Digital sound system modelling and design

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ABSTRACT

As part of a staged redevelopment of the Adelaide Convention Centre, a new Plenary Hall is to be constructed with a capacity of 3150 seats. The Hall can be subdivided and includes an operable theatre in the round within each sub-division. This paper outlines the acoustic performance criteria for the complex operation modes of the Plenary Hall. It provides a review of the sound system design philosophy and electro-acoustic modelling technique used to predict the acoustic performance of the sound system. It also provides an outline of evolving networking technologies used to provide a flexible digital audio network using a standard data network infrastructure.

Keywords: Modeling, prediction and Simulation I-INCE Classification of Subjects Number(s): 76

1. INTRODUCTION

The redevelopment of the existing Plenary Building of the Adelaide Convention Centre includes the design of a highly flexible Plenary Hall which operates in a number of different configurations. In order to service the various configurations a digital sound system design which offers good coverage and flexibility was required to be provided. In delivering this digital sound system acoustic modelling of the space was conducted alongside computational predictions of the sound system performance. Additionally a robust and flexible digital audio network was designed using standard data network infrastructure which allowed the users of the hall to easily send audio to any configuration of the space.

Figure 1: Architectural render of the Adelaide Convention Centre

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2. PROJECT SUMMARY

2.1 Adelaide Convention Centre

The original Adelaide Convention Centre, considered to be the first Convention Centre in Australia, was built on North Terrace in Adelaide Australia in 1987 with a renovation completed in 2001.

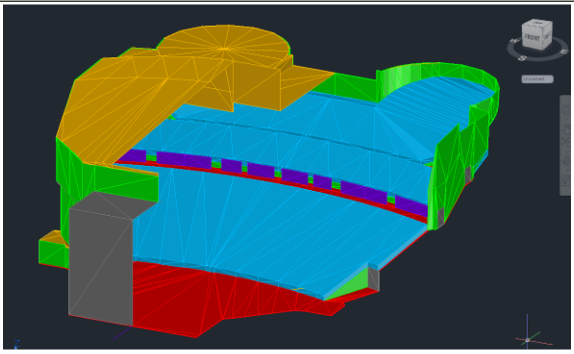
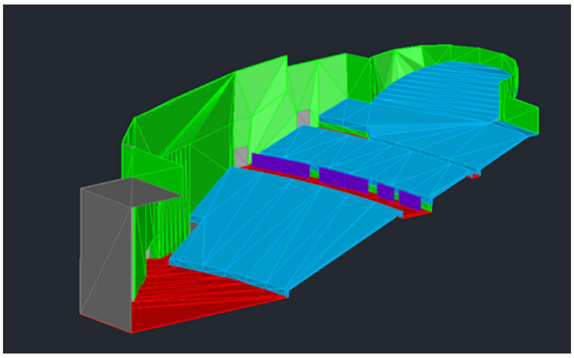
An expansion of the Adelaide Convention Centre was announced in 2011 which included an extension of the existing facility towards over the railway lines, linking in with the Morphett Street Bridge and a second phase consisting of the replacement of the existing plenary building with a multi-purpose, state-of-the-art facility with a capacity of up to 3,500 seats.

The purpose of replacing the original plenary hall is to provide the City of Adelaide with a more functional space which can operate in a highly flexible manner providing a number of different operational modes and configurations. The venue can be subdivided and configured within minutes as a pre-function space, ballroom, exhibition or plenary space. This flexibility is expected to realise the primary financial and economic benefits of the redevelopment.

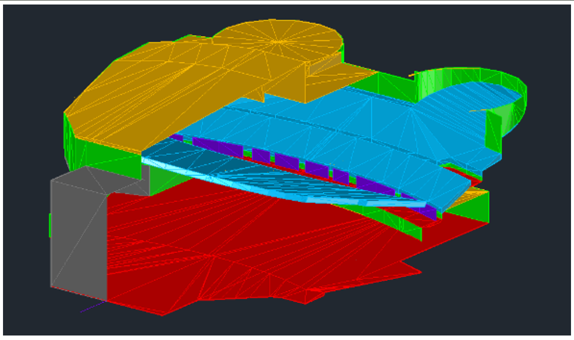
2.2 Operational Modes

The new plenary hall has been designed to operate in a number of configurations. The basic modes of operation are outlined in the following table:

