RAISING THE TONE OF THE DEBATE: SOUND REINFORCEMENT SYSTEMS FOR THE NORTHERN TERRITORY AND NEW ZEALAND PARLIAMENTS.

D Gilfillan ICE Design Australia, Gilfillan Soundwork
G Leembruggen ICE Design, Acoustic Directions, University of Sydney Australia

1 INTRODUCTION

Parliamentary debates and debating chambers place strong demands on audio systems. These demands include a very high dynamic range, high gain before feedback, and high audio quality for transcription and broadcast of both speeches and interjections, as well as excellent intelligibility for the parliamentary members and the public galleries.

Sometimes these demands are contradictory, such as the need to provide a high level of reinforcement as well as “dry” recorded sound.

In 2008, ICE Design undertook a design and commissioning project to provide such a system for the New Zealand Parliament in Wellington, New Zealand. In 2010/11, a similar system was implemented in Darwin in the Northern Territory of Australia. The installed systems overcame the challenges posed and confirmed the conceptual approach.

This paper outlines how the demands of parliamentary chamber systems were met using multi-channel steered loudspeaker arrays for sound reinforcement combined with innovative design and commissioning techniques. The paper focusses on the sound reinforcement aspect of the project.

2 REQUIREMENTS FOR PARLIAMENTARY SOUND REINFORCEMENT SYSTEMS

The Westminster parliamentary system involves opposing political parties debating topics of importance. Special buildings have been built to house debating chambers which provide a forum for these events.

These buildings were mostly built prior to the availability of sophisticated audio-system technology and equipment. Most parliaments have therefore operated for some time without speech reinforcement for parliamentary members.

The days of acoustic oration seem to have largely passed and now speech reinforcement systems are required to provide high levels of speech intelligibility to parliamentary participants (members), the public, transcribers and broadcasters.

The speech reinforcement systems are required to produce the following outcomes:

- Excellent temporal and frequency response to maximise the intelligibility on the chamber floor and the public galleries
- “Grand” sound as befits the importance of a Parliament
- Satisfactory consistency of sound level both on the chamber floor and the public galleries
- Adequate sound pressure levels to overcome shouting and loud interjections from opposing sides of parliament.

- Sufficient gain before feedback to achieve the required Sound Pressure Level
- High reliability
- Effective recording of members’ speeches and interjections
- The ability to maintain the primacy of the Speaker of the House, whose job it is to enforce the rules of parliamentary conduct
- An element of localisation aligning the reinforced speech with the person talking
- “Foldback” and reinforcement of speech to those in the immediate vicinity of the talker
- Simple, effective control and monitoring of the system

Some of these requirements are contradictory and need to be treated carefully, such as the requirement for high sound pressure levels inside the chamber and the maintenance of a high quality broadcast feed. High sound levels inside the chamber cause a more “roomy” sound that is unpalatable to broadcasters.

2.1 Parliamentary system challenges

There are four primary challenges in achieving high quality speech reinforcement systems in Westminster-style parliamentary chambers:

a) Difficult room-acoustic conditions for good intelligibility.

b) The potential for very high background noise levels

c) Aesthetic and heritage constraints

d) “Theatre-in-the-round” type coverage areas for most people speaking

2.1.1 Room Acoustic conditions

Parliamentary debating chambers have very high historical, cultural and political importance. Some chambers were not designed with the inclusion of loudspeakers in mind. Many have not given sufficient thought to the inclusion of audio systems and therefore have limped forward with substandard systems.

In the New Zealand Parliament, for example, architectural details such as large heavy drapes which can be seen in early photographs have been removed from the chamber in order to satisfy a modern aesthetic. While these drapes may or may not have been part of the original architectural design, the current space, with its close and hard side walls and high ceiling, is reasonably hostile environment for a speech reinforcement system.

The addition of extensive public galleries that surround many chambers provides another challenge for the provision of intelligible speech to members. Degradation of the early to late energy ratio is often caused by the addition of numerous, delayed loudspeakers to the reverberant energy in the room.

2.1.2 The potential for very high background noise levels

Parliaments sometimes encourage robust and sometimes antagonistic debate between parties. At times, the level of shouting, interjections and general hubbub rises to a very high level.

To provide intelligible speech to a listener who has another shouting near him/her must be capable of providing concert system levels. To achieve these levels with a talker-to-microphone distance of 300 mm would indeed be marvelous, but is not possible given the constraints of loudspeaker size and requirements for foldback and local coverage which determine the gain margin of the system. However, the system must be able to provide intelligible reinforcement for the majority of debates and speeches, including all but the loudest interjections.

It could be argued that the provision of a high quality, high level reinforcement system simply encourages louder interjections, since members are aware that in order to be heard they are required to shout more loudly than they otherwise might. However, the benefits of such a system: namely high intelligibility under 99% of circumstances, far outweigh these unproven negative effects.

3 REINFORCEMENT SYSTEM CRITERIA: INTELLIGIBILITY

In a mission-critical speech reinforcement system, speech intelligibility can never be too high. However, given the acoustic challenges presented by debating chambers, the following parameters provide guidelines for appropriate system performance. If better results can be achieved, they should not be discarded on the basis of cost as the human dynamics of the parliamentary debates always produce occasions in which the intelligibility is inadequate, due to noisy behaviour of members.

3.1 Early to Late Arriving Energy (C_{50})

Highest intelligibility is achieved with a C_{50} clarity ratio that is high as possible between 125 Hz and 8000 Hz, within the constraints of architecture and coverage.

In the case of both New Zealand Parliament and Northern Territory Parliament, targets were above 2 dB for all octave bands from 200 Hz to 8000 Hz.

While this is relatively easy to achieve at high frequencies, a number of factors combine to make it a challenging target at lower frequencies.

The difficulties arise from:

a) Spill of the direct field of multiple gallery loudspeaker spill into the main chamber

b) Delayed reverberant energy from gallery loudspeakers leaking into the main chamber

c) Size constraints of the loudspeakers limiting loudspeaker directivity at lower frequencies

3.2 Bandwidth and Frequency Response

The target bandwidth for all users on the chamber floor was at least 90 Hz to 13 kHz or greater. For optimal subjective speech intelligibility, systems should not be band limited above 11 kHz frequencies, even though these frequencies will make no difference to the measured STI.

Frequency responses for listeners should be nominally flat, allowing a response window of approximately 5 dB, when viewed at 1/12th octave. The window at higher frequencies should be closer to 3dB if possible. Up to 300 milliseconds of impulse response data should be used to gather the frequency response information, to allow for the reverberant build-up of energy at low frequencies.

3.3 Sound Pressure Level

A crest factor of 18 dB is necessary for speech to ensure that limiting or compression artefacts, which may damage intelligibility, are not present. Noting that interjection noise in the chamber may reach 95LA_{eq}., it is unrealistic to suggest that a speech reinforcement system should be able to overcome this level of noise to provide intelligibility under these circumstances. The target SPL with speech is therefore set to 88LAeq (with 18 dB crest factor).

