"WATT DID THEY JUST ANNOUNCE?" A NOVEL SOLUTION FOR A QUANTUM IMPROVEMENT IN INTELLIGIBILITY ON RAILWAY STATION PLATFORMS.

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1 INTRODUCTION

This paper presents an innovative, benchmarked solution for a sound system that achieves high speech intelligibility in the difficult acoustic environment of multiple railway platforms. RailCorp, the NSW Government department responsible for the operation of the Sydney Metropolitan rail network, recently engaged Acoustic Directions to undertake the design and commissioning of a number of rail sound system projects. These projects have led to a new type of announcement delivery system being conceived and installed at Sydney's Central Railway Station and a roll-out of similar systems is planned for the Sydney suburban rail network.

This paper describes the measured advantages this system shows over other, more conventional systems. The advantages result in part from the use of a customised beam-steered loudspeaker, which is also described.

The issue of excessive announcement levels made in the presence of high background noise arose during commissioning of the system. We present a brief discussion of speech broadcast in the presence of noise.

1.1 RailCorp's requirements

RailCorp's requirements for announcement quality at Central Station Sydney may be summarised as follows:

a) Natural speech-sound is required in order to maximise the intelligibility of announcements and provide a calming influence in the case of an emergency.

b) Provide minimal acoustic spill between platforms that are separated by rail tracks in order to minimise the following:
   - Damage to intelligibility during simultaneous announcements.
   - Annoyance of travellers on adjacent platforms by announcement-sound that is not needed and may be considered unwanted.

c) Provide satisfactory sound coverage of two-thirds the width and entire length of an island platform, as measured from the side of the intended train.

d) Provide adequate intelligibility of an announcement broadcast on one side of an island platform, whilst a different announcement is simultaneously broadcast on the other side.

e) Minimise acoustic spill from announcements to commercial or residential premises beyond the platforms.

f) Provide robust loudspeakers resistant to damage from brake dust, vandalism and environmental effects

g) Provide a cost-effective solution for the long term

RailCorp's requirements for systems at suburban stations were further refined, with emphasis placed on:

a) Minimising annoyance to local residents

b) Minimising spill between platforms

c) Minimising system cost

d) Maximising system longevity

e) Improving maintenance systems

f) Achieving appropriate intelligibility and listener comfort for commuters

2 SYSTEM DESCRIPTIONS

Two loudspeaker systems are initially described and analysed. System 1 is an optimised system where all loudspeakers are connected in parallel without signal delay being applied to any loudspeaker. This is termed the "herringbone" system, due to the arrangement of loudspeaker locations and aiming. The second system uses loudspeakers powered by small locally situated amplifiers. Each amplifier contains digital signal processing that allows delay to be added to the signal of individual loudspeakers. This system is named the "time-sequenced" system.

2.1 Herringbone system

Figure 1 shows the system architecture of the herringbone system, which uses inexpensive horn type loudspeakers driven from a 100 volt line. This arrangement of loudspeakers differs from a standard 100 volt system in that it jointly optimises both coverage and temporal behaviour.

To achieve good performance, it is necessary to space loudspeakers at approximately 5 metre intervals and angle them obliquely across the platform toward the tracks - hence the "herringbone" moniker. Local structures on the station awning are used for loudspeaker support.

In addition to providing good sound coverage, the oblique aiming provides two other benefits:

a) Sound is directed away from the building structure towards the open air thereby reducing the build-up of reverberation.

b) The region of direct-field overlap between the loudspeakers is reduced, resulting in a lower level of late-arriving direct sound.

Using this loudspeaker arrangement, the direct-field coverage at high frequencies over the entire listening area falls within a window of approximately 5 dB at 4 kHz. After equalisation, the averaged frequency response (over the listening area) achieves +/- 3dB between the response-limits of the horn loudspeaker: 200 Hz to 10 kHz.

There are, however, temporal compromises that are significant: the number of loudspeakers in the system and their lack of low-mid pattern control result in a considerable amount of late arriving sound.

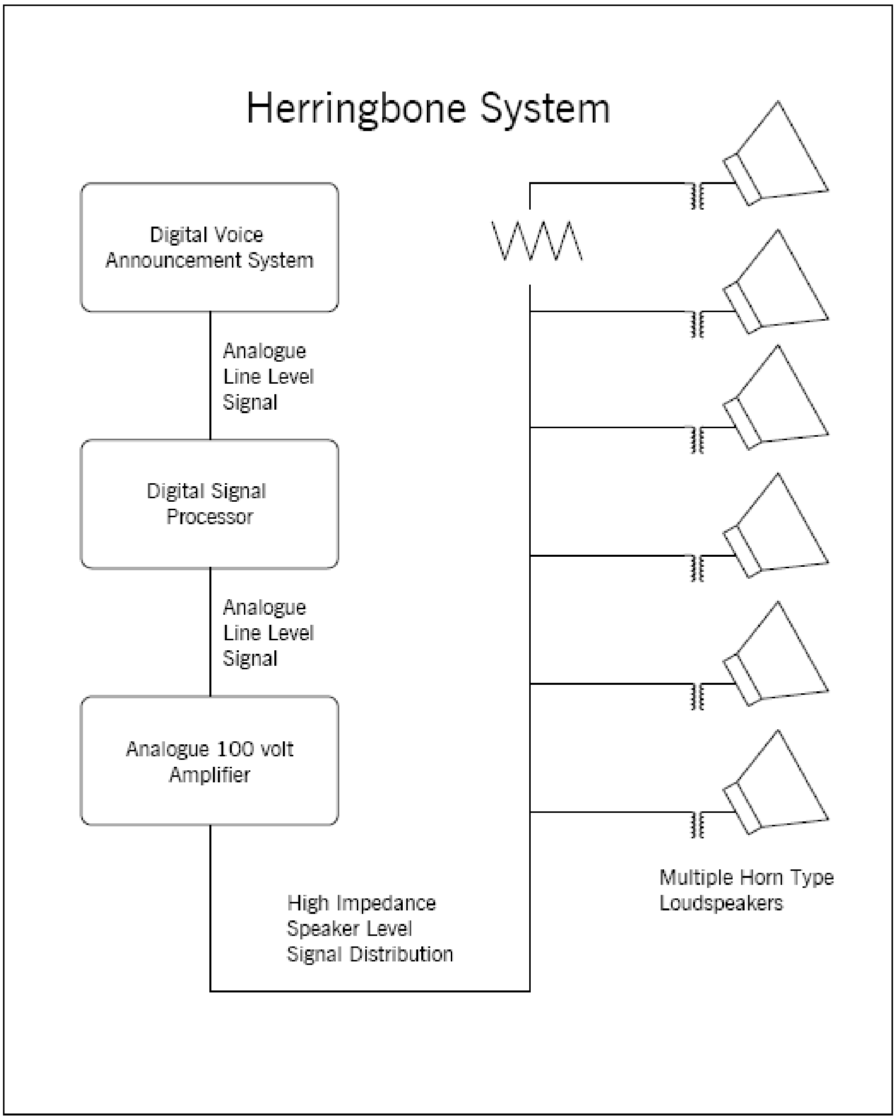

Figure 1: Indicative architecture of herringbone system

2.2 Time-sequenced system

The time-sequenced system uses the same inexpensive horn-type loudspeakers as the herringbone system, but uses an individual processor/amplifier for each loudspeaker. The loudspeakers are arranged so that the main body of sound is projected along (rather than diagonally across) the platform.

To present a time-sequenced arrival pattern from multiple loudspeakers aimed in the same direction, signal delays and level control are applied separately to each loudspeaker. However, to minimise system cost, equalisation is implemented on a global basis.

Figure 2 below shows the indicative architecture of the time-sequenced system.

