A Simple Image Method for Calculating the Distribution of Sound Pressure Levels within an Enclosure

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A Simple Image Method for Calculating the Distribution of Sound Pressure Levels within an Enclosure

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Summary

By use of a computer, a simple image method is used to predict variation in sound pressure level throughout a rectangular room possessing various absorption configurations. A model room was constructed and it is shown that the computer prediction correlates better with measured values than does the classical prediction which assumes a diffuse sound field.

Eine einfache Bildmethode zur Berechnung der Verteilung von Schalldruckniveaus in einem Hohlraum

Zusammenfassung

Mit Hilfe eines Computers wird eine einfache Abbildungsmethode benutzt zur Vorhersage des Schalldrucks in einem rechtwinkligen Raum, der verschiedene Absorptionsfigurationen besitzt. Ein Modellraum wurde konstruiert, und es wird gezeigt, daß die Computer-Vorhersage besser mit gemessenen Werten korreliert als die klassische Vorhersage, welche ein diffuses Schallfeld voraussetzt.

Méthode à image simple pour calculer la répartition des niveaux de pression du son dans une enceinte

Sommaire

En utilisant un calculateur, on emploie une méthode à image simple pour prédire la variation d'un niveau de pression sonore d'un bout à l'autre d'une salle rectangulaire possédant diverses configurations d'absorption.

On construit une salle modèle et l'on montre que la prédiction du calculateur correspond mieux aux valeurs mesurées que ne le fait la prédiction classique qui suppose diffus le champ sonore.

1. Introduction

Acoustics is concerned usually with relating the absorption characteristics of the room boundaries to sound energy density in the room, in both the steady and transient states. Sabine [1], Eyring [2] and Millington [3] have derived formulae for reverberation time, and Knudsen [4] has derived an expression for the steady state condition. Sabine assumed that the sound field was diffuse, i.e. that the energy density was constant throughout the room, and that at any point the energy flow was equally distributed in all directions. Eyring assumed that the acoustic behaviour of a room was determined by the average absorption coefficient $\bar{\alpha}$ of the walls and considered a wave front travelling a fixed distance (the mean free path) between successive reflections at which it lost the same fraction of its incident energy. Millington considered that a room was characterised by the behaviour of a specific sound ray suffering successive reflections at surfaces whose absorption coefficients varied: not unnaturally this approach predicts a reverberation time of zero when the ray under consideration strikes a surface having unity absorption coefficient. Sabine's expression for reverberation time, T,

$$T = \frac{0.16\,V}{S\,\bar{\alpha}}$$

where V is the room volume in m^3 and S its total surface area in m^2, fails when $\bar{\alpha}$ is large, and predicts a reverberation time of $0.16\,V/S$ when $\bar{\alpha}$ is unity.

Eyring's result

$$T = \frac{0.16\,V}{-S\log_e(1-\bar{\alpha})}$$

is more generally accepted. Knudsen attempted to measure the steady state average energy density E_0 for a constant power output Q from the source. Using the relationship,

$$\bar{\alpha} = \frac{4}{c\,S}\,\frac{Q}{E_0}$$

the absorption coefficient of the room could be obtained. The energy density E_0 in question was that of the reverberant part of the sound field and thus readings would need to be taken at a distance from the source.

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In general the sound field in a room is thought of as being made up of two parts; that from the sound source which would have resulted in a free-field condition, and a contribution resulting from many reflections at the enclosure boundaries; the direct and reverberant contributions respectively. An omnidirectional source of sound of power Q watts will give a sound energy intensity of $Q/4\pi r^2$ W/m² at a distance of r metres. The reverberant contribution is calculated as being equal to $4\,Q/R$, where R is the room constant defined by the equation,

$$R = \frac{\bar{\alpha} S}{1-\bar{\alpha}}.$$

The room average reverberant absorption coefficient $\bar{\alpha}$ is modified to

$$\bar{\alpha} + \frac{4\,m\,V}{S}$$

when taking into account the air attenuation coefficient m.

