Evaluation of objective echo criteria

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June 2003

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ACOUSTICS

DIPLOMA THESIS – PROBLEM TEXT

Evaluation of objective echo criteria

" The thesis concerns evaluation of perceptibility of echoes in concert halls and auditoria for music/speech. Criteria for filtered room impulse responses from Odeon Room Acoustic Software are to be evaluated. Mainly established criteria should be evaluated, as presented by Dietsch and Kraak and others throughout the literature.

Suitable sets of impulse responses generated by Odeon are to be auralized with music/speech and used for listening tests. Correlation between listening test results and objective criteria based on the impulse responses generated are examined.

It would also be interesting to study whether input data can be simplified, e.g. reflectograms of lower order reflections only (1-3). "

Acknowledgements

This paper is a diploma thesis at the Norwegian University of Science and Technology (NTNU). The work providing the foundation for the thesis is mainly done at Multiconsult AS in Oslo. Listening tests and preliminary literature research have been conducted at NTNU. The author is presently a student at the Acoustics Group at the Faculty of Information Technology, Mathematics and Electrical Engineering.

Professor Peter Svensson and especially Magne Skålevik have given helpful advice and encouragement throughout the work with the thesis, for which I am most grateful. I would also like to thank the listening test persons for their time and flexibility, and finally Bård Støfringsdal for help with the listening test setup.

Anders Løvstad
Oslo, June 16th, 2003

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SUMMARY

Subjective perceptions of the acoustics of enclosures are based on our ability to perceive incident sound waves. Reflections can add both pleasant enhancements and disturbing effects to sound fields of various complexities. The present paper investigates properties and aspects of reflections that are percepted as separate events, as echoes. In order to be percepted as a separate event must the reflection(s) be of certain amplitude as function of separation time from other adjacent reflections or the direct sound.

The ability to predict whether echoes will arise from room impulse responses or not, would be beneficial with respect to practical and commercial use. An overview of past research on echo perception and proposed criteria is given initially. Dietsch and Kraak[2] formulated the presently most advanced criterion in 1986, based on build-up functions. The criterion is implemented and tested thoroughly.

Listening tests were performed to obtain evaluation references and for use as targets in the development of objective echo criteria for both speech and music motifs. Situations investigated can be divided into three parts: Single echo and multiple echoes in anechoic conditions, respectively, and reverberant situations. A total of four listening test sequences have been carried out. Ten test persons conducted each of the sequences. Listening test results proved consistent throughout. The most noticeable feature observed from the anechoic listening tests is that no disturbing echoes were reported in the listening tests for delays shorter than 30 [ms] for speech and 100 [ms] for music. Higher scattering coefficient of back and sidewalls were found to reduce echo disturbances significantly.

Dietsch and Kraak`s echo criterion has been implemented in two versions, one based on the image source method (ISM) only, while the other on the full length wave-file output generated using both ISM and ray tracing. Both implementations gave unsatisfactory results, mainly caused by the short, rectangular evaluation window employed and insufficiently echo critical test motifs. The most echo critical music motif used in the present paper proved far more critical. Neither is a lower echo delay limit given by Dietsch and Kraak, which is clearly erroneous from the results obtained from the listening tests.

A new criterion is proposed, based on the convolution between energy room impulse responses and a Hanning window simulating the integration properties of the ear. Results obtained from the listening tests verified an integration time of 50 [ms]. An evaluation curve is calculated based on the hearing threshold, and the reverberation curve of the ear and the room to be evaluated, respectively. Consequently is the criterion only valid in reverberant conditions, with a lower room reverberation time limit equal to the reverberation time of the ear (0,4 [s]).

Results are good. A correspondence between calculated values and listening tests results of 100 [%] in a simulated auditorium with speech, and 81 [%] in a concert hall with music test motifs are achieved. Possible improvements are suggested to obtain even better accordance. However, a larger set of reverberant situations must be tested in order to fully validate the criterion.

1 INTRODUCTION

Human perception of sound is complex and depends on several parameters. This paper puts the main focus on echoes as source of nuisance to listeners. The most important parameters in our perception of reflections are delay and amplitude relative to one another and the noise floor. Also the nature of the sound source(s) is critical in our subjective opinion of a sound field.

The easiest and most fundamental case is with one reflection in addition to the direct sound. A reflection that is perceived may not reach the consciousness of the listener if the amplitude is sufficiently low. Increasing the amplitude will cause the reflection to enter the consciousness of the listener. At sufficiently high amplitudes as function of delay relative to the direct sound, is the reflection percepted as a separate event. It becomes an echo. Kuttruff[1] defines echo as "*any sound reflection that is subjectively noticeable as a temporal or spatially separated repetition of the original signal*".

Analyses become more complex as increasing number of reflections are added. Depending on the parameters of the reflections, different impressions arise. If separations between adjacent reflections are adequately short will they be perceived as one impulse due to the inertia of the ears. This phenomenon can be regarded as a simple case of masking. In even more complex cases, such as in a reverberation field, is masking of utmost importance in establishing limits for echo disturbance.

Listening tests need to be performed in order to obtain subjective reference levels, which is to be done both in anechoic and reverberant conditions for a variety of test motifs. As the different samples in the listening tests are reproduced through headphones, must the generated impulse responses be passed through a set of head related transfer functions (HRTF`s).

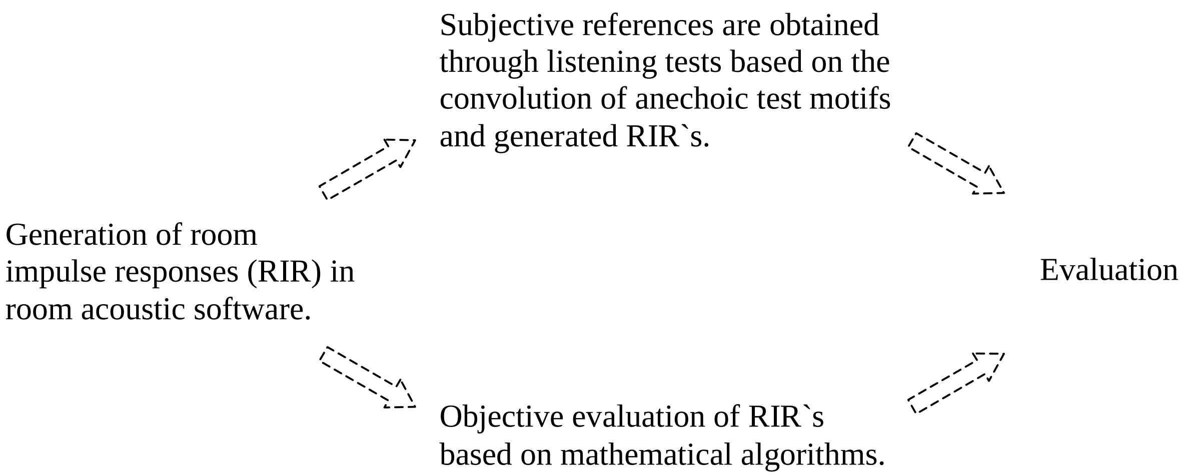

Figure 1: Principle of evaluation of objective criteria based on subjective references.

The principle used in evaluating objective criteria is depicted in Figure 1. Objective algorithms are compared and evaluated with regard to subjectively obtained reference levels. This scheme is adapted in the present paper.

Numerous contributions have been made to enlighten different aspects of our perception and nuisance due to echoes. Dietsch and Kraak[2] put the presently most advanced objective criterion forward in an article published in 1986, which is based on a build-up function iteratively deduced from the center time proposed by Kürer[3]. The criterion is implemented and evaluated in the present paper based on computer simulated room impulse responses (RIR`s) generated by the Odeon room acoustic software.

A new criterion is developed and proposed based on the convolution of energy RIR`s with a Hanning window simulating the integration properties of the ear.

2 THEORY

2.1 Introductory remarks

The next section gives a historic background summary of the most important work on echo perception and on overview of the criteria presently formulated in the literature. The succeeding section discusses and clarifies calculation techniques employed by the computer simulation tool used. Approximations and limitations induced by the complexity of sound fields and limited computational power are explained. Finally, filtering properties of head related transfer functions (HRTF`s) used in headphone reproduction are discussed.

Detailed discussions of other parameters, concepts and notions than the ones mentioned in the previous paragraph are not included. For further theoretical background, see Kuttruff[1].

2.2 Background and previous work

All sound fields consist of a certain set of incident sound waves. Every sound wave except the direct sound are redirected in some manner from other surfaces before arriving at a given receiver position. This path difference, and consequently time gap, between the direct sound and later arriving reflections alter the acoustical impression of sound fields. The total number of reflections as function of arrival time and amplitude forms the room impulse response (RIR). Rooms put their signature on the sound generated by the source.

Preceding reflections can mask later arriving reflections, as mentioned above. A numerical value for the threshold of masking was first given by Petzold[4] as

$$t = 0{,}05 \pm 0{,}01\,[\mathrm{s}] \tag{1}$$

In 1951 Helmut Haas published a famous article investigating the influence of a single artificial echo as a function of a given set of parameters on the audibility of speech[5]. Loudspeakers were used in open air (on the roof of a building), as no damped rooms were available at the time. A baffle board eliminated possible errors from floor reflections. He found that for short delays with equal strength no separate echo source could be detected before it was delayed approximately 40 [ms]. A delay of only 1 [ms] moved the virtual source from center position towards the leading speaker. In the range 1-30 [ms] the echo source was not perceived as a separate source rather a pleasant enhancement and modification of the quality of the sound (later known as spaciousness). Haas found that echo levels must be 10 [dB] stronger than the direct sound in the delay range of 5-30 [ms] if equal loudness should be achieved for speech of average speed.

Increasing the delay dynamics to 0-160 [ms] gave rise to perception of separate echoes depending on certain parameter values. Increasing speed of speech lowered thresholds and increased echo disturbance. 50 [%] disturbance was reported at a delay of 68 [ms] at average speed (5,3 syllables/sec). This speech rate was used in the following experiments. Echo intensities influenced disturbance severely as well. An echo level reduction of 10 [dB] completely removed disturbance effects, an attenuation of 5 [dB] doubled the critical time difference, while an increase of 10 [dB] only reduced the threshold from 68 to 60 [ms], as can be seen from Figure 2.

Higher frequencies were found to cause greater subjective disturbance than lower frequencies. Attenuation of lower frequencies resulted in no noticeable reduction of nuisance. Loudness did not influence thresholds, according to Haas` experiments. Moving into reverberant rooms, the influence of the reverberation time was investigated. Longer reverberation times was found to increase the critical delay difference, as the slower decaying sound level masks later reflections. Haas used a reverberation time of 0,8 [s] in the following listening tests. The next parameter to be evaluated was direction of incidence. Haas found that it did not affect echo thresholds provided that the direct sound was incident from the front. However, he reported that Stumpp[6] found the critical delay time to be 80 [ms] with direct sound and echo coming from the same lateral (90 degrees altered horizontally) for speech, but only 50 [ms] when they arrived from opposite sides.

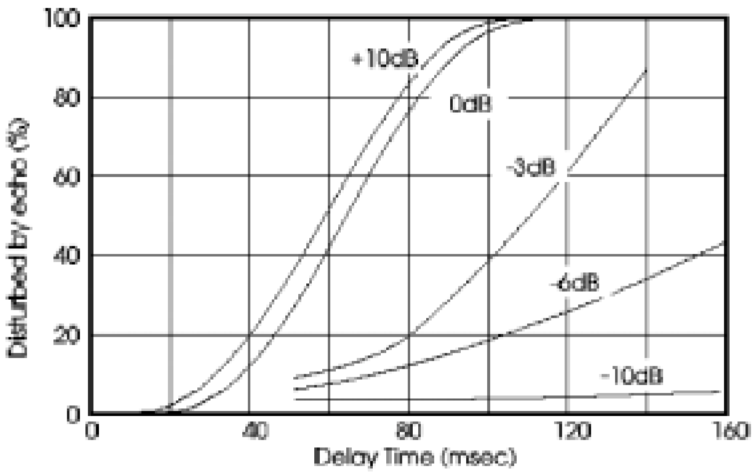

Figure 2: Echo disturbance as function of intensity of reflected sound (after Haas[5]). Curve numbers show the difference between direct sound and echo in [dB].

Bolt and Doak[7] formulated a tentative echo criterion based on the Haas` results as curves of amplitude vs. time delay of each reflection relative to the direct sound. Curves for different percent of listeners being disturbed were presented. Figure 3 shows an example response with threshold curves for 10 and 50 [%] overlaid.

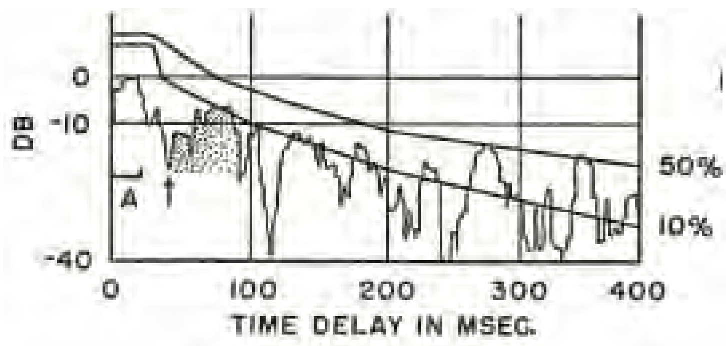

Figure 3: Example room impulse response with echo criterion curves of Bolt and Doak[7] overlaid.

Disturbance contours were proposed connected to reverberation times. Dietsch and Kraak[2] rightfully pointed out that this criterion does not account for the effects of possible reflections arriving before the disturbing echo. The ear integrates the incident energy, so it is not necessarily only one reflection that causes the subjectively perceived echo. It is also inferred that the criterion only is valid in the room in which the reference measurements originate.

Muncey, Nickson and Dubout[8] continued the pioneering work of Haas[5] to include musical sounds and investigated how this would affect the criterion proposed by Bolt and Doak[7]. They used a reverberation time of only 0,15 [s]. Results showed that limits for disturbance were lower owing to the decreased reverberation time for speech. Music was found to be less critical to echoes than speech. Music with fast tempo was more critical than slow tempo pieces. Echo level curves as function of time delay were deduced (Figure 4). The results were reported to be in good agreement with pulse work in motion picture theatres.

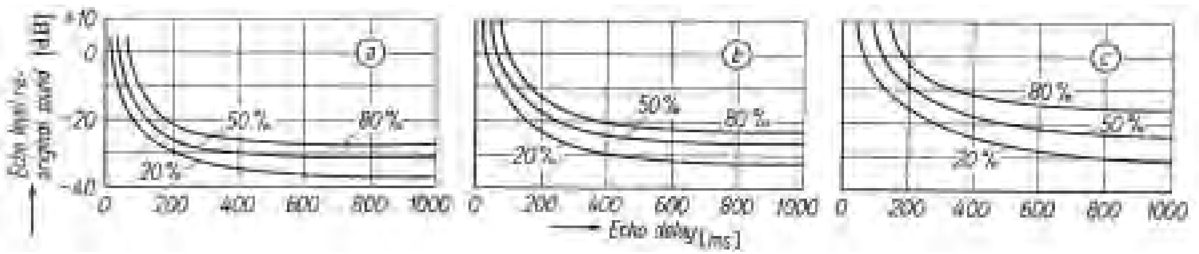

Figure 4: Acceptable echoes for various percentages of listeners disturbed in a non-reverberant room (after Muncey, Nickson and Dubout[8]).
(a) Speech – fast tempo
(b) String music – fast tempo
(c) Organ music – fast tempo

A succeeding article by Muncey, Nickson and Dubout[9] showed that the acceptable echo intensity levels as function of delay time of artificial echoes could be divided into three parts. The first part was associated with Haas` precedence effect. Decay time was reported to determine the echo level in the second period, while in the third section the acceptable echo level was found to be almost constant with increasing delay. Natural echoes in the recording room and the difference in sound level between the loud and quiet parts of the sound material thus determined the two latter periods, respectively. The second part proved close relation between measured reverberation times and reverberation values obtained from fitting the tangent of the initial echo level decay with increasing time delay for some samples. However, with increasing time delay, perceptible echo levels were significantly higher than the linear reverberation decay lines. The authors concluded that attempts "*to translate results of listening tests into criteria for examination of pulse measurements from rooms would produce merely a reflection of the test conditions and not be universally accurate*".

Dubout[10] later formulated a criterion based on correlograms of signal and echo levels. The correlograms showed the attenuation needed to suppress echoes to the threshold of perceptibility, as a function of echo delay. Figure 5 shows an example correlogram. Subjective thresholds obtained experimentally were compared to the correlograms. Similar trends appeared, such as lowered thresholds when reverberation was added to the echo, and raised thresholds when reverberation was added to the signal. At delays lower than 150 [ms] the correlograms showed lower thresholds than the subjective results throughout (Figure 5).

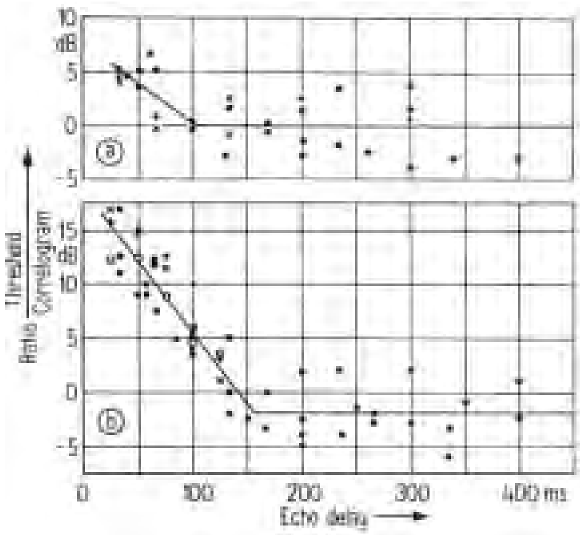

Figure 5: **Ratio of echo threshold to its corresponding correlogram (after Dubout[10]). Solid lines fitted using least squares method.**
(a) Reverberation (0,7 [s]) added to signal.
(b) Reverberation added to echo, and no reverberation added.

This divergence was reported being due to the choice of decay in the correlograms. If the equipment would have more rapid decays than the aural one, divergences will occur. Selecting the decay of the comparator more aural-like would give better agreement.

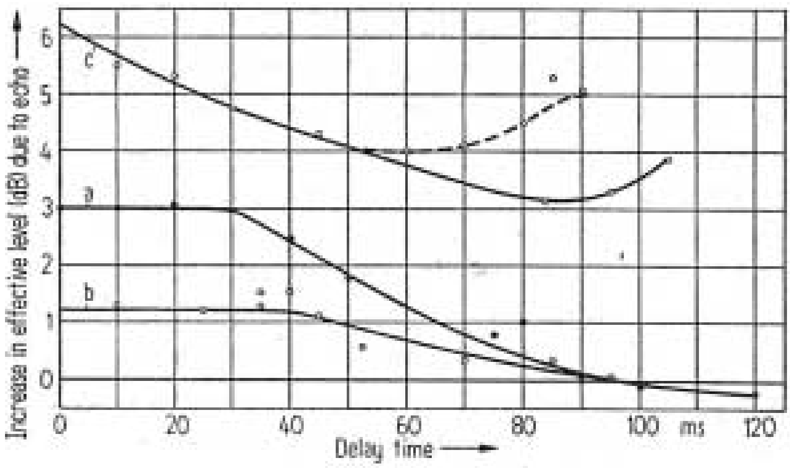

Figure 6: **Integration curves for a single echo (after Lochner and Burger[11]) of**
(a) Equal same intensity as the primary sound
(b) 5 [dB] below the primary sound level
(c) 5 [dB] above the primary sound level

Lochner and Burger[11] performed tests similar to the ones done by Haas[1] in determining thresholds of perception and of equal loudness of echoes delayed 0-120 [ms] using speech and pulsed tones, except these test were performed in a neutral anechoic room. Equal loudness curves show the masking effect of the direct sound upon later arriving reflections, even though a significant part of the total energy is contained in the later reflections. This forms the basis of the second part of the article, in which articulation tests were performed to determine the integration properties of the human ear. By finding the horizontal shift of percentage articulation as function of speech level re. hearing threshold for a given delay compared with zero delay, the integrating properties of the ear could be deduced. This was done with +5, 0 and –5 [dB] levels, and the results are reproduced in Figure 6. With 0 [dB] perfect echo integration was found up to 30 [ms], up to 40 [ms] with –5 [dB], while integration decreases quickly with +5 [dB]. The latter mainly caused by the fact that the observers concentrate on the stronger echo as delay time increase above 20 [ms]. These results were also tested with multiple echoes (three) and fairly good resemblance was found between calculated and measured values.

So far all of the criteria were based on comparison with echo threshold curves. In 1961 Niese[12] proposed a more mathematical criterion, called "Echograd", based on the ratio of signal to noise energy. The criterion is only valid for speech.

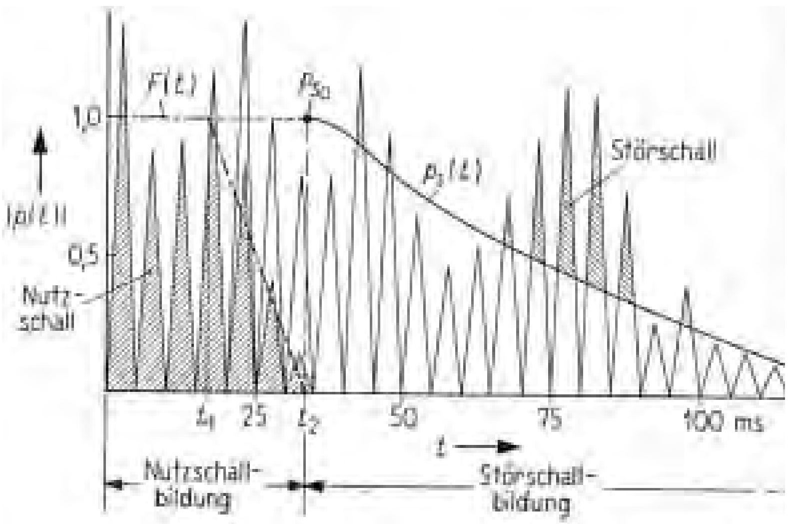

Figure 7: Echograd computation limits (after Niese[12]).

Two weighting functions are used to isolate the so-called useful sound and noise, respectively. The Echograd is given by

$$\varepsilon = \frac{S}{N+S} \tag{2}$$

where S is the noise energy (Störschall) given by

$$S = \int_{t_2}^{\infty} [|p(t)| - p_s(t)]^2 \, dt \tag{3}$$

provided that $|p(t)| - p_s(t) \geq 0$. $p_s(t)$ is a function of the reverberation time of the room. N is the useful energy (Nutzschall) given by

$$N = \int_{t=0}^{t_2} [F(t)|p(t)|]^2 \, dt \tag{4}$$

where *F(t)* is a simple weighting function being 1 from 0-17 [ms] linearly decaying to zero from 16-33 [ms]. Figure 7 shows an example in which these limits and measures are exemplified. The Echograd is given in percent. Considerable simplifications were done to construct the measurement apparatus used in the examples given in the article. No subjective tests were performed to achieve a subjective annoyance reference to evaluate the Echograd results. Dietsch and Kraak[2] infer that the criterion is developed for syllable perceptibility of speech, not for detection of discrete echo disturbances.

Santon[13] used a similar partition into useful and disturbing energy to predict echograms and speech intelligibility. Useful energy was defined as all contributions below the threshold of separate echo perception, while disturbing energy was the portion exceeding the perception value. The method takes the directional distribution of echoes into account. Numerical computations were done and energy accumulation curves were compared using ISM and ray tracing (see Section 2.4). The latter was also compared with measurements done in a given conference room. Good agreement was reported throughout.

Barron[14] investigated the effect of early lateral reflections using music. Subjective effects originating from a single side reflection were explored, resulting in observed effects on level, localization, tone colouration, echo disturbance and spatial impression as function of delay and reflection levels (Figure 8). Meyer and Kuhl[15] have in 1951 observed that the degree of disturbance of delays in the range 50<t<100 [ms] would be much reduced by adding a preceding reflection.

