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# Uncertainties of Measurements in Room Acoustics

A. Lundeby, T. E. Vigran
NTH Trondheim, Norway

H. Bietz, M. Vorländer
PTB Braunschweig, Germany

*Dedicated to Prof. Dr. Heinrich Kuttruff on the occasion of his 65th birthday*

**Uncertainties of Measurements in Room Acoustics**

**Summary**

The influence of several sources of error on room acoustical measurements have been investigated in a collaboration between the Physikalisch-Technische Bundesanstalt (PTB) in Braunschweig, Germany and the Norges Tekniske Høgskole (NTH) in Trondheim, Norway. The measurement method is the technique using maximum-length sequences (MLS). At first, basic impulse response measurements of a one-third octave band filter and of a loudspeaker using quite different MLS-systems are compared and the differences in the results proved to be negligible. Then algorithms for the determination of room acoustical parameters are introduced and the possible influences on the results are discussed. Finally, intercomparison measurements of room acoustical parameters are described, which were performed in the PTB auditorium by participants from seven institutes all using their own equipment.

The algorithms for the determination of room acoustical parameters used by different analyzers introduce systematic differences caused by differences in time-windowing and filtering, in reverse-time integration and in noise compensation. The overall standard deviations of the room acoustical parameters measured in the PTB auditorium by the different teams are about 5 to 10% for $T_{30}$, *EDT* (Early Decay Time), *D* (Definition) and *TS* (Centre Time), and approximately 0.5 dB for *C* (Clarity) and *G* (Strength), all measured in the 1 and 4 kHz octave bands. Larger uncertainties were found in the 125 Hz octave band, and for the parameter *LF* (Lateral Fraction) in all frequency bands.

The results of this investigation show that if measurements of room acoustical parameters are performed according to ISO/DIS 3382, the overall uncertainty is of the same magnitude or a little higher than subjectively perceivable changes in these parameters. However, the draft standard allows various procedures to be applied in the processing of impulse responses. To reduce the overall uncertainty the standard should, in future revisions, be more specific in this respect. This also applies to the specifications of the omnidirectional source.

**Meßunsicherheiten in der Raumakustik**

**Zusammenfassung**

In Zusammenarbeit zwischen der Physikalisch-Technischen Bundesanstalt (PTB) in Braunschweig und der Norges Tekniske Høgskole (NTH) in Trondheim, Norwegen, wurden verschiedene Fehlerquellen bei raumakustischen Messungen untersucht. Die Meßtechnik basiert auf der Maximalfolgenmethode (MLS). Zunächst werden grundlegende Messungen von Impulsantworten eines Terzfilters und eines Lautsprechers mit unterschiedlichen Meßsystemen verglichen. Die Differenzen in den Ergebnissen stellen sich als vernachlässigbar heraus. Dann werden Algorithmen zur Berechnung raumakustischer Parameter eingeführt und mögliche Einflüsse auf die Ergebnisse untersucht. Schließlich werden Vergleichsmessungen raumakustischer Parameter im Hörsaal der PTB beschrieben, die von 7 Instituten mit jeweils eigenen Meßapparaturen durchgeführt wurden.

Die Algorithmen zur Ermittlung raumakustischer Parameter ziehen systematische Unterschiede durch verschiedenartige Fensterung, Filterung, Rückwärtsintegration und Geräuschkompensation nach sich. Die gesamte Standardabweichung der raumakustischen Parameter im Hörsaal der PTB bei Messungen verschiedener Teams beträgt etwa 5 bis 10% für $T_{30}$, *EDT* (Early Decay Time), *D* (Deutlichkeit) und *TS* (Schwerpunktszeit) und um 0,5 dB für *C* (Klarheitsmaß) und *G* (Stärkemaß) im Frequenzbereich 1 und 4 kHz. Bei 125 Hz und für den Parameter *LF* (Seitenschallgrad) wurden größere Abweichungen gefunden.

Die Ergebnisse dieser Untersuchungen zeigen, daß Messungen raumakustischer Parameter nach der Norm ISO/DIS 3382 mit einer Meßunsicherheit in gleicher Größenordnung oder etwas größer als die subjektiv empfundenen Unterschiede dieser Parameter durchgeführt werden können. Die Norm erlaubt jedoch mehrere Möglichkeiten in der Verarbeitung von Impulsantworten. Um die Gesamt-Meßunsicherheit zu verringern, sollte die Norm in zukünftigen Überarbeitungen genauere Festlegungen enthalten. Dies gilt ebenso für die Spezifikationen ungerichteter Schallquellen.

**Incertitudes de mesures en acoustique des salles**

**Sommaire**

Dans le cadre d'une coopération entre le Physikalisch-Technische Bundesanstalt (PTB) de Brunswick en Allemagne et le Norges Tekniske Høgskole (NTH) de Trondheim en Norvège, on a examiné l'influence de diverses sources d'erreurs sur les mesures acoustiques dans une salle, à l'aide de la technique des séquences maximales (MLS). Tout d'abord, nous avons mesuré les réponses impulsionnelles d'un filtre passe-bande d'un tiers d'octave et d'un haut-parleur, à l'aide de systèmes MLS fort différents, mais la comparaison des résultats ne fait apparaître que des différences négligeables. On a

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Siviling. Anders Lundeby, Prof. Tor Erik Vigran, Norges Tekniske Høgskole, N-7034 Trondheim, Norway.
Dipl.-Ing. Heinrich Bietz, Dr. rer. nat. Michael Vorländer, Physikalisch-Technische Bundesanstalt, Bundesallee 100, D-38116 Braunschweig, Germany.

ensuite introduit des algorithmes de détermination des paramètres acoustiques d'une salle en vue d'en évaluer l'influence possible sur les résultats. Enfin, des mesures comparatives des paramètres acoustiques ont été faites, dans l'auditorium du PTB, par des coopérants de sept instituts qui chacun utilisait son propre équipement.

