RETROSPECTIVE ROOM ACOUSTICS

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1 INTRODUCTION

This lecture is to present an overview on the development of room acoustics during the second half of the past century. It begins by describing the state-of-the-art in the early fifties, a time when a certain demand for new and larger auditoria with - of course - excellent acoustics arose. The next part is devoted to the ideas which came up in the sixties and later, and describes how these ideas influenced our present picture of room acoustics. Because of limitations in time and space the author had to restrict his presentation to portraying a few trends which he deemed to be characteristic for that period, he regrets that many other important and interesting contributions had to be left out.

2 FIRST EXPERIENCES (1953 – 1960)

My occupation with room acoustics started in 1953 when I set out to do the experimental work for my diploma thesis. At that time I was a student of the late Professor Erwin Meyer, who was interested in many acoustical fields, particularly in room acoustics. My task was to carry out room acoustical measurements in quite a number of concert halls and broadcasting studios under the guidance of a somewhat elder colleague. The goal of this project was to investigate the significance of some acoustical criteria which were believed to be the objective correlates of certain aspects of listening conditions in a room.

At that time there was a wide gap between the theory of sound waves in rooms on the one hand, and the tools available to an acoustical consultant on the other. Since my professor let me participate in his consulting activities I became gradually aware of this gap.

The theoretical description of sound fields in rooms in terms of normal modes had already reached a high state[1,2] in the fifties, the effects of room shape and wall treatment on the distribution of eigenfrequencies and the shape of normal modes was well-known although difficult to compute. However, this kind of treatment concerns just the physical aspect of room acoustics; neither is it related to what a listener really hears in a room nor does it offer any help to the consulting engineer who is faced with practical problems.

The only undisputed fact was the significance of reverberation and reverberation time, introduced and investigated by W. C. Sabine. Acousticians knew quite well which values of the reverberation time are adequate for a particular type of performance. Moreover, reliable techniques for recording decay curves and evaluating the reverberation time from them were available. And finally, the reverberation time T (in seconds) could be calculated by a simple formula:

$$T = 0.16\frac{V}{A} \quad \text{with} \quad A = \alpha_1 S_1 + \alpha_2 S_2 + \ldots \qquad (1)$$

(V = room volume in m^3, S_i= area in m^2 of i^{th} wall portion with absorption coefficient α_i). It is remarkable that Sabine had derived this formula not on the basis of normal modes, but empirically at first, and later on from simple energetic considerations. However, the condition for the validity of this formula (and similar somewhat more accurate ones) is that the sound field in the room be diffuse. This means that at each point inside the boundary the sound propagation is uniformly distributed over all directions. In this case all wall elements are hit by the same amount of energy per second, and the stationary energy density (or sound pressure level) is constant throughout the whole room (see eq. (6)). The concept of a diffuse sound field is an idealisation, since it requires zero intensity at any point which is impossible in a real room. Fortunately, eq. (1) yields reliable results even if the condition of a diffuse field is not exactly met.

Although the significance of reverberation time (RT) was (and still is) out of question, it was obvious that it does not yield a complete description of listening conditions in rooms. According to eq. (1), the RT should be independent of the shape of the room and of a listener's position in it. This is in contrast to our experience that the subjective acoustical impressions may vary considerably in different parts of a large hall. Therefore great efforts were undertaken to define additional objective parameters suited to fill this gap.

One attempt to gain additional information on the acoustics of a room was to record many decay curves excited by sine tones with a slowly varying frequency and to display these curves in a close sequence[3]. Since each curve shows beats between neighbouring eigenfrequencies which change from one curve to the next mountain-like patterns were obtained which were thought to yield information on "colourations", as are sometimes observed in small broadcasting studios. It turned out, however, that the examination of such patterns did not lead to unanimous and useful results.

Much more successful was a parameter based on the observation that sound reflections from walls etc., following the direct sound with a delay of less than 50 ms are not perceived as separate events but enhance the perceived loudness of the direct sound and often also the apparent width of the sound source (see Figure 1).Therefore they are considered "useful reflections" in contrast to those reflections which reach the listener with longer delay and hence add to what is perceived as reverberation. Accordingly, a parameter called "definition" or "Deutlichkeit"

$$D = \int_0^{50ms} [h(t)]^2 dt \bigg/ \int_0^{\infty} [h(t)]^2 dt \cdot 100\% \qquad (2)$$

was defined[4] in order to characterise the relative contribution of useful reflections, h(t) is the impulse response). In the course of the investigations, D turned out to be highly correlated with speech intelligibility, furthermore, its value showed characteristic differences at various places of a hall since it is not based on the overall sound decay but on the fine structure of h(t).

