

# Fundamentals to perform acoustical measurements

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**Appendix to EASERA**

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# 1 Preface

In order to do acoustic measurements, you have to know **how** to measure and also **what** to measure.

The first point will be covered for all acoustic measurement tools by a good tutorial as part of a well-done operation manual. In the following chapters, we will deal more extensively with the second item, i.e. what must be measured to obtain the right answers for questions and problems that are discovered.

Numerous subjective and objective room-acoustical criteria have been defined and their correlation determined in order to objectify these assessments. However, these individual criteria are closely linked with each other and their acoustic effects can neither be exchanged nor individually altered. They become effective for assessment only in their weighted totality. Based on subjective considerations and well-founded, objective measurement, technical examinations and subjective tests, partially in reverberation-free rooms within artificially generated sound fields, it was possible to define room-acoustical quality criteria that enable an optimum listening and acoustical experience in compliance with the usage function of the room. The wider the spectrum of usage, the broader the limit of the desirable reference value ranges of these criteria. Without extensive variable acoustical measures – also electronic ones – only a compromise brings about a somewhat satisfactory solution. It stands to reason that this compromise can only be as good as the degree to which the room-acoustical requirements coincide with it.

A precondition for an optimum room-acoustical design of auditoriums and concert halls is the very early coordination in the planning phase. The basis here is the establishment of the room's primary structure according to its intended use (room shape, volume, topography of the spectators' and the platform areas). The secondary structure that decides the design of the details on walls and ceilings as well as their acoustic effectiveness has to be worked out on this basis. A planning methodology for guaranteeing the room-acoustical functional and quality assurance of first-class concert halls and auditoriums as well as rooms with a complicated primary structure is reflected in the application of simulation tests using mathematical and physical models.

The acoustical evaluation by listeners and actors of the acoustical playback-quality of a signal that is emitted from a natural acoustic source or via electro-acoustical devices is mostly very imprecise. This evaluation is influenced by existing objective causes like disturbing climatic, seating and visibility conditions as well as by subjective circumstances such as the subjective attitude and receptiveness towards the content and the antecedents of the performance.

Numerous room-acoustical criteria have been defined in order to clarify the terms used for the subjective and objective assessment of a spoken or musical performance. In the following we have listed a relevant selection of them, in which context one should note that there is a close correlation between the individual criteria. Also included are previously published overviews /32/. One single optimally determined parameter might not at all be acoustically satisfactory, because another parameter influences the assessment in a negative way.

In principle, the room-acoustical quality criteria can be subdivided into time and energy criteria. The main type of use – speech or music, then determines the recommendations for the guide values to be targeted. With multi-purpose halls (without available variable measures for changing the acoustics), a compromise is required that should orient itself to the main type of use.

In the assessment of the room-acoustic quality with speech presentations (classroom, auditorium, congress hall, sermon church) and with music presentations (concert hall, opera-house) one distinguishes between "overall parameters" and the source - receiver position-related room-acoustic quality criteria for listeners (seats) as well as with music presentations for musicians (seats on the stage) and for the conductor's position.

The "overall parameters" are, among others:

- the reverberation times  $T_{10}$ ,  $T_{20}$  and  $T_{30}$
- the Early Decay Time EDT as well as
- the Bass-Ratio BR

As a rule, the "overall criteria" are measured with an omni-directional microphone.

The position-related assessment of the room-acoustic quality in the case for **speech presentations** happens with the help of quality criteria such as:

- 50-ms-part or Definition D according to THIELE [1]
- Speech Clarity  $C_{50}$  according to AHNERT [3]
- Articulation Loss  $AL_{cons}$  according to PEUTZ [5]
- Speech Transmission Index STI or RASTI according to STEENEKEN and HOUTGAST [4; 5]
- Center time  $t_s$  according to KÜRER [2]
- Echo criterion  $EK_{speech}$  according to DIETSCH [8] for the perception of (annoying) reflections (echo)

For the listener position-related assessment of the room-acoustic quality in the case for **music presentations**, the following objective room-acoustic quality criteria for the listeners have been developed or suggested:

Sound source-related criteria:

- Direct sound measure  $C_7$  for the sensation of the directness and nearness of the sound source [3]
- Clarity measure  $C_{80}$  according to ABDEL ALIM [9] for the transparency of musical structures (time and register clarity)
- Interaural cross correlation coefficient IACC according to ANDO [29; 30] for the apparent sound source width ASW subjectively felt by the listener.
- Strength measure G according to LEHMANN [11] for the sound volume (level) of music presentation felt at the listener's place.

Space-related criteria:

- Reverberance measure R according to BERANEK [16] for the acoustic "liveliness" of the music presentation supported by the reverberation.
- Lateral Efficiency LE according to JORDAN [18] and Lateral Fraction LF according to BARRON [20] for Apparent Source Width (AWS) and for the Envelopment (from the reflected sound) - Listener Envelopment LEV.
- Lateral Fraction Coefficient LFC according to KLEINER [34] for the Envelopment (from the reflected sound) - Listener Envelopment LEV
- Echo criterion  $EK_{\text{music}}$  according to DIETSCH [8] for the perception of (annoying) reflections (echo) in case of music presentations

For the position-related assessment of the room-acoustic quality in case of **music presentations** the following objective room-acoustic quality criteria for the musicians (conductors) have been developed or suggested:

- Early Ensemble Level EEL according to GADE [21; 22] for the room-acoustic support of the team-play (support of orchestrating) of the musicians on the platform (remark: according to recent investigations the measure Support ST1 instead of EEL is favored by the authors (GADE)).
- Support ST1 and ST2 according to GADE [21; 22] for the room-acoustic support of the team-play (support of orchestrating) and the acoustic sensation of the room response on the concert platform and in the orchestra pit.

## 2 Room-acoustical Overall Criteria for Speech and Music Performances

### 2.1. Reverberation Times EDT, $T_{10}$ , $T_{20}$ , $T_{30}$

#### 2.1.1. Measurement Basis

Basis for **Measuring the Early Decay Time and the Reverberation Times** using the computer-aided reverberation time measurements - like e.g. the measurement of the energy criteria - are the room impulse responses (RIR).

The reverberation time RT is not only the oldest, but also the very best known room-acoustical quantity. It is the time that passes after an acoustic source in a room has been turned off until the mean steady-state sound-energy density  $w(t)$  has decreased to 1/1,000,000 of the initial value  $w_0$  or until the sound pressure has decayed to 1/1,000, i.e. by 60 dB:

$$w(RT) = 10^{-6} w_0 \quad (1.1)$$

Thus the time response of the sound energy density in reverberation /37/ results as:

$$w(t) = w_0 \exp\left(-6 \ln 10 \frac{t}{RT}\right) = w_0 \exp\left(-13.82 \frac{t}{RT}\right) \quad (1.2)$$

The steady-state condition is reached only after the starting time  $t_{st}$  of the even sound distribution in a room (approximately 20 sound reflections within 10 ms) /38/:

$$t_{st} = 1 \dots 2 (0.17 \dots 0.34) \sqrt{V} \quad (1.3)$$

$t_{st}$  in ms  
 $V$  in  $m^3$  ( $ft^3$ )

The defined drop of the sound pressure level of 60 dB corresponds roughly to the dynamic range of a large orchestra<sup>39/</sup>. The listener, however, can follow the decay process only until the noise level in the room becomes perceptible. This subjectively assessed parameter *reverberation time duration* thus depends on the excitation level as well as on the noise level.

The required evaluation dynamic range is difficult to achieve even with objective measuring, especially in the low-frequency range. Therefore, the reverberation time is determined by measuring the sound level decay in a range from -5 dB to -35 dB and then defined as  $T_{30dB}$  (also  $T_{30}$ ). The so-called initial reverberation time (IRT,  $T_{15dB}$  between -5 dB and -20 dB) and the early decay time (EDT according to JORDAN, /18/,  $T_{10dB}$  between 0 dB and -10 dB) are more in conformity with the subjective assessment of the duration of reverberation, especially at low-level volumes. This also explains why the reverberation time subjectively perceived in the room may vary, while the values measured objectively according to the classical definition with a dynamic range of 60 dB or 30 dB are, except for permissible fluctuations, generally independent of the location.

Serving as a single indicator for the principal characterization of the room in an occupied or unoccupied state, the reverberation time is used as the mean value of the two octave bandwidths 500 Hz and 1000 Hz or the 4 one-third octave bandwidths 500 Hz, 630 Hz, 800 Hz and 1000 Hz, and referred to as the *mean reverberation time*.

The desired value of the reverberation time RT depends on the kind of performance (speech or music) and the size of the room. For auditoriums and concert halls, the desired values for the mean reverberation time from 500 Hz to 1000 Hz with a room occupation between 80 % and 100 % are given in Fig. 1.1, and the allowed frequency tolerance ranges are shown in Figs. 1.2 and 1.3. This shows that in order to guarantee specific warmth of sound with musical performances, an increase of the reverberation time in the low frequency range is preferred (see section 1.1.2), while with spoken performances a decrease of the reverberation time is desired in this frequency range (see section 1.2.9).

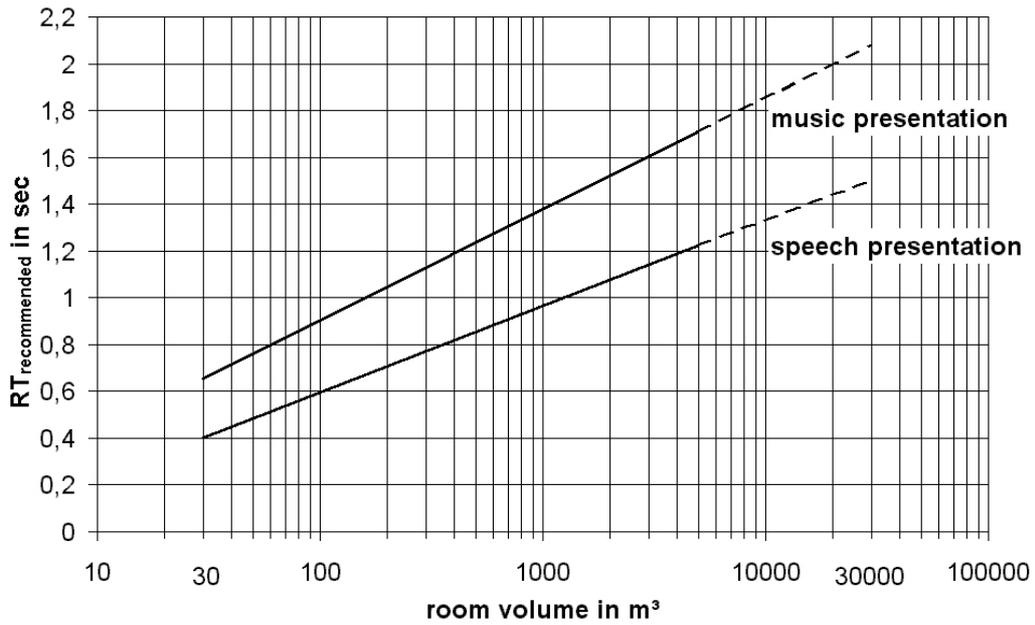


Figure 1.1: Recommended value of the mean reverberation time  $RT_{\text{recommended}}$  from 500 Hz to 1000 Hz for speech and music presentations as a function of room volume  $V$

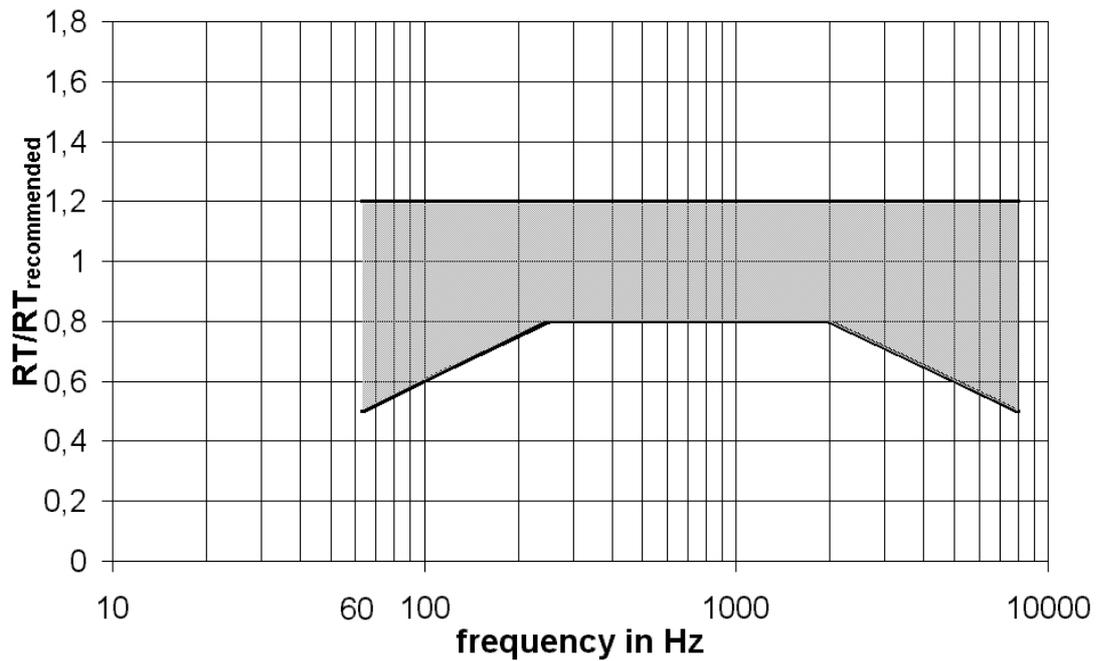


Figure 1.2: Frequency dependent tolerance range of reverberation time  $RT$  referred to  $RT_{\text{recommended}}$  for speech presentations

