AN INVESTIGATION OF ACOUSTIC PROPERTIES IN NON-CLASSICAL / SEMI-OPEN AREAS.

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1 INTRODUCTION

Working as an acoustic consultant can sometimes be very demanding due to high expectations from clients. In most new or refurbishment projects involving PA/VA design work, there are targets to be achieved based on documented standards or custom requirements whether related to speech intelligibility and/or Sound Pressure Levels. A client generally wants to be told whether or not their area(s) will be compliant before potentially undertaking remedial measures such as introducing acoustic treatment in the space or introducing more amplification if applicable.

In the case of semi-open spaces such as sheltered platforms on train stations, our experience has revealed that "border-line" cases regularly occur where the STI prediction results can sometimes fluctuate just above or below the relevant standard requirement. These scenarios can be delicate to handle since marginal errors in the predictions can imply unnecessary costly works to the building or worse still, an area can be said to comply when in fact, measurements later reveal the contrary.

This paper aims to improve acoustic predictions of STI (Speech Transmission Index) by reducing the error margin, using measurement survey data collected at several of these semi-open areas and mathematical predictions based on theory. In this early investigation, the aim is to derive a single multiplying factor to the direct-to-reverberant ratio specific to these non-classical areas.

2 METHODOLOGY

2.1 Theoretical Prediction

The Speech Transmission Index (STI) is a function of the following parameters:

i. Reverberation time,
ii. Signal-to-noise ratio
iii. Direct-to-reverberant ratio
iv. Receiver Q
v. Auditory masking
vi. Absolute hearing reception threshold

Equation 1 gives the formulae used for predicting the 98 indices (*m(F)'s*) as a function of the above parameters. The first factor in the equation is the focal point for the purpose of this investigation.

$$m(F)=\frac{\sqrt{\left(Q_r 10^{DRR/10}+\frac{1}{1+(\omega\tau)^2}\right)^2+\left(\frac{\omega\tau}{1+(\omega\tau)^2}\right)^2}}{Q_r 10^{DRR/10}+1}\cdot\frac{1}{1+10^{-SNR/10}}\cdot\frac{I_k}{I_k+I_{am,k}+I_{rs,k}} \quad (1)$$

where:

DRR is the direct-to-reverberant ratio at the receiver position in dB
Q_r is the receiver Q and acts as a *DRR* modifier
ω is the radial frequency $2\pi F$ in Hz
F is the modulation frequency in Hz
τ is the exponential time constant $T/6ln(10)$ in seconds
T is the reverberation time at the receive position in seconds
I_k is the mean intensity in octave band *k*
$I_{rs,k}$ is the mean intensity of the absolute hearing reception threshold in octave band *k*
$I_{am,k} = I_{k-1} \cdot amf$ is the mean intensity of the audio masking in octave band *k* where *amf* is the auditory masking factor (shown in table 1) for octave band *k* derived from the intensity of octave band *k-1*.

In the first instance, predicting the STI under noiseless conditions is the main objective. The equation element relating to the signal-to-noise ratio is of secondary interest at this stage since the system's output SPL and any ambient noise will simply vary up or down in-sync regardless of the space and its acoustics. On the other hand, the element which holds the key to improving any prediction of STI is held within the direct-to-reverberant ratio (*DRR*) element of the equation. In simple terms, the DRR is defined as the direct SPL minus the Reverberant SPL. When expanded and re-arranged, the equation becomes:

$$DRR = 10\log\left[\left(\frac{Q}{4\pi r^2}\right).\left(\frac{V}{25T}\right)\right] \qquad (2)$$

where:

Q is the source directivity
r is the distance from the source
V is the volume of the space
T is the reverberation time of the space

In the case of a classical type space, all the acoustic energy is assumed to remain inside the volume and the reverberant field is therefore dominant. While previous and more simplistic predictions of STI simply considered canopy areas to be classical, with a volume equal to the audience plane multiplied by the canopy height, it was found that STI results from post-design measurements were often higher than those predicted. The concept of an improved STI calculation method started from the idea of including of a multiplier factor to the effective volume of the space combined with a loss of acoustic power. In more simplistic terms, changes must be made to the reverberant field level such that the *m(F)* is adjusted accordingly.

However, in order to determine the correct *DRR*, the octave band levels of the direct and total fields need to be predicted based on anechoic chamber loudspeaker measurements and reverberant condition measurements respectively.

