Music and acoustics

— an interpretation, by John Gardner

An orchestra playing in the Sydney Opera House will sound quite different when playing the same piece in Adelaide's Festival Hall and different again when playing out of doors. Some influences are obvious, some subtle. John Gardner discusses why.

IF YOU were to creep into the Sydney Opera House in the dead of night, stand on the podium and shout "Eureka" at the top of your voice, you might well end up in a mental institution. If you did the same thing at Woomera you could be investigated as a security hazard. However, before they took you away, you might have a chance to observe that the same word shouted in the two locations sounds entirely different in character.

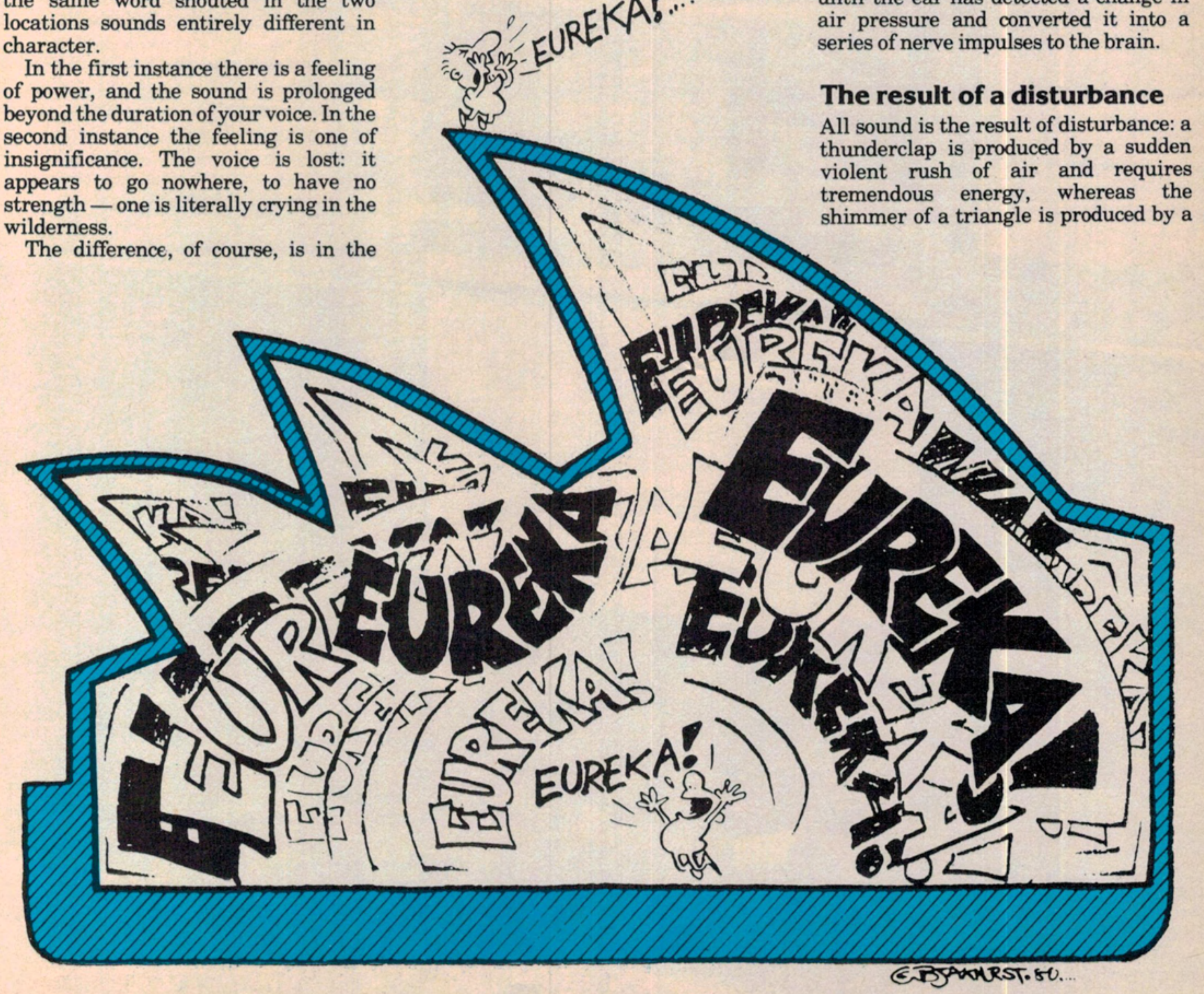
In the first instance there is a feeling of power, and the sound is prolonged beyond the duration of your voice. In the second instance the feeling is one of insignificance. The voice is lost: it appears to go nowhere, to have no strength — one is literally crying in the wilderness.

