11.A Appendix: metrology & instrumentation

In the following we will give a short overview regarding some of the devices and methods necessary for loudspeaker measurements. More extensive information is available from publications of the instrumentation-manufacturers, in particular from Brüel&Kjaer (Technical Reviews).

11.A.1 Measuring microphones

In contrast to microphones used in the recording studio, the transmission coefficient T_{Up} of measuring microphones should be frequency-independent [3]. G_{Up}, the transmission index of a typical free-field microphone (½" B&K 4190), for example, varies by less that ±0.7 dB in the range from 10 Hz to 15 kHz. This range of tolerance is, however, valid only for axial sound incidence; as soon as the direction of the sound deviates from the microphone axis, beaming effects that increase with rising frequency make themselves felt. In the anechoic chamber (AEC), such **beaming** is of no effect since the microphone is directly pointed towards the source. However, in the reverberation chamber (RC) with its diffuse sound field, attenuation towards the higher frequencies occurs that can easily amount to 5 dB at 15 kHz. For this reason, ¼"-microphones are preferred in the RC, and the fact that they are noisier compared to the ½"-microphone is accepted in exchange. The B&K 4135♣ used for our measurements has a beaming error of 0.5 dB at 5 kHz and of 1 dB at 10 kHz – this we deemed acceptable.

Non-linear **distortion** (harmonic distortion) is far below any relevance in the microphones used, and at the sound pressure levels that occurred. The **intrinsic noise** is insignificant for the 4190 (15 dB_A), and marginal for the 4135 (45 dB_A). Not insignificant are the effects of the **microphone mounting**: clamps and stands reflect waves and lead to comb-filter-like interferences♥. With suitable set-ups, such errors could however be kept below ±0.2 dB.

11.A.2 Reverberation time

The time it takes for the diffuse-field SPL to drop by 60 dB in the reverberation chamber after the sound source is switched off is specified as the reverberation time. In order to mainly measure diffuse sound, the microphone must not be located too close to the sound source, and to catch as many room modes as possible, the microphone should move a long a (slanted) circular path. All measurements in the reverberation chamber were done with 50%-overlapping $1/3^{rd}$-octave analysis (IEC 1260 class 0), with the microphone moving along a circle (∅ = 3m) within 80 s. The microphone boom was mounted to a **turntable** (B&K 3299). The latter transmitted a lot of **structure-borne sound** to begin with (equivalent air-borne SPL 84 dB); however, suitable decoupling reduced this value to some just-about-acceptable 45 dB.

Customarily, the reverberation chamber is excited via broadband noise to determine the reverberation time. After switching off the sound, the slope (dB/s) of the level decay is identified, and the reverberation time T_N results from it. Typical values are 2 – 5 s, and up to 10 s in the low frequency range. Since noise processes are of a stochastic nature, it is necessary to average over several decay processes.

♣ Brüel&Kjaer does offer a special pressure microphone (4136) that would be even more suitable.
♥ M. Zollner: Einfluss von Stativen und Halterungen..., Acoustica, Vol. 51 (1982), 268-272.

 Translated by Tilmann Zwicker

Even before this averaging (over several level-decay curves), an **RMS-averaging** is required to make the transition from the sound pressure to the sound pressure level (SPL). **Fig. 11.A1** clarifies this process, first via a decaying sine tone.