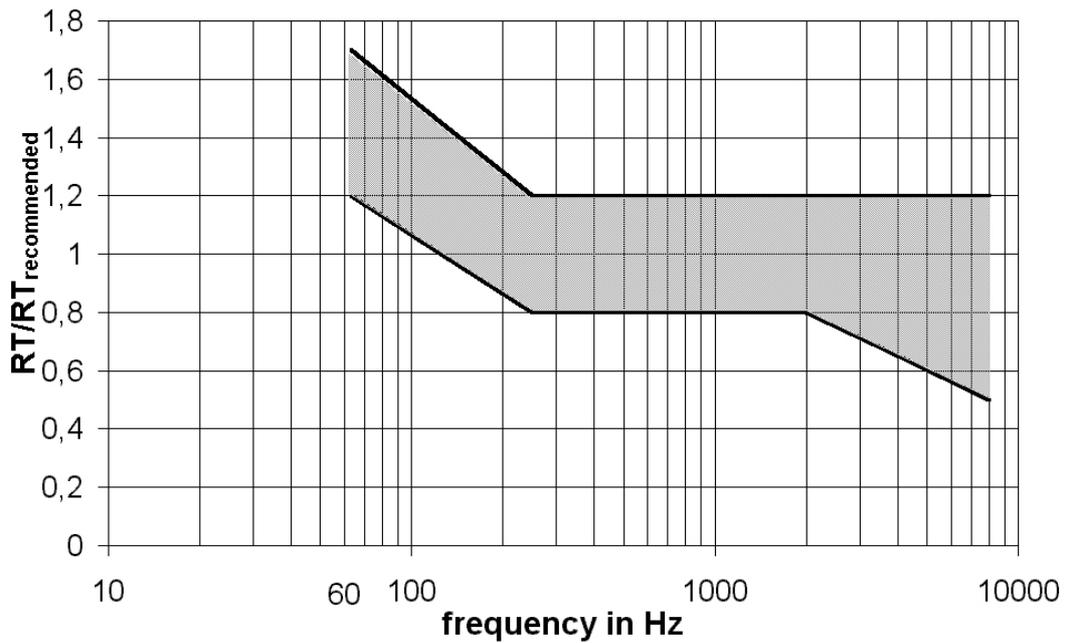


Figure 1.3: Frequency-dependent tolerance range of reverberation time RT referred to  $RT_{\text{recommended}}$  for music presentations

The reverberation time of a room as defined by EYRING mainly depends on the size of the room and on the sound absorbing properties of the boundary surfaces and non-surface forming furnishings:

$$RT = 0.163 (0.049) \frac{V}{-\ln(1 - \bar{\alpha})S_{\text{tot}} + 4mV} \quad (1.4)$$

RT	Reverberation time in s
V	Room volume in m <sup>3</sup> (ft <sup>3</sup> )
$\bar{\alpha} = A_{\text{tot}}/S_{\text{tot}}$	Room-averaged coefficient of absorption
$A_{\text{tot}}$	Total absorption surface in m <sup>2</sup> (sq. ft)
$S_{\text{tot}}$	Total room surface in m <sup>2</sup> (sq. ft)
m	Energy attenuation factor of the air in m <sup>-1</sup> , see Fig. 1.4

The correlation between the mean sound absorption coefficient and the reverberation time for different relations between room volume V and total surface  $S_{\text{tot}}$  is graphically shown in Fig. 1.5.

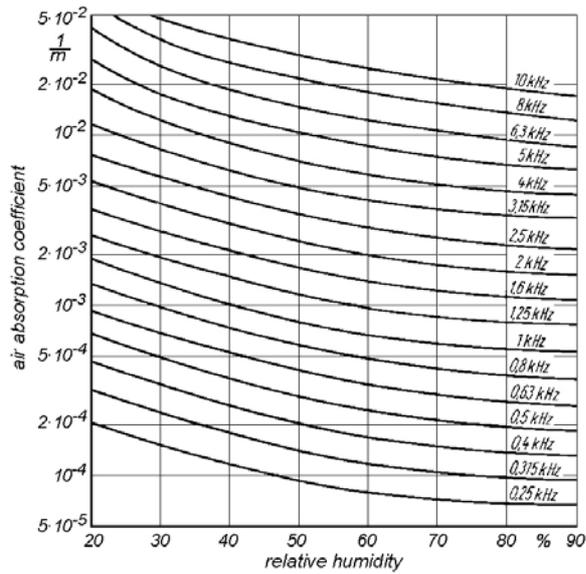


Figure 1.4: Air absorption coefficient  $m$  as a function of relative humidity  $F$

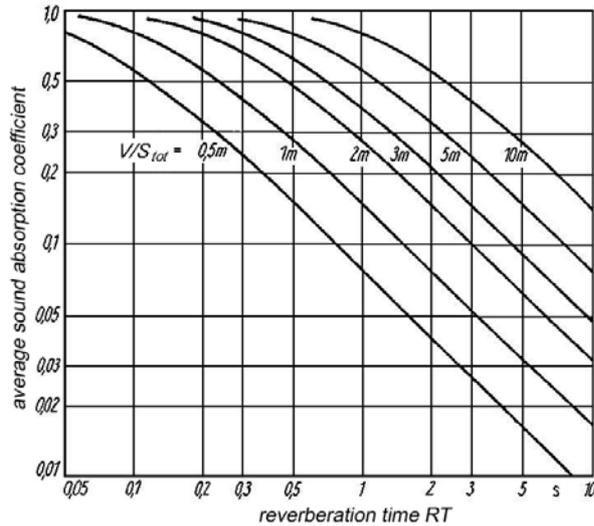


Figure 1.5: Correlation between average sound absorption coefficient and reverberation time for various ratios of room volume  $V$  and room surface  $S_{tot}$

The total sound absorption surface of the room  $A_{tot}$  consists of the planar absorption surfaces with the corresponding partial surfaces  $S_n$  and the corresponding frequency-dependent coefficient of sound absorption  $\alpha_n$  plus the non-surface forming absorption surfaces  $A_k$  consisting e.g. of the audience and the furnishings:

$$A_{tot} = \sum_n \alpha_n S_n + \sum_k A_k \quad (1.5)$$

For an average sound absorption coefficient of up to  $\bar{\alpha} = 0.25$ , the formula (1.4) by EYRING can be simplified using series expansion according to SABINE (see /38/) to:

$$RT = 0.163 (0.049) \frac{V}{A_{tot} + 4mV} \quad (1.6)$$

- |           |   |
|-----------|---|
| RT        | Reverberation time in s   |
| V         | Room volume in $m^3$ ( $ft^3$ )                                 |
| $A_{tot}$ | Total absorption surface in $m^2$ (sq. ft)                      |
| $m$       | Energy attenuation factor of the air in $m^{-1}$ , see Fig. 1.4 |

The correlation between the reverberation time  $RT$ , the room volume  $V$ , the equivalent sound absorption surface  $A_{tot}$  including unavoidable air damping  $m$  is graphically shown in Fig. 1.6.

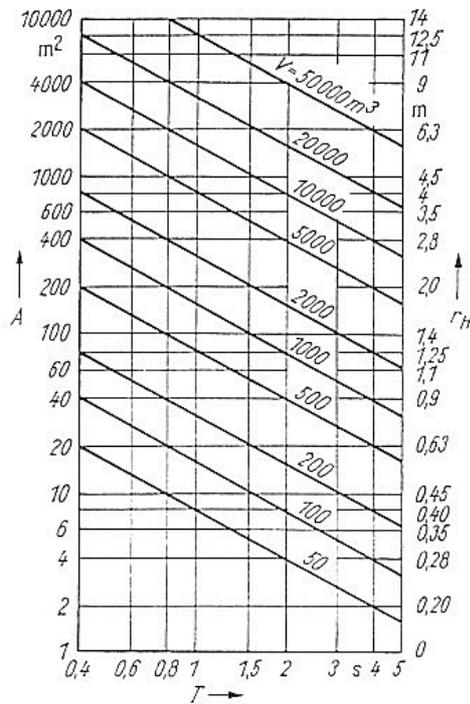


Figure 1.6: Correlation between reverberation time  $RT$ , room volume  $V$  and equivalent sound absorption area  $A$ , according to equation (1.6)

The above stated frequency-dependent sound absorption coefficient has to be determined by measurement or calculation of the diffuse all-round sound incidence. Measurement is generally done in the reverberation room by using formula (1.6). If the sound absorption coefficient is measured by using an impedance tube (or KUNDT's tube) with vertical sound incidence, the results can only be converted to the diffuse sound incidence using the diagrams of MORSE and BOLT /38/, if one can assume that the complex input impedance of the absorber is independent of the angle, i.e. if the lateral sound propagation is inhibited in the absorber (e.g. porous material with a high specific flow resistance).

Properly speaking, the above-mentioned derivatives of the reverberation time from the sound absorption in the room are only valid for approximately cube-shaped rooms with an even distribution of the sound absorbing surfaces within the room. With room shapes deviating heavily from a cube or a parallelepiped, or in case of a necessary one-sided layout of the absorbing audience area, these factors also have a decisive effect on the reverberation time. With the same room volume and the same equivalent sound absorption surface in the room, inclining the side wall surfaces towards the room's ceiling or towards the sound absorbing audience area results in deviations of the measured reverberation time of up to 100 %. For numerous room shapes there exist calculation methods with different degrees of precision, as shown by Kuttruff/38/ for cylinder-shaped rooms. The cause of these differences lies mainly with the geometrical conditions of the room and their influence on the resulting path length of the sound rays determining the reverberation.

The *absorbed sound power*  $P_{ab}$  of a room can be derived from the ratio energy density  $w = \text{sound energy } W/\text{volume } V$  by using the differential coefficient  $P_{ab} = dW/dt$

representing the rate of energy decay in the room and taken from (1.6) and (1.5) (see. /40, p.18/):

$$P_{ab} = \frac{1}{4}cwA \quad (1.7)$$

c	Sound velocity
w	energy density
A	Area

In steady-state, the absorbed sound power is equal to the Power P fed into the room. This results in the average sound energy density  $w_r$  in the diffuse sound field of the room as:

$$w_r = \frac{4P}{cA} \quad (1.8)$$

While the sound energy density  $w_r$  in the diffuse sound field is approximately constant, the direct sound energy and thus also its density  $w_d$  decreases at close range to the source with the square of the distance r from the source, according to:

$$w_d = \frac{P}{c} \frac{1}{4\pi r^2} \quad (1.9)$$

(Strictly speaking, this is valid only for spherical acoustic sources /40/; given a sufficient distance it can be applied, however, to most practically effective acoustic sources).

For the sound pressure in this range of predominantly direct sound, this results in a decline with 1/r. If the direct sound and the diffuse sound energy densities are equal ( $w_d = w_r$ ), the equations (1.8) and (1.9) can be equated, which means it is possible to determine a specific distance from the source, the *reverberation radius (critical distance for omni-directional sources)*  $r_H$ . With a spherical acoustic source this is:

$$r_H = \sqrt{\frac{A}{16\pi}} \approx \sqrt{\frac{A}{50}} \approx 0.141\sqrt{A} \approx 0.057\sqrt{\frac{V}{RT}} \quad (1.10)$$

$r_H$  in m, A in m<sup>2</sup> or sq. ft, V in m<sup>3</sup> or ft<sup>3</sup>, RT in s

With a directional acoustic source (speaker, sound transducer), this distance is replaced by the *critical distance*  $r_R$ :

$$r_R = \Gamma(\vartheta)\sqrt{\gamma} \cdot r_H \quad (1.11)$$

$\Gamma(\vartheta)$ : angular directivity ratio of the acoustic source (the ratio between the sound pressure that is radiated at the angle  $\vartheta$  from the reference axis and the sound

pressure that is generated on the reference axis at the same distance, in other words – the polars)

$\gamma$ : front-to-random factor of the acoustic source

EASERA calculates the reverberation times  $T_{10}$ ,  $T_{20}$ ,  $T_{30}$  in frequency-weighted form (optionally third-octave- or octave-filtered) .

The steepness of the backward-integrated and logarithmic room impulse response (the so-called SCHROEDER PLOT [25]) allows calculating the reverberation time.

In EASERA the reverberation times are determined by this procedure, i.e. according to ISO 3382 [19] across the energy ranges  $-5$  dB to  $-15$  dB ( $\rightarrow T_{10}$ ) ,  $-5$  dB to  $-25$  dB ( $\rightarrow T_{20}$ ) and  $-5$  dB to  $-35$  dB ( $\rightarrow T_{30}$ ).

To minimize measuring errors owing to an insufficient S/N ratio it is necessary to limit the measurement time (integration time). It is recommended that a measurement time of about 0.6 to 0.7 of the expected reverberation time  $T_{exp}$  [19] be chosen.