In reality, the systems installed in New Zealand and Northern Territory Parliaments were able to reach higher SPLs. However, the talker's speaking level at the microphone and the available gain-before-feedback with the talker's distance to the microphone can limit the available SPL.

3.4 Equivalent Acoustic Distance

Equivalent acoustic distance, as described by Davis and Davis[1], calculates the increased SPL at a listener that is achieved by the insertion of a sound reinforcement system into a talker-listener transmission chain. The measure is expressed as a talker-listener distance that would be required to produce the equivalent unaided SPL. It in part indicates gain before feedback.

The target EAD was less than 1.8 metres for all listeners, allowing for a mouth to microphone distance of 400 mm including a feedback stability margin of 6 dB.

3.5 Speech Transmission Index

An appropriate STI was considered to be in excess of 0.65, for electronic system excitation with effectively no noise included in the calculation. This equated to about 0.6 (from experience) when the excitation signal is generated by a NTI Talkbox facing a microphone.

The limitations of STI with regard to frequency response and echo blindness were understood.

4 SYSTEM IMPLEMENTATION

The New Zealand and Northern Territory Parliament audio systems were implemented using similar philosophies. Calculations of statistical STI using the method of Steeneken and Houtgast and modelling using the Aura module in EASE indicated that precise control of loudspeaker polar pattern was necessary at frequencies as low as 200Hz, or lower if possible in order to achieve appropriately high intelligibility.

Microphone use was also paramount to the correct functioning of the system, and therefore fixed, rather than flexible gooseneck extensions were chosen, to reduce the chance of members speaking away from a microphone.

Aesthetic reasons required the loudspeakers to be mounted at a height of 6 m above the floor, and at this height, the acoustic time of flight introduces gap of approximately 20 ms between the amplified and unamplified voice. To ensure that this gap is not elongated, the latency of the DSP device must be ultra-low. The lowest latencies of current state of the art DSP devices with "drag and drop" configuration are of the order of 2.5 ms.

As the DSP device is also used for beam steering of the loudspeakers arrays, it is essential that sample-accurate delay equalisation is provided at all output stages in the DSP.

Another vitally important element for Hansard recording is the ability of automatic mixers to provide accurate sensing of the active microphone(s) and hold all other microphones off.

4.1 Main Loudspeakers

4.1.1 Description

A steered array solution directs the least amount of sound to the ceiling and side walls of space, while covering the audience area effectively. It also allows for a vertical hang and minimises the required size of the array for effective low-mid band pattern control.

We were unable to find any commercial loudspeaker that would fulfil all the acoustic and aesthetic requirements, and it was therefore necessary to design a bespoke loudspeaker. Current steered array loudspeakers on the market attempt to steer frequencies above 3000 Hz but are not particularly successful at doing so due to the small inter-driver spacing required to achieve this. This basic limitation suggests the use of a hybrid array[2] using a physically tilted horn with compression drivers for high frequencies.

Experience with the implementation of bespoke steered array designs in courtrooms and the Australian Parliament gave us confidence that we could model this loudspeaker accurately and make it work according to our predictions.

Three line-arrays of drivers are integrated into a loudspeaker cluster, together with high frequency horns and a down-fill driver:

a) Bass array: Twelve 160 mm diameter drivers, connected in pairs and fed by six amplifier channels, and a implemented as logarithmically spaced, electrically tapered, steered array cover the range from 90Hz to 500Hz.

b) Mid-range array: Twelve 80mm diameter drivers fed by seven amplifier channels and implemented as a logarithmically spaced, electrically tapered, steered array, cover the range from 500 Hz and 1300 Hz. Spatial aliasing due to driver size limits the frequency range of the steering for this array to 1500 Hz.

c) High-mid-range array: Five 45 mm diameter drivers fed by five channels spaced, and implemented as a spaced, electrically tapered, steered array covered frequencies between 1300 Hz and 3000 Hz and fill the gap between the upper limit of steering for the mid-range array and the lower limit of effective operation of the horn elements,.

In order to provide mid and high frequency coverage of the region directly below the array, a 200 mm driver with co-axially mounted horn was located on the underside of the cluster housing. Low frequency coverage for these areas was effectively provided by the wider radiation pattern of the low frequency array.

The following figures show the loudspeaker used to cover the chamber floor of the NZ Parliament, which came to be known as the “Kiwi-tube”.

Figure 1: "Kiwi-tube" Multi-channel cluster of tapered arrays, horns and down-fill

To fit the horns and their compression drivers into the relatively small diameter tube would have resulted in strongly angular in-fill pieces between the horns with odd shapes. To avoid the difficulties with design and implementation of these infill pieces and to mitigate the effects of diffraction of the horn edges on radiation pattern and frequency response, high performance polyester insulation was used fill and smooth the spaces between the horns.

Figure 2: "Kiwi-tube" in situ with perforated metal shroud

Figure 3: Six "Kiwi-tubes" in situ at 6 metres height

4.1.2 Predicted polar performance

The design of the cluster was based on two fundamental requirements:

- Provision of a flat frequency response to every listener
- Minimisation of upward sound radiation

To achieve these requirements, the design process commenced with optimisation of the coverage and frequency response consistency of the high frequency horns with EASE software. The averaged vertical polar plot of these horns then provided the target vertical polar pattern for the arrays in the region up to where the down-fill high frequency radiation pattern becomes dominant. The beam-forming parameters of all arrays were then adjusted until their pattern matched the target pattern and the upward radiation was minimized.

The predicted polar patters are shown in the plots below. It is noted that the target is indicated at 870 Hz. This target represents the averaged polar pattern of the horns below -10 degrees.

The radiation balloons were calculated and imported into EASE to allow confirmation of the frequency response and coverage of the final cluster.