L'utilisation d'algorithmes différents, pour la détermination des paramètres acoustiques, introduit des différences systématiques dues à des différences de troncature temporelle et de filtrage dans l'intégration temporelle inversée et dans la compensation de bruit. Les écarts types globaux des mesures faites sur les paramètres acoustiques de l'auditorium du PTB par les différentes équipes sont de l'ordre de 5 à 10% pour les paramètres $T_{30}$, EDT (temps de décroissance initial), D (définition) et TS (temps central), et d'à peu près 0,5 dB pour C (clarté) et G (puissance), ces grandeurs étant mesurées en bandes d'octave à 1 et 4 kHz. Les incertitudes sont plus grandes dans la bande d'octave de 125 Hz, et pour le paramètre LF (fraction latérale) elles le sont dans toutes les bandes de fréquences. Les résultats de cette étude montrent que, si l'on procède à des mesures acoustiques selon la norme ISO/DIS 3382, l'incertitude globale est de l'ordre de grandeur des variations perceptibles des paramètres, ou légèrement supérieure. Le norme permet certes l'application de diverses procédures de traitement des réponses impulsionnelles, mais dans les futures version il serait bon qu'elle soit plus précise sur ce point si l'on veut réduire l'incertitude globale. Ceci est vrai également pour les spécifications de la source omnidirectionnelle.

## 1. Introduction

In recent years, sophisticated digital techniques for measuring room impulse responses have been developed. Some systems are available commercially, and particularly the maximum-length sequence (MLS) technique or time delay spectrometry (TDS) have been widely used and are well known. Repeatability, reliability in use and thus the accuracy of the MLS systems can be very high. The user, however, must be skilled in performing the measurements because the technique is based on certain important assumptions such as linearity and time invariance. Furthermore, the parameters of the MLS system such as sampling rate and sequence length must be chosen very carefully to avoid systematic errors. These errors can be partly avoided by automatic checks of the validity of the results. This also applies to other techniques involving chirps or band-filtered noise bursts as excitation signals. For further information on the MLS measuring technique the literature [1 – 5] is recommended.

This paper is mainly focussed on the uncertainties in the application of digital measuring techniques in room acoustics. Special interest is taken in the data processing of impulse responses. The basic elements of the room-acoustical measuring equipment such as loudspeakers and filters, the methods for obtaining room acoustical parameters, and the influence of the operator are checked in order to determine the overall reproducibility of the results.

In the following chapter general aspects of MLS systems are briefly summarized. Two different systems were compared, at first, on simple tasks, to investigate possible differences in the results depending on basic equipment and impulse response calculations. Then, new approaches for standardized measurements are discussed. The influences of the evaluation methods and characteristics of the sound source and receiver have multiple dimensions. They are treated in section 3 for measurements of reverberation time and of other room acoustical parameters. Section 4 deals with measurements in a certain room (the PTB auditorium) which was chosen as an example. Eight different equipments were used in this intercomparison by seven independent teams.

## 2. Comparison of PC-based MLS analyzers

The acoustical laboratories of the Physikalisch-Technische Bundesanstalt (PTB) in Braunschweig, Germany and the Norges Tekniske Høgskole (NTH) in Trondheim, Norway joined for an intercomparison of their MLS systems (see Annex A for a detailed description of the equipment used). The studies started with electronic tests of the AD-converters and the electrical signal conditioning.

### 2.1. Comparison of system performance using measurements on filter transfer function and loudspeaker response

A band pass filter (Norsonic type 719) was used as a test object for the first comparison measurements. It has been chosen because it offers a high reproducibility, a good test for precision in the pass-band area and high requirements regarding the dynamic range of the measurement system.

One obvious problem in these comparison measurements was to distinguish between differences in the results related to the measurement systems themselves, and those which were caused by peripheral devices (e.g. amplifiers). The ratios between output voltage and full scale input voltage are different for the two systems. As a consequence, the amplifiers were excluded from the measurement chain, and both systems operated with the same output voltage. The NTH system used a 20 dB attenuator for the input signal (see Annex A), so that both systems worked with the same ratio between input and full-scale voltage.

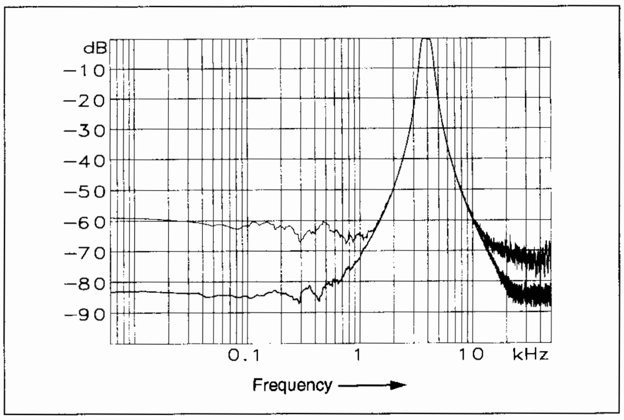


Fig. 1. Transfer function of a one-third octave band filter Norsonic 719 measured at 4 kHz by NTH and PTB (thick line).

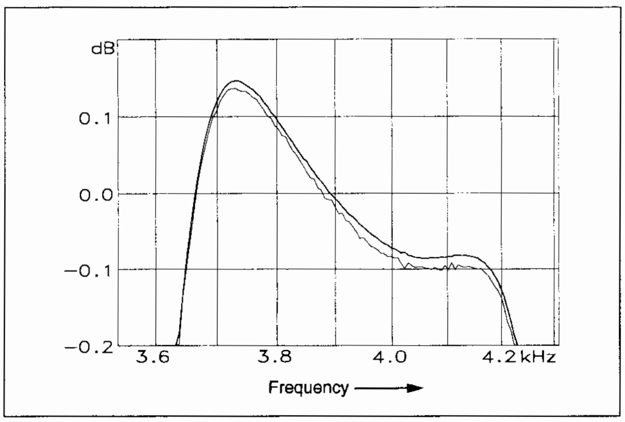


Fig. 2. Zoom plot of passband transfer function of a one-third octave band filter Norsonic 719 measured at 4 kHz by NTH and PTB (thick line).

The results of the measurements of the NE 719 one-third octave band filter at a centre frequency of 4 kHz are shown in Figs. 1 and 2. The most significant aspects are:

1.) In the pass-band area the curves have an offset of only 0.02 dB. Both curves are normalized to the LIN-readout of the filter, so they can be regarded as independent absolute measurements.

2.) In the stop-band areas, the NTH system appears to produce more background noise. It is of interest that the background noise in the frequency range below the pass-band is not reduced by averaging $n$ times by 10 log $n$, but less effectively. This is normally an indication of nonlinearities.