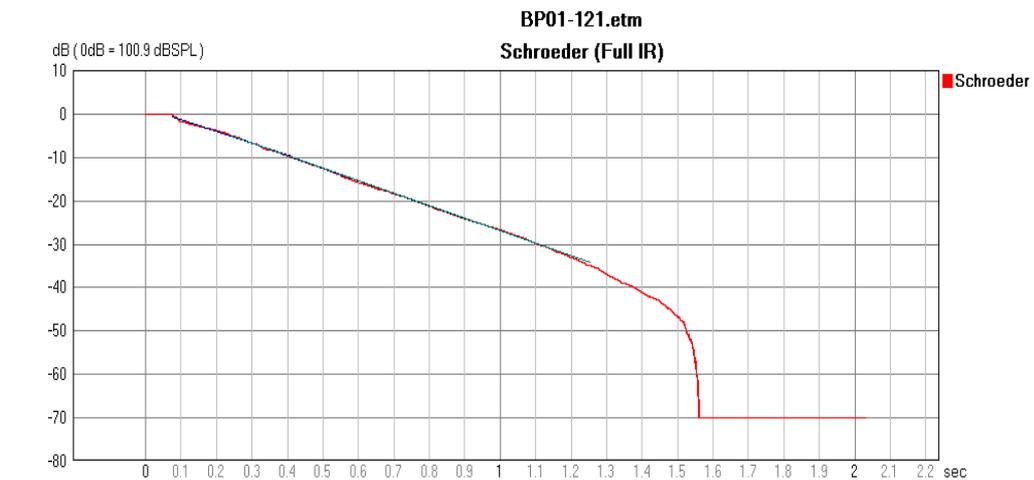
This allows you to obtain an „optimum“ reverberation line with a high correlation degree of the regression line. In addition to limiting the measurement time, there are procedures, as in EASERA, of noise reduction or compensation that can be used.

### 2.1.2. Calculation of the Reverberation Time in EASERA

In EASERA calculation of the reverberation time is done based on the impulse response by using the function **Calculation/Schroeder RT** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



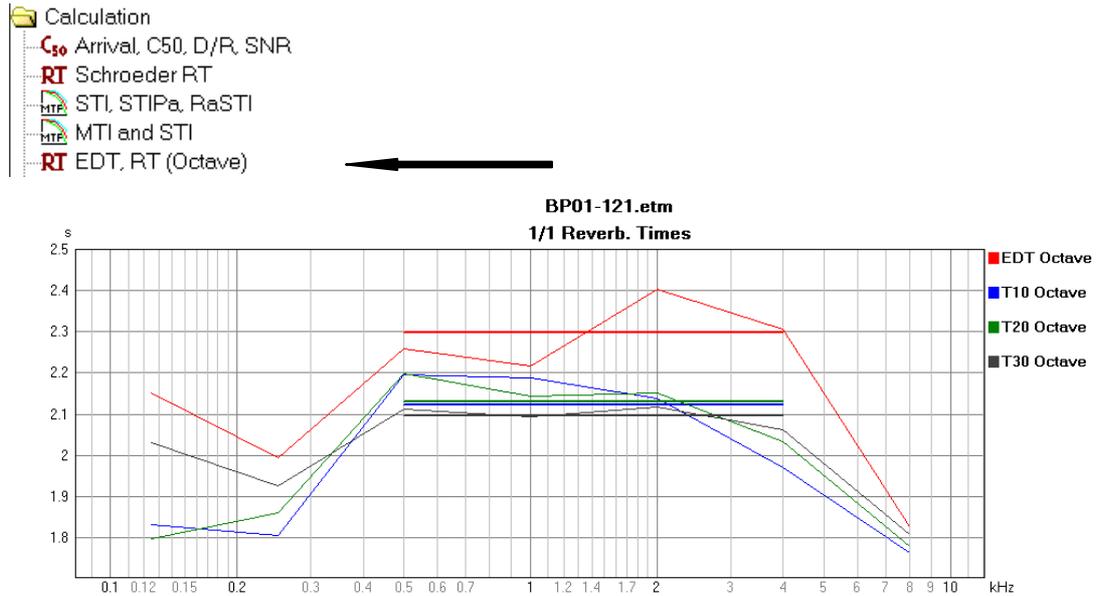
Then the SCHROEDER Plot [25] is shown as a **Graph** (see below) and the corresponding reverberation time values and EDT values are indicated under **Details.Graph** (SCHROEDER Plot)



## Details

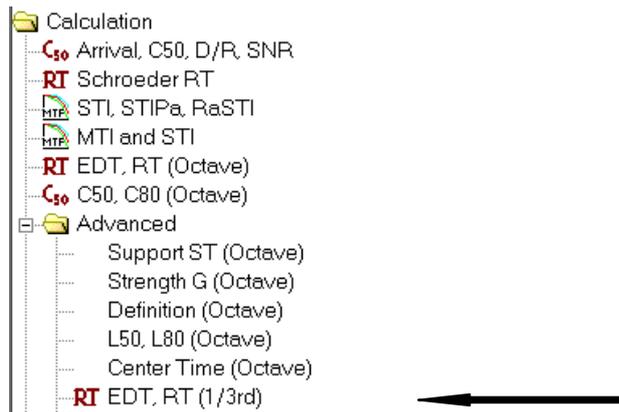
	RT	
EDT	2,20	s
T10	2,02	s
T20	2,07	s
T30	2,07	s

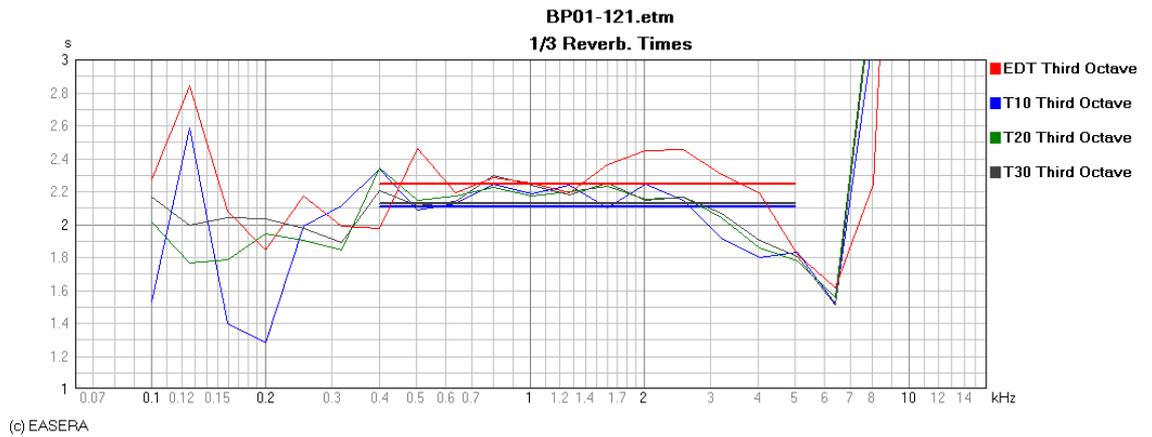
Display of the octave-filtered reverberation times in EASERA using the selection **Calculation/EDT, RT (Octave)**.



(c) EASERA

Display of the third-octave-filtered reverberation times in EASERA using the selection **Calculation/Advanced/EDT, RT (1/3rd)**.





## 2.2. Bass Ratio according to BERANEK [21]

### 2.2.1. Measurement Basis

A single criterion for the reverberation-time frequency response at low frequencies uses the bass ratio according to BERANEK [21]:

$$BR = \frac{T_{20,125} + T_{20,250}}{T_{20,500} + T_{20,1000}} = \frac{T_{20, \text{ lows}}}{T_{20, \text{ mids}}}$$

at  $T_{20,x}$  the index  $x$  indicates the respective octave mid-band frequency or the 2-octave filter range with the frequency-dependent reverberation time measurement or evaluation.

### 2.2.2. Subjective Assessment

For music, the desirable bass ratio is  $BR \approx 1.0$  to  $1.3$ , for speech, on the other hand, the bass ratio should at most have a value of  $BR \approx 0.9$  to  $1.0$ .

### 3 Seat-related Listener Criteria with Speech Performances

#### 3.1. 50-ms Part (D) according to THIELE [1]

##### 3.1.1. Measurement Basis

The room-acoustical criterion for the intelligibility quality of speech performances developed by THIELE [1] was originally referred to as "Definition" and given in %. Owing to the resulting assumption that thereby the speech intelligibility could in effect be measured directly in %, THIELE himself designated the energy ratio as „50-ms-part“.

$$D = \frac{E_{50}}{E_{\infty}}$$

##### 3.1.2. Subjective Assessment

With regard to the subjective assessment of D it is recommended that

$$D = D_{50} > 0.5$$

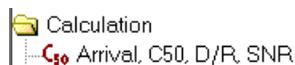
should be frequency independent.

Note:

- Nowadays the criterion is only rarely used, since it was replaced by the Measure of Definition  $C_{50}$  and can also be calculated from it.
- An assessment of the frequency dependence of the „50-ms part“ is not known.

##### 3.1.3. Calculation of the „50-ms Part“ D in EASERA

In EASERA calculation of the „50-ms Part“ D is done based on the room impulse response using the function **Calculation/Arrival, C50, D/R, S/N** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



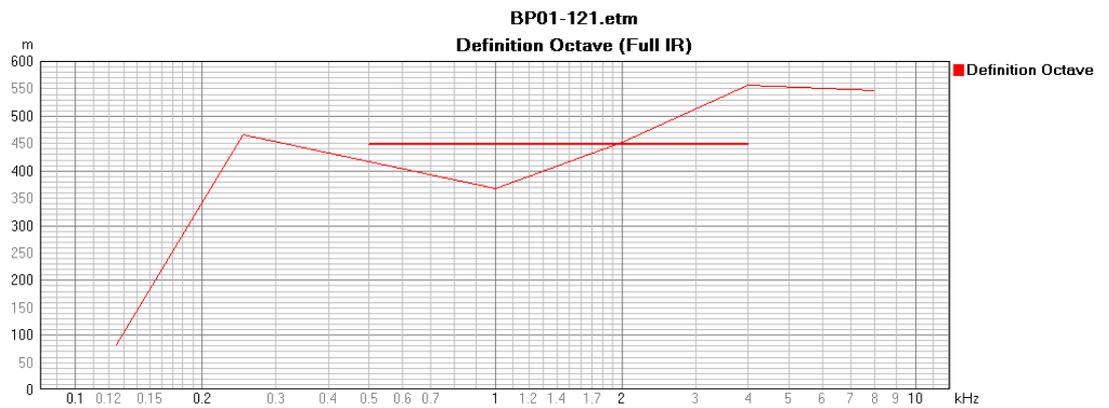
The D-value calculated from the impulse response is then indicated on the left under Data-Details, together with the other energy criteria, according to the above-chosen filter setting (broadband, octave- or third-octave-filtered).

Data		
Arrival	76,29	ms
Distance	26,21	m
C7	-9,0	dB
C50	-1,5	dB
C80	0,0	dB
D	0,414	
Ltotal	118,3	dB SPL
Center Time	136,69	ms



Display of the octave-filtered D-values in EASERA using the selection **Calculation/Advanced/Definition (Octave)**.

- Calculation
  - C<sub>50</sub> Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - MTI STI, STIPa, RaSTI
  - MTI MTI and STI
  - RT EDT, RT (Octave)
  - C<sub>50</sub> C50, C80 (Octave)
- Advanced
  - Support ST (Octave)
  - Strength G (Octave)
  - Definition (Octave)

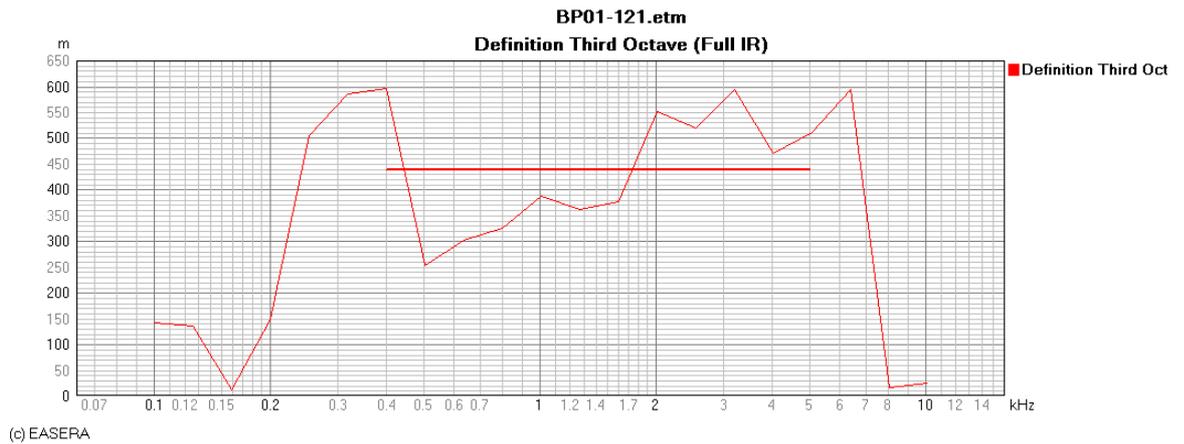


(c) EASERA

Display of the third-octave-filtered D-values in EASERA using the selection **Calculation/Advanced/Definition (1/3rd)**.

- Calculation
  - C<sub>50</sub> Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - MTI STI, STIPa, RaSTI
  - MTI MTI and STI
  - RT EDT, RT (Octave)
  - C<sub>50</sub> C50, C80 (Octave)
- Advanced
  - Support ST (Octave)
  - Strength G (Octave)
  - Definition (Octave)
  - L50, L80 (Octave)
  - Center Time (Octave)
  - RT EDT, RT (1/3rd)
  - C<sub>50</sub> C50, C80 (1/3rd)
  - Strength G (1/3rd)
  - Definition (1/3rd)





## 3.2. Definition Measure $C_{50}$ according to AHNERT [3]

### 3.2.1. Measurement Basis

The measure  $C_{50}$  is relevant for speech intelligibility and calculated from:

$$C_{50} = 10 \log_{10} \left( \frac{E_{50}}{E_{\infty} - E_{50}} \right) \text{ dB.}$$

The EASERA measurement should be carried out using a sound source with the characteristic of a normal talker (front-to-random factor  $\gamma_S \approx 3$ ).