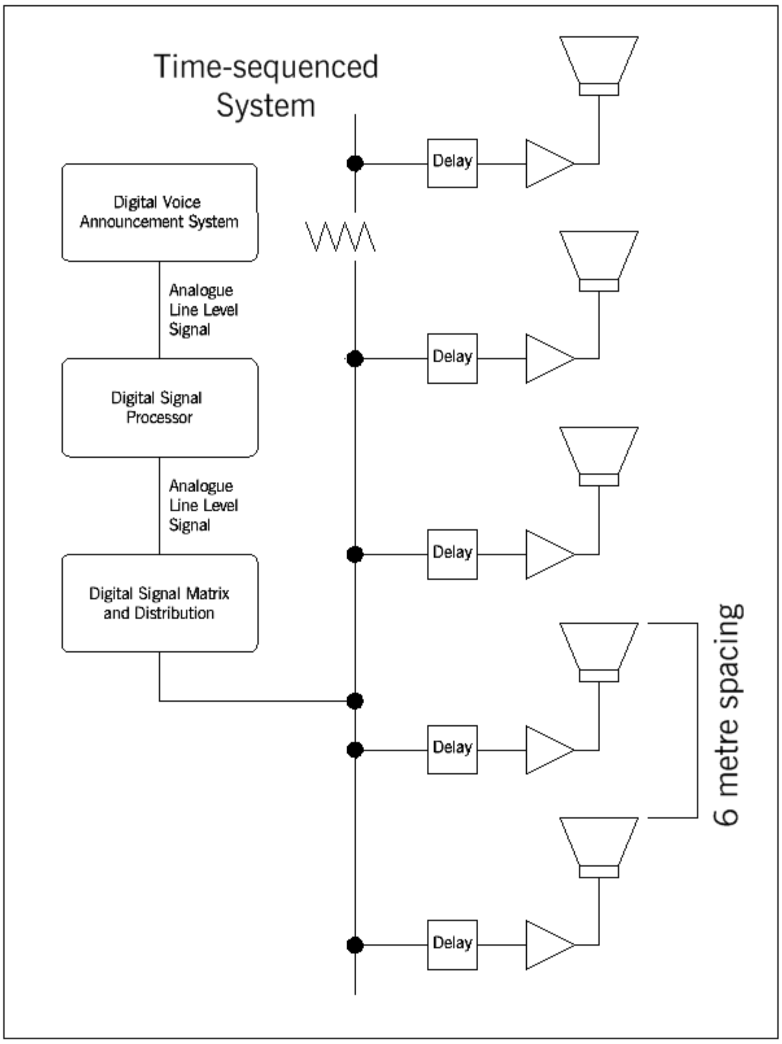

Figure 2: architecture of time-sequenced system

3 SYSTEM PERFORMANCE

The impulse responses of the two systems were measured using WinMLS software to analyse acoustical parameters such STI, clarity ratios, and frequency response.

3.1 Metrics used to assess performance

3.1.1 Frequency response

Good tonal balance of the broadcast sound is necessary in any system that is required to produce good speech intelligibility. It is associated with qualitative measures of "naturalness" as well as complex masking effects. Tonal balance is directly related to a system's frequency response.

The ideal frequency response of a speech system, including the recording, signal processing and playback corresponds to a perfect conduit of the original source. There may be some instances, however, where untrained voices require modification in dynamics or frequency response.

The frequency response of a system is assessed as an average over all listening areas and also at individual points on the listening plane.

3.1.2 Speech Transmission Index (STI)

Speech Intelligibility Index (STI)[1] is a metric commonly used in the assessment of speech intelligibility that can be derived from an impulse response of a system under test. It produces a matrix of modulation values for a range of modulation frequencies present in seven octave bands. Weights are applied to the matrix to calculate a final result between 0 and 1. Noise may be added to the calculations in octave bands to determine the STI in the presence of noise.

While STI is well known and often used, it is relatively blind to a number of important factors in both frequency and time [2,3,4].

STI has been used as one measure to assist interpretation of the performance of the system, with its limitations understood. Both systems perform extremely well in terms of STI, which perhaps points to further unresolved deficiencies in that metric.

3.1.3 The ratio of early-arriving sound to late-arriving sound

High intelligibility requires the total amount of early arriving speech sound to be at a higher level than the total amount of late-arriving sound from distant loudspeakers, reverberation or echoes. A metric useful in determination of the ratio of early-to-late arriving sound is the clarity ratio.

One time limit that is often-used to differentiate between useful and non-useful sound is 50 milliseconds resulting in the C_{50} clarity ratio. However, 35 milliseconds is used here as the time limit in order to accentuate the effects of late-arriving direct-sound and its degrading effects on intelligibility.

3.1.4 Environmental spill

Whilst meeting intelligibility requirements on the platforms, the sound levels at nearby residential and commercial receivers are required to be as low as possible to meet noise criteria. Criteria are derived from the local background noise at various times of the day and night for both L_{eq} and L_1 of announcements.

3.2 Results

3.2.1 Tonal balance

A comparison of the average frequency response of the two systems is shown in Figure 3 below. It can be seen that both systems exhibit reasonably flat responses through the pass-band of the horn loudspeakers: 200Hz to 10000Hz.

A total of 58 measurements (herringbone system) and 61 measurements (time-sequenced system) were averaged for this analysis. Measurements were taken within the platform coverage zones for the two systems.

The process of tonal balance equalisation was achieved by measurement (WinMLS) of average and spot responses and critical listening using music and speech. Both systems were equalised without time constraints and with the intention of achieving the best possible outcome.

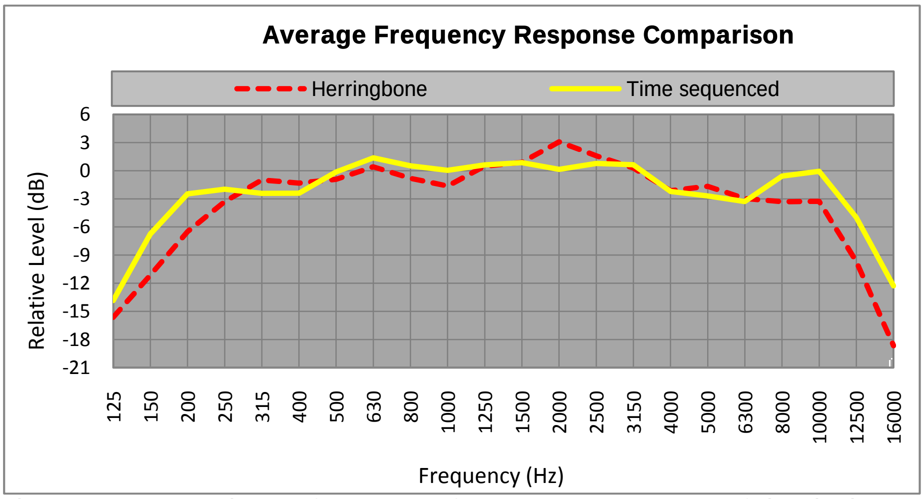

Figure 3: Comparison of average frequency responses of herringbone and time-sequenced systems.

Figure 4 and Figure 5 show the individual frequency responses on each system with the least and most variation from the average within the frequency range shown, together with the standard deviation from the mean response of each system. Statistically, the two systems appear very similar, with the majority of responses lying within a 6 dB window around the mean.