After the measurements of the filter, one could say that the agreement is almost perfect. For a second comparison, the measurement of a loudspeaker frequency response in a small anechoic chamber was performed. The results are shown in Fig. 3. They are identical in the working frequency range of the loudspeaker. Also here,

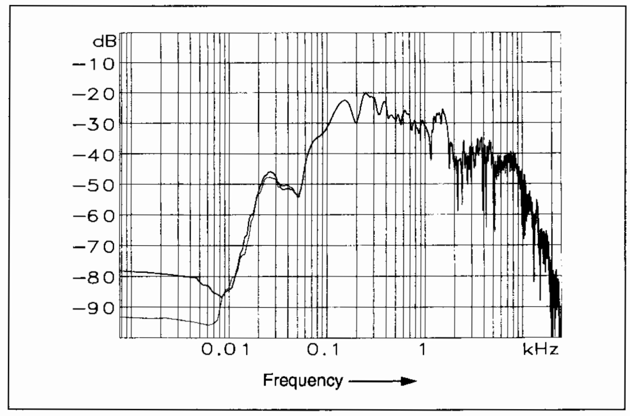


Fig. 3. Free field frequency response of a loudspeaker (example) measured by NTH and PTB (thick line).

the agreement is much better than one could expect. As a conclusion from the first comparisons, it can be said that both measurement systems offer a high level of precision and repeatability for a wide range of practical applications. Only in cases where the requirements on the dynamic range are extremely high, do the PTB systems appear to have some advantages, probably due to the better electrical properties of the A/D conversion board. In view of the great similarity of the systems the study could be focussed on procedures for room acoustical measurements without distinguishing between PTB's and NTH's basic equipment.

## 3. Aspects of measuring room acoustical parameters

Impulse measurements in room acoustics have been state of the art for many years. Sufficient comparability of measurements in different halls, however, requires well-defined parameters and measurement conditions [6]. These conditions are now prescribed in ISO/DIS 3382 from 1994 [7]. The parameters investigated in this study are listed in Table I.

### 3.1. Reverberation time

The basic measurement results from MLS systems are impulse responses. All other quantities or functions must be derived from the impulse responses by subsequent transformations. The elementary transformation to obtain reverberation times is the backward (reverse-time) integration of the impulse response. The reverse-time integrated impulse response is *the* decay curve. The decay curve in the room at the source and receiver points under consideration is unique. It is identical with the result from an infinite number of decay curve averages (ensemble average) using interrupted random noise [8]. The first non-trivial problem to be solved is the calculation of the

Table I. Parameters for description of the acoustical quality in rooms (ISO/DIS 3382).

| Parameter | Definition |
|---|---|
| $T_{30}$/s | Reverberation time, derived from $-5$ to $-35$ dB of the decay curve |
| *EDT*/s | Early decay time, derived from 0 to $-10$ dB of the decay curve |
| *D*/% | Deutlichkeit (definition), early (0–50 ms) to total energy ratio |
| *C*/dB | Clarity, early (0–80 ms) to late 80 ms–∞) energy ratio |
| *TS*/ms | Centre time, time of 1. moment of the energy impulse response |
| *G*/dB | Sound level related to omnidirectional free-field radiation over 10 m distance |
| *LF*/% | Early lateral (5–80 ms) energy ratio, $\cos^2$ (lateral angle) |
| *LFC*/% | Early lateral (5–80 ms) energy ratio, cos (lateral angle) |

reverse-time integrated impulse response. Measured impulse responses are noise-contaminated and must be truncated in a reasonable way [9, 10]. In the following chapter a new algorithm is described which allows the impulse response to be processed automatically. It is presumed that the impulse response is filtered prior to the integration. The influence of the filter is discussed below (see section 3.2.).

### 3.1.1. Estimation of decay slope and background noise

The algorithm described here is based on the use of a short-time averaged impulse response to estimate both the late energy (background noise) and the decay constant of the impulse reponse. The optimum length of the local time intervals will depend on the frequency. Starting with a time-interval in the range of 10–50 ms proved to be a reasonable compromise (see Fig. 4). Larger intervals are related to low frequency bands.

The interval length should be dependent on the slope and we are therefore using this estimate of the slope to choose new averaging time intervals, 3–10 for each 10 dB of decay. The impulse response must then be re-arranged in these new local time intervals. The final estimation of the slope and the background noise level is the result of an iteration which becomes stable after 3 to 5 steps (see Fig. 5). The complete procedure is described in detail in Annex B.

### 3.1.2. Calculation of rest energy assuming exponential decay

Truncation of the impulse response before reverse-time integration leads to a systematic error if the energy from the truncation point to infinity is lost. However, we can "invent" a curve based on extrapolation of the regression

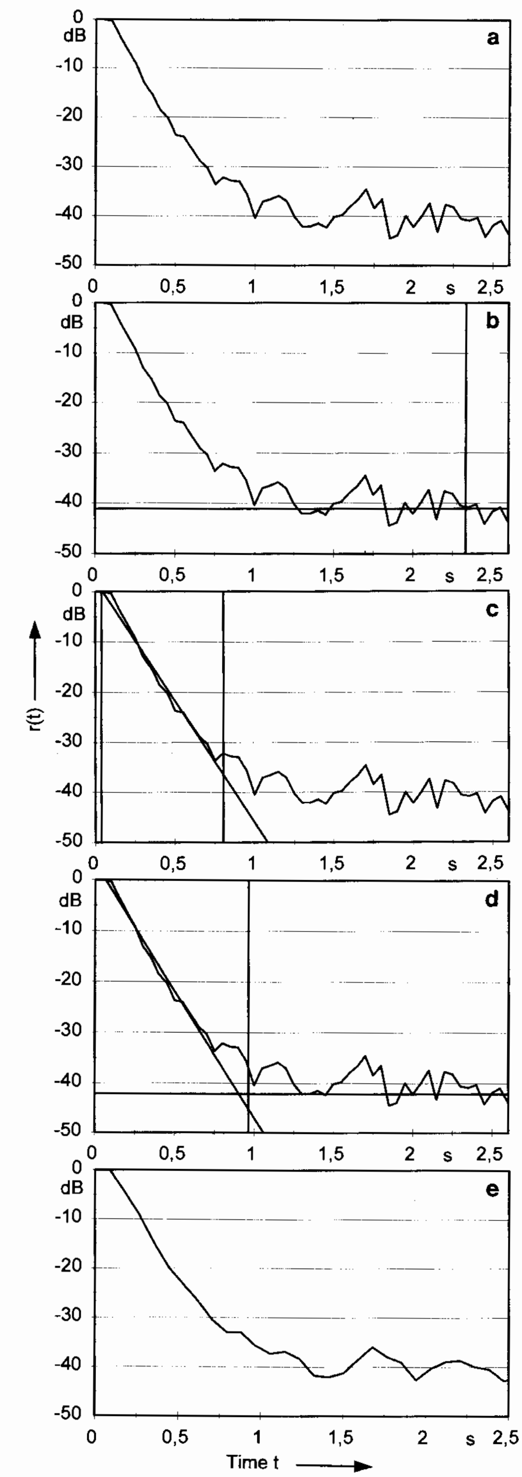


Fig. 4. Iterations in processing impulse responses. a) Average in local time intervals, b) estimate noise level, c) estimate decay slope, d) find preliminary crosspoint, and e) average in new local time intervals.

