New Method of Measuring Reverberation Time

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A new method of measuring reverberation time is described. The method uses tone bursts (or filtered pistol shots) to excite the enclosure. A simple integral over the tone-burst response of the enclosure yields, in a single measurement, the *ensemble average* of the decay curves that would be obtained with bandpass-filtered noise as an excitation signal. The smooth decay curves resulting from the new method improve the accuracy of reverberation-time measurements and facilitate the detection of nonexponential decays.

INTRODUCTION

THE accuracy with which reverberation times can be determined from decay curves is limited by random fluctuations in the decay curves. These random fluctuations result from the mutual beating of normal modes of different natural frequencies. The exact form of the random fluctuation depends on, among other factors, the initial amplitudes and phase angles of the normal modes at the moment that the excitation signal is turned off. If the excitation signal is a bandpass-filtered noise, the initial amplitudes and phase angles are different from trial to trial. Thus, for the same enclosure, and identical transmitting and receiving positions within the enclosure, different decay curves are obtained—the differences being a result of the randomness of the *excitation signal*, not of any changes in the characteristics of the enclosure.

A method frequently used to minimize the effect of the fluctuations in decay curves on the measured reverberation-time values is to repeat the reverberation experiment many times and to average the reverberation times (or decay rates) obtained from the individual decay curves. However, this method is not only inefficient but it fails to reveal the true nature of the decay. In particular, it is often impossible to detect the existence of multiple decay rates—especially high initial decay rates that persist for only a few decibels. This shortcoming of conventional decay measurements is regrettable because the initial portion of the decay contains much valuable information. For example, decays with multiple slopes point to a lack of sound diffusion—an important factor both for absorption measurements in reverberation chambers and the evaluation of concert-hall acoustics. Furthermore, in reverberation chambers in which the diffusion decreases during the decay, it is the *initial* decay rate that is important for the determination of the statistical absorption coefficient of the test material. Finally, there is increasing evidence that the initial decay rate of the reverberation process in halls for speech and music is at least as important for subjective "reverberancy" as the later portions of the reverberation decay. To extract more, and more useful, information from decay curves, many such curves (obtained under identical physical conditions) should be averaged—not just the decay rates or reverberation times obtained from individual decay curves. Unfortunately, averaging over many decay curves is rather impractical because of the labor involved and the lack of suitable automatic devices with sufficient storage capabilities.

In the following, a new method for measuring reverberation time is described, which, in a single measurement, yields a decay curve that is identical to the average over infinitely many decay curves that would be obtained from exciting the enclosure with bandpass-filtered noise. Thus, the above-mentioned difficulties are avoided.

I. THEORY AND EXPERIMENTAL METHOD

Let $n(t)$ be a "stationary white noise." "Stationarity" means that its autocovariance function $\langle n(t_1)\cdot n(t_2)\rangle$ (the brackets denote "ensemble average") depends only on the time *difference* t_2-t_1. "Whiteness" means that the autocovariance function is zero everywhere, except for $t_1=t_2$. Thus, one may write

$$\langle n(t_1)\cdot n(t_2)\rangle = N\cdot\delta(t_2-t_1), \tag{1}$$

where N is the noise power per unit bandwidth and $\delta(t_2-t_1)$ is the Dirac delta function.

In reverberation-time measurements, the noise is usually filtered to give it the desired spectrum with a bandwidth covering typically 1 or $\frac{1}{3}$ octave. The filtered noise is radiated into the enclosure. When a "steady state" is reached, the noise at the input to the filter is switched off. The signal received at a receiving point in the enclosure is then

$$s(t)=\int_{(-\infty)}^{0} n(\tau)\cdot r(t-\tau)d\tau, \quad (2)$$

where $r(t)$ is the *combined* impulse response of the system consisting of the noise filter, amplifiers, transducers, *and* the enclosure between the transmitting and receiving points. The upper limit in the integral is the time at which the noise is switched off ($\tau=0$). The notation for the lower limit ($-\infty$) is meant to indicate that the noise is switched on sufficiently before the onset of the decay for the sound field in the enclosure to have reached a "steady state." In practice, ($-\infty$) might be -2 sec.

The square of the received signal may be written as a double integral as follows:

$$s^2(t)=\int_{(-\infty)}^{0} d\tau \int_{(-\infty)}^{0} d\theta\, n(\tau)\cdot n(\theta)\cdot r(t-\tau)\cdot r(t-\theta). \quad (3)$$

By averaging the squared received signal over the ensemble of noise signals and applying Eq. (1), one obtains

$$\langle s^2(t)\rangle=\int_{(-\infty)}^{0} d\tau \int_{(-\infty)}^{0} d\theta N\cdot\delta(\theta-\tau)\cdot r(t-\tau)\cdot r(t-\theta). \quad (4)$$

Since $\delta(\theta-\tau)$ vanishes, except when $\theta=\tau$, and since the integral over the delta function equals unity, integration over θ yields

$$\langle s^2(t)\rangle=N\cdot\int_{(-\infty)}^{0} r^2(t-\tau)d\tau, \quad (5)$$

or, after substituting the new integration variable x for $t-\tau$,

$$\langle s^2(t)\rangle=N\cdot\int_{t}^{(\infty)} r^2(x)dx. \quad (6)$$

According to Eq. (6), the ensemble average of the squared noise decay $\langle s^2(t)\rangle$ is identical to a certain integral over the squared impulse response $r^2(t)$ of the bandpass filter (including amplifiers and transducers) connected in series with the enclosure. In practice, the ensemble average would require a large number of measurements. By contrast, the right-hand side of Eq. (6) requires only a single measurement.