2.2 Direct field Measurements

With the knowledge of the loudspeaker models used on the stations and the inverse square law of "-6dB per doubling in distance", we are able to approximate the far-field of the loudspeakers under anechoic conditions through measurement. The process involves monitoring a pink noise signal from the loudspeaker at an on-axis point far-away from the source (within the limits of the chamber boundaries) while gradually reducing this distance by one-half until the far-field properties appear to break down. For accuracy, this process was repeated several times using a different initial distance between the loudspeaker and receiver microphone. It was eventually found that the frequency response and overall total SPL patterns began to change at distances below 400mm from the column loudspeakers and 300mm in the case of the recessed ceiling type loudspeaker.

The direct field of the loudspeakers was then measured in-situ on a covered train station platform where the space acoustics is known to be essentially reverberant. Figures 1 and 2 below give a comparison of the two frequency response visualisations of the column loudspeakers' direct field under anechoic conditions and on-site respectively, 400mm away from the loudspeaker grille. With the exception of some minor fluctuations, the overall responses tie up closely, hence confirming the potential for measuring the direct sound field of the loudspeakers installed at any site.

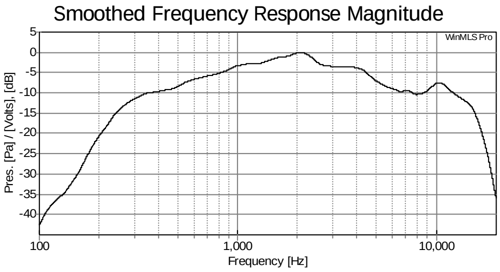

Figure 1 - Anechoic direct field response @ 400mm

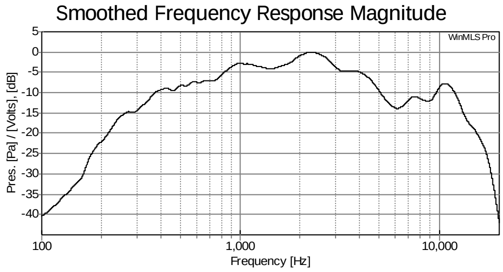

Figure 2 - On-site direct-field response @ 400mm

2.3 Site Survey and Measurements

Two main types of canopy platforms were considered during this investigation. Most of the areas measured (referred to as semi-open) consist of a stand-alone platform covered by a roof and closed off by a wall at one side (the opposite side being open to the rail track). These were all fitted with column loudspeakers mounted to the back wall and facing towards the platform edge. The second type of area (referred to as fully-open) consists of a covered island platform with both sides open to free-space. At these sites, the platforms were covered by ceiling loudspeakers recessed into the roof structure.

Using a CD player and a pre-amplifier set to 0dBu electrical output, two pre-recorded signals were injected through the station's PA zone of interest. The first is a continuous speech-shaped noise signal used to measure the octave-band SPL of the system, while the second is the WinMLS swept sine wave signal used to derive most acoustic parameters of the space such as Reverberation Time, frequency response and, to some degree, STI. The WinMLS signal used on site is a 12-second logarithmically swept sine wave with flat frequency response ranging from 40Hz to 22 kHz.
While monitoring the direct SPL from the loudspeakers at 400mm on-axis with the speech-shaped noise signal, the "normal" site measurements (representative of a real audience) were taken at randomly chosen positions on the platforms at a height of 1.5m affl. In this case, both speech-shaped noise and swept sine wave signals were recorded onto DAT for post-processing and further analysis.

In addition to the above site measurements, detailed notes about the PA system setup and layout (loudspeaker positions, angles, heights, etc…) and field measurement locations were made for producing an acoustic model of the in-situ measurement scenario.

2.4 Acoustic Modeling

For the third step of the investigation process, an accurate 3D acoustic model replica of each area was built with the CAD software CATT Acoustics. Having gathered all the necessary information from the site survey as described in section 2.3, the acoustic model sound sources were calibrated based on the direct field information obtained from the close-up measurements. Once the virtual loudspeakers are adjusted, potential amplifier non-linearity and/or system loss along the signal path is no longer of concern, since all has been accounted for as part of the measured direct SPL.

After the receiver (audience) positions have also been replicated in the model, their direct sound pressure level octave bands (125Hz to 8kHz) are captured in the model for each receiver. The direct levels for each position (Figure 3) are gathered and substituted into equation 2 for the calculation of the *DRR*.