The total energy intensity at a point r metres from the sound source is then:

$$I = Q\left(\frac{1}{4\pi r^2} + \frac{4}{R}\right).$$

Using the definition of sound pressure level,

$$S.\,P.\,L. = 10 \log_{10} \frac{p^2}{4 \cdot 10^{-10}}$$

that of sound power level as,

$$P.\,W.\,L. = 10 \log_{10} \frac{Q}{10^{-12}}$$

the sound pressure level within the room is given by,

$$S.P.L. = P.W.L. + 10 \log_{10}\left(\frac{1}{4\pi r^2} + \frac{4}{R}\right). \qquad (1)$$

From eq. (1) it is seen that on plotting contours of sound pressure level at intervals of 1 dB regular circles are produced (Fig. 1). The spacing between the contour lines increases with increasing distance from the source, indicating that the $S.\,P.\,L.$ is tending asymptotically to a finite value at large distances. This value is given by the equation,

$$S.P.L. = P.W.L. + 10 \log_{10} \frac{4}{R}.$$

Common experience has shown that the sound field in most enclosures is not diffuse. Appreciable variations in the angular and spatial distribution of the steady-state and transient sound energy have been observed [5], [6], [7].

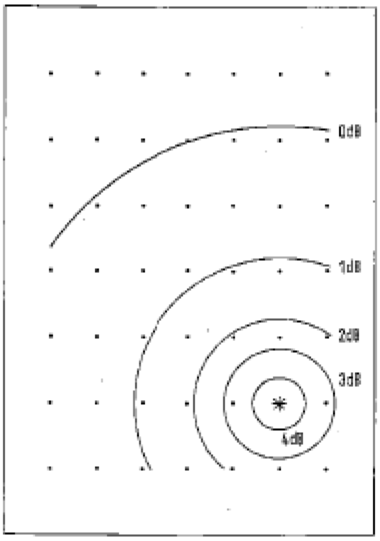

Fig. 1. *S.P.L.* distribution predicted classically for the single and six patch configuration. The asterisk indicates the omnidirectional sound source.

This non-diffuseness, with the resultant departure from classical theory, can be ascribed to several phenomena. Often, in enclosures, the absorbent material is not distributed evenly over the surface area. In any situation in which the wavelength of sound is of the same order of magnitude as the room dimensions then the wave characteristics of the sound must be considered. Standing waves will be set up and, in the case of reverberation time measurement, anomalies will result from the non-uniform damping of the room modes.

In a situation in which one room dimension is greatly different from the other two the concept of the mean free path of the sound 'rays' becomes less meaningful. GOMPERTS [8] evolved a modification to EYRING's reverberation formula which did not rely on assuming a constant mean free path. The resultant formula was strongly dependent upon the ratio of the room dimensions. An example of an enclosure of widely differing dimensions is that of an open plan office. DAY [9] has shown that in this case the sound intensity does not fall off with distance from the source as predicted classically. Anomalies produced in this form of enclosure must also result from the uneven distribution of absorbants since only the floor and ceiling are used as absorbing surfaces.

The image approach uses the optical analogy in which a sound source placed in front of a plane, smooth, reflective surface will produce an image sound source at an equal distance from the plane. The effective power of the image source is determined by the absorption of the plane.

POWER [10] used this assumption to predict the acoustic response of large offices. If the source has a power Q then an image source was assumed to have a power given by,

$$Q_n = (1-\bar{\alpha})^n Q$$

where n is the order of the image. To simplify the problem the ceiling was assumed to be totally absorbent. The images of the source were therefore confined to two planes; one through the real source and the second resulting from the first reflection from the floor. The energy density E_p at the receiver was given as the summation of the energy densities due to each point,

$$E_p = \sum_{n=0}^{\infty} \frac{Q_n}{4\pi r_n^2 c}$$

where r_n is the distance from the receiver to the n-th image source. A time dependent expression was used to obtain the reverberation time.

$$E_{pt} = \sum_{r=ct}^{\infty} \frac{Q_n}{4\pi r_n^2 c}$$

where E_{pt} is the energy density at a time t after the sound source is removed.

In a rectangular enclosure with no extreme dimension a three dimensional regular array of image sources will result from introducing a real source into the room. MAYO [11] generalized this three dimensional array into line and plane arrays and was able to show that two characteristic frequencies existed, one of which was the set of eigentones. DOAK [12] considered a three dimensional array of images in assessing the reverberant contribution to the sound field. By considering the attenuation of sound by the inverse square law, air and boundary absorption, the spatial average value of the contribution from the image sources was given by the integral,

$$\langle P_s^2 \rangle = \frac{1}{V} \int_R P_0^2 \left(\frac{r_0}{r}\right)^2 (1-\bar{\alpha})^{\bar{N}} \exp(-m_a r)\, \mathrm{d}r$$

where R is the volume outside the room occupied by the image sources and $\bar{N}$ is the number of reflections experienced by a ray travelling from the image source to the receiver. The exponential term describes the air absorption. Using the classical value of $4V/S$ for mean free path, $\bar{N}$ has a value $rS/(4V)$. Considering an effective radius of the room as being $(3V/4\pi)^{1/3}$ and using EYRING's reverberation time formula, the expression becomes,

$$\langle P_s^2 \rangle = 302.84\, P_0^2 r_0^2 \left(\frac{T}{V}\right) \exp\left[\left(\frac{-0.0257}{T}\right) V^{1/3}\right]$$

where P_0 is the value of pressure obtained at a distance r_0 in free field conditions.