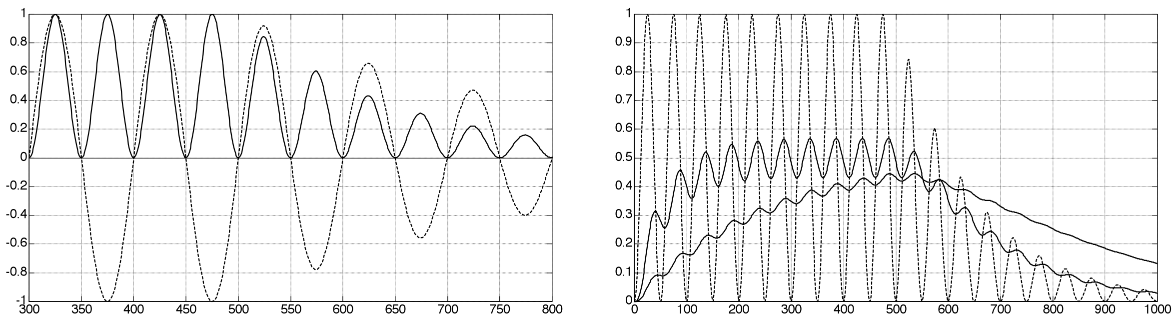

Fig. 11.A1: RMS-averaging. Left: exponentially decaying sine-tone (----), the square of this result (**——**). Right: exponentially averaged decay process; two different time-constants. Not yet subjected to the square root.

To get to the RMS (Root-Mean-Square) value, the signal needs to be squared (S) in a first step, subsequently averaged (M = mean), and last the square root (R) needs to be applied. While the squaring is an unambiguous step, taking the average is not. In metrology, two averaging methods are predominantly used: the so-called exponential averaging, and the so-called linear averaging – the latter should more appropriately be termed arithmetical averaging. Averaging devices in the above sense are linear low-pass filters that are described unambiguously by their impulse response. To achieve exponential averaging, a straight-forward (1^{st}-order) RC-lowpass is used; its impulse response is a decaying *e*-function. The linear averaging happens in the gap-lowpass that features the unipolar rectangular pulse as its impulse response. Both approaches to averaging may be described by one parameter each: by the time constant τ for the exponential averaging, and by the duration of the rectangular pulse (block length) T. Even for $\tau = T$, the results are not the same.

The general problem of every averaging process becomes clear from Fig. 11.A1: with too short a time constant, the smoothing is insufficient, and with a long time constant, the decay-process representation holds errors. **Fig. 11.A2** shows corresponding level-graphs: it is evident that for the exponential averaging, the slope of the flanks depends on the time constant. With the linear averaging, the decaying slope is merely delayed but its slope remains. For determining the reverberation time, this implies that linear averaging has advantages. It still is not without problems: in particular during the **early decay** phase, the slope is too shallow – this could lead to the calculation of too long a reverberation time. This early decay range is important, since real reverberation curves do not decay as ideally as given in this example but show a degressive decay (i.e. an increasingly shallow curve).

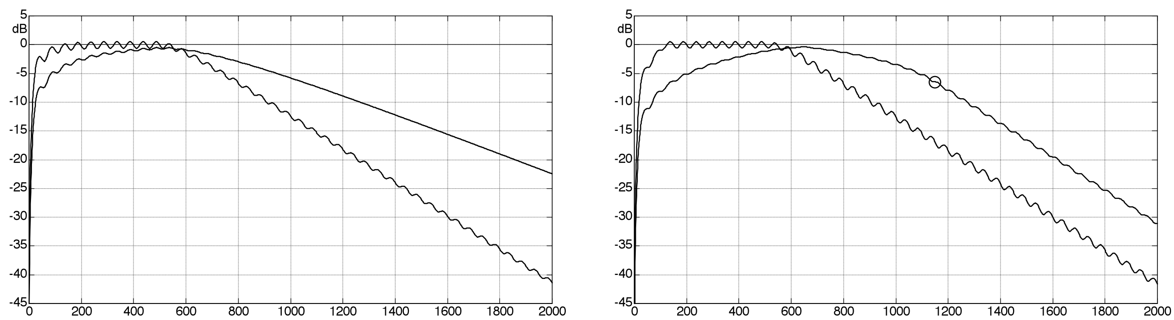

Fig. 11.A2: Level-decay for exponential averaging (left, two different time-constants), and for linear averaging (right, two different time-constants).

