfaces below the galleries are highly absorptive because of acoustic panels on all walls, carpeted floors, and padded seats, and a semianechoic environment with weak early reflections results, similar to the open air. As much of the ceiling is parallel to the floor (consisting of perforated plasterboard with a low percentage of open area fronting absorptive material), audibly disconcerting reflections from the ceiling occur with natural speech, causing sibilance. Energy–time curves [1], [4] measured with a small loudspeaker simulating a voice showed that these reflections were only 12 dB down, arriving at around 90 ms. As early decay times (EDT) were less than 1.5 s, these late reflections were a more important factor in determining intelligibility than the RT_{60} values [5].

1.2 Background Noise, Microphones, and Physical Constraints

Debate in the Australian Parliament is characterized by a large amount of sporadic uproar (mainly in question time) with measured sound pressure levels (SPL) as high as 100 dBA rms at the main table and lasting for several seconds.

The House of Representatives had 138 microphones while the Senate had 94 (Fig. 1). Many microphones were to be either underneath or partly orientated toward the loudspeaker system. To accommodate poor microphone techniques, a standard distance of 700 mm between orator mouth and microphone was adopted.

The loudspeakers had to be as physically narrow as possible and were to hang from existing positions at a height that allowed uninterrupted sightlines between the galleries. Existing winches were to be retained, specifying a maximum loudspeaker cable diameter of 10 mm and a minimum cable length of 60 m.

2 SYSTEM REQUIREMENTS

The following performance specifications were proposed to apply to seating areas only, including future expansion of one row. No formal specification was applied to intelligibility, since our experience and Davis and Davis [5], [6] indicated that high intelligibility would result if the specifications were met. A prime requirement was naturalness of voice sound under all circumstances.

2.1 Acoustic Gain and Spectral Response

A sound system provides acoustic gain and, as long as ambient noise at the microphone is low, decreases the apparent listener–orator distance to the equivalent acoustic distance (EAD) [7]. At the 700-mm microphone distance the maximum EAD was to be 3–3.5 mm, with a 6-dB gain margin before the onset of acoustic feedback.

The bandwidth requirements for speech reproduction fidelity are often believed to be substantially less than those for music. Our experience is that, aside from reduced energy below 100 Hz and above 12 kHz, the requirements are identical as long as the frequencies critical for intelligibility [5] are assigned prime importance. The specified low-frequency limit was supported by analyzing the voice of two male politicians as well as other studies [2], [8]. The frequency response of a loudspeaker's direct field needed to be similar at all seats, as there would be insufficient early reflections (arriving within 50 ms) from other radiation angles to compensate for any dips in the loudspeaker's direct-field frequency response (greater than one-third octave wide) at a given seat.

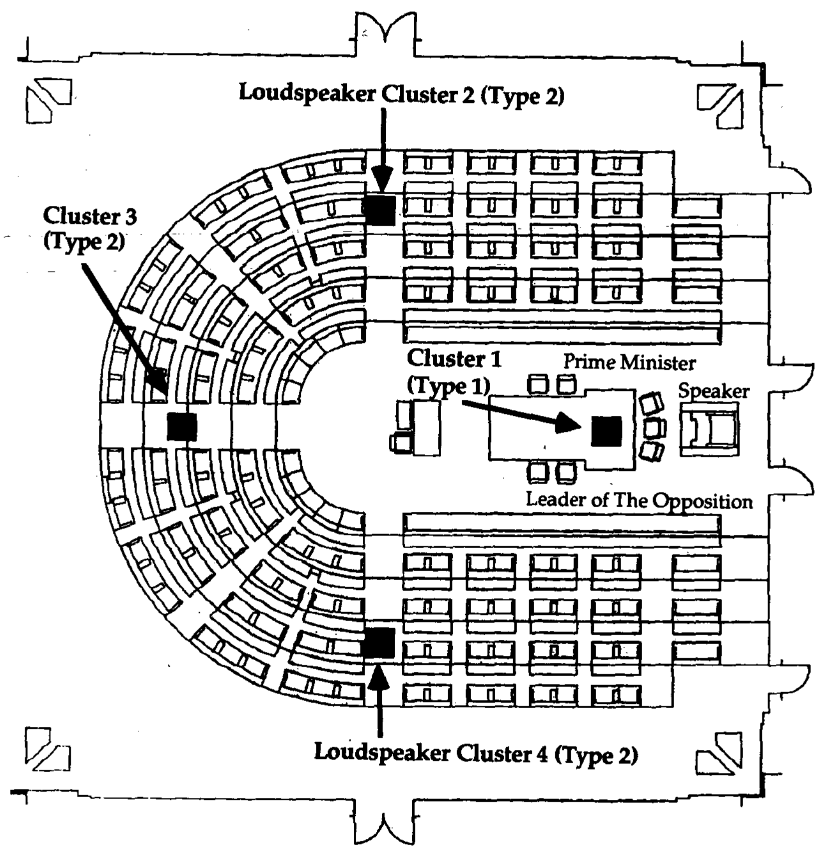


Fig. 1. Layout of the House of Representatives.

For 90% of the seats the specified frequency response of the direct field +50 ms (measured with one-third-octave smoothing) was 80 Hz to 15 kHz ± 3 dB. For the remaining 10% the limits were ±6 dB. Any peaks and troughs exceeding this margin were to be narrower than one-third octave. While a flat frequency response at each seat was considered more important than consistency of the broad-band SPL, the variation in broad-band SPL over the chamber was to be within ±4 dB.

2.2 Vertical Beamwidth

The loudspeaker radiation pattern was to be as constant with frequency and as narrow as possible to ensure that a minimum level of sound with a smooth frequency response reaches microphones. A vertical −6-dB beamwidth above 300 Hz of ±20° was necessary to 1) hold ceiling reflections to at least 20 dB below the direct field [5], 2) ensure that the early decay time is not increased [8], 3) minimize audible coloration by presenting the listener with a constant ratio of direct to reflected or reverberant sound, and 4) produce the required acoustic gain.

2.3 Maximum Undistorted Sound Pressure Level and Source Localization

For a 10% loss of intelligibility in a room whose broad-band EDT is 1.2 s Peutz [8] gives a minimum speech signal-to-noise ratio of 25 dB. As high intelligibility is not required during uproar conditions, a maximum system rms SPL of 107 dBA with a crest factor of 8 dB would give an adequate rms signal-to-noise ratio of 7 dB.

The loudspeaker system (type 1) used by the Speaker, the prime minister, and the ministers was to deliver a short-term peak SPL of 115 dBA at the seating perimeter with pink noise. Each system (type 2) used by the back-bench politicians was to produce 109 dBA SPL peak. Although these levels would be used only occasionally, the requirement for voice naturalness precluded the use of preemphasis or clipping to overcome the high noise levels, and the frequency response was to remain flat with nonlinear distortion below 10% at all frequencies.

Source localization allows the perceived origin of amplified speech to be at the orator. For minimum psycho-acoustic disturbance due to echo, the maximum time between the arrivals of the natural and the amplified sounds should be less than 25 ms for all orator–listener combinations. Where intervals are greater than 25 ms, the difference between amplified and unamplified sound levels becomes important [2], [5].

The existing system used eight loudspeaker clusters to achieve high localization, but to minimize visual impact, we proposed th use of only four loudspeakers (one type 1 and three type 2). To confirm the adequacy of localization, software was written to compare the levels and flight times of an orator's unamplified and amplified speech arriving at a listener. A total of 19 000 combinations of orator, listener, and loudspeaker were tested. If the modulus of the arrival time difference was greater than 25 ms, the level difference was compared against the criteria [2], [5]. Results indicated that the echo disturbance would be adequate to very good at every seat.