Table 1 – Plenary Hall modes of operation

Full Hall Mode: This mode consists of a typical plenary hall which can cater for an audience of up to 3,150 patrons. To accommodate the maximum number of people the full hall mode includes the two drum meeting rooms at the rear of the hall, two tiers of raked seating and flat floor seating in front of the presenters' position. The hall can also operate with the circular drum meeting rooms at the rear of the space sectioned off with an operable wall to act as independent meeting spaces if the additional seating is not required.	
Half Hall Mode: This mode can operate in a similar fashion to the full hall mode, however a large operable wall is deployed through the centre of the hall to subdivide the hall into two 'half halls' which can individually operate. As with the full hall mode the circular drum meeting rooms can be removed from the space through the deployment of an operable wall. The digital sound system has been designed such that the halls can operate independently from one another	

Flat Floor Mode: For events where a flat floor is preferred (i.e. exhibitions, banquets, etc) the hall has been designed such that the lower raked seating tier can pivot up and be effectively 'removed' from the space. The operable wall between the ground floor meeting rooms and the rear of the plenary hall can be opened so that the ground level spaces become one large space.



2.3 Design Targets

2.3.1 Acoustic Design Criteria

2.3.1.1 Background Noise Level

Background noise levels are required for sound system modelling as these represent the ambient level of noise that is typically expected to be present in the hall during normal operation. The background noise level is used when designing the operational noise levels of the sound system and when assessing intelligibility of the system.

The projects' acoustic consultant specified that in the Plenary Hall the background noise level was to be 35 dB(A). A NR30 curve was used as the spectral equivalent for the modelling of background noise levels in the hall.

2.3.1.2 Reverberation time

The reverberation time of a space refers to the length of time it takes sound to decay by 60dB and is related to the volume of a space and the absorption of the materials used for finishes in the space. Reverberation time contributes to how intelligible signal from the sound system is.

The projects' acoustic consultant has specified the mid-frequency (500Hz – 2000 Hz) reverberation time targets for the plenary hall is 1.4 seconds for the full hall mode and 1.1 seconds for the half hall mode. No requirement was specified by the projects' acoustic consultant for the flat floor mode.

The location and types of finishes were agreed between Aurecon and the projects' acoustic engineer based on modelling results.

2.3.2 Sound System Design Criteria

The sound system designer for the conceptual stage did not specify any sound system performance criteria in the design brief report. The performance criteria were proposed by Aurecon based on previous project experience (eg. Darwin Convention Centre and Exhibition Centre, Adelaide Entertainment Centre etc) and are provided in Table 1.

Table 1 – Performance criteria for the Plenary Hall Sound System

Parameter	Criteria
System Frequency Response	45 – 18kHz within ± 3dB
System SPL	105 dB(lin) peak
Broadband SPL Coverage	Within ± 3dB for > 90% of audience plane
Speech Transmission Index	0.65 ("very good" to "excellent")

3. FOH DESIGN - SOUND SYSTEM

3.1 Design Intention

The plenary hall has been designed to provide optimal flexibility offering the client the ability to transform the space between its different modes with a minimum of hassle.

Designing one sound system which can satisfy the design criteria in all the halls' configuration options without being altered was not deemed possible. Alternatively, designing standalone sound systems for each configuration would result in a high cost to the project in the purchase of equipment, increased storage requirements for keeping loudspeaker elements not in use and potentially extended down-time between hall mode changeovers for change-over of equipment and re-rigging.

It was clear that there was a need to strike a balance between flexibility and cost without compromising the ability to meet the sound system design targets.

3.2 Loudspeaker Selections

3.2.1 Why Line Arrays?

A typical PA system can fulfill the sound pressure level requirement of the hall but it is difficult to achieve the desired flat frequency response using such a system, and even more difficult to achieve the required intelligibility (defined in this case in terms of STI).

In order to achieve all three design parameters a line array is considered to be the best solution.

A line array consists of loudspeakers that are installed in a vertical array in order to produce coverage patterns that are unattainable from a single loudspeaker or loudspeaker system acting alone.

The loudspeakers installed in line arrays typically implement various design features to normalize the propagation distance from the throat to the mouth, minimizing interference between horns and allowing the array to approximate a line source rather than a number of stacked discrete point sources.

Arraying, depending upon the techniques involved, can increase or decrease the total coverage angles. Wide vertical coverage can be provided to meet the particular needs of auditoriums with balconies, or narrow vertical coverage can be achieved for very deep spaces requiring long throws and low trim heights.

