# ACOUSTIC MODELLING – APPROXIMATIONS TO THE REAL WORLD.

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## 1. INTRODUCTION.

The study of sound propagation is a complex subject. That is especially true for enclosed spaces. In nearly all branches of acoustics, models[1] are constructed, either conceptually, physically or, more often nowadays, numerically as algebraic expressions, in order to make the problems of design and prediction tractable. All such models involve abstractions and simplifications, which differ depending on the basis of the model.

This paper is intended to be an introduction to the different types of model and their individual departures from the realities of the physical system that they are attempting to describe. It begins with a brief review of the fundamental properties of sound propagation, especially in the context of reproduced sound. It includes discussions of conceptualisation, physical scale modelling, ray-tracing, image-based methods and finite-element representations.

The discussion is limited to propagation in air, over moderate distances and in spaces that have a number of more or less solid boundaries. The assumed conditions are characteristic of those which apply to human beings listening to sound, for entertainment or enjoyment, in a room, theatre or concert venue. It is not a deeply theoretical study, nor is it concerned with propagation over long distances, where the statistical properties of the transmission path play an important rôle. It is a personal view, intended to stimulate discussion.

## 2. SOUND PROPAGATION IN ENCLOSED SPACES.

Sound propagation in an elastic medium begins when a disturbance takes place at some point in the medium. The energy transferred to the medium by the disturbance travels from one point in the medium to adjacent points according to the properties of compliance and inertia of the medium. The propagation in three dimensions can be described by the well-known wave equation ...

$$\nabla^2 p = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} \qquad (1)$$

Equation 1 describes in mathematical terms a sound wave travelling in straight lines at velocity *c*, which is typically around 340 $ms^{-1}$. It is, theoretically, a full and complete description of the sound propagation.

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[1] *model* - OED, Second Edition 1989, definition 'e' – "A simplified or idealized description or conception of a particular system, situation, or process (often in mathematical terms: so *mathematical model*) that is put forward as a basis for calculations, predictions, or further investigation."

However, even by this early stage, we have a description that can only be solved analytically in the simplest cases and most certainly not in any realistic, practical situation. Furthermore, it is limited to sound waves of infinitesimal amplitude in a homogeneous, isotropic, frictionless medium at rest [1].

From this point onwards in any practical situation, simplifications and approximations begin to accumulate. Even before any boundary is reached by the sound wave, the calculation is compromised by local distortions of the material properties, for example by thermal inhomogeneities, energy loss causing progressive attenuation and winds and drafts from various causes. It is also compromised by the sound having to have finite amplitude to be of any practical value.

Readers may protest that ignoring these details is perfectly justifiable; it involves reasonable simplifications that have little impact on the perception of the sound. To some extent that is true, but mainly because they are a natural part of the sound propagation and we usually accept their implications without conscious thought. However, these effects can have a large impact on the objective behaviour of the sound. That is what most acoustic models model and, by ignoring them, can make substantial objective errors. Whether or not the errors are subjectively significant is a separate issue – the topic is the underlying accuracy of the different modelling processes.

When any part of the propagating sound reaches a boundary, the situation at once becomes more complicated still. In most spaces used for listening, that happens within about two or three milliseconds. At a boundary, some of the incident sound energy is absorbed, either transmitted through the new medium to the other side or dissipated in the internal friction of the material. The remainder is reflected back more or less from whence it came in the original medium. The balance between absorption and reflection depends strongly on the relative physical properties of the two media and also on the shape of the junction between them, in relation to the wavelength of the sound wave. These properties are also strong functions of frequency, so that the effects are different for different parts of the frequency spectrum.

The absorbed and transmitted energies are simply lost to the system. The effect may be a complicated function of frequency, but at least that energy no longer needs to be considered as part of the calculation. The reflected energy on the other hand continues to exist within the system and must be accounted for in any calculations. It almost always has an extremely complex distribution pattern, in both spatial and frequency domains.