3 LOUDSPEAKERS

3.1 Overview

The type 1 and 2 loudspeakers comprise two full-range systems called "main" and "downfill," housed in a common structure. Both types utilize the same arrangement of drivers, enclosures, and filter polynomials. Consisting of an electrically tapered low-frequency array and a high-frequency driver, the main system provides coverage to most of the chamber away from its associated microphones. The downfill system is much smaller and irradiates areas below the loudspeaker, facilitating orator foldback. As the downfill system is equalized separately, less feedback equalization is required for the main system, substantially reducing tonal degradation for the bulk of the chamber sound. Fig. 2 shows the layout of the drivers. Both systems have separate enclosures lined with 50-mm-thick acoustic foam. The heavily braced plywood structure is constructed around a steel space frame to which the winch cable is attached. Each box hangs at 7.5 m above the floor, tilted at 35°. Special multicore loudspeaker cable systems were designed and manufactured, and cable losses are less than 1 dB.

3.2 Low-Frequency Drivers

In the main system two adjacent vertical arrays of six 250-mm JBL 2123 drivers provide output up to 1500 Hz. The 2123 driver provided the required combination

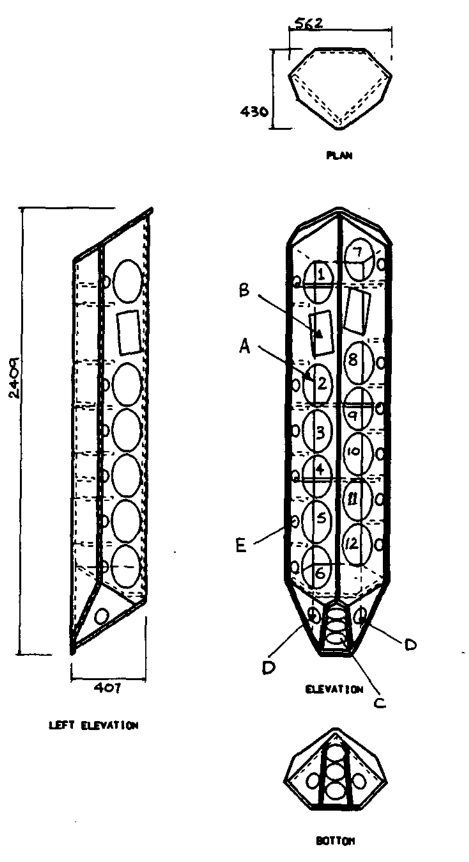


Fig. 2. Driver layout in loudspeaker enclosure. A—2123 drivers; B—HP94 horns; C—2105 drivers; D—2404 horn drivers; E—vents.

of high sensitivity (99 dB/W), high thermal power rating (250 W), and suitable Thiele–Small parameters and frequency response flatness/smoothness, both on and off axis. Thermal compression [9] is not a problem as the high SPLs have short duration. At 45° off axis and 1.5 kHz the response of the 2123 is -5.5 dB relative to on axis, and thus the interaxial angle between the arrays was made 90°.

While the array length provides directivity down to 80 Hz, electrical tapering is applied to the drivers to produce a more constant radiation pattern and less off-axis lobing at higher frequencies [10]. This arrangement produces a narrow vertical (40°) and wide (180°) horizontal radiation pattern over much of the frequency range. Active electrical tapering is used for the higher SPL capacity type 1 system. Passive tapering in the type 2 systems allows a lower quantity and rating of power amplifiers per loudspeaker.

Classical tapered arrays utilize small drivers with logarithmic spacings (see [11]), but the need to mount all woofers and horns as closely as possible resulted in nonideal woofer spacings.

3.3 High-Frequency Drivers

For frequencies above 1500 Hz, two Electro-Voice N/DYM-1 compression drivers (76-mm diaphragm) are used with two EV HP94 constant-directivity horns (90° horizontal by 40° vertical and 50-mm throat). This combination was necessary for minimal distortion, diaphragm longevity and pattern control at 1500 Hz, and maximum SPL. To minimize phase cancellations in the regions where the horn radiation patterns overlap, the horns are located side by side with an interaxial angle of 90°. While the neodymium magnets of the N/DYM-1 allow close spacing of all drivers, the horn throats need to overlap horizontally and be offset vertically.

Mark IV acoustic modeling software ACOUSTACADD was used for setting up the orientations of the horns to give optimum coverage. Fig. 3 shows the predicted direct-field coverage pattern of the main system at 2 kHz (power summation). Prediction of phase cancellations was not possible with that version of ACOUSTACADD and the cancellations were worse than expected, resulting in poorer coverage in the overlap areas than shown. During the testing of the prototype in the chambers, minor adjustments to the horn orientations were made by trial and error to improve the consistency of coverage, and the worst cancellations were arranged to occur in the inner areas not occupied by listeners. Better coverage without cancellations would have resulted if a small constant-directivity horn with a nominal pattern of 170° horizontal by 55° vertical had been available. There appears to be a need for an expanded range of commerically available horns.

3.4 Low-Frequency Response

A nonstandard sixth-order vented alignment [12] is used for the main system and provides a good compromise between enclosure volume, frequency response (-3 dB at 80 Hz), and displacement-limited SPL. All 12 woofers share the acoustic volume. The associated second-order high-pass filter (f_{res} = 70 Hz, Q = 1.8) is located in the external signal processor, compensating for the small volume (11.5 L per driver) and the low system Q_1 [13, eq. 49] of 3.3. The measured Thiele–Small parameters of four 2123 woofers were different from the manufacturer's data, so their average was used.

In vented-box systems the cone acceleration has a notch at the box resonant frequency f_B and the system impedance reflects a minimum near f_B [13]. Initially six unevenly spaced vents were used in the prototype, and the driver acceleration notches (measured with an accelerometer of 1-g mass), ranged from 87 to 125 Hz. The minimum impedance occurred at 105 Hz with series–parallel connection of the drivers. The box was modified to use 12 evenly spaced vents, and the notches ranged

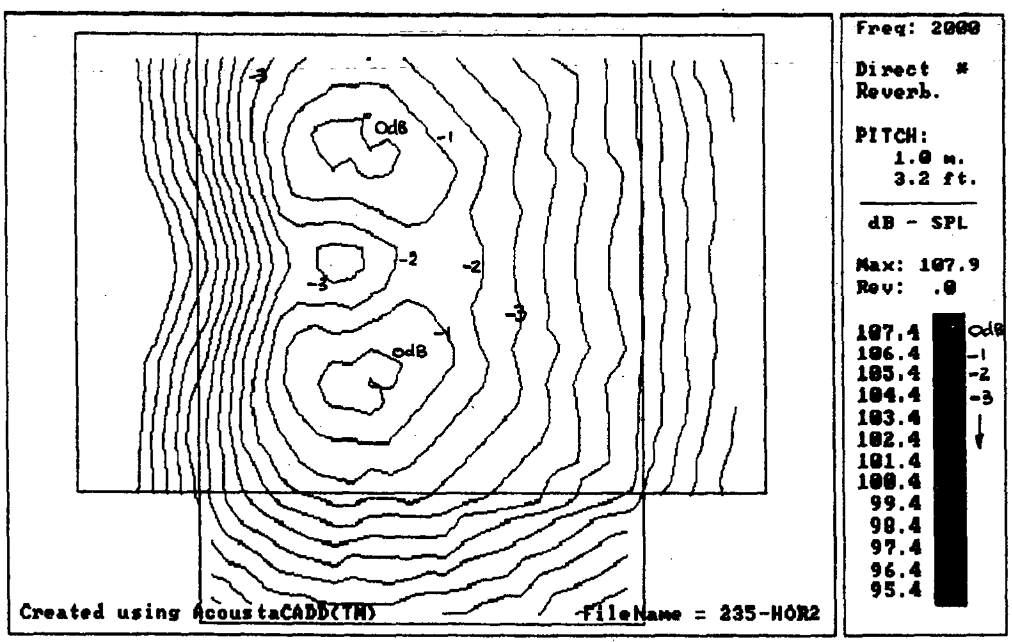


Fig. 3. Direct-field coverage pattern predicted by ACOUSTACADD.

from 79 to 105 Hz, with the minimum impedance at 85 Hz. The acceleration notch frequencies were found to be independent of the driver, and it is anticipated that the range of nulls results from differences in each driver's apparent acoustic volume. Further research is required for systems in which a number of drivers share a common acoustic volume with multiple vents.