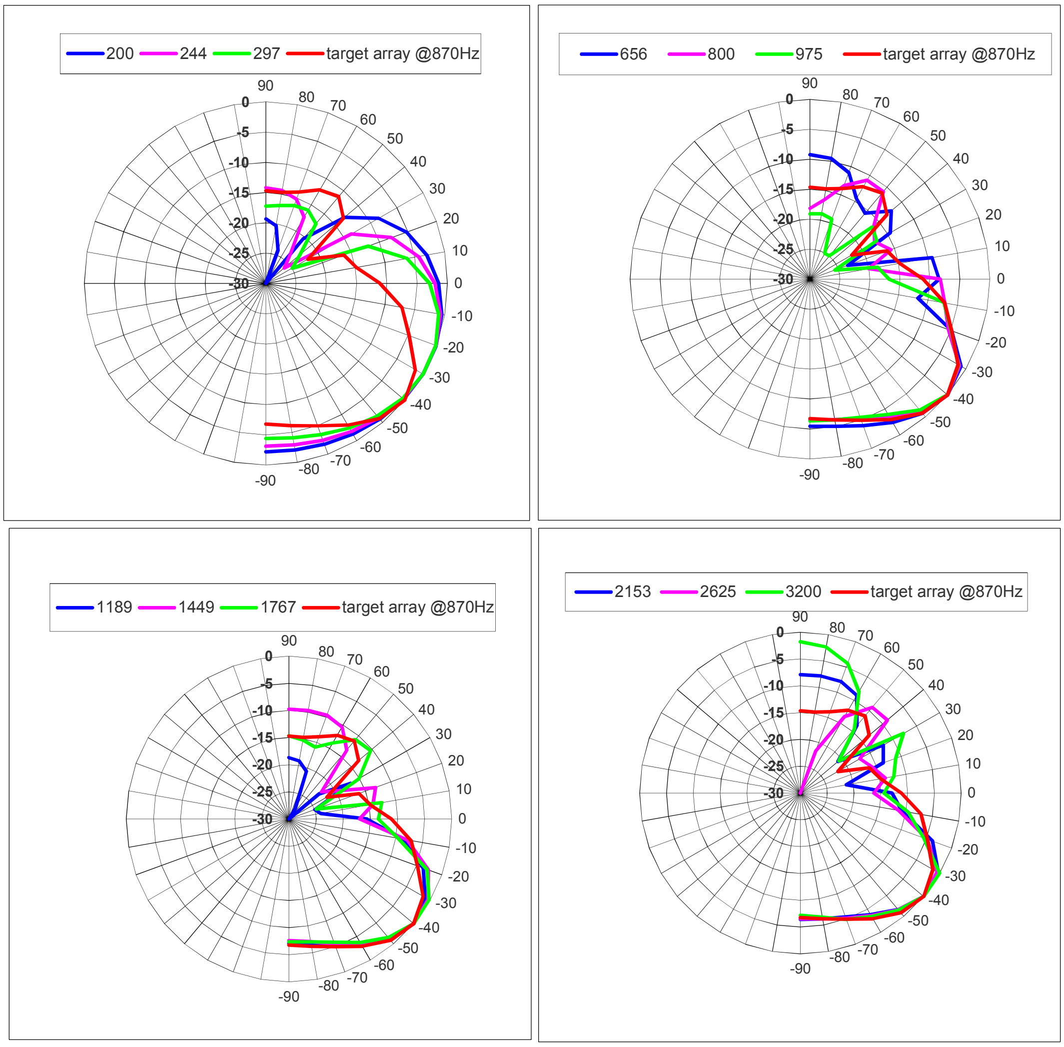


Figure 4: Predicted vertical polar plots, 30 degree angle of steer at 10 metres distance, various frequencies indicated

4.1.3 Gallery Loudspeakers

Different approaches to the implementation of sound reinforcement in the public galleries were required in the two chambers.

In New Zealand, the public galleries are narrow with low ceilings and bulkheads. Advantage was taken of the acoustic shielding provided by the balcony front and the ceiling bulkheads by using numerous small loudspeakers fixed behind these structures.

In the Northern Territory, the public galleries are open and had very high ceilings. Larger steered array loudspeakers mounted in the ceiling were considered and proven by modelling to produce less spill into the main chamber than large numbers of small loudspeakers located behind the backs of seats in the galleries.

In both cases, the build-up of sound from gallery loudspeakers impacted negatively on the intelligibility in the main body of the chamber.

4.1.3.1 NEW ZEALAND PARLIAMENT

In the New Zealand chamber, choices of loudspeaker positions for effective gallery coverage and shielding were limited, and therefore a number of small loudspeakers were developed to fit available positions.

Some of the gallery loudspeakers that were developed are shown below:

Figure 5: Balustrade Type Loudspeaker

Figure 6: Cornice Type Loudspeaker taking advantage of bulkhead shielding

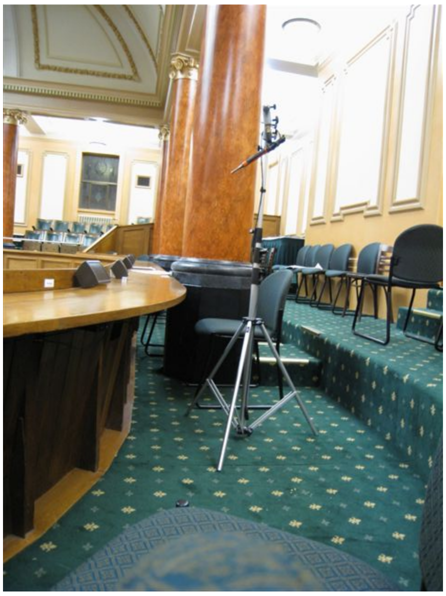


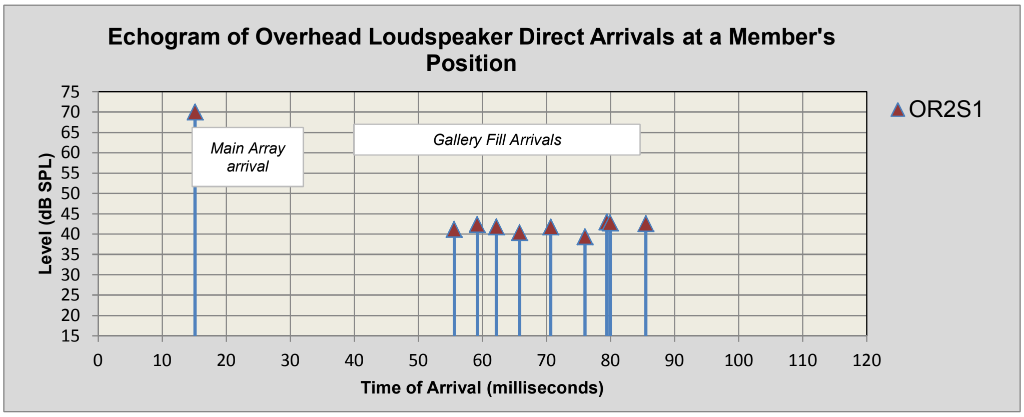
Figure 7: Press Gallery Type Loudspeakers. Note that shield was no available here

Figure 8: "Tube" type loudspeakers covering the open rear “Speaker’s Gallery”

4.1.4 Northern Territory Parliament

Comparisons were made between an over-head steered-array system and small seat-back mounted loudspeakers for gallery sound reinforcement in order to determine which system was less likely to degrade audio in the chamber.

The following echograms compare the relative sound spill of a ceiling-mounted steered-array and seatback loudspeakers at a listener on the chamber floor. These echograms indicate the relative levels and arrival times of Gallery loudspeakers on the main chamber floor. Perceptually, the gallery loudspeakers would have produced significantly greater degradation of the sound on the main chamber floor than those in New Zealand Parliament.