Based on a diffuse „statistical“ sound-field structure, the known room volume  $V$  and the predicted reverberation time  $RT_{60}$ , it is possible to compute the **anticipated value**  $C_{50,E}$  for the definition measure  $C_{50}$  as a function of the distance between the sound source - listener seat ( $r_x$ ). The formula is:

$$C_{50,E} = 10 \log_{10} \frac{\gamma_S \left( \frac{r_H}{r_x} \right)^2 + 1 - e^{-\frac{13.8 \cdot 0.05}{T}}}{e^{-\frac{13.8 \cdot 0.05}{T}}} \text{ dB}$$

$r_x$	distance sound source (talker) → listener seat in m
$r_H$	Half-room diffuse-field distance $r_H = 0.057 \cdot \sqrt{\frac{V}{T}}$ in m
$V$	Volume in $\text{m}^3$
$T$	Reverberation time in s
$\gamma_S$	Front-to random factor of the speaker characteristic

### 3.2.2. Subjective Assessment of the Definition Measure $C_{50}$

For the definition measure  $C_{50}$  there does not exist any normative room-acoustical rules. Assessment rules establishing a qualitative relation between speech intelligibility

and definition measure  $C_{50}$  are known notwithstanding. These rules show that  $C_{50}$  should be  $\geq -2$  dB to avoid the syllable intelligibility decreasing below 80 %. Phrase intelligibility (text intelligibility), however, which thanks to the context is higher than the syllable intelligibility, amounts still to approx. 95 %. A value of  $C_{50} = -2$  dB is therefore considered as the bottom "admissible" limit value for a good speech or text intelligibility.

### 3.2.3. Subjective Assessment of the Frequency Dependence of the Definition Measure $C_{50}$

The subjective assessment of the frequency dependence of the definition measure  $C_{50}$  is not yet sufficiently investigated. Initial attempts in this respect can be found reported by HOFFMEIER [24].

### 3.2.4. Calculation of the Definition Measure $C_{50}$ in EASERA

In EASERA calculation of the definition measure  $D_{50}$  is done based on the room impulse response using the function **Calculation/Arrival, C50, D/R, S/N** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



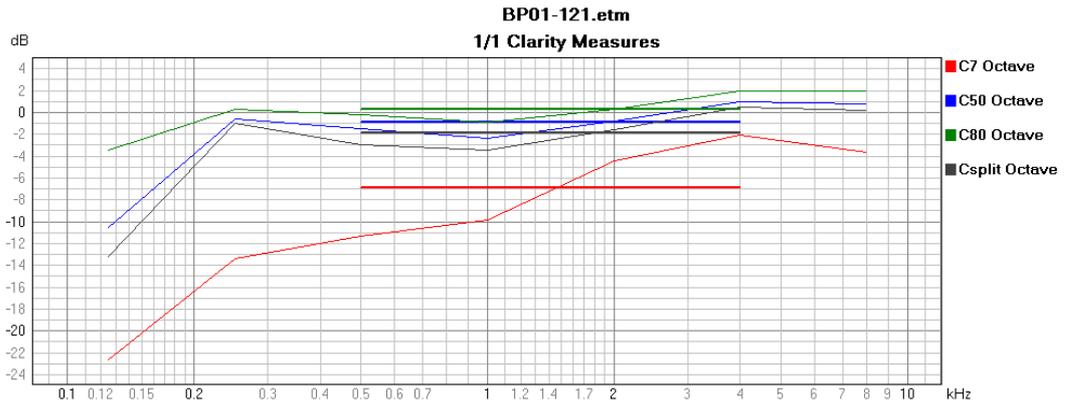
The  $C_{50}$ -value calculated from the impulse response is then indicated on the left under Data-Details, together with the other energy criteria, according to the above-chosen filter setting (broadband, octave- or third-octave-filtered).

Data		
Arrival	76,29	ms
Distance	26,21	m
C7	-9,0	dB
C50	-1,5	dB
C80	0,0	dB
D	0,414	
Ltotal	118,3	dB SPL
Center Time	136,69	ms



Display of the octave-filtered  $C_{50}$  -values is done in EASERA together with the other „clarity measures“ using **Calculation/C50, C80 (Octave)** → blue curve.

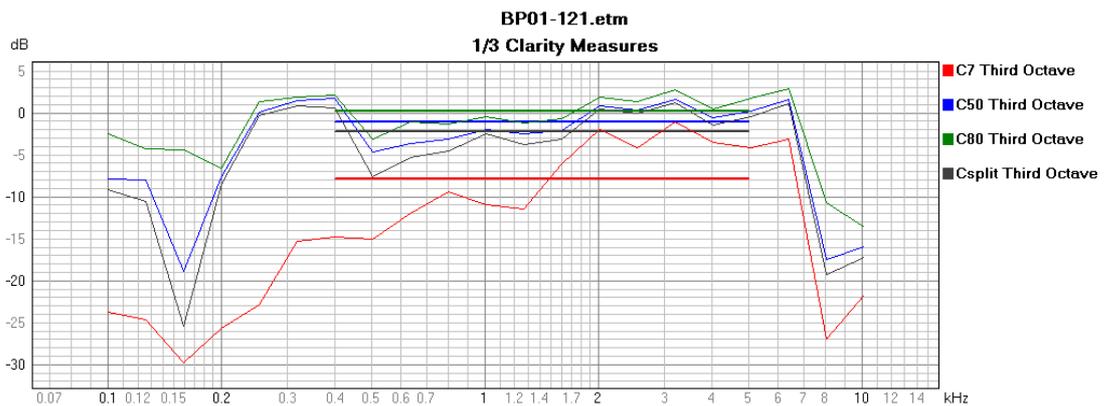




(c) EASERA

Display of the third-octave-filtered  $C_{50}$  -values is done in EASERA together with the other „clarity measures“ using **Calculation/Advanced /C50, C80 (1/3rd)** → blue curve.

- Calculation
  - $C_{50}$  Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - STI, STIPa, RaSTI
  - MTI and STI
  - EDT, RT (Octave)
  - $C_{50}$  C50, C80 (Octave)
- Advanced
  - Support ST (Octave)
  - Strength G (Octave)
  - Definition (Octave)
  - L50, L80 (Octave)
  - Center Time (Octave)
  - EDT, RT (1/3rd)
  - $C_{50}$  C50, C80 (1/3rd)



(c) EASERA

### 3.3. Articulation Loss of Consonants $Al_{cons}$

#### 3.3.1. Measurement Basis

The objective criterion  $Al_{cons}$  (Articulation Loss of Consonants) is an alternative measure in addition to  $C_{50}$  for objective assessment of speech intelligibility in rooms or with sound reinforcement systems.

PEUTZ and KLEIN [7] have found that the articulation loss of spoken consonants  $AL_{\text{cons}}$  is useful for the evaluation of speech intelligibility in rooms.

Calculation of  $AI_{\text{cons}}$  from the room impulse responses determined by EASERA is done using the **Speech Transmission Index (STI)** developed by HOUTGAST and STEENEKEN [4; 5] or using the "fast" variant, the **Rapid Speech Transmission-Index (RASTI)**.

### 3.3.2. Subjective Assessment of the $AI_{\text{cons}}$ Values

The calculated  $AI_{\text{cons}}$  values can be combined with a rating scale so as to allow a **verbal** assessment of the speech intelligibility.

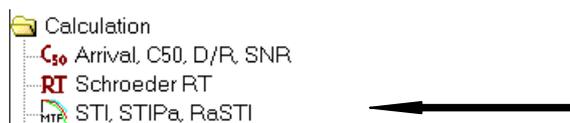
Assigning the results to speech intelligibility yields:

$AI_{\text{cons}} \leq 3\%$	ideal intelligibility
$AI_{\text{cons}} = 3$ to $8\%$	very good intelligibility
$AI_{\text{cons}} = 8$ to $11\%$	good intelligibility
$AI_{\text{cons}} > 11$ to $20\%$	poor intelligibility
$AI_{\text{cons}} > 20\%$	worthless intelligibility (limit value 15%)

Long reverberation times cause an increased articulation loss. With the corresponding duration, this reverberation acts like noise on the following signals and thus reduces the intelligibility.

### 3.3.3. Calculation of $AI_{\text{cons}}$ in EASERA

In EASERA calculation of the  $AI_{\text{cons}}$  according to item 3.3.1 is done using the function **Calculation/STI, STIPa, RaSTI** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



The calculated  $AI_{\text{cons}}$  value is then indicated on the left under Data-Details, together with other intelligibility criteria.

STI	0,495
$AI_{\text{Cons}} [\%]$	11,681
STI (Male)	0,492
STI (Female)	0,500
RaSTI	0,476
Equiv. STIPa (Male)	0,504
Equiv. STIPa (Female)	0,510

### 3.4. Speech Transmission Index STI and RASTI according to STEENEKEN and HOUTGAST [4; 5]

#### 3.4.1. Measurement Basis

The determination of the STI-values is based on measuring the reduction of the signal modulation between the location of the sound source, e.g. on stage, and the reception measurement position with octave center frequencies of 125 Hz up to 8000 Hz.

The authors proceeded on the assumption that not only reverberation and noise reduce the intelligibility of speech, but generally all external signals or signal changes that occur on the path from source to listener. For ascertaining this influence they employ the Modulation Transfer Function (MTF) for acoustical purposes. The available useful signal S (signal) is put into relation with the prevailing interfering signal N (noise). The determined modulation reduction factor  $m(F)$  is a factor that characterizes the interference with speech intelligibility:

$$m(F) = \frac{1}{\sqrt{1 + (2\pi F \cdot RT / 13.8)^2}} \cdot \frac{1}{1 + 10^{-\frac{S/N}{10 \text{ dB}}}}$$

F	modulation frequency in Hz,
RT	reverberation time in s,
S/N	signal/noise ratio in dB.

Modulation frequencies from 0.63 Hz to 12.5 Hz in third octaves are used for the calculation. In addition, the modulation transfer function is subjected to a frequency weighting (WMTF - weighted modulation transfer function), in order to achieve a complete correlation to speech intelligibility. In doing so, the modulation transfer function is divided into 7 frequency bands which are each modulated with the modulation frequency. This results in a matrix of  $7 \times 14 = 98$  modulation reduction factors  $m_i$ .

In order to render this relatively time-consuming procedure to be practical so that it can be applied in "real-time operation", the RASTI-procedure (rapid speech transmission index) was developed from it in cooperation with the company Brüel & Kjaer [36]. The modulation transfer function here is calculated only for two octave bands (500 Hz and 2 kHz) that are especially important for the intelligibility of speech and for select modulation frequencies, i.e. in all for 9 modulation reduction factors  $m_j$ .

To obtain the RASTI value the (apparent) effective signal-noise ratio X can be calculated afterwards from the modulation reduction factors  $m_i$  :

$$X = \frac{1}{9} \sum_{i=1}^9 X_i \qquad X_i = 10 \lg \left( \frac{m_i}{1 - m_i} \right) \text{ dB}$$

(To obtain STI you have to consider all 98  $m_i$  values in this calculation).

According to definition the (RA)STI-value then is:

$$(RA) STI = \frac{X + 15 dB}{30 dB}$$

(compare also IEC 60268-16:2003)

### 3.4.1.1. Subjective Assessment of the (Ra)STI Values

Based on the comparison of subjective examination results with a maximum possible intelligibility of syllables of 96%, the (RA)STI-values are graded in subjective values for syllable intelligibility according to the following table:

Syllable intelligibility	(Ra)STI-value
poor	0 to 0.3
satisfactory	0.3 to 0.45
good	0.45 to 0.6
very good	0.6 to 0.75
excellent	0.75 to 1.0

### 3.4.2. Calculation of the (Ra)STI Values in EASERA

In EASERA calculation of the (Ra)STI values according to item 3.4.1 is done using the function **Calculation/STI, STIPa, RaSTI** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



The calculated (Ra)STI values are then indicated on the left under Data-Details, together with other intelligibility criteria.

#### Data-Details

STI	0,495
AICons [%]	11,681
STI (Male)	0,492
STI (Female)	0,500
RaSTI	0,476
Equiv. STIPa (Male)	0,504
Equiv. STIPa (Female)	0,510

## 3.5. Center Time $t_s$ according to KÜRER [2]

### 3.5.1. Measurement Basis

This quality criterion is relevant for speech intelligibility and musical clarity and measured and calculated according to KÜRER [2] using EASERA as follows:

$$t_S = \frac{\int_0^{\infty} t \cdot p^2(t) dt}{\int_0^{\infty} p^2(t) dt}$$

### Anticipated Value

In the „statistic“ sound field, i.e. for listener seats located at larger distance from the sound source ( $r_x \gg r_H$ ) the **anticipated value**  $t_{s,E}$  for the center time  $t_s$  according to KÜRER [2] is:

$$t_{s,E} = \frac{RT}{13.8} \text{ms}$$

RT

Reverberation time in ms

### 3.5.2. Subjective Assessment

A relationship between the (subjective) syllable intelligibility  $V_S$  and the center time  $t_s$  is given by KÜRER [2] by the following correlation rule:

$$V_S = 96 \cdot \left(1 - t_s^2 \cdot 10^{-5}\right) \%$$

$t_s$  in ms

For a syllable intelligibility of  $V_S \geq 80 \%$  the results according to the above-mentioned rule are  $t_s \leq 130$  ms.