3.5 Driver Arrangement and Tapering Filter

The active–passive electrical tapering filters create three signal chains—low, mid, and high. At low frequencies all three chains share the output equally, but as the frequency increases, the signal is progressively transferred to the high chain by rolling off the low and mid chain signals. As the vector sum of the three chains is constant, overall equalization is not required, allowing a higher maximum SPL. Software tested and eliminated the possibility of tapering with 12-dB per octave low-pass filters (requiring equalization), as a suitable radiation pattern could not be produced with the available driver spacings.

In Fig. 2 the 2123 drivers 1, 7, 6, 12 handle the low chain, drivers 2, 8, 5, 11 handle the mid chain, and drivers 3, 9, 4, 10 cover the high chain. The distance between horizontally adjacent drivers results in some phase cancellations in the coverage area where the driver radiation patterns overlap. At around 600 Hz the inner four drivers produce most of the acoustic output, whereas at 1200 Hz one driver per side is rolled off by a 6-dB per octave passive low-pass filter located in the loudspeaker box. The rolloff reduces the level of radiation pattern lobing caused by a vertical driver pair on each side and is mostly counteracted by the inherent, rising response of the unfiltered 2123 drivers.

Nonstandard driver spacings necessitated a first-principles approach using in-house software to design the tapered array. The time constants of the tapering filter, the 1200-Hz low-pass filter, and the location of the HP94 horns were chosen by trial and error to optimize the total radiation pattern. The software functions as follows:

Look-up tables hold the location and the pan and tilt angles of each 2123 driver, along with the 2123's amplitude and phase response at one-sixth-octave frequency intervals at 15° polar increments. Using direction cosines, the angle of a listener point relative to each driver's axis is calculated and quantized to 15°. The total vector of each driver's acoustic output is computed at this point from the combination of amplitude loss due to geometric spreading, phase shift due to flight time, and the complex frequency responses of the tapering filter and driver at the relevant 15° polar. All driver vectors are summed to give the system response at this frequency and point, and the process is repeated at one-sixth-octave intervals and appropriate spatial points. Fig. 4 shows the predicted behavior of the array at a number of points.

The passive tapering filter is located in the type 2 loudspeaker enclosure. Constant load impedances are required for this passive filter, and equalization of the motional and inductive impedance of each of the three driver chains is provided. Fig. 5 shows the circuit diagram of the type 2 passive network.

3.6 Maximum SPL Calculations

Using estimates of speech spectral energy handled by the three low-frequency and the high-frequency signal chains, calculations were made to ensure that the required direct-field SPLs at 15 m were available under conditions of driver thermal and excursion limiting [9], [13] and amplifier clipping. An advantage of the tapered array is that all drivers provide low-frequency output, allowing drivers with a small $x_{\max}$ rating to produce high SPLs.

3.7 Downfill System

The downfill system is located on the lowest surfaces of the enclosure (Fig. 2), and this imposes severe limitations on the size and layout of the drivers. Due to reduced directivity, the main system supplies energy below 150 Hz to the downfill areas. Frequencies below 3 kHz are covered by three 125-mm JBL 2105 drivers located in a linear array. Above 800 Hz the outputs of two of these drivers are rolled off by a passive 6-dB per octave low-frequency filter to reduce radiation pattern lobing. On axis the rolloff is mostly counteracted by the rising frequency response of the third unfiltered driver, which is located at the front of the array. In front of the array, a partial null is produced intentionally at midfrequencies. Frequencies above 3 kHz are reproduced by two JBL 2404 constant-directivity (100° horizontal by 100° vertical) horn–driver assemblies fitted with nonstandard diaphragms that extend their lower frequency response. Their main axes are inclined at 45° below horizontal. The crossover from the 2105 to the 2404 drivers is by passive second- and third-order networks, incorporating high-frequency response equalization for the falling power response [14] of the 2404 and impedance equalization. This crossover arrangement produces the best overall acoustic summation in the downfill area.

4 MICROPHONES

All microphones in the chambers were replaced by AKG CK 1 capsules with C451 preamplifiers, which were chosen from six candidates (after testing in an anechoic chamber) for their extended and smooth frequency response and their consistency of polar pattern. The position of the Speaker (chairperson) of each chamber is reasonably static, allowing AKG CK8 minishotgun types to be used for their microphones, which must always be open.

5 PREDICTION OF EAD AND GAIN MARGIN

Eq. (1), from Davis and Davis [7], gives the equivalent acoustic distance (EAD) at the listener,

$$\mathrm{EAD} = 2D_s \frac{D_1}{D_m} \tag{1}$$

This equation is based on a geometric spreading loss of the direct field. It assumes one open microphone, a 6-dB gain margin, and omnidirectional loudspeaker, and microphone polar patterns, and it ignores early reflections and the reverberant field. Only microphone–loudspeaker distances less than the critical distance are valid. Eq. (2) gives the available acoustic gain from microphone to loudspeaker under these conditions,

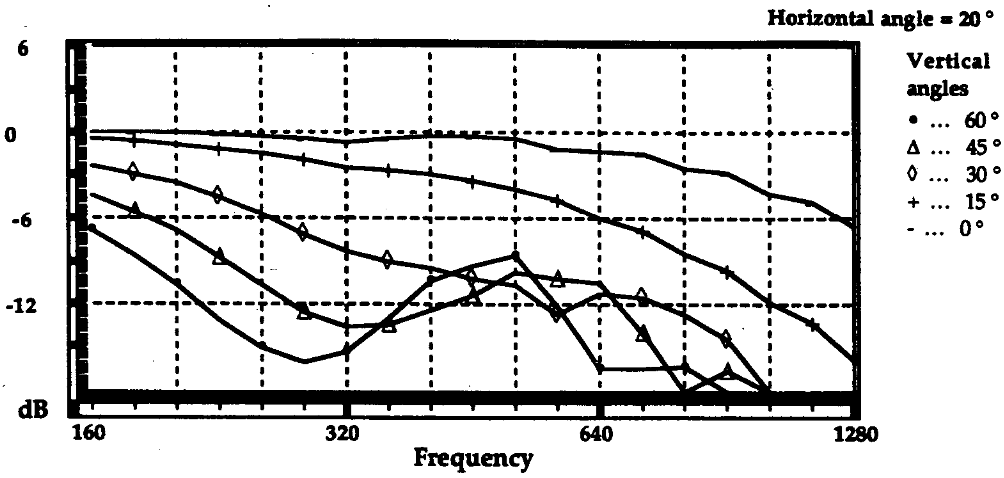
$$G_{av} = 20 \log \frac{D_m}{2} \tag{2}$$

In these equations D_s is the distance from the microphone to the orator, D_l is the distance from the loudspeaker to the listener, and D_m is the distance from the microphone to the loudspeaker.

A more accurate prediction was needed for each microphone's gain margin with both the reverberant field and the directionality of microphones and loudspeakers considered. Software was written to determine the margin at octave frequencies using the gain set by Eq. (2). The modeling of early reflections was beyond the scope of the project.