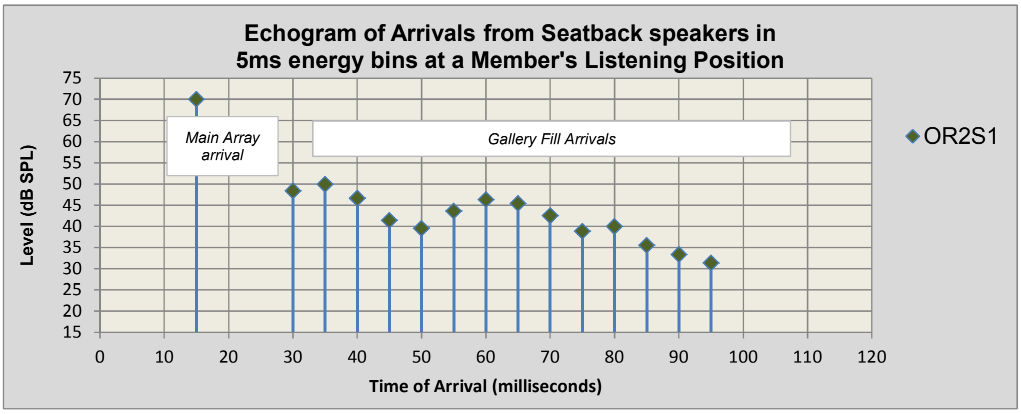


Figure 9: Echogram predictions of Gallery Loudspeaker arrival times and levels

4.2 Microphones

4.2.1 Microphone requirements

Microphone selection was critical to the success of the system. Well behaved microphones are important to gain-before-feedback (therefore EAD), tonal quality and intelligibility. A number of microphone features were tested in order to inform the selection. Parameters included in the testing were:

a) Frequency response on axis

The response required is flat on axis at 400mm between 100 Hz and 12000 Hz.

b) Frequency response off axis (polar response) and rejection off axis

Polar response (relative to on axis) and off axis response are critical parameters and uniform attenuation with frequency is paramount.

c) Noise Floor

In large spaces with background noise the noise floor of a microphone is usually inaudible to listeners. However in smaller rooms and in broadcast recordings, microphone self-noise must be low.

d) Pop filter efficacy

Microphones were tested for response with and without their standard pop filter, and the efficacy of the filter to stop vocal popping was tested via listening.

e) Listening tests

Critical listening was carried out with a member of the New Zealand Parliament broadcast team to compare the short-listed microphones. This process helped dispel myths surrounding the need for very expensive microphones.

4.2.2 Microphone Selection

From the above tests, a suitable microphone was selected which fulfilled these requirements, with manufacturer agreeing to make customised goosenecks for the purpose.

One important conclusion that resulted is that competing gooseneck microphones with similar cost have vastly different performance characteristics, with some having much worse performance than that indicated by their data sheets.

Examples of a non-compliant microphone are shown below:

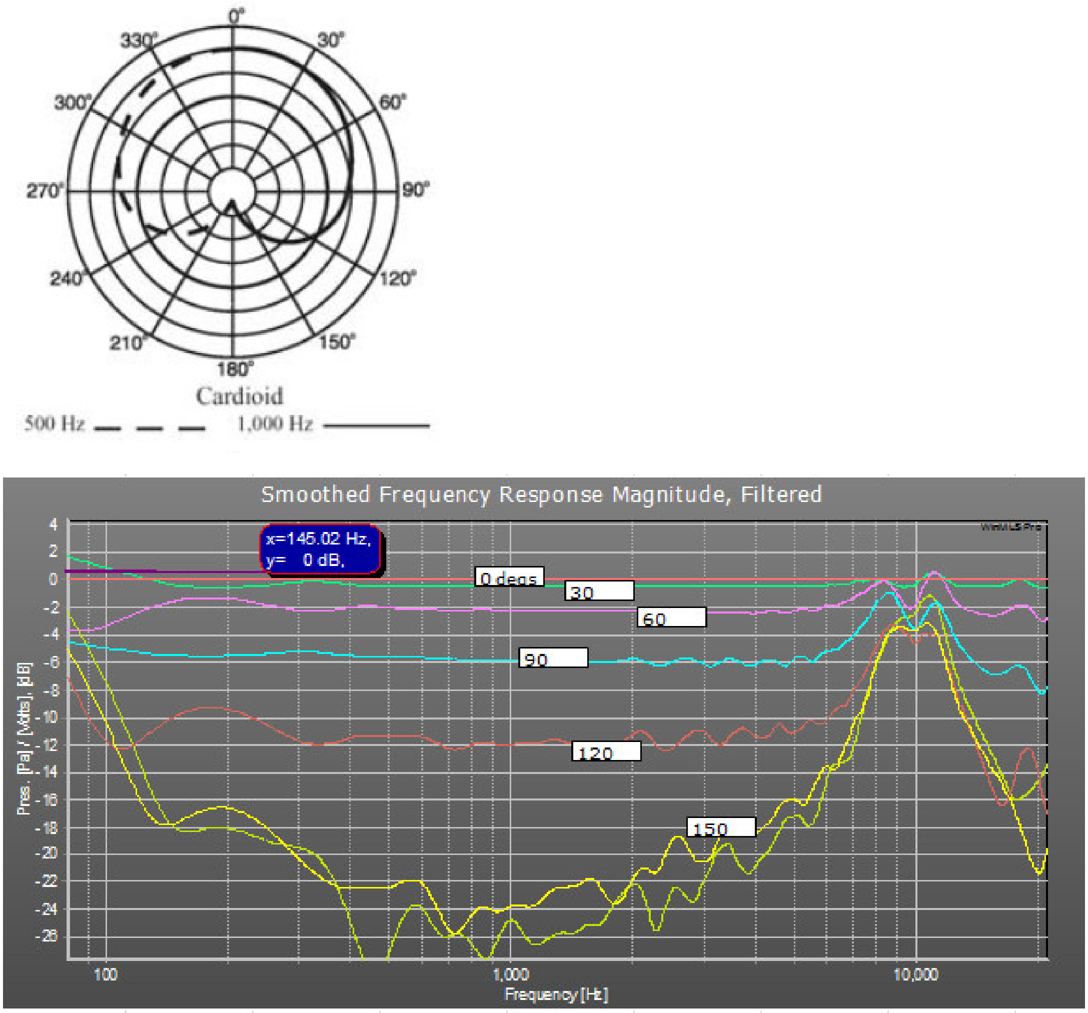


Figure 10: Example 1 - Manufacturer’s published polar pattern and normalised off-axis responses of a non-compliant microphone. The angles of incidence are noted.

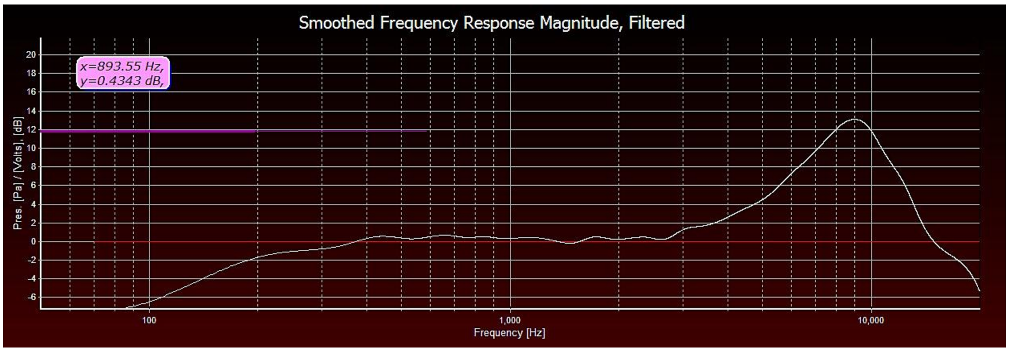


Figure 11: Example 2- Non compliant microphone frequency response on axis

5 MEASUREMENT RESULTS

5.1.1 Microphone on axis responses

All microphones were measured on axis in order to confirm their functionality. The overall gain window allowable was +/- 1.5dB, which appears to be an achievable level window. Below is a sample of on axis frequency responses of microphones in Northern Territory Parliament, measurement limited to 10 kHz as a limitation of the NTI Talkbox. Some proximity effect is evident in the increased amount of low-mid response due to the measurement distance of 300 mm. The ripples in the response are due to the rectangular window used to truncate the impulse response before the arrival of the first reflection.