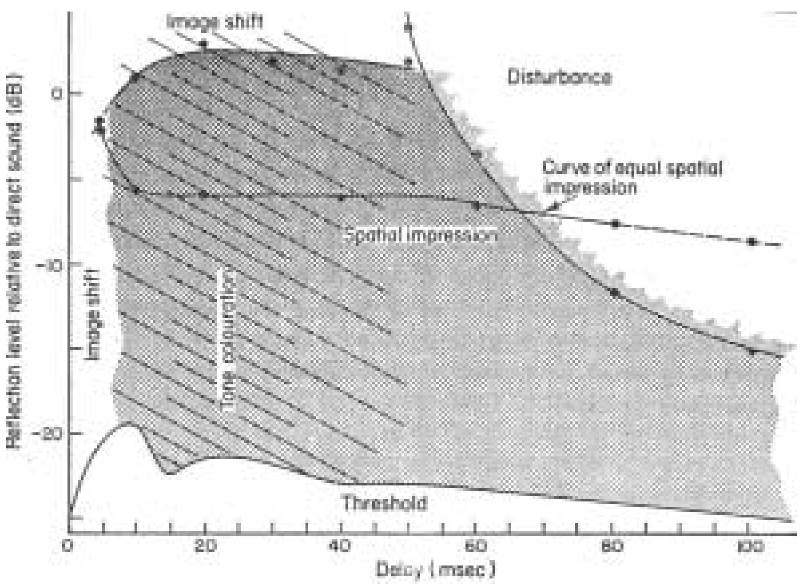

Figure 8: **Subjective effects of a single side reflection (α=40°) of variable delay and level using music (after Barron[14]).**

Spatial impression was in Barron's study subjectively determined as the only positive effect of early reflection (other than loudness). He found that delay was "*relatively unimportant for reflections delayed between 10 and 80 [ms]*" relative to the direct sound. Ceiling reflections did not produce the same effect, but only tone coloration. Delay of single side reflections proved to be the principle determinant of spatial impression. The degree of spatial impression for multiple reflections related to the powers of lateral reflections, and could be measured by the ratio of lateral to non-lateral energy arriving at t<80 [ms].

Ando[16] also investigated the preferred delay gap between the direct sound and the first reflection and the preferred angle of incidence. He used a long-time autocorrelation function (ACF) and the interaural cross correlation (IACC) of the sound field. The preferred delay was found by coherence between ACF and the amplitude of the echo reflection. The preferred angle of incidence was found to be 55 degrees by minimizing the IACC. Four different music motifs were used.

Rakerd, Hartmann and Hsu[17] investigated echo suppression in the horizontal and median sagittal planes for continuous speech. Listening and correlation tests were performed for both echo and masked thresholds in the horizontal and median sagittal planes. Differences are detected by interaural level and time differences in the horizontal plane, and through spectral properties in the median sagittal plane. Thresholds were found to be comparable for the two cases, especially with respect to delay time dependence; the masked thresholds being 8-15 [dB] lower throughout. In the median sagittal plane both cases showed a weakness for overhead location, relative to front and back.

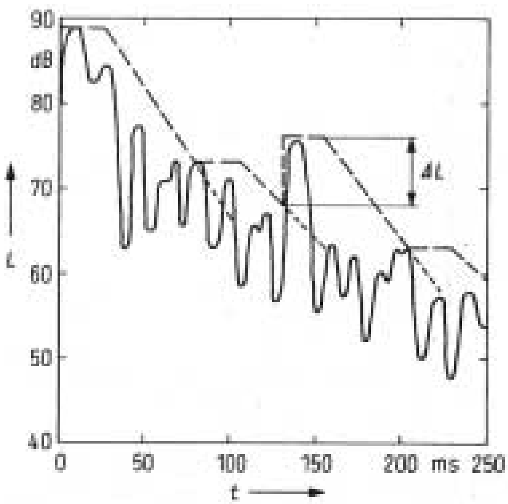

Figure 9: Example of the Yamamoto[18] echo criterion. Stippled line is the evaluation curve (after Dietsch and Kraak[2]).

Dietsch and Kraak[2] refer to an echo criterion formulated by Yamamoto[18] where the room impulse response is compared with an evaluation curve. The evaluation curve will remain at a constant level for a certain time after the initial direct sound peak to simulate the masking effect. The evaluation curve then falls off with a certain gradient, until it hits the room impulse response, and the sequence repeats itself (Figure 9). If the evaluation curve hits the room impulse response below a peak, the difference ΔL between evaluation curve to the peak are compared to a set of echo critical threshold values. Yamamoto used 10 [ms] long bursts as excitation signal, whether the method is valid for speech or music is not verified.

2.3 Echo criterion formulated by Dietsch and Kraak

Dietsch and Kraak published an article proposing an objective echo criterion for both music and speech in 1986[2]. Subjective judgments were compared with the respective impulse responses in both synthetic sound fields and measured ones. The present paper evaluates the criterion based on computer simulated sound fields and impulse responses generated.

The relation $\left|p_v\right|^n t_v$ is assumed constant throughout the deduction of the criterion, where p_v is the amplitude of the sound pressure, t_v the delay time and n a weighting exponent. Kürer[3] formulated a build-up function defined as the first moment of the squared impulse response (center time). Exchanging the upper integration limit from infinity to a variable one yield

$$t_s = \frac{\int_{t=0}^{\tau} t\left|p(t)\right|^n dt}{\int_{t=0}^{\tau} \left|p(t)\right|^n dt} \tag{5}$$

where the exponent n equals 2 in the traditional center time formula. *The* final criterion is given by the ratio of increase of energy to a given time interval, $\Delta t_S(\tau)/\Delta\tau$ in Figure 10. Comparing the maximum value of this ratio to computed threshold values completes the criterion.

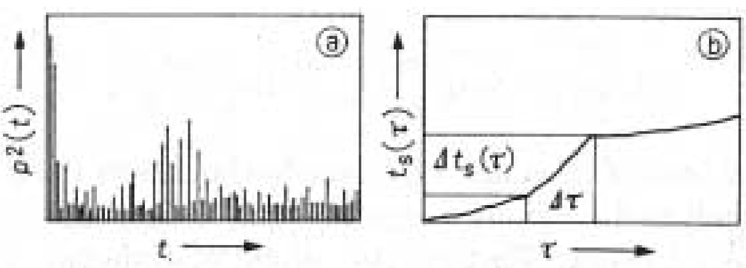

Figure 10: An impulse response (a) and its corresponding build-up plot with a given $\Delta\tau$ (b) (after Dietsch and Kraak[2]).

An echo will appear if Δt_S would be above the echo threshold Δt_{SE} in a given period $\Delta\tau$. Using a single time delayed echo forms the mathematical basis in achieving threshold values of Δt_S for speech and music. The direct sound and the echo reflection are then given by $p_D = \hat{p}_D f(t)$ and $p_E = \hat{p}_E f(t - t_v)$, respectively. The increase of Δt_S due to p_E then becomes

$$\Delta t_s = \frac{\int_{t=0}^{\infty} t\left|p_E(t)\right|^n dt}{\int_{t=0}^{\infty}\left|p_D(t)\right|^n dt + \int_{t=0}^{\infty}\left|p_E(t)\right|^n dt} = \frac{\widehat{p}_E^{\,n} t_v}{\widehat{p}_E^{\,n} + \widehat{p}_D^{\,n}} \qquad (6)$$

Threshold values were found from the results from the subjective tests for the different motifs when the ratio between direct and echo reflections equals zero. Equating the ratio p_D *to* p_E to zero gives

$$\begin{aligned} \Delta L_E(t_V) &= 20\log\frac{\hat{p}_E}{\hat{p}_D} = \frac{20}{n}\log\frac{\Delta t_S}{t_V - \Delta t_S} = 0 \\ \log(t_V - \Delta t_S) &= \log(\Delta t_S) \\ \Delta t_{SE} &= \tfrac{1}{2} t_V \end{aligned} \qquad (7)$$

Threshold curves for the different test motifs used are plotted in Figure 11. The value from the most rhythmic motif was chosen for music (Δt_{SE} = ca. 25 [ms]) shown by filled triangles (▲), while × represents the one motif for speech. Choice of exponent n was done by iteration. n = 1 was chosen for music, while n = 2/3 and Δt_{SE} = ca. 9 [ms] was chosen for speech.

Dietsch and Kraak commented being satisfied by being on "the secure side" for time delays below $t_{VE} \approx 60$ [ms] when choosing n = 2/3 for speech, which in practice means that the criterion will exceed the echo threshold more often than the subjective tests. Figure 12 shows the fit done with speech.

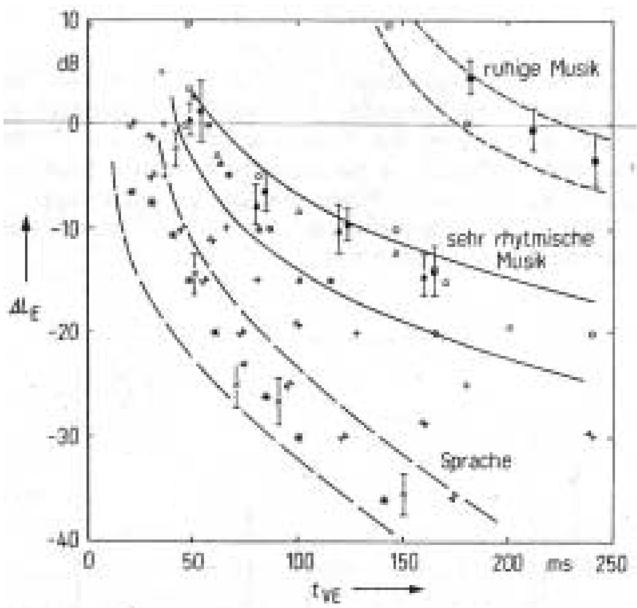

Figure 11: Level difference as function of time delay (after Dietsch and Kraak[2]).

The critical period $\Delta\tau$, in which the perception of a reflection or a set of reflections must grow at least Δt_{SE} to be considered an echo, was also determined using impulse responses and center time. Whether the echo was caused by a single strong reflections or a set of reflections should be indifferent. Dietsch and Kraak used the following formula[19] to calculate the reverberation sound pressure as a function of time

$$\left|p_N(t)\right|^n = \left|p_{NI}\right|^n e^{-\frac{6{,}9nt}{T}} \tag{8}$$

where p_{NI} is the amplitude of the reverberation at the end of the impulse response, and T is the reverberation time.

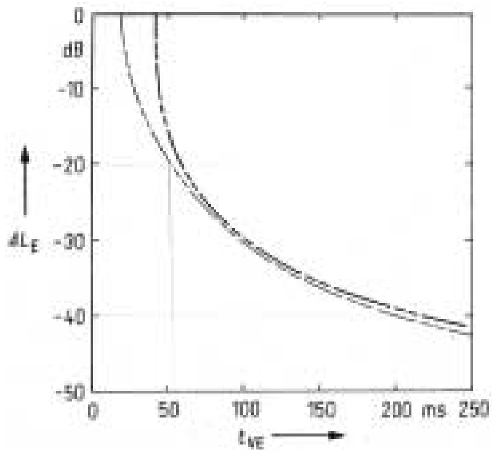

Figure 12: Comparison between subjective listening results (upper) and calculated threshold (lower) (after Dietsch and Kraak[2]).

Inserting equation (8) into equation (5) gives

$$t_s = \frac{\int_{t=\tau_N}^{\tau} t\left|p_N(t)\right|^n dt}{\int_{t=\tau_N}^{\tau} \left|p_N(t)\right|^n dt} \tag{9}$$

which was used to find numerical values for $\Delta\tau_E$ for the different motifs. Values were independently of the reverberation time found to be $\Delta\tau_E$ = 9 [ms] for speech and $\Delta\tau_E$ = 14 [ms] for the most critical music motif. The final echo criterion was then given by the ratio

$$EK(\tau) = \frac{\Delta t_S(\tau)}{\Delta\tau_E} = \frac{t_S(\tau) - t_S(\tau - \tau_E)}{\Delta\tau_E} \tag{10}$$

as shown in Figure 13. Table 1 sums up the limits found for both music and speech.

Motif	***n***	$\Delta\tau_E$	***EK*** **(10 [%])**	***EK*** **(50 [%])**	**Bandwidth of test signal**
Speech	2/3	9 [ms]	0,9	1,0	700-1400 [Hz]
Music	1	14 [ms]	1,5	1,8	700-2800 [Hz]

Table 1: Echo threshold from Dietsch and Kraak`s echo criterion

Minimum separation between multiple echoes were also given as $\Delta\tau_{Emin}$ = 50 [ms] for speech and $\Delta\tau_{Emin}$ = 80 [ms] for music. Flutter echoes will arise if echo thresholds are broken regularly for $\Delta\tau_E > \Delta\tau_{Emin}$.

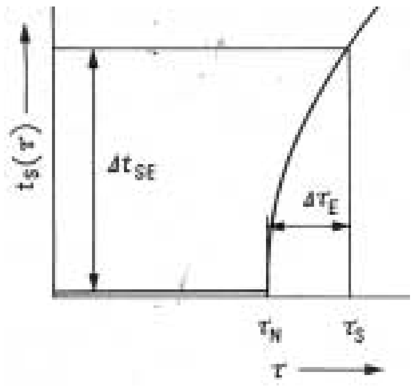

Figure 13: Illustration of the method used to determine $\Delta\tau_E$ (after Dietsch and Kraak[2]).

2.4 Calculation techniques

In the calculation of any room acoustic response of some complexity, approximations must be used. Calculation methods exist both in the frequency and time domain. Only the methods implemented in Odeon Room Acoustic Software[20],[21], which is the simulation program used in the current work, will be investigated below. Odeon employs both the image source method (ISM) and the ray tracing technique, both of which are based on geometrical acoustics.

2.4.1 The Image Source Method (ISM)

All geometrical acoustic models use the concept of sound rays instead of sound waves in their description of sound propagation. The image source method (ISM) is based on geometrical acoustics. Replacing any planes or surfaces that will cause a reflection of an incident sound ray by a virtual source is the foundation of ISM (see Figure 14). This virtual source is called an image source. A right angle is shown in the figure, but if the edge is not perpendicular, the angle of the reflected ray will be twice the angle of the incident ray. Every wall generates one image source, after which every image source must be checked for its visibility. All areas are not visible for an image source, as shown in the upper part of Figure 14. Every visible image source generates a reflection. If a receiver is situated in the area outside the visibility limits of a source, it will not receive an impulse. The sound ray generated by an image source can only be seen within the reflection limits of the same wall that connects it with the preceding (image) source.

As multiple reflections occur, image sources of higher order are constructed. The subscripts in Figure 14 denote the order of the image sources. It is intuitive that as rays propagate and reflect, their amplitudes are decreased depending on the absorption of the reflecting surfaces. The amplitude of an image source is given by

$$p_i^2 = P \cdot \frac{\rho_0 c}{4\pi r_i^2} DF_i \cdot \prod_1^j (1-\alpha_j) \tag{11}$$

where *DF* is the directivity factor in direction *i*, α_j the absorption coefficient of reflection *j*, and r_j the distance to the receiver.

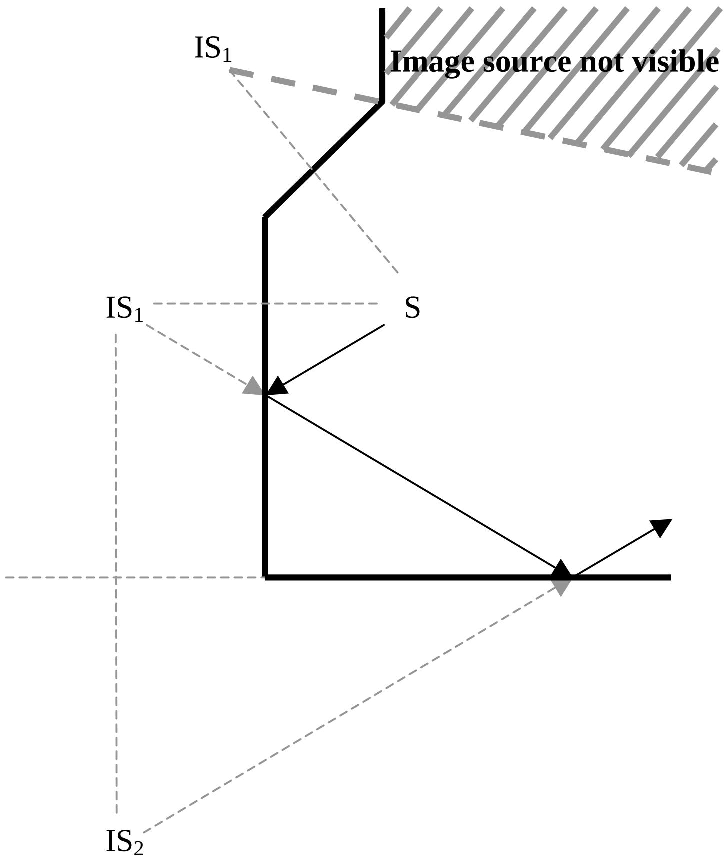

Figure 14: Concept of the image source method (ISM).

The principle is simple and iterative, but the number of image sources will increase exponentially with the number of reflections. Practical implementations using geometrical acoustics normally convert to another method called ray tracing after 6-8 orders at most, as the number of image sources increase by

$$N = N\frac{(N-1)^{i_0} - 1}{N-2} \qquad (12)$$

where N is the number of image source and i_0 sets the order.

ISM does not handle curved surfaces or edge diffraction. Some room acoustic software programs employ an approximate technique called “edge diffusion”[31],[32] to simulate edge diffraction effects. This technique puts Lambert diffusive sources[1], Section 4.5 within a quarter of a wavelength from edges and transforms specular reflections to diffuse ones. Edge diffusion will not compute the correct sound field in shadow zones, but simulates edge diffraction by reducing the specular reflection and give rise to a more diffuse sound field in the vicinity of the edge.

In particular, Odeon employs scattering by assigning diffusion coefficients ranging from 0 to 1 to each surface, determining the part of the proportion of specularly reflected energy. Lambert diffusion is optional, and the number of early scattered rays in ISM can be chosen manually.

2.4.2 Ray tracing

Ray tracing is also based on geometrical acoustics, as it relies on specular and diffuse reflections. The first authors to use ray tracing in a digital computer simulation to concert hall acoustics were Krokstad et al.[22] in 1968. Sources release numerous rays in all directions at a certain time, and as the rays propagate and are reflected by different walls, they lose a certain proportion of their energy. There are two ways in which this has been implemented. In the first one, each ray is multiplied with the absorption factor of the given wall (1-α) it hits. When the energy of a ray has fallen below a certain threshold value, the next ray is traced. The second way is to consider the absorption factor as the probability of a reflection occurring, i.e. either the ray is reflected in a perfect specular manner, or it is not reflected at all.

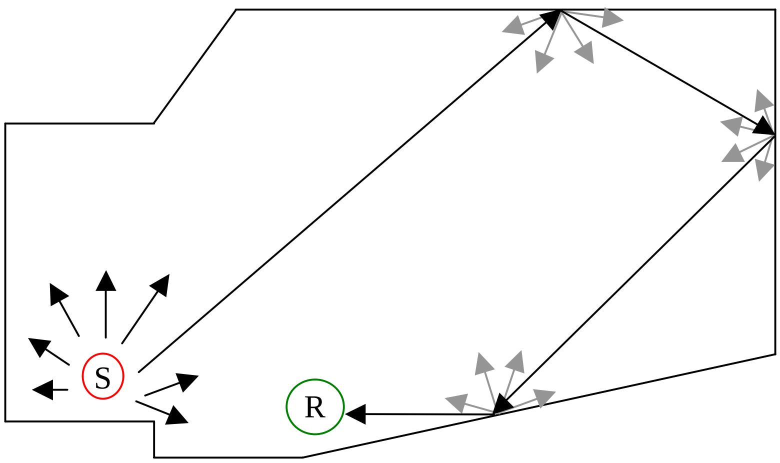

Figure 15: Principle of ray tracing.

Reflections can be both specular and diffuse, and the method also handles curved surfaces. As in ISM, edge diffraction is not accounted for. Another limitation is that the rays diverge. Depending on the number of rays emitted, the distance between them become larger as they propagate, and they start to miss small walls and even receivers. This calls for a large amount of rays. The resulting rays to arrive at receiver position are calculated by a sphere being an area or a volume. Whenever a ray crossed the sphere, its energy and time delay are stored. As all the rays have been calculated or lost, the energy is added up, and an impulse response can be constructed. The procedure is shown in Figure 15, where *S* denotes source, *R* receiver, grey arrows diffuse reflections and black arrows specular reflections, respectively. A similar method to ray tracing is cone tracing, where rays carry a circular area that increase as the carrying ray diverge. Receivers can thus be modelled as a point.

2.5 Auralization and binaural reproduction

The concept of auralization is based on the idea that reproduction of the correct sound impression of a sound field can be done by reproducing the sound field at the two eardrums. Kleiner et. al.[23] first introduced auralization, and proposed the following definition: "*Auralization is the process of rendering audible, by physical or mathematical modeling, the sound field of a source in a space, in such a way as to simulate the binaural experience at a given position in the modeled space.*" It is in other words the process were predicted room impulse responses (RIR) are converted to binaural impulse responses. Hammershøi[24] have shown that the sound pressure can be measured at the ear canal entrance plane if the ear canal is blocked. This is based on the assumption that the transmission through the ear canal is independent of the angle of incidence of the sound waves, a simplification that reduces the computation time and data significantly.

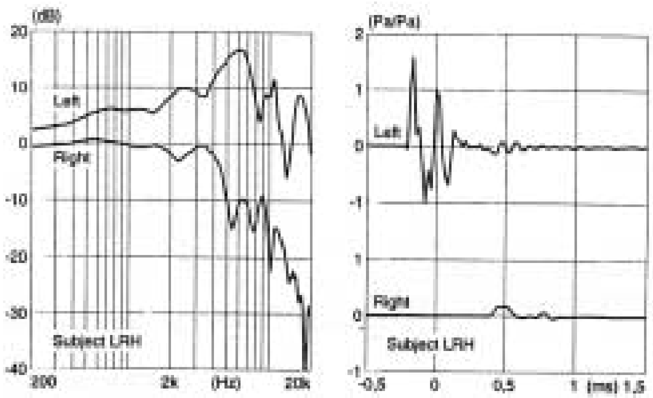

Figure 16: Example of head related transfer function (HRTF) in the frequency domain (left) and in the time domain (right) (after Møller et al.[26]).

In order to obtain binaural impulse responses, one needs a set of head related transfer functions (HRTF`s). The HRTF`s describe the effects of a persons head and shoulders on an arriving impulse response in free field. Our perception of sound is based on the analysis of time-, level- and spectral differences, all of which are governed by the HRTF set if recorded perfectly. As head shapes differ from person to person, perfect reproduction is not presently possible, but good approximations can be made through recordings done with an artificial head. HRTF`s are given by

$$HRTF(\phi,\theta) = \frac{sound \quad pressure \quad at \quad the \quad ear \quad of \quad the \quad listener}{sound \quad pressure \quad at \quad head \quad center \quad pos. \quad (listener \quad absent)}(\phi,\theta) \qquad (13)$$

where (ϕ,θ) indicate the incidence angle of the sound wave. A complete set of HRTF`s include ratios for both left and right ear, $HRTF_{left}(\phi,\theta)$ and $HRTF_{right}(\phi,\theta)$. An example of a HRTF is shown in Figure 16. Clear differences can be seen between the ears, both in time, amplitude and frequency content.

Gardner and Martin[25] recorded a set of 710 different positions at elevations from -40° to +90° degrees. They also measured were the impulse response of the speaker in free field and several headphones placed on the artificial head. This set of HRTF`s, named KEMAR, is used in the present paper as it is contained in Odeon Acoustics.

By convolving the resulting binaural response with anechoically recorded sound, an impression of how the music would sound if replayed in the modeled room or concert hall is obtained.

3 TEST SITUATIONS AND GEOMETRIES

Evaluation of room acoustical parameters based on computer simulations become more and more common as computational power increase and the available tools develops. Several authors have investigated echo thresholds of single echoes for a variety of motifs. Most listening tests have been performed using loudspeakers as sound source. The present paper employ headphones to reproduce the listening test sequences. The test situations investigated can be divided into three parts:

1) Single echo in an anechoic environment.
2) Multiple echoes in an anechoic environment.
3) Reverberant conditions.