Look-up tables provide locations and pan and tilt angles for the microphones and loudspeakers, octave-band room constants, and directional data for the loudspeakers and microphones quantized to 15° increments. The calculated directional data for the low-frequency array were used and the array Q was calculated using [15]. Measured and published data were used for the microphones and high-frequency horns, respectively. The downfill system was not included.

Using direction cosines, the direct sound's angle of exit from the loudspeaker and angle of entry into the microphone are determined and quantized. The relevant attenuations due to directionality are found from the look-up tables and added to the direct-field geometric loss from loudspeaker to microphone. Using the orator SPL at the microphone, the loudspeaker and microphone sensitivity, and the electrical gain, the effective direct and reverberant field levels from the loudspeaker arriving at the microphone are determined and summed. The



Australian Parliament, Type 1

Conditions: Listener distance—10 min; midpoint (m) x, y, z—0.824, 0, 0; driver directionality considered.
6-dB per octave constant voltage tapering filter.
Low chain capacitor—130 μF; mid chain capacitor—118 μF; phase lead resistor—0; phase lead capacitor—0.
Additional low-pass filter at 6 dB per octave—driver 4 at 1200 Hz.
Additional low-pass filter at 6 dB per octave—driver 10 at 1200 Hz.
Array layout for 12 JBL 2123 drivers.

Driver	Group (H/M/L)	Attenuation (dB)	Height (m)	−L/R+	Position (m) Forward	Pan	Tilt
1	L	0	1.545	−0.113	0	45	0
2	M	0	0.98	−0.113	0	45	0
3	H	0	0.735	−0.113	0	45	0
4	H	0	0.49	−0.113	0	45	0
5	M	0	0.245	−0.113	0	45	0
6	L	0	0	−0.113	0	45	0
7	L	0	1.648	0.113	0	−45	0
8	M	0	1.103	0.113	0	−45	0
9	H	0	0.857	0.113	0	−45	0
10	H	0	0.612	0.113	0	−45	0
11	M	0	0.367	0.113	0	−45	0
12	L	0	0.125	0.113	0	−45	0

Fig. 4. Array response prediction output.

difference between this total and the orator level is the gain margin. This margin is typically 9 dB, with 5.5 dB the lowest value, and in general it is limited by the reverberant field.

6 ELECTRONIC EQUIPMENT

Fig. 6 shows the block diagram of the signal-processing equipment. Programmable one-sixth-octave equalizers for the main and the downfill systems allow equalization for tonal balance and gain margin maximization. Each main system uses a type 1 or type 2 processor and a signal delay for correction of the time offset between the 2123 and the N/DYM-1 drivers.

The downfill system is protected from thermal overload and overexcursion in long-term shouting situations by the limiter and a 160-Hz 12-dB per octave high-pass filter in the equalizer. The downfill delay provides partial alignment of the arrival times of the main and downfill systems, optimizing the acoustic summation in those areas where the systems overlap.

6.1 Processors, Types 1 and 2

The processor is integral to the main system, providing accurate and tamper-proof filtering. Fig. 7 shows a block diagram of the type 1 processor. The active tapering filtering is omitted in the type 2 unit. High-rate (30-dB per octave) crossover filters prevent large amounts of radiation pattern lobing in the crossover region, which would be exacerbated by the large distances between the low-frequency and high-frequency drivers. Computer modeling showed that with the drivers' relative amplitude and phase responses, offset fifth-order Butterworth low- and high-pass filters centered at 1.5 kHz would produce the best spatial frequency response compared to other crossover types.

Active boost and notch filters equalize the falling high-frequency power response of the horn–driver combination [14], producing a fairly flat on-axis frequency response out to 15 kHz without major peaks off axis. This network provides 7 dB per octave of boost, but has sharp transition zones, resulting in 10 dB of boost at 15 kHz. This is significantly more boost than conventional schemes [16], which use a single pole–zero combination for a nominal 6-dB per octave boost.

A second-order all-pass filter gives the main system

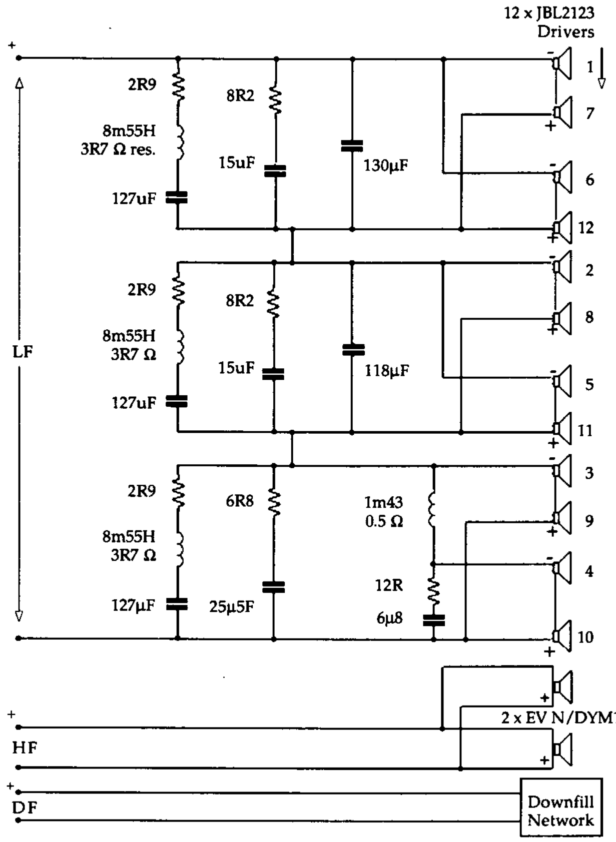


Fig. 5. Circuit diagram of type 2 passive filter network.

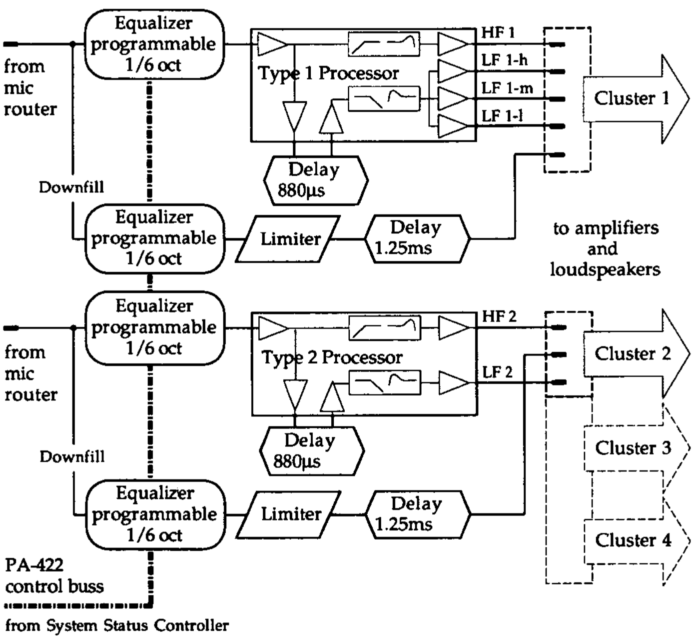


Fig. 6. Block diagram of signal-processing electronics.

an additional 180° of phase shift at 400 Hz compared to the downfill signal to improve the summation between the main and downfill systems directly below the loudspeaker. Boost and notch filters equalize minor aberrations in the main system's frequency response, and a 40-Hz high-pass filter provides additional low-frequency protection.

6.2 Power Amplifiers

The type 1 system uses six power amplifier channels. The tapered low and mid chains, high frequency, and downfill, each use one 360-W per 8-Ω channel, while the high chain uses two channels in bridge mode (1620 W per 8 Ω). In the type 2 system the tapered array uses two channels in bridge mode (1200 W per 8 Ω) while the high frequency and downfill each use a 360 W per 9-Ω channel.