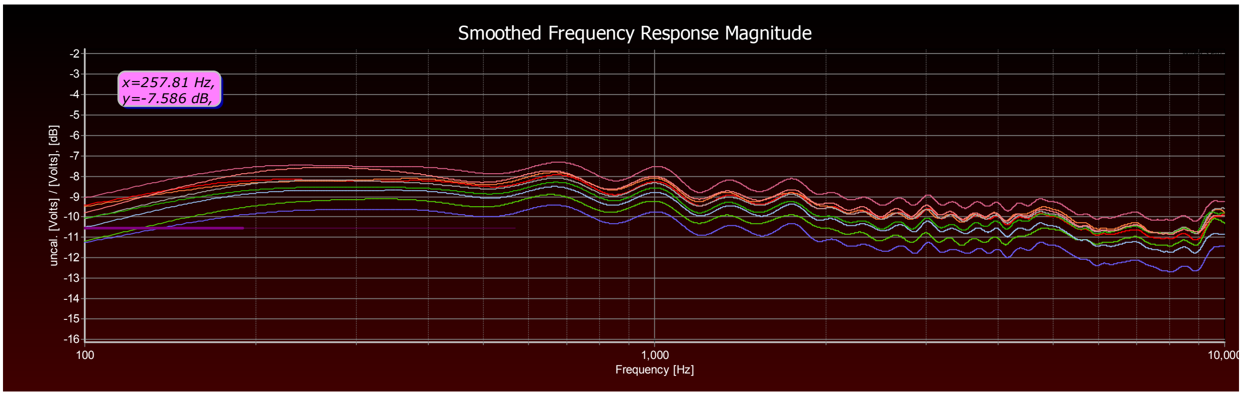


Figure 12: Sample of on-axis microphone responses at Northern Territory Parliament

5.2 Acoustic Measurement results

5.2.1 Loudspeaker integration

A rigorous testing regime was required to allow the correct functioning of the complex loudspeaker clusters. Drivers were tested in the near field before steering algorithms were inserted and tested.

Figure 13 shows the 4 elements of the main steered cluster as individual elements in transfer function.

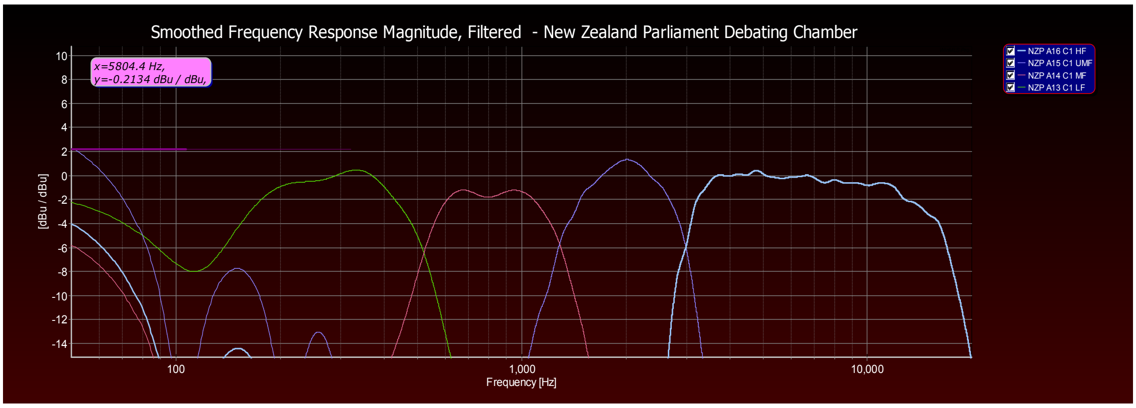


Figure 13: Low, Low-mid, high-mid and high frequency elements in "Kiwi-tube" cluster, during integration phase. The out-of-band spectral leakage due to truncation of the impulses should be ignored.

Our first approach to integration of the down-fill element with the main array using an iterative process of acoustic measurements at listening positions proved unsuccessful.

A method was then developed using measurements at a number of positions of the complex transfer functions of both the main array and down-fill elements. The amplitude and phase data was exported and manipulated using a combination of delay, level, and high-pass and all-pass filters to produce the best compromise between frequency responses at all the measured positions.

The following plot shows the equalisation magnitude applied to the down-fill element in order to achieve the average response across the listening positions, which is also shown. The equalisation process included three 2nd order all pass filters applied to the down-fill element.

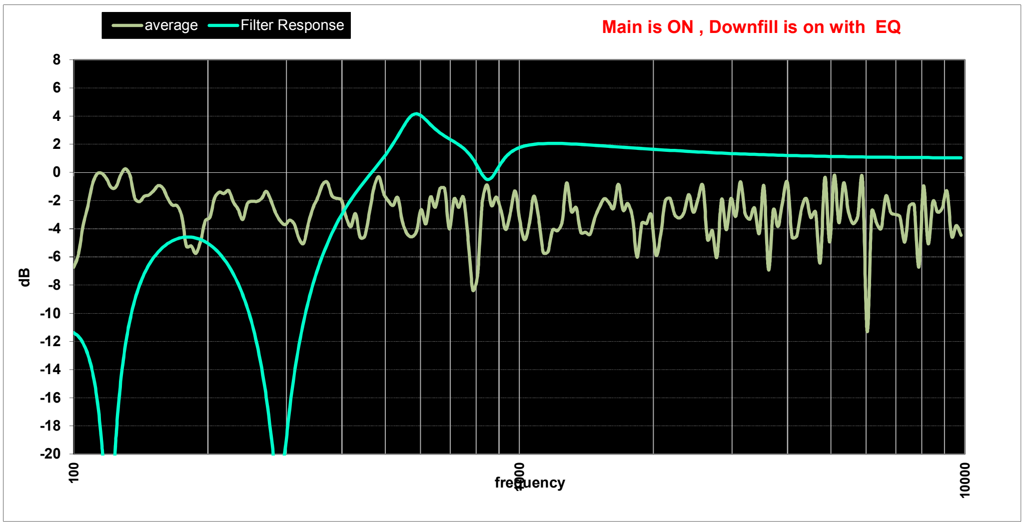


Figure 14: Average response of 8 positions under array with applied equalisation

The following figures show the frequency response on the axis of steering of a loudspeaker cluster in the two parliaments. Low frequency response is limited by a window length of about 6 ms in the Northern Territory response.

