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SOUND LEVEL DISTRIBUTION IN ROOMS: IS IT REALLY THE SAME EVERYWHERE ?

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1 INTRODUCTION

Practitioners regularly use classical statistical theory that predicts the same reflected sound level at all points in a room. For auditoria, measurement data[1,2] show this to be an oversimplification as the reflected levels significantly decrease with increasing distance from the source. From simple reasoning that the onset of reflected sound cannot occur prior to the arrival of the direct sound, Barron and Lee[2] developed a revised theory that predicts the average behaviour of sound levels in auditoria more accurately. So far this revised theory has only been applied to auditoria, where there is a particular acoustic environment with virtually all the acoustic absorption (the audience) on one surface. Variations of revised theory have been proposed for application in reverberation rooms[3] and specific types of churches[4,5].

A diffuse sound field is taken as the reference condition for room acoustics and a model investigation is reported here that tests whether this revised theory is applicable to proportionate spaces with diffuse sound fields. Two commonly accepted ways to promote a diffuse sound field are non-parallel geometry and scattering surfaces. The results given below are for measurements in two physical acoustic scale models, each having one of these basic characteristics. Results are also presented from acoustic prediction software.

2 REVISED THEORY

Sound received in a space from a single source can be split into a direct and reflected component. The prediction of the direct component is common in all approaches and accepted as giving an accurate result. The following discussion considers only the reflected component, which does differ significantly between prediction methods. To obtain the total sound level by any of these methods would require the addition of the direct component to the expressions given below for the reflected component.

Equation (1) is the classical statistical expression for the reflected sound intensity, i. Intensity is taken here with its usual room acoustics meaning as the sound power per unit area arriving at a point from <u>all</u> directions.

$$i = \frac{4W}{A} \quad (1)$$

where W is the sound power of the source and A is the total acoustic absorption. This equation does not account for receiver (or source) locations and it therefore predicts the same reflected sound level throughout a space. Equation (1) is usually derived through consideration of the balance of acoustic energy introduced into and absorbed in a room. An alternative derivation comes from an image model, which predicts exponentially decaying sound in a space. The reflected level is obtained by summing the energy arriving at a point, from the time of direct sound leaving the source to infinity. Implicit in this alternative derivation is that the onset of the reflected sound occurs simultaneously at all points throughout the space at the moment the direct sound is emitted.

Revised theory[2] can be derived by reasoning that, as reflected sound cannot arrive earlier than direct sound, the arrival time of the direct sound can also be taken as the onset time of the reflected sound at that location. During a sound decay the instantaneous sound level at any moment is still assumed to be the same at all points in the space and the sound is still taken to decay exponentially. Therefore, the longer the delayed onset the lower the reflected sound level will be at a given point. This gives a revised value for reflected intensity:

$$i = \left(\frac{4W}{A}\right)e^{-0.04r/T} \quad (2)$$

where r is the distance from the source to the receiver and T is the reverberation time. The effect of eq. (2) is that the reflected sound level decreases at a rate of 0.174/T dB/m as one moves away from the source. As the average absorption coefficient in a space decreases, this sound level reduction with distance becomes less. With a small average absorption coefficient eq. (2) gives a very similar result to eq. (1). For spaces such as reverberation chambers there is little benefit in using eq. (2) rather than eq. (1) whereas for auditoria, which have a higher average absorption coefficient, there is a significant difference between the two results.

Vorländer[3] proposes that rather than starting the onset of reflected sound at the arrival of the direct sound it should be started at the arrival of the first reflection. He argues that the average arrival time of the first reflection can be approximated as the time for sound to travel the mean free path. This gives equation (3), which is proposed primarily for reverberation chambers where the measurement results are spatially averaged.

$$i = \left(\frac{4W}{A}\right)e^{-A/S} \quad (3)$$

where S is the total area of the room's surfaces. Vorländer goes on to show that if the Eyring absorption exponent is used, ignoring air absorption so that $A = S\ln(1-\bar{a})$, then eq. (3) can be written as:

$$i = \left(\frac{4W}{A}\right)(1-\bar{a}) \quad (4)$$

Equation (4) is a well established and commonly used modification of eq. (1), notionally to account for the decrease in reflected level due to sound absorbed at the first reflection, but there has been no robust derivation for this prior to Vorländer's work.