After a short time, all parts of the original wavefront will have reached some boundary and undergone a degree of absorption and reflection. The human hearing system, finely tuned to the acoustics of the local environment by natural selection (the innate desire to avoid becoming a meal for some predator) is very sensitive to the detail and nuances of these acoustic reflection patterns. Early discrete reflections, those reaching the ears less than about 50 ms after the direct sound, give clues of direction and local spatial details. Later reflections, those after about 80 ms, are heard as discrete echoes and inform the listener about the physical aspects of the space on a larger scale. In enclosed spaces and at times after about 80 ms, the complicated reflection pattern is heard as an ensemble, called "reverberation", provided that it has no large discrete components.

Such patterns of sound distribution, as functions of time, frequency and direction, are extremely complicated. No method of conceptualisation, calculation or modelling is likely to come even close to predicting or representing the details accurately. First, the radiation characteristics of the source will not be known accurately. Next, the vagaries of the air and its variations in temperature and velocity (globally and locally) must be accommodated. Not even the basic sound velocity can be assumed to be everywhere the same and constant, especially for outdoor venues. Once the sound wave has reached an obstruction, the detailed physical shape and acoustic properties that govern the absorption and reflection will be equally unknown, in detail. Fig. 1 shows the change in response over two instances separated by about one hour in a still room. Over the frequency range from 2 kHz to 4 kHz, variations of more than ±15 dB can be seen. Even under those exceptionally comparable occasions, significant objective variations occur. How then can we compare the responses of a computed model and the real room and expect to find reasonable differences?

Even if we were given the radiation characteristics of the source and the variations in temperature and velocity, the description of the propagating sound would require immense storage and calculation. As a rough estimate, if this were to be done using some form of digital representation, using a spatial quantisation interval of 7 mm – the distance travelled by a sound wave in one digital sampling interval, we can calculate the size of the problem. After just 1 ms, the sound propagation sphere from a point source would be 680 mm in diameter. It would contain about 480,000 elemental points. For each of those points, the wave equation would have to be solved for the particular local conditions. That would be difficult, but probably not impossible today. However, after a further 99 ms, the number of elemental points would be $4.8 \times 10^{11}$. Even that time is only about $^1/_5$th of the reverberation time in a small room. Thus, by any reasonable yardstick, accurate objective prediction must be either impossible or impractical. The acoustician is forced to rely on simplifications and approximate models in order to resolve the design problem.

# 3. CONCEPTUAL AND ELEMENTARY MODELS.

The severe difficulties involved in the exact description of sound propagation mean that, in practice, most of the complications are simply ignored. Even in just thinking about propagation, acousticians make some assumptions – mostly without even consciously realising that they are doing so. In each person's own special field, a set of learned assumptions specific to that field are automatically invoked. Unless special conditions arise that cause the assumptions to be questioned, they rarely are. This results in conceptual models that are adequate for the purpose – until something out of the ordinary arises.

Typically, an acoustician working in reproduced sound will assume that the sound energy spreads outwards from a small source and travels in straight lines until intercepted by boundary or internal surfaces. Local departures from homogeneous conditions are rarely, if ever, considered. Propagation loss is generally accommodated into formulae for calculating "reverberation time", along with the average absorption coefficient of all of the boundary and internal surfaces. Those values, in turn, come from other models of sound propagation that were used to make "measurements" of those properties.

Based on properties of the human hearing system, the time domain is divided into three times zones – 'early' when the few discrete reflections affect localisation, 'late', characterised by reverberation or discrete echoes, and the middle interval (which doesn't seem to have a catchy, popular name), which primarily affects timbre.

The immense detail of diffraction and reflection are subsumed into an overall image of a more or less confused combination of specular reflection and "diffusion". It is clear that, on some scales, acoustic reflection behaves rather like optical reflections in a mirror, on other scales more like omnidirectional reflection from a uniform diffuser. The boundary between these depends on the scale of the space and the wavelength of the sound. It is common to get an apparently clear and well-defined reflection over a wide band of frequencies from a very irregular surface, if that surface is large and at a great distance (an echo from the façade of a distant large building). On a smaller scale, specular reflections are only possible at high frequencies from acoustically smooth surfaces.