The difference, of course, is in the acoustic environment: and if we go to another extreme and sing in the bath, the voice immediately becomes more vibrant and we wonder why we were not offered a recording contract years ago. What applies to the voice applies, to a greater or lesser extent, to music: so what significance has the acoustic environment for the music lover? There are several answers to that, depending on whether one is a performer, concert-goer or record enthusiast.

Before we consider these we need to know something about the nature and propagation of sound waves. It is also important to realise from the outset that a sound as such does not exist — not until the ear has detected a change in air pressure and converted it into a series of nerve impulses to the brain.

The result of a disturbance

All sound is the result of disturbance: a thunderclap is produced by a sudden violent rush of air and requires tremendous energy, whereas the shimmer of a triangle is produced by a ▶



gentle vibration. If a simple continuous tone (say from a tuning fork) were picked up by a microphone and the electrical output displayed on an oscilloscope, the trace would appear as in the upper section of Figure 1.

This is really a graph of voltage (vertical) against time (horizontal). The signal produced follows a symmetrical path, rising to a maximum value, and so on.

The distance between successive peaks is a wavelength, and the pitch (or frequency) of a note is determined by the number of wavelengths passing a given point in one second. That is, if 440 wavelengths pass this point in one second, the sound has a frequency of 440 cycles per second, usually known as Hertz (Hz). If the top trace of Figure 1. represents Middle A (440 Hz) with a wavelength in air of 0.75 metres, then the bottom trace is one octave higher, that is, the frequency is doubled to 880 Hz and the wavelength reduced to 0.375 metres.

This representation is a convenient way of expressing sound in visual terms, and it is significant that almost all descriptions of sound quality or texture are analogies of sight and touch. Sound is warm, muddy, coloured, clinical, neutral, clean, fuzzy, plummy, hard, dark brown, open, cold, lush, edgy, ragged, spiky, and so on. This is probably because we cannot easily convey to one another what we hear except by reference to other experiences we have in common.

Sound travels

Wé must now consider how sound travels from A to B. Figure 1 showed the rise and fall of voltage (or air pressure) with time; it did not show how the sound wave moves from point to point through air. This is better demonstrated with a child's toy — the Slinky Spring. If one end is held whilst the other is moved to and fro, a series of ripples appears to move along the spring. What actually happens is that the individual turns of the spiral move to and fro about a mean position and give the illusion of continuous onward movement. This is an almost exact analogy of sound wave propagation.

Imagine a loudspeaker cone vibrating, or the skin of a bass drum. In either case a membrane is oscillating about a central position and alternately moving out into the free air and then back into the body of the instrument. Each excursion into the air causes marginal compression — or a localised increase in pressure. Each incursion causes a marginal reduction of pressure, known as a rarefaction.

