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FIFTEEN YEARS EXPERIENCE WITH A COMPUTERIZED RAY TRACING PROGRAMME

KROKSTAD, A., & STRØM, S., & SØRSDAL, S.

THE LABORATORY OF ACOUSTICS/ELAB-SINTEF, THE TECHNICAL UNIVERSITY OF TRONDHEIM, N-7034 TRONDHEIM-NTH, NORWAY

1. Introduction

After one and a half years work Svein Strøm and Svein Sørsdal completed the first version of our ray tracing programme in 1967. The programme was supplemented and revised in 1973. Minor supplements and modifications are still made, but the main principles and great part of the software have remained the same during 15 years. The programme is implemented on Univac 1100-series. The estimated cost of the software is about 4 man years, documentation and users guide included.

The main features is presented in ref. 1 and in a somewhat popular form in a pamphlet [ref. 2] which may be obtained from the laboratory.

The program has been used by all major projects in room acoustics at our laboratory; 4 research projects, 32 consultancies and 4 thesis works.

2. Examples on use

2.1. Sound distribution in existing halls [ref. 3].

Calculations were done on simplified models of 10 halls from different categories in "Music, Acoustics and Architecture" by L. Beranek. The study is used as a reference for the evaluation of sound distributions of halls under design.

2.2. Sound distribution and geometry [ref. 4].

70 distributions were calculated for a systematic study on the influence of room shapes, wall shapes and source positions. Audience area and room volume were kept constant. The results are an importante basis for initial discussions by new projects.

2.3. "Hjertnes" Concert Hall, Sandefjor, Norway [ref. 5].

Our laboratory was the responsible acoustic consultant. The computer study was used for optimizing the room cross section of a rectangular shaped hall.

2.4. Grieg Memorial Hall, Bergen, Norway.

Our laboratory cooperated with the acoustic consultant by designing reflectors at walls and in the ceiling for improving the sound distribution in the partly fanshaped hall. The graphical output from the computer was of great help by discussions with the architect.

2.5. Grand Studio

Calculations were done for the acoustic consultants for studying the response in different microphone positions above the stage and the influence of source positions.

2.6. Sound reinforcement

For designing the central loudspeaker cluster a Sanson-Flamsteed projection of the hall as seen from the speaker is generated by the computer. Those directions from the speaker and those surfaces in the room giving early and late reflected sound to different regions of the audience area computed.

2.7. Stage acoustics.

The opening from an orchestra enclosure to the hall may be considered a secondary sound source and directional and time distribution of energy over this area may be calculated.

3. Summary of main features of the program

- the program is a three dimensional ray tracing program based on energy response in time and direction, and its space distribution, for impulsive exitation.
- a deterministic approach is used, repeated calculations on the same room results in exact the same response.
- any source directivity may be modelled, even multisources such as distributed loudspeaker systems.
- output may be presented in a variety of different ways, both graphically and numerically.
- the "history" of individual rays are stored.
- architectural drawings, different projections, of the room used in the calculations are made by the computer.
- the precision is only limited by economy and storage space.

4. Software features

The software consists of 16 programs using a common data base. The individual programs may be changed or supplemented without affecting the main software.

No programming knowledge is needed by the user. A users guide [ref. 6] presents a step by step procedure for the preparation of input data and run information.

The current limitations are 500 corners in a half room, 150 plane surfaces and 15000 generated rays in one run. These limitations are deliberately imposed and may be changed.

5. Room acoustic parameters

Our policy has been to implement all parameters which are unambiguously defined and may be calculated from the data for energy impacts on measuring surfaces. The list of current parameters include :

1) Total energy on linear and logaritmic scale, giving the stationary sound intensity.
2) "Schwerpunktzeit", 1.order momentum of the echogram, as defined by Kürer [ref. 7].
3) "Deutlichkeit" or "Definition" with two time limits, 50 ms after Thiele [ref. 8] and 80 ms after Lochner and Burger [ref. 9].
4) Early decay time (T_{10}) after Jordan [ref. 10].
5) Lateral parameter after Barron [ref. 11].

6) "Klarheitsmass" after Reichardt [ref. 12].