The first situation is the most classic one, with one single echo of certain amplitude at various delays. Increasing the number of separate echoes to three forms the basis of situation two, while the last test situation comprises echoes in a complex reverberant auditorium and in a concert hall for speech and music, respectively. In practice, conditions are more or less reverberant as exemplified in the latter situations.

Some geometric measures have been indicated in the various figures. The complete geometry can be built and seen from the .Par files included in Appendix F – CD.

3.1 Single echo in an anechoic environment

Echo threshold for single echoes in anechoic conditions is the simplest possible situation, and has consequently been investigated thoroughly (see Section 2.2). Most of these have been done using loudspeaker reproduction, as mentioned. Two different single echo situations were investigated. Two separate omni-directional sources were used to generate the direct sound and the echo, respectively, in the first case. The second case uses only one omni-directional source, and echoes are generated as reflections from a single wall in an otherwise perfectly absorbent room.

3.1.1 Single echo generated by a separate source

By using two separate sources can both delay and amplitude easily be controlled and adjusted to the wanted levels. However, the frequency content of the direct sound and echo will be the same, which deviates from what is expected if a wave propagates and reflected from walls in a room. The situation resembles loudspeaker reproduction where one speaker is delayed.

The direct sound source was set at 0° elevation and an azimuth angle of 0° (see Figure 17). Echo reflections in many practical situations come either from the front (reflected towards speaker from the back wall of auditoria) or from the back (reflected to listeners (facing forwards) from back wall in rooms). Angle dependence of reflections is not considered in this paper. The echo source was set at 180° azimuth in the horizontal plane (elevation angle of 0°). Both sources were placed at equal distance from the receiver in a virtual room with absorption coefficient of 1 on all walls (perfectly anechoic).

The delays used for speech were 18, 30, 50, 100, 150, 200 and 400 [ms]. 18 [ms] was chosen because this is the delay where the criterion proposed by Dietsch and Kraak says that the annoyance threshold is when both direct sound and echo have equal amplitudes. 400 [ms] were chosen as longest delay, during which a sound wave propagates about 1400 [m]. As speech is known to be more sensitive for echoes than music, were the 18 [ms] samples excluded and a set of 30, 50, 100, 150, 200 and 400 [ms] used.

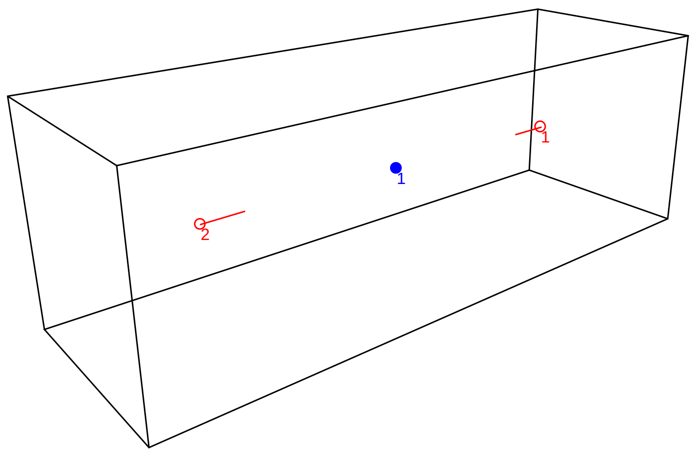

Figure 17: Single and multiple echo situation in Odeon.

Dietsch and Kraak`s criterion was also used as basis to determine the amplitudes of the echoes. Inserting the left hand equality of equation (6) into the left hand equality of equation (10) gives

$$EK(\tau)\cdot\Delta\tau_E = \frac{\int_{t=0}^{\infty} t\left|p_E(t)\right|^n dt}{\int_{t=0}^{\infty}\left|p_D(t)\right|^n dt + \int_{t=0}^{\infty}\left|p_E(t)\right|^n dt} \tag{14}$$

which for single echoes give

$$EK(\tau)\cdot\Delta\tau_E = \frac{\left|p_E\right|^n t_E}{\left|p_D\right|^n + \left|p_E\right|^n} \tag{15}$$

Inserting values from Table 1 (50 [%] annoyance) gave the limit values for the Dietsch and Kraak criterion (i.e. lowest echo amplitudes where echoes are predicted). The amplitudes were then adjusted to an appropriate range for each motif.

Due to the nature of the listening tests, discrete steps in amplitude levels of the echo must be used. Mainly 3 or 5 [dB] steps were used. An iterative process where the author adjusted the amplitudes to a certain range and then performed some preliminary tests to ensure that ranges were well adjusted was performed. The complete set of amplitudes for all delays are presented in Appendix D – Listening Tests and Calculated Results.

3.1.2 Single echo generated by a single wall reflection

The second case with single echoes was done using a single wall with fixed source position and a set of receiver positions. Only music motifs were investigated in this test. As the echo now is a reflected version of the direct sound, and no longer has a separate origin from the direct sound, the properties and position of the wall relative to the receiver are decisive factors.

Octave band	63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1k [Hz]	2k [Hz]	4k [Hz]	8k [Hz]
α	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10

Table 2: Absorption coefficient for back wall in single echo case.

An initial idea in working with this paper was to try to find information from single echo cases that can form the basis of a criterion that also is valid in reverberant conditions. It was therefore natural to use the same receiver and source positions as used in the reverberant concert hall cases described in Section 3.3.2. All surfaces were set 100 % absorbing, except the back wall which was given the absorption coefficients shown in Table 2. Two different cases were investigated, with scattering coefficient of the back wall of 0,1 and 0,7 respectively.

Receiver	x-axis [m]	y-axis [m]	z-axis [m]	Receiver	x-axis [m]	y-axis [m]	z-axis [m]
1	4	0	1,14	10	22	0	1,75
2	6	0	1,21	11	24	0	1,82
3	8	0	1,27	12	26	0	1,89
4	10	0	1,34	13	28	0	1,96
5	12	0	1,41	14	30	0	2,03
6	14	0	1,48	15	32	0	2,10
7	16	0	1,55	16	34	0	2,17
8	18	0	1,62	17	-1,5	0	3,00
9	20	0	1,69				

Table 3: Receiver positions in concert hall and for single reflections from the back wall.

The audience floor tilt was set so that the floor at the back wall was at equal height with the stage floor (1,2 [m]). The receivers were kept at a constant height of 1 [m] above the floor, and separated by 2 [m], as can be seen from Table 3. An omni-directional source was used. The situation is shown in Figure 18. Listening tests were only performed for a selection of the receiver positions.

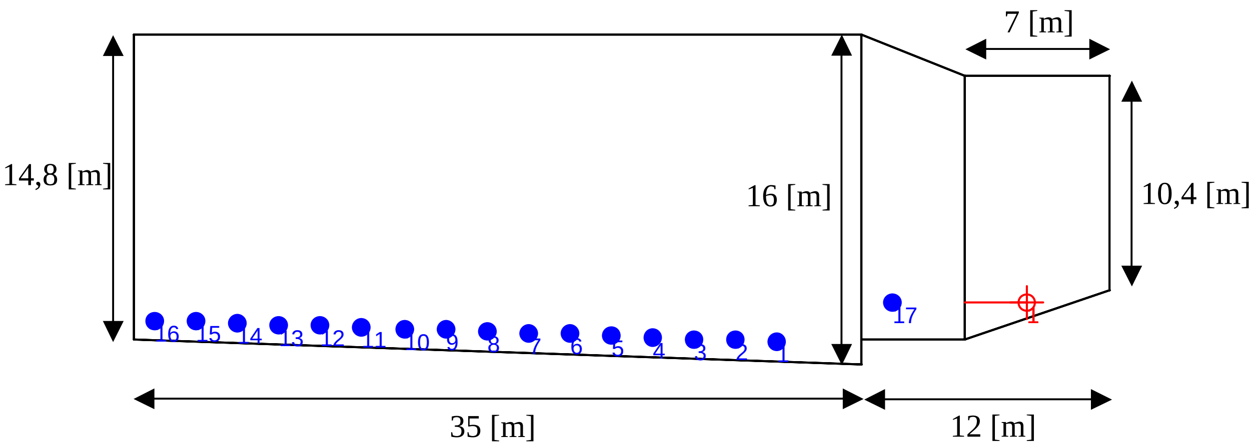

Figure 18: Receiver and source positions in concert hall. All walls except the back wall have α=1 in all octave bands.

3.2 Multiple echoes in an anechoic environment

Generation of the multiple echo test sections were done in a similar manner as with single echoes. Three echo sources were placed at 180° azimuth in the horizontal plane at an elevation angle of 0°. The direct sound source was set at 0° azimuth and 0° elevation. All sources were situated at equal distance from the receiver. This is shown in Figure 17, except that there are three sources on top of each other in one of the source positions depicted.

Dietsch and Kraak employ an evaluation window of length $\Delta\tau_E$ = 9 [ms] for speech and $\Delta\tau_E$ = 14 [ms] for music motifs. Echo separation times were chosen with these proposed window lengths in mind to assess their validity. Three test series were used both for the music and speech motifs, the first of the three echoes arriving at 18, 100 and 200 [ms] delay relative direct sound in both cases. Separation times of 4, 8, 10, 25 and 50 [ms] were selected for speech. With a separation of 4 [ms] will all three echoes arrive within the evaluation window, only two of them will do so when separated by 8 [ms], while only one echo will be included in the cases of the three longest separations. It might be inferred that this is pushed to the extremes, a fact that is done intentionally[1]. It is intuitive that the subjective difference between 8 and 10 [ms] separation is minimal, which is not how it is evaluated using a rectangular window. The two longest separation times are selected to assess the integration time of the ear, as it is approximated around 50 [ms] in the literature. The separation times used with the music motifs are 4, 13, 15, 25 and 50 [ms]. All situations and delays are summed up in Table 4 for convenience.

[1] Triple reflections within a short time interval have previously been proven equivalent to single reflections of same strength[27], p. 482.

The amplitudes of these echoes were first approximated from Dietsch and Kraak`s formula in a similar manner as for single echoes. In each case were the three echoes of equal strength, calculated from equation (14), which for three echoes give

$$EK(\tau)\cdot\Delta\tau_E = \frac{|p_E|^n (t_{E1}+t_{E2}+t_{E3})}{|p_D|^n + 3|p_E|^n} \tag{16}$$

where the subscript denote the arrival order of the reflections (echoes). Adjusting the amplitudes to appropriate ranges was done through iteration and preliminary listening tests, as for single echoes.

Receiver	Speech Separation [ms]	Speech Echoes [ms]	Music Separation [ms]	Music Echoes [ms]
18	4	18 – 22 – 26	4	18 – 22 – 26
	8	18 – 26 – 34	13	18 – 31 – 44
	10	18 – 28 – 36	15	18 – 33 – 48
	25	18 – 43 – 68	25	18 – 43 – 68
	50	18 – 68 – 118	50	18 – 68 – 118
100	4	100 – 104 – 108	4	100 – 104 – 108
	8	100 – 108 – 116	13	100 – 113 – 126
	10	100 – 110 – 120	15	100 – 115 – 130
	25	100 – 125 – 150	25	100 – 125 – 150
	50	100 – 150 – 200	50	100 – 150 – 200
200	4	200 – 204 – 208	4	200 – 204 – 208
	8	200 – 208 – 216	13	200 – 213 – 226
	10	200 – 210 – 220	15	200 – 215 – 230
	25	200 – 225 – 250	25	200 – 225 – 250
	50	200 – 250 – 300	50	200 – 250 – 300

Table 4: Delay times of multiple echoes.

3.3 Reverberant conditions

Most practical situations of echo annoyance occur in reverberant enclosures. As first reported by Haas[5], increasing reverberation times heighten echo thresholds. Two different geometries will be investigated, the first being an auditorium mainly constructed for speech performances, while the second is a larger concert hall.

3.3.1 Properties and geometry of the auditorium

The auditorium is modeled as a typical auditorium for teaching purposes, with a tilted audience area. Two different cases are implemented, where the scattering coefficient of certain walls are changed from 0,1 to 0,5 in between them. The reverberation times are approximately 0,95 and 0,7 [s], respectively, in the two cases. The auditorium is shown in Figure 19.

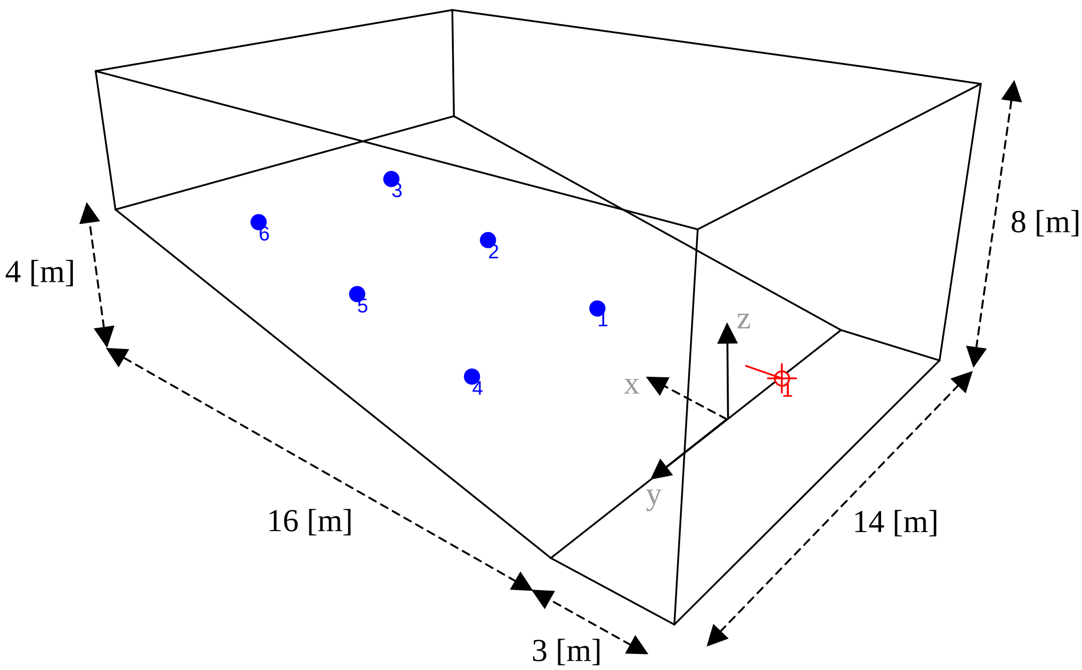

Figure 19: Auditorium for speech performances.

Table 5 shows the properties of the surfaces. Notice that the front and back wall and the sidewalls have two sets of scattering coefficients (see coefficients shown in bold in Table 5), as mentioned above.

Surface	Absorption coefficient, α 63 [Hz]	125 [Hz]	250 [Hz]	500 [Hz]	1k [Hz]	2k [Hz]	4k [Hz]	8k [Hz]	Scat. coeff. Case 1	Case 2
Stage floor	0,40	0,40	0,30	0,20	0,17	0,15	0,10	0,10	0,1	0,1
Audience floor	0,60	0,60	0,74	0,88	0,96	0,93	0,85	0,85	0,5	0,5
Front wall, stage	0,28	0,28	0,22	0,17	0,09	0,10	0,11	0,11	**0,1**	**0,5**
Back wall	0,28	0,28	0,22	0,17	0,09	0,10	0,11	0,11	**0,1**	**0,5**
Sidewalls	0,28	0,28	0,22	0,17	0,09	0,10	0,11	0,11	**0,1**	**0,5**
Ceiling	0,45	0,45	0,55	0,60	0,90	0,86	0,75	0,75	0,5	0,5

Table 5: Properties of surfaces in the auditorium.

The source was placed at center position of the flat stage floor, 1,8 [m] above ground to simulate a normal person. An omni-directional directivity was used for simplicity. The receiver coordinates are given in Table 6, origo is shown in Figure 19. All receivers are 1 [m] above the tilted audience floor. Receivers 1-3 are placed at center of the auditorium in the y-direction (y = 0) along the x-axis, while receivers 4-6 are placed 2 [m] from the nearest sidewall in the y-direction (y = 5).

Receiver no.	**x-axis [m]**	**y-axis [m]**	**z-axis [m]**
1	4	0	2
2	8	0	3
3	12	0	4
4	4	5	2
5	8	5	3
6	12	5	4

Table 6: Receiver positions in auditorium.

3.3.2 Properties and geometry of the concert hall

A concert hall can be built in a wide variety of shapes. However, a rectangular box is usually the basic starting point for most halls. Such a shape makes sure of sufficient early reflections if the hall is not too wide, as reported by Barron[14]. The concert hall used in simulations in the present paper is attempted kept simple. Two separate simulations have been carried out with scattering coefficients of 0,1 and 0,7 on the back wall of the hall, respectively (see bold figures in Table 7).

		Absorption coefficient, α								Scat. coeff.	
Surface	**No.**	**63 [Hz]**	**125 [Hz]**	**250 [Hz]**	**500 [Hz]**	**1k [Hz]**	**2k [Hz]**	**4k [Hz]**	**8k [Hz]**	**Case 1**	**Case 2**
Audience floor	1	0,60	0,60	0,74	0,88	0,96	0,93	0,85	0,85	0,5	0,5
Stage front	2	0,40	0,40	0,30	0,20	0,17	0,15	0,10	0,10	0,1	0,1
Flat stage floor	3	0,40	0,40	0,30	0,20	0,17	0,15	0,10	0,10	0,1	0,1
Tilted stage floor	4	0,60	0,60	0,74	0,88	0,96	0,93	0,85	0,85	0,5	0,5
Side walls, hall	5	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10	0,5	0,5
Side walls, mid	6	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10	0,5	0,5
Side walls, stage	7	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10	0,5	0,5
Stage back wall	8	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10	0,5	0,5
Stage flat ceiling	9	0,45	0,45	0,55	0,60	0,90	0,86	0,75	0,75	0,5	0,5
Tilted ceiling	10	0,45	0,45	0,55	0,60	0,90	0,86	0,75	0,75	0,5	0,5
Hall ceiling	11	0,45	0,45	0,55	0,60	0,90	0,86	0,75	0,75	0,5	0,5
Hall back wall	12	0,30	0,30	0,12	0,08	0,06	0,06	0,10	0,10	**0,1**	**0,7**

Table 7: Properties of surfaces in the concert hall.

An outline of the hall geometry is shown in Figure 18, with some measures denoted. A three dimensional view of the hall is shown in Figure 20. The ceiling between the hall and the stage (surface 10) house is tilted, as are the floor where musician(s) or an orchestra would play (surface 4).

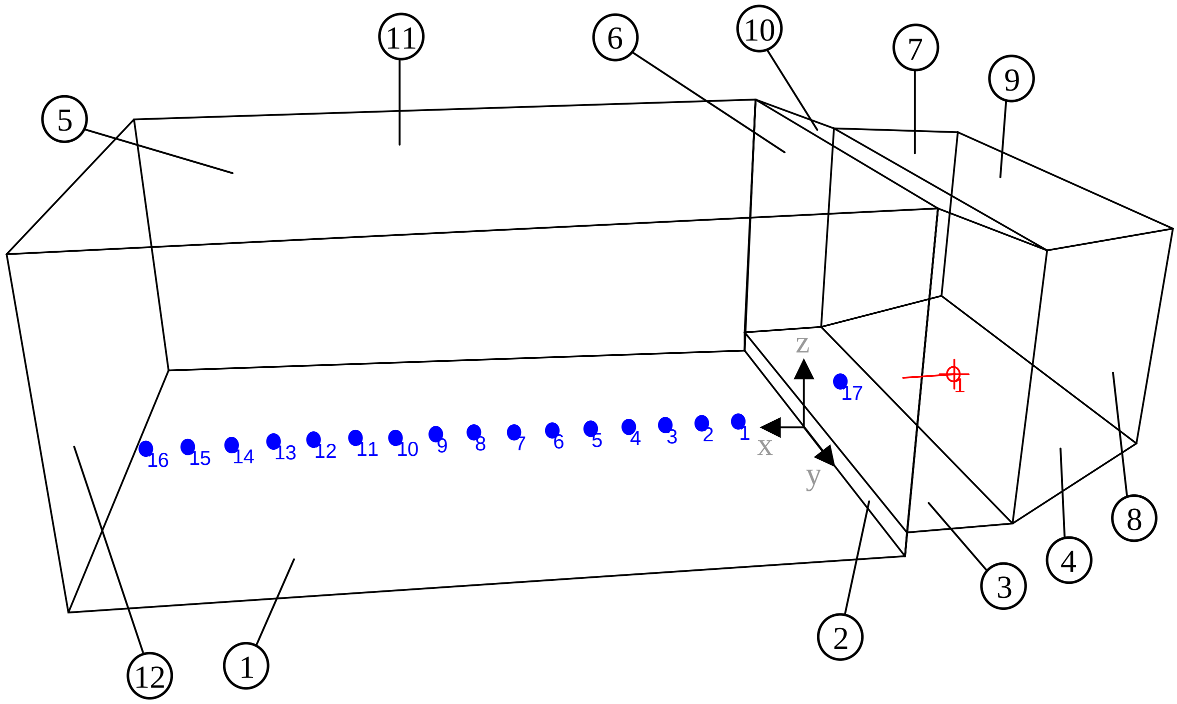

Figure 20: Concert hall geometry.

All surfaces of the concert hall are listed and numbered in Table 7, with their corresponding absorption and scattering coefficients. The numbers found in Table 7 have corresponding numbers in Figure 20. Reverberation times differ somewhat dependent on position, but are approximately 1,7 [s] with the lowest scattering coefficient set, while it decreases to about 1,45 [s] with the higher scattering coefficient set.

4 LISTENING TESTS

To be able to evaluate objective echo criteria, reference values and levels must be found from subjective listening tests. Listening tests were performed using both speech and music.

4.1 Choice of test motifs for listening tests

The nature of the test motifs forms the basis of any subjective evaluation done by listeners during listening tests. Threshold values are dependent on the speed and rhythmic properties of the motifs. The most echo critical situations must be used in establishing thresholds. Motifs for both speech and music sequences are chosen.

Anechoic recordings are available only in a limited extent. The recordings used are taken from the Bang & Olufssen[28] compact disc (CD) "Music for Archimedes", "Anechoic orchestral music recording" from Denon[29], and "Impact 2" from Japan Audio Society[30]. The music sequences chosen are given in Table 8.

Music Type	CD	Track	Composer	Excerpt	Duration
Cello	Archimedes	20	Weber	Theme	9,0 [s]
Guitar	Archimedes	12	F. Tárrega	Capriccio Arabe	15,3 [s]
Female chorus	Impact 2				13,1 [s]
Orchestra	Denon	9	Händel/Harty	No.6 – Water music suite, bars 1-11	18,1 [s]
Trumpet	Archimedes	34	Haydn	Cadance from concert (Eb Cornet)	12,8 [s]

Table 8: Music motifs for listening tests.

The frequency and dynamic properties of the different instruments vary greatly. Some instruments will not sound as dry as others even when played in an anechoic room. The bodies of instruments vibrate as strings of cellos and guitars or membranes of drums are excited. The most echo critical cases of the above are the trumpet and the female choir.

The dynamics of the music pieces chosen are also different in between them, as dynamics and frequency content of the tones are important in echo detection. Plots of amplitudes of the music and speech motifs are shown in Appendix A – Test Motifs.

Choice of test motifs was done with care, and all motifs are short excerpts of longer songs or music pieces. Listening sessions must be kept shorter than a maximum length of 30-35 minutes to avoid fatigue. The duration of each listening example should therefore be attempted kept as short as possible without changing the outcome of the subjective evaluation. I would argue that the motifs presented to the listeners are of sufficient length. As long as the excerpts are chosen with care, the parts of the total music pieces that are most critical will be included. It will still be these excerpts that become decisive if the music pieces where presented in original and full length. There are no indications that the durations of test sequences have any influence on echo sensitivity. However, it is important that the excerpts are cut at natural breaks so that it sounds natural.