3.2.2 Preliminary Speaker Layout and Selection

The design decision was made to conduct modelling for the full hall mode using a modular line array approach and then examine ways where the same line array modules could be reconfigured and re-aimed to meet the design criteria in the other hall arrangements.

The client had indicated a preference for externally powered loudspeakers and separate amplifiers to allow re-patching of systems on the fly in case of amplifier failure rather than the need to disassemble the array to service/replace a faulty unit.

Using proprietary line array calculation software provided by a loudspeaker manufacturer, an array configuration was decided upon for use in the detailed modelling.

In the full hall mode, the left/right arrays selected consist of eight cabinets and two subwoofer cabinets placed either side of the stage at a height of 9m above floor level (when measured from top of the array). The half hall mode and flat floor mode utilize reconfigurations of these same cabinets.

As the output of the sound system design was a performance specification document we note that it is possible for an alternative selection to be made by a sound system contractor on the provision that proof is provided of their alternative selection being able to satisfy the performance criteria.

3.3 Modelling Software Used

3.3.1 Proprietary Line Array Calculation Software

The preliminary aiming, gain settings and cabinet sizing was conducted using the manufacturer's proprietary software. This proprietary software allows the user to roughly size and layout the audience planes with respect to the speaker locations and review the coverage prior to exporting the loudspeaker data for the array to an EASE compatible format.

The program generates coverage pattern plots like the example shown in Figure 2.

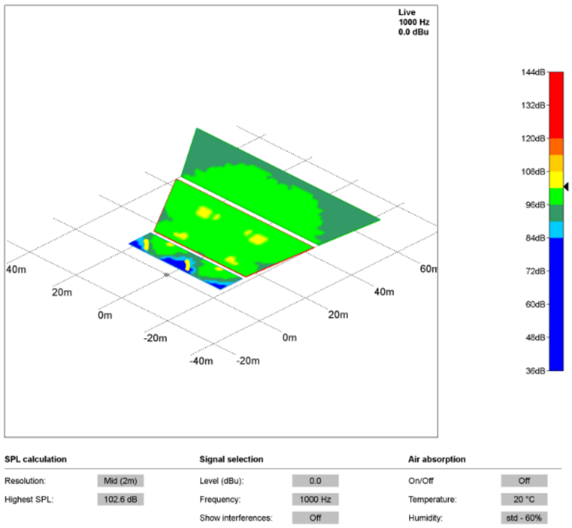


Figure 2: Sample output from proprietary line array calculation software

3.3.2 EASE + AURA

To model the sound system performance in each of the configurations, modelling software which was able to accurately model room response was required. The Industry preferred software package for modelling sound systems is EASE (Enhanced Acoustic Simulator for Engineers) developed by AFMG. EASE allows the sound system designer to virtually place loudspeaker selections into a 3D model of a room and model the sound system performance. The modelling of the performance space includes assigning acoustic properties to surfaces such as absorption coefficients, scattering coefficients and whether the surface is one or two faced (i.e. whether a surface can be 'seen' by sound on the front and rear or not).

The developers of EASE have also released an acoustic ray tracing module AURA which calculates the room responses more accurately than using EASE as a standalone prediction tool.

The program generates coverage pattern plots like that shown in Figure 3.

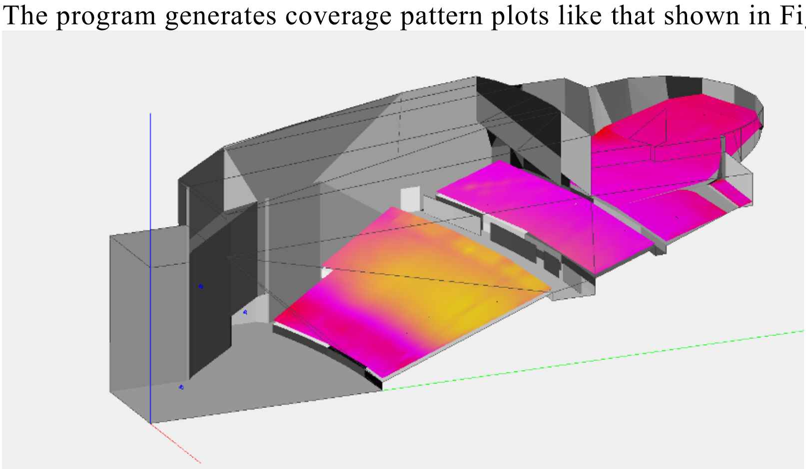


Figure 3: Sample output from EASE AURA 3D mapping software

AURA is composed of a hybrid Ray Tracing engine, which uses both a deterministic Image Model (Cone Tracing) and Stochastic Ray Tracing methods.