Several recent measurement exercises in religious buildings[4-7] also show reflected sound levels to generally decrease with increasing source-receiver distance, contrary to classical theory, eq. (1). However, the reflected levels in these religious buildings generally fall below those predicted by revised theory and in their study of Gothic-Mudejar churches Sendra et al[4] introduce an adjustment parameter, ß, to equation (2) giving:

$$i = \left(\frac{4W}{A}\right)e^{-br/T} \quad (5)$$

For each octave band, average values of ß are empirically derived from the measurement data and equation (5) is then proposed for general use in this specific type of church. During a study of Apulian-Romanesque churches Cirillo and Martellotta[5] theoretically derive an alternative adjustment to equation (2). Like Vorländer, they identify that the onset of reflected sound does not in reality start at the arrival time of the direct sound. Their solution is to assume that the delay is proportional to the source-receiver distance and derive a correction on this basis. Cirillo and Martellotta also derive a correction to account for the specific early reflection pattern. While these corrections do not rely on empirical data, the formulation of them makes many approximations and results in a complex set of expressions only tested for these Apulian-Romanesque churches.

Other approaches include Cotana's proposal[8] to subdivide a room into zones and then predict the sound level in each zone using a weighting factor calculated from the strength of the first order reflections to that zone for a given source location. The use of this weighting for all reflections is not theoretically sound and there is no evidence of the accuracy of the model other than Cotana's own test space. For a computer simulation of a cuboid space, Franzoni and Labrozzi[9] find an empirical relationship that predicts the average cross sectional value of the reflected sound level as it decreases with distance from the source. Labrozzi[10] also proposes an alternative equation for this situation from energy conservation principles but it remains complex, relatively unproven, and limited in possible applications.

Revised theory, eq. (2), does not account for the distribution of absorption or room geometry and defines receiver positions solely in terms of their distance from the source. These assumptions allow revised theory to retain most of the simplicity of the classical statistical formula. As previously mentioned, it has been shown to be significantly more accurate than classical theory for auditoria. The question behind this investigation is whether these simplifying assumptions still allow a sufficiently robust prediction for other spaces and specifically in the reference condition of a proportionate space with a diffuse field. Alternative methods, illustrated by those presented above, tend to have limited application and introduce such complexities that they represent a more detailed level of analysis that would perhaps be better handled by standard computer modelling packages.

3 MODELLING

3.1 Measurement System

The use of physical scale modelling is an established technique in room acoustics[11], although for this investigation key characteristics of the University of Bath's modelling system were reviewed: verification measurements were made of transducer directivity, spark source linearity, and air absorption corrections. In this investigation a greater accuracy is desired than is needed for say auditorium modelling.

The modelling system uses two microphones. To account for slight variations in the spark source level the energy of the direct sound is measured with the second reference microphone. This also allows for the direct sound being obscured at some locations of the first microphone. For both models the spark source and two microphones are suspended in adjustable height tubes held in randomly distributed sockets on the models' top surfaces. Impulse responses are obtained by direct measurement of spark discharges, then stored and sequentially filtered in third octaves / octaves. All normal room acoustics parameters are obtained using the standard reverse integration but with tail correction to extend the dynamic range available.

The upper frequency used in the models is determined by the directivity of the 1/8" microphones, which is omnidirectional to within 2dB up to the 16kHz third octave band. The lower frequency limit (5kHz third octave band) is determined by the energy from a single spark that still maintains a suitable signal to noise ratio. In the context of this investigation the absolute full size equivalent frequencies are not important so the scale factor used is slightly arbitrary. For convenience, 1:25 was chosen since it gives results in a familiar frequency range: 250Hz and 500Hz octave bands. To avoid confusion, results and discussion below are always for the full size equivalent frequencies and distances.