There are strong correlations between some of these parameters, but both the numerical values and their differences for different room and room positions give useful information. At least three parameters must be considered.

6. Acoustic simplification and limitations

The most important limitations are:

- the spherically emitted sound wave is quantized in direction
- the room is defined by only plane surfaces
- specular reflections and angle independant absorption is assumed
- energy addition is assumed
- no limitation in bandwidth or stretch of the pulse length due to ringing of boundaries.

Due to spherical spread the distance between rays is increasing with transmission distance and fewer rays will illuminate a surface after a long transmission distance or transmission time. By matching the density of rays to the details in the room and the dimensions of the measuring surfaces, any precision may be obtained. Measuring surfaces are penetrable and not affecting sound distribution.

Speculary reflected sound will be the dominating energy contribution by planes of linear dimensions greater than one wavelength and roughness less than a quarter wavelength. When the highly reflecting surfaces in a room consists of planes of dimensions greater than 1.5 meter and roughness less than 2 mm, specular reflections will result in reasonable approximations to real distributions over the frequency range 250 Hz - 4000 Hz, which is the range of greatest importance for speech and music.

Sound diffraction, edge radiation and more complicated laws of reflection may be implemented in the program. Kuttruff [ref. 13] is using cosine-law for directional distribution of reflections.

The principle of energy addition has a psychoacoustic foundation. Two observable effects are lost; coloration due to interference, importante for time delay less than 10 ms, and standing waves, importante at low frequencies.

7. Computer model compared with real model

Building and measurements on real models are the only practical alternative to a computer simulation when the hall under design deviate from previous accepted shapes. Compared with a computer model, a real model have these benefits:

- more correct reflection simulation, especially by complex shapes where diffraction, diffusion and edge reflections are of great importance
- listening test may be performed on model, if a rather great scalefactor is used and special equipment is available.

The benefits of a computer model is:

- cheap and quick to make
- easy to change

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- flexible excitation
- detailed output data
- direct analysis and presentation of data
- no laboratory and special equipment necessary

Planning of the calculations and preparation of input data may by our program take from 4 to 15 hours for a trained user, dependent on the complexity of the hall.

Running one set of complete calculations may be done in one to two hours. Modifications and repeated runs may be done quickly so 6-10 calculations may be performed during one day.

The source may be given any directivity and the influence of varying directivity and of directions may be studied systematically.

The different types of output data from the same run cost very little and are of great advantage when communicating with the architect, and for the consultant in analyzing and explaining the results.

Our program is based on a large computer, but only commonly used peripherals; an alphanumeric terminal, a printer and a pen plotter.

Due to flexibility and low cost we have found the use of our ray tracing program a very suitable tool in the initial study of a project. Having decided the main shape of a hall, the control of details and the nearly finished design may be done by a real model. The two types of design tools are supplementing, not excluding each other.

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THE PHYSICS OF THE INTERACTION BETWEEN THE MUSICIAN, HIS INSTRUMENT, AND HIS ACOUSTIC ENVIRONMENT.

CHARLES TAYLOR.
UNIVERSITY COLLEGE, CARDIFF.

I feel somewhat diffident about presenting this paper because it takes me into fields that are outside my general experience. However, it is sometimes helpful for an audience of experts to be reminded of some of the simpler principles on which their subject is based, especially if the reminder involves demonstrations. It is very much in that spirit that I agreed to give this lecture.

The essence of music is the communication to the listener of ideas, experiences, or emotions originating with a composer. The various links in the chain of communication involved are: the eye-brain system of the performer in reading and interpreting the score; the psycho-motor system of the performer in controlling his interaction with the instrument; the physics of the instrument itself including its radiation characteristics; the propagation and modification of the resultant sound waves in the total acoustic environment; the resultant feed back to the performer via his ear-brain system; and the response of the ear-brain system of the listener. It might seem most logical to begin at the beginning of the chain and to follow the process through. However, the musical instrument itself is really the heart of the whole process and is the physical link that places the most restraints on the nature of the communication; and the ear-brain systems of both player and listener are links in the system whose properties we must accept as given. I shall therefore begin with these elements and then consider their combined influences on the effects of a given acoustic environment. Those familiar with the tricks of the demonstration-lecturer will recognise that another (possibly the major) reason for this sequence is that it will allow me to put in some demonstrations at a very early stage!