A simple image approach using a computer and attempting to rectify some of the classical postulates whilst preserving ease of application is now described.

In attempting to extend the image theories described in predicting the spatial variation of sound energy several assumptions are necessary. To eliminate the wave characteristics of the sound the ratio of wavelength: room dimension must be sufficiently small and the source must be wide band. Use of a frequency of 2 kHz within a room of smallest dimension 2.24 m gives a ratio of approximately 0.08 which is thought adequately small. By using a one third octave band of white noise as source, the contributions from the image sources could be combined by simple addition. Finally the sound source and receiver are assumed to be omnidirectional.

2. Theory

Consider a rectangular room of dimensions α, β, γ (Fig. 2) metres in the x, y, z directions of a rectangular co-ordinate system, the origin of which is at the room centre. Into it are placed a sound source, at co-ordinates x_s, y_s, z_s, and a receiver at x_r, y_r,

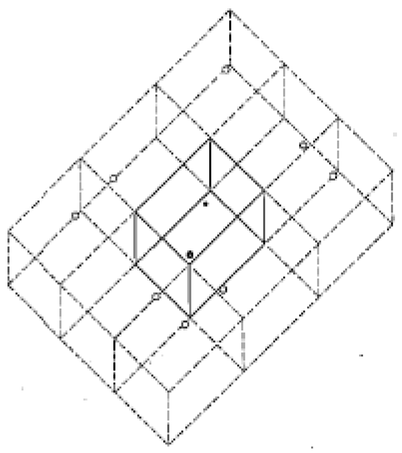

Fig. 2. Nearest neighbour image cells in the plane of the room.
● Real sound source,
○ image source,
• receiver.

z_r. If the six surfaces are smooth and have a reasonably low value of absorption coefficient, a three dimensional array of image sources will be produced outside the room volume, each in its own image room. To describe each room or cell the integers l, m, n will be used. Thus the real room will be described as the 0, 0, 0 cell while its neighbour along the positive x axis is the 1, 0, 0. A cell is a 'mirror' image of its neighbour and will be orientated as the real cell only if l, m, n are all even.

The distance vector $\mathbf{d}$ from the real receiver to an image source in cell l, m, n will have components given by,

$$d_x = (\alpha\, l \pm x_s) - x_r,$$
$$d_y = (\beta\, m \pm y_s) - y_r,$$
$$d_z = (\gamma\, n \pm z_s) - z_r.$$

The positive term in the bracket must be used when l, m, n are positive, similarly for negative integers the negative sign is used.

The distance is given simply by,

$$d_{l,m,n} = \sqrt{(d_x^2 + d_y^2 + d_z^2)}.$$

For an image sound source of power Q watts at a distance d metres from the receiver the intensity at the receiver is given by,

$$I = \frac{Q}{4\,\pi\, d_{l,m,n}^2}\ \mathrm{W/m^2}. \tag{2}$$

The number of boundary reflections suffered by a sound ray assumed to have come from an image source in cell l, m, n can be seen to be equal to $N_b = |l| + |m| + |n|$. Therefore modifying eq. (2) to include boundary absorption gives,

$$I = \frac{Q}{4\,\pi\, d_{l,m,n}^2}\,(1-\alpha')^{N_b}$$

where α' is the absorption coefficient of the surface averaged over all angles of incidence. Multiplication by an exponential term describes the air absorption,

$$I = \frac{Q}{4\,\pi\, d_{l,m,n}^2}\,(1-\alpha')^{N_b}\exp(-m_a\, d_{l,m,n}).$$

Summation of all the contributions from real and imaginary sources gives the total sound intensity at the receiver,

$$I_t = \sum\sum\sum \frac{Q}{4\,\pi\, d_{l,m,n}^2}\,(1-\alpha')^{N_b}\exp(-m_a\, d_{l,m,n}). \tag{3}$$

The range of values of the integers l, m, n can be determined with respect to the desired accuracy of the prediction.