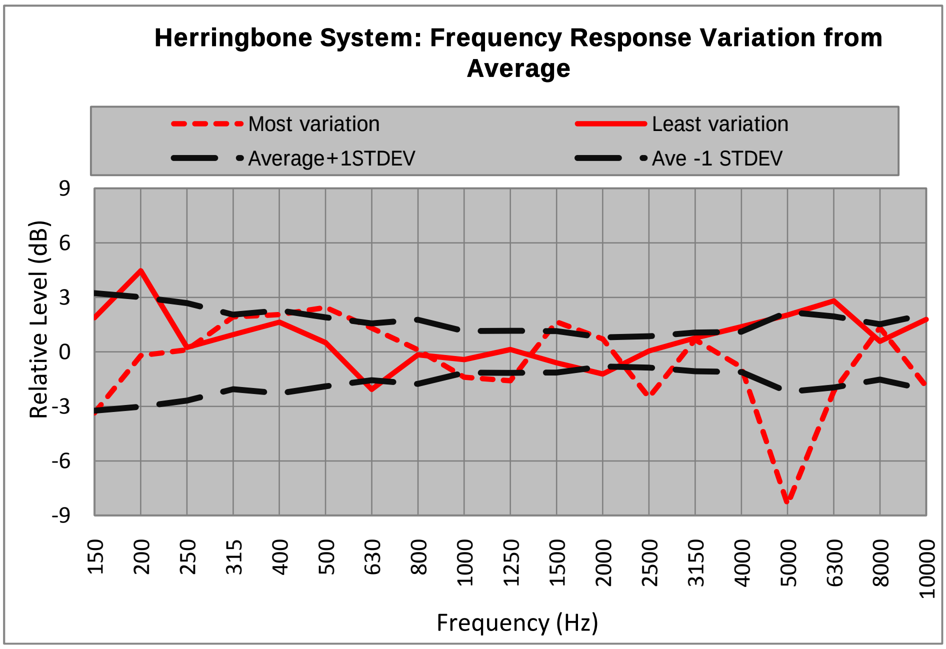

Figure 4: Frequency Response variability in herringbone system

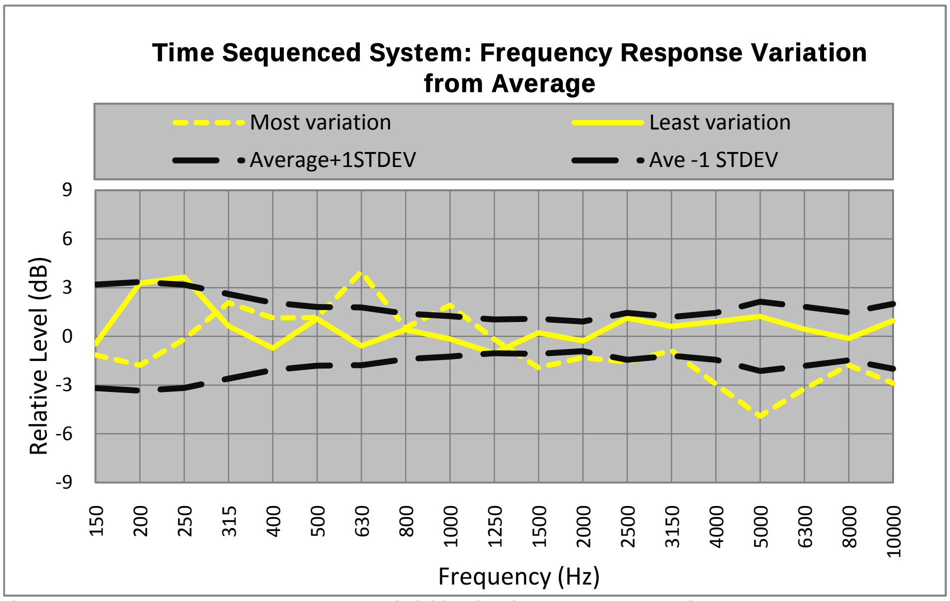

Figure 5: Frequency response variability in time-sequenced system

Figure 6 compares the time sequenced and herringbone systems in terms of their most and least variable responses. The variations indicate differences between the systems, with the best responses of the time-sequenced system being flatter than those of the herringbone system. The dip at 5000Hz in both the herringbone and time sequenced systems indicate that the polar response of the horn loudspeaker is narrow for this application in the 5000 Hz region.

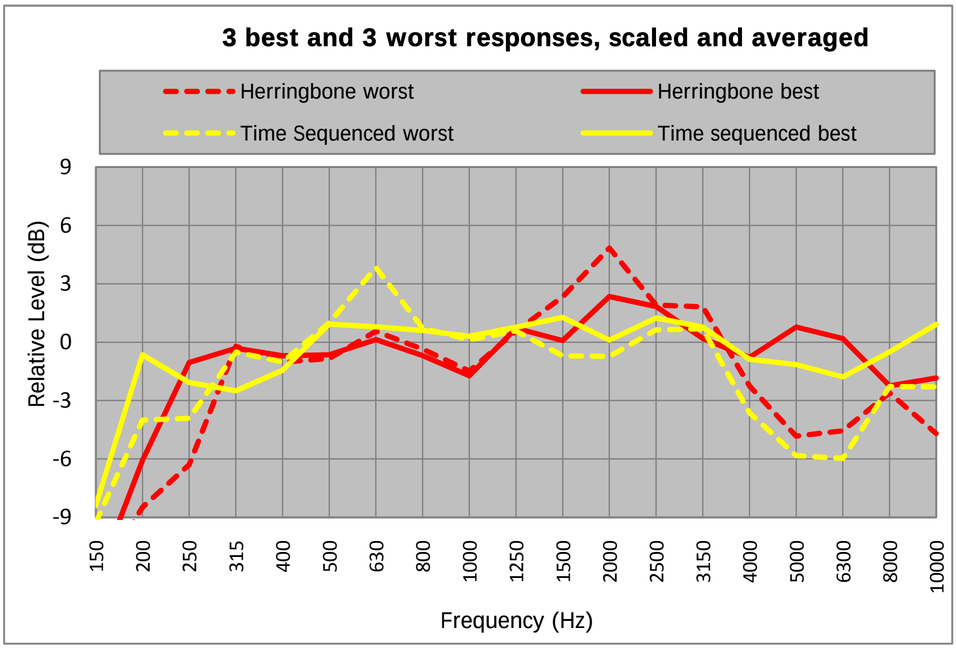

Figure 6: Average of 3 best responses and average of 3 worst responses of each system, scaled

3.2.2 STI

The noiseless STIs were calculated from the impulse responses with the average Modulation Transfer Indices (MTI) being shown in Figure 7. The MTI values are high in all octave bands from 250Hz to 8 kHz. The 125 Hz octave band did not exhibit a high enough signal to noise ratio to be reliably included in the calculation. Both systems show excellent noiseless STI: 0.87 for herringbone and 0.88 for time sequenced.

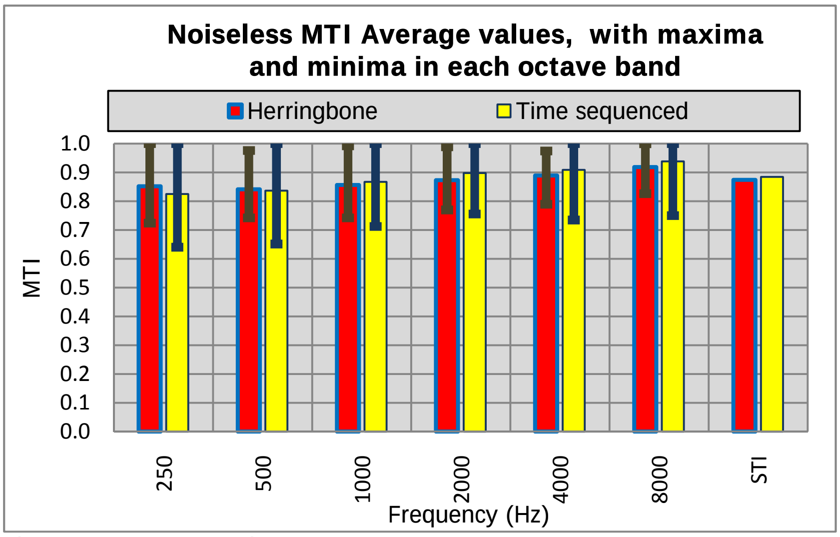

Figure 7: Average noiseless MTI values

The addition of train noise to the MTI calculation shows the time-sequenced system as superior, with the worst case MTI values being noticeably higher for this system than for the herringbone system. STI figures for herringbone were 0.56 and time sequenced 0.57.