3.4 Detailed Speaker Layout

The results of the modelling conducted with EASE are presented in the following sections for each of the different scenarios.

Note that for the prediction of reverberation time in each mode the source used is an omnidirectional source placed on stage, off the center line rather than the loudspeaker array being modelled.

3.4.1 Full Hall Mode

The full hall mode consisted of two line arrays consisting of 8 speaker cabinets and two subwoofer cabinets each. The line arrays were flown to a height such that the top of the array is at 9m above floor level. Figure 4 shows the location of the line arrays in the full hall mode.

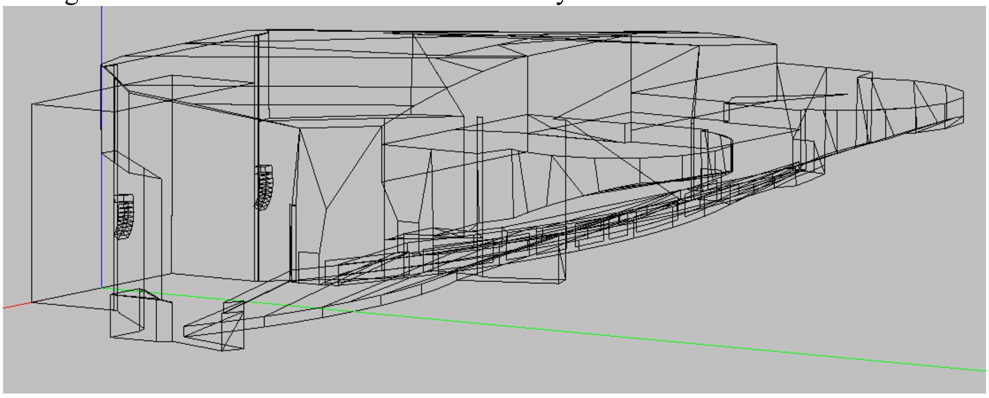


Figure 4: Full Hall mode, two line arrays either side of proscenium

3.4.2 Half Hall Mode – using a single line array

The half hall mode using a single line array is essentially the full hall mode split directly in two with no changes made to the sound system at all. It is unlikely the room would be used in this loudspeaker configuration; however there was an interest to see whether coverage could be achieved in this case.

The sound system consisted of one line array consisting of 8 speaker cabinets and two subwoofer cabinets per half, which was essentially the full hall mode split directly in two with no changes made to the sound system at all. The line arrays were flown to a height such that the top of the array is at 9m above floor level and aimed towards the centre of the audience plane. Figure 5 shows the location of the single line array in half hall mode.

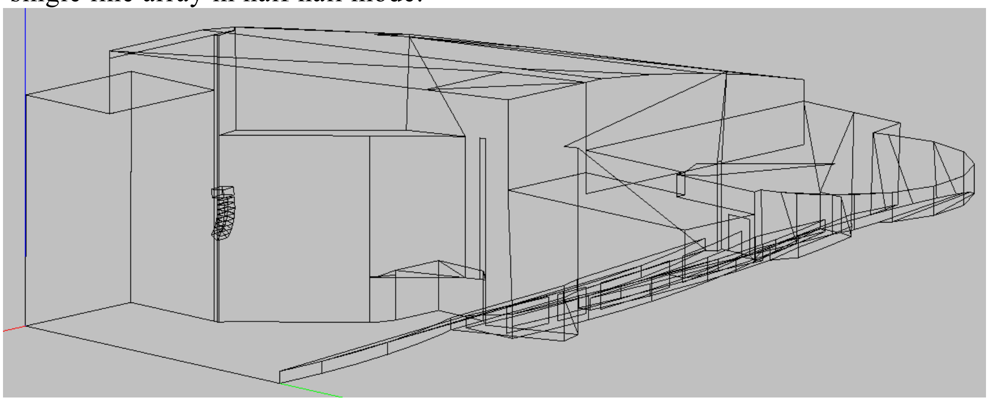


Figure 5: Half Hall mode, single line array one side of proscenium

3.4.3 Half Hall Mode – using two smaller line arrays

The half hall mode consisted of splitting on of the full hall arrays into two smaller line arrays, each consisting of 4 speaker cabinets and one subwoofer cabinets. The line arrays were flown to a height such that the top of the array is at 9m above floor level either side of the presenter position.