From a systems-theory point-of-view, averaging is filtering: the signal to be averaged is convolved with the impulse response of the averager. In other words: every averaging is a weighted (convolution-) integral over a *range*. For the linear averaging (= block-averaging or arithmetic averaging), this integration happens over the block-length T ahead of the center-point (in time) of the averaging. For T = 0.4 s, the linear average indicated at the point in time of t = 2 s specifies the integral over 1.6 … 2 s. Therefore, the linear average value measured at the time when the sound source is switched off (t = 0) does not constitute an averaging over the decay process. Staying with T = 0.4 s: 50 % of the linear average measured at t = 0.2 s is determined from the steady-state process, and the remaining 50% are captured via the decay process. Only the linear average measured at t = T captures 100% of the decay process♣. It is exactly from that point on that the slope of the level-decay is correctly shown when linear averaging is used (Fig. 11.A2, right). Every averaging (over time) needs to happen over a *range* (as elaborated above). If the duration of this range is set too short, the averaging cannot serve its purpose: the convolution with a Dirac pulse results in the unchanged signal. Since the averaging needs to happen over a range (the duration of which needs to be larger than zero), all averages arrive delayed after the signal to be averaged.

If the decay process were an exponentially decaying sinusoidal oscillation as presented in the first figure, and if the travel time in the averager were known, the slope of the process could be precisely determined. However, already an envelope composed of two e-functions (see **Fig. 11.A3**) renders determining the slope problematic: if the slope changes significantly within the block-length T, a linear averaging will not be adequate to reliably detect this.

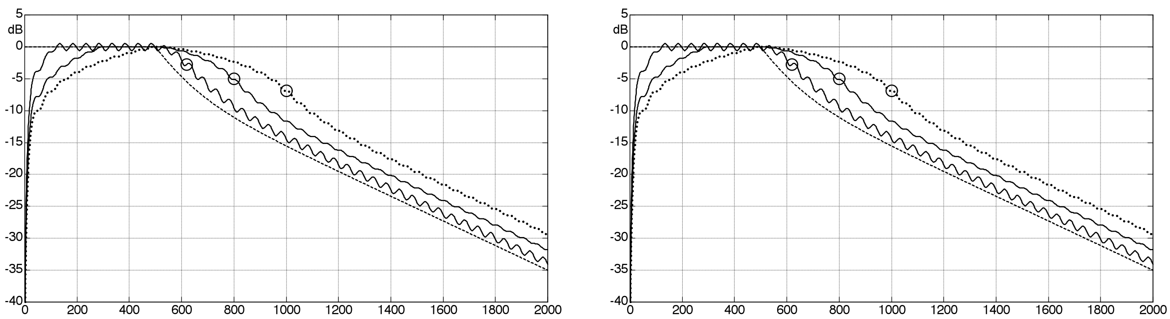

Fig. 11.A3: Level decay for exponential averaging (left, three different time-constants) and for linear averaging (right, three different block-lengths). Degressive decay-curve.

As consequence, the averaging time T, or the time-constant τ, respectively, needs to be short enough not to too much distort the shape of the space-impulse-response (or the space-step-response, respectively), but on the other hand it needs to be long enough to average out the stochastic fluctuations of the noise. This is because, contrary to the figures presented so far, decay curves of reverb are not determined using sine-tones, but with noise (of a bandwidth of an octave or of 1/3rd of an octave). If a linear averaging over e.g. 50 ms is not good enough to average out the noise sufficiently, and if a longer averaging time would prohibit measuring the early decay of the curved decay curve, then a further dimension remains as a way out: the **ensemble-averaging** across different realizations of the noise process. This simply entails averaging over several decay curves – however not with identical noise-excitation but using different excerpts from a noise signal (e.g. 1/3rd-octave noise).

♣ Strictly speaking, the travel time of the sound needs to be considered, as well.

In **Fig. 11.A4** we see, in the left hand section, 4 decay curves that were determined from the squared SPL-signal via linear averaging over a block-length of 50 ms. For this measurement, microphone and loudspeaker were in fixed locations such that the signal fluctuations are to be attributed predominantly to the stochastic of the noise. In the right-hand graph, a multitude of such curves is included – as is the averaging curve derived from them. For orientation, the dashed line represents the decay for a reverberation time of 1.6 s. We can see that the latter approximately corresponds to the early decay, while the remaining curve is shallower. The fluctuations that still remain in the decay curve are not primarily associated with the noise stochastic with the room. The superposition of many decay processes (with many frequencies and different dampening) does not result in a single decay-time-constant; rather we get a curve of any arbitrary complexity that can normally only be approximated to a straight line in sections. For power measurements in the reverberation chamber, it is not the level range between -5 and -36 dB that is to be captured, but rather the initial slope♣.