If one writes eq. (3) in the form

$$I_t = K\, I_0$$

then if

$$I_0 = \frac{Q}{4\pi}\exp(-m_a)$$

is the intensity at a distance of 1 m from the sound source in free field conditions and

$$K = \Sigma\,\Sigma\,\Sigma\, d_{l,m,n}^{-2}\,(1-\alpha')^{N_b}\exp m_a(1 - d_{l,m,n})$$

the total sound pressure level at the receiver is given by

$$S.\,P.\,L._t = S.\,P.\,L._{1m} + 10\log_{10} K. \tag{4}$$

2.1. *Room with one surface covered by highly absorbent material*

Let one wall, say the one described by the expression,

$$x = \frac{\alpha}{2}$$

be covered by a material possessing an absorption coefficient α'' greater than that of the remaining five surfaces. For an image cell l, m, n there will be a total of $N_b = |l| + |m| + |n|$ reflections of which l_2 will involve the wall with the higher absorption coefficient. l_2 is given by the expression,

$$l_2 = l/2 \qquad \text{for } l \text{ even}$$
$$l_2 = (l \pm 1)/2 \text{ for } l \text{ odd} \begin{cases}\text{positive}\\ \text{negative.}\end{cases}$$

Therefore the factor K in eq. (4) now becomes

$$K = \Sigma\,\Sigma\,\Sigma\, d_{l,m,n}^{-2}\,(1-\alpha')^{N_b - l_2}(1-\alpha'')^{l_2} \times \times \exp m_a(1 - d_{l,m,n}).$$

2.2. *Room with six patches*

Let the single patch of the previous section be divided into six equal areas of absorber, one placed on each room surface. The sound ray path in the room represented by the distance vector $\mathbf{d}_{l,m,n}$ will suffer a reflection involving an absorbing patch only if the vector cuts a real or image patch. The coordinates of intersection between the distance vector and image plane must be calculated. If the point of intersection lies within the area of the image patch then the absorption coefficient of that reflection assumes a value α'' otherwise the reflection involves an absorption coefficient α'.

Let the patches, which are placed centrally on each wall have dimensions a, b, c in the x, y, z directions respectively; they are assumed to have no thickness. Considering the reflections involving the $x = \pm\alpha/2$ walls; of the l reflections experienced by a ray represented by an image ray from cell l, m, n, let l_2 be the number of these reflections occurring at a patch of absorber. The image planes of concern can be represented by,

$$x = a(l_i \pm 1/2) \quad \text{for} \quad l \gtreqless 0$$

where integer l_i has values given by

$$-l \leqq l_i \leqq l.$$

On each of these planes will exist an array of patches expressed by the inequalities

$$\begin{aligned} &\left(m_i\beta - \frac{b}{2}\right) \leqq y \leqq \left(m_i\beta + \frac{b}{2}\right) \\ &\left(n_i\gamma - \frac{c}{2}\right) \leqq z \leqq \left(n_i\gamma + \frac{c}{2}\right) \quad \text{where} \\ &-m \leqq m_i \leqq m\,, \quad -n \leqq n_i \leqq n\,. \end{aligned} \tag{5}$$

By calculating the co-ordinates of the intersection of a ray and a plane and substituting the values into the inequalities it is possible to deduce whether a reflection occurs at a patch or a bare wall area.

To derive the relationship between the co-ordinates of intersection x, y, z consider the distance line d from an image source x_s, y_s, z_s to a receiver x_r, y_r, z_r.

The line has direction cosines λ, μ, ν given by

$$\begin{aligned} \lambda &= (x_s - x_r)/d \\ \mu &= (y_s - y_r)/d \\ \nu &= (z_s - z_r)/d\,. \end{aligned}$$

Thus, any point x, y, z on this line must obey the relationship

$$(x - x_r)/\lambda = (y - y_r)/\mu = (z - z_r)/\nu\,.$$

Expressing y and z in terms of x

$$\begin{aligned} y &= \frac{\mu}{\lambda}(x - x_r) + y_r \\ z &= \frac{\nu}{\lambda}(x - x_r) + z_r\,. \end{aligned}$$

Thus, knowing x the y and z values are deduced and substituted into inequalities (5).