The coverage achieved by this configuration was seen to be more even than that provided by a single array. Figure 6 shows the locations of the two smaller arrays in half hall mode.

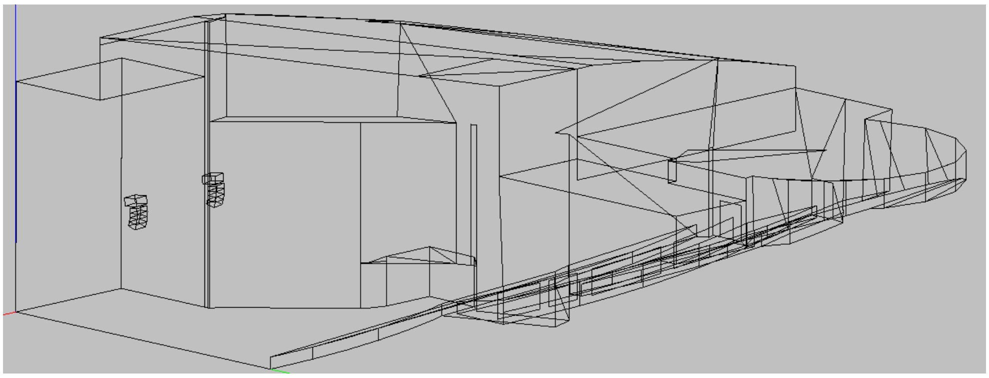


Figure 6: Half Hall Mode, two smaller line arrays, either side of centre position

3.4.4 Flat Floor Mode – line arrays only

The flat floor mode consisted of two line arrays consisting of 8 speaker cabinets and two subwoofer cabinets each as per the full hall mode. The line arrays were flown to a height such that the top of the array is at 6m above floor level.

The coverage and clarity across the hall was seen to be uneven with both level and clarity being significantly lower under the balcony overhang than at the floor closer to the stage position. Figure 7 shows the locations of the two line arrays in flat floor mode

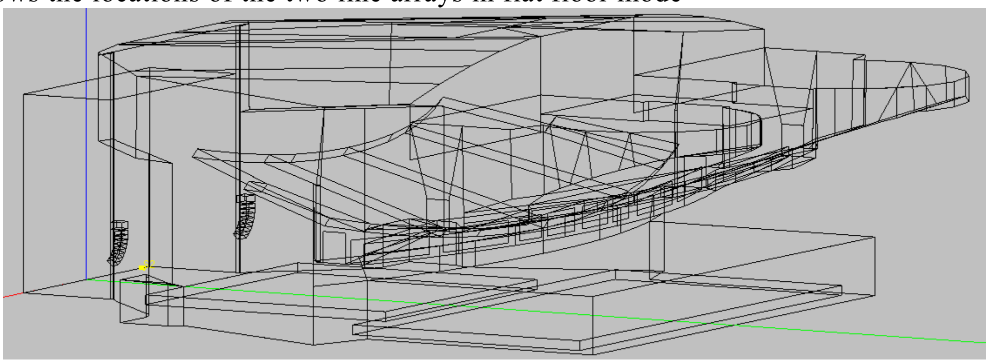


Figure 7: Flat Floor Mode, two line arrays either side of proscenium

3.4.5 Flat Floor Mode – line arrays plus additional cabinets mounted under the seating overhang

In order to improve the evenness of coverage and intelligibility, an option of modelling the flat floor mode as per the option above with additional cabinets mounted to the underside of the overhang was investigated.