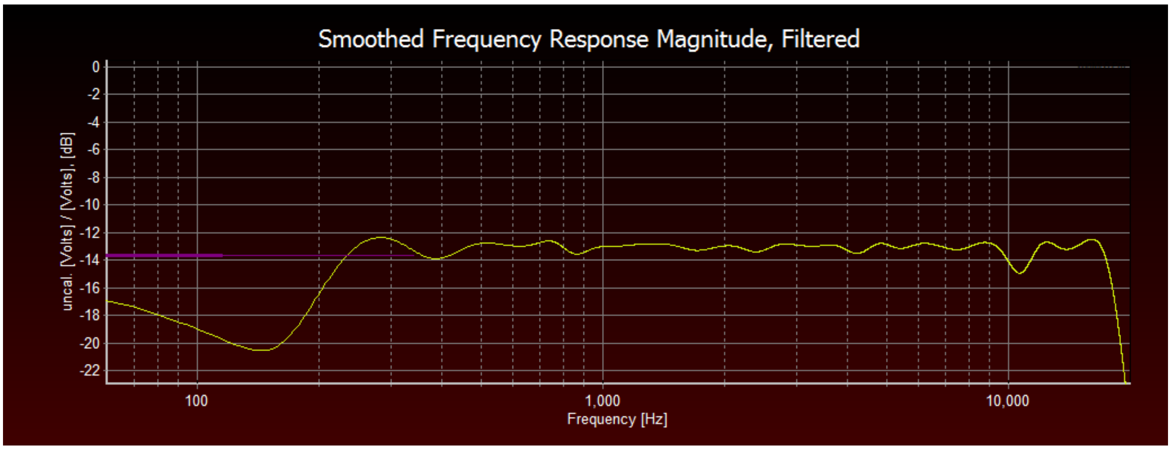


Figure 15: On-axis response of loudspeaker cluster with 6ms time window

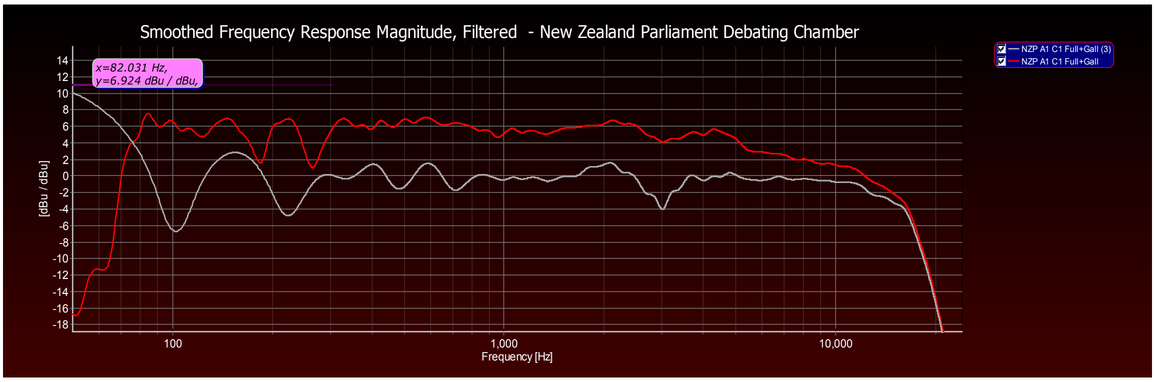


Figure 16: On axis cluster response in New Zealand Parliament, with 10 ms and 1000 ms time windows.

Note that in
Figure 16, there is a large build-up of energy after the initial arrival. In general, our process would aim for a flat response with 300 ms of time data and a Hanning window.

5.2.2 Equalisation, Spectral Balancing and Spatial Averaging

Following the testing and insertion of parameters into the clusters, equalisation was achieved through a process of spatial averaging followed by critical listening. Multiple listening positions within the loudspeaker's coverage area were measured using a swept sine-wave excitation. The responses were derived using a long (300 ms) time window with narrow band smoothing (1/15th octave).

In order to ensure good frequency response at each listener rather than have the loudest responses dominate the average, the results were normalised in level: the average level between 400 Hz and 4000 Hz was used for each measurement and set to 0 dB. The average response was compared to each measurement to ensure that single measurements were not dominating the average response.

The average of these normalised responses was then taken, and exported to a spread-sheet. This spread-sheet contained simulations of the filters contained in the signal processing units, and these filters were used to flatten the average response, as shown in the following figure. Given even high frequency coverage with a 3 dB window, flat average response provides the best basis to commence critical listening.

Critical listening commenced with listening to known musical tracks and progressed through other material including pre-recorded anechoic speech and less well recorded speech.

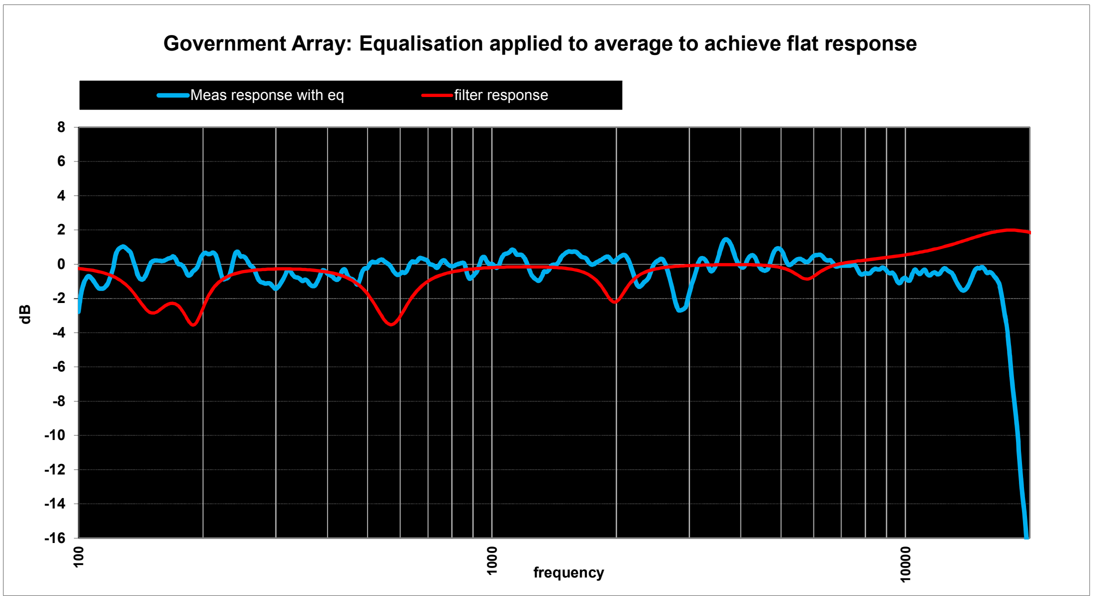


Figure 17: Filters applied to the average response (red), and the resultant average response (blue)

Open Loop Gain Response

A sound reinforcement system is a closed feedback loop, with some frequencies in regeneration and others in degeneration. Loop gain describes the maximum stable amplification that can be achieved by a microphone in a system. Measurement of loop response: the response of a talker's microphone in situ using the sound system as the source, helps find narrow band frequencies that are prone to feedback.