To account for the frequency variation of air absorption, previous modelling at the University of Bath has been conducted in dry air or nitrogen. However, for this investigation, the larger sizes of the models make these options less practical and instead, the measurements presented here were made in air with numerical corrections for absorption from ISO 9613-1[12]. Equations in ISO 9613 allow calculation of the absorption of sound per metre travelled for given temperature, humidity and pressure. The absorption due to nitrogen and oxygen is calculated separately so it is straightforward to calculate for 100% nitrogen. To correct for air absorption the attenuation is calculated for air in standard conditions (20C, 50% R.H., 1013.25 mb) and the actual measurement conditions. Each time step of the measured impulse response is then corrected using these values to give the equivalent response in air at standard conditions. The validity of these equations is demonstrated in Figure 1, which shows sample decays from measurements that were conducted in air and then repeated in nitrogen. The first graph shows the two decays with no correction for air absorption and the second graph shows the same decays following application of the appropriate corrections. These graphs illustrate that a penalty of measuring in air rather than nitrogen is that there is a significant loss of dynamic range in the corrected decay.

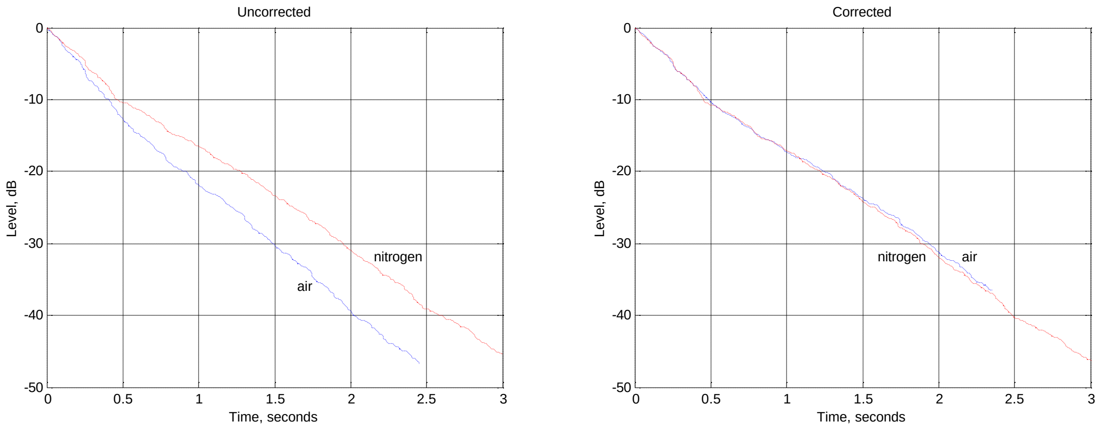

Figure 1 – Uncorrected/corrected decays in air and nitrogen

3.2 Model Constructions

Certain ratios of dimensions for rectangular spaces encourage a good distribution of modes and hence promote a diffuse sound field[13]. Two of these ratios are the basis for the models in this investigation: 1 / 0.7 / 0.59 (Model 1), 1 / 0.83 / 0.65 (Model 2). In Model 1, the ratio is only used to obtain a reasonable starting point as there is further distribution of modes by angling three of the walls by 5° (one from each pair of opposite walls). In Model 2 there are not simple internal dimensions, due to the depth of the semi-cylindrical scattering surfaces, and there is some deviation from the chosen ratio. The longest dimension of both models is just over 1m actual size.