Propagation nonlinearities are not usually considered, except in the field of public address or large music venues where the sound level near to the sources can be high enough to cause significant distortion.

# 4. PHYSICAL MODELS.

The earliest physical models used 'ripple tanks' [2, 3, 4]. In these, a two-dimensional 'slice' through a space was placed so that the reflections from the surfaces could be made visible, either by half filling the model with a liquid or using Schlieren photographic methods in air. In principle, such models could produce little more than a draughtsman could achieve on a drawing board, but did allow dramatic and convincing visualisations to be created. They were beneficial in demonstrating, and thus avoiding, the worst sorts of acoustical shadowing and focussing. Once the initial model section had been made, it was reasonably easy to assess the effects of changes.

The most obvious way of making a full acoustic model is to construct a real, three-dimensional scaled facsimile of the space. This has been done many times in the past [5 - 18]. Apart from air absorption, all of the acoustic aspects of such a model behave at the scale size just like their prototypes do at full size. By scaling the time, and thus the frequency range, by the scale factor, all of the acoustic effects remain in proportion. Air absorption is more difficult to model. It happens, by mere coincidence, that the absorption characteristic of dry air (< 1% RH) is a good model of the behaviour of real air under 'normal' conditions at a scale factor of around 8:1. This is not true for other scale factors [19, 20, and 21].

An 8:1 scale model, especially of a concert hall, is large and expensive to build. It may, however, have other uses that can partly justify the cost, such as convincing clients, testing air conditioning system layouts, verifying public address system design, etc. Building such a model is certainly a serious undertaking, requiring large amounts of skilled labour and taking a great deal of both consultant's and calendar time.

In the past there were problems with instrumentation. Objective measurements of reverberation times and other forms of time and frequency history could be made almost as easily as at full scale. However, the main point of a model of any kind is for people to experience it directly. That required a source of suitable audio material without any of its own acoustic environment [22]. The material could be speeded up by the scale factor and played through small loudspeakers into the model. The sound could then be recorded through small microphones and slowed down again for listening. The problems involved in doing that, with any reasonable final quality, were severe. The model loudspeakers, in particular had to be special. They needed a frequency response extending up to at least 100kHz. They had to be small, but still deliver a power output similar to that of real loudspeakers to give a reasonable overall signal-noise ratio. Any colourations had to be of short duration and low level because their effects would be both reduced in frequency and increased in duration by the scale factor [23, 24].

Today, instrumentation is not such a problem. Impulse methods can be used to measure transfer functions from a nearly perfect sound source, though microphones still do constitute a potential limitation. Averaging can be used to make an arbitrarily low signal-noise ratio. The impulse responses can be used to create the 'sound' of the model in real time. Even the three-second reverberation time of a large hall, requiring something like a 150,000-tap convolution filter, can nowadays be processed in real time.

The fundamental problem with physical scale models (apart from the cost) arises in the selection of materials, particularly the internal finishes. It is just about impossible to find out everything there is to know (acoustically) about normal materials at full-scale. Of course, absorption coefficients can be measured to some degree of accuracy, but the result of a measurement of absorption coefficient is dependant on the physical arrangements used for the measurement and the behaviour of the sound field changes in a different environment. It is less easy, but still possible, to measure the other important aspects of the materials, especially their reflection and diffraction properties. Those can then be modelled in the same way as the absorption. Much of this can be done by physical scale modelling of the assemblies, though it is important to verify the physical properties of the basic component materials also scale correctly. To do this accurately is time-consuming and expensive. Not to do it removes most of the justification for building the model. There is no point in building an inaccurate physical model if it is going to depart from the real thing by as much as any other inaccurate method would do.

## 5. RAY TRACING.

The principle of the ray-tracing model is deceptively simple. It begins by visualising the propagation as narrow 'rays' of sound, emanating from the source. Each ray can then be 'traced' through its interactions with surfaces until it becomes insignificant. In principle, and maybe in some examples of practice also, the method can accommodate local variations in the medium and propagation non-linearities. In those respects, the method could be as close to ideal as necessary. For longer paths, such as those encountered in large outside venues, the local effects of wind shear could also be accommodated.