We used a process of gain maximisation within minimal tonal impact that is a refinement of the method described by Leembruggen and Connor.[3]

With the microphone activated at a level where feedback is just commencing, we measured the response of the microphone under test in both the open-loop (connection between microphone and loudspeaker open) state and the closed-loop state (connection made). A person stood in front of the microphone during testing to simulate the effects of a person using the microphone during normal operation. The gain margin before sustained feedback used was approximately 3dB.

The following plots show a sample of open and closed loop responses.

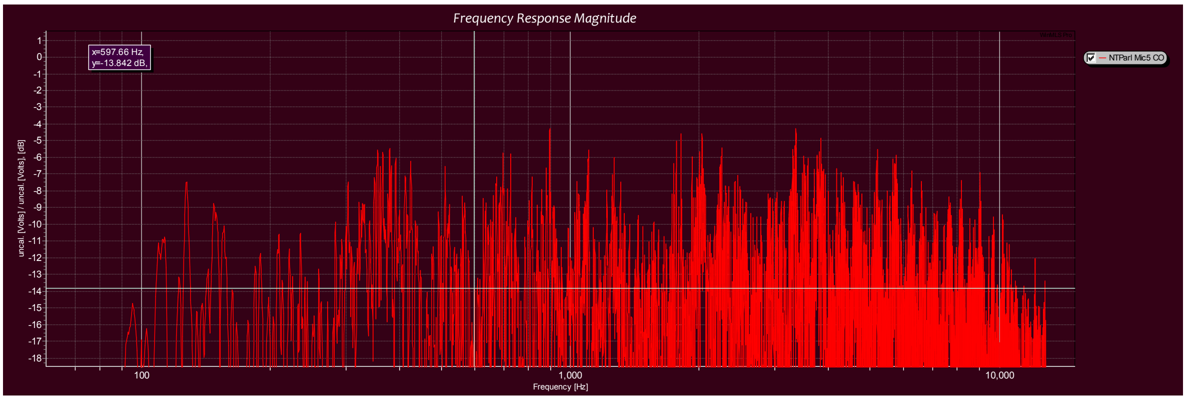


Figure 18: Open loop microphone response

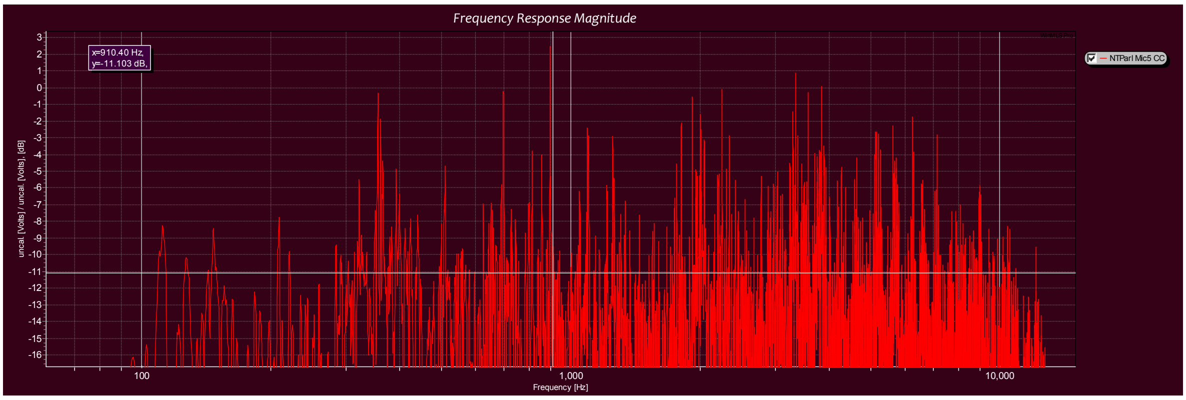


Figure 19: Closed loop response

Examination of the open loop response (magnitude and phase) indicates those frequencies which will regenerate or degenerate. Comparison of the closed loop and open loop magnitude responses shows the frequencies that are regenerating due to peaks in the open-loop response and the open-loop phase being closed to 0 degrees. The figures below show the comparison of the exported responses and the resultant equalisation applied to maximise the gain before feedback with minimal tonal colouration.

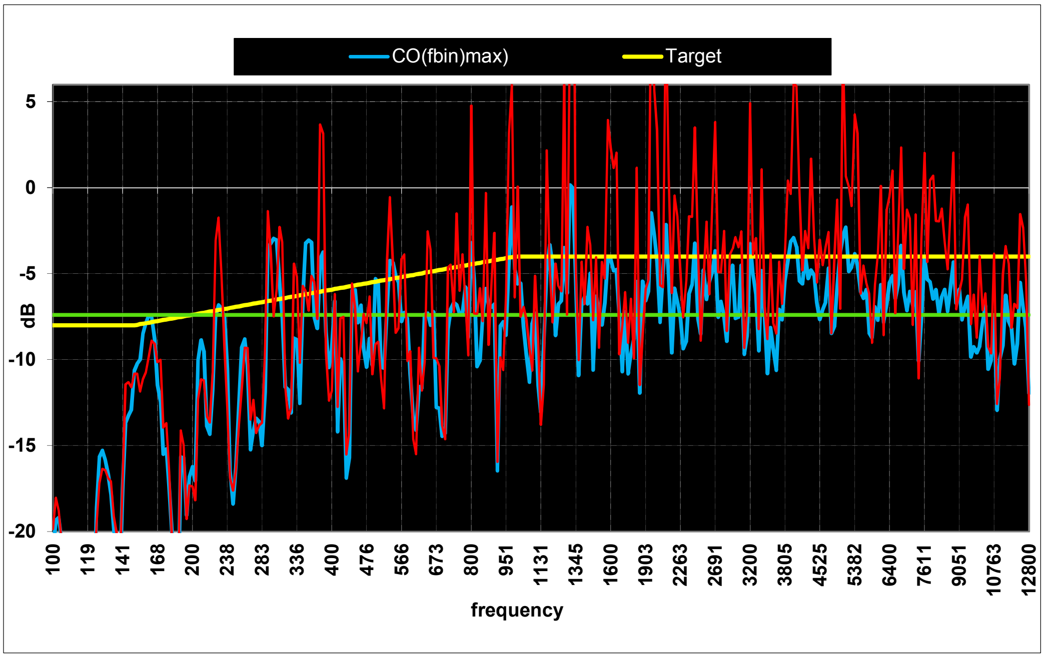


Figure 20: Example closed-loop response (red) and open-loop response (blue)