In order to manage the difference in distance between the arrays and the supplementary cabinets under the overhang, a time delay was applied to the signal being sent to these additional cabinets to achieve the speech intelligibility targets. Figure 8 shows the locations of the two line arrays and additional speakers installed underneath the seating overhang

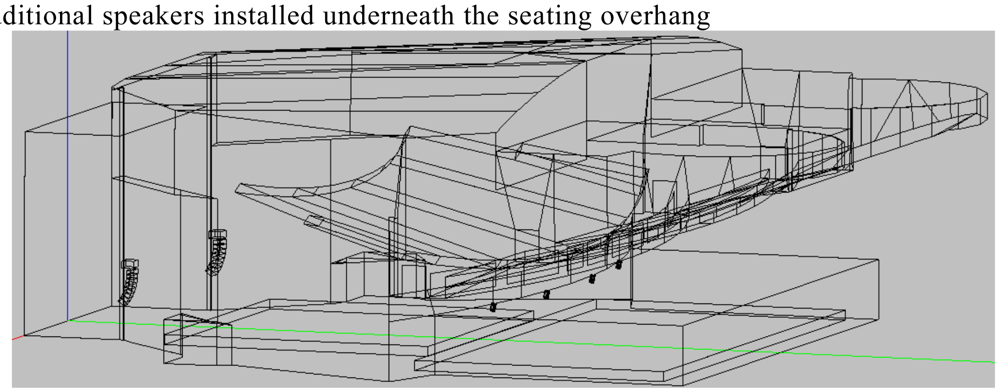


Figure 8: Flat Floor Mode, two line arrays either side of the proscenium plus four additional cabinets mounted under the seating overhang

3.5 EASE Modelling Results

A summary of the modelling results for each of the different modes is provided in Table 2 below.

Table 2 – Assessment of sound system configurations

Mode	**Sound System Configuration**	**Criterion**			
		RT	**SPL**		**STI**
TARGET		**1.4 sec (full hall) or 1.1 sec other modes**	**Min peak 105dB(lin)**	**Within ± 3dB for >90% of audience plane**	**0.65 ("very good" to "excellent")**
Full Hall Mode	Left and Right arrays, each consisting of: - 8 cabinets and - 2 subwoofer cabinets Flown to 9m above floor level (when measured from top of array)	1.1 s	105.5	± 5.5dB	Meets
Half Hall Mode	Single array consisting of: - 8 cabinets and - 2 subwoofer cabinets Flown to 9m above floor level (when measured from top of array)	1.0 s	105.5	± 5 dB	Meets
	Left and Right arrays, each consisting of: - 4 cabinets and - 1 subwoofer cabinets Flown to 9m above floor level (when measured from top of array)	1.0 s	105	± 5 dB	Meets
Flat Floor Mode	Left and Right arrays, each consisting of: - 4 cabinets and - 1 subwoofer cabinets Flown to 6m above floor level (when measured from top of array)	0.7 s	110	± 4 dB	Meets
	Left and Right arrays, each consisting of: - 4 cabinets and - 1 subwoofer cabinets Flown to 6m above floor level (when measured from top of array) Four additional cabinets mounted under the seating overhang with delay added to provide more even coverage to the rear of the Plenary Hall when in flat floor mode.	0.7 s	113	± 3.5 dB	Meets

4. BOH DESIGN – NETWORK INFRASTRUCTURE

Analogue audio inputs are converted to digital, processed and distributed around the space subject to the configuration of the venue. Input mixing, routing plus loudspeaker management and signal processing including, gating, filtering, equalisation, limiting, time alignment delay and ambient noise compensation will be performed in the digital domain. The digital signals will be converted from digital to analogue and passed into power amplifiers either separate or integrated within self-powered loudspeakers as at designated locations within the venue as noted above. Users preference was to maintain separate amplifiers and loudspeakers rather than integrated, given the potential for either component to fail and the practical issues associated with replacement (ie. Line arrays are cumbersome to access due to the nature of rigging).

Consideration was given to the protocol used to transmit digital audio over Ethernet, with Harman providing a useful summary of available protocols (3). Ethernet is divided into seven layers, from the physical layer (the Ethernet switch / cable – layer 1) to the application layer (an email software program, for example – layer 7), as shown below on Figure 2. Generally, network data is transmitted on layer 3 as IP (Internet Protocol) traffic, including some proprietary audio and video protocols. Audio-video transmission will always be prone to interruption from email, web and other internet or local area use unless implemented on a separate, dedicated network. Even in this situation, often complex network management is still required to protect audio and video traffic from non-media data and itself.