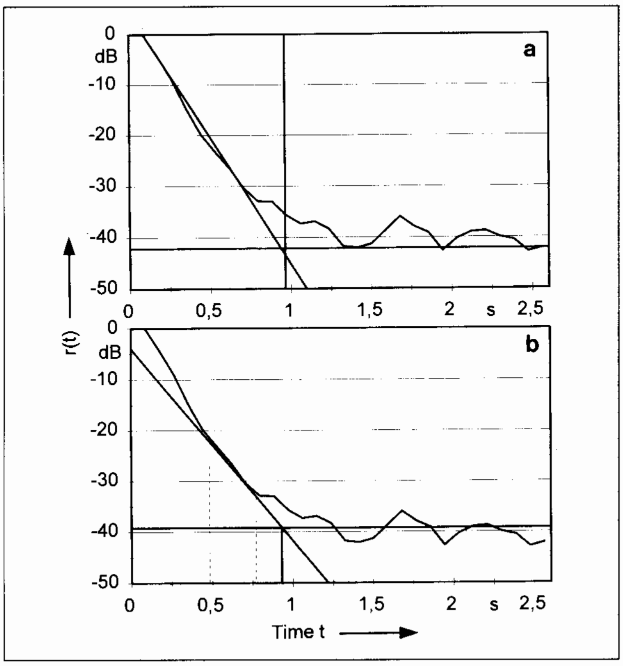


Fig. 5. Iterations in processing impulse responses. a) Preliminary slope and background noise level and b) "final" slope and background noise level after 5 iterations.

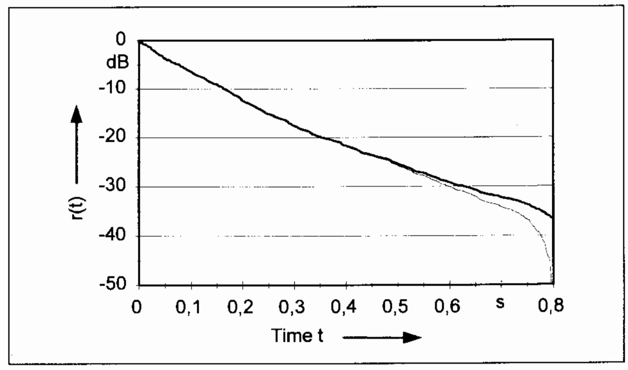


Fig. 6. Integrated impulse response with (thick line) and without noise compensation.

line, and then calculate the total energy from the truncation point to infinity. This is subsequently called the total compensation energy $E_{comp}$. It is indeed essential that the regression line corresponds to the late decay slope near the truncation point. Otherwise, in non-straight decays errors would be made in the extrapolation.

Having the coefficients of the regression line, we can find an expression for the assumed exponential decay. A simple way to do this is using the equivalent energies at two points on the time axis, the zero point and the point where the regression line meets the noise level, $t_1$, eventually compensated with a safety margin. The energy density at the zero point is simply the coefficient $B$ of the regression line, and the energy density at $t_1$ is known as the mean noise energy density, $N$ (in this case, energy density means not spatial density but "temporal density" which is calculated by integrating over short time intervals). Care must be taken to ensure correct normalisation of block energies into sample energies.

The expression for the assumed exponential decay of the energy density can be stated as:

$$r(t) = B\,e^{At}. \tag{1}$$

The constant $B$ is already known, and $A$ is given by:

$$A = \frac{\ln(N/B)}{t_1}. \tag{2}$$

Since $A$ is always a negative value, the total compensation energy $E_{comp}$ can be found:

$$E_{comp} = B\int_{t_1}^{\infty} e^{At}\,dt = -\frac{B}{A}c^{At_1}. \tag{3}$$

Fig. 6 shows an integrated impulse response with and without correct truncation and compensation. The signal to noise ratio of the smoothed impulse response is equal to a little less than 40 dB. The truncation point lies at − 36 dB. Without noise compensation $T_{30}$ is underestimated by about 8%. The signal to noise ratio needed for the evaluation of $T_{30}$ is 35 dB plus a few decibels safety margin. Without noise compensation a minimum signal to noise ratio of 45 dB would be necessary for the evaluation of $T_{30}$ [10].

### 3.2. Filtering, time-windowing and integration for the calculation of energy ratios

Some of the parameters defined in ISO/DIS 3382, Annex A, require time-windowed and octave-band filtered impulse responses. The integrations needed for the calculation of $D$, for example, must cover the range from zero to infinity and from zero to 50 ms ([7]). When applying the MLS technique, the excitation and the whole measurement procedure is mostly of a broad-band type. Filters are normally introduced at a later phase of the signal processing in the analyzer. This causes substantial differences compared with measurements with band-filtered noise bursts, where intermediate broad-band results are not available.

The filter delays create problems in the determination of the start of the impulse response and the window length. The question is therefore when to apply the integration intervals, before or after filtering. This aspect is particularly important in the low freuency range. It is normally considered the best approach to cut the broadband impulse response before any filtering. The problems concerning filter delays are then avoided.

In the same manner as for reverberation times, the parameters *D*, *C*, *TS* and *G* (see Table I) are based on impulse responses measured using microphones with omnidirectional sensitivity. For their determination, at first, the time axis has to be re-defined. The zero point is no longer the time when the (imaginary) excitation impulse leaves the source, but that time when the direct sound arrives at the receiver.

An example from a reverberation room measurement shows that time-windowing before filtering can lead to unexpectedly large energy in the late part of the impulse response. Here, the late energy is calculated with starting limits ranging from 0 to 60 ms, integrated to infinity and plotted as a function of (left) time limit in Fig. 7. The impulse responses filtered with and without time-windowing are shown in Fig. 8. For the time-windowed and filtered response, presumably a suppressing reflection in the broad-band response is removed. Hence, the total energy of the time-windowed impulse response is, surprisingly, larger than the energy without time windowing.