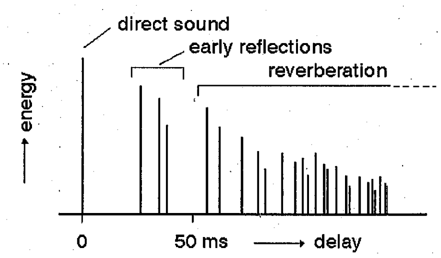

Figure 1 Impulse response (schematically)

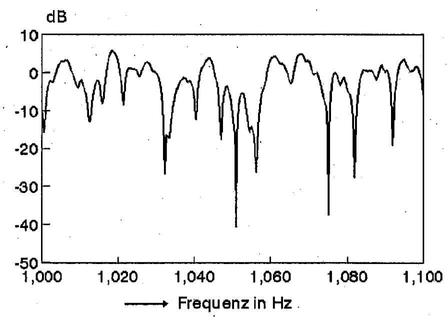

Figure 2. Section of a frequency curve

All the parameters discussed so far were related to the transient behaviour of a room, i. e. to the time domain. As an alternative, several authors[5,6] had proposed to consider the irregularities of a room's frequency transmission function instead. In fact, there were good reasons for looking at the frequency characteristics since every sound engineer tries to avoid linear distortions. Although "frequency curves" (see Fig. 2), measured at different locations in a hall or in different enclosures are considerably different in detail, their statistical parameters do not show characteristic variations: The range of fluctuations is about 10 dB on the average, and the mean frequency distance $\overline{(\Delta f)_m}$ of adjacent maxima is inversely proportional to the RT. This behaviour is due to overlap of the resonance curves associated with the eigenfrequencies. According to M. R. Schroeder[7] this overlap occurs for frequencies exceeding a certain limit (Schroeder frequency):

$$f_s = 2000\sqrt{\frac{T}{V}} \quad \text{(in Hz)} \tag{3}$$

Hence, evaluating single normal modes and eigenfrequencies may be useful for small rooms and/or low frequencies, in all other cases it is not worthwhile to know them "personally".

In the early fifties of the past century it was observed that the acoustics of some newly constructed concert halls fell behind the high expectations although there were no obvious acoustical faults and although their reverberation times were all right. Several acoustical experts believed that the excellent acoustics of some old halls were due to their interior style, in particular to the rich decorations which are typical of such halls but were out of fashion in the fifties. Such wall irregularities scatter the impinging sound rather than reflecting it specularly and create what was called a diffuse sound field above, and the idea was that a uniform directional distribution is responsible for what is known as "spatial impression" nowadays. To check this idea we measured the directional sound distribution in many halls and hence the degree of sound field diffusion by scanning all directions with a highly directional receiver namely a concave mirror with 1.20 metres diameter. The results can be summarized as follows: Wall scattering leads to very uniform distributions of the sound energy, spatially as well as directionally. However, with regard to the spatial impression the results were rather disappointing in that halls with highly scattering walls and ceiling do not convey a exceptionally strong spatial feeling; on the contrary, everything sounds quite clear and direct in them.

So we can conclude that three out of four attempts to find objective correlates to subjective acoustical impressions had failed. The reason for this failure was that these parameters – except "definition"- were not based on systematic listening experiments but merely on assumptions which were plausible, but not more.

3 MORE RECENT DEVELOPMENTS

3.1 Concert Hall Acoustics, Spaciousness

As mentioned above a new subjective category had emerged in the fifties, namely the spaciousness of a sound field, or the spatial impression, which was believed to be an important ingredient to good concert hall acoustics. Although this category had turned out not to be related just to sound field diffusion, it was obvious that certain features of the directional distribution of sound are responsible for it.

At this point a publication of J. West[8] should be mentioned. By analysing the data contained in L. L. Beranek's famous collection of concert halls and opera houses[9] he observed that in highly renowned concert halls the ratio of width to height is relatively small. This was probably the first indication on the importance of lateral sound reflections. In the following A. H. Marshall[10] demonstrated clearly that spatial impressions are due to sound reflections from side walls which are particularly strong in narrow halls while in wide halls they are masked by other reflections.