6.3 Equalizers and System Status Controller

When an operator activates a microphone, an equalization specific to that microphone is applied to its associated loudspeaker by a programmable equalizer. Memory selection in each equalizer is controlled by the custom-designed system status controller via a PA-422 link, according to the microphone number code output by the operator's console. The only programmable one-sixth-octave equalizers available during the project were White 4710, which have 10 memories. Memory 1 holds the equalization that optimized the tonal balance of the system and formed the basis for equalizations to minimize feedback. The remaining memories hold the total equalizations (tonal balance plus feedback) for nine groups of microphones, after individual microphone equalizations were grouped according to similarity.

The controller holds a look-up table that gives the main and downfill equalizer memory number corresponding to each microphone and its priority level. If more than one microphone is open, the equalizers are set for the microphone with the lowest gain margin.

7 LOUDSPEAKER PERFORMANCE TESTING

A prototype loudspeaker and processor were constructed and tested in a large anechoic chamber. The performance of the tapered array was reasonably similar to the predictions, and effects such as diffraction and differing driver radiation impedances were assumed to account for the unpredicted aspects of the performance. The prototype was evaluated in different positions in both parliamentary chambers, and refinements were made in the orientation of the high-frequency drivers to improve the coverage. Other tests were carried out on the final installation.

7.1 Spectral Output and Coverage

Fig. 8 shows the frequency response of the prototype type 1 system (with ±2 dB of equalization) measured with the MLSSA v 7.1 FFT analyzer at two locations using the first 50 ms of impulse response data [2]. The low-frequency response at these locations is slightly less than specified but was not pursued, as other measurements showed that the lower −3-dB point was 80 Hz. The dip at 210 Hz was not a loudspeaker effect and was not aurally significant.

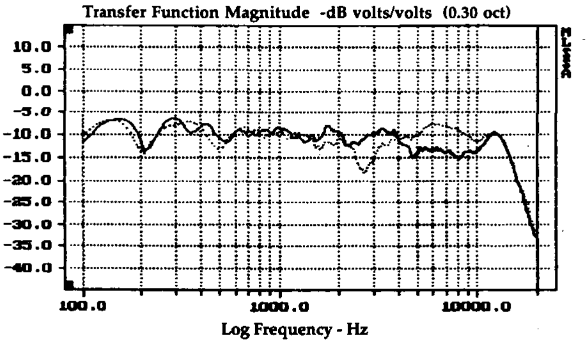


Fig. 8. 50-ms frequency response of prototype loudspeaker at two seating positions.

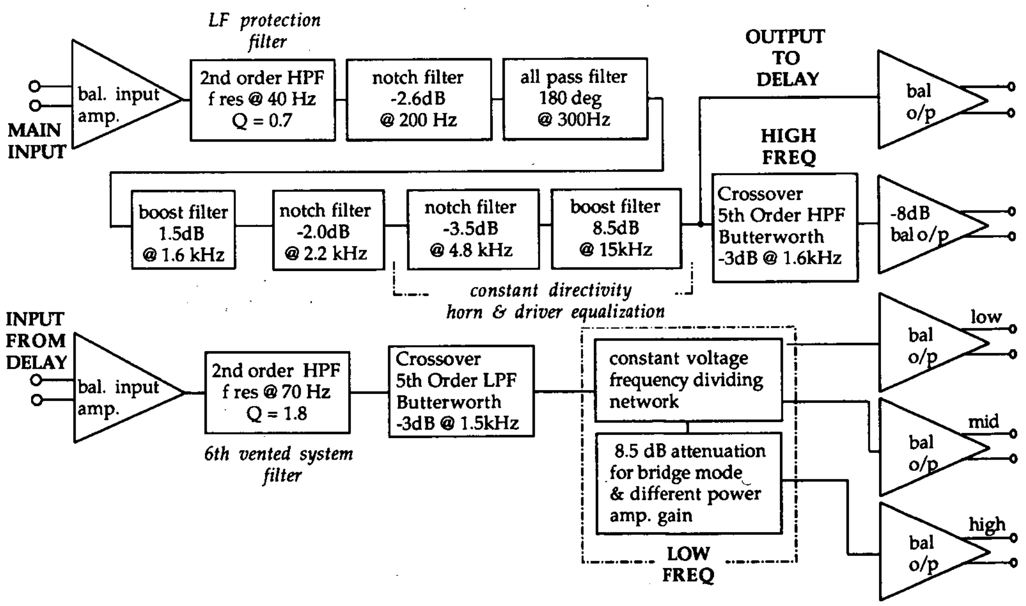


Fig. 7. Block diagram of type 1 signal processor.

The B&K 2143 multimemory real-time analyzer was used with pink noise to measure the one-third-octave spectral output up to 4 kHz at all seats in the chamber. This analyzer was able to rapidly acquire, store, and output large amounts of data compared to the then current MLSSA version. As there was strong agreement between the pink noise and MLSSA 50-ms responses up to 4 kHz, and above 4 kHz the MLSSA results showed a flatter frequency response, the use of the "stationary" pink noise signal was deemed adequate. The variation in broad-band SPL between seats was within the specified ±4 dB, with more than 92% being within ±3 dB.

Standard deviation was used to quantify the frequency response envelope at all seats because it provides useful information about aural performance. If the response at measurement position A has all one-third-octave bands at 0 dB, except for one at +6 dB and another at −6 dB, it has the same absolute envelope of ±6 dB as position B, which has, say, 10 bands at ±6 dB. Position A will have a much lower standard deviation than position B, and the sound at A will be more natural and consistent. The maximum standard deviation was 3.5 dB, with the typical value at 2 dB. Whilethere are a few seats whose response is outside the absolute specification, their aural result is satisfactory.

7.2 Equivalent Acoustic Distance

With the total system equalized, the EADs were measured at five seats for each of four microphones. A small loudspeaker was set up at 400 mm from the microphone, fed with band-limited (150-Hz to 7-kHz) pink noise, and equalized (flat ±2 dB). The broad-band SPL was then measured at the microphone and scaled for a source at 700 mm. With the gain of the microphone set at 6 dB below constant regeneration, the level of the amplified noise as measured at each seat. The contribution of the unamplified noise was removed from the received level and the difference between scaled source and receive levels was converted to an EAD value. Analysis of the clarity ratio C_{50} at the seats indicated that the contribution of the reverberant field (inherent in the measurement) would decrease the EAD results by less than 8%.

The measured EADs for the type 1 system with the Speaker's microphone and one other microphone open averaged 2.1 m, the worst case being 3.2 m. The type 2 systems gave similar results with one microphone open. Since the results were as predicted without the downfill operating, the benefits of the separate downfill system had been realized.

7.3 Maximum SPL, Intelligibility, Directionality, and Localization

The maximum SPL produced by the type 1 prototype was measured at the onset of amplifier clipping using clipped pink noise with a 10-dB crest factor. At the most distant seat from the loudspeaker, the peak SPL was 118.5 dB lin (116.3 dBA).

According to the experienced listeners present, the voice sound quality was natural, warm, dry, and intimate. While reverberant sound was natually audible, late reflections from the ceiling were not audible, even with percussive music. The source localization experience was satisfactory.

The intelligibility of the final system was measured at a few locations in the chamber (while it was quiet) using the STI routine in the MLSSA v 7.1 analyzer, and was found to be "fair." However, this result was contrary to all aural impressions, where after extended listening to live speech and prerecorded speech and vocal music, the client group concluded that intelligibility was very high. Limited time precluded the use of standard articulation-loss tests using words embedded in carrier phrases. Thus as the STI results did not reflect the remarkable improvement in intelligibility and clarity over the old system, more research appears to be required in this area.