In a conventional symphony orchestra there are three basic types of musical instrument. One group consists of one or more devices each of which vibrates and is directly coupled to the air around it. Examples of these would be mostly in the percussion section - gongs, tubular bells, glockenspiel, etc. The second group consists of resonant tubes, pipes or volumes which are set in vibration by means of a primary vibrator but in which the resonant part of the system has the major influence on pitch. Wind instruments in which the primary vibrator is the reed or the lips of the player are the most obvious examples. The third group has a primary vibrator which is the dominant influence on the pitch, coupled to a mechanical amplifier which has negligible influence on pitch but a major effect on loudness and tone quality. The whole of the string family belongs in this group.

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In whichever of these three categories the instrument falls there are probably four characteristic acoustical properties that we need to consider in relation to the theme of this paper. All are relatively easy to demonstrate. The first is the actual steady-state wave form produced by the instrument sounding a given note and its analysis into harmonic components. The second is the so-called formant characteristic which in some cases results in a relative consistency of tone quality throughout the pitch range (e.g. in the strings) and in others (e.g. the woodwind) may result in characteristic timbre changes in different registers. The third characteristic describes transient changes that occur during the build up and decay of a given note - sometimes called the envelope characteristic. Finally, we need to consider the radiation characteristic which determines what particular frequency components of the resultant sound are radiated in which direction.

The most important feature to be brought out from demonstrations of the steady state wave forms is that, while a genuinely repetitive wave form can be analysed into harmonic components by Fourier methods, no mechanical instrument ever produces a truly repetitive wave form. Pre second-world war text books tended to reproduce wave forms of instrumental sounds lasting only perhaps 1/100 sec and, over this period, the repetition is very good. However, over a period of one second - or even of a tenth of a second - irregularities soon become apparent. They arise principally from the interaction between the player and the instrument; breath control, bow control, etc. all require the feed back system in which the player listens to the sound being produced (and to some extent 'feels' the response of the instrument through lips or fingers) and uses the signals in a kind of servo system to maintain control. But there is always a certain amount of hunting and even the most expert player cannot eliminate this. This effect is one of the powerful reasons why electronic sounds - especially with pre-1950 electronic organs - sound 'electronic'- the repetition is too perfect and the ear-brain system of the listener recognises this.

The formant characteristic can be demonstrated in many ways but perhaps one of the most effective is the aural illusion of the continuously rising tone. My example is computer generated and involves a set of notes an octave apart which are all stepped up a tone at a time; the resultant is effectively processed by an amplifier that has a gaussian frequency characteristic which peaks at about 1 kHz. In other words at low and high frequencies the amplification is very small indeed and the notes 'creep' in and out unnoticed; but the fact that there is the formant peak at 1 kHz dominates the sound and gives the illusion that the rising notes never get anywhere. A useful demonstration of the other aspect of the formant uses a recording of a sequence of notes a fifth apart on, say, a bassoon, starting at the bottom of the register. The recording is then played back on a variable speed machine so that each note in turn is brought down to the pitch of the lowest note, when the change in timbre as one moves through the registers is very marked.

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The effect of envelope shape is most easily demonstrated with a synthesiser, and the initial transient by artificially removing the first fifth of a second from the recorded note of an instrument. It turns out that of all the characteristics discussed here the initial transient is probably the most important in that it triggers off the recognition process in the brain. If this is indeed the case then the effect of the acoustic environment on the initial transients is clearly going to be of great significance.

The radiation characteristic and its variation with frequency can be usefully demonstrated with the so-called 'magic wand' (D.F. Gibbs). This is a miniature microphone adjacent to a pea-lamp at the end of a 'wand'; the microphone feeds the pea-lamp through an amplifier. By adjusting the input level at which the lamp lights, rough loudness contours can be plotted. The radiation polar diagrams (e.g. Meyer 1981) of different instruments must clearly be taken into account in any acoustic design.