Speech is of a different nature than music, although percussive music can also be very articulated. Echo critical limits depend on the speed of speech. The 8 seconds of speech presented to the listeners is male speech at the speed of 4,0 syllables per second (see Table 9). 4,0 syllables per second can be argued to be below average, which in the literature are said to be about 5,0 syllables per second. On the other hand, it is equally important that the speech sequence is clear and distinct, and contains breaks and transient consonants as for example *k*, *p*, *q*, *t* and *x*. The sequence chosen reads: "*Germanys decision followed eight years later, and the Scandinavian states and Russia changed in eighteen seventy-five*".

Speech Type	CD	Track	Language	Speed	Duration
Male	Archimedes	5	English	4,0 [syllables/sec]	8,0 [sec]

Table 9: Speech motif for listening tests.

4.2 Listening test setup and accomplishment

A total set of four different listening sequences were used, covering the six different motifs in the various geometric situations. Odeon generates wave-files as output from convolving the anechoic motifs with the room impulse responses of the different rooms. Anechoic samples were separated from reverberant ones to isolate the parameters wanted in the evaluation. The different motifs were also separated for the same reason.

	Section	Motif	Conditions	Samples
Sequence 1:	1	Speech	Anechoic – single and multiple	117
	2	Speech	Reverberant	12
Sequence 2:	1	Trumpet	Anechoic – single and multiple	121
	2	Trumpet	Reverberant	14
Sequence 3:	1	Cello	Anechoic – single and multiple	107
Sequence 4:	1	Guitar	Anechoic – single and multiple	33
	2	Female chorus	Anechoic – single and multiple	13
	3	Female chorus	Reverberant	14
	4	Orchestra	Anechoic – single and multiple	35
	5	Orchestra	Reverberant	8

Table 10: Composition of the listening test sequences.

If all the sequences were to be mixed randomly together, would the actual echo annoyance be reduced to only one of several parameters changed between each sample. The reference created as one hears the same motif repeatedly would consequently be blurred or altered as result of the variety of instruments and conditions mixed together.

Table 10 shows the composition of the listening test sequences. A random order of the samples was generated for each listening person within each section. The samples for the different sections were put in separate folders, and generated randomly using MatLab[2] (see m-files[3] included in Appendix F – CD and also Appendix C – MatLab m-Files). Examples were played before each section, to present the echo span of the following test section. An anechoic sample was played first, before a given number of samples ranging from the strongest echo sample to the weaker were played. Breaks were included before and after every example and test section. The MatLab m-script generated a txt-file of the sample order after each sequence was finished.

Listening test persons were presented with a range of three different echo grade options. A lot of consideration was given to the number of options made available to the test persons. It was important to make it as simple as possible. Simplicity ensures better assessment and separation of the parameter to be evaluated. The combination of setting the number of options down to three and presenting the anechoic sample as well as the range of echo levels before commencing every test section, was done to secure a common reference level. Use of larger evaluation scales can disturb the common echo reference level, and cause test persons to only use a limited range the total scale. The three options presented to the test subjects were:

1) Echo **N**ot audible (N)
2) **A**udible, but not disturbing, echo (A)
3) **D**isturbing echo (D)

Still a certain margin for variations in individual decision levels is inevitable. This is why a certain number of test persons must take part to give a satisfactory statistical basis for evaluations and conclusions to be drawn. Ten listening tests were conducted for each test sequence. A total of twelve persons were used. Consequently, all test persons did not participate in all sequences, which is no necessity. The number of test sequences performed per day by one test person was attempted limited to one, once again to avoid fatigue. However, some test persons did two test sequences in one day, separated by several hours.

All test persons were carefully instructed to concentrate on the (possible) echoes. Adding a delayed version of a music or speech sequence might cause annoyance due to other reasons than an echo being present, especially in anechoic conditions where no reverberant masking is present. What could be heard in some of the samples, particularly in cases with an echo separation of 30 and 50 [ms] for the trumpet, was a kind of shrill interference effect. Despite not being a pleasant sound, it was still the annoyance of separate echoes (see definition from Kuttruff[1] in Section 1) that should be evaluated.

The test persons gave their answers on paper instead of typing them continuously in on the computer. Examples of the answering sheet layout can be seen in Appendix B – Listening Test Answering Sheets. Even though listening sequences were kept short, fatigue and unawareness can occur, which in turn can cause errors. As samples occasionally can get

[2] Info available at http://www.mathworks.com
[3] m-files speechtest, trumpettest, cellotest and mixtest

mixed up when listening to the same short motif over and over again, a counter was displayed on the computer screen. The counter corresponded with the numbering on the answering sheet so that test persons could check the synchronization between their answering and the sample flow continuously. Inserting breaks between each section also helped synchronization. Results from the answering sheets were manually compared with the corresponding samples in the txt-file generated by the m-scripts.

All of the listening tests were completed in the semi-anechoic lab at the Norwegian University of Science and Technology to ensure that the listening situation was kept as quiet and neutral as possible. A Soundblaster PCI 128 Gold sound card[4] was installed in the computer, and the wave-files was sent through a Yamaha 01V mixer[5] before being played by a set of Beyer Dynamics DT990 headphones[6]. The test persons were allowed to adjust the volume on the mixer to a suitable level individually.

[4] See http://www.americas.creative.com/support/manuals/Files/AudioPCI128.pdf for details.
[5] See http://www.yamaha.co.jp/product/proaudio/homeenglish/index.htm for details.
[6] See http://www.beyerdynamic.de/com/product/index.htm for details.

5 DEVELOPMENT AND IMPLEMENTATIONS OF OBJECTIVE ECHO CRITERIA

The main purpose of this paper is to investigate and evaluate objective echo criteria based on subjectively acquired reference data. Two different versions of Dietsch and Kraak`s criterion are implemented, as explained theoretically in Section 2.3. Also a new criterion is developed, based on the convolution of the inertia of ear and the room impulse response. The implementation is done in MatLab, and all m-files can be viewed in Appendix C – MatLab m-Files and are included in Appendix F – CD.

All the criteria start out from input room impulse responses (RIR) obtained from Odeon. Despite being a well-developed computer simulation program, certain approximations and limitations exist, as discussed in Section 2.4.

5.1 Implementation of Dietsch and Kraak`s echo criterion

Two different versions of the criteria are implemented, as mentioned above. The difference between implemented versions of the Dietsch and Kraak criterion is the basis on which the room impulse response inputs from Odeon are generated.

One version of the RIR`s are generated from only ISM and not passed through the HRTF filter set (see Section 2.5). The other version receives RIR`s in wave-format, which is filtered by the KEMAR head related transfer function set. These are the full-length impulse responses that the listening test motifs are convolved with, and consequently what the listeners actually heard and intuitively should be the basis of any criteria.

The MatLab m-files "*dkism.m*" and "*dkwav.m*" calculates Dietsch and Kraak`s echo criterion based on output .txt- and .wav-files, respectively. Both files can be viewed in Appendix C – MatLab m-Files and are also included in Appendix F – CD. Both files naturally have many similarities, as it is the same criterion that has been implemented.

5.1.1 Algorithm based on RIR calculated using ISM

Excluding reflections of higher order than those calculated using the image source method is an ambitious approximation. The degree of error depends on the transition order from ISM to ray tracing. Naturally, the transition order is insignificant in any anechoic environment. The reverberant cases are investigated with transition orders ranging from 1-4. Errors will be more severe with decreasing absorption factors, as little energy will be lost as the transmission and absorption components of reflections become larger.

Odeon is not compatible with MatLab, and mat-files cannot be generated. Neither can the ray tracing part of the room impulse response generation be switched off to get wave-file output based on ISM only. However, amplitudes, delays and angles of incidence of all ISM calculated reflections can be exported in ASCII-format to txt-files from the reflectogram folder in "Single point response" for the highlighted job in the job list in Odeon. Sound pressure levels (SPL) are exported in octave bands, which must be added together before entered into the Dietsch and Kraak echo algorithm. Calculations and evaluations are not performed in octave bands separately. Figure 21 shows an example of the normalized reflectogram of receiver 11 in the concert hall (see Figure 20 and Table 3). ISM up to fourth order is used, with a scattering coefficient of the back wall of 0,1 (surface 12 in Table 7). Only the first 15000 samples (≈340 [ms]) are shown. The direct sound is normalized to 1 in all impulse responses.

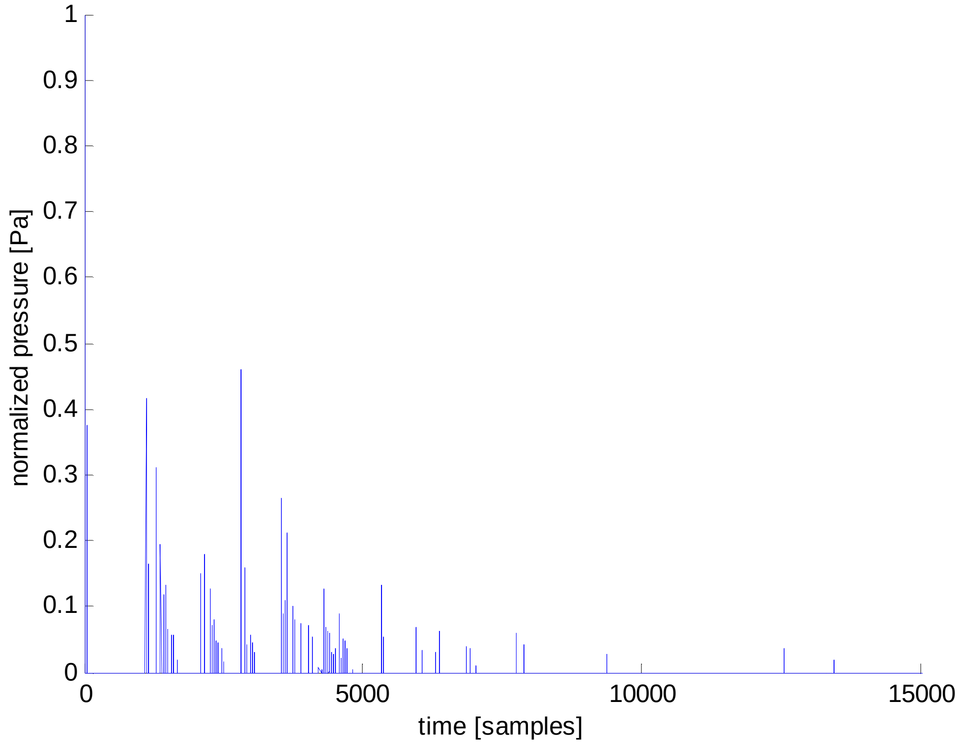

Figure 21: Example room impulse response calculated from ASCII output of reflectogram from Odeon.

All files to be evaluated in a given run must be put in a folder specified in the m-file. As Dietsch and Kraak have specified different parameters for speech and music (see Table 1), only files calculated for speech or music should be calculated at the same time. Whether speech or music motifs are to be evaluated must be typed in manually for each run (asked by MatLab, see line 26 of *dkism.m*. If both speech and music motifs are to be included, this line (and the simple *if*-loop in line 37-40) must be moved inside the main *for*-loop beginning at line 46. As a result, the calculation parameter must be typed in for all files separately).

One must also choose whether the resulting Dietsch and Kraak levels shall be plotted. Finally, after all calculations are completed, a file containing the maximum Dietsch and Kraak level for each impulse response file calculated can be constructed if wanted. Name of the file must be typed in the command window in MatLab when asked (line 192 xyz in *dkism.m*).

Figure 22 shows temporary results of an example calculation from the ISM implementation of Dietsch and Kraak. The figures on the y-axis in graphs a)-c) are simply ratios relative to one another resulting from the normalization done earlier in the computation. The calculation shown is based on the reflectogram in Figure 21. The nominator shown in a) has smaller amplitude than the denominator in b) because, as can be seen in Equation (5), the nominator is multiplied with a time factor. Lines 103-132 of *dkism.m* show the computation of the nominator and denominator. The final build-up function is the ratio of the nominator in a) to the denominator in b), and is shown in part c) of Figure 22.

Echo disturbance is dependent on steps in the build-up functions within a certain time interval, as outlined in Section 2.3. Steps in the build-up function in Figure 22 c) can therefore be seen as peaks in the resulting EK(τ) values plotted in part d) of the figure. The example room impulse response will not contain reflections that will be percepted as echoes, according to this implementation of Dietsch and Kraak. The highest EK-value in the given situation is 1,018, which is well below the limit value of 1,8.

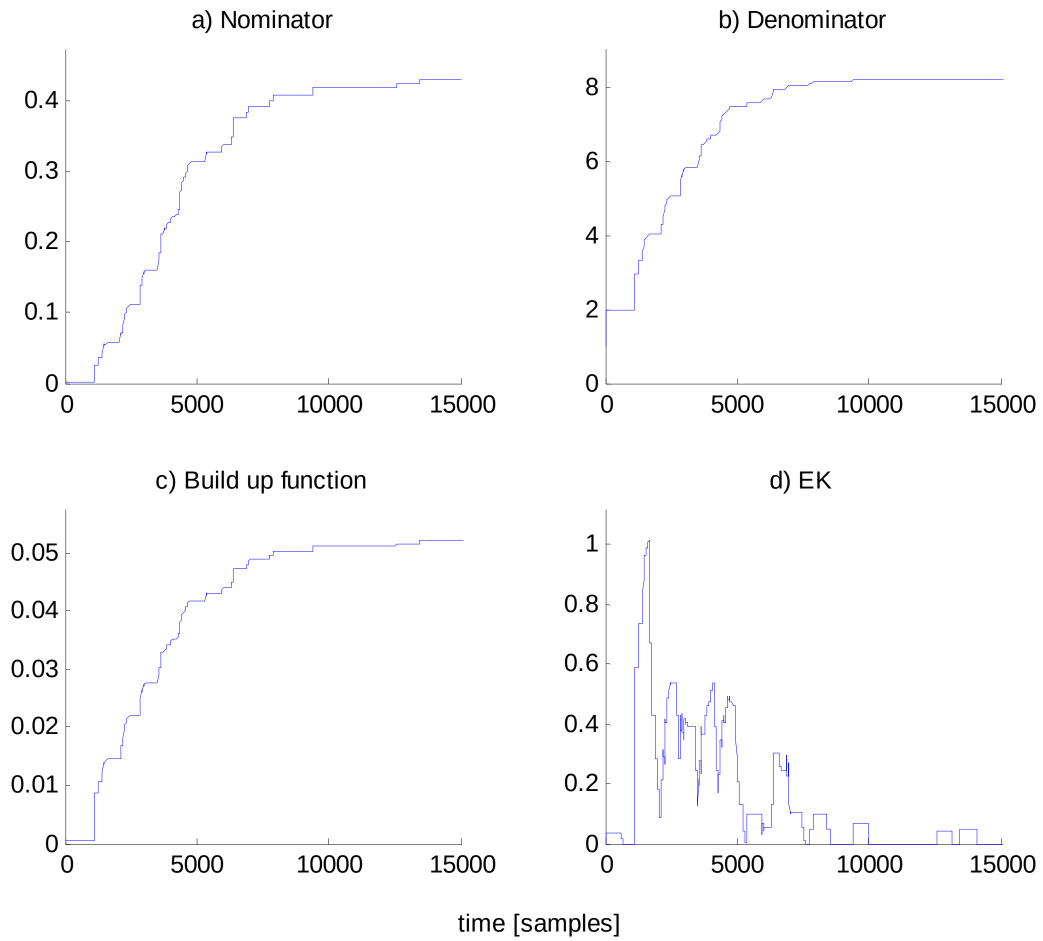

Figure 22: Temporary results of the Dietsch & Kraak echo criterion. RIR reflectogram of Figure 21 from receiver 11 in the concert hall (Figure 20) used as example.
(a) Nominator of Equation (5).
(b) Denominator of Equation (5).
(c) t_s of Equation (5).
(d) Resulting Dietsch and Kraak values from Equation (10).

Discrete reflections computed using ISM, as seen in the reflectogram above, cause abrupt steps in all graphs plotted in Figure 22. A feature caused by the abruptness of the rectangular window proposed by Dietsch and Kraak. A reflection is either inside or outside the time interval. No scaling is contained in the window qualities. This in turn causes the angular shape of EK(τ). The two reflections seen around sample 13-14000 in the reflectogram in Figure 21, has corresponding squares visualized around the same samples in Figure 22 d). As the reflections are separated by more than the evaluation period of 14 [ms] given for music by Dietsch and Kraak (Table 1), the reflections gets a shape similar to the window used. The effects of the rectangular window will be further discussed in Section 6.3.

5.1.2 Algorithm based on wave-file output of RIR from Odeon

The difference introduced by including whole RIR`s calculated by Odeon compared to only lower order ISM is potentially huge. Sound waves that have been reflected more than the threshold number of times given by the transition order chosen now play a major role in the total impulse response. Besides, all RIR`s were convolved with the KEMAR HRTF set which is a necessity with headphone reproduction of the sound samples for listening tests. The complete m-file can be viewed in Appendix C – MatLab m-Files, and are also included in Appendix F – CD.

Most of the code used in evaluating the input wave-files is the same as with the txt-files. The implementation of the Dietsch and Kraak echo criterion in itself is the same, but the preprocessing is different. As with the txt-files, all files to be evaluated must be stored in a folder set in advance in the m-file. RIR`s will be evaluated for the same type of motif (speech or music) as this option is set as a constant before the main calculation of all the RIR`s begins.

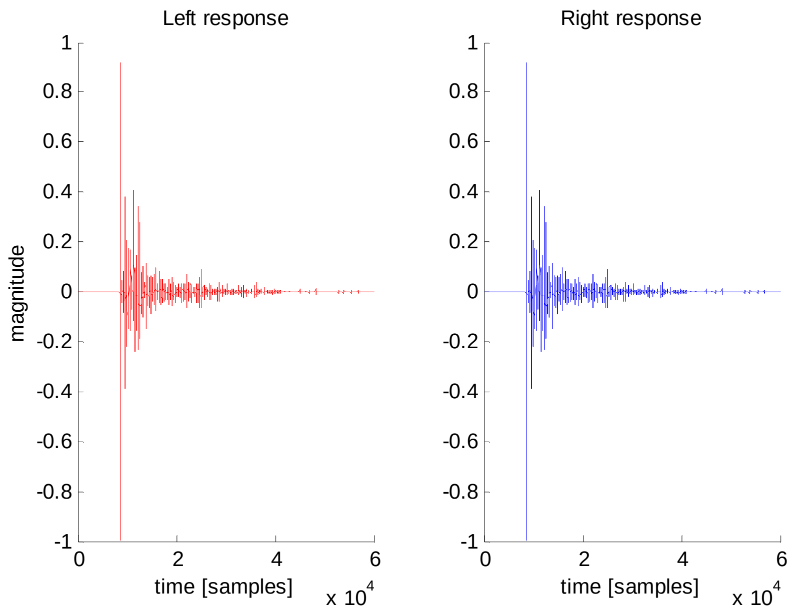

Figure 23: Binaural room impulse response from receiver 11 in the concert hall.

First of all, the wave-files contain binaural impulse responses that have been passed through the HRTF filter. Figure 23 shows the output BRIR`s for the same receiver position and parameters in the concert hall as exemplified in Section 5.1.1. Both channels of the BRIR`s are rectified before they are added coherently. It is thus pressure values that are added ($\left|p_{left}\right| + \left|p_{right}\right|$). 16 [bit] integer quantization is used in generating wave-files in Odeon, causing a weak noise floor to be present. Removal of this error is performed in lines 73-80 xyz. All BRIR output generation from Odeon are normalized to a max level of 1, as can be seen in Figure 23.

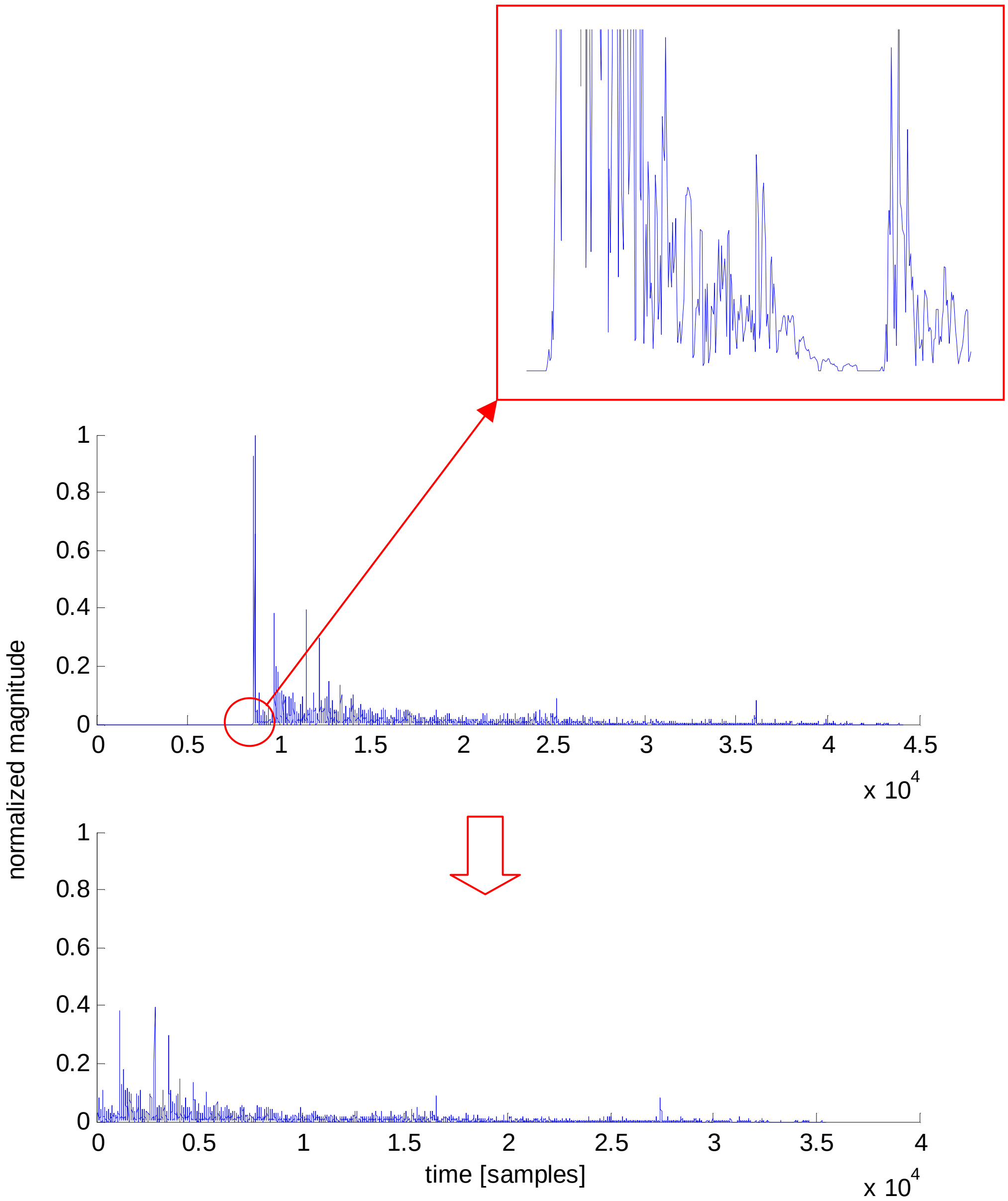

Figure 24: Adjustment of time axis of summed BRIR.

A time axis relative to the arrival of the direct sound must be established, as the BRIR`s begin in time at source generation. Since the arriving impulse responses are quite complex, adjusting the time axis is not straightforward. The enlarged part of the upper response shown in Figure 24 shows the direct sound response. It is quite complex compared with the corresponding ISM calculated response in Figure 21. As quantization errors have been removed, the RIR has values of zero until the direct sound arrival. As the direct sound no longer arrive at t=0, it will impose an effect also on the nominator of the build-up function in Equation (5). The time axis is adjusted so that the first sample exceeding a certain value is set as first sample, which give the RIR shown in the lower part of Figure 24. Other solutions like summing the direct sound contribution to one sample or setting all samples within direct sound contributions to zero in the nominator of the build-up function cannot be validated. The basis of evaluation must be the sound field that arrives at the two eardrums of the listener.

The succeeding calculation is the same as explained in Section 5.1.1. However, as the input RIR are more complex, the output EK(τ) become correspondingly complex.