CobraNET (developed originally by Peak Audio and then later by Cirrus Logic.) is a layer 2 network protocol and has been the standard protocol for audio-networking up until about 2005. Since then a range of other protocols have developed with extended capabilities. Layer 2 protocols limit any interruption of audio-video transmission. AVB (Audio Visual Bridging) is also a layer 2 protocol, and it is an open standard as opposed to commercial protocols, and is the industry standard. While AVB adoption increases, there is still a need within the industry to support a suitable layer 3 technology which can be carefully managed over non-AVB Ethernet switches to provide high channel count solutions. Other than AVB, Dante is really the only other current-generation Gigabit transport to have been adopted by several manufacturers across the pro audio industry.

Given the critical nature of ensuring uninterrupted, low latency, synchronised (time stamped) audio (and video) distribution, AVB network switches with Dante network protocol was selected as the basis for design to allow for future-proofing. Routing solutions using Dante enabled processors allow for simplified reconfiguration of the venue depending on use.

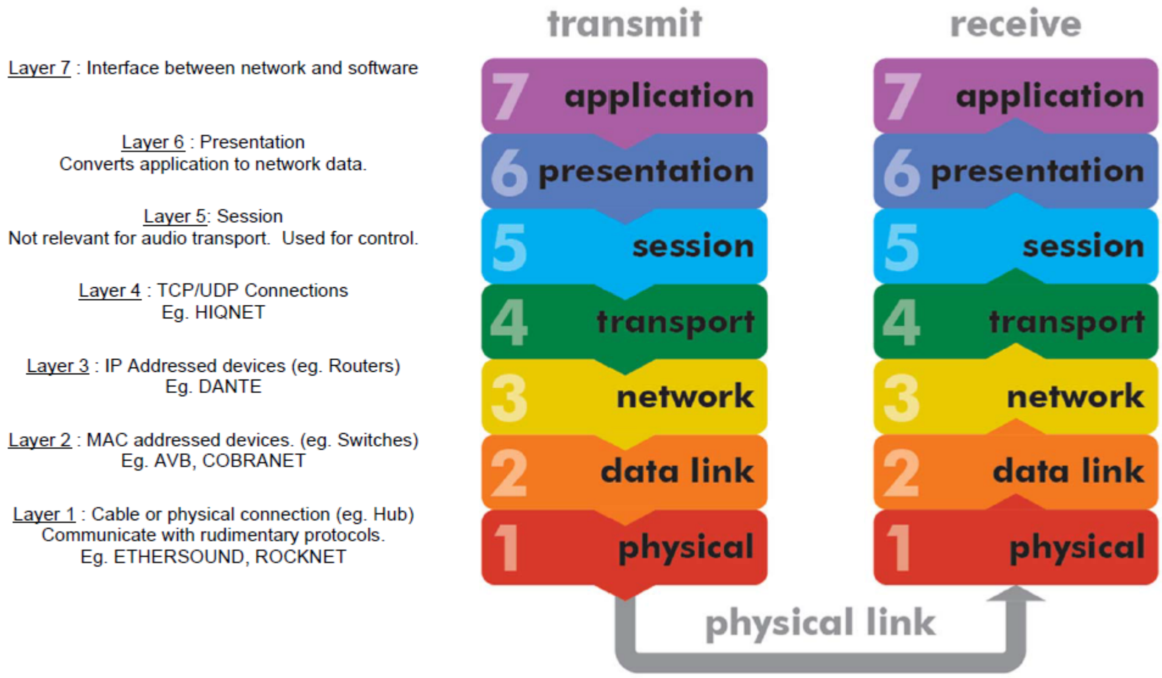


Figure 9 – Ethernet layers (3)

5. CONCLUSION

The redevelopment of the existing Plenary Building of the Adelaide Convention Centre includes the design of a highly flexible Plenary Hall which operates in a number of different configurations. In order to service the various configurations a digital sound system design which offers good coverage and flexibility has been designed. In delivering this digital sound system, acoustic modelling of the space was conducted alongside computational predictions of the sound system performance. The sound system design delivered allowed the user to use the one set of components to achieve a number of different configurations to match the various room uses. While separate sound systems individually selected and designed for each room mode could potentially provide further optimized performance results, the cost benefits and flexibility provided using the approach of one reconfigurable sound system balances the slight shortcomings.

A robust and flexible digital audio network was designed using standard data network infrastructure which allowed the users of the hall to easily send audio to any configuration of the space.

6. REFERENCES

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3. Harman Corp, “HiQnet guide to audio networking”

7. ACKNOWLEDGEMENTS

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