In practice, there are two computational limitations. The first is that the number of rays needs to be very large indeed if 'adequate' resolution is to be maintained after a number of reflections. The second is that the 'rays' are not rays but, if they can be described as anything at all, are like cones of propagation, progressively increasing in cross-section and, potentially, encountering different conditions at different parts of the ray's wavefront.

For the former, given a requirement to model up to a relatively modest 2 kHz limit, object sizes down to about 40mm (¼-wavelength) need to be represented. Not only does that imply a large database for the geometry but, after 100ms, there would be about $10^7$ objects of that size at the position of the sound wavefront, not including any objects already passed. Of course, many surfaces and objects could be represented by larger elements, but the basic resolution must be of that order if any attempt is to be made to model the effects of diffraction. (The equivalent figure for a 1 second time span and a 10 kHz limit – figures not infrequently found in ray tracing 'packages' – is $2 \times 10^{10}$.)

For the second, the problem can be moderated to an arbitrary extent by increasing the initial number of rays. Then, the likelihood of any part of the wavefront of each ray encountering anything different to the other parts becomes smaller, in the limit vanishingly small, and the ray-like assumption ever more reasonable. The problem with this approach is that huge computational resources would be expended on calculating effects at points in space which are extremely close to each other and between which there can be no significant acoustic differences. One obvious answer is to subdivide the rays progressively. At the beginning, in the space immediately surrounding the source, only a few rays need to be considered. Either arbitrarily progressively or specifically as each ray encounters an obstacle, it can be subdivided. Some of those subdivisions may continue in other directions, perhaps depending on partial intersections of the original ray with the edge of an object. All of this eases the computational load.

Ultimately, the real limitation to ray tracing is the effect of diffraction and the consequential collapse of the model into chaos. Ideally and excluding the rare examples of perfect, specular reflection, each new boundary interaction should be modelled by a complete new source, with the radiation pattern determined by the incident sound ray and the diffraction and reflection pattern of the boundary. This is especially true for edge contacts and partial intersection of the rays. That would cause the calculation size to increase rapidly, to the extent that it would almost immediately become intractable. One major effect of approximating to diffraction is that it introduces an element of chaos into the computation. To predict whether a second-order reflection would hit or entirely miss a 1-metre target across even a moderately sized room, the angular accuracy required is of the order of

±3°. What acoustician would be prepared to predict complex reflection patterns to that accuracy? That serious non-linearity, when repeated a few times, leads to chaotic behaviour.

The net result of the ray tracing calculation is usually presented as an impulse response. In reality, the closely timed arrival of huge numbers of individual echoes is not perceived as such, at least after about the first 30ms. The effects are perceived in the frequency domain. Therefore, to make a meaningful interpretation, these arrivals have to be summed as vectors for each frequency. The final result is then the sum of a large number of such vectors. The prediction of their amplitudes is dependent on the accurate knowledge of the reflection properties of all of the surface interactions along the way, including the effects of diffraction from the limited sizes of the reflecting objects. It may, however, be reasonably accurate, given good data and minor contributions from diffraction. It is practically certain that their phases will be close to random. It is entirely unreasonable to suppose that phase coherence could be maintained in a predictable way through a sequence of reflections, each with its own complex dependence on frequency, and airborne propagation paths, with the vagaries of local temperature and velocity variations. The complex sum of all of these vector contributions is likely to depend more on generalised statistical properties than on any element of predictability.

Finally, ray tracing, like all other modelling methods, suffers from the lack of exact information about the materials and constructions. The main effect for ray tracing is the error introduced by uncertainties in diffraction and reflection and the consequential chaotic behaviour of the later 'reflections'.

## 6. IMAGE METHODS.

Image methods have much in common with ray tracing. In principle, instead of generating a limited selection of initial rays, the method seeks to calculate all of the visible reflections, as though the sound behaved in the purely geometric, optical 'sense'. Then, the contributions from all of the images of the source via all of the paths to the receiver can be summed. Most of the limitations and difficulties of the ray tracing method also apply to the method of images. It is obviously difficult to add the concepts of diffraction and partial obstruction to such a model, more so than for the ray tracing method.