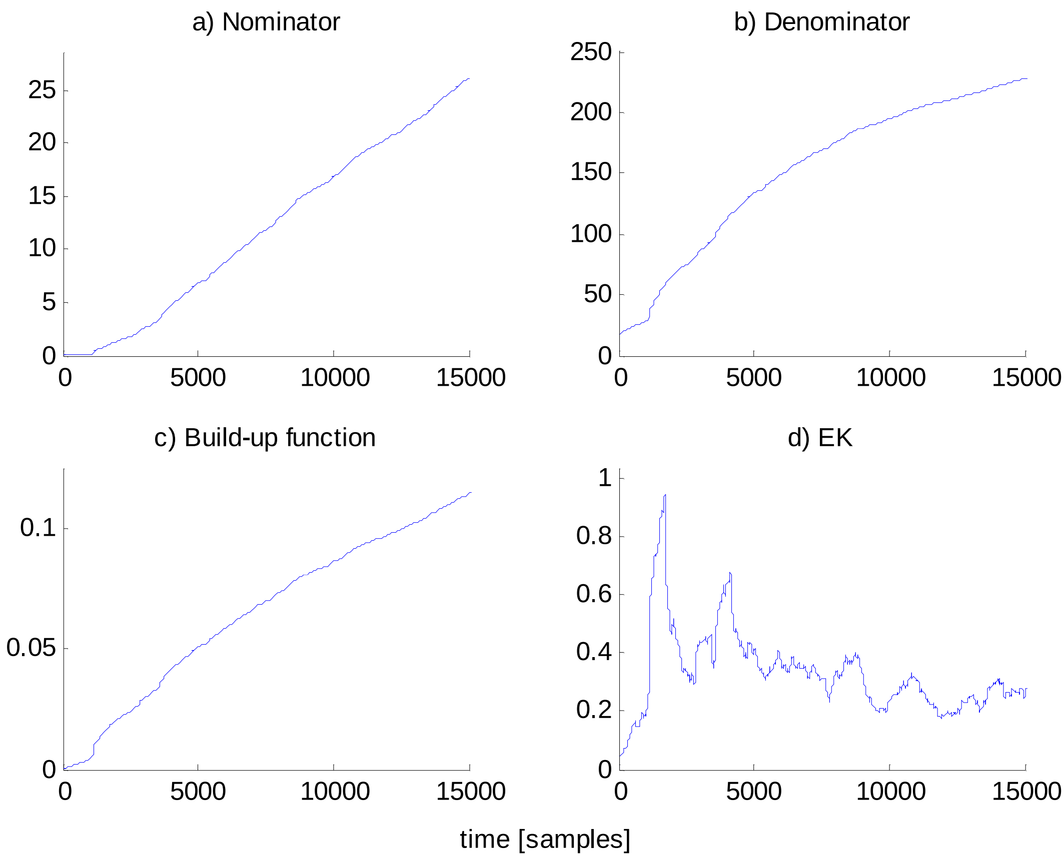

Figure 25: Temporary results of the Dietsch & Kraak echo criterion. Lower RIR shown in Figure 24 used in the computation.
(a) Nominator of Equation (5).
(b) Denominator of Equation (5).
(c) t_s of Equation (5).
(d) Resulting Dietsch and Kraak values from Equation (10).

Visual differences between all parts of Figure 22 and Figure 25 are conspicuous. Using wave-files as input create more continuous results. However, the shape and values of the resulting EK(τ) is similar. Despite keeping a higher level throughout due to the additional energy arriving at receiver, the peaks are at comparable levels with the ones from the ISM calculation. The results from wave-file RIR`s will not contain reflections that will be percepted as echoes either. The highest EK-value in the given situation is 0,9426, which is well below the limit value of 1,8, but not so different from the maximum value of 1,018 from Figure 22 d).

5.2 Development of an alternative echo criterion

As will be evident in Section 6, the echo criterion formulated by Dietsch and Kraak did not prove satisfactory agreement with the results from the listening tests. In formulating an alternative criterion, the idea of using a build-up function of the entire impulse response as the basis of evaluation was rejected. A more intuitive approach was adapted based on the energy impulse response. Evaluation of the sound fields the listeners actually were presented with appeared to be the best starting point. Consequently wave-file RIR`s were used throughout.

The human ear integrates incident sound energy, and contributions arriving within the integration time of the ear will be percepted as one. Sound waves arriving within the integration time can produce effects of increased spaciousness, tone colouration or image shifts (see Figure 8). However, separate discrete echoes cannot be distinguished. Amplitude, frequency spectrum and angle of incidence are decisive parameters of our perception of incident sound waves. The HRTF`s used to generate BRIR`s covers these filter effects, which are caused by the alteration of incident sound waves due to our head and shoulders. Relating the RIR to the arrival of the direct sound is done in the same way as described in Section 5.1.2. Wave-files are imported and preprocessed to obtain pressure RIR as depicted in the lower part of Figure 24 in a similar as for the Dietsch and Kraak criterion. Squaring the BRIR after they have been rectified converts pressure to energy:

$$p^2 = \left|p_{left}\right|^2 + \left|p_{right}\right|^2 \tag{17}$$

This resembles the function of the human ear.

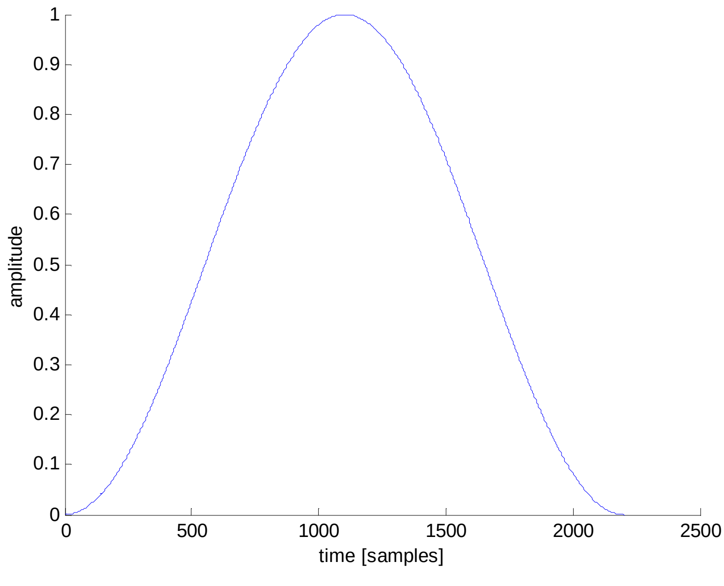

Figure 26: Hanning filter of length 50 [ms].

Much work has gone into determining the limits of perceptibility of separate reflections (also referred to as *the merging limit* in the literature), as discussed in Section 2.2. The single and multiple echo parts of the listening test in the present paper also assess these limits. An integration time of 50 [ms] is adapted to simulate the time integration of the human ear. 50 [ms] is a conservative and acknowledged integration time agreed upon by several authors[4]-[6], [8]-[11] and[14]. Results from the listening tests done in this paper also acknowledge 50 [ms] as a reasonable integration time.

Reflections separated shorter in time than the inertia of the ear will not be percepted as a separate event, and can consequently not be distinguished as one. An evaluation window of 9 and 14 [ms] for speech and music motifs, respectively, as employed by Dietsch and Kraak, seemed to be too short. It is reasonable to infer that a different integration time is valid when evaluating a build-up function. However, it is still the difference within the integration window that is evaluated, and it can therefore be interpreted as such. On the other hand, also the shape of the integration window has significant impact in evaluating the incident sound waves. In constructing a filter to simulate the inertia of the ear, must not only the length of the filter be determined, but also the filter coefficients.

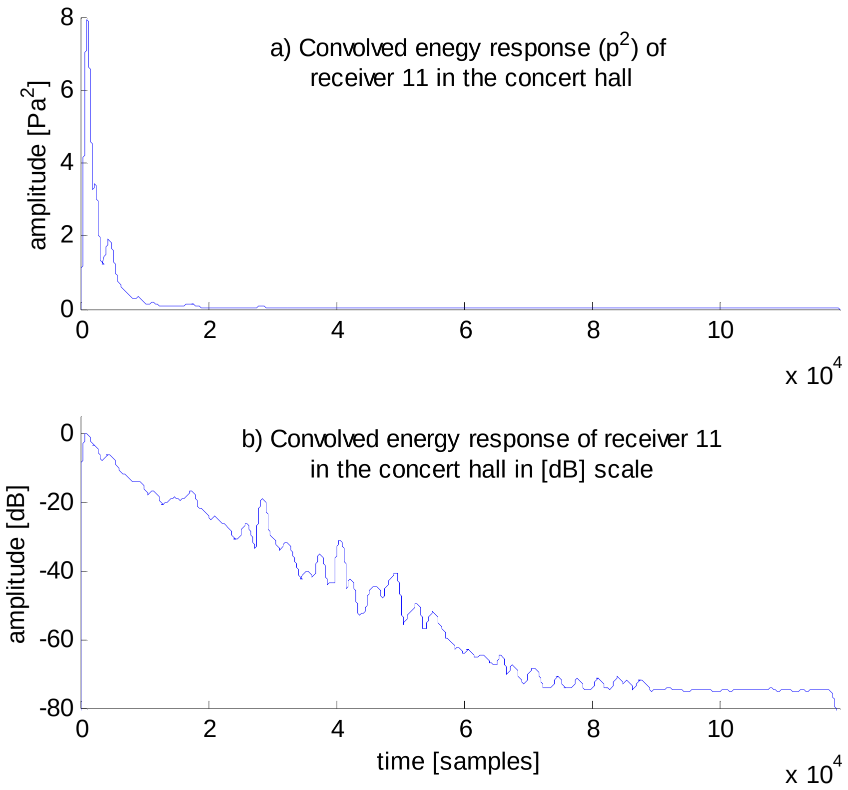

Figure 27: Energy response of receiver 11 in the concert hall in (a) [p^2] and (b) in [dB].

To employ a complicated filter seems unnecessary, as a large number of room impulse responses must be evaluated to validate and discriminate small variations caused by the filter from variations caused by other parameters that change as geometries are varied. The filter used here is a simple Hanning filter of length 50 [ms], which is a discrete time- and amplitude shifted cosine wave. The mathematical expression is given by

$$w(k+1) = 0{,}5(1-\cos(2\pi \frac{k}{n+1})), k = 0,....,n-1 \qquad (18)$$

where n must be a positive integer. Multiplying the sampling frequency and filter length in seconds give the correct filter length in samples. Figure 26 shows a Hanning window of length 50 [ms]. Convolving the Hanning window with the energy RIR provides the basis for further evaluation. Reflections arriving within the width of the Hanning window will contribute to the total simulated sound energy percepted by the receiver as one impulse. Forward and backward masking is assumed equal in time and amplitude, which is an approximation of the integration and masking curve found throughout the literature.

Figure 27 (a) shows the resulting response after squaring the RIR shown in the lower part of Figure 24 and subsequently convolving it with the Hanning window depicted in Figure 26. Quantization errors from the wave-file generation are not removed in the following. Removal of the error is not necessary, as its level is so low that it will make no significant impact on the criterion. A quantization noise floor of approximately –75 [dB] is introduced. Figure 27 (b) shows the corresponding result, converted to a [dB] scale and scaled so the maximum sound level equals 0 [dB]. The RIR length of the example in hand is a bit shorter than 3 [s] as the sampling frequency is 44100 [Hz].

Development of the echo criterion was an iterative process, in which logical and theoretically supported ideas formed the platform that investigations were based upon. The following section includes some initial implementations and considerations that led to the final result.

5.2.1 Initial unsuccessful implementations and considerations

The listening test material was developed with that in mind to verify Dietsch and Kraak`s echo criterion in both anechoic and reverberant conditions. If the criterion formulated by Dietsch and Kraak proved not satisfactory, the anechoic listening test result could be used to develop a new criterion. The criterion could then be tested on the simplified reverberant situations simulating an auditorium and a concert hall.

Amplitude thresholds were found as function of time delay between direct sound and echo, as given in Section 6.2. Evaluation of peaks rising above the constant reverberant decay curves based on amplitude thresholds from anechoic conditions was used as a starting point in the development of the new criterion. The Hanning window was applied in anechoic conditions as well, ensuring a simulated sound perception length as the echo pulses arrive. The peak level will still be the same as generated in Odeon as long as reflections are separated by a minimum time interval equal to the window length. It occurs when the Hanning window has its peak level of 1.

The energy impulse responses were converted to [dB], setting the global peak level to a reference value of 0 [dB], as shown in Figure 27 (b). In reverberant and more complex situations extremal points on the [dB] energy decay curve (local and global maxima and minima) must be available if such an evaluation should be possible. This was implemented in addition to level step from minimum to their respective maximum points. Also separation times from last equal energy level from all maxima points were found.

Extremal points of the response shown in Figure 27 (b) are included in Figure 28. Green, red and black circles mark maxima, minima and separation time from last equal sound energy level, respectively. Only points in the first 60000 samples are included as the curve fluctuates rapidly in the late response, which are mainly due to the quantization error mentioned above. Odeon outputs are not reliable at such late and weak levels either.

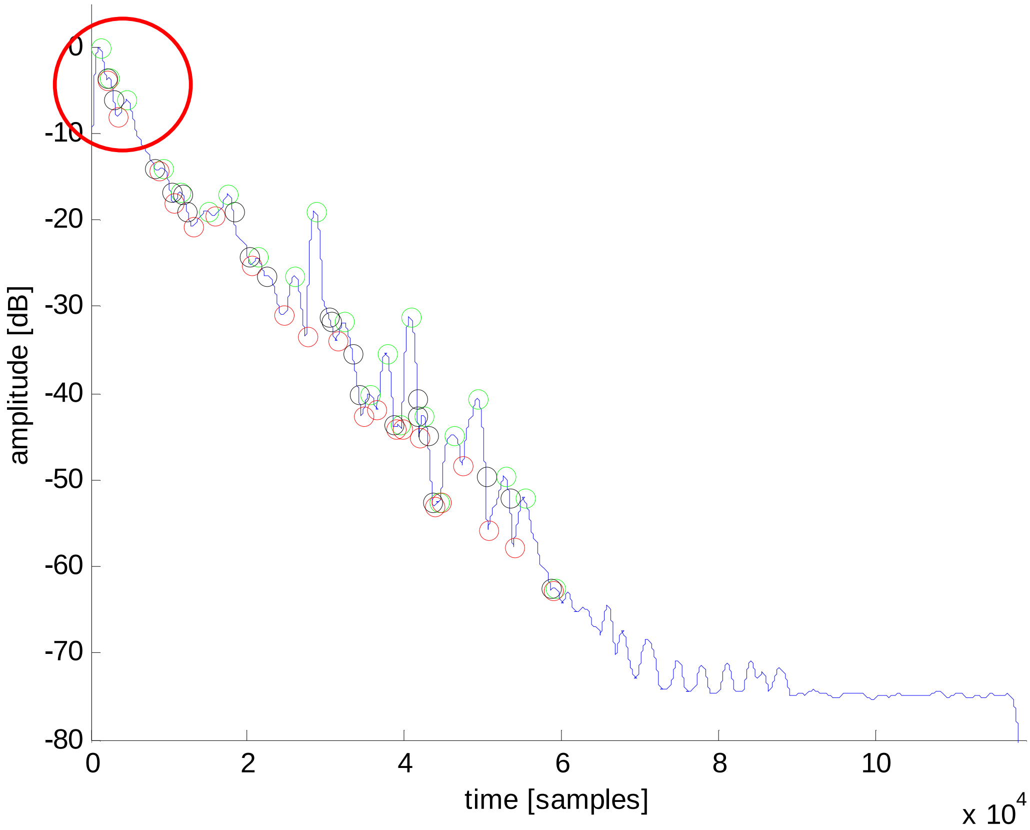

Figure 28: Response of receiver 11 in concert hall, extremal points marked by circles.

Peaks that rise above the reverberation decay are clearly visible. A magnified version of the part of the energy response in the bold red circle in Figure 28 is shown in Figure 29. Evaluating each peak along the decay curve based on step values from minimum to maximum value as function of time was the initial idea. As time parameter was the separation time since the last equal energy level attempted used (t_{sep} in Figure 29). Echo perception will depend on the area under the curve in the rectangular box indicated in Figure 29. Averaging the area in the box and adding the resulting value to the minimum value will give a smaller step between minimum and maximum values and compensate for masking energy arriving shortly before the peak. How much above the minimum value the new minimum reference level indicated by the red line in Figure 29 should be elevated thus depend on the shape of the energy curve within t_{sep}. ΔL_{step} becomes the level step to be evaluated.

Subsequently comparing ΔL_{step} with reference levels obtained from single echo listening tests for the time delay t_{sep} were meant to complete the criterion. The criterion would then be applicable in any anechoic or reverberant conditions. However, results were not satisfactory. For one, the criterion does not consider anything outside the rectangular evaluation window. Dependent on the steepness of the curve before entering window, the reverberation time of the ear itself may mask the entire peak, for instance. Also, the adjusted average level of the red line will rise as the echo amplitude rise, and consequently cause ΔL_{step} to have small variations due to the amplitude of the peak.

An attempt to only use the area under the curve up to the minimum point (in time) as basis for calculation of ΔL_{step} did not prove satisfactory results either. A final alteration of the criterion was not to convert the energy response to [dB] scale, but use the linear scale instead. The same problems as above visualized themselves. Two single echo cases with the echo arriving after i.e. 200 [ms] came out almost equal even with an amplitude deviation of 10 [dB] between the two.

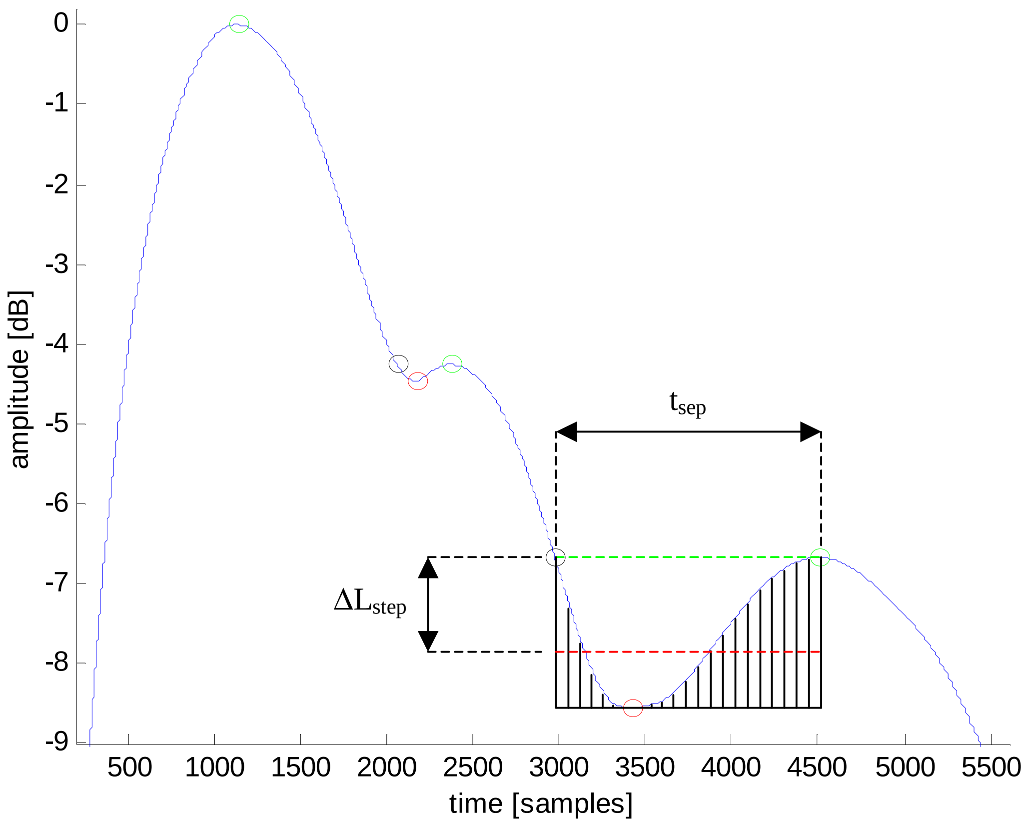

Figure 29: Evaluation of extremal points to determine echo disturbance.

5.2.2 Development and implementation of an alternative echo criterion

Despite the unsatisfactory results of the previous discussed echo criteria, echo critical situations could be seen from a visual inspection of the logarithmic energy response of all receiver positions as given in the lower part of Figure 27. Curves for receiver positions and parameters where listening tests had proven to produce disturbing echoes were seen to contain higher peaks, more fluctuations, and deviating from the ideal reverberant decay.

A clear advantage of any criterion is to be independent of as many input parameters as possible. Odeon computes T_{30} based on decay from –5 to –35 [dB]. It now seemed most reasonable to use the reverberation time as a starting point for further development of the criterion, in contrast to the previous discussion. An algorithm could be implemented to compute the reverberation time in MatLab (based on i.e. backward Schroeder integration). To reduce computational time and since the reverberation time already is available from Odeon, was the reverberation time set as the sole input parameter required to be typed in manually.

Figure 30 shows the energy response after it has been convolved with the Hanning window. A reverberation decay line of is also plotted. It has been set to start its decay at the global maximum of the logarithmic energy response for convenience. Its starting point can be arbitrary in the following discussion as long as the decay is correct. The black curve plotted in the upper part of Figure 30 shows the difference between the reverberation decay and the energy response.

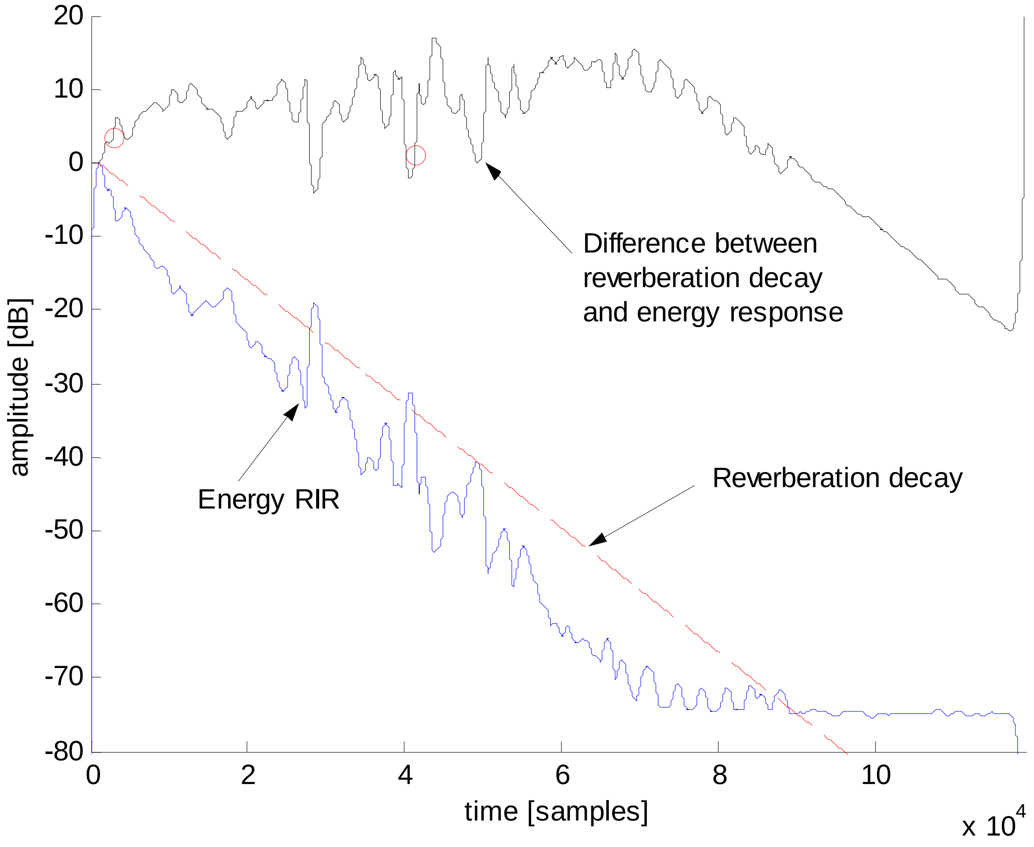

Figure 30: Difference between reverberation decay and energy RIR of receiver 11 in the concert hall.