The function $r(t)$ can be considered the response of the enclosure when excited by the impulse response of the bandpass filter. The impulse response of a bandpass filter used as an excitation signal may be thought of as a filtered pistol shot or "tone burst" with a spectrum identical to the spectrum of the filtered-noise signal. Thus, one may summarize the result of the foregoing theoretical analysis as follows. The value of the ensemble average of squared noise responses at time t after the onset of the decays equals the squared tone-burst response integrated from time t to "infinity" (in practice, several seconds), provided that the tone-burst energy spectrum is identical to the noise-power density spectrum.

Equation (6) also contains the answer to the frequently posed question concerning the relation between the decays obtained with tone bursts (or filtered pistol shots) and bands of noise. The answer is that there is no direct relationship but that an *identity* exists between the average squared noise decay and an integrated squared tone-burst response.

The integration in Eq. (6) can be evaluated by hand with a planimeter or automatically on a digital computer. For automatic evaluation without a digital computer, the variable *lower* limit t in the integral poses a problem. It can be solved, however, by recording the sound-pressure responses on magnetic tape and reversing the direction of time by playing the recording backward. Thereby, the variable lower-limit integration in Eq. (6) becomes variable upper-limit integration,

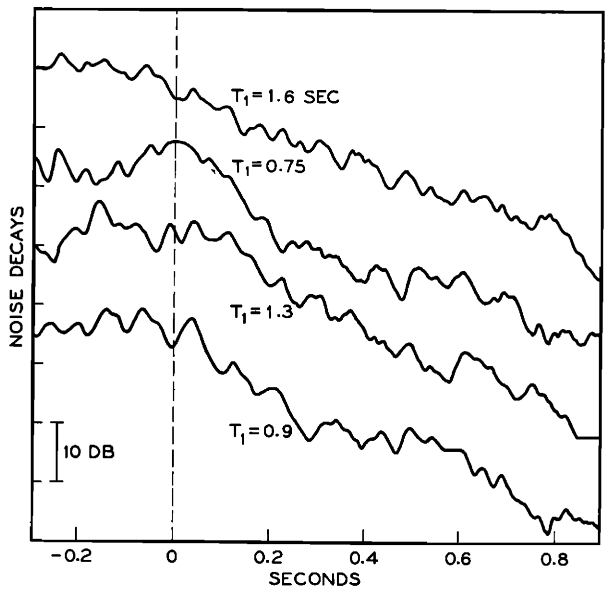

FIG. 1. Four noise-decay curves in Philharmonic Hall, New York (19 October 1963), obtained under identical conditions with $\frac{1}{3}$-oct band of noise centered at 167 cps. Noise source was near center stage, receiving point on Second Terrace (Seat A-221). Both loudspeaker and microphone had omnidirectional characteristics. The dashed vertical line indicates the time at which the noise signal was switched off. Note the large differences in the four decay curves that result from the randomness of the excitation signal. The values of T_1 are reverberation times based on straight-line fits to the first 10 dB of the decays.

which can be instrumented, for example, by a simple RC-integrating circuit.

In summary, then, the new method, called "integrated tone-burst method," works as follows. Tone bursts whose spectra cover the desired frequency bands are radiated into the enclosure from a loudspeaker. The response of the enclosure to each tone burst is picked up by a microphone whose output is recorded on magneic tape. The tape recording is played back in reversed-time direction. The output signal from the tape recorder is squared and integrated by means of an RC-integrating network. The voltage on the capacitor is then the desired decay curve (on a reversed-time scale). If the decay curve is to be converted to a logarithmic amplitude scale, care must be taken that the integration start after the reverberation signal exceeds the noise background.