Normally, one would expect a monotonically decaying curve, which is the case when the time windowing is performed after the filtering. The deviation from monotonicity can be explained as destructive interference in the complete filtered impulse response, with one or more of the interfering reflections removed by time-windowing before filtering. The fact that wall reflections interfere coherently is normally not a feature of broad-band measurements in rooms. The coherence is introduced by filter responses. They last a few periods of the midband frequency. These effects change reflection peaks into wavelets of a certain length. If the temporal length is greater than the delay between two reflections, the overall energy sum may increase or decrease, depending on the phase relation between the reflections.

The results obtained in a reverberation room using 100 Hz one-third octave band filtering are surely not representative of normal room acoustical parameter measurements, but should be taken into account when comparing different algorithms. This shows that time-windowing before filtering may *not* be the method which gives the best inter-laboratory agreement. The interference effect cannot be totally ignored. The approximation using a "window correction" [11] when cutting the filtered impulse response might be a better approach for standardization. For this correction the impulse responses of the filter must be known. Its half-length is then taken into account in finding the points needed for the integration of early and late energy (see [7]).

### 3.3. Influence of the loudspeaker

The directivity of the loudspeaker must be as uniform as possible [12]. In Table A 1 of the draft standard ISO/DIS 3382 maximum deviations from uniformity of up to

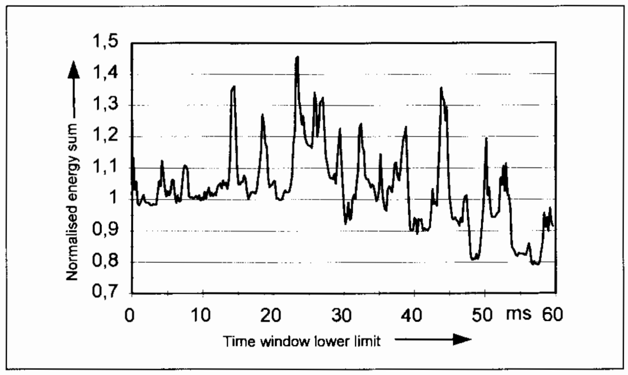


Fig. 7. Late energy from $t_1$ to infinity as a function of $t_1$, normalized to the integral from 0 to infinity.

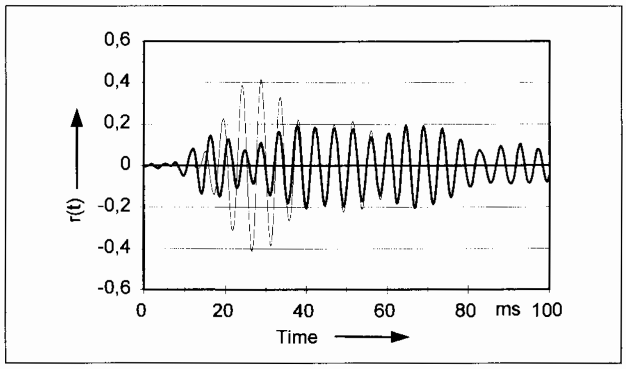


Fig. 8. Early part of an impulse response measured in a reverberation chamber in 100 Hz one-third octave band. Curve 1 (thick line) filtered, curve 2 first time-windowed (arbitrarily at 23.5 ms) and then filtered.

$\pm$ 6 dB in the 4 kHz octave band are allowed. This means that a 4 kHz single reflections in the squared impulse response may vary by a factor of four, depending on the orientation of the source. This variation is much too large for a satisfying reproducibility of details of squared impulse responses. The single number parameters, however, might not necessarily be affected as much by the directivity. To check this particular influence, two types of sources were used, having deviations from omni-directionality as listed in Table II. The influence of the sound source on the room acoustical parameters was investigated as part of the measurements in the room which served as an example (see next section).

## 4. Overall uncertainty and tests in PTB's auditorium

As the example of a room a speech auditorium, located on the premises of PTB in Braunschweig, was chosen because it was easy to access for this study. The hall has a seating capacity of 274, a volume of about 1800 m$^3$, an approximately rectangular ground area of $22 \times 14$ m$^2$, and a seating area inclined by 12° (see photograph in Fig. 9). The average reverberation time is about 1.1 to 1.2 s.

As mentioned above, the particular influence of the loudspeaker directivity was investigated on two "omnidirectional" sources with regular spherical symmetry (cube and dodecahedron). Quantitative relations between loudspeaker directivity and standard deviations of parameters are checked by discussing the results from repeated measurements with the sources turned in steps of approximately 20° (see Table II). PTB's measuring equipment was used for this test. The variation of the parameters is investigated in the low, mid and high frequency ranges at 125 Hz, 1 kHz and 4 kHz, respectively. The standard deviations of repeated measurements with turned sources shown in Fig. 10 are surely not representative, but they clearly indicate the influence of the directivity patterns with increasing frequency. However, the standard deviations at 4 kHz of parameters *C* and *TS* are still tolerable, compared with subjectively perceivable changes [13]. If a significantly smaller loudspeaker than the commercial dodecahedron type were to be used, the influence of its directivity could be reduced, particularly at 1 kHz (see Table II). The low power output of small loudspeakers can be easily compensated by synchronous MLS-averaging.

Table II. Maximum deviations from omni-directionality for octave bands of pink noise.

| Frequency Hz | Tolerances in ISO/DIS 3382 dB | Source 1 (dodecahedron, 45 cm ∅) dB | Source 2 (cube, 16 cm edge) dB |
|---|---|---|---|
| 125 | ±1 | 0 | 0 |
| 250 | ±1 | 0 | 0 |
| 500 | ±1 | 0 | 0 |
| 1000 | ±3 | 1.5 | 0 |
| 2000 | ±5 | 4.0 | 3.0 |
| 4000 | ±6 | 2.5 | 3.5 |

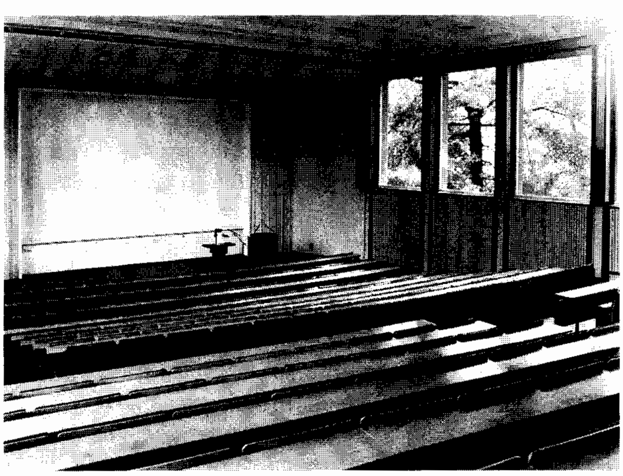

Fig. 9. Auditorium at PTB in Braunschweig.