Parallel to these more empirical studies various groups of researches tried to find out how variations of the sound field structure influenced the degree of spaciousness. They used synthetic sound fields produced by a number of loudspeakers which were arranged in an anechoic room. Each of them was fed by an electrical signal the level and delay of which could be chosen independently in order to simulate a particular reflection. Additionally, the signals could be reverberated, for instance, by use of a reverberation chamber[11,12]. The general result was that just a few reflections are sufficient to create subjective spaciousness provided these reflections arrive from lateral directions and their delays with respect to the direct sound are not too long. Probably the most comprehensive investigation of this kind has been carried out by Barron[13], again by using a synthetic sound field. He found that all lateral reflections with delays between 5 ms and 80 ms contribute to the spatial impression, and that the amount of their contribution is proportional to their energy and to cos θ where θ denotes the angle between an imagined axis through the listener's ears and the direction of sound incidence. This result is independent of the reverberation time. As a suitable measure for the spatial impression Barron proposed

$$\mathrm{LEF} = \int_{5\mathrm{ms}}^{80\mathrm{ms}} [h(\theta,t)]^2 \cos\theta \, dt \Bigg/ \int_{5\mathrm{ms}}^{80\mathrm{ms}} [h(\theta,t)]^2 \, dt \tag{4}$$

This quantity is the "lateral energy fraction", its definition shows immediately that reflections arriving from the median plane of the listener's head do not contribute to the spatial impression. Alternatively, the laterality of reflected sounds can be characterised by the so-called "interaural cross correlation" (IACC)[14]: Consider the impulse responses $h_r(t)$ and $h_l(t)$ received at a listener's right and left ear and their cross correlation function

$$\varphi_{rl}(\tau) = \int_0^{0.1\mathrm{s}} g_r(t) g_l(t+\tau) dt \Bigg/ \left\{ \int_0^{0.1\mathrm{s}} [g_r(t)]^2 dt \int_0^{0.1\mathrm{s}} [g_l(t)]^2 dt \right\}^{1/2} \tag{5}$$

From this the IACC is obtained as the maximum of $|\varphi_{rl}|$ in the range $|\tau| \leq 1$ ms. Since it is a measure of the similarity of both ear signals, it is, in contrast to the LEF, negatively correlated with the spatial impression.

Nowadays it seems that the spatial impression has two components, both of them being caused by lateral reflections. If their delay times are less than 80 ms they increase the "apparent source width" while those with longer delays contribute to what is called "listener envelopment" [15,16].

The investigations reported so far are unsatisfactory in a way because they do not yield information on the relative weight of the various parameters, on individual differences in the subjective rating of auditive sensations, or if the assessment "good or poor acoustics" has further components not considered so far. Several groups of researchers have tried to tackle this problem by applying modern psychometric methods. The raw material of such studies constitute of subjective assessments collected from test persons which were asked to compare dummy head sound recordings from different concert halls, either by using bipolar rating scales ("dull – brilliant", "cold – warm" etc) or simply by asking them which of two presentations they prefer. Such data were subjected to multidimensional factor analysis, which typically yielded three to six independent perceptual "factors" or dimensions[17,18,19]. The next (and more difficult) step is to correlate these factors with objective sound field parameters. However, the results from different researchers were not quite unanimous. There was agreement that the reverberation time is still a major attribute of good concert hall acoustics. Furthermore, the clarity of a presentation seems to be of high significance, and the IACC, which turned out to be nearly uncorrelated with reverberation time shows high (negative) correlation with preference. However, researchers do not agree on the relative significance of these parameters, and also individual listeners seem to give different weight to them which indicates differences in personal taste. Moreover, the variations within a single hall may be larger that the variation from hall to hall which means that no definite ranking of concert halls with respect to their acoustics is possible.

3.2 Prediction of Sound Fields, Auralisation

For the acoustical design of rooms of any kind reliable methods of sound field prediction are needed. (The modal theory is not of much use for this purpose; only nowadays normal modes can be computed by using FEM or BEM.)

The traditional formula for the steady state energy density u_{st} is based on Sabine's theory and reads,

$$u_{st} = \frac{P}{4\pi c}\left(\frac{1}{r^2}+\frac{1}{r_h^2}\right) \rightarrow \frac{4P}{cA} \text{ for } r >> r_h \text{ with } r_h^2 = \frac{A}{16\pi} \tag{6}$$

(P = power of the sound source, assumed as non-directive); it predicts constant energy density and hence a constant level for all distances r exceeding the characteristic distance r_h which is typically a few meters in a large hall. Obviously, this formula often fails to yield true results, everybody knows that the sound level in a large hall, in a theatre or in a factory is all but uniform. The reason for this failure is lack of sound field diffusion.

