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CURRENT DEVELOPMENTS IN ANALOGUE ACOUSTIC MODELLING

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Although all our analysis at Cambridge is performed with analogue devices, it is important to appreciate that even with computer systems much of the equipment used has to be analogue and many decisions are based on limited transducer capabilities. All the remarks in this paper relate to auditorium models and most of the discussion is concerned with objective measuring techniques. Only air will be considered as the model medium.

Frequency limitations in modelling

There is an effective upper frequency limit in auditorium model work of about 100kHz irrespective of scale. Apart from transducer limitations, it is instructive to discuss the role of air absorption. Of the two components of air absorption, the classical and the molecular, the latter can be reduced to minimal values by using a dehumidified atmosphere. The air absorption in the model will be dominated by the unavoidable classical component and if this matches the scaled-up molecular absorption at full-size, one is in the fortunate position of having perfect acoustic modelling (in which the sound field in the model is a replica in miniature of that at full size with all processes speeded up). This desirable state of affairs will generally occur at some frequency for any scale factor; at a scale of one eighth it occurs over virtually all the audible frequency range. The reason for this is as follows:

The molecular absorption coefficient per wavelength has a maximum at the relaxation or Napier frequency. At 20°C, 50% r.h. the Napier frequency is 43kHz, i.e. significantly above the audio range. At frequencies well below the Napier frequency the molecular absorption per unit length is a function of frequency squared. However the classical absorption coefficient is also a function of frequency squared and the optimum scale factor to obtain identity between the relevant constants is close to 8. Even at the full size frequency of 16kHz, the effective error in the air absorption coefficient in a dried model (at 2% r.h.) is less than 4%. Errors at this scale are in fact largest at low frequencies.

At larger scale factors high frequency air absorption is excessive in the model and corrections have to be applied where this is possible. Obviously the effects due to excessive air absorption must remain small relative to effects being measured.

Measurement of reverberation time

Requirements formeasurement of reverberation time (R.T.) are not critical. To measure R.T. in a model requires a facility for temporal transposition as level recorders cannot operate fast enough. This is generally done with a multi-speed tape recorder with a pre-recorded series of noise bursts.

Impulsive measurements in models

Apart from the measurement of steady state sound level, most other measurements use impulsive type signals. To perform impulse tests two facilities are necessary which are frequently absent from standard equipment: trigger delay systems and peak reading meter systems. The general analysis system consists of a gating system and a squarer (to produce voltage squared) followed by an integrator. This enables the energy content of temporal segments of the received response to be calculated to measure, for instance, the ratio of early-to-late energy.

The same measuring system can be used for measurement of Early Decay Time (E.D.T.) A convenient method is to store the impulse response on a digital store (similar to B and K Digital Event Recorder) and use the difference method due to Kuttruff and others [1] to derive the time reversed integral.

With current electronic devices operating comfortably up to 200kHz the analysis of signals from models presents no particular problems. The problems of modelling are substantially those of transducers. For impulsive testing a single source transducer cannot cover 5-6 octaves with an omni-directional characteristic and two distinct testing systems are used at Cambridge for 1:8 scale, as summarised in Table 1. The frequency limit is principally determined by problems of producing an omni-directional loudspeaker.

Frequency:	Below 3kHz	Above 3kHz
Source:	Loudspeaker	Spark
Microphone:	Electret or $\frac{1}{4}$" condenser	$\frac{1}{8}$" condenser

Table 1: Transducers for 1:8 scale models

Loudspeaker sources

Because loudspeaker magnets cannot be scaled down without serious loss of output, model loudspeakers are difficult to produce with omni-directional characteristics. A small loudspeaker unit is required for omni-directionality, yet small units tend to have high lowest operating frequencies. In line with current practice a dodecahedron array is used, yet the individual units with 50mm diameter cones can barely handle frequencies below 500Hz and the array is already becoming directional at 4kHz.

The alternative for an omni-directional source is to use a tube mounted on a driver. Longitudinal resonances in the tube can be compensated for electronically, but the system is temperature sensitive and therefore clumsy in use. For ultrasonic frequencies, electrostatic transducers are available. When placed in a spherical array they constitute an omni-directional source, but the polar response is not sufficiently uniform for objective testing.

Loudspeakers, of course, do have the unique property of accepting any signal which makes them extremely flexible and there remains much scope for further development of model loudspeaker systems.