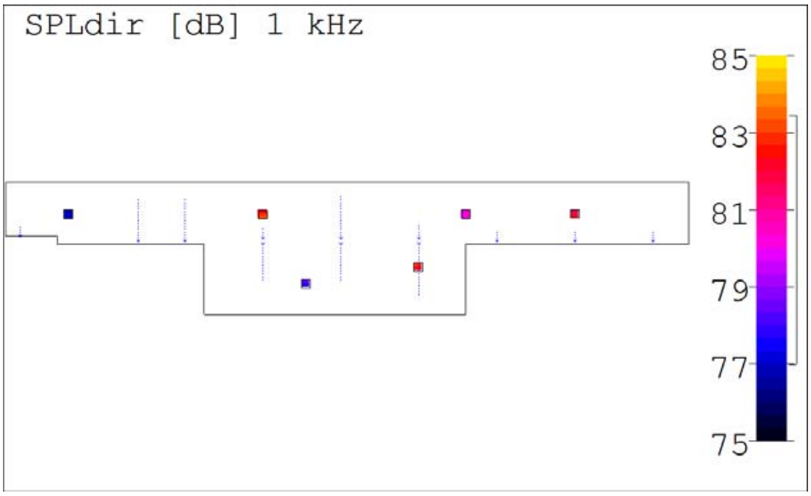

Figure 3 - Calibrated CATT model replica of site measurements

2.5 Revised STI Predictions

Since the true STI value for each position is known from the site measurements, the STI prediction calculations can be revisited at each receiver location until the actual STI is reproduced.
After adjusting the DRR for each position within the area, an average of the correction factors is calculated and thereafter used in a new set of STI predictions for the area as a whole. This final calculation also takes into account the average EDT for the area as well as any relevant deviation.

3 RESULTS

From the calculation process described in section 2.5, the following results were obtained for each semi-open canopy considered. For comparison purposes, Table 1 also includes the STI results obtained when no reverberant field correction factor is used (Old Classical Approach).

Area	Canopy Type	Loudspeaker Type	Old Predicted Average STI	Measured Average STI	New Predicted Average STI	Avg Rev-field Adjust (dB)
A	Semi-open	Column 1	0.617	0.655	0.656	-6.7
B	Semi-open	Column 1	0.600	0.633	0.630	-6.3
C	Semi-open	Column 1	0.653	0.690	0.689	-6.5
D	Semi-open	Column 1	0.671	0.705	0.705	-5.3
E	Semi-open	Column 2	0.597	0.650	0.644	-6.3
F	Semi-open	Column 2	0.623	0.657	0.653	-4.7

Table 1

With the exception of areas D and F which require a slightly inferior reverberant field adjustment parameter, the above results appear to follow a similar trend and, themselves, give an overall total average of -6dB. Such a reverberant field adjustment could be translated into a classical space with an effective volume four times greater than that initially assumed with the old method.

It is however strongly believed that a loss in acoustic power also plays its part in such areas, hence ruling out the possibility of an effective volume change alone. In terms of STI results, Table 1 clearly shows an improvement in the predictions using the adapted method, with an average prediction error of less than 0.06 compared to the measured STI.

Figure 4 shows, in graphical form, the position specific STI values (measured and predicted) for Canopy A using the derived reverberant field correction of -6.7dB. This visualisation also highlights the improvement in using the newly modified *m(F)* equation.

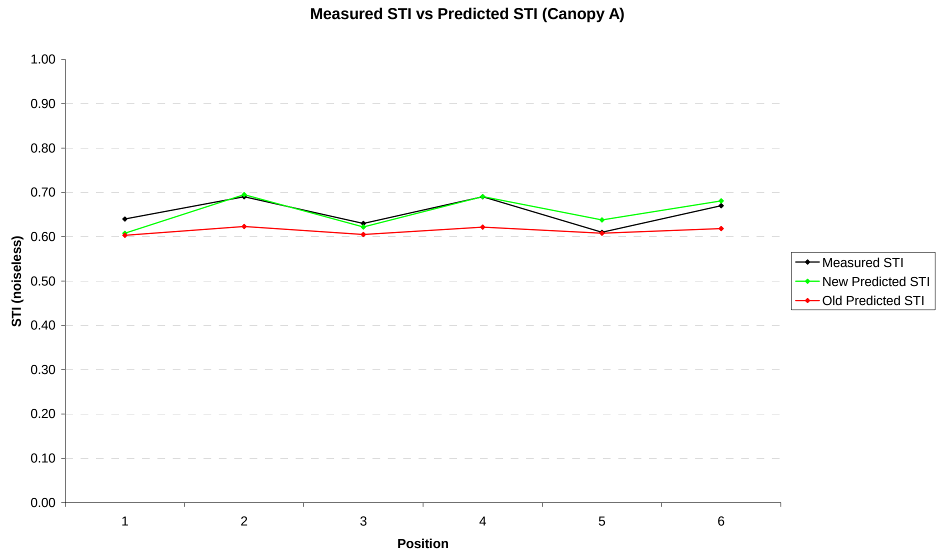

Figure 4 - STI comparison for Area A

Using the derived average reverberant field correction of -6dB for areas A through F, the STI for all positions was calculated again. The resulting figures are plotted against a best-fit line representing the target STI values extracted from measured data. As Figure 5 shows, there is once again an overall improvement when adjusting the reverberant field by -6dB, where the majority of green dots sit closest to the best-fit line.