However, by far the most serious problem for image methods is the great redundancy in the calculation of valid paths. In simple cases, such as a uniform rectangular room, the situation is trivial. As the number of surfaces increases and their angles depart from the perpendicular, the total number of potential images increase rapidly. Of those, only a tiny proportion constitutes valid, 'visible' images. Yet the total has to be investigated in order to separate them. An estimate for a one-second response in a medium-sized room (400 $m^3$) with just 11 surfaces (six enclosing walls and a single six-sided object standing on the floor) produced $10^{37}$ potential images with just 53,000 of them visible. Just to calculate the visibility criteria would have taken $10^{25}$ years at 100MHz. Clearly such things are not reasonable. Even for a 100 $m^3$ room and a response time of 0.1s, the number of images is over $10^{12}$ and the number visible about 450 (less than 1 in every 1,000,000,000).

One author [26] has argued that it is, in any case, incorrect to exclude images that are 'invisible'. On the grounds that their total number is so much greater than the visible ones, it is argued that their effects are dominant anyway. In an ideal, purely geometrical world that would be nonsense. However, in the real world, the real-life effects of uncertainty about the geometry and acoustic properties of the model and, especially, diffraction ensure that theoretically 'invisible' images have some effect. Their much larger number ensures that their effects do indeed dominate the response, at least for reflection orders of the second or third or higher.

Thus, while the image and the similar ray tracing methods might be reasonable approaches to the prediction of early responses, they become poor models for predicting the later part of room responses.

## 7. FINITE ELEMENT MODELS (FEM, BEM, FDTD).

Finite element approaches are common in many areas of scientific endeavour, where analytic methods are too complex or where numerical representations are more appropriate. The problem domain is subdivided into small regions, where the physical properties can be reasonably accurately represented by a uniform distribution or simple interpolation functions. Relatively simple expressions can be used to describe the physical properties with reasonable accuracy. A complex problem can thus be reduced to an array of repeated simple calculations. Sometimes the array can be very large. In global atmospheric studies, as many as $5 \times 10^{14}$ elements may be involved, each with perhaps a dozen parameters. That represents the state of the art, requiring the most powerful computers available for their solution. Such computers are not usually available for the solution of acoustic problems.

Acoustic FEM attempts to carry out the analytic solution to the wave equation by subdividing the acoustic space into small elements, rather like the calculation described in Section 2. Each element is assumed to be small enough that the wave equation can be applied to the element. The number of elements required depends on their order. With higher order elements, fewer may be required, but the calculation of each is more complicated. Generally, there appears to be a net benefit in going to moderately high order elements. Second order elements, with quadratic interpolation functions are commonly used. With second order elements, six elements per wavelength are usually though to be necessary.

A small listening room, for frequencies up to, say 12kHz, requires $1.5 \times 10^{9}$ elements. That may be feasible today, but is computationally demanding. It is not easy to envisage dealing with significantly larger rooms on this basis.

The basic FEM method computes the steady-state sound pressure distribution. It is not applicable to transient responses. Furthermore, the calculation has to be repeated for each frequency. An alternative is to solve the equation mesh for the eigenmodes. The response at each frequency can then, in principle, be calculated from the sum of the eigenmode responses. That process is susceptible to small errors because of the large number of contributions included in the summation.

An alternative approach, using the physical properties of the medium expressed in terms of their time domain behaviour (Finite Difference Time Domain) can resolve time domain effects. That requires a similarly large number of elements.

One of the serious limitations of FEM is that the whole acoustic space has to be modelled, even if no results are required for the majority of the locations within it. The boundary element method (BEM) can model the far field responses of larger spaces, but the component calculations are more complex.

Combinations of structural FEM and acoustic BEM have been highly successful in modelling smaller objects like loudspeakers and especially dealing with the boundary coupling between the mechanical properties of the structures and the radiation into the air.

Given a large enough number of elements of adequately small size, these finite element approaches ought to be able to approach the theoretical to an arbitrary degree of accuracy. However, like other methods, these models are also limited by lack of detailed knowledge on the mechanical and acoustic properties of the materials. At the present time, the number of elements required is also a severe limitation especially for larger spaces.