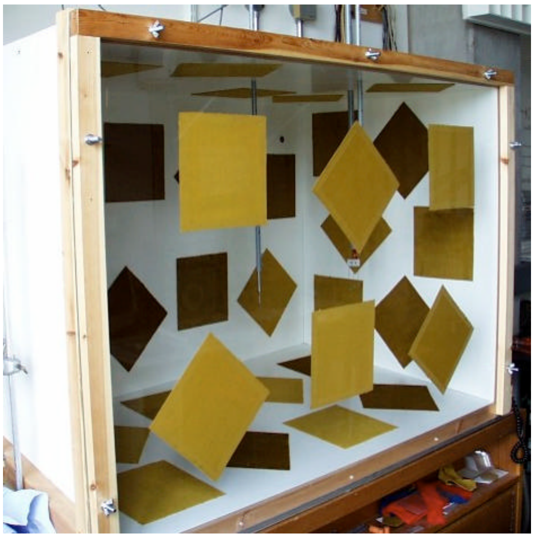

Figure 2 – Photograph of Model 1

Revised theory only predicts a significant difference (>1dB) from classical theory when the average absorption coefficient in a space is greater than about 0.2 (as is the case in auditoria). For the purposes of testing revised theory a reasonable amount of absorption is therefore included in both models. This absorption is evenly distributed on all surfaces to maintain a diffuse sound field. Model 1 is shown in Figure 2 (the front surface is transparent acrylic), with 200mm square patches of velvet material providing absorption. Figure 3 shows Model 2 during construction; when completed, the 40mm deep wells, between the black semi-cylinders, are covered by strips of velvet material as porous absorbers.

Figure 3 – Photograph of Model 2 during construction

4 RESULTS

This investigation is concerned with revised theory, which predicts a difference only in reflected and not direct sound levels when compared to classical theory. Therefore, the key parameter here is the reflected sound level, denoted G_r.

$$G_r = 10\log_{10}\frac{\int_5^\infty p^2(t)dt}{\int_0^5 {p_{10}}^2(t)dt} \tag{6}$$

where p is the pressure at the receiver position and p_{10} is the pressure that would be at a position 10m from the source. The G_r measurement results have been plotted against source receiver distance as shown in Figure 4 for the 500Hz octave band in Model 2. This graph also shows the theoretical values for G_r according to classical (purple line) and revised (green line) theories and also the best-fit line through the measurement data (dotted red line). Results in Model 1 and at 250Hz in Model 2 have the same overall trends as seen in Figure 4.

For both models and octave frequencies overall statistics are given in Table 1. These are taken from the linear regression line for measured reflected sound levels against source-receiver distance and the revised theory predictions. Differences between measured and predicted values are included. The first set of values in this table is the mean reflected sound levels from the regression line and revised theory at the mean measurement distance, which happens to be the same in both models at 13.4m. The second set of values is the slopes of the revised theory prediction lines and regression lines, giving the decrease in reflected sound levels with distance from the source.