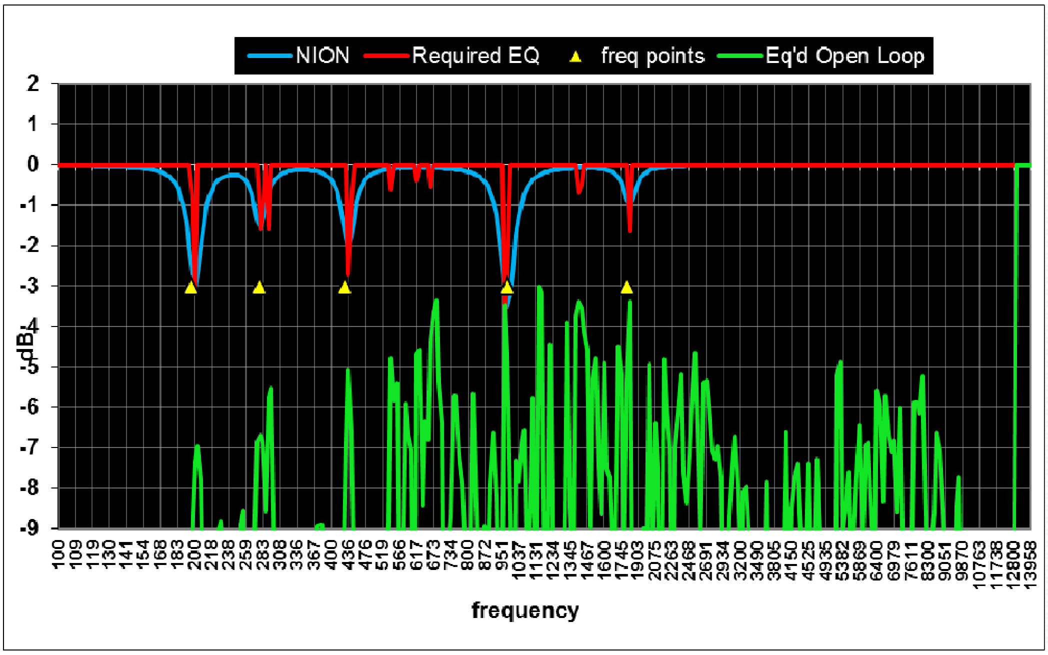


Figure 21: Equalisation applied to microphone (blue) and the resultant response (green)

6 INTELLIGIBILITY MEASURES

6.1 Early to late arriving sound ratio (C_{50})

The following tables show average C_{50} values for a number of listening positions in the Northern Territory and New Zealand Parliament Chambers. All measurements were at member's positions on the floor of the chamber. Note, despite the efforts to isolate the gallery sound from the chamber floor, the impact of the gallery loudspeakers is relatively high in both cases. Perceptually, however, the impact was far more marked in the New Zealand chamber.

As the New Zealand chamber has mid-band RTs in the order of 1.8 seconds, it presents a more difficult environment than the Northern Territory chamber; which mid-band RTs of 1.2 seconds, and in this context, we were surprised by the higher C_{50} ratios of the New Zealand system at 500 Hz and below. However, the New Zealand chamber is narrower than the Northern Territory, and therefore more early reflections will contained within the 50 millisecond period.

NTP C50	125	250	500	1000	2000	4000	8000	Ave 250-8k
Average C50 no Galleries	-0.65	-0.38	0.52	3.98	4.98	5.80	9.75	4.11
Average C50 with Galleries	-0.45	-1.35	-0.47	1.65	2.83	4.07	7.47	2.37

NZP C50	125	250	500	1000	2000	4000	8000	Ave 250-8k
AVERAGE C50 no Galleries	1.08	2.42	1.68	2.24	2.28	3.54	7.12	3.21
AVERAGE C50 with Galleries	0.72	1.60	0.22	-0.02	0.38	1.74	5.18	1.52

Figure 22: Average C_{50} results in New Zealand and Northern Territory debating chambers.

6.2 Sound Pressure Level

Adequate Sound Pressure level was available from the loudspeaker system. Levels in excel of 88LZea for speech were measured.

6.3 Equivalent Acoustic Distance

Equivalent Acoustic Distance was calculated at 3 listening positions at Northern Territory Parliament. The source was a Talkbox using a speech track of multiple voices at the Speaker of the House's microphone at a distance of 400mm. Results were 2.1 m, 2.0 m and 2.1 m at listening positions 14, 3 and 19 respectively.

6.4 Speech Transmission Index

The following tables show the measured STI in the New Zealand and Northern Territory parliaments using two methods of measurement.

Method 1 injected the excitation signal electronically into a microphone point and measured the result at listener positions. Method 2 used a simulated talker (NTI Talkbox) in front of a microphone as the excitation signal and measured the result at listening points. Method yielded STI results on average 0.01 and 0.02 lower that the direct electronic injection.

The effect of the gallery system in intelligibility in the main chamber can be easily seen here, with a reduction of 0.04 in STI when the gallery system was switched on in New Zealand. Similarly in Northern Territory, the reduction in STI with the addition of the gallery systems was 0.04. However, the impact was markedly less perceptually in the Northern Territory.

	125	250	500	1000	2000	4000	8000	STI
AVERAGE MTI/STI no Galleries	0.63	0.63	0.60	0.61	0.62	0.67	0.77	0.64
AVERAGE MTI/STI with Galleries	0.61	0.61	0.56	0.55	0.58	0.62	0.73	0.60

New Zealand Parliament, electronic injection

	125	250	500	1000	2000	4000	8000	STI
Average MTI/STI with Talkbox signal injection to microphones	0.61	0.59	0.55	0.54	0.56	0.60	0.71	0.58

New Zealand Parliament, "acoustic" injection to microphones

NTP STI	125	250	500	1000	2000	4000	8000	STI
AVERAGE MTI/STI no Galleries	0.60	0.59	0.62	0.70	0.71	0.73	0.84	0.70
AVERAGE MTI/STI with Galleries	0.58	0.55	0.59	0.64	0.66	0.69	0.79	0.66

Northern Territory Parliament, electronic injection

NTP STI	125	250	500	1000	2000	4000	8000	STI
Electronic Injection	0.58	0.55	0.59	0.64	0.66	0.69	0.79	0.66
Acoustic Injection	0.43	0.40	0.44	0.47	0.50	0.52	0.59	0.65

Northern Territory, difference between electronic and acoustic signal injection

Figure 23: STI measurement results

The results indicate that intelligibility was slightly higher in Northern Territory Parliament, and perceptually that is indeed the case.

7 CONCLULSIONS

The sound reinforcement system discussed represents a highly engineered solution to the difficult environments posed by parliamentary debating chambers. The systems achieve the highest possible speech intelligibility while maintaining a "grand" sound befitting a parliamentary chamber. This achievement has proven a great challenge and has necessitated the design of customised loudspeakers, attention to all minor design details and the development of very thorough and original commissioning techniques.

8 REFERENCES

1. Davis, D and Davis, C., Sound System Engineering, 2nd Ed, SAMS, Indiana, 1994.

2. Thompson, A., Leembruggen, G., Gilfillan, D. "Hybrid line arrays – a viable alternative." Proceedings of the Institute of Acoustics, presented at Reproduced sound 25, Brighton, 2009.

3. Leembruggen, G and Connor, D. Design and Commissioning of sound reinforcement systems for the Australian Parliament - A Holistic Approach. J. Audio Eng.Soc. Vol 44 No.10 1996 October.