Usage of the reverberation curve as basis in constructing an evaluation curve for the echo criterion requires it to be shifted to an appropriate level relative the energy response input automatically. This must be done correctly for any input energy response, regardless of its shape and reverberation time. The interval of the RIR curve used to shift the reverberation curve must be the same as the interval used by Odeon to calculate T_{30}. Two red circles are indicated on the black difference curve, indicating the first time the energy response reaches a level of –5 [dB] and the last time it reaches –35 [dB] re. direct sound, respectively. Subtracting the average value of the difference curve from the reverberation decay curve on this interval will move it to a centre position of the decay (Figure 31).

Shifting the reverberation curve will cause it not to start from the direct sound in its decay, which is intuitively wrong. The sound level inside the ear decreases with the reverberation time of the ear (T = 0,4 [s]). No echo can occur before a sound wave has arrived. Thus the evaluation curve is set to zero in the interval before the direct sound arrives. To connect the direct sound peak with the evaluation curve constructed from the reverberation time of the room, is a decay of T = 0,4 [s] used. The steeper reverberation decay of the ear is adjusted so that it becomes tangential to the filtered energy response of the direct sound.

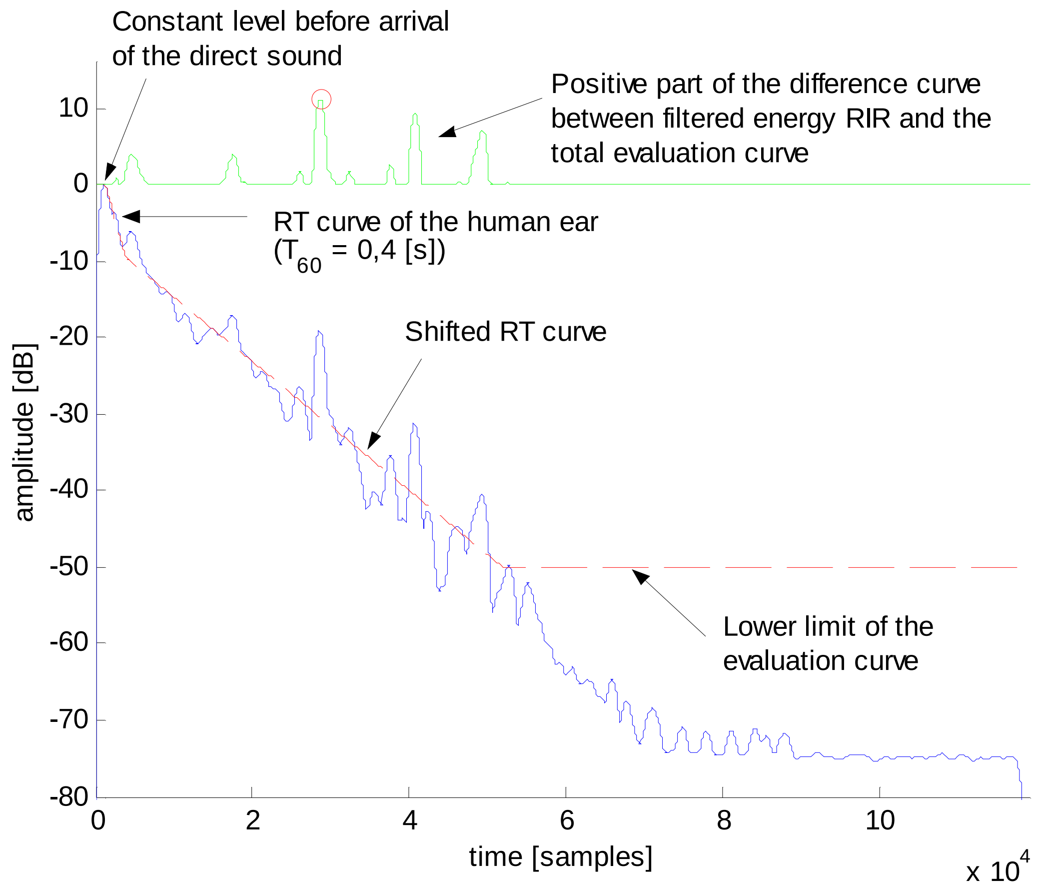

Figure 31: Total evaluation curve applied to energy RIR of receiver 11 in the concert hall. Positive levels of the difference between the resulting evaluation curve and the energy RIR is shown in the upper part.

Muncey, Dickson and Dubout showed that acceptable echo levels could be divided into three parts, as function of delay time[9] (see Section 2.2). The resulting evaluation curve is a combination of the curves discussed in the previous paragraphs, and is of a similar nature as proposed by Muncey, Nickson and Dubout. With a noise floor present will the shifted reverberation curve eventually fall below it as it continues its decay with time. The background noise floor will not be a source of any perceptible, separate echo reflections. Thus a lower limit for which the criterion is applicable must be established. This was done using the results from 400 [ms] delayed single echo listening tests and adjusted during the iterative process of establishing threshold values for how much the energy RIR could exceed the evaluation curve before an echo was audible. A lower limit of –50 [dB] re. direct sound was found.

Threshold limits for the difference curve were determined as best fit from the limited reverberant material available. The criterion does only apply in reverberant conditions, a feature that is further discussed in Section 6.4. Anechoic threshold limits were found from the listening tests, and tested throughout on both single and multiple echoes.

6 RESULTS AND DISCUSSION

The test material can be divided into three parts: Single echoes, multiple echoes and reverberant cases. Results will be presented and discussed in the same order. First, some comments and considerations on the listening test results are appropriate.

6.1 Listening test considerations

Construction properties and test motif background of the listening tests are discussed in Section 4. Test answers of all the test persons that participated can be viewed in Appendix D – Listening Tests and Calculated Results. Invalid test results are drawn through with a dotted red line. This applies to two test sections performed by the author (trumpet and speech motifs), and also a total of one other listening sequence for the trumpet motif. The latter are excluded as the test person felt he/she had evaluated the samples based on wrong criteria. Still a total number of ten tests are performed for each motif.

Results are not discussed in this chapter as such, but some comments on the validation of the results are appropriate. Echo threshold levels are given in Section 6.2. Using much time in the development of the listening tests by adjusting echo levels, parameters and geometries proved successful. As an example is the results from the 400 [ms] delayed single echo case for speech reproduced in Table 11.

Amp. re. direct [Db]	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	D	D+A	N	50 [%]	10 [%]
-34,1		D	D	D	D	D	D	D	D	D	D		100 %	100 %	0 %	D	D
-39,1		A	D	D	A	A	D	A	D	A	D		50 %	100 %	0 %	D	D
-43,1		A	D	D	A	A	D	A	D	A	D		50 %	100 %	0 %	D	D
-46,1		A	N	D	A	A	D	A	N	A	D		30 %	80 %	20 %	A	D
-49,1		N	N	N	N	N	N	N	N	N	A		0 %	10 %	90 %	N	A

Table 11: Example results for 400 [ms] delayed single echo in anechoic conditions for the speech motif.

The important thing exemplified in Table 11 is that all test persons have been consistent in their answers. Subjective variations of how strong an echo must be to be evaluated as disturbing (D) or audible (A) are naturally present. The individual scale used by the different test persons to move from not audible echoes (N) to disturbing echoes (D) will inevitably vary. However, a monotonous increase in percepted echo level can be seen throughout, despite the fact that all test persons have heard the samples in a different randomised order.

A correlation analysis has been done to validate the results from the listening tests. Answers given in each specific sample case are compared. Thus, answers are validated in an additional manner than the monotonous increase in echo disturbance discussed above. Numerical correlation values of all combinations of test persons for all test motifs can be viewed in Appendix E – Correlation Analysis of Listening . The appendix also includes, perhaps more importantly, how each persons answer correlate with the average answer for each test situation. Finally the average level of this figure is given for all test subjects, and also reproduced in Table 12 for convenience. The correlation coefficients range from –1 to 1, where 1 is perfect resemblance. A negative correlation coefficient indicates a tendency of answers developing in opposite and conflicting directions.

	Average correlation coefficient	
	Anechoic	**Reverberant**
Cello	0,76	-
Chorus, female	0,88	0,85
Guitar	0,68	-
Orchestra	0,79	0,22
Speech, male	0,80	0,90
Trumpet	0,75	0,61

Table 12: Average correlation coefficient computed as the average of each persons correlation coefficient with the average answers in each situation.

The correlation coefficients are seen to be satisfactory high. However, the reverberant orchestra case stands out, with a value of only 0,22. This can partially be explained by the fact that only eight reverberant samples where included in the listening test. Deviations in each answer will therefore have larger impact on the total average correlation coefficient. The statistical basis is weak. None of the answers were evaluated as disturbing when demanding 50 [%] disturbance. Answers are sorted with increasing source distance. In a situation where one person answers audible echo (A) in the receiver position closest to the source and another does the same in the situation farthest from the source, while both persons answer not audible echo (N) in the other positions, give a negative correlation coefficient. The analysis is in other words very sensitive. Still, the correlation coefficient is low.

A second correlation coefficient that deviates significantly can be seen in Appendix E – Correlation Analysis of Listening , where test person 4 (TG) could not hear an echo in any of the samples presented for the guitar samples. This lowers the guitar motif correlation to 0,68, as seen from Table 12. The test samples presenting the span were played up twice before starting the test, but no audible or disturbing echoes were detected in any of the test samples presented. Despite this was the test run included, as reducing the selection of the test sequences to be evaluated will be statistically wrong.

6.2 Single echo results

Several studies have been carried out to establish single echo thresholds. Generating the impulse responses using a computer simulation program and reproducing test sequences through headphones differ from previous research. The accuracy in generating simple impulse responses using computer simulations in anechoic conditions is excellent, as no approximations in calculating reflections and sound paths deteriorate it.

Echo thresholds from the listening test for 50 [%] disturbance and from Dietsch and Kraak are shown in Figure 32 for speech. The motifs used are quite similar. Dietsch and Kraak employed male speech of 4,6 syllables/sec (duration 6,5 [s]) in determining their threshold values, while the speech motif used in the present paper is male speech of 4,0 syllables/sec (duration 8,0 [s]). To assume a similar frequency content thus seem reasonable. All three curves have similar shape, the threshold from the present listening test being about 5 [dB] less echo critical for delays longer than 100 [ms] than the analytic and ISM calculated values using Dietsch and Kraak. Decreasing the delay time increase the difference between the two curves, reaching about 10 [dB] around 50 [ms] delay, before the values close in again approaching 30 [ms] delay. Threshold values calculated using wave-files as input to the Dietsch and Kraak algorithm are higher than ISM calculated ones. The HRTF filter causes this, as the direct sound no longer has no impact on the build up of the nominator of Equation (5). The direct sound therefore lowers the effect of later arriving reflections on the build-up function. Consequently, echo thresholds are increased.

Due to the unavoidable error introduced by using discrete steps in the echo amplitudes, and not continuous adjustable echo amplitudes in the listening test, a certain quantization error apply. The same quantization is used in establishing values for the Dietsch and Kraak echo criterion when wave-files are used as input. Nevertheless, clear tendencies can be seen.

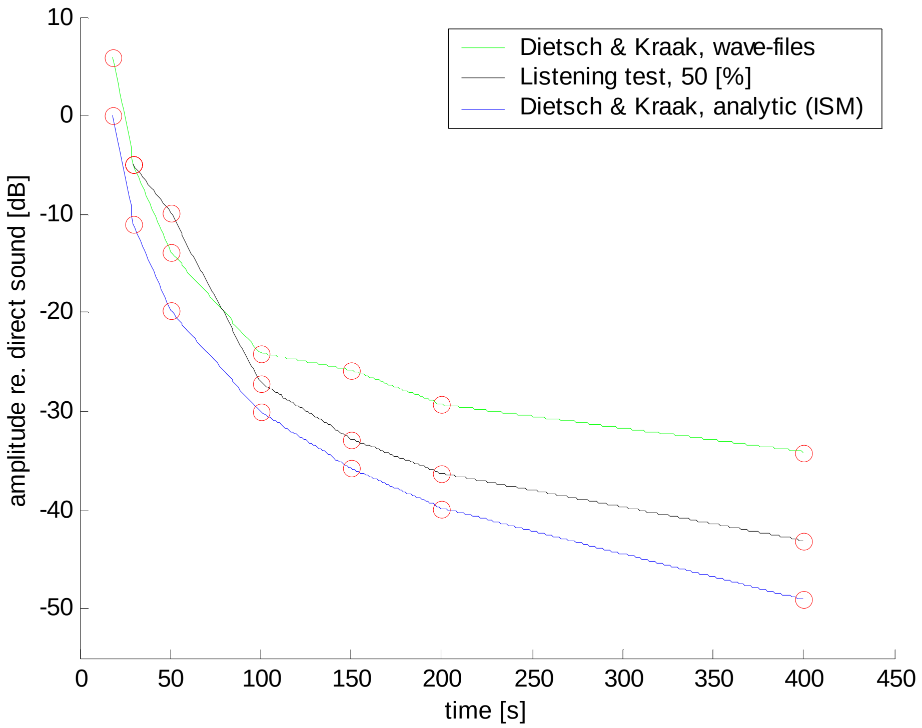

Figure 32: Single echo thresholds for speech.

A small proportion of the deviation can be traced back to the different speed of the speech motifs. Natural breaks in the speech sequence and transient consonants also affect echo sensitivity. No threshold limit can be determined with a delay of 18 [ms] from the present listening test. Dietsch and Kraak reported echo thresholds also with only 18 [ms] delay. 50 [%] of the test persons feel disturbed by an echo with the same amplitude as the direct sound, as can be seen from Appendix D – Listening Tests and Calculated Results. But as the amplitude of the echo gets higher or lower is no echo disturbance reported. The same tendency can be seen throughout for the music motifs as well (with delays of 30 and 50 [ms]), but not reaching higher disturbance than 20 [%] in cases where this phenomenon can be seen. Weaker echo reflections will cause it to be masked by the direct sound, while stronger echo reflections introduce backward masking, and the direct sound is masked. Both situations lead to the same result; no disturbing echo is percepted.

A bigger variety of test motifs were included to investigate music thresholds, as discussed in Section 4.1. Figure 33 shows the resulting threshold values for the different motifs. Test results vary greatly with different instruments and music motifs, as expected. Dietsch and Kraak based their choice on the most echo critical motif in their study. Test results once again seem relatively similar, if the trumpet motif is excluded. The guitar prove good resemblance with Dietsch and Kraak`s analytic values on the interval 100-400 [ms] echo delay. Results from wave-file input to Dietsch and Kraak`s echo criterion show the same increase in threshold levels as discussed above.

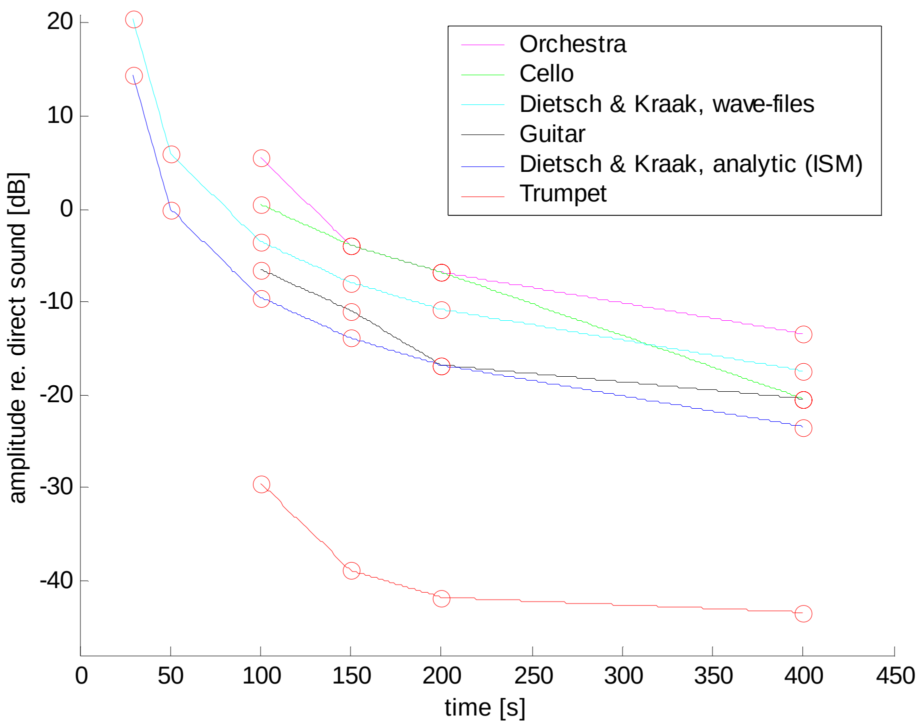

Figure 33: Single echo thresholds for music.

The dynamic properties of the trumpet visually appear to be more critical for echoes than speech, as can be seen from Appendix D – Listening Tests and Calculated Results. However, our perception of echoes is increased by the fact that speech carries a more precise and distinct message. The resulting echo levels for the present motifs are roughly equal within the valid trumpet delay interval of 100-400 [ms]. From the properties of the different motifs seen in Appendix A – Test Motifs, does the female chorus show similar dynamic properties as the trumpet, and can be expected to have come out with similar values. The guitar can be seen to have a reverberant tail, which is very prominent with the present guitar motif. Moving further upwards from the most echo critical motif in Figure 33, can the cello motif be seen to have a dynamic origin, but the reverberance of the body introduce a strong masking effect. The same is true for the orchestra test motif.

A criterion must always be based on the most critical situation. In this case, the most critical motif is the trumpet. An equally sensational aspect of the test results presented in Figure 33 is that no disturbing echoes were reported for delays below 100 [ms], regardless of amplitude. It can be inferred that no test were performed for delays between 50 and 100 [ms], but a certain quantization was inevitable. Disturbance percentages do not rise above 20 [%] except for the guitar motif where 40 [%] were disturbed by the strongest echoes added for 50 [ms] delays.

A second situation described in Section 3.1.2 generates a single echo reflection from a single wall reflection. The back wall of the concert hall used to generate the reverberant music situations was given to different scattering coefficients, and impulse responses were recorded at a selection of the receiver positions. Only the trumpet, female chorus and orchestra motifs were used. Complete results are given in Appendix D – Listening Tests and Calculated Results, and the main figures are reproduced in Table 13 for convenience.

	Scattering coefficient 0,1						Scattering coefficient 0,7					
	Test results						Test results					
Receiver	Trumpet	Chorus	Orchestra	D&K ISM	D&K wave-file	New Criterion	Trumpet	Chorus	Orchestra	D&K ISM	D&K wave-file	New Criterion
17	D	D	A	N	N	D	D	D	-	N	D	D
1	D	D	A	N	N	D	D	D	-	N	D	D
7	D	D	N	N	D	D	D	D	-	N	D	D
9	-	**D**	-	N	D	D	-	D	-	N	D	D
11	A	N	-	N	N	D	**D**	A	-	N	D	D
13	A	N	-	N	N	N	A	N	-	N	D	D
14	N	-	-	N	N	N	N	-	-	N	N	N
15	N	-	N	N	N	N	N	N	-	N	N	N

Table 13: Results from single echo generated by a single wall reflection.

Receiver positions included in testing the orchestra motif was limited, as no disturbing echoes could be heard in the preliminary listening tests. As for single echoes generated by an additional source, test answers are remarkably consistent. The trumpet and female chorus motifs gave similar results. The receiver closest to the back wall that was evaluated as disturbing is shown in bold in Table 13. This is in agreement with the classical integration time of the human ear (50 [ms]). Receiver 9 and 11 are 11 and 7 [m] from the back wall, which are delays of approximately 65 and 47 [ms], respectively.

The situation with a single echo caused by a wall reflection proved to be a situation Dietsch and Kraak`s criterion did not handle well. EK values were highest at positions in the middle of the hall, decreasing towards the back wall and the stage (values can be viewed in Appendix D – Listening Tests and Calculated Results). The ISM calculation did not give a single echo situation. Using wave-files as input gave good results with a scattering coefficient of 0,7, but did not work so well with a scattering coefficient of 0,1. Despite giving satisfying results in some situations, is it not good that echo values decrease towards the stage from receiver positions in the middle of the concert hall. Subjective values increase with distance to the back wall. Sound waves will only be attenuated as function of distance, with one reflection from the back wall and otherwise anechoic conditions. The attenuation due to distance is minimal in the present situation.

That more people are disturbed with the bigger scattering coefficient is not intuitive either. It is probably caused by calculation approximations in Odeon, where larger scattering coefficients secure a larger amount of the incident energy to be reflected back to the receivers. Lower scattering coefficients cause energy to be reemitted in larger portions in certain directions. The same tendency can be seen from the listening tests, Dietsch and Kraak`s criterion, and also from thresholds from the new criterion.

The impulse responses have been convolved with the Hanning window of Figure 26 before they are compared with threshold limit values shown in Figure 33 that also have been convolved with the same window. This is done to account for the integration time of the ear. However, as shown in Figure 33, no echo thresholds are given below 100 [ms]. The integration window still adds a width equal to the integration window to all reflections.

6.3 Multiple echo thresholds

Our perception of successive echoes includes a huge possibility of parameter variations. Directional dependencies are not considered, and all echoes are generated at an angle of 180° azimuth and 0° elevation. Incident reflections arriving at receiver positions after being reflected from the back wall or generated by loudspeakers are common. A detailed explanation of the test situation is given in Section 3.2, and echo delays are summed up in Table 4. In addition the speech motif are only two music test motifs (cello and trumpet) included in the multiple echo test due to the extent of the test sections. Disturbance levels of 50 [%] are used throughout in the listening test results presented. Numerical results from the multiple echo tests can be viewed in Appendix D – Listening Tests and Calculated Results.

Values from multiple echo tests with the speech motif are reproduced in Table 14. All values are in [dB] re. direct sound level. In evaluating the new criterion have all impulse responses been convolved with the Hanning window. Separation times are given in the leftmost column, indicating separation times between the three echoes generated in each case. The first of the three echoes is delayed by the time seen in the top row of the columns. A separation time of 0 indicate the single echo case, with one echo arriving after 18, 100 or 200 [ms], respectively.

No disturbing echoes were reported in the listening tests until echoes were separated by a minimum of 25 [ms] when the first echo was 18 [ms] delayed. Once again, an integration time of the ear of approximately 50 [ms] is verified. Dietsch and Kraak`s echo criterion give perceptible echo limits for all delays, as for the single echo cases, which are erroneous. Wave-file input to the Dietsch and Kraak criterion results in the highest thresholds. This tendency was also seen in the single echo test results.

The new criterion is based on limit values obtained from the single echo tests in anechoic conditions. Echo thresholds of the new criterion are not broken until the three successive echoes are separated by a minimum interval of 25 [ms] with a delay of the first echo of 18 [ms], where the threshold amplitude is 3 [dB] higher than the result from the listening tests. A comparison between the results from the new criterion and the listening test show good resemblance, deviations of more than one quantization step do not occur.

	18 [ms]				100 [ms]				200 [ms]			
Separation [ms]	**Listening test**	**D&K ISM**	**D&K wave-file**	**New Criterion**	**Listening test**	**D&K ISM**	**D&K wave-file**	**New Criterion**	**Listening test**	**D&K ISM**	**D&K wave-file**	**New Criterion**
0	-	0,0	3,0	-	-27,1	-30,1	-24,1	-27,1	-36,8	-39,8	-29,8	-36,8
4	-	-19,0	-9,0	-	-25,8	-40,8	-33,8	-30,8	-39,8	-48,8	-39,8	-39,8
8	-	-16,5	-11,5	-	-31,3	-38,3	-31,3	-31,3	-35,0	-49,0	-35,0	-35,0
10	-	-17,6	-12,6	-	-34,6	-28,6	-25,6	-31,6	-35,2	-35,2	-30,2	-40,2
25	-18,7	-12,7	- 8,7	-15,7	-33,3	-28,3	-23,3	-28,3	-41,1	-36,1	-31,1	-36,1
50	-28,1	-28,1	-25,1	-28,1	-35,7	-35,7	-30,7	-35,7	-37,5	-42,5	-32,5	-37,5

Table 14: Results for the speech motif in triple echo situations (50 [%] disturbance thresholds). Separation times are given to the left, first reflection delayed as set on top of the columns. Separation time of 0 indicate single echo situation. All values are given in [dB] re. direct sound unless otherwise stated.