II. RESULTS AND DISCUSSION

The reduction of randomness in decay curves achieved by this method is illustrated in Figs. 1 and 2. Figure 1 shows four noise decays obtained for the same transmitting and receiving positions and identical physical conditions in a large concert hall (Philharmonic Hall, New York, 19 October 1963; noise source near the center of the stage, receiving point on Second Terrace, Seat A-221). The noise band had a center frequency of 167 cps and a 3-dB bandwidth of 38 cps, corresponding to a third of an octave. The time at which the noise source was switched off is indicated by a vertical dashed line. The most severe random fluctuations in the decay were smoothed out by the finite writing speed of the level recorder (100 mm/sec). Even so, the remaining fluctuations do not allow one to make meaningful statements about the nature of the decay and the initial and final decay rates on the basis of a single decay curve. For example, the initial decay rates defined by best straight-line fits over the first 10 dB of the decays vary over a range of more than 2:1. The corresponding reverberation times T_1 vary from 0.75 to 1.6 sec. Also, it is difficult to say on the basis of the four noise decays whether the logarithmic decay curve is essentially linear or double-sloped. The topmost decay in Fig. 1, for example, looks quite linear, while the decay directly below it seems to stem from a strongly nonlinear (nonexponential) reverberation process. What are the actual characteristics of the reverberation process in this hall (for the stated frequency band and locations) when the decay is not "masked" by the irrelevant randomness resulting from the exciting-noise signal? What are the reverberation times corresponding to the initial and final decay rates and are they different?

The answers to these questions are provided by a measurement according to the integrated-tone-burst method described above. The upper half of Fig. 2 shows the tone-burst response for the same loudspeaker and microphone locations that were used in the noise decays

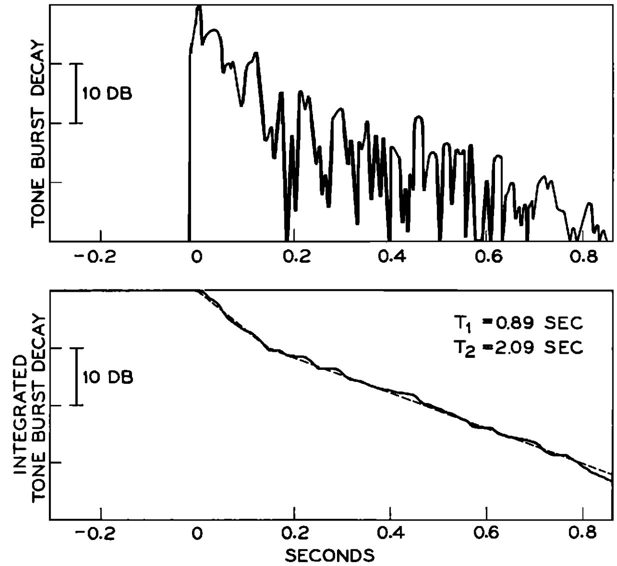

FIG. 2. *Upper half:* Tone-burst response for same loudspeaker and microphone positions as in Fig. 1. Tone-burst spectrum identical to spectrum of noise signals used for the noise decays shown in Fig. 1. *Lower half:* Squared tone-burst response integrated from time t to 1 sec. Double-sloped nature of decay is clearly visible. Reverberation times T_1 and T_2 are obtained by best straight-line fits to first 10 dB and remainder of the decay, respectively.

shown in Fig. 1. The tone-burst spectrum was identical to that of the noise signals used in the noise decays. The lower half of Fig. 2 shows the integral over the squared tone-burst response from time t to 1 sec. The greater regularity of the integrated tone-burst decay is immediately apparent. This was to be expected because, according to Eq. (6), the integrated tone-burst decay is the average of infinitely many noise decays like those shown in Fig. 1. Also, because the squared tone-burst response is a *positive* function of time, its integral is a *monotonically decreasing* function of time—in contrast to noise decays that fluctuate both down and *up* (see Fig. 1). The fact that the decay curves obtained by the integrated tone-burst method are monotonically decreasing is in accordance with intuition, which requires that the sound energy in an enclosure should always *decrease* with time when no energy is being radiated into the enclosure.

The monotonicity of the decay of the integrated tone burst can be seen in the lower half of Fig. 2. The double-sloped nature of the decay is also clearly revealed. On the basis of such decay curves, reverberation times can be determined with great precision. In the example, the "initial" reverberation time T_1 (for the first 10 dB of the decay) is 0.89 sec and the "final" reverberation time T_2 (for the remainder of the decay) is 2.09 sec.

It is particularly noteworthy that even the first few decibels of the decay show little deviation from a straight line. Thus, when using the integrated-tone-burst method to measure reverberation time, one does not have to omit this important portion of the decay.

The remaining small fluctuations in the integrated tone-burst decay result entirely from properties of the enclosure (arrival-time distribution of the echoes) in the chosen frequency band and for the loudspeaker and microphone positions selected within the enclosure. If desired, these fluctuations could be further reduced by averaging over several different paths within the enclosure and by employing wider frequency bands for the tone bursts. (At higher frequencies, the bandwidths are usually larger than the 38 cps, used for the decay shown in Fig. 2, anyhow.)

The greater precision of measured reverberation times, obtained by the integrated-tone-burst method, should also benefit both the accuracy and reproducibility of measurements of sound-absorption coefficients in reverberation chambers. In addition, the easily discernible occurrence of double-sloped decays could serve as a reliable indicator of insufficient sound diffusion.

More importantly, it is hoped that the new method will lead to a better understanding of the relation between measured reverberation times and subjectively experienced reverberation.