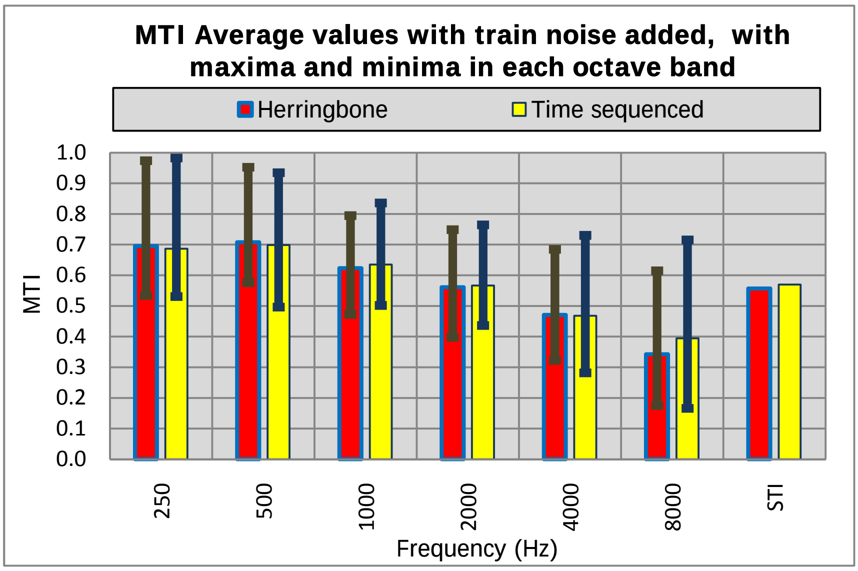

Figure 8: MTI values in the presence of train noise

3.2.3 Clarity ratio

A comparison of the two systems in terms of C_{35} is shown in Figure 9, below. The MTIs of the time-sequenced system are superior by at least 1.5 dB in each octave band above 250Hz. The results indicate that not only are the average C35 results better for the time sequenced system, but the best results for this system are over 5dB better than those of the herringbone system.

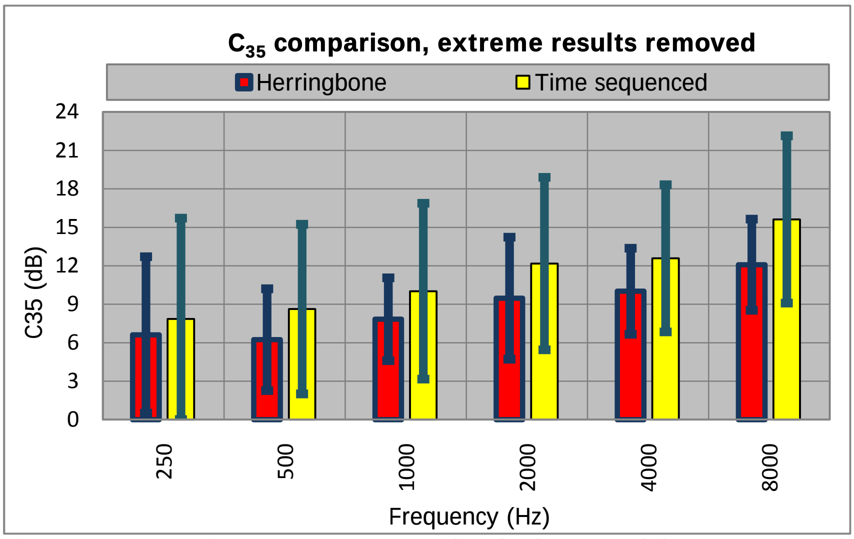

Figure 9: Comparison of average C_{35}, herringbone and time-sequenced systems with maximum and minimum values shown

3.2.4 Environmental spill

Measurement of environmental spill was done over a number of evenings, with minimum levels being taken as the basis calculations including background noise. Announcement levels were scaled from known platform levels using a shaped noise stimulus.

Figure 10 shows the positions used for measurement and subsequent analysis.

Figure 10: Spill measurement positions relative to platform

Results for spill are shown in Figure 11. The high level for the "45 degree" position is due to the dominance from loudspeakers in uncovered areas. Spill is indicated relative to the minimum background noise measured (LA_{eq}). Announcements are adjusted for average speech spectra.

In each case, the performance of the time sequenced system is either equal or superior to the herringbone system.

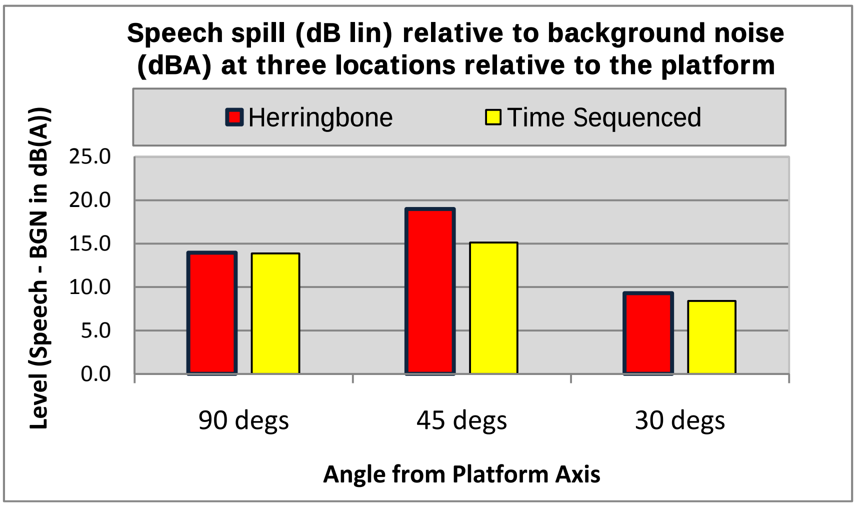

Figure 11: Spill relative to background noise at positions relative to the platform

3.3 Qualitative performance

Both systems performed well in both listening tests and provided a great improvement in performance over the systems traditionally installed at railway stations.

However, despite the similarities, there were noticeable differences in the quality of audio produced by the two systems.

The time-sequenced system sounded "intimate" and "present". The delivery of messages felt personal and direct with this system. By contrast, the herringbone system sounded more "roomy", and slightly confused, with the direct nature of announcement delivery being slightly lost to a more "PA" type sound.

These differences were noted by Railcorp stakeholders.

3.4 Other performance indicators

A number of other performance indicators were taken into account in the final choice of system.

3.4.1 Cost

The time-sequenced system was slightly more expensive to purchase and install than the herringbone system, but the cost implications were deemed to be off-set in the medium term by reduced maintenance costs and improved performance.

3.4.2 Maintenance

The time-sequenced system allows off-site monitoring of each loudspeaker and amplifier's health, allowing specific and targeted repairs to be carried out. The time sequenced system therefore has advantages in the area of maintenance.

3.4.3 Longevity

Both systems provide a robust solution in the longer term.

3.4.4 Commuter expectations

The increasing expectations of commuters regarding the quality of announcement delivery are better met by the time-sequenced system due to its more intimate and personal sound.

3.5 Conclusions

The time-sequenced system is an approach to the delivery of announcements on railway platform that yields significant benefits when compared to more traditionally implemented systems. While the herringbone system performed well (perhaps above expectations), RailCorp determined that the time-sequenced system was preferable and adopted its use. However, the limited bandwidth of the horn loudspeaker was a limitation to the system in terms of fidelity of sound and therefore listener comfort.

4 DEVELOPMENT OF CARDIOID-PATTERN LOUDSPEAKER TO IMPROVE THE TIME-SEQUENCED SYSTEM PERFORMANCE

A number of improvements in the performance of the time-sequenced system were desired:

a) Increased system bandwidth.

- The bandwidth of the horn system, although serviceable, is not ideal for the reproduction of natural sounding speech. A good target for bandwidth was considered as 100 Hz to 12.5 kHz, which is not achievable with low-cost horn-type loudspeakers.

b) Improvement of high frequency coverage.

- The horn loudspeakers showed very narrow coverage angles at some high frequencies (5 kHz to 6.3 kHz), which degraded both coverage at these frequencies and the formulation of an average equalisation for the platform. High frequency coverage angles should be reasonably even throughout the loudspeaker's range.

c) Increase of low frequency loudspeaker directivity.

- Modelling indicated that improvements to intelligibility could be achieved by reducing late-arriving low frequency sound emanating from loudspeakers facing away from a listener. While the horn loudspeakers have significant directivity at higher frequencies, their directivity at the lower frequencies is quite low.

To address these improvements, a loudspeaker with a cardioid-shaped radiation pattern in the low and low-mid frequency ranges was developed for use with the time-sequenced system.

4.1 Required characteristics

The following characteristics were required for the loudspeaker.

a) Low rear radiation.

Reducing rear radiation would increase intelligibility and intimacy of sound at the listeners and reduce spill to other platforms and to outside the station. Using a small transducer, this can only be achieved by processing multiple loudspeaker drivers.

b) Appropriate width of forward radiation pattern to cover platform.

c) Bandwidth of at least 125 Hz to 12 kHz in order to reproduce natural sounding speech.

d) Directivity as constant as possible.

e) Sensitivity of at least 85 dB per 2.83 volts drive to each transducer.