We should now turn to the psycho-physical aspects of our theme; that is to the problem of how the brain in interpreting aural signals helps us to survive, gives us a great deal of enjoyment, but, at the same time, puts up considerable barriers that need to be overcome in research in this area. The first point to be made is very simple and very obvious but at the same time of enormous importance. It is simply that all the various sound sources to which either the performer or the listener is exposed all produce independent changes in the pressure of the air; when the separate pressure waves arrive at the ear they all add together algebraically and a single resultant wave causes the movement of the ear drum which sets in motion the train of events which we collectively describe as aural perception. The brain, however, is usually capable of disentangling the components so that we can respond to each separately. This process of disentangling is a complex one and by no means properly understood. That it _is_ performed is easily demonstrated by a mono tape recording - which quite obviously records the variation of only one parameter with time - of a group of instruments or noises which the listener can easily identify separately. A glance at the screen of a cathode ray oscilloscope displaying the corresponding wave form soon confirms the complexity of the task. Equally important is the need to be aware that some combinations of sound cannot be disentangled. The so-called masking effect which arises when sounds within the same critical band are heard simultaneously is an example. Current theories of hearing suggest that a 'pre-sorting' of the incident sound into bands occurs early in the hearing chain and that sounds in the same band interact strongly, whereas sounds in widely separated bands are unaffected by each other. Schouten (1962) has prepared a splendid demonstration using a simple tune in which each note is played twice, first as a pure tone and secondly as a tonal complex (that is a group of high frequency tones which give the required note as a resultant). When low-frequency noise around the frequency of the pure tones is introduced they are masked, but the tune using the tonal complexes can still be clearly heard as the high frequency tones are entering by a different critical band and only being compounded later in the chain. Conversely when high frequency noise is used the pure tones can be heard and the complexes are masked.

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Perhaps the stereo sense is the next that should be considered as it is widely misunderstood. The over-simplified view that our directional sense stems from the time difference between signals arriving at our left and right ears can easily be called into question by a simple home experiment. Lie on the carpet in front of and between the two speakers of a stereo system so that one ear is vertically above the other and the ears are exactly equidistant from the speakers. A stereo test record still works perfectly well. Similarly a stereo illusion can be created with a single loudspeaker if certain frequencies are filtered out to correspond to those eliminated by the diffraction pattern of the head (Bloom 1976).

Most of us have experienced the frustration of performing tests with individuals or with audiences and it is essential to remember how powerful and rapid is the learning-memory-retrieval system of the brain. Unless this is borne in mind and exhaustive, very carefully planned tests are used, little reliance can be placed on test results; yet clearly there can be no objective substitute for audience and performer reaction in determining response to such a subjective phenomenon as music. My favourite demonstration of this consists of a piece of rather crude speech synthesis, so incomplete in fact that an audience fails to understand the sentence being synthesised. However if the sentence is then revealed to the audience and the recording played again they immediately understand perfectly and are amazed that they could not understand the first time.

This section of the subject is a particularly fascinating one and we could easily spend a great deal more time on it; so far we have barely scratched the surface. However though time and space are pressing there is one more vital topic that must be discussed before moving on; it concerns the ability of the ear-brain system to record small time intervals either consciously or sub-consciously. Various ways of demonstrating this using artificial echo on tape and other devices lead to the observation that two single clicks very close together (typically as close as a fifteenth of a second) can still be detected as separate clicks; on the other hand identical lines of speech separated by time intervals as long as about a third of a second are fused into a single line. Here the time interval is recorded sub-consciously <u>not</u> as a time interval but as an apparent variation in the size of reverberation of the room in which the recording was made.

Thus we realize that the ear-brain system adopts a kind of forumla which says "If you hear the same thing several times in very rapid succession its probably the result of reflection from walls with different time delays and is obviously meant to be a single sound; therefore hear it as a single sound". I suppose in some ways this is one of the most important formulae that designers of Acoustic Environments need to recognise. The extraordinary thing is that the ear brain system can distinguish between a repetition with <u>no</u> diminution in intensity from one in which there is a gradual reduction. My demonstration recording with eight repetitions at 1/15 sec intervals all at the same intensity sounds strange and unnatural. The same eight repetitions at the same time intervals but with a 3 db reduction between each sounds completely fused with just a little reverberation.