A traditional way to examine the acoustical effects of a particular room shape was the construction of sound rays and their reflections. This method yields information on concentrations of the sound energy and also indicates the risk of echoes if delay times are considered. It fails, however, when the reflecting surfaces are not smooth, and it becomes too involved if reflections of higher than first order are to be included.

Another well-established method was the experimental examination of scale models. At first, optical models have been employed for this purpose with a little light bulb simulating the sound source; the walls were made of metal with a glossy surface, and the energy distribution was measured with photosensitive paper. Of course, optical models cannot yield information on transient effects such as echoes or reverberation. This limitation is not present when acoustical models are used. They have the additional advantage of including scattering and diffraction phenomena. In an acoustical model the wavelength of the test sounds must also be subjected to the scaling rules, accordingly their frequency must be elevated by the scaling factor. The absorption of the walls and the audience as well as the attenuation in the medium including its frequency dependence must be carefully modelled, otherwise one cannot expect the impulse response in the model to have more than a superficial similarity with that obtained in the real room.

The advent of digital computers has ended the era of acoustical measurements in scale models. The first authors who applied digital simulation to concert hall acoustics were Krokstad et al.[20] who determined the spatial and temporal distribution of sound by a method which is known as "ray tracing" today. The process starts with numerous hypothetical "sound particles" being released from a sound source with a prescribed directional distribution. Each of them travels with sound velocity and carries a certain amount of energy. Whenever it hits a smooth wall a new direction is determined either from the law of specular reflection (smooth wall) or from two random numbers modelling the scattering characteristics of the wall. The absorption can be accounted for by reducing the particle's energy by a factor of $(1-\alpha)$. The particle is followed up until its energy has fallen below a certain limit. Once in a while it will cross a previously assigned "counting volume" where its arrival time, its energy, and, if needed, its direction is registered. The result of a complete run can be represented as a histogram (see Figure 3) which is an approximation of the energetic impulse response of the room at a given place and from which many important parameters (decay time, definition, clarity, lateral energy fraction etc.) can be evaluated.

Quite a different method of sound field prediction is based on the old concept of image sources. It may be explained by considering an infinite flat room formed by two parallel planes (Figure 4). A ray reflected from one plane seems to originate from a virtual source, which is the mirror image S' of the original source S with respect to the reflecting plane. Similarly, a ray which has undergone two reflections seems to originate from a second order image, which is the mirror image of a first order image source. By repeating this process over and over infinitely many image sources of increasing order are created, they are equidistant on a straight line if the original source S is in the middle between both planes. After having constructed all relevant image sources the resulting energy density in a point P is obtained by adding the contributions of all image sources, the absorption of the boundaries is taken into account by multiplying the energy of each contribution with the appropriate factor (1-α) whenever the ray crosses the image of one of the original planes.

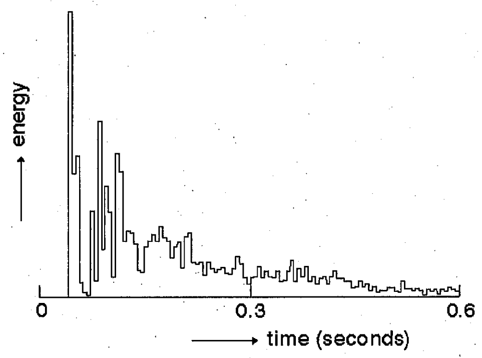

Figure 3 Result of a ray tracing run

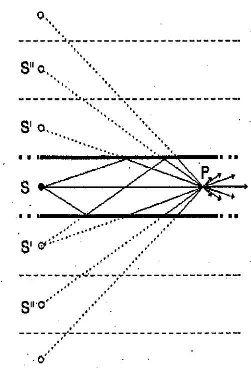

Figure 4 Flatroom and its image sources

Of course, this method can as well be applied to more realistic rooms shapes provided their boundaries are made up of plane and smooth wall sections, which results in a spatial pattern of image sources. However, most of these are "invisible" from a given receiving point, i. e. there is no real path connecting this point with the sound source via realistic reflections. Therefore special procedures to check the validity or visibility of image sources have been developed[21]. Since then, the image source model has undergone many modifications including combinations with ray tracing. Nowadays it is possible to determine quite realistic impulse responses of rooms which exist only on the drawing board or in the designer's imagination.