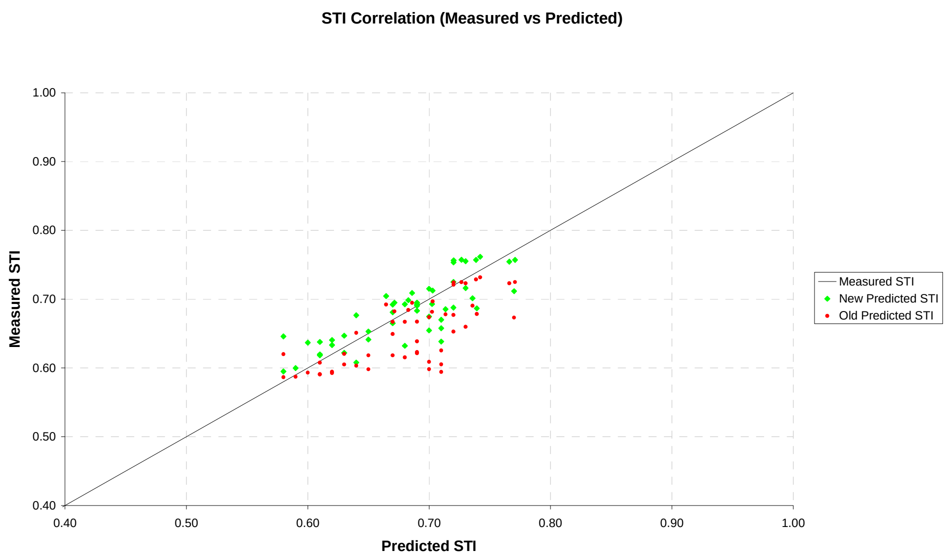

Figure 5 - STI Correlation for all receiver positions

In the case of canopy areas which are open to free space on either side (Island Platforms), the new predicted STI figures in Table 2 also prove to be a close match with those measured on site. However, in these cases, a reverberant field correction of around -2.5dB appears to be sufficient for obtaining accurate STI results.

Area	Canopy Type	Loudspeaker Type	Old Predicted Average STI	Measured Average STI	New Predicted Average STI	Avg Rev field Adjust (dB)
G	Fully open	Ceiling	0.706	0.729	0.729	-2.6
H	Fully open	Ceiling	0.712	0.734	0.733	-2.4

Table 2

4 CONCLUSIONS

The overall procedure of this investigation has highlighted the improvement that can be achieved between predicted STI figures and measured data. Consequently, averaging the individual reverberant field adjustment parameters for each position within an area has proven to be a good method for greatly improving mathematical STI predictions.

The current results derived from the measurements and calculations for semi-open canopies show a repeatable trend with regards to the reverberant field correction parameter. It is therefore believed that a mathematical increase of up to 6dB in DRR can be applied to the *m(F)* calculations.

In the case of canopy areas open on both sides, the early results lean towards a promising start, although it would be premature to draw any firm conclusions at this early stage of the investigation. However, these results contradict what a logical approach would expect, i.e. that the required reverberant field adjustment would be greater than for semi-open areas. A preliminary justification would point towards a combination of much lower reverberation times of the space and the very different type and positioning of the loudspeakers.

5 FURTHER RESEARCH

With the main objective of confirming the promising first set of results, further measurements would be carried out on more areas presenting similar acoustic features than those investigated to date.
So far, this research has been limited to distributed PA systems and using only three different loudspeaker types. Further work would involve investigating various loudspeaker layouts in order to compare the results with those presented in this paper. As part of this process, an in-depth research, as to why smaller reverberant field adjustments are required for open canopy areas, would be beneficial to further understand the acoustics of such areas.
Finally, predictions of acoustic parameters such as Reverberation Times would be performed through statistical calculations and ray-tracing while introducing the findings from this paper.

6 REFERENCES

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