Spark sources

A spark source produces an intense, concentrated, short duration pulse which makes it ideal for high frequency work where an omni-directional characteristic is required. As a source it is however so intense that non-linear propagation will occur in the vicinity of the source. This probably makes its use unsuitable in experiments involving diffraction near the source, such as diffraction round screens, but it is felt not to be too serious in the auditorium situation, in which propagation close to the source is not critical, being frequently dominated by a simple reflection from the floor. The spark energy determines two basic characteristics: the frequency, f_m, of maximum energy (i.e. also pulse duration) and the limit of non-linear propagation, r_o. Precise relationships are not possible since the characteristics of the source are also determined by the electrode shape and spacing, which also influence the acoustic efficiency. Approximate relations are as follows:

$$f_m \approx k_1 E^{-1/3} \quad ; \quad r_o \approx k_2 E^{1/3}$$

where E is the electrical energy dissipated in the spark, and k_1 and k_2 are constants. This implies that non-linear propagation becomes less serious (at scale distances) for larger scale factors. Small energy discharges should be used commensurate with the frequency spectrum required. For 1:8 scale we use a 2J discharge which is considered to be linear beyond 1.25m (= 10m full size) and has peak energy around 12.5kHz.

Microphones

Microphones become directional at frequencies lower than generally appreciated. The maximum frequency for use as an omni-directional microphone for objective tests is about 15kHz for a ¼" microphone and 30kHz for a ⅛" microphone. The use of a nose cone improves the situation somewhat, but at the expense of some stretching of the impulse response. Smaller microphones have the disadvantage, of course, that they are significantly less sensitive.

For audio frequencies miniature electret microphones are very attractive, being low in cost, with a directionality similar to a ¼" microphone but with lower noise levels.

Test signals

For good temporal discrimination a short duration signal should be used for impulsive tests. Since results are generally presented in octaves, a signal with a flat spectrum within the octave band of measurement is desirable, since a single measurement per octave can then be made. A Dirac delta pulse or its square-wave approximation has a constant percentage bandwidth spectrum of +3dB/octave increase below the frequency of maximum energy. It is possible to compensate for the impulse response of a loudspeaker by computing the optimum signal by Fourier transformation.

Since a concert hall is a linear system, compensation for the signal can be undertaken either before the source transducer or on the received signal. The signal from a spark source approximates an N-wave with a +9dB/octave spectrum. If an integrator (with a -6dB/octave characteristic) is introduced prior to analysis, the effect is equivalent to the signal being positive going only with a +3dB/octave spectrum. The impulse response can then be analysed in octaves.

Subjective model testing

The opportunity to test auditoria subjectively is particularly attractive. This can be done at 1:8 scale with both music and speech with acceptable signal-to-noise ratios. Again it is the transducers which present the problems. For music, the BBC research dept. [2] developed a scale version of a monitor loudspeaker for model studios. This works well in concert hall models though the directionality of the orchestra is not particularly well reproduced. A transducer for speech with the correct directionality for a human speaker has been developed by Orlowski [3]. Briefly, the requirement was for sound to radiate from a small aperture at a point remote from the loudspeaker magnet. This was achieved by placing a tapered horn on a tweeter unit, with an electronic compensation system to suppress reflections along the pipe.

¼" microphones are the obvious choice as microphones. These have to date been mounted on either side of a scale head. Though subjectively the results sound acceptable, attempts to simulate the human outer ear system more closely are required. To do this without a serious reduction in the signal-to-noise ratio is likely to be difficult.

Acoustic modelling with large scale factors

The most significant recent development for auditorium model testing has been the investigation of models with scale factors of greater than 1:8. A substantial study of models at 1:50 scale has been undertaken, including a comparison exercise between a small auditorium and its model. This has already been reported and is fully discussed in [4]. It was found that reverberation time could be measured up to the 2kHz octave equivalent and point-to-point impulse responses and their subsequent analysis up to the 1kHz octave equivalent. A comparison, for instance, between the model and full size auditorium of the ratio of early-to-late energy showed good agreement.

Larger models enable much more detailed analyses on auditoria to be made, including the very valuable subjective facility. Objective testing in 1:50 models promises to provide a worthwhile complementary function. The small model can be built early in the design programme, and the effect of more substantial changes in design can be investigated economically.

Conclusions

The influence of room shape and detailed design on the acoustic characteristics of an auditorium is still little understood. Computer models are still unable to deal with the complexities associated with wave acoustics. The much greater knowledge available after testing small and large acoustic models should enable us to provide good acoustics with much greater confidence.

References

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