8 COMMISSIONING THE SOUND SYSTEM

The process of commissioning the sound system is as important as the design stage, because it ensures that the system is working at its full potential.

8.1 Gain Structure

To optimize the system gain structure, the signal chain from the orator to the most distant listener was optimized mathematically using an iterative nonlinear optimization routine. Incorporated in the model were the gain, normal operating level, maximum output level, and residual noise level of each electrical, electroacoustic, and acoustic element in the chain. With an EAD of 2.5 m set at the listener, the gains of the microphone preamps, distribution amplifiers, and attenuators were determined so that the targets of 1) minimum total noise levels less than 5 dB above ambient, 2) maximum headroom, and 3) 115-dB peak total maximum SPL were achieved. Fig. 9 shows the level diagram of the high-frequency chain for which the target maximum SPL was 109 dB.

8.2 Spectral Balancing

Equalization for spectral balancing was carried out using measurements with the MLSSA analyzer and by listening. To reflect the subjective tonal balance, the frequency response was measured using the first 50 ms of impulse response data [5], [17]. By postprocessing the MLSSA data on a computer, the response at each of 17 seats was smoothed to one-twelfth octave, scaled to 0 dB based on the average level between 250 Hz and 8 kHz, and the average response was determined. As consistency of the frequency response between positions is more aurally critical than consistency of broad-band level (within the allowed variation), the scaling ensured that the average response was based only on spectral differences and not on the overall level.

Initially the average response of the only main system over its coverage area was determined and equalized. The downfill was then introduced, and after measurement of the combined response in the downfill area, the downfill response and gain were equalized. The measurement adjustment procedure was then repeated with both systems on. Live speech, and speech and vocal and instrumental music from compact discs were then played to assess the tonal balance and audibility of the measured peaks and troughs throughout the chamber. Minor subjectively based adjustments were made, and the measurement–adjust–listen–adjust process was repeated for all positions.

8.3 Feedback Equalization

The large number of microphones meant that adjustment of the equalizers by trial and error for maximum gain margin was not practical. A method was designed to predict the required equalization from the measured closed- and open-loop transfer functions of each microphone. This method together with some supporting laboratory experiments are presented in the Appendix.

As the air temperature and the location of each microphone are fixed, the main variable in the acoustic loop gain is the proximity of a politician to the microphone. To account partially for the acoustic effects introduced by a politician, a person of average height stood facing the microphone under test. With the setup described in the Appendix (see Fig. 10) and the tonal balance equalization in place, the system gain was set to 2 dB below the gain that the most susceptible microphone required for constant regeneration. Using 712 ms of impulse response data, measurements were made for each microphone of the closed- and open-loop transfer functions of the combined system and the open loops of the main and downfill systems.

Software (described in the Appendix) was written to extract the required main and downfill equalizations for each microphone from these measurements. If a major acoustic cancellation at a certain frequency is occurring between the main and downfill systems, then an acoustic phase change resulting from a change in an orator's position could upset this cancellation and reduce gain margin stability. To produce more operational stability, a routine is included that assesses the extent of cancellations. The target level L_v for the equalization is the required open-loop gain stability margin and is initially set at -7 dB. If the level difference between the main and downfill signals is significant, then the lower of these two signals has only a minor effect on the combined level, and equalizing it may result in the unnecessary removal of energy. A look-up table provides a weighting factor W_t that corresponds to the level difference and reduces the equalization applied to the lower signal.

8.4 Allocating Equalizations to Memories for Back-Bench Groups

As each back-bench group contains up to 50 microphones, a process was required to compress the group's microphone equalizations into nine subgroups corresponding to the nine available main and downfill equalizer memories. A subgroup's equalization at each one-sixth-octave frequency would be determined by the microphone needing the most attenuation at that frequency. To minimize the amount of unnecessary equalization applied to each microphone, a combination of computer correlation and hand sorting was used to form the most appropriate subgroups.

To determine the similarity of the equalization for microphone M with equalizations for microphones 1 to N, all equalizations in each one-sixth-octave band were quantized to 1.5-dB increments. Then using Eq. (3), the similarity scores were calculated and arranged in increasing order. Those with low scores were possible candidates for a subgroup. The allocation process was

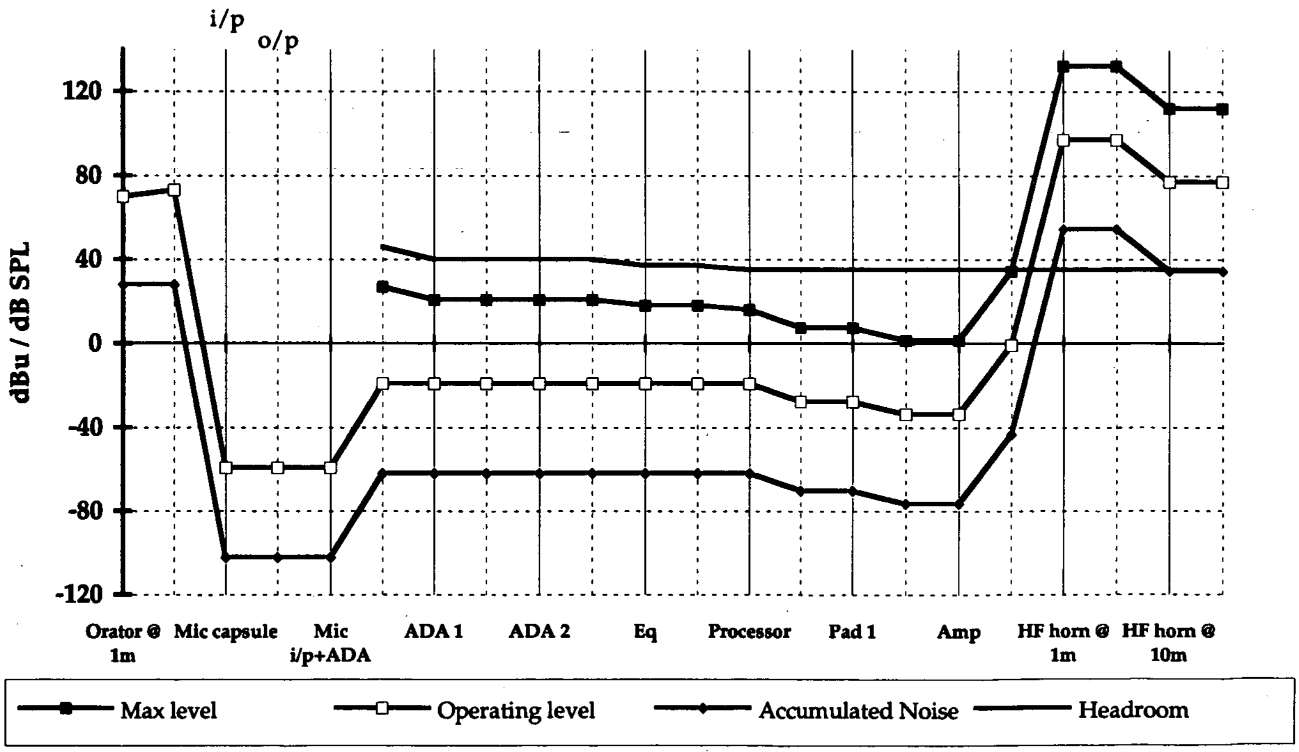


Fig. 9. Signal level diagram for high-frequency chain.

then finished by hand sorting.

$$S_{M,N} = \sum_{i=1}^{j} |\mathrm{Eq}M_i - \mathrm{Eq}N_i| \qquad (3)$$

where $C_{M,N}$ is the similarity between the equalizations for microphones M and N, and EqM and EqN are the values of equalization for microphones M and N in the ith one-sixth-octave band.