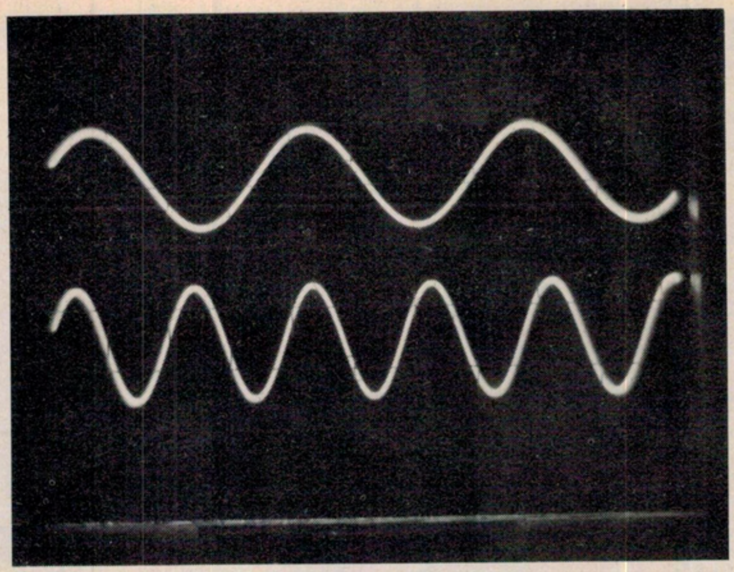


Figure 1. **The rise and fall of a voltage with time — a wave (here, a sine wave).**

So the pulsating drum or loudspeaker is rhythmically disturbing the surrounding air and, as with the Slinky, the disturbance has a 'knock-on' effect, carrying the sound energy away from the source, although any given particle of air is only marginally affected. If it were otherwise, a performance of the "Ride of the Valkyries" would subject you to a gale-force blast and make for rather uncomfortable listening!

We are now ready to discuss what happens in the concert hall. It is well known that musicians are happier and play better in some locations than others. Equally, the concert-goer has preferred venues for particular performances; but the musician and the listener will not necessarily agree what constitutes a good location.

Musical environments

As an extreme example of this we can cite an outdoor symphony concert which, on a beautiful day in a picturesque spot, may seem superb to the listener. In fact he will hear a very unbalanced sound, but his judgement will be impaired by the visual stimuli. The musician will be less enamoured of the location: he will find it hard to judge his touch and his loudncss, and he will have difficulty playing in unison with his colleagues. To assist the musicians in these circumstances, the bandstand or concert platform is composed of hard rostra and there is usually a shell of some sort behind and to the sides of the players so that some of their sound is reflected back and they can hear what they are playing.

What a musician requires, therefore, is an environment that gives something back: if he is playing in an unsympathetic hall he might just as well be outside.

The difference between a good hall and a poor one is not very great: but that small difference depends on subtle and elusive factors that are not easy to define. There are known formulae to determine how 'live' or 'dead' a room will sound, and the strength of various reflections can be predicted. What is

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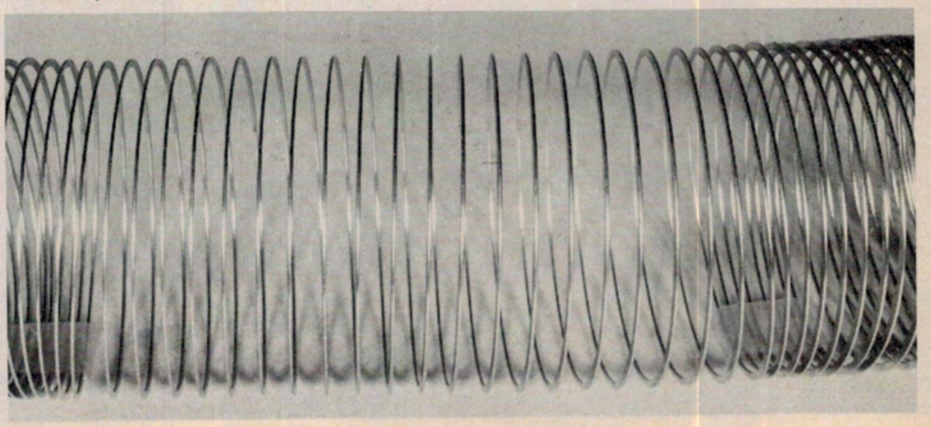
The Slinky Coil, or Tumblebug, can illustrate how sound waves travel, as discussed here.



less easy to predict is whether in practice a hall will be acoustically satisfactory, despite any amount of theoretical calculation.