This requirement allows small amplifiers located at the loudspeakers to produce sufficient SPL at the listeners.

4.2 Development of the loudspeaker

4.2.1 Transducer type

To improve the radiation in high frequencies while keeping the loudspeaker footprint small, an inexpensive dual concentric driver with co-axial tweeter and passive crossover was chosen as the most suitable transducer.

4.2.2 Enclosure

Development of the cardioid-like loudspeaker commenced with a cylindrical enclosure to house the drivers, as this was the simplest enclosure to manufacture that was aesthetically acceptable. The tube length was set to 200 mm, being a compromise between inter-driver distance and enclosure volume (for low frequency response).

With the drivers at each end of the tube, the polar patterns were measured for one driver, with the results below 1600 Hz shown in Figure 12. These results show a strong resemblance to the classic directional properties of a piston in a long tube.

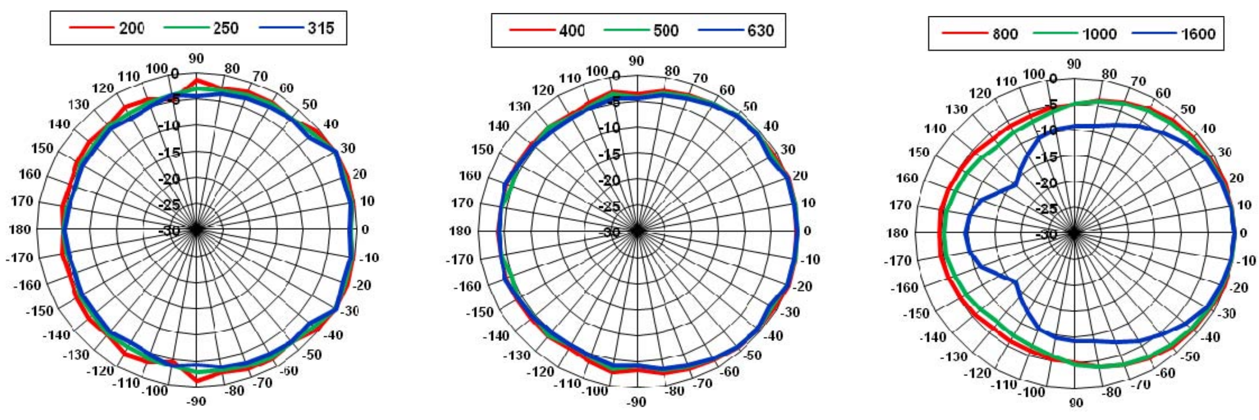

Figure 12 Polar pattern of driver in the 200 mm long cylindrical tube

Although it is straightforward to achieve cardioid performance at low frequencies with the simple tube system, it was difficult to achieve satisfactory cardioid-like beam forming above 630 Hz, without compromising the polar pattern below 400 Hz. Two factors contributed to this:

a) Equal path lengths from the front driver around the tube to the rear driver allow a build-up of energy at each end of the tube.

b) The relatively long acoustic path lengths between the front and rear drivers at frequencies between 500 Hz and 1 kHz created difficulties for the signal processing to reduce the concentration of rear-radiated energy.

When elliptical baffles were added to the tube and the drivers laterally offset in each baffle, the path lengths to the rear were less equal, and the increased baffle area produced less rear radiation at the higher frequencies. With this arrangement, signal processing was better able to achieve a more consistent radiation pattern up to the frequency at which the natural directivity of the driver dominates the radiation pattern. Figure 13 shows the polar pattern up to 1600 Hz of the unprocessed system. At frequencies above 1600Hz, the intrinsic radiation pattern of the woofer and tweeter dominated the pattern, as shown in Figure 14.

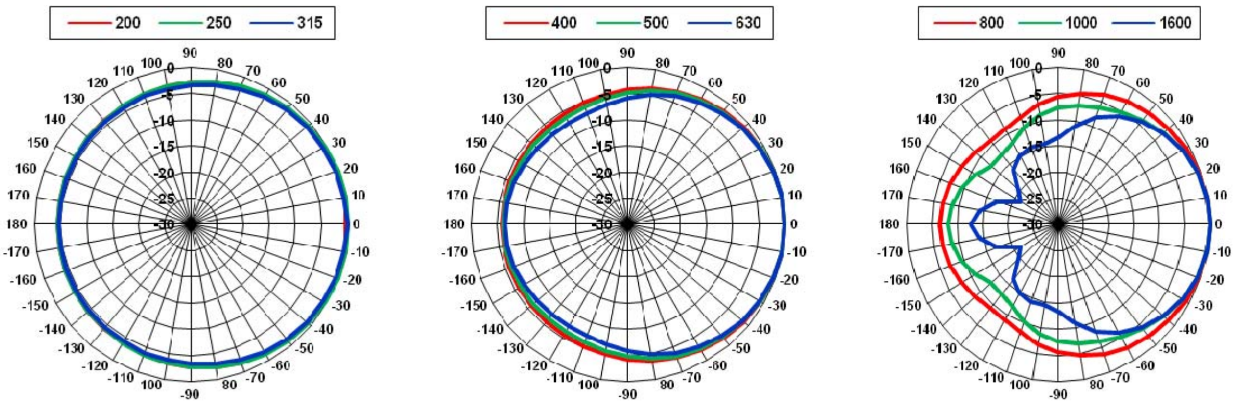

Figure 13 Polar pattern of driver in the 200 mm long tube with elliptical baffle and offset drivers

When simple signal processing comprising delay, low pass, allpass and notch filtering was applied to this system, the polar patterns of Figure 14 resulted.

Although it was not possible to produce a perfect cardioid or super-cardioid pattern using IIR filters, a worthwhile reduction in rear radiation has resulted, especially at frequencies up to 630 Hz. As this frequency range carries the majority of the power in speech, this decrease in rear-radiated energy reduces the tonal colouration and potential loss of intelligibility due to late arriving sound at these frequencies.

In-situ photographs of the final implementation of the cardioid loudspeakers at Sydney Central Station are shown in Figure 15.