This process is repeated for the other wall pairs. To summarise, a ray from an image cell l, m, n suffers N_b reflections of which N_p reflections are at the surface of an absorbent patch. The equation for the factor K in eq. (4) now becomes,

$$\begin{aligned} K = \Sigma\Sigma\Sigma\, d_{l,m,n}^{-2}(1-\alpha')^{N_b - N_p}(1-\alpha'')^{N_p} \times \\ \times \exp m_a(1 - d_{l,m,n}) \ldots \end{aligned} \tag{6}$$

where $N_p = (l_2 + m_2 + n_2)$ and $N_b = |l| + |m| + |n|$).

2.3. *Modifications due to angular variation in absorption coefficient and diffraction effect*

It may be desirable to investigate the variation of absorption coefficient with angle of incidence of the sound ray.

Considering an image ray line and an image plane; the former is expressed by the vector,

$$\boldsymbol{d} = (x_s - x_r)\,\boldsymbol{i} + (y_s - y_r)\,\boldsymbol{j} + (z_s - z_r)\,\boldsymbol{k}$$

and the latter by,

$$\boldsymbol{P} = x\,\boldsymbol{i} + y\,\boldsymbol{j} + z\,\boldsymbol{k}$$

where $\boldsymbol{P}$ is the distance along the normal of the plane from the origin. The angle Θ between line vector and the normal of the plane is simply,

$$\Theta = \cos^{-1}\frac{|\boldsymbol{P}|\,|\boldsymbol{d}|}{\boldsymbol{P}\cdot\boldsymbol{d}}\,.$$

The planes described by,

$$x = a(l_i \pm \tfrac{1}{2})$$

have their normals parallel to the x-axis, thus

$$\Theta_x = \cos^{-1}\left(\frac{x_s - x_r}{d}\right).$$

Similarly, for the z and y planes,

$$\begin{aligned} \Theta_y &= \cos^{-1}\left(\frac{y_s - y_r}{d}\right) \\ \Theta_z &= \cos^{-1}\left(\frac{z_s - z_r}{d}\right). \end{aligned}$$

An absorption coefficient $\alpha(\Theta)$ related to the reverberant absorption coefficient $\bar{\alpha}$ by,

$$\bar{\alpha} = 2\int_0^{\pi/2} \alpha(\Theta)\,\cos\Theta\,\sin\Theta\,\mathrm{d}\Theta$$

could therefore be substituted directly into the eq. (6). The sound distribution in the model room was calculated assuming a random incidence and also assuming that the absorption coefficient varied with angle of incidence as postulated by Paris [13]. The difference between these two distributions was too slight to justify the additional computing time.

The phenomenon of diffraction can be interpreted as an effective increase in the area of the absorbing patch. A patch on the x wall would thus have an apparent area increase given by

$$(b + \Delta b)\,(c + \Delta c)$$

and the inequalities (5) are modified to

$$\begin{aligned} &\left[m_i\beta - \left(\frac{b+\Delta b}{2}\right)\right] \leqq y \leqq \left[m_i\beta + \left(\frac{b+\Delta b}{2}\right)\right] \\ &\left[n_i\gamma - \left(\frac{c+\Delta c}{2}\right)\right] \leqq z \leqq \left[n_i\gamma + \left(\frac{c+\Delta c}{2}\right)\right]. \end{aligned}$$

These simple modifications are difficult to apply because of the lack of reliable information as to the nature of the angular variation of $\alpha(\Theta)$ and also the diffraction effect.

3. The computer program

A computer makes possible the rapid calculation of simple equations and would thus seem ideal for the problem of evaluating the total intensity at different room positions.

Although the number of image sources is considered infinite, it is necessary to consider only the contributions from images within a finite sphere of influence, the centre of which coincides with the real room. The radius of this sphere of influence is dictated by the room absorption, the required accuracy of prediction, and the computer time available. The last point is important in that if the sphere radius is doubled the computation time is increased eight-fold.

To calculate the intensity at a receiver from an image at a distance r, simplify eq. (3) to give,

$$I = \frac{Q}{4\,\pi\,r^2}\,(1-\alpha')^{r/\bar{l}}\exp(-m_a\,r)$$

where $\bar{l}$ is the classical mean free path.
The total contribution from a sphere of radius a is obtained by integration,

$$I_a = \frac{Q}{V}\int_0^a (1-\alpha')^{r/\bar{l}}\exp(-m_a\,r)\,\mathrm{d}r\,.$$