### 3.5.3. Subjective Assessment of the Frequency Dependence of the Center Time $t_s$

The higher the center time  $t_s$  is, the more spatial is the acoustic impression at the listener's position. The maximum achievable center time  $t_s$  is based on the optimum reverberation time. According to HOFFMEIER [24], there is a good correlation between center time and intelligibility of speech with a frequency evaluation of the 4 octaves bands 500 Hz, 1000 Hz, 2000 Hz and 4000 Hz.

For **music**, the desirable center time  $t_s$  is:

$$t_s \approx (70 \text{ to } 150) \text{ ms} \quad \text{with a 1000 Hz octave}$$

and for **speech**:

$$t_s \approx (60 \text{ to } 80) \text{ ms} \quad \text{with four octaves between 500 Hz to 4000 Hz}$$

### 3.5.4. Calculation of the Center Time $t_s$ in EASERA

In EASERA calculation of the center time  $t_s$  is done using the function **Calculation/Arrival, C50, D/R, S/N** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



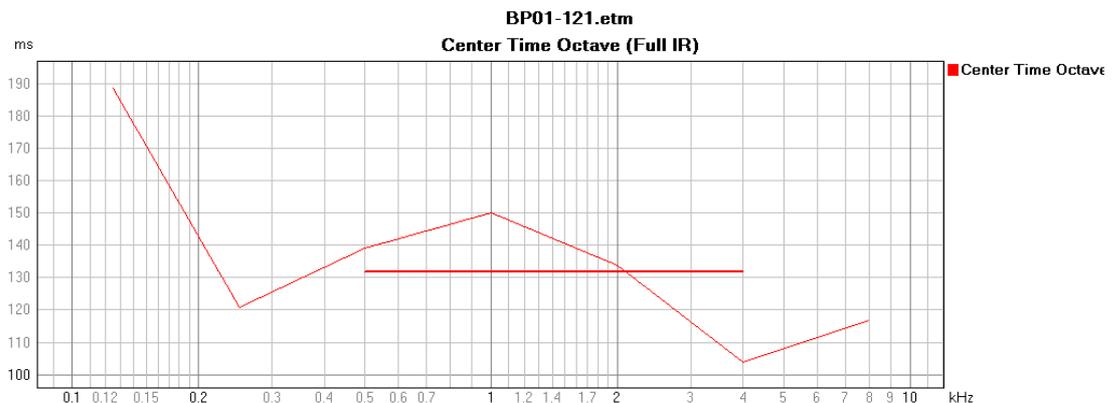
The  $t_s$  value calculated from the impulse response is then indicated on the left under **Data-Details**, together with other energy criteria according to the above-chosen filter setting (broadband, octave- or third-octave-filtered).

Data		
Arrival	76,29	ms
Distance	26,21	m
C7	-9,0	dB
C50	-1,5	dB
C80	0,0	dB
D	0,414	
Ltotal	118,3	dB SPL
Center Time	136,69	ms

Display of the octave-filtered  $D_{50}$  -values is done in EASERA using **Calculation/Advanced/Center Time (Octave)**.

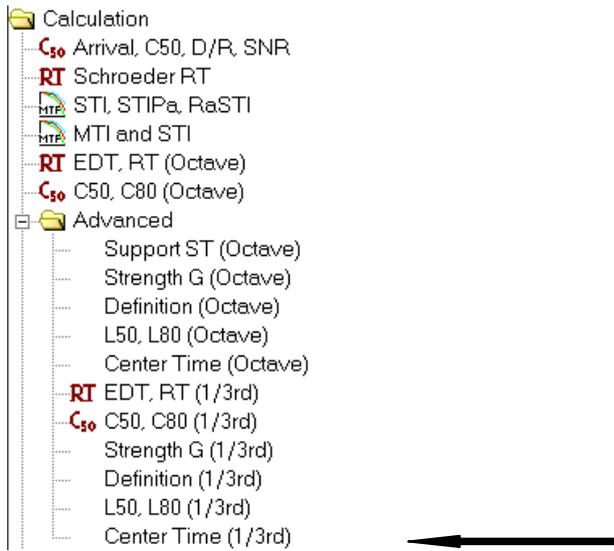


Then the graph Center Time is shown using the octave-filter midband frequency.

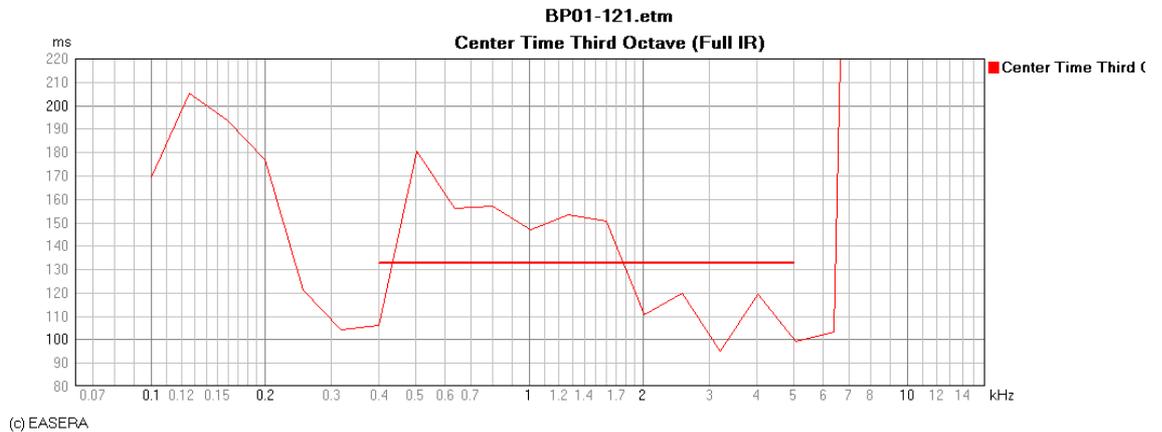


(c) EASERA

Display of the octave-filtered  $t_s$ -values is done in EASERA using **Calculation/Advanced/Center Time (1/3rd)**.



Then the graph Center Time is shown using the third-octave-filter midband frequency.



### 3.6. Echo criterion $EK_{\text{Speech}}$ according to DIETSCH [8]

#### 3.6.1. Measurement Basis

In addition to the above-mentioned criteria, the reflection sequence is also of importance for assessing the acoustic overall impression of a room. The reflectograms show in which temporal sequence and with which intensity reflections arrive at a listener's position.

When planning a hall one tries to design the surfaces of the room in such a way that at all seats the reflections sequence is as uniform and dense as possible and that no **high-energy late reflections (echoes)** occur.

The designer examines the sound-pressure records as to whether this uniformity is given and whether the intensity of the reflections decreases along with the increasing temporal distance from the direct sound.

Strong reflections which in speech performances arrive later than 50 ms after the direct sound and which are not preceded by any or only few weaker reflections, are subjectively perceived by the ear as signals not related to the direct sound, i.e. as an echo.

According to KUHL [27] an **Echo** is defined as a subjectively „clearly audible repetition of the direct-sound phenomenon“.

A certain echo impression (but not in the sense proposed by KUHL) may also come about when low-energy groups of reflections occur at a late moment during the decay process. Such a phenomenon „amplifies“ the spatial impression in a certain way and may even be considered as positive in a room-acoustical sense.

The decision as to whether a group of reflections is an echo can frequently not be reached on the basis of the **sound-pressure reflectogram**, but only by assessing the behavior of the **sound-level decay process** at a receiving position. This is why not only the sound-pressure behavior over time is recorded and evaluated at a receiving position, but also the sound-level behavior and the sound intensity behavior.

A further possibility for recognizing echoes in the reflectograms is offered by the so-called DIETSCH criterion [8].

This is an echo criterion EK which DIETSCH calculates, differently weighted for speech and music performances, as follows:

$$EK_{Speech} = \frac{\Delta t_s(\tau)}{\Delta \tau_E}$$

with:

$$t_s(\tau) = \frac{\int_0^{\tau} t |p(t)|^n dt}{\int_0^{\tau} |p(t)|^n dt}$$

For speech performances use these values:

$$\Delta \tau_E = 9 \text{ ms}$$

$$n = 2/3$$

$$EK_{limit} = 1$$

### 3.6.2. Subjective Assessment of the Echo criterion $EK_{Speech}$ and Frequency Dependence of the same

An echo occurs when  $EK_{max} > EK_{limit}$ .

If  $EK_{max} > EK_{limit}$  occurs periodically (periodicity 50 ms with speech, 80 to 100 ms with music), a **flutter echo** becomes audible.

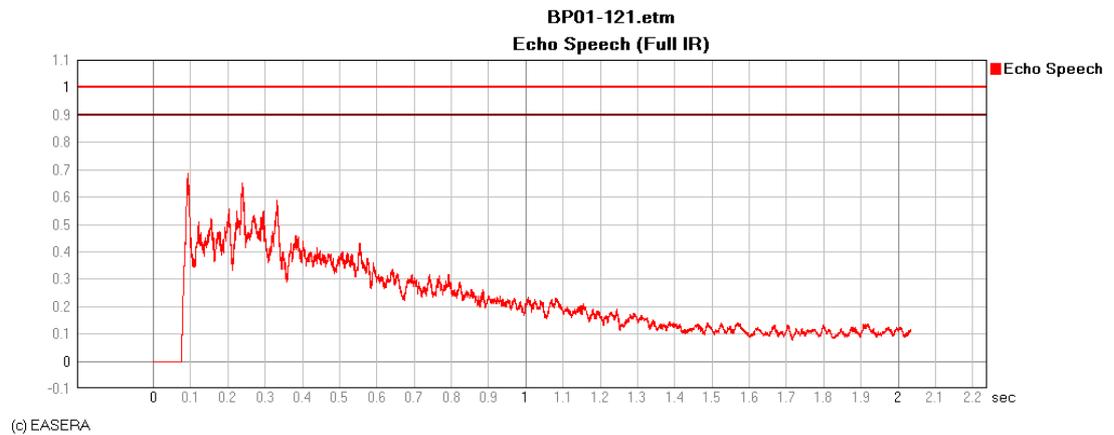
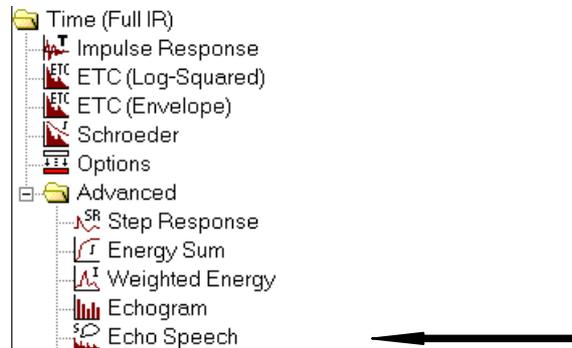
With band-limited evaluation of the room impulse responses you have to keep in mind that especially the **high-frequency** signal components tend to cause echo disturbances.

According to DIETSCH [8] it is, however, sufficient to employ:

- for **Speech**: Test signals with a bandwidth of one octave and a midband frequency of  $f_M = 1 \text{ kHz}$ ,

### 3.6.3. Calculation of $EK_{\text{speech}}$ in EASERA

In EASERA calculation of  $EK_{\text{speech}}$  graphs is done based on the room impulse response using the function **Time (Full IR)/Advanced/Echo Speech** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



(c) EASERA

A frequency assessment does not take place, but the  $EK_{\text{speech}}$  graph is calculated from the **broadband** impulse response (Full IR) by default. The octave and third-octave filtered responses can also be viewed.

## 4 Seat- and Sound Source-related Listener Criteria with Music Performances

### 4.1. Direct Sound Level $C_7$

#### 4.1.1. Measurement Basis

This measure reflects the sound energy component of the direct sound in relation to the sound energy of the reflections and of the reverberation arriving after the direct sound at the listener position.

This measure is relevant for the subjective perception of "nearness" or "directness" of the sound sources (singers, orchestra register, soloists).

The measuring rule is [15]:

$$C_7 = 10 \log_{10} \left( \frac{E_7}{E_\infty - E_7} \right) \text{ dB}$$

made with monaural measurements using an omni-directional microphone K:

$$E_{x,K} = \int_0^{x \text{ ms}} p_K^2(t) dt$$

the cumulative energy of the squared room impulse response  $[p_K^2(t)]$  until  $x$  ms after the direct sound.

#### 4.1.2. Subjective Assessment

The direct sound measure  $C_7$  should in correlation to the distance from the sound source - listener not fall below a range of -10 to -15 dB.

An assessment method for the frequency dependence of the direct sound level is not yet known.