Having determined the impulse response of a large room either by computation or by measurement one may wish to use it for creating an audible impression from that room. Procedures of this kind are known as "auralisation" nowadays. Its can be achieved by a filter which convolves a "dry" music sample with the impulse response of the room (Figure 5). In order to produce binaural impressions it must have two output terminals namely one for each ear of the listener. The presentation can be achieved by headphones or, preferably, by two loudspeakers in an anechoic room. In the latter case "cross talk cancellation" (CTC) is applied which makes sure that the right (left) ear receives only sound produced by the right (left) loudspeaker.

The first experiments with auralisation have been carried out by Spandöck and his coworkers[22]. Their auralisation filter was a carefully designed scale model of the room. Unfortunately, the quality of reproduction was far from what we are used to today. This is not amazing keeping in mind the twofold frequency transformation involved and the imperfections of the electroacoustic transducers which had to operate in a very wide frequency range. Particular difficulties were caused by the complicated frequency dependence of sound attenuation in air. The mentioned authors tried to solve this problem by reducing the relative humidity of the air to about 3%.

The first successful auralisations with purely digital room models were carried out by Allen and Berkley[23]. Today auditive impressions achieved with auralisation are very realistic. Hence, these new and fascinating techniques can be used for the immediate acoustical comparison of different halls, or of different places in one and the same hall. Furthermore the acoustical consultant can employ them to convince himself (or the architect) on the necessity or effectiveness of certain acoustical measures such as modifications in shape or material.

A purely analytical way to determine the steady state or transient distribution of sound energy in rooms is the radiosity method, expressed by an integral equation. Let dS and dS' denote two boundary elements dS and dS', R the length of a straight line connecting them, ϑ and ϑ' the angles between this line and the wall normals of both elements, then the energy arriving at dS (per second and m^2) is obtained by collecting all contributions from other boundary elements

$$B(\vec{r},t) = \iint_S \rho(\vec{r}')K(\vec{r},\vec{r}')B\left(\vec{r}', t - \frac{R}{c}\right)dS' + B_0(\vec{R},t) \quad \text{with} \quad K(\vec{r},\vec{r}') = \frac{\cos\vartheta\cos\vartheta'}{\pi R^2} \qquad (7)$$

B_o is the energy directly contributed by a sound source, and $\rho = 1 - \alpha$. The function K contains the $1/R^2$ law and the fact, that the boundary is assumed to scatter all incident sound according to Lambert's cosine law. Closed solutions of this integral equation are available for a few simple room shapes only (see next section). Generally, however, it must be numerically solved[24].