Let $\qquad (1-\alpha')^{1/\bar{l}}\exp(-m_a) = C\,.$

The integral becomes $\qquad I_a = \dfrac{Q}{V}\displaystyle\int_0^a C^r\,\mathrm{d}r$

which gives $\qquad I_a = \dfrac{Q}{V\log_e C}\,(C^a - 1)\,.$

If the sphere is of infinite radius, the integral becomes,

$$I_\infty = -\frac{Q}{V\log_e C}\ \text{where}\ \log_e C < 0\,.$$

The $S.P.L.$ difference between the two values is given by

$$\Delta L = 10\log_{10}\left(\frac{1}{1-C^a}\right)\mathrm{dB}\,.$$

The radius a can therefore be expressed by the equation,

$$a = -\frac{\bar{l}\log_e\left(1-\dfrac{1}{\operatorname{antilog}_{10}(\Delta L/10)}\right)}{m\,\bar{l} - \log_e(1-\alpha')}\ \text{metres}$$

where ΔL is the arithmetic difference between the two values of $S.P.L._\infty$ and $S.P.L._a$.

To summarise, the radius of influence, a, is determined by the required accuracy of prediction and the average absorption coefficient of the room. It is seen that the radius decreases with increase in $S.P.L.$ difference or α'. It was decided that in any computer prediction the sound pressure level at a receiver position should only be 0.1 dB below the value which would have been obtained from summing this contribution from an infinite number of images. This difference would be well within experimental accuracy. In order to reduce the radius of influence and thus to save computer time, it was decided to increase the average absorption coefficient of the room.

To test the validity of the computer program produced a program was written for the rectangular room described in the previous section. Of dimensions 4.4 m × 3.08 m × 2.27 m the room surface had an absorption coefficient of about 0.27 except for the patches of high absorber which had a value of 0.88 (DF.120 50.8 mm sponge rubber). Results were obtained for the single and six patch configuration. The receiver was assumed to be at approximately one third room height and results were obtained for forty nine positions. Contour diagrams (Figs. 3 and 4) illustrate the nature of the distribution for the single and six patch configuration. The values of the reverberant absorption coefficients of the materials used were calculated, using a normal impedance concept, from impedance tube measurements.

As an approximation to the computer room model, a room was constructed of the above dimensions which was surfaced by celotex 12.5 mm wallboard

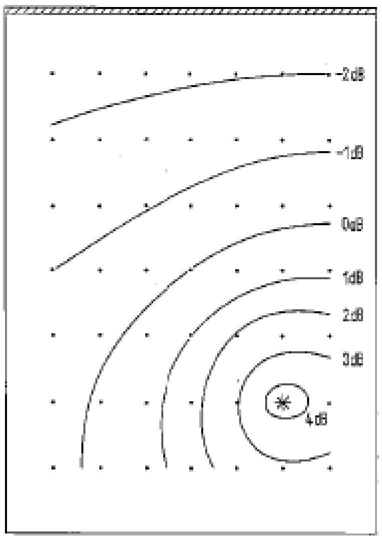

Fig. 3. $S.P.L.$ distribution predicted by computer for the single patch configuration. The shading indicates the 50.8 mm sponge rubber.

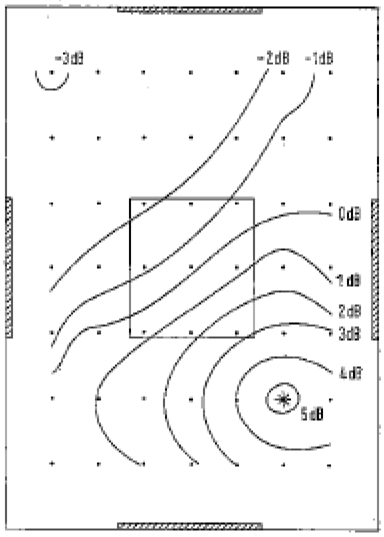

Fig. 4. *S.P.L.* distribution predicted by computer for the six patch configuration.

except for areas of the sponge rubber used in the single and six patch configurations. The omnidirectional sound source used was a pipe speaker which produced reasonable directional characteristics (Fig. 5) at 2 kHz.

The omnidirectional receiver used was a Bruel and Kjaer half-inch microphone. The microphone positions corresponded to the forty nine receiver positions considered in the computer programs. The sound source was filtered white noise of bandwidth 300 Hz centred on 2 kHz. The receiver circuit included a one-third octave filter (Fig. 6) and the sound pressure levels were recorded by a level recorder.