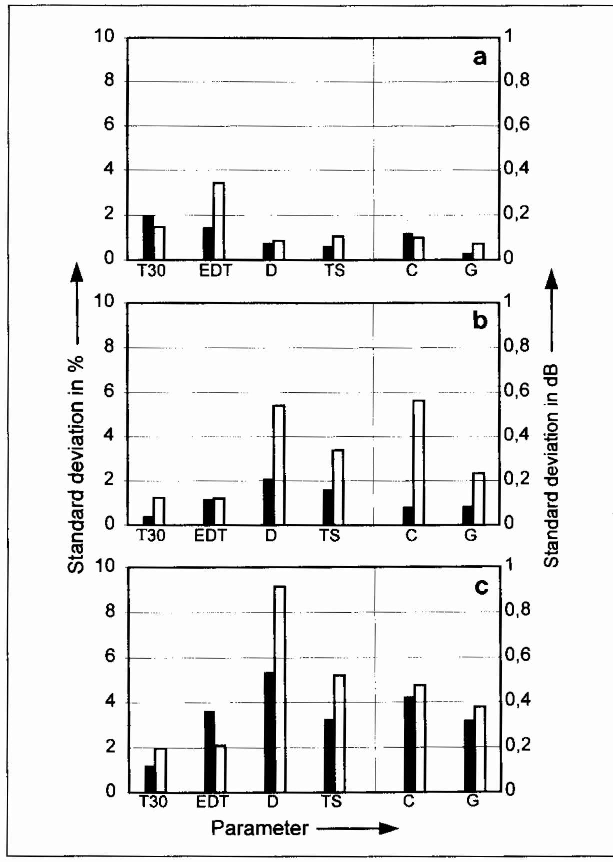


Fig. 10. Standard deviations of measurements with two different sources rotated stepwise by 20°. a) 125 Hz, b) 1 kHz and c) 4 kHz octave bands.

To investigate the overall reproducibility, together with the PTB and NTH, colleagues from RWTH Aachen, IEMB Berlin, HEAD acoustics Herzogenrath (all in Germany), Chalmers University, Gothenburg, Sweden, and University of Parma, Italy, participated in measurements in order to obtain reference data for an international "round robin" on computer simulations. However the results were not only used for the above-mentioned purpose, but were also reviewed concerning the reproducibility of measurement results according to ISO/DIS 3382. Of particular interest was the question, as to whether the error influences discussed in section 3 were also found when different measurement teams used different loudspeakers and hard- and software. Table III contains the specifications of the different equipment in summarized form.

Two source positions and five receiver positions were chosen. The resulting ten combinations were named S1R1, S1R2, ..., S2R5. The measurement results were compared in octave bands of 125 Hz, 1 kHz and 4 kHz.

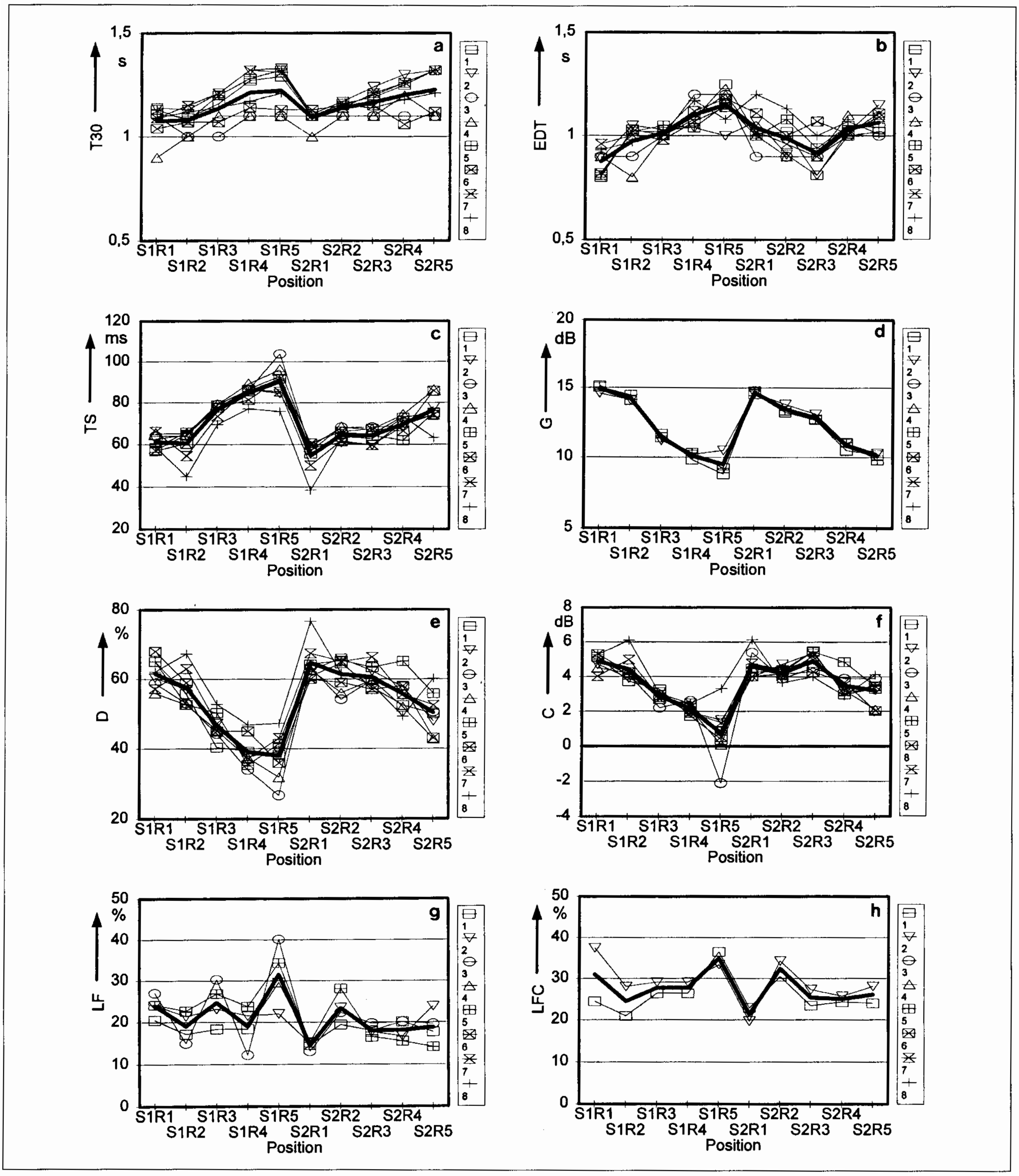


Fig. 11. Individual results and average (thick line) of parameters for the 1 kHz octave band measured by different teams using different equipment (see Table III).