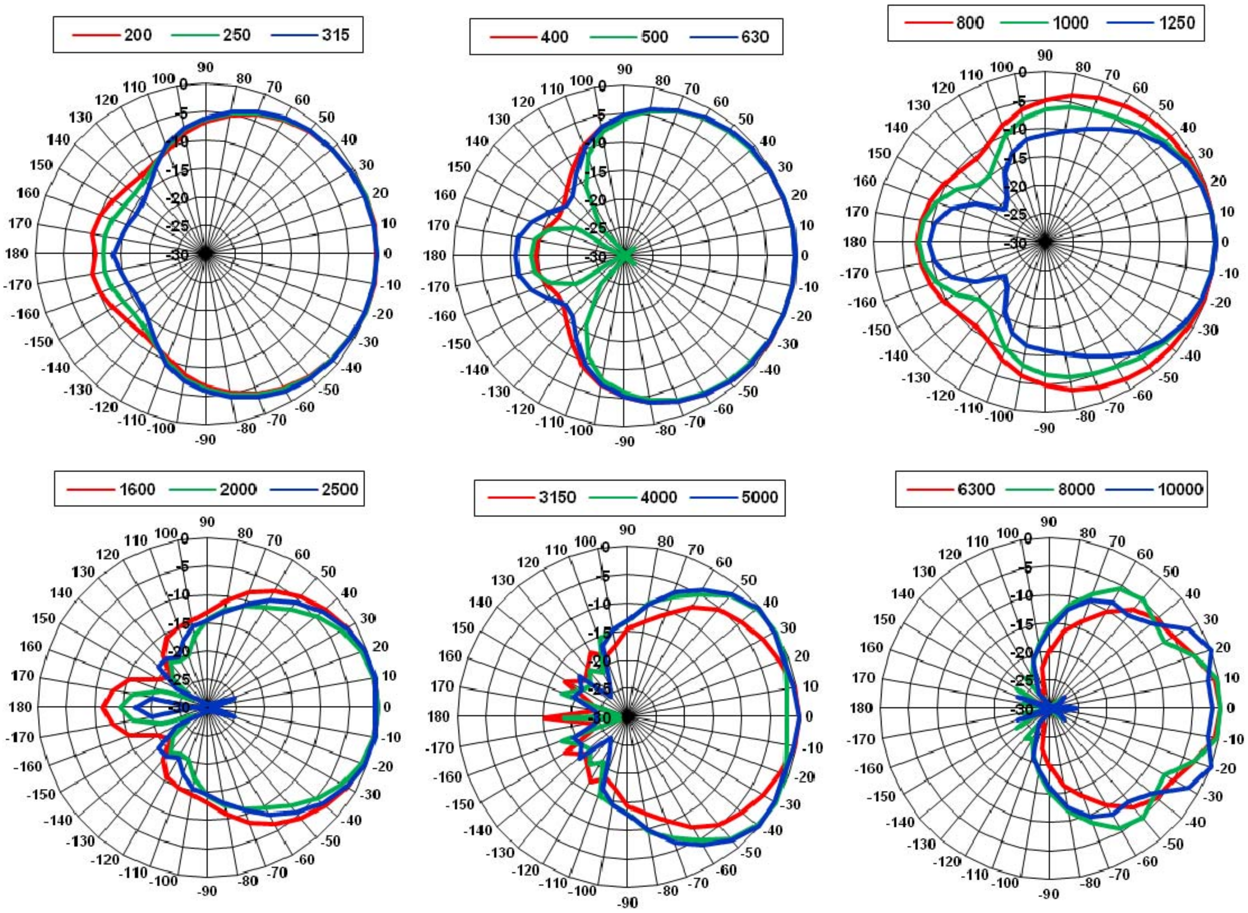

Figure 14 Polar patterns of final system

Figure 15: Photographs of steered time sequenced system as installed at Sydney Central Station

5 IMPLEMENTATION OF THE STEERED TIME SEQUENCED SYSTEM

5.1 Spatial separation of stimuli

A requirement of the system performance was that announcements with different content could be simultaneously made to each side of an island platform and satisfactorily understood by patrons.

The time-sequenced system architecture using steered loudspeakers provides a way to achieve this by giving a strong directional component to the announcements. Listeners are able to "lock onto" the relevant information, much as the phenomenon known as the "Cocktail Party" effect allows people to gather intelligibility in noisy environments.

The effect of spatial separation of competing sources on intelligibility has been reported by numerous authors [5,6,7]. The generalised findings are that spatial separation of competing sources improves intelligibility by some margin. Hawley et al [7] report a keyword error rate of 40% for coincident sources, reducing to around 5% for sources separated by 30 degrees.

5.2 Time-sequenced steered system processing and amplification

The time-sequenced system required delivery of two amplified signals to the front driver and rear drivers of each loudspeaker. The arrival time of these signals was required to be synchronised precisely. Processing required for the cardioid pattern: filtering, polarity reversal and delay and basic equalisation was achieved on a global basis, with the front and rear driver signals being distributed via a digital buss to the appropriate amplifiers. DSP cards located with each amplifier provided adjustment of local delay for every loudspeaker. Figure 16 indicates the system implementation in diagrammatic form.

6 RESULTS

6.1 Steered loudspeaker time-sequenced system ("Steered time-sequenced system")

6.1.1 STI

The steered time-sequenced system compared favourably with the horn time-sequenced system. Figure 17 shows a comparison of the MTIs of the two systems with background noise removed. The poor MTI of the horn system at 125 Hz is due to the poor signal to noise ratio in that band, as the horn loudspeaker has low output capacity in this frequency range. The increased MTI at 250 Hz from the cardioid system probably reflects its higher directivity at lower frequencies. The slightly higher MTIs of the horn system at higher frequencies may reflect its narrower radiation pattern at these frequencies.

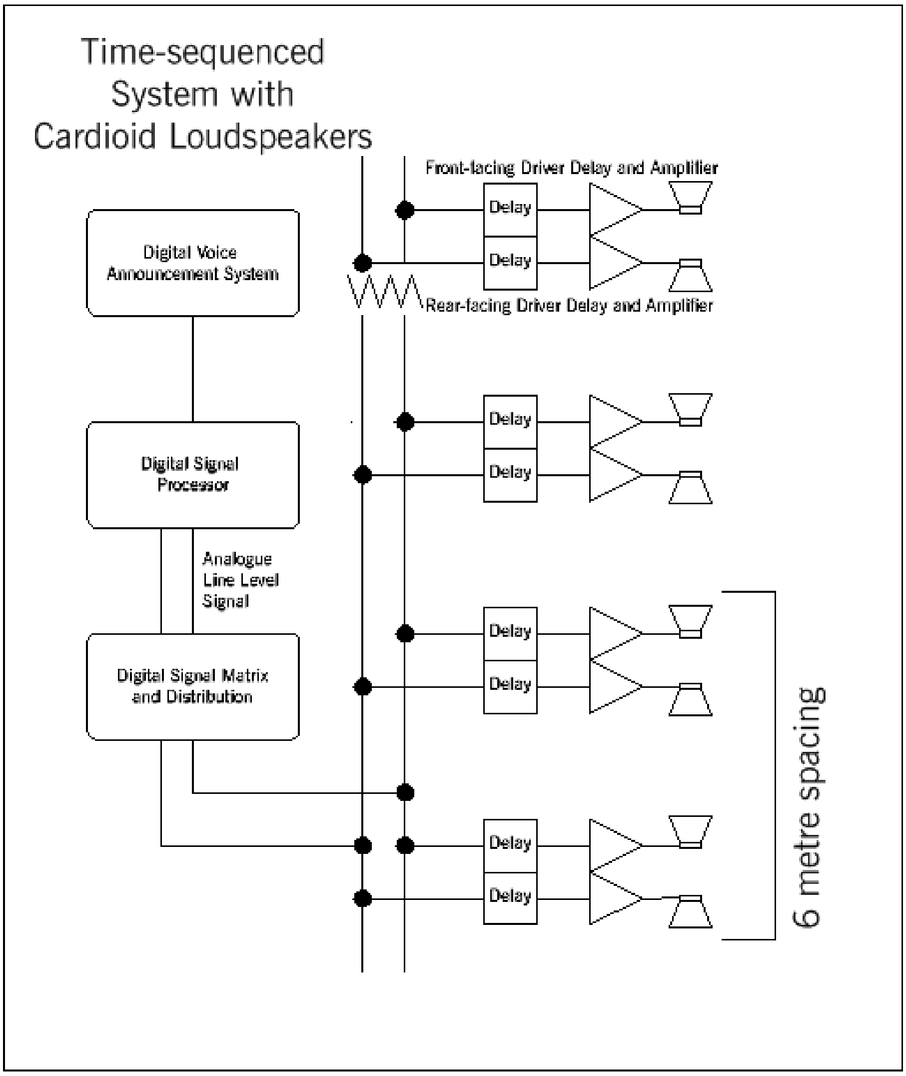

Figure 16: Signal Processing Architecture for steered time-sequenced system

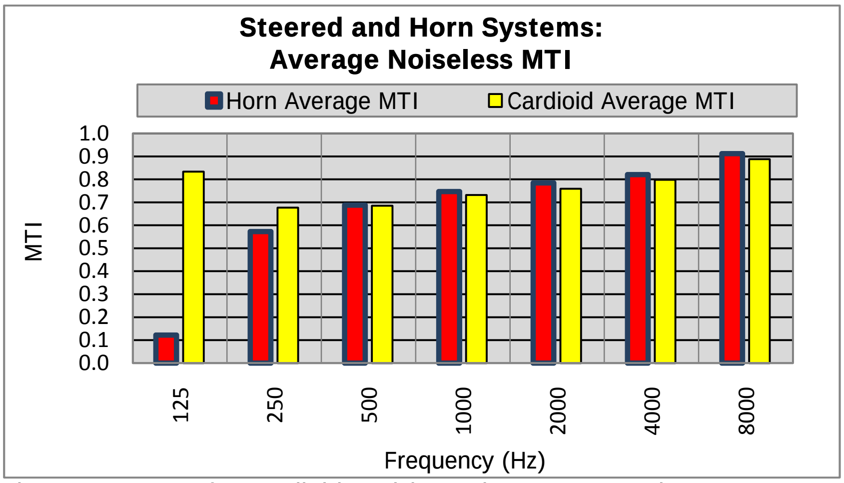

Figure 17: MTIs for cardioid and horn time-sequenced systems on adjacent platforms at Central Station

6.1.2 Frequency response

The responses of the steered system and time-sequenced horn system are compared in Figure 18 and Figure 19. These graphs indicate the increased bandwidth capability of the cardioid loudspeaker. The dip in the frequency response at 250 Hz is due to equalisation for the response of the automated announcement system, which was not ideally recorded or dynamically processed.