Figure 34 shows a graphical plot of the results given in Table 14 for the cases when the first echo is delayed by (a) 100 [ms] and (b) 200 [ms]. The bold dotted red lines, as seen from the legend, show reference values obtained from the listening tests. Threshold values are seen to decay slowly with increasing separation time, as expected. Dietsch and Kraak`s criterion once again show higher values with wave-file input, especially with longer separation times.

The variations in threshold values for separation times of 4, 8 and 10 [ms] computed by the Dietsch and Kraak criterion are the most evident feature of Figure 34. Especially the differences observed by only increasing the separation time by 2 [ms] (from 8 to 10 [ms]) are conspicuous. The large variations can be traced back to the rectangular evaluation window lengths employed, as explained in Section 5.1. Scaling values are either 0 or 1, introducing a large quantization step. Incident energy are either evaluated by its full strength if it has arrived within the given integration time, or it is not considered at all. Steps of up to 14 [dB] can be seen in Figure 34 (b) (ISM implementation). Differences are not so severe when wave-files are used as input, but still clearly present.

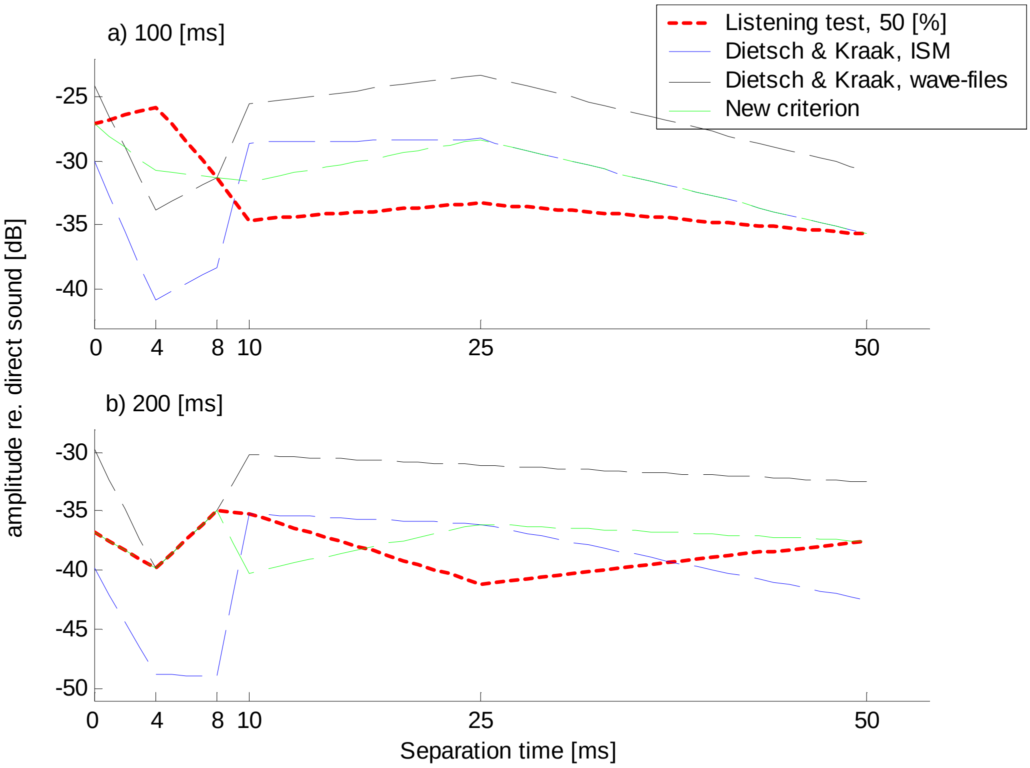

Figure 34: Echo thresholds as function of separation time between the three successive echoes for speech. First echo delayed by (a) 100 [ms] and (b) 200 [ms].

Figure 35 shows the effect of using a finite length rectangular window. Three different separation times are shown, with the first echo arriving 200 [ms] after the direct sound. Separation times are seen from the figure. The window length for speech motifs determined by Dietsch and Kraak is 9 [ms]. Consequently will all three echoes be evaluated within the integration window for a short time interval if they are separated by 4 [ms] (a), only will be inside the window with a separation time of 8 [ms] (b), while finally neither of the echoes will be inside the window simultaneously if echoes are 10 [ms] apart (c). Such abrupt time separation limits are clearly non-physical. Another feature of any function based on build-up (center time) functions also evident from Figure 35, is the reduced EK levels of later arriving peaks due to earlier energy contributions.

The number of music motifs was reduced compared with the single echo tests due to the extent of the multiple echo listening tests. Only tests with the trumpet and cello motifs were carried out. Table 15 shows an extract of the values from multiple echo tests with music, a complete set of numerical results from all criteria and the listening tests can be viewed in Appendix D – Listening Tests and Calculated Results. All values are in [dB] re. direct sound level unless stated otherwise.

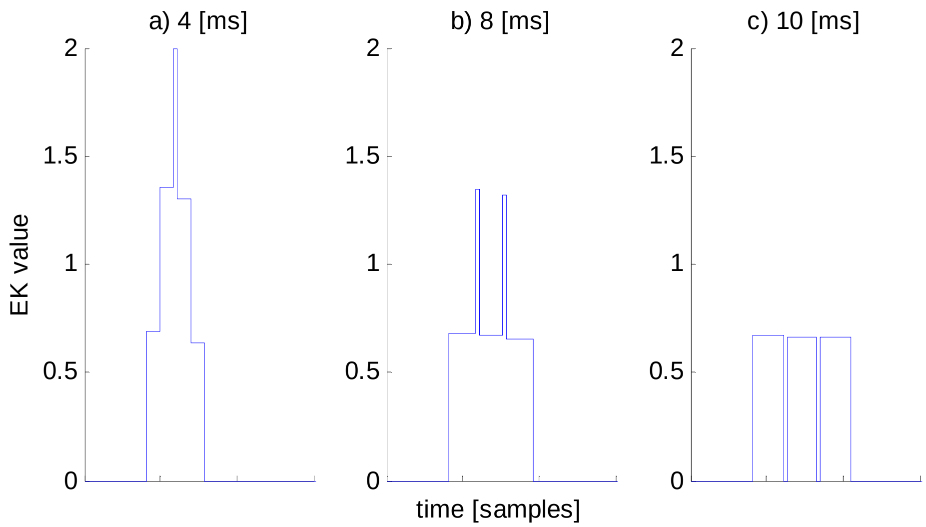

Figure 35: Effect of rectangular finite length integration windows ($\Delta\tau_E$). Three different separation times between three echoes are shown. First echo arriving 200 [ms] delayed re. direct sound.

Results are calculated and shown in the same way as explained above for the speech motif. Delays of 18 [ms] are not included as no echoes were reported in the trumpet listening tests. However, listeners reported disturbing echoes for the cello test motif with a separation time of 50 [ms] at much higher echo amplitudes (echoes only 2 [dB] below direct sound level, which is very unlikely). Amplitudes need to be unlikely strong if echoes are to be detected, also for Dietsch and Kraak`s criterion. It can therefore be claimed that Dietsch and Kraak`s echo criterion have a certain validity for short delayed multiple echoes of musical origin.

	100 [ms]					200 [ms]				
Separation [ms]	**Listening test, trumpet**	**Listening test, cello**	**D&K ISM**	**D&K wave-file**	**New Criterion**	**Listening test, trumpet**	**Listening test, cello**	**D&K ISM**	**D&K wave-file**	**New Criterion**
0	-29,5	0,5	-9,5	-3,5	-29,5	-41,8	-6,8	-16,8	-19,8	-41,8
4	-33,1	-6,1	-16,1	-13,1	-33,1	-48,3	-12,3	-22,3	-22,3	-45,3
13	-33,9	-6,9	-13,9	-9,9	-33,9	-42,7	-12,7	-22,7	-15,7	-42,7
15	-34,1	-	-7,1	-4,1	-34,1	-42,8	-12,8	-15,8	-12,8	-42,8
25	-34,8	-7,8	-7,8	-4,8	-37,8	-46,2	-13,2	-16,2	-16,2	-40,2
50	-39,5	-9,5	-4,5	-4,5	-39,5	-43,2	-16,2	-13,2	-13,2	-40,2

Table 15: Results for the trumpet and cello motifs in triple echo situations. Separation times are given to the left, first reflection delayed as set on top of the columns. Separation time of 0 indicate single echo situation. All values are given in [dB] re. direct sound unless otherwise stated.

The results reproduced in Table 15 divide themselves into two different ranges for both 100 and 200 [ms] delays. Listening test thresholds with the trumpet motif resemble well with the new criterion, while both Dietsch and Kraak calculated curves lie in the vicinity of results from listening tests with the cello motif. The same tendency was observed for single echoes in Section 6.2. Plots of the echo amplitude values of Table 15 are given in Figure 36.

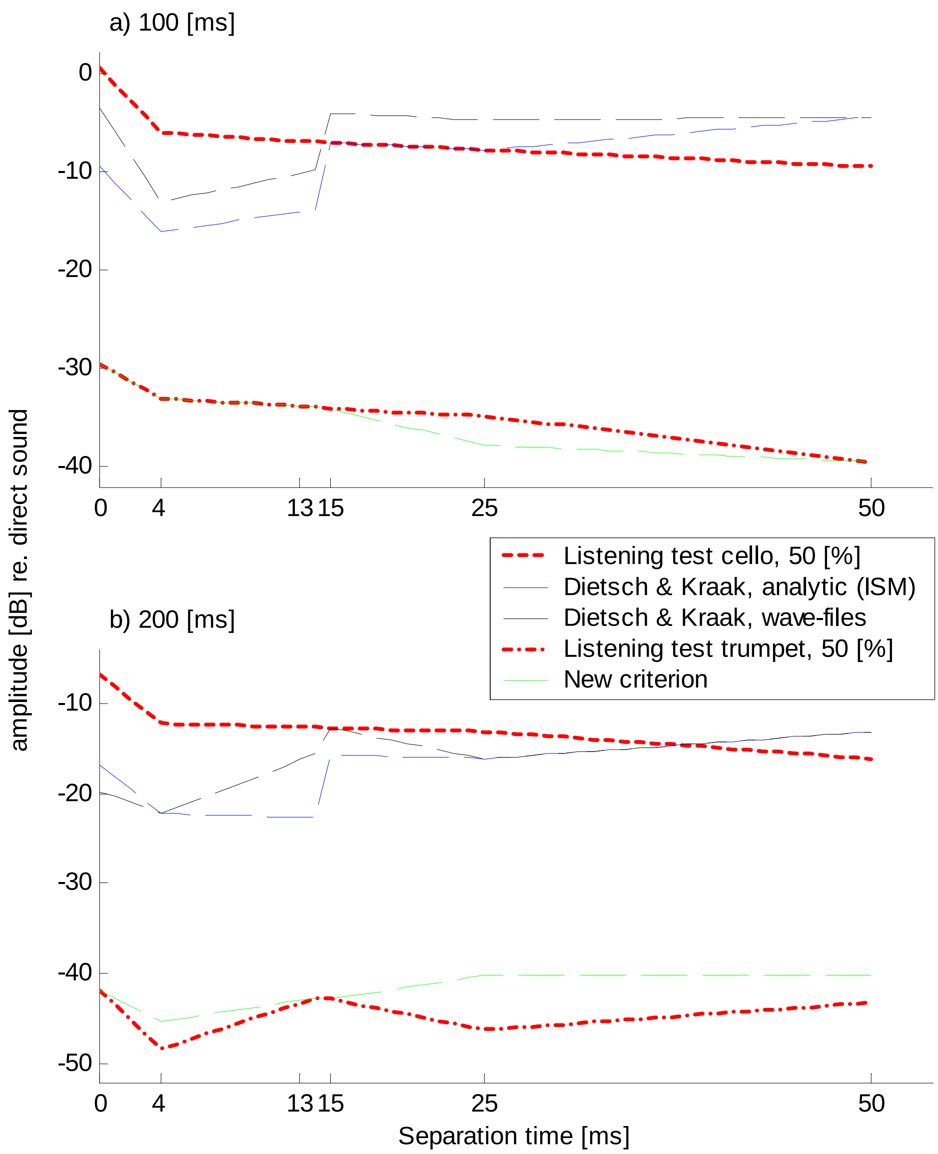

Figure 36: Echo thresholds as function of separation time between the three successive echoes for music motifs. First echo delayed by (a) 100 [ms] and (b) 200 [ms].

It is a reasonable assumption that Dietsch & Kraak have set their limit values based on far less echo critical motifs than the trumpet motif included in the present tests. Abrupt steps in echo thresholds caused by the rectangular integration window are also evident from Figure 36. The window length used for music motifs deviates from the length used for speech motifs, being 14 and 9 [ms], respectively (see $\Delta\tau_E$ in Table 1, Section 2.3). Separation times are varied accordingly, to highlight the effect of the rectangular window.

Except for the sudden step caused by the integration window are threshold values seen to be relatively constant for the different separation times. However, a slight negative gradient can be seen for most of the curves in Figure 36 (and Figure 34). Increasing separation times compensate for lower echo thresholds, resulting in slowly decaying threshold limits, as seen for single echoes.

The good resemblance between the new criterion and the trumpet listening tests are no surprise. Single echo thresholds from the trumpet listening test were used set the limit curves of the new criterion. Consequently will results be within a certain range. The main feature evaluated when the criterion are applied to a set of multiple echo situations is the length of the Hanning filter used to simulate the inertia of the human ear. Once again an integration time of approximately 50 [ms] are verified. However, the criterion is only valid in anechoic conditions.

		Dietsch and Kraak echo criterion													
Scat. coeff.	**Rec.**	**Listening test**		**1st order**		**2nd order**		**3rd order**		**4th order**		**Wave-files**		**New criterion**	
			%		**EK**		**EK**		**EK**		**EK**		**EK**		**Max**
10	1	D	50	N	0,64	N	0,94	D	1,19	D	1,19	N	0,89	D	5,9
	2	D	100	N	0,54	N	0,85	D	1,00	D	1,12	N	0,75	D	7,7
	3	D	90	N	0,41	N	0,66	N	0,86	N	0,86	N	0,68	D	9,0
	4	D	80	N	0,40	N	0,58	N	0,77	N	0,93	N	0,62	D	6,8
	5	D	100	N	0,44	N	0,59	N	0,64	N	0,65	N	0,64	D	9,2
	6	D	90	N	0,41	N	0,67	N	0,70	N	0,70	N	0,60	D	10,1
50	1	N	0	N	0,61	N	0,88	D	1,09	D	1,09	N	0,77	N	2,6
	2	N	0	N	0,51	N	0,83	D	1,01	D	1,01	N	0,66	N	2,3
	3	N	20	N	0,65	N	0,87	N	0,87	N	0,87	N	0,75	N	2,8
	4	N	0	N	0,43	N	0,63	N	0,69	N	0,70	N	0,72	N	3,2
	5	N	40	N	0,49	N	0,65	N	0,71	N	0,70	N	0,72	N	4,8
	6	D	50	N	0,45	N	0,68	D	1,19	N	0,71	N	0,65	D	5,5

Table 16: Echo disturbance in the auditorium with the speech motif. (D = disturbing, N = no echo percepted).

6.4 Results from reverberant conditions

Single echoes form a theoretical threshold limit, and can aid in understanding fundamental properties of the ear and our annoyance due to a small number of incident reflections. However, reverberant conditions apply in most practical and commercial cases. As concert halls, auditoria or other reverberant rooms are planned and constructed, would it be beneficial to have an algorithm detecting possible echoes from the room impulse response obtained from computer simulations.

Properties of the listening test are discussed in Section 4, and the final test material is summed up in Table 10. Three test motifs are used in generating test sequences from the concert hall (trumpet, female chorus and the orchestra). The trumpet and female chorus motifs are tested more extensively than the orchestra motif. The speech motif is simulated in the auditorium. All results can be viewed in Appendix D – Listening Tests and Calculated Results.

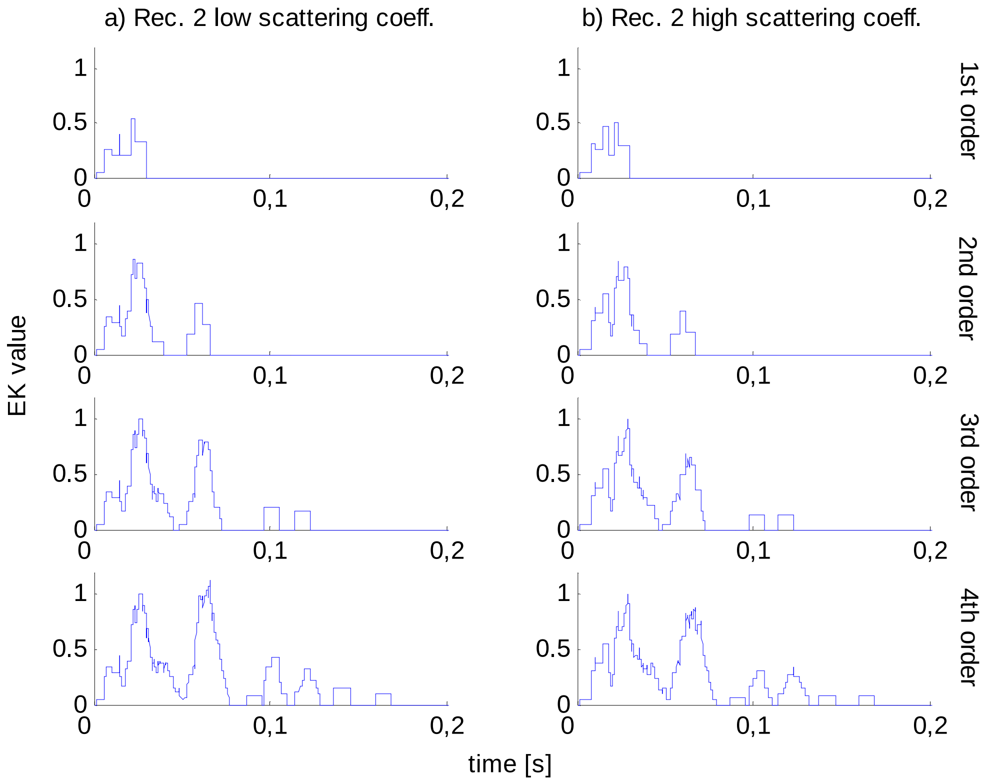

Figure 37: EK values computed with different orders of the image source method. Computations with low and high scattering coefficients on side and back walls of the auditorium are shown in (a) and (b), respectively.

6.4.1 Auditorium results

Echo disturbance levels obtained from the reverberant speech tests are reproduced in Table 16. The geometry of the auditorium is shown in Figure 19 and properties of surfaces and receiver positions listed in Table 5 and Table 6 respectively.

Dietsch and Kraak`s echo criterion was implemented in MatLab for both txt-file input based on purely ISM, and for wave-file input based on ISM and ray tracing. txt-files based on ISM calculated impulse responses were generated to investigate possible simplifications of criteria. The order of ISM was varied from 1-4. Figure 37 (a) and (b) show the development of EK as an increasing number of image source orders are added for the same receiver, but with two different sets of scattering coefficients. The properties of the surfaces in the given examples can be viewed in Table 5, case 1 being (a) and case 2 (b).

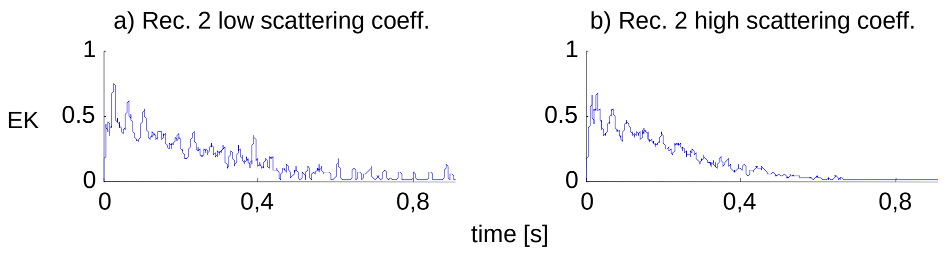

Figure 38: EK values computed from Odeon wave-file output. Computations with low and high scattering coefficients on side and back walls of the auditorium are shown in (a) and (b), respectively.

Dietsch and Kraak proved poor resemblance with the listening test results. It is clear that severe errors occur as the ISM order is reduced. Neither do the EK values change correctly compared with the subjective disturbance percentages, so adjustments of EK limits will not prove adequate.

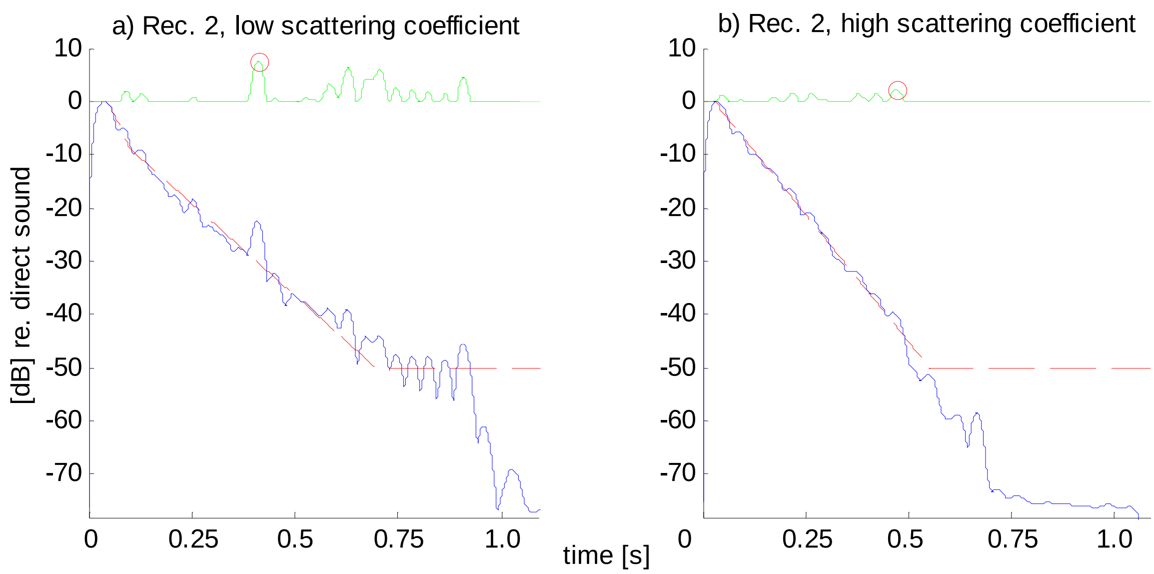

Figure 39: Results from receiver 2 in the auditorium calculated using the new criterion. The surfaces have low (a) and high (b) scattering coefficients, respectively.

EK values computed from Odeon wave-file output for the corresponding situations as depicted in Figure 37 are shown in Figure 38. Notice the different scaling on the axis. As the complete calculation is included do both EK-curves have a longer time span than with the ISM only. EK values can be seen to be lower with the higher scattering coefficients, especially in the interval 0,4-0,8 [ms]. Still, results deviate from the listening tests.

Figure 39 shows results calculated using the new criterion. The green line in the upper part of the figure shows positive deviations from the dotted red threshold line. The red circle indicates the maximum positive deviation, which is the most echo critical part of the decay. Lower scattering coefficients on surfaces lead to higher reverberation times as wall reflections become more specular and less energy is absorbed or transmitted. The reverberation times of the auditorium with the two parameter sets of Figure 39 a) and b) are 0,91 and 0,67 [s], respectively.