The BBC has tried to overcome this by making models of proposed new studios in order to investigate their acoustic properties. One of the problems with a model is that if it is scaled down by, say, a factor of ten, then the music must be scaled up by a factor of ten in order to maintain the relationship between room demensions and the sound wavelengths.

Wavelengths

We mentioned wavelengths earlier and it is worth examining them further because they have a direct bearing on the reflection and absorption of sound at various frequencies. Compared with radio and light waves, the dimensions of sound waves are large, especially at low frequencies. For example, a 50 Hz note has a wavelength of 3.4 metres, for 1000 Hz it is 0.3 metres, and for 10 000 Hz it is 0.03 metres.

This means that obstacles such as pillars, seats, people, drapes, panelling and even small rooms are of comparable dimensions with some instrumental sounds. Where an obstruction is large compared with the wavelength of a sound, that sound tends to be reflected, particularly from hard surfaces. A soft surface tends to absorb part of the energy from the wave and so weaken the reflection.

When an obstruction is small in comparison with a sound wavelength, the sound will go around it as though there was nothing in the way. It can be seen, then, that to modify the acoustic character of a studio or hall, low frequencies require large structures, whereas the higher frequencies can be treated selectively with various types of absorbent material.

The most efficient way of absorbing sound is to release it unobstructed into the fresh air. But, as we have seen, fresh air does not make a good playing environment, so we have to create an artificial acoustic that is neither too like a cavern, nor too like the great outdoors.

Of course, some halls exist that are naturally suited to the playing of live music, such as the Sofiensal, Vienna, or St John's, Smith Square in London. Others are specifically treated to make them so.

If an echoey room is filled with a sufficient quantity of the right type of sound absorbing material it becomes 'dead' with a total absorption and therefore no reflection. This is the anechoic chamber, or dead room, used in the development and testing of microphones and loudspeakers. It is uncanny and unnatural in a dead room and the heart seems to beat with sinister power. It is like being in the world of Poe.

It is apparent that by absorbing the right amount of sound at the right frequencies, by controlling reflections so that the orchestral sound is full, but not confused by multiple echoes, a satisfactory acoustic can be created. In order to fix accurately the source of a sound, the ear must hear the direct sound fractionally before the first reflection, so the dimensions of the hall and the seating arrangements are also critical.

Hall's effects

One indication of the suitability of a hall for music is its reverberation time. If a steady tone is sounded in a room until a sound field of constant intensity is created, and then the tone is abruptly stopped, it is possible to measure the decay of sound. Reverberation Time (RT) is the time it takes for the sound intensity to fall to one millionth of its steady value. (To the sound engineer this is a drop of 60 dB).

Good acoustics are not made by slavishly following mathematical formulae, but there is a body of evidence that suggests that the RT should rise in approximate proportion to the volume of the hall. Choral and large scale orchestral works require longer RT than small ensembles. Speech requires as little as 0.3 or 0.4 seconds.

Before it was treated, the Albert Hall in London was a particularly difficult environment. The vast surface area of the dome, the many arches, alcoves, and the numerous pillars all served to disperse the sound so that such reflections as there were had a long time delay. Apart from making playing difficult, a long time delay destroys musical clarity for the listener.

Another fault of the Albert Hall, but common to many others, was that the sound quality varied from one part of the hall to another. In one place the sound would be ill-defined and ragged: elsewhere the musical texture would have the listener wondering what everyone was complaining about.