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We must now begin to draw together the various effects discussed so far to discover their combined effect on the musician and the listener in a particular environment. The study of the physics of the establishment of a sound field in an auditorium begins with the simple idea of surrounding a source with reflecting surfaces which produce sound images of the source in much the way that mirrors would with light. These source-images are at different distances from the observer and consequently each note or sound is heard many times. The combining of these repetitions into a single sound but with a marked change in quality depending on the nature and positioning of the surfaces, occurs both for the instrumentalist and for the listener. Why the change in quality? The two primary reasons are (a) that the wavelengths corresponding to the most used range of audio frequencies (about 6 m to about 30 mm) are comparable with the kinds of dimensions that occur in the room. Waves with a wavelength of 30 mm would be almost specularly reflected by a 1 square metre patch of flat surface; the same surface would produce considerable diffraction effects at 6 m. And (b) that standing wave systems or eigentones are set up in the three-dimensional space enclosing the sound source and, depending on the position of the source may enhance or diminish particular frequencies. Diffraction round an object and the existence of eigentones may all be demonstrated by means of the magic wand already mentioned.

In an extreme example it may happen that components in a particular frequency range may be reflected strongly from a particular wall and components in another range may be absorbed. This can be very disturbing for a performer. I recall that when the assisted resonance system at the Royal Festival Hall was first introduced, one of the complaints by a solo harpsichord player was that notes of different frequencies seemed to be 'coming back to him' from different directions. The precise radiation directions from each instrument in relation to absorptive and reflective elements in the environment must be looked at here. A relatively small change in position or angle of a particular performer can in some circumstances make a considerable change in the tone quality. The possibility becomes even more significant when locating microphones for making recordings.

Orchestra pits present very special problems because, for the higher frequencies, the wavelength is such that relatively little deviation occurs due to diffraction and only those parts of the audience that have a direct sight line to the performers will hear higher frequencies well. The low frequencies however can be diffracted by the edge of the stage and the wall of the pit nearest to the audience and may be distributed relatively uniformly round the hall. The parts of the audience who do not have good sight lines may thus hear mostly the lower frequencies. Conversely the players themselves in the pit will tend to hear greatly distorted sounds if the ceiling and floor are more or less parallel and fairly close together as is often the case. Very strong eigen tones can occur and not only will they hear their own instruments with distorted harmonic content but their hearing of the rest of the orchestra and of the singers on stage will be very badly unbalanced.

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Earlier in the paper the importance of the transient in enabling the listener to identify an instrument was mentioned. There is also, of course, a transient in the excitation of the room itself and this may become more significant than the instrumental transient. I remember some 20 years or so ago a new type of mechanical instrument was developed by a French group called Les Structures Sonores. This was excited by stroking glass rods with wet fingers and it was possible to produce a pleasing variation in the apparent tone by changing the attack from a legato to a staccato effect. However when the group gave a concert during the York Festival at the Minster the long reverberation time of the building became dominant and whatever the stroking technique a slow build-up resulted.

One of the consequences of this kind of effect on the performer is that he may need to adopt a different tempo. In a very 'dry' environment (that is one with little reverberation) rapid staccato passages can be performed with high precision; slower passages may sound unsatisfactory if there is insufficient reinforcement. Conversely, in a reverberant enclosure slow passages may sound well and staccato passages will need to be slowed down if the individual notes are to be preserved.

As we consider these complications we can readily see the source of some of the criticisms of existing concert halls and other acoustic environments both by listeners and performers. No space can be right for all kinds of instruments, all kinds of music, all locations for listeners and performers; there must always be compromise and it is impossible to please everyone. I would therefore contend that the physics and psycho physics are relatively well understood; the real problem facing the acoustic designer is that of identifiying the most influential musicians likely to use the hall, and the seats where the most influential listeners will sit so that at least a reasonable acoustic behaviour can be assured for some people!

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