#### 4.1.3. Calculation of the Direct Sound Measure $C_7$ in EASERA

In EASERA calculation of the direct sound measure is done based on the room impulse response using the function **Calculation/Arrival, C50, D/R, S/N** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



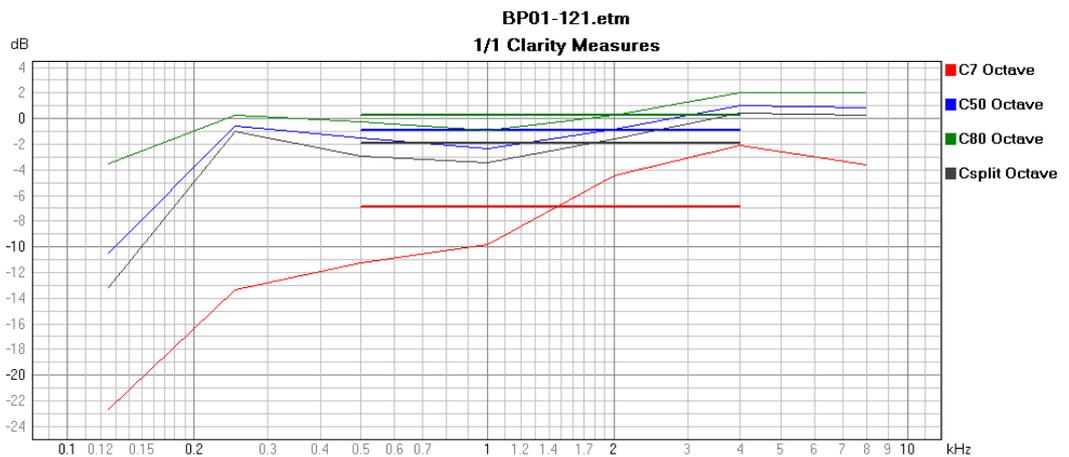
The  $C_7$  value calculated from the impulse response is then indicated on the left under Data-Details, together with other energy criteria according to the above-chosen filter setting (broadband, octave- or third-octave-filtered).

Data		
Arrival	76,25	ms
Distance	26,03	m
$C_7$	-9,0	dB
$C_{50}$	-1,5	dB
$C_{80}$	0,0	dB
$C_{35}$	-2,5	dB
D	0,414	
Total SPL	118,3	dB SPL
Center Time	136,74	ms



Display of the octave-filtered  $C_7$  values is done in EASERA, together with that of the other "clarity measures, via **Calculation/C50, C80 (Octave)** → red curve.

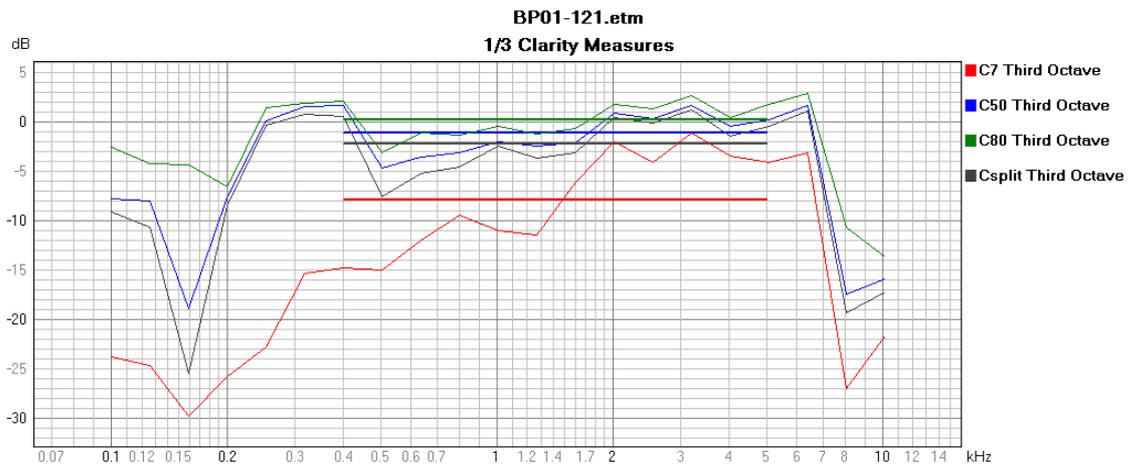
- Calculation
  - $C_{50}$  Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - STI, STIPa, RaSTI
  - MTI and STI
  - EDT, RT (Octave)
  - $C_{50}$  C50, C80 (Octave)



(c) EASERA

Display of the third-octave-filtered  $C_7$  values is done in EASERA, together with that of the other "clarity measures, via **Calculation/Advanced/C50,C80 (1/3rd)** → blue curve.

- Calculation
  - C<sub>50</sub> Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - STI, STIPa, RaSTI
  - MTI and STI
  - RT EDT, RT (Octave)
  - C<sub>50</sub> C50, C80 (Octave)
- Advanced
  - Support ST (Octave)
  - Strength G (Octave)
  - Definition (Octave)
  - L50, L80 (Octave)
  - Center Time (Octave)
  - RT EDT, RT (1/3rd)
  - C<sub>50</sub> C50, C80 (1/3rd)



(c) EASERA

## 4.2. Clarity Measure C<sub>80</sub> according to ABDEL ALIM [9]

### 4.2.1. Measurement Basis

The clarity measure C<sub>80</sub> is relevant for temporal and the register clarity of music performance, especially of rapid musical passages.

It is calculated from:

$$C_{80} = 10 \log_{10} \left( \frac{E_{80}}{E_{\infty} - E_{80}} \right) dB$$

Based on the assumption of a diffuse „statistical“ sound-field structure, the known room volume V and the predicted reverberation time T, it is possible to compute the **anticipated value C<sub>80,E</sub>** for the clarity measure C<sub>80</sub> as a function of the distance from the sound source - listener seat (r<sub>x</sub>). The formula is:

$$C_{80,E} = 10 \log_{10} \frac{\left( \frac{r_H}{r_x} \right)^2 + 1 - e^{-\frac{13,80,08}{T}}}{e^{-\frac{13,80,08}{T}}} dB$$

$r_x$	Distance sound source (orchestra) → listener seat in m
$r_H$	Half-room diffuse-field distance $r_H = 0,057 \cdot \sqrt{\frac{V}{T}}$ in m
$V$	Volume in $m^3$
$T$	Reverberation time in s

#### 4.2.2. Subjective Assessment of the Clarity Measure $C_{80}$

For an "optimum" clarity measure  $C_{80}$  there does not yet exist any normative room-acoustical rules.

According to the papers of ABDEL ALIM [9] a sufficient musical clarity should be obtained with:

$C_{80} \geq -1,6$ dB	for classical music (Mozart, Haydn)
$C_{80} \geq -4,6$ dB	for romantic music (Brahms, Wagner).

The requirement:

$$C_{80} \geq -3 \text{ dB}$$

is an acceptable compromise.

For sacral music even:

$$C_{80} \geq -5 \text{ dB}$$

can be accepted.

An assessment method for the frequency dependence of the clarity measure is not yet known.

#### 4.2.3. Calculation of the Clarity Measure $C_{80}$ in EASERA

In EASERA calculation of the clarity measure  $C_{80}$  is done based on the room impulse response using the function **Calculation/ Arrival, C50, D/R, S/N** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



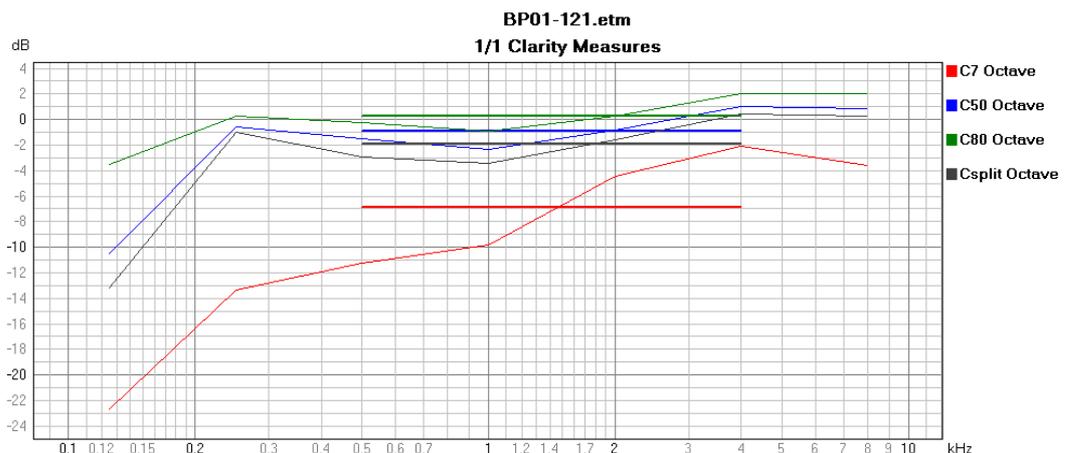
The  $C_{80}$  value calculated from the impulse response is then indicated on the left under Data-Details, together with other energy criteria according to the above-chosen filter setting (broadband, octave- or third-octave-filtered).

Data		
Arrival	76,25	ms
Distance	26,03	m
C7	-9,0	dB
C50	-1,5	dB
C80	0,0	dB
C35	-2,5	dB
D	0,414	
Total SPL	118,3	dB SPL
Center Time	136,74	ms



Display of the octave-filtered  $C_{80}$  values is done in EASERA, together with that of the other "clarity measures", using **Calculation/C50, C80 (Octave)** → green curve.

- Calculation
- $C_{50}$  Arrival, C50, D/R, SNR
- RT Schroeder RT
- STI, STIPa, RaSTI
- MTI and STI
- EDT, RT (Octave)
- $C_{50}$  C50, C80 (Octave)



(c) EASERA

### 4.3. Interaural Cross Correlation IACC according to BERANEK [29;30]

#### 4.3.1. Measurement Basis

According to BERANEK [29; 30] the value  $\rho = (1 - IACC_E)$  correlates with the subjective perception of the "width" of the sound source (AWS: „**A**pparent **S**ource **W**idth“) and the value  $\varepsilon = (1 - IACC_L)$  correlates with the subjective perception of being "enveloped by the sound" (LEV: „**L**istener **E**nvelopm“).

With binaural measurements (e.g. with dummy-head microphones) it is possible to calculate interaural correlation measures.

From the room impulse responses captured by the right and left "ears" ( $p_R(t)$  and  $p_L(t)$ ) the interaural correlation measures according to ISO 3382 [26] are calculated as follows using the interaural cross correlation function  $IACF(\tau)$ :

$$IACF_{t_1; t_2; F}(\tau) = \frac{\int_{t_1}^{t_2} p_L(t) \cdot p_R(t + \tau) dt}{\sqrt{\int_{t_1}^{t_2} p_L^2(t) \cdot \int_{t_1}^{t_2} p_R^2(t) dt}}$$

$t_1$  and  $t_2$                       Integration time limits in ms  
 for  $IACC_{E(early)}$ :  $t_1 = 0$  ms;  $t_2 = 80$  ms  
 for  $IACC_{L(ate)}$ :  $t_1 = 80$  ms;  $t_2 = 500$  to  $2000$  ms  
 for  $IACC_{A(II)}$ :  $t_1 = 0$  ms;  $t_2 = 500$  to  $2000$  ms

$F$                                       Frequency range in Hz  
 e.g.  $IACC_{E3B}$      $IACC_{E,average}$  across 3-octave frequency ranges  
 500, 1000 and 2000 Hz,  $t_1 = 0$  ms;  $t_2 = 80$  ms

The interaural cross correlation coefficients  $IACC$  are calculated using the interaural cross correlation functions  $IACF(\tau)$  as follows:

$$IACC_t = |IACF_t(\tau)|_{\max} \quad \text{for } -1 < \tau < +1 \quad (\tau \text{ in ms})$$

**Note:** Complying with BERANEK's suggestions [29; 30], deviations from ISO 3382 [26] were introduced in the following items of the calculations of the  $IACC$  values carried out in this respect:

$IACC_L$ :                               $t_1 = 80$  ms;  $t_2 = 500$  ms  
 $IACC_A$ :                               $t_1 = 0$  ms;  $t_2 = 500$  ms  
 $IACC_{E,L,A,F}$                        $F = T$  for 2-octave frequency range „low“ (88 - 353 Hz)  
      $F = M$  for 2- octave frequency range „middle“ (353 - 1414 Hz)  
      $F = H$  for 2- octave frequency range „high“ (1414 - 5656 Hz)

#### 4.3.2. Subjective Assessment of $IACC$ incl. Frequency Dependence of the Same

For the values of  $IACC_{E3B}$  or  $\rho = (1 - IACC_{E;500,1000,2000Hz})$  BERANEK [29; 30] specifies the following quality categories for concert halls:

Category „Excellent“ to „Superior“	$IACC_{E;500,1000,2000Hz}$	0.28 to 0.38
	$\rho = (1 - IACC_{E;500,1000,2000Hz})$	0.62 to 0.72
Category „Good to Excellent“	$IACC_{E;500,1000,2000Hz}$	0.39 to 0.54
	$\rho = (1 - IACC_{E;500,1000,2000Hz})$	0.46 to 0.61

Category „Fair to Good“

IACC<sub>E;500,1000,2000Hz</sub>)

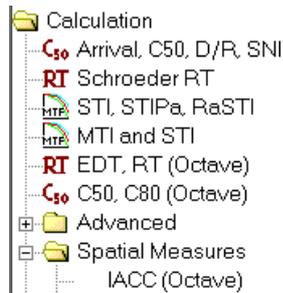
0.55 to 0.59

$\rho = (1 - \text{IACC}_{E;500,1000,2000\text{Hz}})$

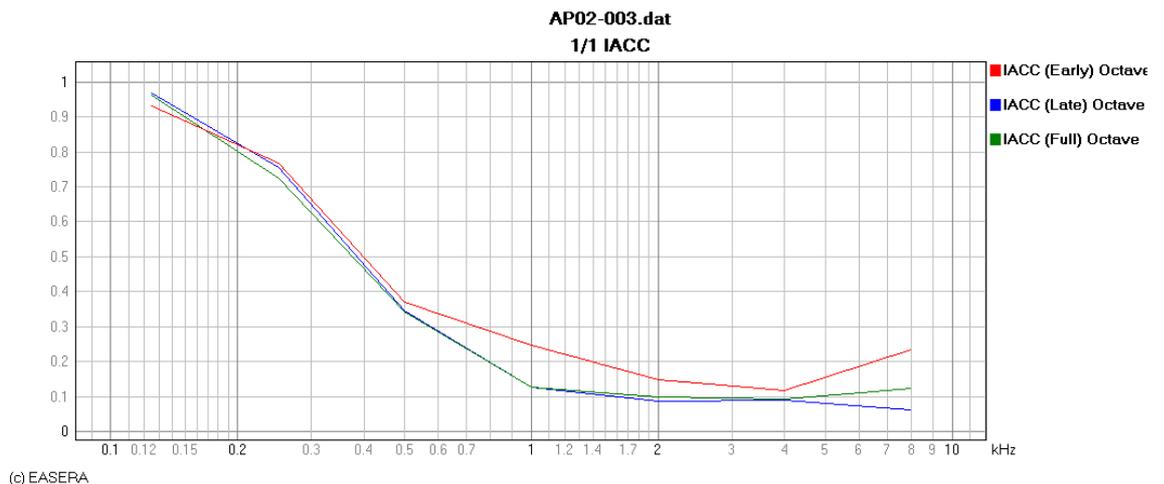
0.41.. 0.45

### 4.3.3. Calculation of IACC in EASERA

Calculation of IACC of the octave-frequency-filtered RIR is done in EASERA using the function **Calculation/ Spatial Measures/IACC (Octave)** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



Graphs show IACC<sub>early,oct</sub> (red curve), IACC<sub>late,oct</sub> (blue curve) and IACC<sub>full,oct</sub> (green curve).