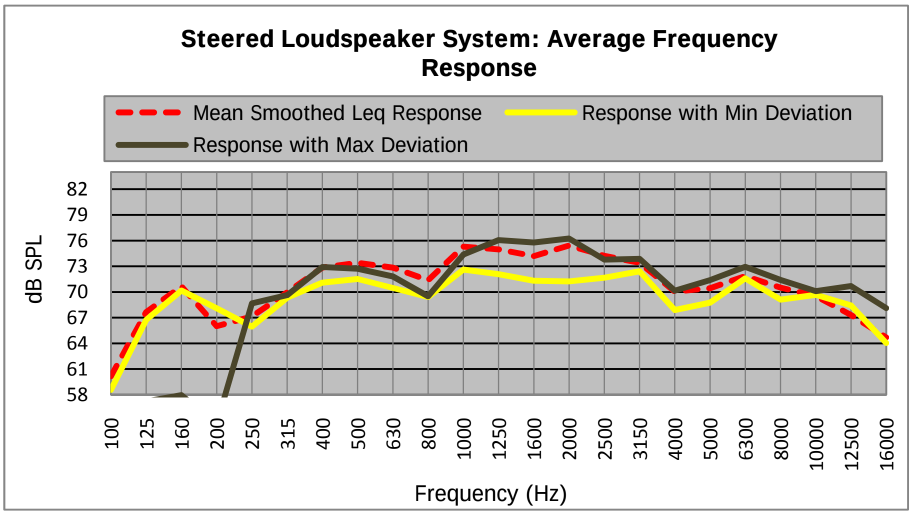

Figure 18: Average frequency response and responses with maximum and minimum deviations of steered time-sequenced system on Central Station. The data was measured as Leq in each 1/3rd octave band.

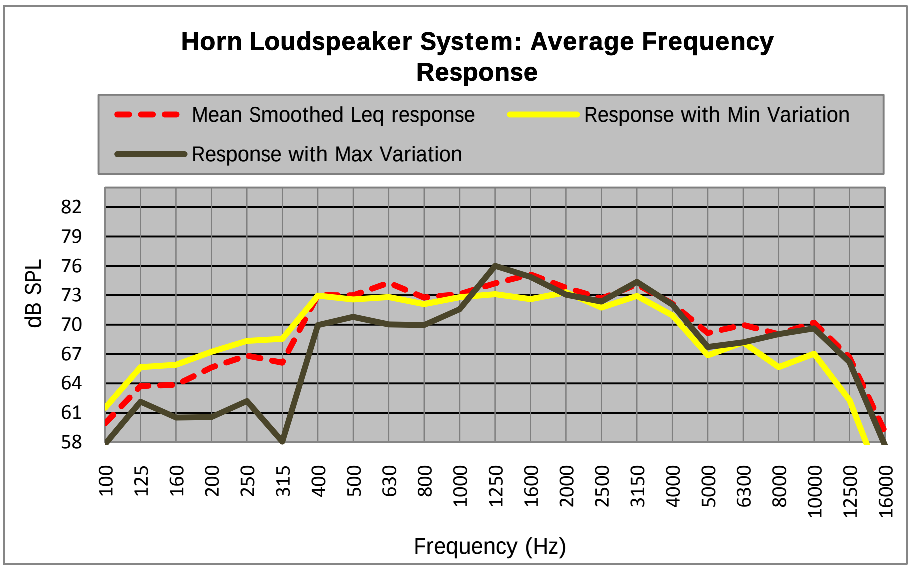

Figure 19: Average frequency response and responses with maximum and minimum deviations of horn time-sequenced system on Central Station. The data was measured as Leq in each 1/3rd octave band.

6.1.3 Spill to other platforms

Minimising spill between platforms was considered an important indicator of system suitability, as lower spill between platforms allows commuters to concentrate on local announcements and minimises confusion.

The following figures indicate small differences in spill between the two systems. Features of Figure 20 are the dip in the centre of the platform due to shielding from platform buildings, and the build-up of energy toward at the end of the platform toward which the loudspeakers were facing.

The average A weighted speech levels of the systems on neighbouring platforms are identical, as shown in Figure 21. It is of interest that the steered loudspeaker system produced more low-mid energy at the listening positions without any penalty in terms of spill.

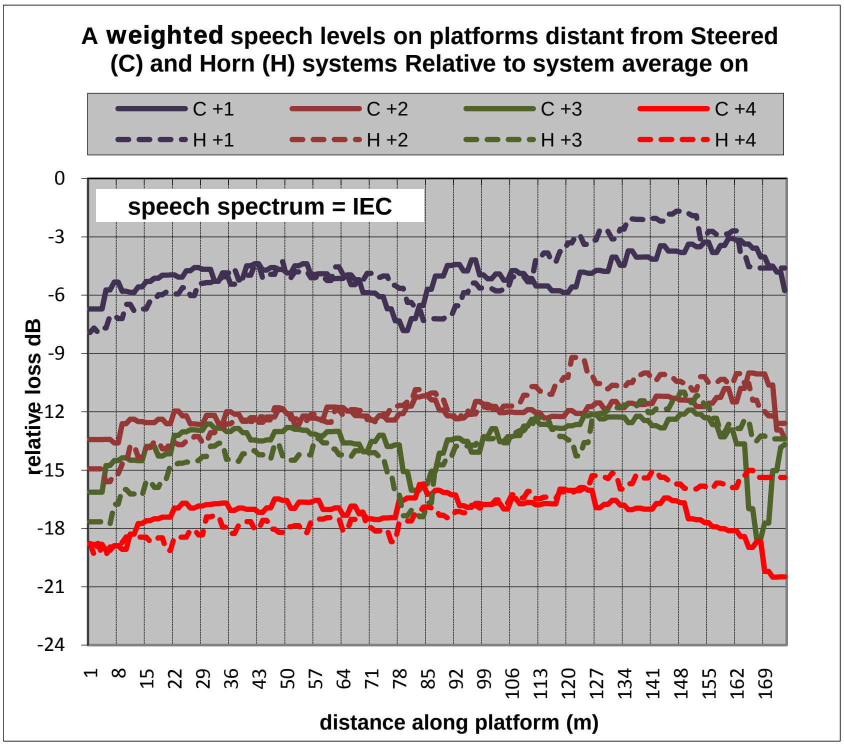

Figure 20: Comparison of inter-platform spill, A weighted, between horn and Cardioid systems

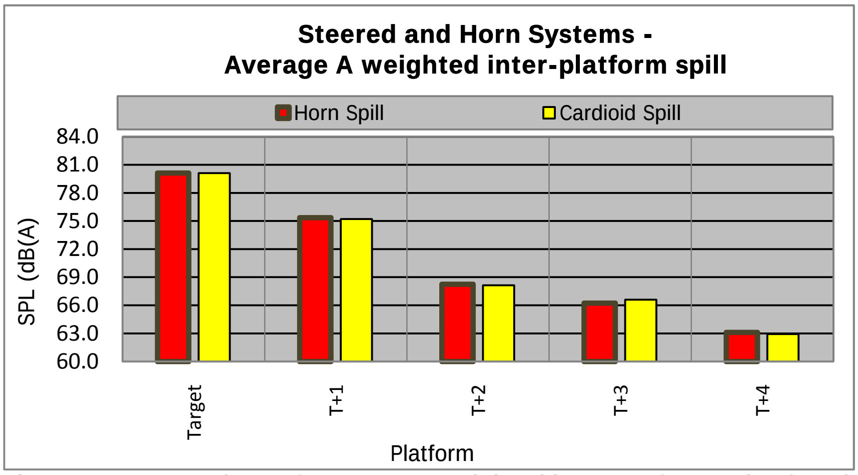

Figure 21: Comparison of average A weighted inter-platform spill of each system.
"T+n" indicates the number of platforms away from the platform with the loudspeakers.