8.5 Results

After setting up the equalizer memories to the required responses, the actual feedback stability margins were verified by measurement. Typically an additional 2.5–4 dB in gain margin was obtained. As only 11% of the microphones needed minor individual adjustment to improve their stability, the algorithm appeared to have merit. As expected, there was a slight degradation in the aural tonal balance of the back-bench microphones due to the grouping of feedback equalizations, but the overall tonal quality was quite satisfactory.

9 CONCLUSIONS

Using a holistic approach that recognized the interdependence of macro and micro components, a sound system was designed to meet the difficult requirements of the Australian Parliamentary Chambers. A high-powered, processed loudspeaker utilizing a tapered array was designed. In commissioning the system, novel computational methods were used to design the system gain structure and to determine equalizations that optimized the tonal balance and increased the acoustic gain margin. Aural and measured results illustrate the merits of the approach. The system provides the required level of stable acoustic gain together with sutiable coverage, high tonal quality, and high SPL capacity. The intelligibility and performance of the system have satisfied the users.

10 ACKNOWLEDGMENT

The authors wish to thank the Sound and Vision Department of the Australian Parliament House for their vision and commitment to both excellence and a long-term solution that is based on detailed engineering design and verification, rather than the latest software or loudspeaker.

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APPENDIX 1
BASIS OF THE METHOD

To investigate aspects of acoustic feedback, equipment was set up in our laboratory as shown in Fig. 10. In the closed-loop configuration the system oscillated at 1189 Hz. The system's open-loop amplitude and phase response is shown in Fig. 11 (calculated from the system's open- and closed-loop impulse responses using 350-ms time data, with a half Blackman–Harris window). Only at 1190 Hz is the Nyquist condition for regeneration fulfilled, although the loop gain at 1550 Hz is higher. In contrast, with only 45 ms of impulse response data, the phase at 1189 Hz was 43°.

At a different location and with the attenuator adjusted

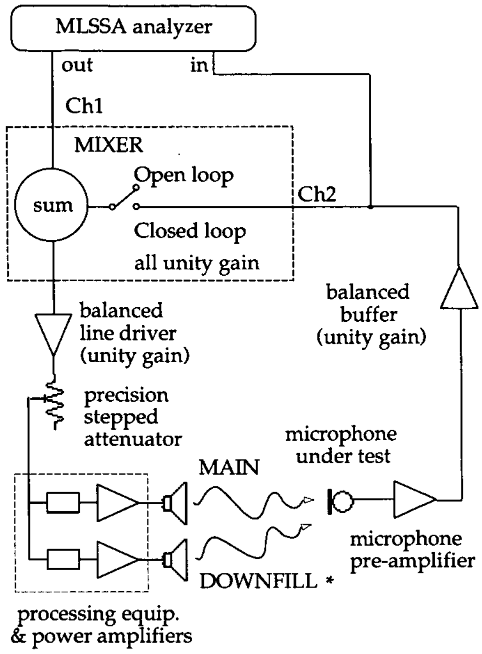


Fig. 10. Setup for measurement of system open- and closed-loop gain. *Not used in laboratory setup.

so that the system was ringing but stable, the system was excited and the power spectrum of Fig. 12 measured. Dominant spectral lines at 353, 482, 550, and 6600 Hz are evident. An additional 3-dB attenuation was switched in and the open- and closed-loop transfer functions were measured. Fig. 13 shows the open- and closed-loop amplitude responses, and the regeneration at 353, 482, and 550 Hz is clearly visible. Therefore with sufficient impulse response data and a linear system, frequencies subject to regeneration can be identified by the difference between the closed- and open-loop amplitude responses. It is important to recognize that a sound reinforcement system is always in regeneration and degeneration. When the stability margin is greater than 10 dB, the regeneration cannot be heard, as the decay rate is high and factors such as background noise and reverberation and auditory masking overwhelm it. If the polarity of a microphone is reversed, the gain margin can increase or decrease, and this depends on the bandwidth of the open-loop response peaks and the number of loop phase rotations that are present within each peak Uzzle [18] gives further information.

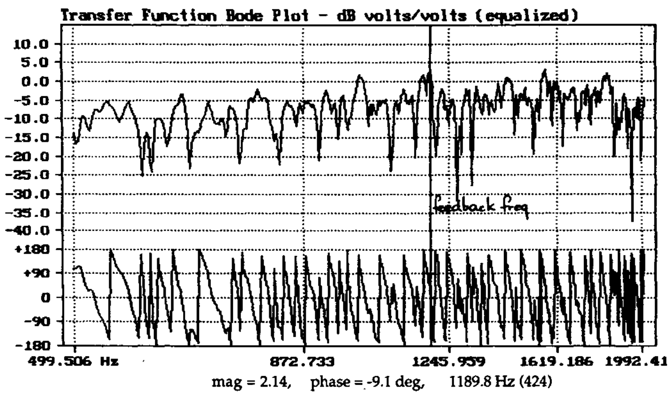


Fig. 11. Measured open-loop amplitude and phase response of loudspeaker–room–microphone system.

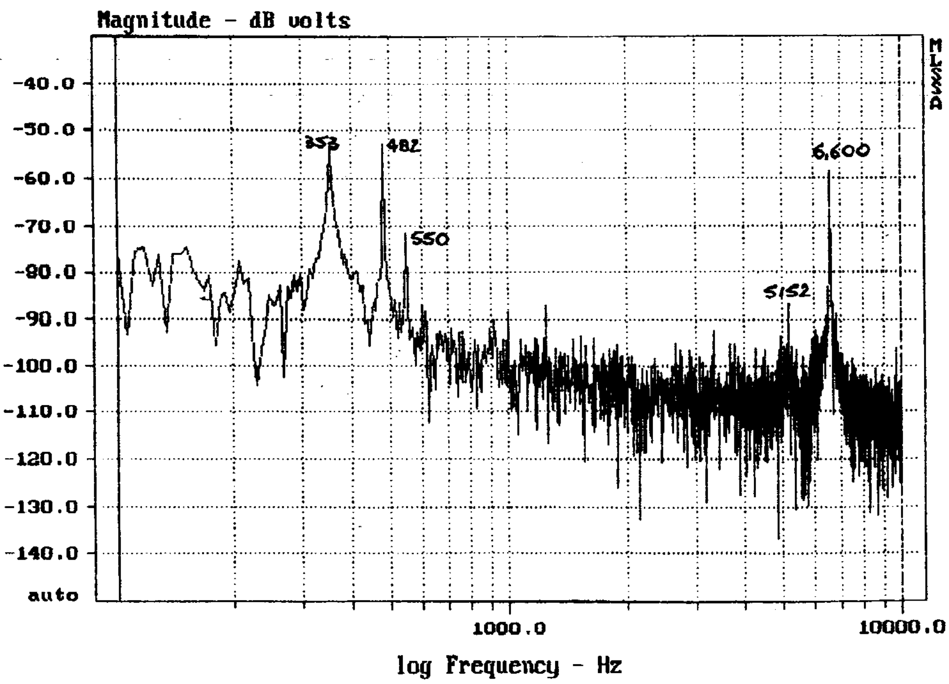


Fig. 12. Power spectrum of stable "ringing" feedback.