## 4.4. Strength Measures G according to LEHMANN [11]

### 4.4.1. Measurement Basis

If one relates the sound level measured at the listener position to a sound level equivalent to the acoustic performance of the sound source, one obtains, given an omni-directional characteristic of the source, the strength measure G calculated according to LEHMANN [11] by the following formula:

$$G = 10 \log_{10} \frac{\int_0^{\infty} p^2(x, t) dt}{\int_0^{\infty} \gamma \cdot p^2(s, t) dt} - 10 \log \left( 4\pi \frac{s^2}{m^2} \right) \text{ dB}$$

- s Reference distance (approx. 10 m)
- x Distance of the measuring position from the sound source, in m
- $\gamma$  Front-to-Random factor of the sound source ( $\equiv$  ratio of the sound-pressure square in the main radiation direction to that which would result with omnidirectional uniform radiation at the same distance and with equal overall performance of the sound source). Measurements made with a (non-directional) dodecahedron loudspeaker produced  $\gamma \approx 1$ .

Note:

Provided the acoustic output of the source remains constant over the time which a measurement series runs, a one-off measurement of the same **before** starting the series (if need be also a control measurement **after** conclusion of the series) is sufficient.

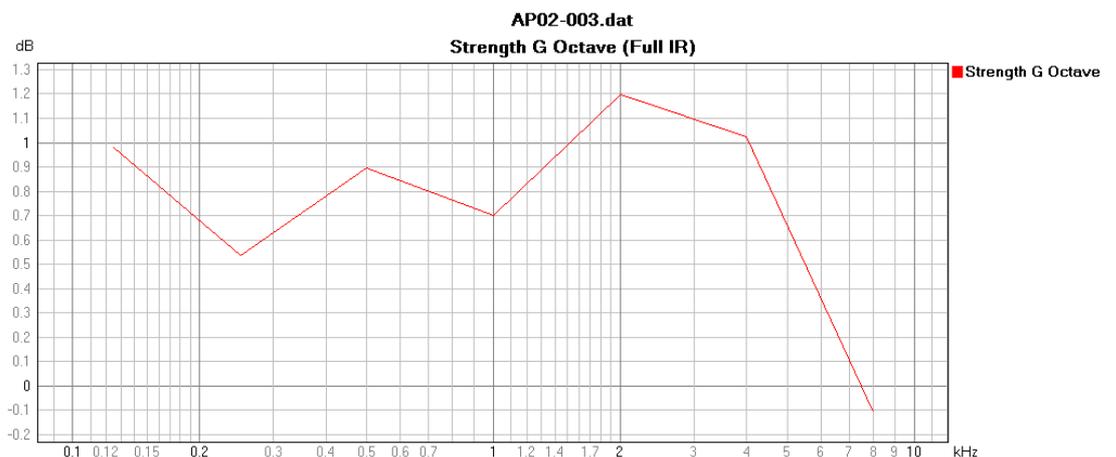
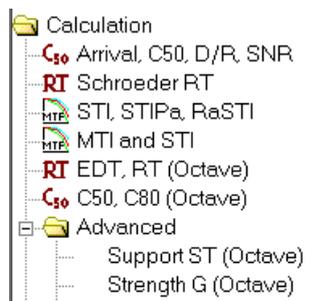
#### 4.4.2. Subjective Assessment of the Strength Measure G and Frequency Dependence of the Same

For the strength measure G the recommendation is for  $G \geq 0$  dB (in the mean frequency range of 500 to 1000 Hz).

An assessment method for the frequency dependence of the strength measures is not yet known.

#### 4.4.3. Calculation of the Strength Measure G in EASERA

Calculation and display of the octave-filtered strength measures is done in EASERA via **Calculation/Advanced/Strength G (Octave)** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



Calculation and display of the third-octave-filtered strength measures is done in EASERA using **Calculation/Advanced/Strength G (1/3rd)** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.

- Calculation
  - C<sub>50</sub> Arrival, C50, D/R, SNR
  - RT Schroeder RT
  - MTI STI, STIPa, RaSTI
  - MTI MTI and STI
  - RT EDT, RT (Octave)
  - C<sub>50</sub> C50, C80 (Octave)
  - Advanced
    - Support ST (Octave)
    - Strength G (Octave)
    - Definition (Octave)
    - L50, L80 (Octave)
    - Center Time (Octave)
    - RT EDT, RT (1/3rd)
    - C<sub>50</sub> C50, C80 (1/3rd)
    - Strength G (1/3rd)



## 5 Room- or Seat-related Criteria

### 5.1. Reverberance Measure R according to BERANEK [16]

#### 5.1.1. Measurement Basis

The reverberance measure R is with music performances relevant for the acoustic liveness or "reverberance" of the acoustic impression.

The formula is:

$$R = 10 \log_{10} \left( \frac{E_{\infty} - E_{50}}{E_{50}} \right) \text{dB}$$

Based on the assumption of a diffuse „statistical“ sound-field structure, the known room volume V and the predicted reverberation time RT, it is possible to compute the **anticipated value R<sub>E</sub>** for the reverberance measure R as a function of the distance from the sound source - listener seat (r<sub>x</sub>). The formula is:

$$R_E = 10 \log_{10} \frac{e^{\frac{13.8 \cdot 0.05}{RT}}}{\left( \frac{r_H}{r_x} \right)^2 + 1 - e^{\frac{13.8 \cdot 0.05}{RT}}} \text{dB}$$

r <sub>x</sub>	Distance sound source (orchestra) → listener seat in m
r <sub>H</sub>	Half-room diffuse-field distance $r_H = 0.057 \cdot \sqrt{\frac{V}{T}}$ in m
V	Volume in m <sup>3</sup>
RT	Reverberation time in s

#### 5.1.2. Subjective Assessment Reverberance Measure

According to the papers of BERANEK and SCHULTZ [16], the values for the reverberance measure should be within the range of:

$$+6 \text{ dB} \geq R \geq +2 \text{ dB.}$$

An assessment method for the frequency dependence of the reverberance measure is not yet known.

### 5.1.3. Calculation of the Reverberance Measure R in EASERA

Calculation of the reverberance measure R is not explicitly implemented in EASERA. The reverberance measure R may, however, be easily calculated on the basis of  $C_{50}$  through the relationship

$$R = - C_{50}$$

## 5.2. Lateral Efficiency LE according to JORDAN [18] and Lateral Fraction LF according to BARRON [20]

### 5.2.1. Measurement Basis

The objective measure defined by JORDAN [18] for the acoustic overall impression of a room is calculated as:

$$LE = \frac{\text{lateralEnergy}(25...80ms)}{\text{totalEnergy}(allSides)(0...80ms)}$$

According to BARRON [20] it is the sound reflections arriving from the side at a listener's position within a time window from 5 ms to 80 ms that are responsible for the acoustically perceived extension of the musical sound source (contrary to JORDAN who considers a time window from 25 ms to 80 ms).

BARRON [20] calls the Lateral Efficiency LE according to JORDAN [18] Lateral Energy Fraction LF giving the following formula:

$$LF = \frac{E_{80BI} - E_{5BI}}{E_{80}}$$

$E_{xBI}$  Sound energy component, measured with a bi-directional (figure8) microphone (gradient microphone).

For the subjective assessment of the apparent extension of a musical sound source, e.g. on stage, the early sound reflections arriving at a listener's seat from the side are of greatest importance, as compared with all other directions. Therefore the ratio between the laterally arriving sound energy components and those arriving from all sides, each within a time of up to 80 ms, is determined and its tenfold logarithm is calculated.

If one multiplies the arriving sound reflections with  $\cos\Phi$ , with  $\Phi$  being the angle between the direction of the sound source and that of the arriving sound wave, one achieves the more important evaluation of the lateral reflections. With measurements this angle-dependent evaluation is achieved by employing a microphone with bi-directional characteristics.

The higher the lateral efficiency, the acoustically broader the sound source appears.

For obtaining a uniform representation of the energy measures in room acoustics, LE and LF can also be defined as lateral efficiency measure  $LEM = 10 \log LE$  dB and as lateral energy fraction measure  $LFM = 10 \log LF$  dB, respectively.

## Note

The overall measurement of e.g. a concert hall with diverse microphone arrangements such as the combination of omni-directional and figure 8 microphone and dummy-head microphone is frequently not possible for time and financial reasons.

Thus it stands to reason to content oneself with only one microphone arrangement for a complete measurement run.

The Lateral Fraction LF proposed by BARRON [20] may then be determined based on the dummy-head room impulse responses using this formula:

$$LF = \frac{\int_5^{80} p_L^2(t) dt + \int_5^{80} p_R^2(t) dt - 2 \int_5^{80} p_L(t) \cdot p_R(t) dt}{\int_0^{80} p_L^2(t) dt + \int_0^{80} p_R^2(t) dt}$$

(see also ISO 3382 /26/)

### 5.2.2. Subjective Assessment of the Lateral Efficiency LE and LEM or LF and LFM

Then the favorable range is between:

$$0.3 < LE < 0.8$$

$$-5 \text{ dB} < LEM < -1 \text{ dB.}$$

It is of advantage if LF is within the following range:

$$0.10 < LF < 0.25,$$

or, with the logarithmic representation of the lateral fraction measure LFM = 10 lg LF, within:

$$-10 \text{ dB} < LFM < -6 \text{ dB}$$

#### Note:

According to BARRON and MARSHALL:[35] the lateral reflections are responsible for subjective effects varying with the frequency range as per the following correlation:

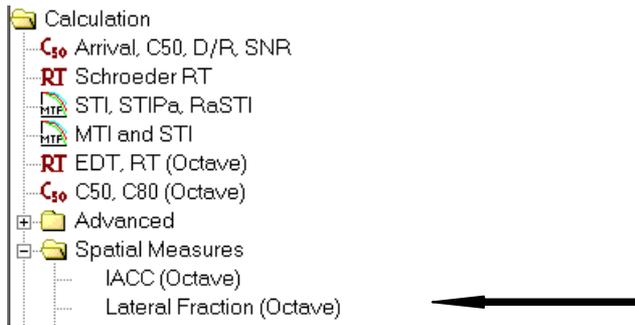
LF-octave frequency range 125 Hz  $\geq$  LF  $\geq$  500 Hz: **Envelopment**

LF-octave frequency range 500 Hz  $>$  LF  $\geq$  4000 Hz: **Source broadening**

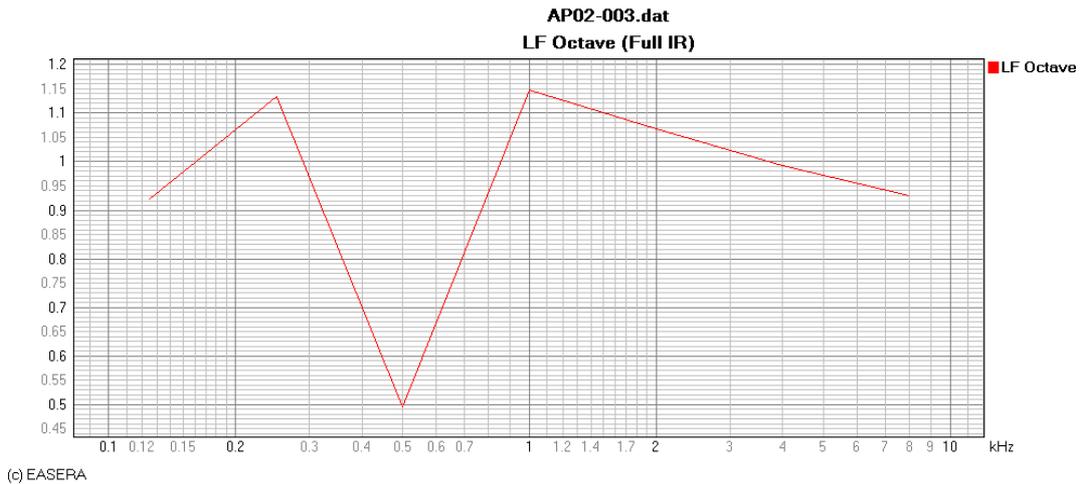
LF-octave frequency range LF  $>$  4000 Hz: **Image shifting**

### 5.2.3. Calculation of LF in EASERA

Calculation of the LF of the octave-frequency-filtered RIR is done in EASERA using the function **Calculation/Spatial Measures/Lateral Fraction (Octave)** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



Graphs show LF as a function of the midband frequency of the octave-filter.