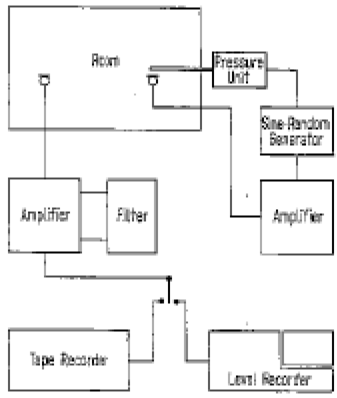

Fig. 6. Diagram of apparatus.

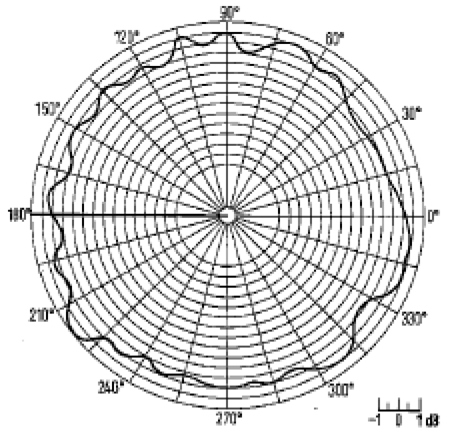

Fig. 5. Directional characteristic of the pipe speaker sound source at a frequency of 2 kHz.

4. Results

The results are plotted, as with the classical and computer predictions in the form of contour diagrams (Figs. 7 and 8) and it is apparent that the range of values predicted by computer approximates more closely to the experimental results than do the classical predictions.

Figs. 9 to 12 show the *S.P.L.*'s measured at each of the 49 positions, plotted against the predictions

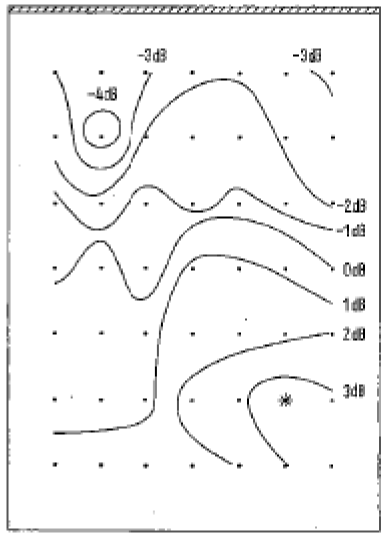

Fig. 7. Measured *S.P.L.* distribution for the single patch configuration.

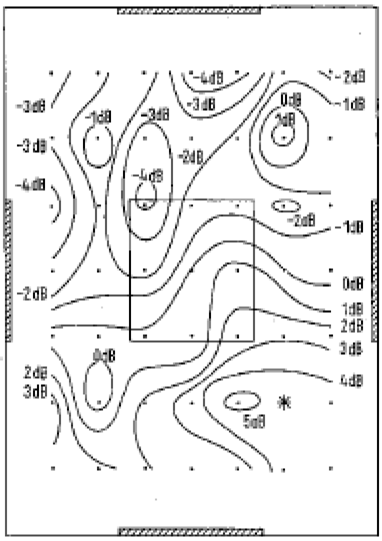

Fig. 8. Measured *S.P.L.* distribution for the six patch configuration.

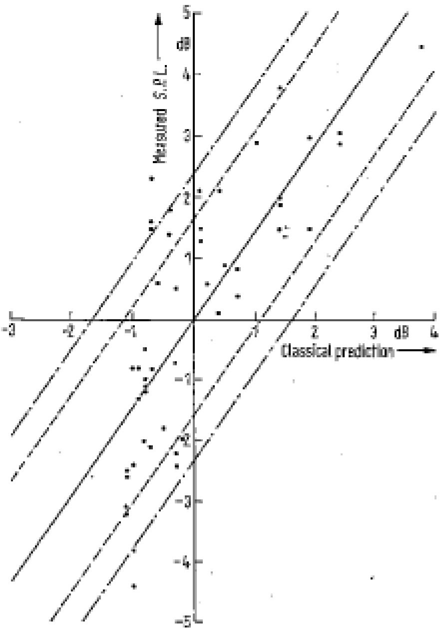

Fig. 9. Regression line and confidence limits of the measured *S.P.L.* distribution against that predicted classically for the single patch configuration. Frequency of 2 kHz.
— — — 75%,
—·—·— 90%.

computed from classical theory and from this image approach, together with the regression lines Table I shows the corresponding correlation coefficients, slopes of regression lines, and confidence limits.