6.1.4 Response consistency

Figure 22 plots the distribution of the consistency of frequency responses for the two systems over approximately 300 measurement points. It indicates that the variance in the frequency responses of the cardioid system from the mean steered loudspeaker system response is considerably lower than that for the horn system. The steered (cardioid) time-sequenced system therefore sounds better in more positions than the horn-based system.

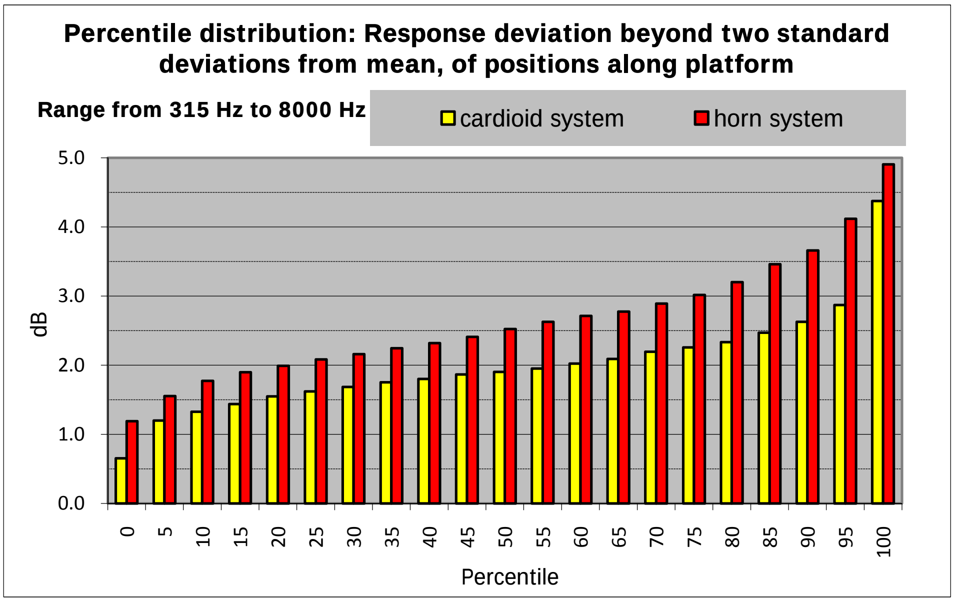

Figure 22: Response Deviation for Steered and Horn systems

6.1.5 Qualitative assessments

Listening tests were conducted, including (mostly) untrained listeners from Railcorp. Conclusions from these listening tests were that the steered (cardioid) system was an appreciably better sounding system. This was possibly due to the extra bandwidth presented at the listening position, and the increased consistency in frequency response.

7 DISCUSSION OF AUTOMATIC NOISE COMPENSATION

7.1 Implementation of noise compensation system

The background noise levels on the platforms at Central station are highly variable with conditions ranging from a late-night state with no trains or commuters to a noisy train at the platform during peak-hour. The levels range from 45 dBA to upwards of 75 dBA (LA_{eq} 30 s) or 85 dBC (80 dBA) lasting for some 5 s.

To provide satisfactory intelligibility and a comfortable listening experience under this wide range of conditions, it was necessary to provide a system that automatically adjusted the level of speech to match the noise environment.

Sensing of the platform noise levels was provided by two pressure zone microphones located at 1/3 and 2/3 the length of the platform on the underside of the platform canopy. These locations were found to be useful in achieving a balance between the noise of trains arriving and leaving.

The noise compensation algorithm was implemented in digital signal processing with adjustments of ratio and time constant being available. The sensed noise was band limited to frequencies between 135 Hz and 5000 Hz, to prevent low-frequency rumble and short-term high-frequency squeals and hiss from affecting the level of speech.

7.2 Intelligibility and noise compensation ratio

On the basis of maintaining a satisfactory STI performance in the presence of a train, it would seem appropriate to increase the volume of the sound system by one dB for every one dB of increase in background noise. This of course ignores the effects of the upward-masking algorithm in STI, which reduces the STI as the level of speech progressively increases. However, the effect of this masking reduction only becomes significant at speech levels L_{eq} above 90 dBA, and therefore this effect can be essentially ignored here.

Announcement levels were adjusted to lie within a comfortable range for listening. It was found that when the announcement levels tracked background noise levels at a ratio of 1:1, the announcements were intolerably loud, even in the presence of high levels of background noise. The comfort of listeners was compromised and to a small extent, the subjective intelligibility was also degraded.

It was found that a 0.58 dB increase in announcement level for every 1 dB increase in background noise level satisfied the requirements of listener comfort and intelligibility. The softest levels of announcement were appropriate for the quietest times of the day.

For the Central Station platforms, highest intelligibility without discomfort was found with LC_{eq} speech levels ranging between 62 dBC and 79 dBC (58 dBA to 76 dBA) measured over the duration of the announcement). Note that the use of dBC is preferred, as the A weighting scale removes a significant portion of the speech energy from the measurement.

Although comparing subjective signal to noise ratios with transient events such as speech and train noise is fraught with difficulty, in broad terms, the ratios of the speech and noise range from some 16 dB with quiet conditions to -5 dB with noisy conditions. The result of -5 dB is an unusual outcome for the rail environment in which STI measurements mostly form the basis of performance.

8 CONCLUSIONS

The systems presented in this paper represent a significant improvement in performance from previous sound systems on railway platforms in New South Wales and probably most parts of the world.

All three systems ((i) herringbone, ii) time-sequenced with horn loudspeaker and iii) time-sequenced with steered loudspeaker) provide excellent, measureable results and provide customer and client satisfaction. However, system ii) provides incremental improvements over system i) with system iii) providing incremental improvements over system ii). These improvements may be summarised as follows:

a) Improvements of time-sequenced horn system over herringbone system

- Greater ability to "lock-onto" the sound by commuters to aid intelligibility during simultaneous announcements on both sides of an island platform

- Improved perception of intelligibility due a shorter window of arrival times
- Less "confused" sound due to the small time window for arrivals and the directional component of the sound
- Generally better average response due to more consistent interference effects between near-by loudspeakers
- Fewer loudspeakers on the platform
- Improved ability to tailor the system to the local environment to avoid unnecessary spill to local residents
- Improved possibilities for enhanced maintenance regime due to monitoring of individual loudspeakers and amplifiers
- Higher clarity ratio

b) Improvements of time-sequenced system with steered loudspeakers over the time-sequenced horn system:

- Improved bandwidth allows more natural sounding voice replay without increased spill
- Well controlled high frequency dispersion characteristics allowing greater consistency of platform coverage and frequency responses.

The time-sequenced cardioid system was installed on the 23 above ground platforms at Central Station Sydney. The implementation of this system has been a great success due to the consistently high intelligibility achieved under all operational circumstances.

While the cost of the steered time-sequenced system was more than that of the time-sequenced system using horn loudspeakers, the benefits gained of commuter satisfaction and robust intelligibility have been considered to outweigh its additional cost.

9 REFERENCES

1. J.M. Steeneken, T. Houtgast. A physical method for measuring speech-transmission quality. *Journal of the Acoustical Society of America.* 1980, Vol. 67, 1, pp. 318-326
2. G. Leembruggen, A. Stacy. Should the Matrix be Reloaded? *Proc IOA.* 2003.
3. P. Mapp. Some Effects of Equalisation on Sound System Intelligibility and Measurement. *Preprint AES 115th Convention.* 2003.
4. T. Steinbrecher, Speech Transmission Index: Too weak in time and frequency? *Proc IOA.* 2008, Vol. 30, Part 6
5. M.A. Ericson and R. McKinley. 'The intelligibility of multiple talkers separated spatially in noise'. United States Air Force Research Laboratory, Human Effectiveness Directorate, Final Report for Period December 1987 to September 1993.
6. A.W. Bronkhorst, 'The cocktail party phenomenon: A review of research on speech intelligibility in multiple-talker conditions, AUSTICA Vol86 (2000), pp117-128.
7. M.L. Hawley, R. Litovsky, H.S. Colburn, Speech Intelligibility and localization in a multi-source environment, J. Acoustic Soc. Am. 105 (6), June 1999, pp 3436 – 3448.