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IMPULSE TESTING OF CONCERT HALL ACOUSTICS:

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1. INTRODUCTION

Progress in concert hall acoustics over the last 30 years rests on the division of the sound arriving at a listener into an early and a late component. To make objective measurements involving this division clearly involves impulse techniques. The techniques used have not changed greatly over the years, though clearly the advent of computers offers advantages. The current emphasis in objective concert hall acoustics is to evaluate integrated energy. A characteristic of concert hall measurements is that they are generally made with the intention of expressing results in octave bands, the five octaves between 125Hz and 2KHz being frequently used. Before considering integrated energy measurement, the traditional impulse response of a room displayed on an oscilloscope screen will be discussed.

2. VISUAL IMPULSE RESPONSE

To produce an impulse response the signal which is used to excite the room is generally a spark discharge or a pulse fed to a loudspeaker, which for concert hall use generally has an omnidirectional characteristic.

Impulse responses clearly vary from one seat position in a hall to another and between one hall and another. It is tempting to imagine that some method is available to interpret impulse responses in terms of subjective effects, yet extensive study under Meyer at Göttingen and elsewhere during the 1950's yielded very little of diagnostic value. The most obvious characteristic of impulse responses in a hall is the relative importance of direct sound to reflections. As illustrated diagrammatically in Figure 1, for receiver positions near the source (a) the direct sound is markedly more intense than first reflections, which only arrive after several tens of milliseconds. At positions near the back of the hall (b) the reflection density is high, the first reflections have an intensity only slightly less than than that of the direct sound and they arrive soon after the direct sound

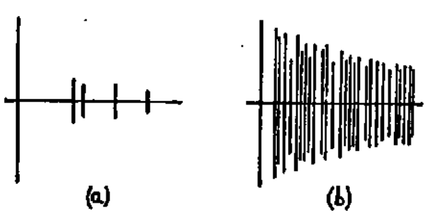

Figure 1

There are subjective differences between the sound at the front and rear of concert halls, but experience of listening in concert halls suggests that the inevitable continual transition between the extremes in Figure 1 as a function of source-receiver distance is unlikely to have any overriding subjective correlate. A more hopeful comparison is between impulse responses at similar source-receiver distances, though this obviously restricts the usefulness of the impulse response as an indicator. A study, containing subjective assessment as well as impulse response measurements, was made in 1974 in the Gulbenkian Great Hall, Lisbon [1]. As the stage was highly variable, different stage configurations could be tested with two listening positions in the auditorium. Comparison between the impulse responses and subjective responses is instructive. It is possible in many instances to point to characteristics in the impulse response which probably account for the perceived subjective change but to predict from impulse responses would seem rash. There were clear examples of obvious changes in the impulse response with minimal corresponding change in the subjective assessment.

Some of these criticisms of the value of impulse responses relate only to the admittedly qualitative assessment of an oscilloscope trace. Statistical analyses of reflections have however been largely unsuccessful. The measures based on integrated energy described below are now considered more valuable guides to subjective quality. The impulse response however remains a simple indicator of strong early reflections. Clearly however interference between several reflections arriving simultaneously can complicate interpretation.

The ideal impulse response would originate from an ideal impulse, or Dirac pulse waveform produced in the hall. A measured impulse response is the convolution of this (ideal) room impulse response and the time signature of the actual acoustic pulse introduced. Interference is clearly influenced by this time signature, particularly its duration. The frequency spectrum of the impulse is also very important. A short duration pulse is desirable to avoid interference, yet the shorter the duration, the more high frequency components will dominate the observed response, and reflections off acoustically small surfaces barely important subjectively become visible. A pulse with a duration of about 1 ms seems a reasonable compromise. Display of impulse responses is currently often made with pressure squared; not only does this correspond with integrated energy concepts but also by suppressing low level sound discrete reflections become easier to identify.

3. TEMPORAL ENERGY ANALYSIS OF THE MONOPHONIC IMPULSE RESPONSE

Many objective measures have been proposed which relate to the monophonic impulse response. Foremost among these is the concept of the ratio of early to late energy or the early energy fraction. The latter was originally proposed by Thiele in 1953, and is expressed mathematically as

$$D = \frac{\int_0^{80ms} p^2(t).dt}{\int_0^{\infty} p^2(t).dt} \qquad \text{— (1)}$$

where p(r) is the instantaneous pressure at time t at the listening point and t = o corresponds to the arrival of the direct sound.

The choice of impulse signal proves not to be too critical for this measurement; this aspect will be discussed below. The measurement also requires a squarer and integrator. Currently for music the time interval of 80 ms is generally used for the early sound; the measure relates to the perceived balance between clarity and reverberance.

A very valuable conceptual link was revealed by Schroeder in 1965, which relates the reverberant decay of a noise excitation to the response of the room to an impulse. The problem of the noise decay is that it contains random fluctuations superimposed on the decay. The exact form of these fluctuations is a function of the time history of the excitation signal prior to switch-off.