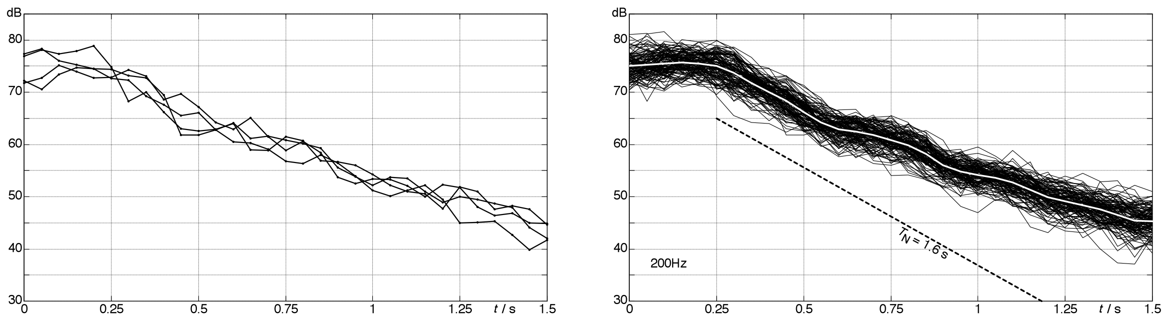

Fig. 11.A4: Decay curves; composed from 1/3rd-octave noise (f_m = 200 Hz) via linear averaging (50 ms). In the graph on the right, the white line represents the mean of the ensemble.

In conclusion a short comment regarding the **Hilbert transform**, since it occasionally is accredited the capability to do ideal averaging. For a decaying sine-tone it is indeed possible to derive, from the sound pressure and using the Hilbert transform, the analytical signal, and from this a smooth decay curve. However, given the narrow-band noise commonly used for reverberation measurements, the Hilbert transform is not an option – at least as long as it alone is put to use (**Fig. 11.A5**).

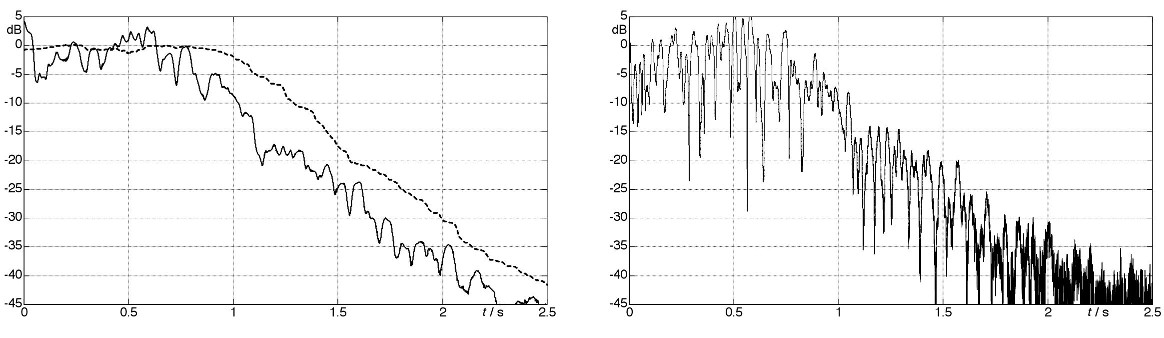

Fig. 11.A5: Left: decay curved filtered with a bandwidth of 1/3rd-octave (f_m = 200 Hz), linear average, T = 50 ms and 500 ms, respectively.
Right: Level of the analytical signal belonging to the signal on the left (also termed "magnitude").

♣ H. Larsen, Technical Review Nr. 4, Brüel&Kjaer, !978.