Table III. Specifications of the equipment used by the participants.

| | Signal type, loudspeaker (size), hardware, resolution, (software) |
|---|---|
| PTB | MLS, cube (16 cm edge), Data Translation board, 16 bit, (MEGAFFT) |
| ITA | MLS, dodecahedron (20 cm ∅), self-made board, 12 bit, (MEGAFFT) |
| NTH | MLS, dodecahedron (45 cm ∅), Loughborough board, 16 bit, (MATLAB) |
| IEMB | Noise burst, "pulsating sphere", MUSYCS board, 16 bit, (MUSYCS) |
| IEMB | MLS, "pulsating sphere", MLSSA board, 12 bit, (MLSSA) |
| HEAD | MLS, dodecahedron (40 cm ∅), HEAD DSP5+ EBU board, 16 bit, (RAS) |
| UNIPA | MLS, dodecahedron (45 cm ∅), MLSSA board, 12 bit, (MLSSA) |
| CHA | MLS, single speaker box (volume 1 l), MLSSA board, 12 bit, (MLSSA) |

In Fig. 11 the individual results for the 1 kHz octave band are shown. The average standard deviation in the results is shown in Fig. 12. These data were calculated from the inter-laboratory standard deviations at the individual positions. To obtain a single figure expressing the average overall uncertainty, the ten "local" standard deviations were averaged.

The standard deviation in the low frequency range is caused by filtering and different methods for integrating the early and late parts. All parameters are affected by this influence. This uncertainty can be as large as 25% in the case of parameter $D$, or 1 dB for parameter $C$. At mid and high frequencies the situation changes and the magnitude of the average standard deviation of the parameters indeed is rather small even at 4 kHz, although for some locations of source and receiver it is significantly higher, particularly at positions with large distances between the source and receiver. The standard deviation of the parameter $LF$ is about 20% in the 1 kHz octave band. This number can only, however, serve as rough estimate because only four measurements were made using figure-of-eight microphones. This large uncertainty in $LF$ is perhaps caused by inadequate microphone performance, and also microphone orientation during the measurement. Indeed, in the determination of the $LF$, all the possible error influences coincide: the problem of time-windowing and filtering, the directivity of the loudspeaker plus the calibration and orientation of the microphone(s).

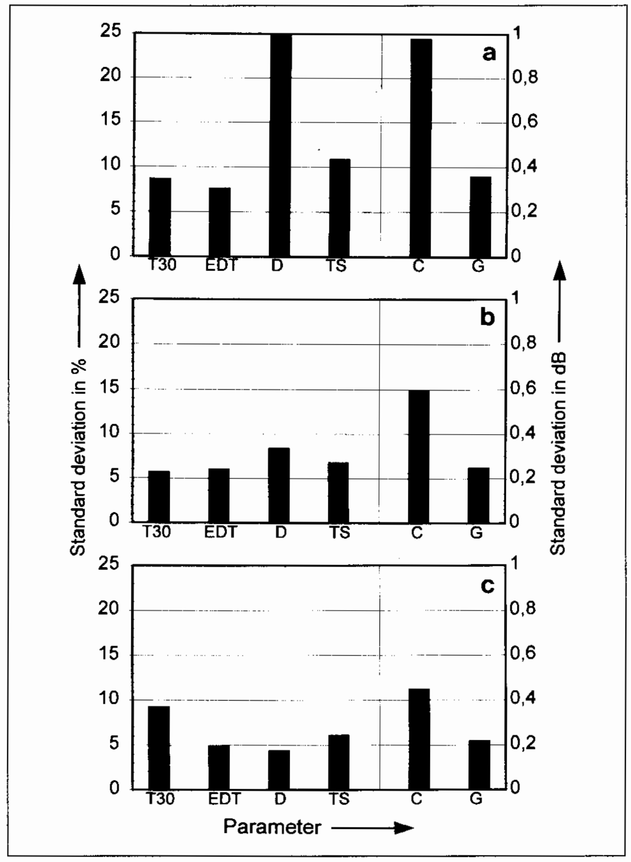


Fig. 12. Mean relative standard deviation in % of parameters measured by seven teams from different laboratories (for parameters $C$ and $G$ the mean absolute standard deviation in dB is shown). a) 125 Hz, b) 1 kHz and c) 4 kHz octave bands.

## 5. Conclusions

In this study the influence of several errors on room acoustical measurements were investigated. The basic measurement method was the technique using MLS-sequences. Basic impulse response measurements using two quite different MLS-systems (see Annex A) were compared and the differences in the results proved to be negligible. Then algorithms for the determination of room acoustical parameters were introduced and the possible influences on the results were discussed. Finally, intercomparison measurements between seven institutes were performed in an auditorium.

The main results can be summarized as follows:

1.) The basic measurements using MLS-sequences were performed very successfully. The broad-band impulse responses are highly reproducible, regardless of being measured by different teams and equipment.

2.) The algorithms for the determination of room acoustical parameters introduce systematic differences caused by differences in:

- time-windowing and filtering,
- reverse-time integration, and
- noise compensation.

It is recommended to restrict the method of filtering and time-windowing in the ISO standard 3382 to only one method.

3.) Reverse-time integration can be performed automatically if the truncation point of the impulse response lies at least 10 dB below the (right) interval limit of the evaluation range of $T$ (45 dB for $T_{30}$). With noise compensation this margin can be reduced to about 5 dB or less (less than 40 dB for $T_{30}$).

4.) It is essential that the noise compensation is based on estimates of that part of the impulse response near to the background noise. If the decay slope is derived from the total impulse response (above background noise), large underestimates of the compensation energy occurred even in only slightly curved decays.