One way to eliminate these fluctuations, though it requires a sophisticated storage facility to do it, is to sum a large number of noise decays taken under otherwise identical conditions. Schroeder showed that the ensemble average of the squared noise decays $\langle n^2(t)\rangle$, is identical to a certain integral of the squared impulse response, $r^2(\tau)$, of the enclosure plus filters included in the measuring system.

$$\langle n^2(t)\rangle = K.\int_t^{\infty} r^2(\tau).d\tau = K.\left[\int_0^{\infty} r^2(\tau).d\tau - \int_0^{t} r^2(\tau).d\tau\right] \quad -(2)$$

The constant K simply takes account of level differences. The time interval in the integral is between t and infinity. Conceptually easier to understand is the integral between time zero and t (as contained in equation (1)) and it is clear that the integral between t and infinity is identical to the total integrated energy minus the integral from zero to t. This is illustrated in Figure 2, which shows the complementary relationship between sound build-up and decay. As well as providing a link between noise decays and impulse responses, Schroeder's analysis provides a method for measuring the reverberation time without random fluctuations inherent in the standard noise method. The impulse response is a unique description of a linear system; the reverberant decay is related to the squared integrated impulse response.

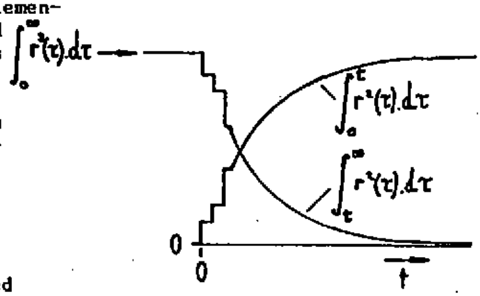

Figure 2

By measuring the integrated energy at t = 50 ms on a decay curve, such as in Figure 2, and the total integrated energy, one can derive the value of D (Equation (1)). There is however the problem that r(t) includes the response of filters in the measuring system. Again the signal for analysis is the convolution of the impulse responses of the filter(s) and the room. The effect of a bandpass filter will be slurring of the signal envelope as well as a delay in the response. Since this delay is between 5 and 6 periods for a $\frac{1}{3}$ octave filter (and until recently octave filters), this becomes significant particularly at low frequencies. To measure the early energy fraction we have to measure integrated energy at a time which includes filter delays;. there is also the problem of determining t = o which requires great care.

A convenient alternative method for measuring the proportion of early energy is to use a short duration signal, to gate the response and then filter before measuring the energy. This is valid as long as the gating introduces minimal spectral components, which is the case for sampling durations of 50 ms or longer.

4. CHOICE OF IMPULSE SIGNAL

The requirements for an impulse signal are as follows:

(a) short duration
(b) adequate energy
(c) reasonably flat spectrum in the measurement range

An additional requirement is introduced if no filtering is envisaged in the analysis, namely that the bandwidth be narrow. It is perhaps easiest to discuss this situation first. What is required is a signal with a small duration-bandwidth product. The useful signal envelopes are familiar from window functions for non-stationary signals. Kurer used a Gaussian envelope, which has the virtue of having no side lobes. Atal et al propose the use of a Hamming envelope, that is a single cycle of a cosine^2 on a small pedastal, which has the property of low level side lobes. The signals clearly have to be radiated from a loudspeaker. In terms of maintaining good temporal resolution within a specified bandwidth they are near-optimal

For wide band signals of short duration, spark sources have been used for many years and they remain the most popular measuring signal. They do however have two significant problems associated with them. Firstly being intense and physically compact sources, the initial propagation is non-linear, a situation about which several aspects remain unclarified. The second problem relates to the spectrum: the time signature of a spark signal is generally similar to a single cycle of a sine wave. At low frequencies the spectrum decreases 9dB/octave on a constant percentage bandwidth basis. This can cause inadequate signal-to-noise ratio at low frequencies and implies a non-flat spectrum for octave measurements. Inclusion of a 6dB/octave low pass filter (or integrator) on analysis is a simple solution to the spectrum shape problem with no penalty in the form of stretching of the impulse response.

With a loudspeaker source a single cycle sine wave can also be used with the same integrator compensation system to enable several octaves to be measured from a single impulse. Signal-to-noise ratio is likely to be poor at low frequencies, a low frequency single cycle pulse is generally required for these octaves. A more elegant technique than compensating for spectral characteristics of the source signal is to use a pulse approximating a Dirac delta pulse. A single polarity pulse, such as a half-cycle sine wave between zero crossings, of suitable duration is satisfactory. Loudspeakers however generally ring when subjected to such signals. Winter et al have shown that in certain cases the loudspeaker impulse response may be compensated by feeding a signal to the loudspeaker with an inverse (complex) spectrum calculated by Fourier transform.

1: M. BARRON 1978 Journal of Sound and Vibration 59, 481-502. The Gulbenkian Great Hall, Lisbon, II: an acoustic study of a concert hall with variable stage.