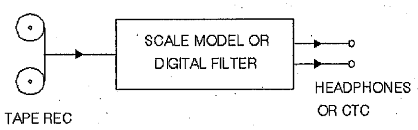

Figure 5 Principle of auralisation

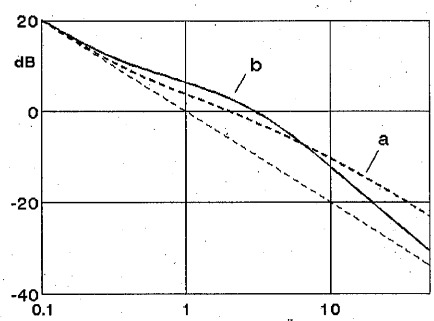

Figure 6 Sound level in a flatroom (h=height)
a) smooth walls, b) rough walls

3.3 Sound Propagation in Work Spaces

This last section is devoted to a rather prosaic branch of room acoustics: sound in work spaces. The interest in it stems from the fact, that most people spend more of their lifetime in offices, workshops or factories than in concert halls and theatres. The goal of acoustic design in work spaces is noise reduction, and hence it is the steady-state level which is in the foreground of interest. A first approach to calculate it could be eq. (6) which predicts constant energy density except for locations close to the sound source. In reality, the sound level usually decays with increasing distance from the sound source. As mentioned, this discrepancy is due to the lack of sound field diffusion, which is particularly poor if one dimension of the enclosure is much larger or smaller than the remaining ones, and/or the sound absorbing material (if there is any) is concentrated on the ceiling of the room. This is easily seen from Figure 4 which depicts an idealized flat room: All contributions to the energy density at point P are traveling to the right side resulting in a non-vanishing intensity vector (which should be zero in a diffuse field). – The dependence of the sound pressure level on the distance from the sound source, obtained by adding the contributions of all image sources, is shown in curve (a) of Figure 6. However, real offices, workshops or factories are not empty but contain many objects such as furniture, screens, machines, stockpiles etc. which scatter the sound waves and redistribute the energy into new directions. One way to account for this effect is by assuming the bounding planes not as smooth but as having irregular surfaces and thus reflecting the impinging sound diffusely. Then the radiosity method (see eq. (7)) yields a closed solution for the idealized flat room (see curve (b) in Figure 6). It shows this redistribution which results in higher levels near the source and lower levels far from it. In his comprehensive investigation M.Hodgson[25] reported on numerous measurements which he performed in factories, both full size and in model scale, which demonstrated this effect very clearly.

Another approach to sound propagation in fitted work spaces is by assuming the scattering objects distributed uniformly in the room. Then the noise propagation can be treated as a kind of diffusion process (S. Jovicic 1971). Now an additional parameter is needed to characterise the effectiveness of scattering: the mean free path length of a sound "particle" with respect to scattering, $\lambda_s = 1/nQ$, with n denoting the number of scatterers per m^3 and Q their scattering cross section. The results show, at least in many cases, close agreement with sound pressure levels measured in factories[25]. Alternatively, ray tracing procedures can be adapted to noise propagation in fitted work rooms as has been done successfully by (Ondet and Barbry[26]).

4 REFERENCES

1 P. M. Morse, P. M., Vibration and Sound. Mc Graw-Hill, New York 1948.
2 P. M. Morse and R. H. Bolt, Rev. Mod Physics **16**, (1944), 70.
3 T. Somerville, and G. L. S. Gilford, BBC Quart. **7** (1952), 80.
4 R. Thiele, Acustica **3** (1953), 291.
5 E. C. Wente, J. acoust. Soc. Amer. **7** , (1935), 123.
6 R. H. Bolt and R. W. Roop, J. acoust. Soc. Amer. **22** , (1950), 280.
7 M. Schröder, Acustica **4** (1954), 594. K. H. Kuttruff and M. R. Schroeder, J. acoust. Soc. Amer. **34** , (1962), 76.
8 J. E. West, J. acoust. Soc. Amer. **40** , (1966), 1245.
9 L. L. Beranek, Music, Acoustics and Architecture. John Wiley New York 1962.
10 A. H. Marshall, J. Sound Vib. **5** (1968), 116.
11 W. Reichardt, and W. Schmidt, Acustica **17** (1966), 65; Acustica **18** (1967), 274.
12 E. Meyer, W. Burgtorf and P. Damaske, Acustica **15** (1965), 339, Acustica **19** (1967/68), 199.
13 M. Barron, J. Sound Vib. **15** (1971), 475.
14 P. Damaske and Y. Ando, Acustica **27** (1972), 232
15 W. de Villiers Keets,.: Proc. 6th ICA, 1968, vol. III, section E, 49
16 J. S. Bradley and G. A. Souloudre, J. acoust. Soc. Amer. **98** , (1995), 2590.
17 R. J. Hawkes and H. Douglas, Acustica **24** (1971), 235.
18 H. Wilkens and G. Plenge in: R. Mackenzie (ed.) Auditorium Acoustics, Applied Science, London 1974
19 M. R. Schroeder, D. Gottlob and K. F. Siebrasse, J. acoust. Soc. Amer. **56** , (1974), 1195.
20 A. Krokstad, S. Strøm and S. Sørsdal, J. Sound Vib. **8** (1968), 118.
21 M. Vorländer, J. acoust. Soc. Amer. **86**, (1989), 172.
22 F. Spandöck, Ann d. Physik **20** (1934), 345; P. Brebeck, R. Bücklein, E. Krauth and F. Spandöck, Acustica **18** (1967), 213 .
23 J. B. Allen and D. A. Berkley, J. acoust. Soc. Amer. **65**, (1979), 943.
24 R. N. Miles, J. Sound Vib. **92** (1984), 203.
25 M. R. Hodgson, Ph. D. thesis, Univ. of Southampton, England 1983.
26 A. M. Ondet and J. L. Barbry, J. acoust. Soc. Amer. **85**, (1989), 787.