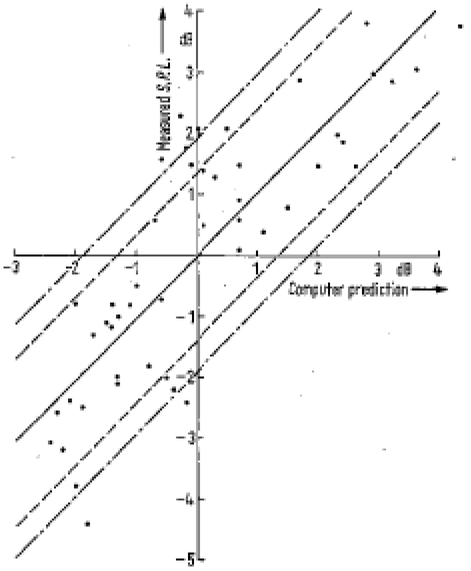

Fig. 10. Measured *S.P.L.* against the computer prediction for the single patch case.
— — — 75%,
—·—·— 90%.

It will be seen that both for the case of the absorption covering one wall and for the case when the absorption was distributed between all six surfaces of the room, that the prediction from the image approach gives significantly better agreement with the measured results. This improvement is most marked when the absorbent is distributed amongst all six surfaces of the room.

5. Conclusion

To summarise, the computer results gave better agreement with observed results than did the classical prediction. This improvement, however, was gained at the expense of a somewhat long computing time, especially for the six patch configuration.

It would appear however that the approach may well be useful in estimating sound pressure level variations in open plan offices, i.e. where the distribution of absorber is confined to the floor and ceiling areas and is such that those two most important absorbing areas can be described by two values of absorption coefficient.

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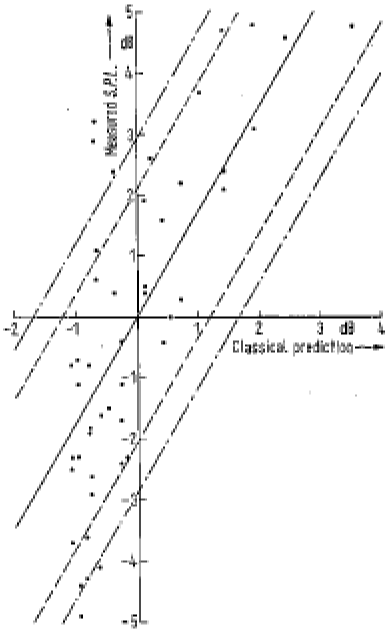

Fig. 11. Measured *S.P.L.* against that predicted classically for the six patch case.
——— 75%,
—·—·— 90%.

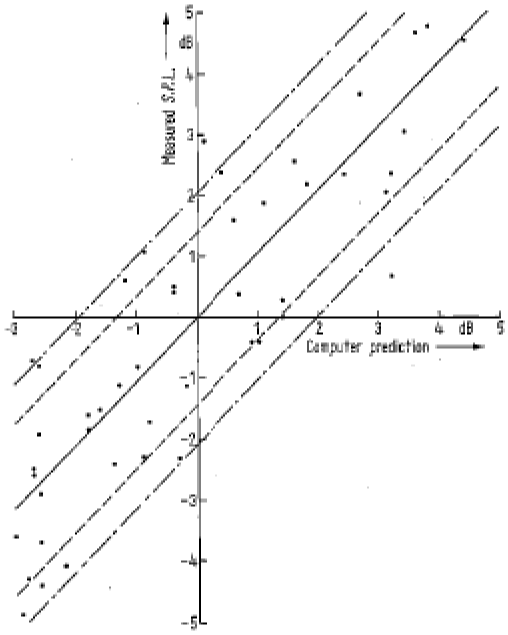

Fig. 12. Measured *S.P.L.* against the computer prediction for the six patch case.
——— 75%,
—·—·— 90%.

Table I.
Regressions and confidence limits.

Absorption configuration	Ordinate	Abscissa	Correlation coefficient	Regression	Confidence limits 75% dB	90% dB	95% dB	99% dB
Single patch	measured distribution	Classical prediction	0.759	1.453	1.7	2.4	2.8	3.8
	measured distribution	Computer prediction	0.845	1.305	1.3	1.9	2.3	3.1
Six patch	measured distribution	Classical prediction	0.790	1.960	2.0	2.9	3.5	4.6
	measured distribution	Computer prediction	0.898	1.062	1.4	2.0	2.5	3.3

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