## 8. SUMMARY AND CONCLUSIONS.

This paper has attempted to provide an overview of acoustic modelling methods. Beginning with the basic theoretical principles of sound propagation, the progressive departures from the mathematically ideal have been enumerated. Full and accurate mathematical descriptions of the behaviour of sound energy are impracticable in anything other than the simplest and idealised cases. It has been shown that, even at the simplest level of application, basic approximations are invoked to make the calculation of acoustic responses tractable. This occurs even in fundamental conceptualisations of acoustics by 'experts'.

As soon as the purpose includes objective modelling of acoustic responses in realistic environments, the level of approximation becomes significantly higher. In particular, the complications introduced by diffraction will have a significant impact and are difficult to calculate comprehensively.

All modelling methods also suffer from lack of objective data about the geometry and acoustic properties of the enclosure and its contents. In any real situation, adequate information about the acoustic properties of surfaces and structures cannot be obtained. Such data as are readily available, such as average absorption coefficients, are themselves obtained from models that may only be approximations anyway.

Although much has been written in criticism of the various modelling methods, that is not say that any or all are useless. Without the first conceptual stage of modelling it is doubtful whether much progress could ever have been made in the development of room acoustics. Similarly, the objective modelling methods lead to ideas and developments that might not have been possible without the insights provided by the modelling, even though they are themselves only relatively crude approximations to the underlying physical process. It is for the user of the modelling method to decide, in the local context, to what extent the approximations represent the ideal.

All of the modelling methods described here have been developed to a much greater degree than has been indicated in this brief review. Vast libraries of complication have been added by many individuals, far too many to do justice to in any reasonable bibliography. However, it remains true that the great complexities of real-life acoustic propagation, especially diffraction, remain poorly represented.

It is certain that all modelling methods lead to results that are, objectively, far from a description of reality. Great danger lies in the confusion of the model with reality and in assumptions that the one is an accurate description of the other.

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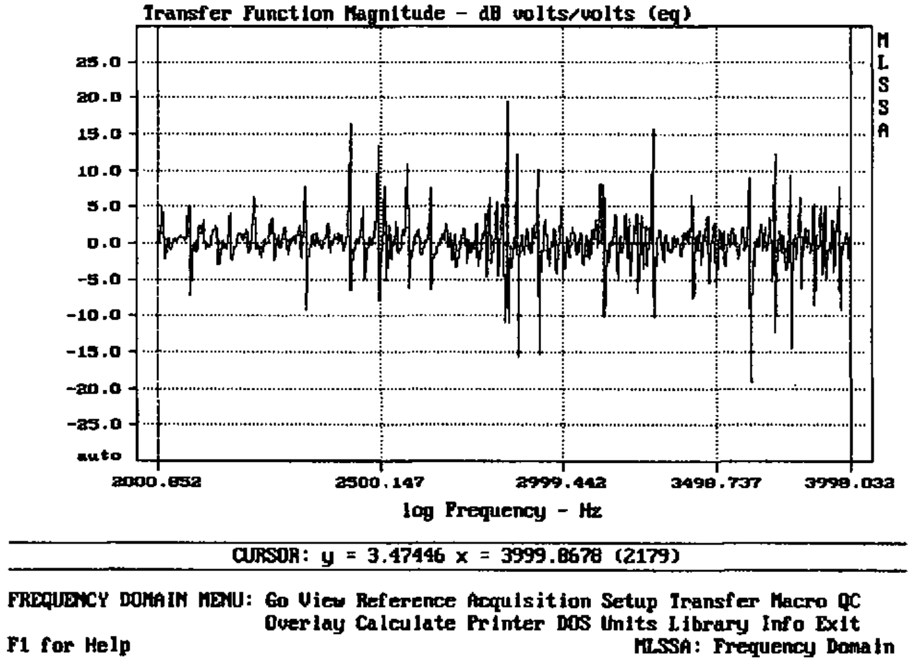


Fig . 1. Response change in one hour in still room.