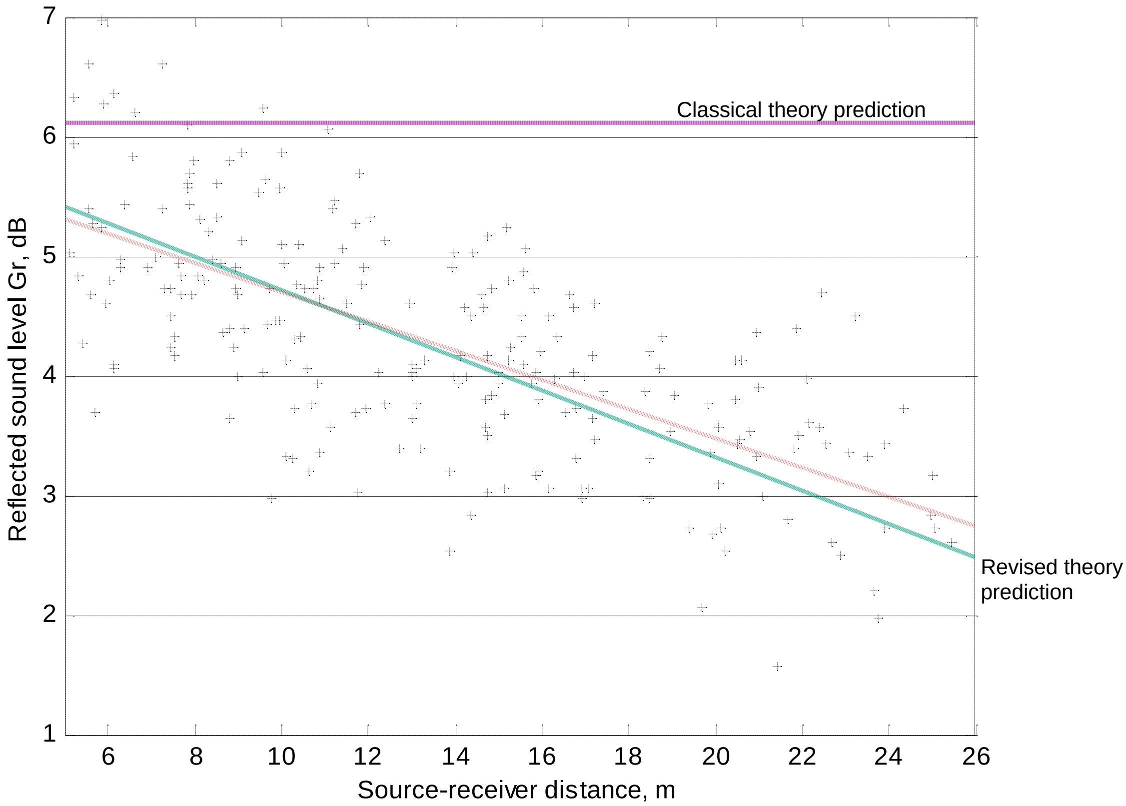

Figure 4 – Reflected sound levels measured in Model 2 (500Hz octave band)

		Mean level (dB)			Slope (dB/m)		
		Measured	Predicted	Difference	Measured	Predicted	Difference
Model 1	250Hz	9.7	9.7	0.0	-0.05	-0.05	0.00
	500Hz	6.9	6.6	-0.3	-0.08	-0.09	0.01
Model 2	250Hz	4.2	4.6	0.4	-0.17	-0.13	-0.04
	500Hz	4.3	4.2	0.0	-0.12	-0.14	0.02

Table 1 – Reflected sound level statistics

In parallel with the physical acoustic scale modelling, the spaces were analysed by computer. Figure 5 shows the same situation as Figure 4 but for predictions made using proprietary acoustic software (CATT Acoustic). The reflected sound levels have been normalised to account for the differing reverberation time predicted by CATT and measured in the model. These differences are attributable to difficulties in assigning the correct absorption and scattering coefficients to Model 2's complicated surfaces and also the inaccuracy of modelling undulating surfaces as a flat planes.