5.) The influence of the loudspeaker directivity on the room acoustical parameters is significant at 1 and 4 kHz. It should be recommended to average the results with the sources turned at least three times, if the individual results differ more than 5% or 0.5 dB.

6.) The overall standard deviation of the room acoustical parameters measured in PTB's auditorium are about 5 to 10% for $T_{30}$, *EDT*, *D* and *TS* and approximately 0.5 dB for *C* and *G*, all measured in frequency bands 1 and 4 kHz. Larger uncertainties were found in the 125 Hz octave band, and for the parameter *LF* in all frequency bands.

The results of this investigation clearly show that a considerable step towards standardization of room acoustical measurements has been made. If measurements are performed according to ISO/DIS 3382, the overall uncertainty is of the same magnitude or a little higher than subjectively perceivable changes in the room acoustical parameters. The standard, however, allows much freedom in processing impulse responses, and should be more specific in future revisions. This also applies to the specifications of the omnidirectional source.

This study was not focussed on measurements of the interaural cross-correlation coefficient (IACC). The reason is that IACC was not included in the initial research programme on room acoustical computer simulations for which the measurements served as a reference. It is surely necessary to study uncertainties in IACC measurements in a separate investigation since the problems involved there are more diverse, mainly due to the comparability of different dummy heads. The same holds for the usage of dummy heads as sound sources simulating human speakers.

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**Annex A – Description of equipment**

PTB and NTH use MLS systems which are very similar in their principal arrangement. Each system consists of a PC 486 fitted with an AD/DA conversion board and an external junction box. The main difference is in the approaches to the software which is used for measurement control and data processing.

The NTH dual channel system includes a Loughborough DSP96002 board with Motorola AD/DA-converters of 16 bit resolution, maximum sampling rate of 100 kHz per channel and a full-scale voltage range of $\pm 1$ V for both A to D and D to A conversion. The junction box includes a 20 dB attenuator which allows an expansion of the voltage range up to $\pm 10$ V. Also included is a filter board from R.C. Electronics Inc. with programmable analogue filters intended to work as anti-aliasing filters. The software employed was developed at NTH under the MATLAB environment, for data acquisition as well as for further data processing. The MLS sequences are generated by software and sent through the D to A section of the board. Sequences up to order 17 (131.071 samples) can be used. With a 50 kHz sampling rate this offers time periods of 2.6 s which is generally enough for measurements in concert halls or auditoria.

The system used at PTB includes a Data Translation DT 2823 board which offers 16 bit resolution and 100 kHz maximum sampling rate for the A/D-converter. The sample and hold amplifier can be configured with a maximum of 4 inputs in time multiplex mode. The voltage range is $\pm 10$ V for both input and output. An external junction box, which also contains an anti-aliasing filter, is used. Data acquisition and data processing is performed by software developed in Pascal and Assembler (MEGAFFT), mainly at the Institute for Technical Acoustics of the RWTH Aachen and partly at PTB. The software has been developed for MLS and FFT measuring techniques and offers a large number of possibilities concerning data acquisition and MLS sequence generation including complex pre-emphasizing [13]. The sequence length is only dependent on the RAM size. Having 32 Mbyte RAM installed, sequences up to order 20 (1.048.575 samples) can be used. With 50 kHz sampling rate, time periods of 21 s can be reached, which is enough for measurements in all kinds of rooms including, for example, reverberation chambers or churches.

To make any comparisons of measurements possible, it was first necessary to provide data exchange between the two systems. This was realized with MATLAB import and export routines.

Concerning measurement of room acoustical parameters, the equipment must be calibrated particularly for measuring the parameter $G$ (relative sound pressure level or "strength") since $G$ expresses an energy ratio between diffuse field (in the room to be tested) and free field conditions. This was performed by NTH and PTB in a small anechoic chamber. Impulse responses with fixed amplifier settings were obtained at a distance of 2 m in a free sound field. These results were related to those at a distance of 10 m and used as reference for the measurements in the example rooms (see section 4). The calibration of the figure-of-eight microphones used for obtaining lateral measures was performed in a similar way.

## Annex B – Algorithm for determination of integration limit and noise compensation of impulse responses

Reverse-time integration of an impulse response should theoretically start at infinity. In practice the integration should be performed from a point where the impulse response meets the background noise level. The determination of this point requires estimates of the late decay slope and background noise level.

The background noise level is considered to be stationary and can be determined from the later part of the response, where noise dominates over the signal. It is desired to use as much of the tail as possible, in order to minimize statistical variations. On the other hand, the true signal should be kept at a safe distance. The slope of the decay curve should be estimated from the later part of the decay. This is particularly important when the decays are curved. On the other hand, one should not go too close to the background noise level.

We can see that the estimation of these two quantities, the background noise level and the decay slope, are mutually dependent. This leads to an iterative algorithm (see Fig. 4 which shows the first six steps of the procedure):

1. Average squared impulse response in local time intervals

The interval length should be 10–50 ms, to give a smooth curve without losing short decays. For low frequency octave bands, long intervals are better.

2. Estimate background noise level using the tail

The last 10% in time of the impulse response should be regarded as noise, giving a reasonable statistical selection and not too large a systematic error if the decay continues to the end of the response.

3. Estimate slope of decay from 0 dB to noise level

The "left" point is set at 0 dB. Search for the "right" point 5–10 dB above noise level. Determine the slope of the regression line between the two points.

4. Find preliminary crosspoint

5. Find new local time interval length

The interval length chosen is dependent on the estimated slope. Use 3–10 intervals per 10 dB decay. For low frequency octave bands, few and long intervals are better.

6. Average squared impulse response in new local time intervals

7. Estimate background noise level

Allow a safety margin from crosspoint corresponding to 5–10 dB decay, but use a minimum of 10% of the impulse response. This is a trade-off between stability and the possibility of integrating the whole response when no noise is significant.

8. Estimate late decay slope

A dynamic range of 10–20 dB should be evaluated, starting 5–10 dB above the noise level. (The evaluation range must be determined using the former estimated slope, giving the corresponding number of intervals).

9. Find crosspoint

10. Repeat step 7, 8 and 9 until convergence of crosspoint is achieved

It is assumed that 5 iterations are sufficient in all cases.

The steps 1–6 are shown in Fig. 4. The result of step 1–6 plus 5 iterations of step 7–10 are shown in Fig. 5. After performing this algorithm, the late decay slope and the truncation point are known for further processing.

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