### 5.3. Modified LF by Consideration of the Angle of Incidence with the: LFC according to KLEINER [34]

#### 5.3.1. Measurement Basis

Both lateral efficiencies LE and LF have in common thanks to using a gradient microphone, that the resulting contribution of a single sound reflection to the lateral sound energy behaves like the square of the cosine of the reflection incidence angle, referred to the axis of the highest microphone sensitivity.

KLEINER [34] therefore defines the lateral fraction coefficient LFC in better accordance with the subjective evaluation, whereby the contributions of the sound reflections vary like the cosine of the angle  $\Phi$ :

$$LFC = \frac{\int_0^{80} |p_{BI}(t) \cdot p(t)| dt}{E_{80}}$$

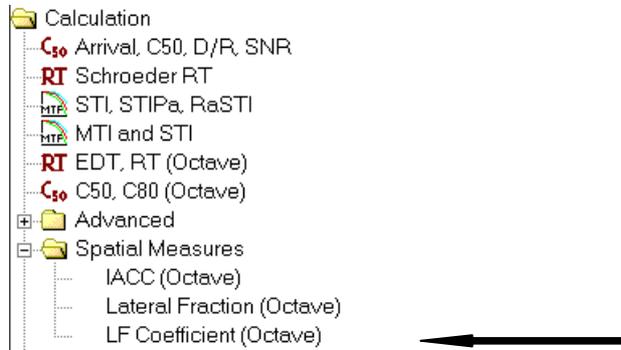
(see also ISO 3382 /26/)

#### 5.3.2. Subjective Assessment of LFC

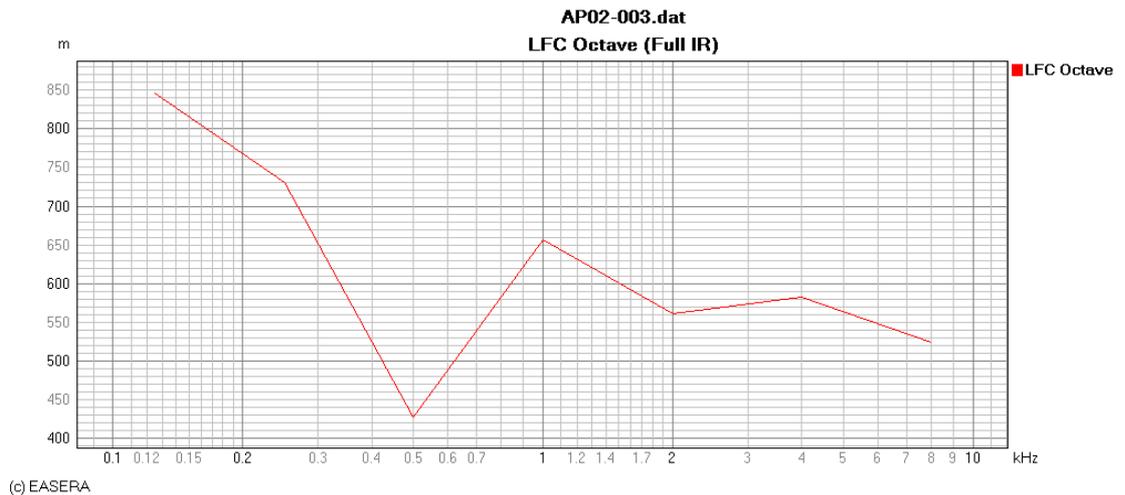
Though a subjective assessment is not known explicitly, it can be assumed, however, that the ranges mentioned under 5.2.2 for LE and LF are by approximation applicable also to LFC.

### 5.3.3. Calculation of LFC in EASERA

Calculation of the IACC of the octave-frequency-filtered IR is done in EASERA using the function **Calculation/Spatial Measures/LF Coefficient (Octave)** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



The graph shows LFC as a function of the midband frequency of the octave-filter.



## 5.4. Echo Criterion $EK_{\text{music}}$ according to DIETSCH [8]

### 5.4.1. Measurement Basis

See clause 3.6: Echo Criterion  $EK_{\text{speech}}$ ,

### 5.4.2. Subjective Assessment of the Echo Criterion $EK_{\text{music}}$ and its Frequency Dependence

An echo occurs when  $EK_{\text{music,max}} > EK_{\text{music,limit}}$ .

For musical performances use these values:

$$\Delta\tau_E = 14 \text{ ms}$$

$$n = 1$$

Mozart (most critical motive)

$$EK_{\text{music,limit}} = 1.8 \text{ to } 2$$

Piano music  
 "Sustained" music (Wagner)

$EK_{\text{music,limit}} = 3$   
 $EK_{\text{music,limit}} = 7$

If  $EK_{\text{max}} > EK_{\text{limit}}$  occurs periodically (periodicity 80 to 100 ms with music), a **flutter echo** becomes audible.

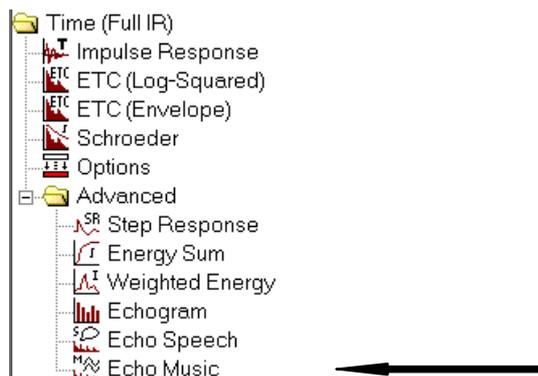
With band-limited evaluation of the room impulse responses one has to keep in mind that especially the **high-frequency** signal components tend to cause echo disturbances.

According to DIETSCH [8] it is, however, sufficient to employ:

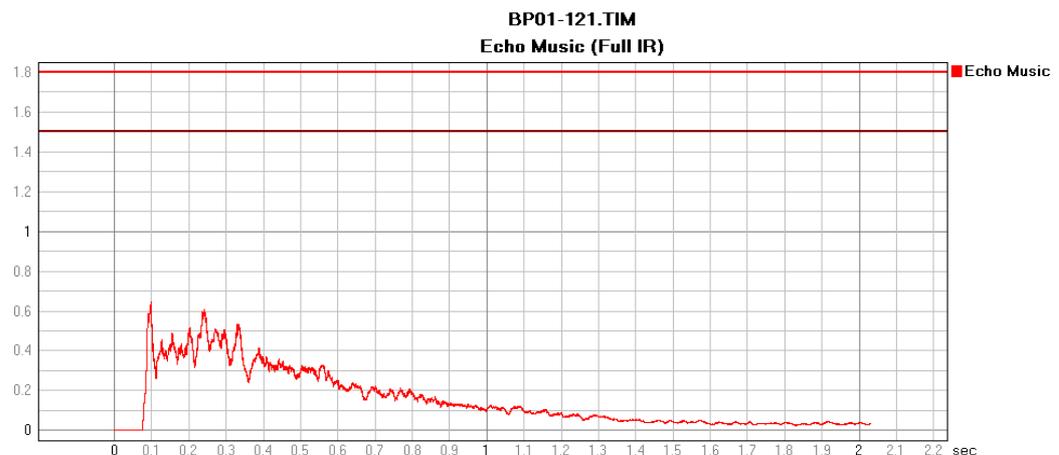
- for **Music**: Test signals with a bandwidth of two octaves and a midband frequency of  $f_M = 1.4$  kHz.

### 5.4.3. Calculation of $EK_{\text{music}}$ in EASERA

Calculation of the of the  $EK_{\text{music}}$  –graphs from the room impulse response is done in EASERA using the function **Time (Full IR)/Advanced/Echo Music** selected from the tree as shown here, or by selection of the corresponding menu item or tool bar button.



### $EK_{\text{Music}}$ –graph



(c) EASERA

A frequency assessment does not take place, but the  $EK_{\text{music}}$  –graph is calculated from the **broadband** impulse response (Full IR) by default. The octave and third-octave filtered responses can also be viewed.

## 6 Musicians' Criteria

### 6.1. Mutual Hearing (Monitoring) - Early Ensemble Level EEL according to GADE [16;17]

#### 6.1.1. Measurement Basis

According to GADE [16; 17], the energy measure EEL is considered as a measure for the "mutual hearing" among the musicians.

It is calculated by:

$$EEL = 10 \cdot \log_{10} \frac{E_{x(0..80 \text{ ms})}}{E_{5 \text{ ms}(1 \text{ m})}} \text{ dB}$$

In this case  $E_{x(0 \text{ to } 80 \text{ ms})}$  is the accumulative sound energy measured in the receiver-source position coupling (x) until 80 ms after the direct sound, referred to the direct sound energy  $E_{5 \text{ ms}(1 \text{ m})}$  measured at 1 m distance from the sound source.

#### 6.1.2. Subjective assessment of EEL and Frequency Dependence of the same

Investigations by GADE in 14 European concert halls have revealed that the EEL values at middle frequencies (500...1000 Hz) lie in the range of  $EEL = -15 \text{ dB}$  to  $-10 \text{ dB}$ . A quantitative correlation of these values to the quality of mutual hearing can not yet be established. This applies also to the frequency dependence of the EEL values.

### 6.2. Room Support ST1(2) according to GADE [16; 17]

#### 6.2.1. Measurement Basis

The energy measures ST1 and ST2 are regarded, according to GADE [11], as measures for the acoustical (subjective) perception of the musicians that the room „answers“, „carries“ or supports the playing.

They are calculated as:

$$ST1(2) = 10 \log_{10} \frac{E_{x(20..100(200) \text{ ms})}}{E_{5 \text{ ms}(1 \text{ m})}} \text{ dB}$$

In this case  $E_{x(20 \text{ to } 100 \text{ ms})}$  is the accumulative acoustic energy measured at 1 m distance from the musician's location x within the period from 20 to 100ms or 200ms after the direct sound, referred to the direct acoustic energy  $E_{5 \text{ ms}(1 \text{ m})}$  (measured at 1 m distance).

#### 6.2.2. Subjective assessment of ST1(2) and it's Frequency Dependence

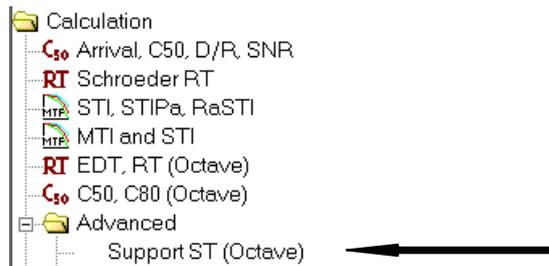
Typical ST1 values measured on the platforms of 14 European concert halls by GADE [23] lie within a range of:

$$-15 \text{ dB} < \text{ST1} < -12 \text{ dB}.$$

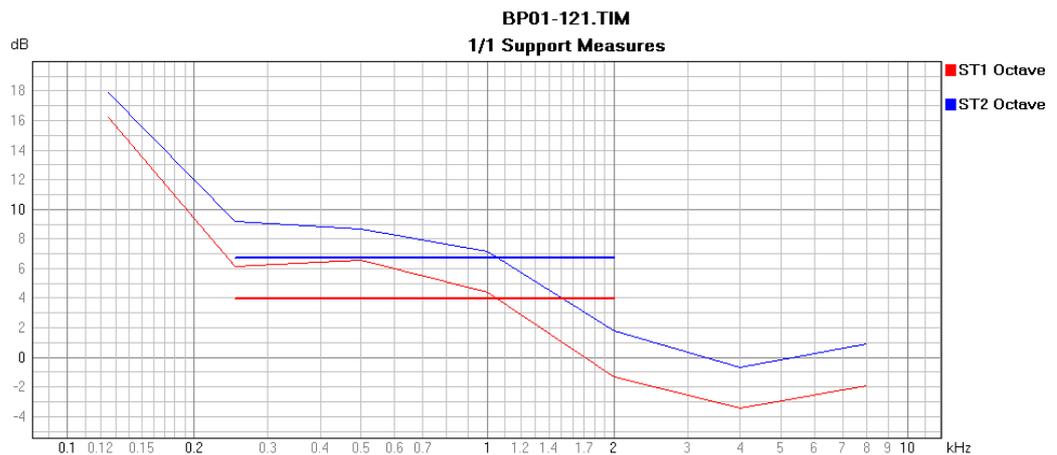
A quantitative correlation of these values to the quality of room support can not yet be established. This applies also to the frequency dependence of the ST1(2) values.

### 6.2.3. Calculation of ST1(2) in EASERA

Calculation of the of the ST1(2) –graphs from the room impulse response is done in EASERA using the function **Calculation/Advanced/Support ST (Octave)** selected from the tree as shown here.



Graphs show ST1(2) as a function of the midband frequency of the octave-filter



(c) EASERA

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