The large saucer-shaped reflective panels, which were suspended in the dome area some time ago, have considerably improved the acoustics of the hall. However, in our opinion, it remains better suited, visually and acoustically, to the LSO playing the "1812 Overture" or Beethoven's "Ninth Symphony" than to the chamber sound of Neville Marriner and the Academy. Of Queen Elizabeth Hall one might say the opposite.

Naturally a conductor is more sensitive than most to the ambience of a hall. If he is worried by late echoes, or by the deadness of the acoustic, he will be unsettled and his confidence will be affected. This feeling will be communicated to the orchestra and add to their own difficulties. In such circumstances an inspired performance is not to be expected.

Recording the sounds

The ear and the brain together form a complex computer capable of analysing, interpreting, and rejecting stimuli they receive. They can create an image which is more accurate than seems theoretically possible from the information presented. The brain sifts incoming nervous impulses from the two ears and deduces the end product. No one is quite certain how this happens.

What we do know is that neither microphones nor any of the recording equipment has a satisfactory electronic replica of this faculty to discriminate. No matter how many recording microphones are used — and frequently there are too many — they cannot analyse, they cannot reject confusing reflections: they are designed to respond accurately to all changes in air pressure. Whether that change of pressure is due to an incident sound wave, its umpteenth reflection, or to the conductor tapping his foot, the microphone will respond to it and it will be recorded.

Once recorded, most of the 'clues' that the ear used to make its accurate deductions are so scrambled on the tape that they cannot satisfactorily be recovered.

The best we can do is to produce a spatial effect so that we can mentally position the orchestra (stereo): or we can, by recording some of the sound reflections, make a crude attempt at ▶

reproducing the concert hall ambience. The latter is variously known as surround sound, quadraphony, and ambiophony, and it is by his enthusiastic pursuit of the necessary associated hardware that the fool is readily parted from his money!

External influences

However, even if the balance engineer has the good fortune to be working under ideal acoustic conditions, it is unlikely that the location will also be entirely soundproof. This is particularly true of the many halls in and around central London.

One hazard is heavy traffic, which produces a steady pulsating murmur that can occasionally break through to the microphones. Worse by far is the structure-borne vibration generated by Underground trains. This deep-throated rumble is a constant menace to recording engineers throughout the Metropolis: it marches through countless yards of recording tape like a fifth column of frustrated music critics.

Obviously the engineer would frequently prefer to use a custom built studio, acoustically tailored to his needs and also isolated from air and structure-borne noise. He then has more control over the quality of the end product and possibly has more technical facilities available.

Technical compromises

The trouble is that few recording or broadcasting organisations can afford to build a particular type of studio to meet each type of demand. As a result they tend to build general purpose studios having a rather low reverberation time. The theory is that reverberation can be added artificially if required (by the use of reverberation plates and springs, or echo rooms) but that it cannot be taken away.

Many studios are, therefore, too dead for music to be played and recorded naturally and, as we have seen, this deadness causes problems for the players. So, although the balance engineer may get a better sound and feel more in command of the situation, the quality of the performance may suffer.

Light music and pop are exceptions to the argument and, because of the specialist techniques required, these are almost invariably recorded in a studio. Reverberation is added artificially and selectively to the different instruments. Often several reverberation plates will be used, set to different times, so that the delay, say on a vocal, will not be the same as that on the strings.

There are endless permutations of the facilities and, by using a multiple microphone array, nature is defeated and a type of sound is created on tape which would be impossible in the concert hall.

These techniques are a commendable extension of the recording art when confined to the right repertoire. Unfortunately, many a classical record producer allows himself to be hypnotised by the electronic gadgetry: he will exaggerate reality and highlight an instrument that would normally be heard only as part of an ensemble, thus distorting the composer's intention.

However, because many record enthusiasts prefer to demonstrate their hi-fi equipment than listen to music, such producers tend to enjoy a degree of success out of proportion to their taste! ●

Of course, there are those who take the "wall of sound" approach, in which case venue acoustics matter nought (. . . Ed.).