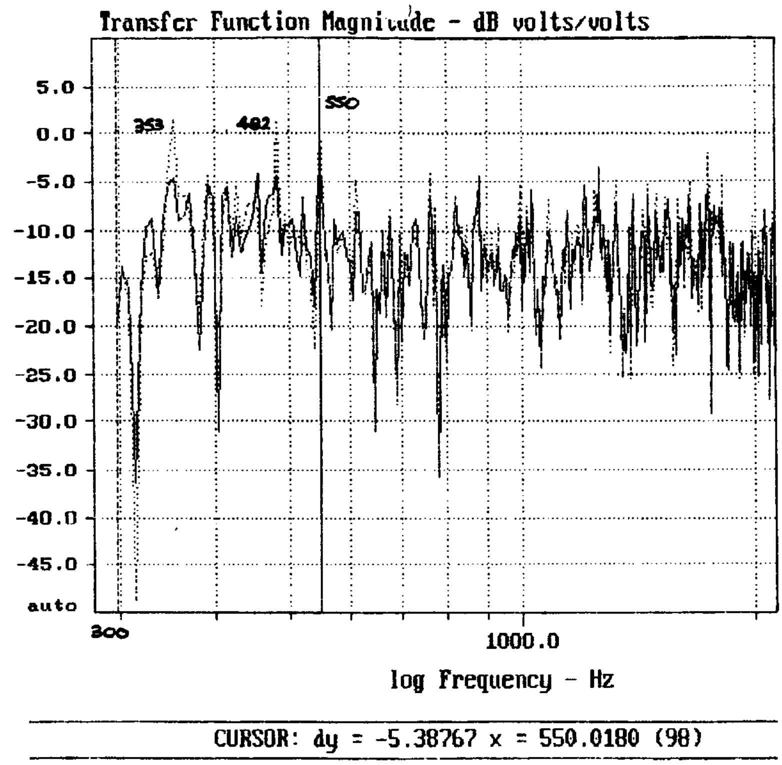


Fig. 13. Closed- and open-loop responses of loudspeaker–room–microphone system with 3-dB attenuation. −3 dB below regeneration—--- closed loop; ——— open loop.

APPENDIX 2
ALGORITHM TO DETERMINE THE EQUALIZATION TO MAXIMIZE SYSTEM STABILITY

Test Condition	Action	Decision	Comment
	Take in 8192 data point FFT. Place frequency points in $^1/_{12}$-octave wide octave bins.		
1. *Detection of regeneration*			
	Calculate $CC(f) - CO(f)$.	For each frequency point in each $^1/_{12}$-octave bin.	
	Find $CO(f)_{max}$, $MO(f)_{max}$, and $DO(f)_{max}$.	Within each $^1/_{12}$-octave bin.	Maximum values in each bin.
$CC(f) - CO(f) > 0$.		System is regenerative.	
2. *Determine target value for* L_v			
	Find $aveCO_{max}$ and $maxCO_{max}$ between 250 Hz and 8 kHz.		Average and maximum levels of CO_{max}.
	Calculate $A = maxCO_{max} - aveCO_{max}$.		See Note.
$A < 8$ dB.		$L_v = -7$ dB.	
$A \geq 8, 10, 12$ dB.		$L_v = -8, -9, -10$ dB.	
3. *Determine basic equalization* $EQ1_{main}$ *and* $EQ1_{downfill}$			
	Calculate $EQ_{req} = L_v - CO_{max}$.	Depending on which is higher, apply to main or downfill.	Finds basic equalization.
	Calculate $B = abs(MO_{max} - DO_{max})$.		abs(β) is the absolute value of β.
$B < 5$ dB.	Apply EQ_{req} to lower of main or downfill.		
$5 < B < 15$ dB.	Apply $W_t * EQ_{req}$ to lower of main or downfill.	Find W_t from table.	
$B > 15$ dB.	Apply $W_{t\ min} * EQ_{req}$ to lower of main or downfill.	$W_t = W_{t\ min}$.	Minimum weighting.

4. *Detection and handling of cancellations*

$CO_{max} < MO_{max}$ and/or $CO_{max} < DO_{max}$.		A cancellation is present.	
$f < 400$ Hz and $A_2 > 5$ dB.		Ignore cancellation	Not likely to be upset by small path-length changes.
$f < 400$ Hz and $A_2 < 5$ dB.	$EQ_{2\ main} = L_v - MO_{max}$; $EQ_{2\ downfill} = L_v - DO_{max}$.	MO_{max} and/or DO_{max} is reduced to L_v.	
$f > 400$ Hz and $A_2 < 5$ dB.	$EQ_{2\ main} = L_v - MO_{max}$; $EQ_{2\ downfill} = L_v - DO_{max}$.	Reduced MO_{max} and/or DO_{max} to L_v.	
$f > 400$ Hz and $5 < A_2 < 10$ dB and MO_{max} or $DO_{max} > L_v$.	Calculate $EQ_{2\ main/downfill}$ and apply to higher signal; apply $W_t * EQ_2$ to lower signal.	Reduce MO_{max} and/or DO_{max} to L_v and find W_t from look-up table.	
$f > 400$ Hz and $A_2 > 10$ dB.		Ignore cancellation	Not likely to be upset due to level difference.

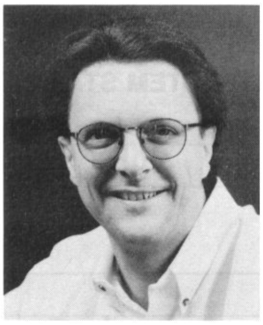
5. *Determine final equalization*

		Combine $EQ_{1\ main}$ with $EQ_{2\ main}$ to form EQ_{main}. Repeat for downfill.
		Combine 1/12-octave equalizations into 1/6-octave bands by taking the larger of adjacent 1/12-octave bands.

* $CC(f)$—combined system closed-loop amplitude response; $CO(f)$—combined system open-loop amplitude response; $MO(f)$—main system open-loop amplitude response; $DO(f)$—downfill system open-loop amplitude response.

Note: Equalization is a compromise between removing useful energy and providing stability. The higher the value of A, the lower the number of frequencies that are possibly in regeneration, and additional stability without significant loss of audible energy can be gained by lowering the target gain and equalizing only a few frequencies.

THE AUTHORS



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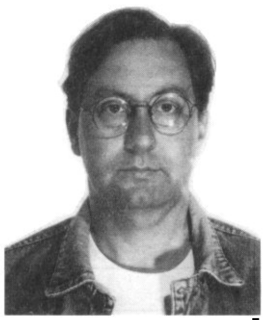

D. Connor

Glenn Leembruggen was born in Sydney, Australia, in 1955, and received a B.E. from Sydney University in electrical engineering in 1977. After six years of designing audio test instrumentation for AWA, he co-founded Elecoustics Pty Ltd in 1983, and is its director and chief engineer. Elecoustics are consulting engineers, specializing in the design of loudspeakers, sound systems, and acoustics related to speech and music. In 1993 and 1995, the company received Achievement Awards from the AES for their sound system designs for the upper and lower Houses in the Australian Parliament and the High Court of Australia. On three recent occasions, Elecoustics-designed hi-fi loudspeakers won Loudspeaker of the Year awards from the Australian body CESA.

Mr. Leembruggen is a member of the AES and the Australian Acoustical Society and is currently lecturing at Sydney University in loudspeaker design. He also serves on the audio engineering subcommittee of Standards Australia. Among his interests are playing saxophone and clarinet and local issues of conservation and planning.

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David Connor has worked in the audio industry for over 20 years. After studying electrical engineering and science at university, he commenced working in live theater and music production, coordinating and engineering musical productions for audience, live television, radio, and record. In 1983 he founded Elecoustics with Glenn Leembruggen, bringing the discipline of electro-acoustics to Australian commerical engineering, gaining awards for their loudspeaker and systems design.

While with Elecoustics, Mr. Connor also achieved Australian and international recognition as a freelance recording engineer and producer. He is currently audio operations manager for the Sydney Opera House, and is responsible for management of staff and equipment resources, and design and implementation of sound and vision systems.