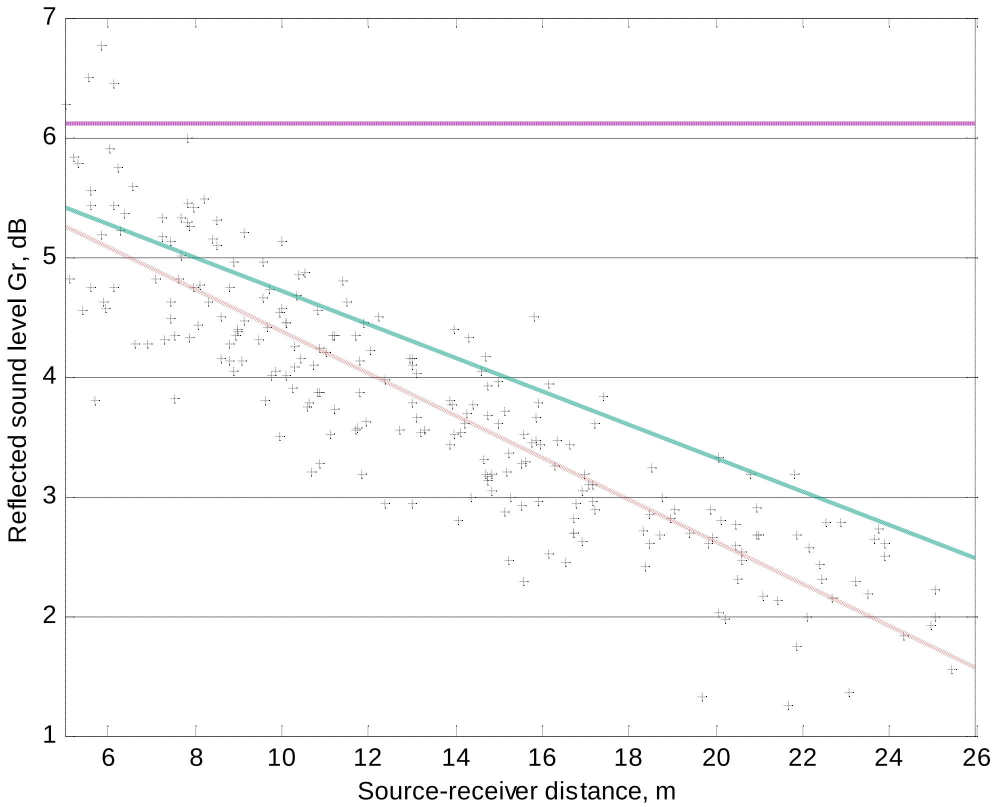

Figure 5 – Reflected sound levels predicted by CATT Acoustic for Model 2 (500Hz octave band)

5 DISCUSSION

From the results of the scale model measurements it appears that revised theory is substantially more accurate than classical theory for predicting sound levels in proportionate spaces with diffuse sound fields. For each model and each frequency there is clear evidence that reflected sound level decreases with distance from the source. Although the best-fit lines through the data often differ slightly from revised theory no systematic difference was identified and the mean differences are very small.

For the measured data, despite the close agreement of the average behaviour with revised theory, it can be seen in Figure 4 that there is significant scatter of the results around the best-fit line. While this initially caused some concern, it is explained by the theoretical fluctuation of sound levels within a space. This was investigated many years ago by Lubman[14] and Schroeder[15], who both arrived at expressions that lead to the following equation for the standard deviation of sound levels measured in a room:

$$s_{G_r} = \frac{4.34}{\sqrt{\left(1+\left(\frac{B\times T}{6.9}\right)\right)}}\,dB \qquad (7)$$

where B is the bandwidth (Hz) and T is the reverberation time (seconds). The derivations for this theoretical standard deviation start with a reverberation room, where the low average absorption coefficient means that revised theory would predict negligible variation with source-receiver distance (same result as classical theory). However, for this investigation the standard deviation also includes the intrinsic variation of sound levels with source-receiver distance, so no longer gives a true measure of the scatter. An alternative adopted here is to use the standard error taken from the linear regression of the measured reflected sound levels against source-receiver distance. Although interchanging standard deviation with standard error has not been proven theoretically, in the absence of an equation for sound level scatter that accounts for sound level decrease with distance, equation (7), used in this way, still gives useful values for comparison.

The theoretical standard deviation of the 500Hz octave band sound levels in Model 2 using eq. (7) is 0.54dB and the standard error of the measured data is 0.70dB. The standard errors for measurements in Model 1 and for the 250Hz octave band in Model 2 are also consistently higher than the theoretical standard deviation by a similar degree.

To investigate eq. (7) in more detail it is necessary to use data with a wider range for the two variables: reverberation time and bandwidth. This data was available from measurements made during the preparation of Model 1. The absorbing patches were originally added in 3 stages and measurements made at each stage, giving results for a wide range of reverberation times. These impulse responses have been analysed for the two octave bands at 250Hz and 500Hz and the six third octave bands from 200Hz to 630Hz, giving a range of bandwidths. The standard error for these measured sound levels is plotted below against the predicted standard deviation using eq. (7). For each set of data the two left hand points represent the 500Hz and 250Hz octave bands and the other six points to the right are for the third octave bands from 630Hz to 200Hz (there is an exception for the set of data on the left, 'no absorption', where there is no 500Hz octave band point). This data shows a general agreement between the measured standard error and predicted standard deviation, although it also highlights the consistent deviation at the two frequency bands of prime interest in this study: 250Hz and 500Hz.

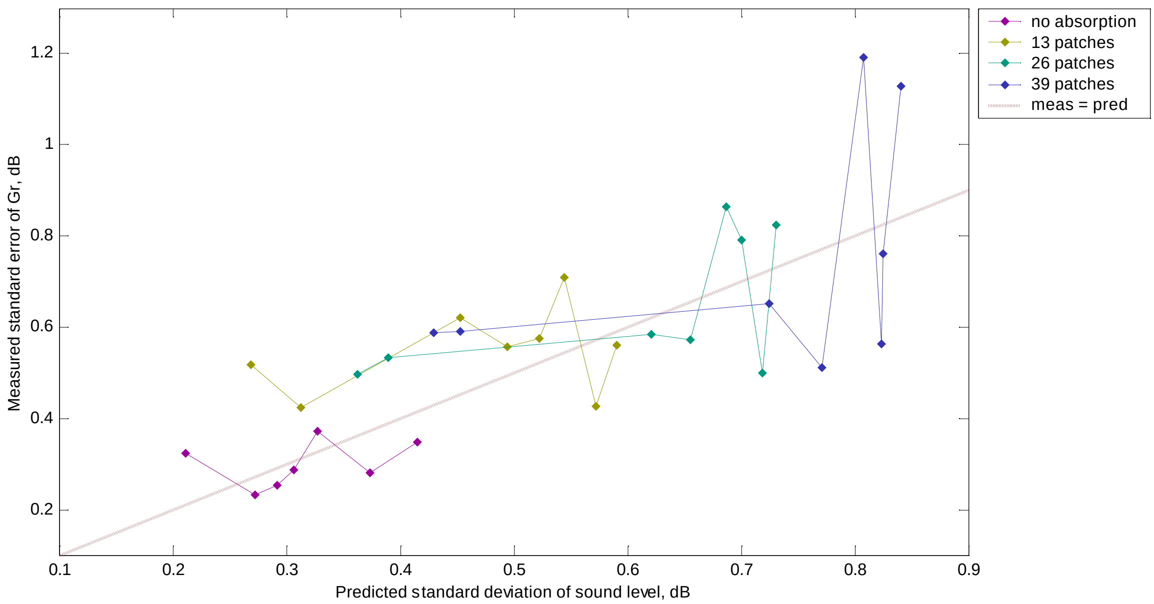

Figure 6 – Standard error of reflected sound levels compared with theory

This exercise has demonstrated that the scatter of sound levels in diffuse sound fields is significant and can be predicted to an extent by an existing theoretical model. However, the theoretical model originated for reverberation rooms, with low average absorption coefficients, and there remain discrepancies with values measured in the spaces studied here, which need to be resolved.

Looking at Figure 5 for the CATT predictions it can be seen that there is significantly less scatter than for the scale model measurements in Figure 4. This CATT data (and CATT data for Model 1) has a lower standard error than the theoretical standard deviation, particularly at lower frequencies although not significantly at higher frequencies, which are of most interest. The reason for the lesser scatter was assumed to be due to the exclusion of phase information in the computer modelling. To test this hypothesis, Model 1 was reanalysed in CATT using the output from the auralisation module, which includes an approximation for phase information.

The results below show the theoretical standard deviation and measured/predicted standard errors for the sound levels in Model 1. These results show that the most likely reason for the lower scatter of the original CATT results is due to the omission of phase in the standard calculations.

	250Hz	500Hz
Theoretical standard deviation from Equation (7)	0.46dB	0.44dB
Standard errors		
Measured in scale model	0.56dB	0.56dB
Predicted in CATT Acoustic	0.19dB	0.46dB
Predicted in CATT Acoustic with phase approximation	0.71dB	0.70dB

When phase is included in CATT, results for individual positions do not match measured values any closer as the inclusion of phase is only an approximation. This is probably why there is still a significant difference between scatter of CATT predictions and measured results. As accuracy of individual results is not improved, there is no benefit in introducing phase approximations to routine computer modelling, although for lower frequencies in particular one should be aware of the prediction uncertainty due to sound level scatter.

6 CONCLUSIONS

Revised theory has previously been proven to predict reflected sound levels in auditoria significantly more accurately than classical theory. From measurements in two scale models of proportionate spaces with diffuse sound fields it has now been shown that revised theory is also significantly more accurate than classical theory in this situation. The revised theory predictions are close to the best-fit lines through the data and it is concluded that it can be used for analysis of the average behaviour in these types of spaces. The scatter of reflected sound levels measured is significant but it is of a similar order to a theoretical value calculated using an equation for reverberation rooms. The omission of phase in computer modelling programmes reduces the scatter of predicted sound level results at lower frequencies.

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