		Listening test			Dietsch and Kraak echo criterion												
Scat. coeff.	Receiver		Trumpet %	Female chorus %	1st order	EK	2nd order	EK	3rd order	EK	4th order	EK	Wave-files	EK	New Criterion	Max
10	17	D	70	80	N	0,43	N	1,02	N	1,04	N	1,02	N	1,00	D	10,1
	1	D	50	10	N	0,78	N	1,23	N	1,29	N	1,37	N	0,97	D	7,5
	4	N	40	-	N	0,73	N	1,03	N	1,06	N	1,14	N	0,65	D	7,5
	7	N	20	0	N	0,72	N	0,9	N	0,93	N	0,93	N	0,75	D	7,4
	10	D	-	90	N	0,64	N	0,95	N	0,98	N	0,98	N	0,81	D	12,0
	11	D	70	80	N	0,63	N	0,94	N	0,97	N	0,97	N	0,94	D	11,3
	13	D	10	80	N	0,62	N	0,87	N	0,89	N	0,89	N	0,88	D	10,9
	16	D	60	90	N	0,42	N	0,64	N	0,71	N	0,71	N	0,75	D	8,4
70	17	D	10	60	N	0,43	N	0,61	N	0,73	N	0,73	N	1,49	N	3,5
	1	N	20	10	N	0,80	N	1,07	N	1,11	N	1,11	N	1,59	N	6,3
	4	N	0	-	N	0,57	N	0,89	N	0,89	N	0,89	N	0,97	N	3,4
	7	N	10	0	N	0,67	N	1,00	N	1,03	N	1,03	N	1,03	N	2,9
	10	D	20	70	N	0,69	N	1,00	N	1,03	N	1,03	N	1,05	D	8,7
	11	N	-	0	N	0,68	N	0,99	N	1,02	N	1,02	N	0,99	N	4,6
	13	N	10	0	N	0,65	N	0,89	N	0,92	N	0,92	N	0,93	N	3,9
	16	D	10	70	N	0,50	N	0,71	N	0,75	N	0,75	N	0,78	D	9,2

Table 17: Echo disturbance in the concert hall with music motifs. (D = disturbing, N = no echo percepted). Disturbance percentage >= 50 [%] = D, < 50 [%] = N.

A visual inspection of Figure 39 a) and b) indicates that a) should be the most echo critical one. A distinct peak can be seen at a delay of approximately 0,4 [s], which also is the largest positive deviation from the evaluation curve. This peak is not present with the larger back wall scattering coefficient set. The repeated peaks seen below the hearing threshold are flutter echoes between the sidewalls of the auditorium. An iterative approach was used to determine the threshold value of positive deviation from the evaluation curve. It is set to 5 [dB] based on test results from the auditorium. However, it is increased to 7 [dB] when RIR`s and listening tests from the concert hall form the basis of evaluation. A difference probably not caused by the fact that a speech motif were used in the listening tests in the auditorium, while music motifs were employed in the concert, rather the deviating reverberance times.

Maximum deviations calculated with the different receivers and surface parameters are given in the leftmost column of Table 16. Comparing threshold values with disturbance percentages obtained from the listening tests show similar trends. If the disturbance level is high, is the peak deviation correspondingly high. Results from the new criterion and from the listening tests give equal results for all the tested receiver positions and scattering parameters.

6.4.2 Concert hall results

Echo disturbance levels obtained from the reverberant music tests are reproduced in Table 17. Properties of surfaces and receiver positions are listed in Table 7 and Table 3, respectively. The geometry of the concert hall and receiver and source positions are shown in Figure 20.

As for the auditorium are all responses simulated with varying orders of the ISM and also with wave-file output from Odeon. Dietsch and Kraak`s echo criterion did not predict an echo for any of the simulated situations, and consequently corresponded poorly with the listening test results. Results from the orchestra test motif are not included in Table 17.

Evaluations of all situations in the listening tests are done with respect to the motif where the higher disturbance percentage was observed. As the orchestra test motif is far less echo critical than the trumpet and female chorus motifs, is no result included. Disturbance percentages exceeding the required 50 [%] were not found in any situations. The female chorus and trumpet motifs have roughly the same dynamic properties. Conducting tests based on both motifs are important to remove time dependencies within each motif masking possible echo reflections. As two approximately equally echo critical motifs are used, the possibility that time dependencies in both motifs mask the same reflections in a given room is thus reduced. Rhythmic properties may explain large disturbance differences between motifs (see Table 17).

Figure 40 (a) and (b) shows the development of EK as an increasing number of image source orders are added for two different sets of scattering coefficients. The properties of the surfaces in the given examples can be viewed in Table 7. The same tendencies as in Section 6.4.1 are evident: Severe errors occur as the ISM order is reduced, and the EK values do not change correctly compared with the subjective disturbance percentages. Another feature more evident with the concert hall RIR`s shown is the effect of the rectangular integration window, which can most clearly be observed from the 2nd order case of Figure 40 a).

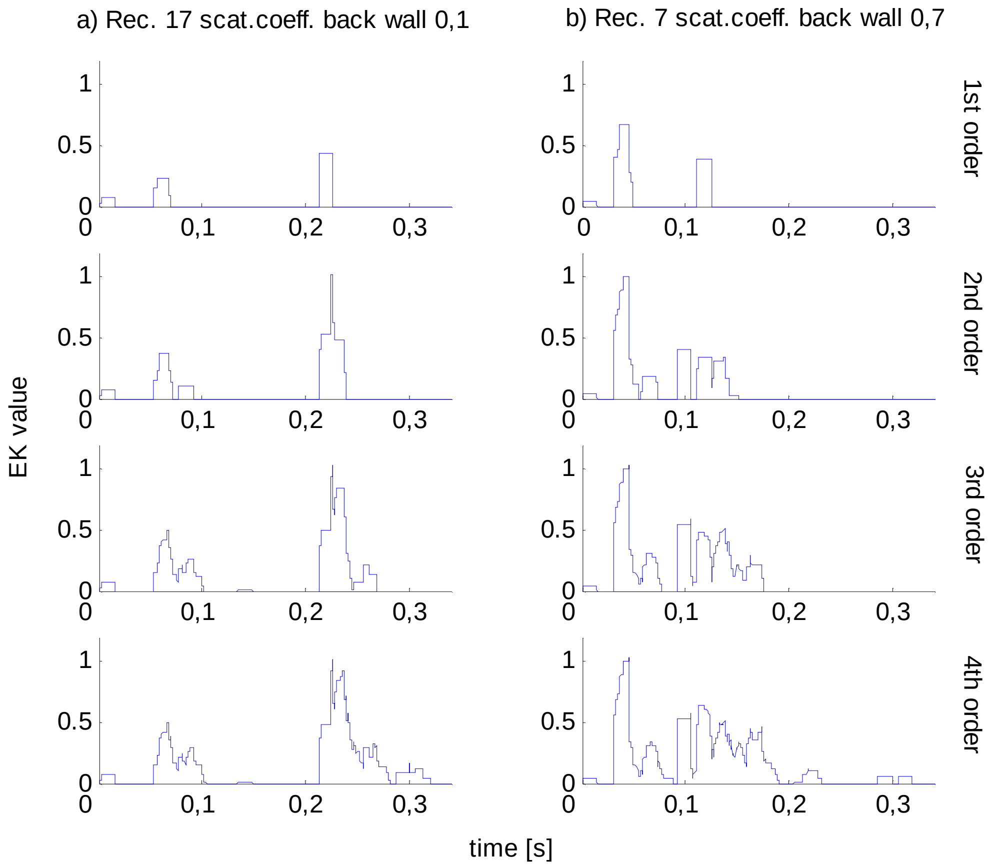

Figure 40: EK values computed with different orders of the ISM. Computations with low and high scattering coefficients on side and back walls of the auditorium are shown in (a) and (b), respectively.

EK values computed from Odeon wave-file output for the corresponding situations as depicted in Figure 40 are shown in Figure 41. Notice the different scaling on the axis between the figures. Inclusion of ray tracing makes the RIR`s longer than with ISM only. An intuitive analysis of the two curves in Figure 41 based on visual inspection shows that is it a) that are most echo critical. A distinct peak can be seen after approximately 0,3 [s], but since no EK thresholds are broken will no echo be predicted from the algorithm.

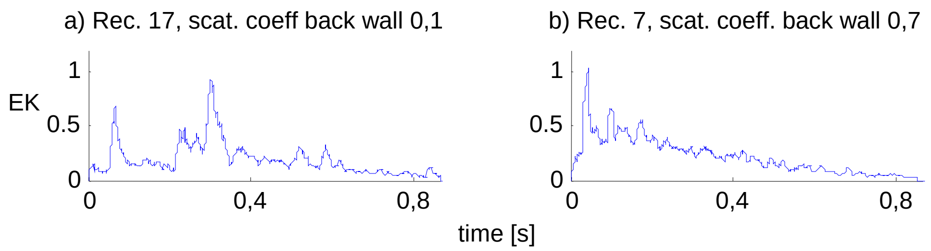

Figure 41: EK values computed from Odeon wave-file output. Computations with low and high scattering coefficients on the back wall of the auditorium are shown in (a) and (b), respectively.

Figure 42 shows example results calculated by the new criterion. The reverberation times of the concert hall with the two parameter sets (see Table 7) for situation a) and b) are 1,82 and 1,47 [s], respectively. The threshold value is determined iteratively to be 7 [dB]. Maximum deviations calculated with the different receivers and surface parameters are given in the leftmost column of Table 17.

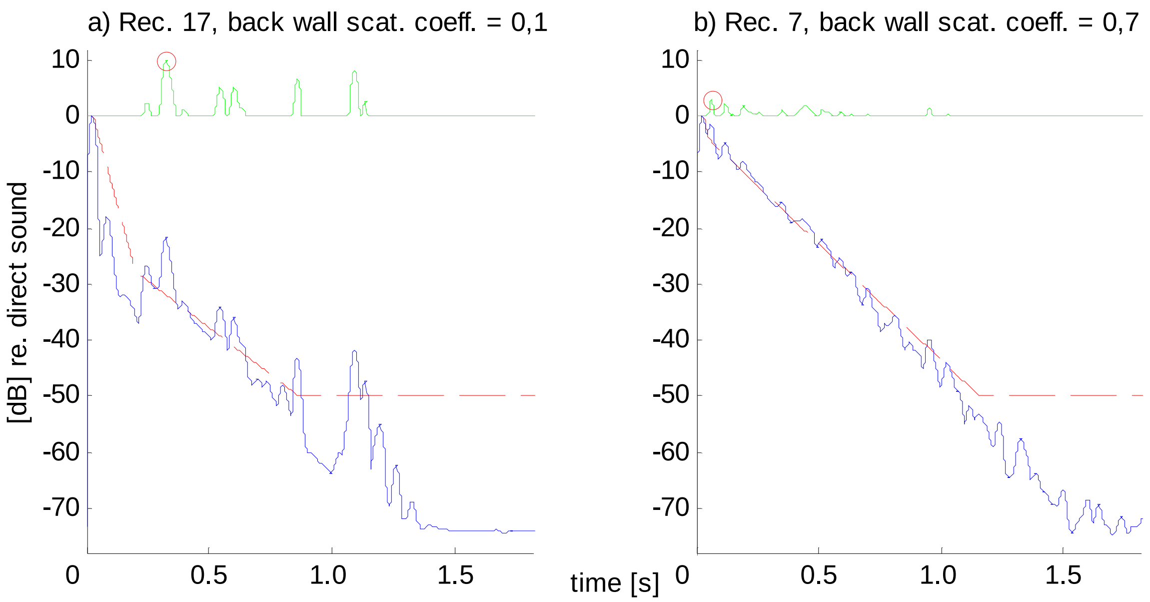

Figure 42: Results from receiver 17 and 7 in the concert hall calculated using the new criterion. The back wall has scattering coefficients of 0,1 (a) and 0,7 (b).

Results from the new criterion and from the listening tests give equal results for 13 of the 16 tested situations (81 [%]). The effect of letting the evaluation curve follow the reverberation curve of the ear from the direct sound peak until it crosses the shifted reverberation curve of the room is evident from Figure 42 a). Receiver 17 simulates the conductor, who often gets a strong direct sound contribution, as the distance to the orchestra is short. The same situation applies for a single musician or the musicians in orchestras on stage. Problems and causes of the three situations that deviate from the listening test result are discussed in the following section.

6.4.3 Problems and shortcomings of the new criterion

The new criterion did not prove 100 [%] percent equal results as the listening tests. Two of the three situations predicted disturbing echo when the listening tests indicated the opposite. Consequently was the disturbing threshold limit not broken in the last erroneous situation, while test persons felt disturbed. The case where echoes were falsely predicted is discussed first. Figure 43 shows the two erroneous situations, where the maximum deviations are 7,5 and 7,4 [dB] in (a) and (b), respectively.

The similarity between Figure 43 (a) and (b) is conspicuous. The maximum positive deviation from the evaluation curve occurs in the vicinity of the transition from the shifted reverberation curve of the room to the hearing threshold limit. It is hardly the lower threshold level that causes the error, rather the preceding transition from the room reverberation decay. The lower level of the evaluation curve is optimized iteratively from the reverberant situations available, and is therefore validated to the degree possible. Odeon calculate reverberation times as T_{30}, based on the decay from –5 to –35 [dB] re. direct sound. That the same reverberation decay is valid at lower levels is no matter of course. A more gentle decay of the evaluation curve from the last –35 [dB] re. direct sound sample to the lower limit would lower the exceeding peaks.

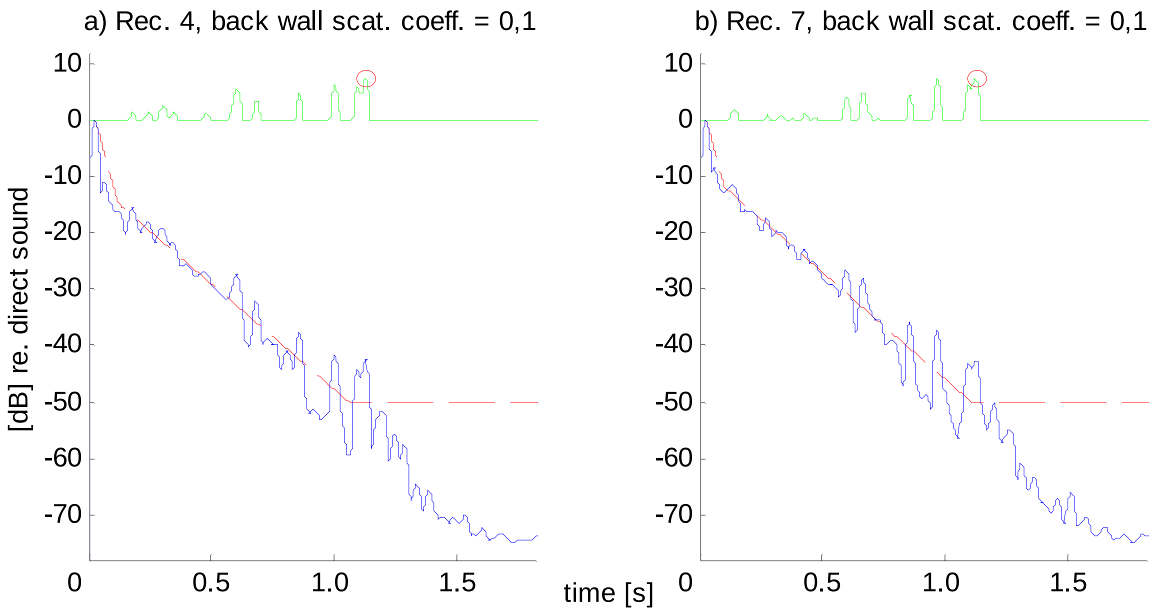

Figure 43: Situations where the new criterion erroneously predicts a disturbing echo.

Receiver position 17 with a scattering coefficient of the hall back wall of 0,7 was the only situation where test persons reported a disturbing echo, while the new criterion failed to predict so. Figure 44 shows the calculated response. The maximum positive deviation level is only 3,5 [dB], but the probable cause of the erroneous echo prediction is the peak arriving directly after the direct sound. It is partly masked by the reverberation decay of the ear. Small variations in either reverberation time or Hanning window length will have decisive impact on the deviation level between the evaluation curve and the filtered RIR, as would adjustments in the reverberation time of the ear.

Frequency content of the impulse responses is not considered except for the natural filtering occurring in wall reflections and in the HRTF`s. Reverberation times can vary significantly between octave bands. Also, disturbance rates of 10 and 60 [%] with the female chorus and trumpet motif, respectively, show that the response shown in Figure 44 lies at the boundary of the 50 [%] limit. A faultless criterion is not to be expected from the somewhat limited set of reverberant situations available and used. Despite the limited reverberant test set are results good, being 100 [%] and 81 [%] in the auditorium and concert hall situations, respectively.

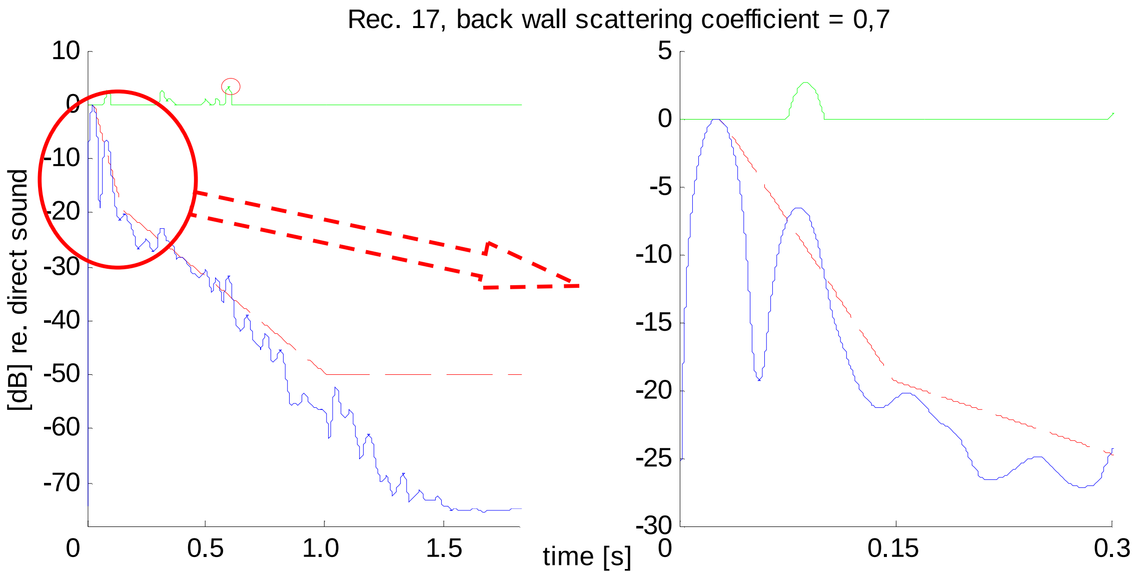

Figure 44: **Situation where test persons felt disturbed by an echo, while the new criterion fails to predict echo.**

6.5 Further work and developments of the criterion

There are many possible improvements to be investigated for the new criterion. The preceding section pointed out and discussed some of them.

(1) Development of the transition between the shifted reverberant decay and the lower hearing threshold. Odeon calculates reverberation times based on the decay from –5 to –35 [dB]. Fitted decay lines based on reverberation times are consequently an even rougher approximation outside this interval.

(2) Investigation of possible frequency dependencies. Room impulse responses should be evaluated in octave bands, to address possible simplifications as function of frequency content. The reverberation time of the ear is also frequency dependent and an octave band implementation could prove a better fit.

(3) Optimisation and improvement of the ear filter implemented as a simple Hanning window in the present paper.

(4) Implementing an absolute threshold to account for anechoic conditions.

In addition to the above listed improvement possibilities is it important to conduct a second listening test with a more numerous set of reverberant cases. Reverberation times in the present paper covers only the ranges 0,7-1,0 [s] and 1,4-1,8 [s]. Other complicating factors will have to be considered as reverberation times of rooms approach the reverberation time of the ear (0,4 [s]). A lot of research remains to be done in order to complete the transition from T_{60}`s of about 0,5 [s] to anechoic conditions. First of all, a bigger RIR set must be available.

Motifs should be echo critical and contain varying rhythmical properties to eliminate masking of echoes at certain delay intervals.

6.6 Dependence on simulation software

The assumption that outputs from Odeon Room Acoustic Software are correct is made throughout this paper. Calculation approximations are discussed in Section 2.4. Listening tests based on simulations from Odeon or other room acoustic software should be tested against corresponding measured impulse responses. This would further validate the use of room acoustic software. Particularly late prominent reflections like the ones shown in Figure 42 (scattering coefficient = 0,1) should be addressed. However, impulse responses computed through software simulations have undergone severe testing, so the assumption made is reasonable.

7 CONCLUSION

This paper has explored disturbance due to echoes in both anechoic and reverberant conditions. Odeon room acoustics software has been used to generate room impulse responses (RIR) for further evaluation of objective echo criteria. Five different music motifs were used, and one speech motif.

Listening test results proved very consistent throughout. Single echo thresholds for speech were comparable with corresponding levels calculated using Dietsch and Kraak`s algorithm, where disturbance levels are given as EK-levels. The implementation using wave-file inputs was less critical to echoes than the subjective reference, while the ISM implementation showed stricter threshold limits. No disturbing echoes were reported for delays shorter than 30 [ms], due to the inertia of the ear. Dietsch and Kraak failed to give a lower delay limit. The same feature could be seen for music motifs, as the 50 [%] disturbance limit were unbroken for delays shorter than 100 [ms]. Thresholds obtained from the trumpet motif proved far more echo critical than Dietsch and Kraak calculated values. Echo disturbance thresholds should thus be adjusted for music motifs.

Increasing the number of echoes to three was done to address the length of the rectangular evaluation window ($\Delta\tau_E$) suggested by Dietsch and Kraak and to verify the integration time of the ear. Dietsch and Kraak proved good resemblance with the less critical cello motif for longer delays. However, as separation times between the adjacent echoes were varied around the window length, huge differences occurred by adjusting the discrete echoes only few [ms], which is clearly non-physical. This feature is also evident in reverberant situations where reflections are separated just within the window evaluation length.

A selection of the music motifs was evaluated in a simplified concert hall, while the speech motif was evaluated in an auditorium. Both implementations of Dietsch and Kraak`s echo criterion showed low correspondence with listening test reference values. Adjustment of limit values due to the echo critical trumpet motif would prove inadequate, as increased subjective annoyance did not reflect itself in increased EK values.

A new criterion has been developed based on the convolution between energy RIR`s and a Hanning window of length 50 [ms]. An integration time of 50 [ms] simulates the energy integration of the human ear. Results from the single and multiple echo sections verified this integration time. Validation of window shape and length in detail is not possible from a limited number of test situations. A huge number of situations are needed if parameters of the window are to be assessed separately.

An evaluation curve consisting of an initial decay from the direct sound equal to the reverberant decay of the human ear, and a middle section where the room reverberation curve is shifted to fit the mean of the decay from –5 to –35 [dB] re. direct sound was deduced. Finally the hearing threshold forms the lower limit of the evaluation curve. Through an iterative process were levels for maximum positive deviation between the evaluation curve and the filtered energy RIR given as objective echo criteria.

Results from the auditorium proved perfect correspondence with the subjective reference, while 81 [%] was the result with the concert hall geometry. Causes and possible improvement possibilities are discussed. More experimental work must be performed to validate the criterion on a more general basis. However, initial results are promising.

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APPENDIX A – TEST MOTIFS

Appendix A includes time and amplitude dynamics of both output channels of all test motifs. Depending on the dynamic properties of each motif can reflections be masked by interference with later arriving test signals. Amplitudes should not be compared between different motifs, as they are scaled and normalized in the succeeding generation of listening test samples. However, amplitudes within each test motif remain constant relative one another. It is also important to keep in mind that the time scales used are different in between the figures throughout the appendix.

The dynamics of the male speech motif shown in Figure A1 is 8,0 seconds long. Echo critical silent periods are present, and also distinct peaks without long succeeding decays can be seen.

8,0 [s]

Figure A1: Male speech motif.

The cello test motif in Figure A2 has a constant sound level throughout due to the reverberation introduced by the body of the instrument. Possible echo generating peaks are therefore partly masked. A constant sound floor is also present in the orchestra test motif, as seen in Figure A5. Peaks seem more transient with the orchestra test motif due to the extended time interval shown in the figure.

Clear reverberant tails are also evident from the guitar test motif in Figure A3. However, the reverberant tails have more rapid decays than seen with the cello test motif. Long silent periods are also evident, causing the motif to be more critical to possible echoes.

The female chorus and the trumpet are found to be about equally sensitive to possible echoes. Their dynamic properties, as seen from Figure A4 and Figure A6 respectively, are similar.

Amplitudes within each of the motifs are of roughly equal strength. The main difference is three long periods with constant amplitude levels seen from the trumpet test motif.

Figure A2: Cello test motif.

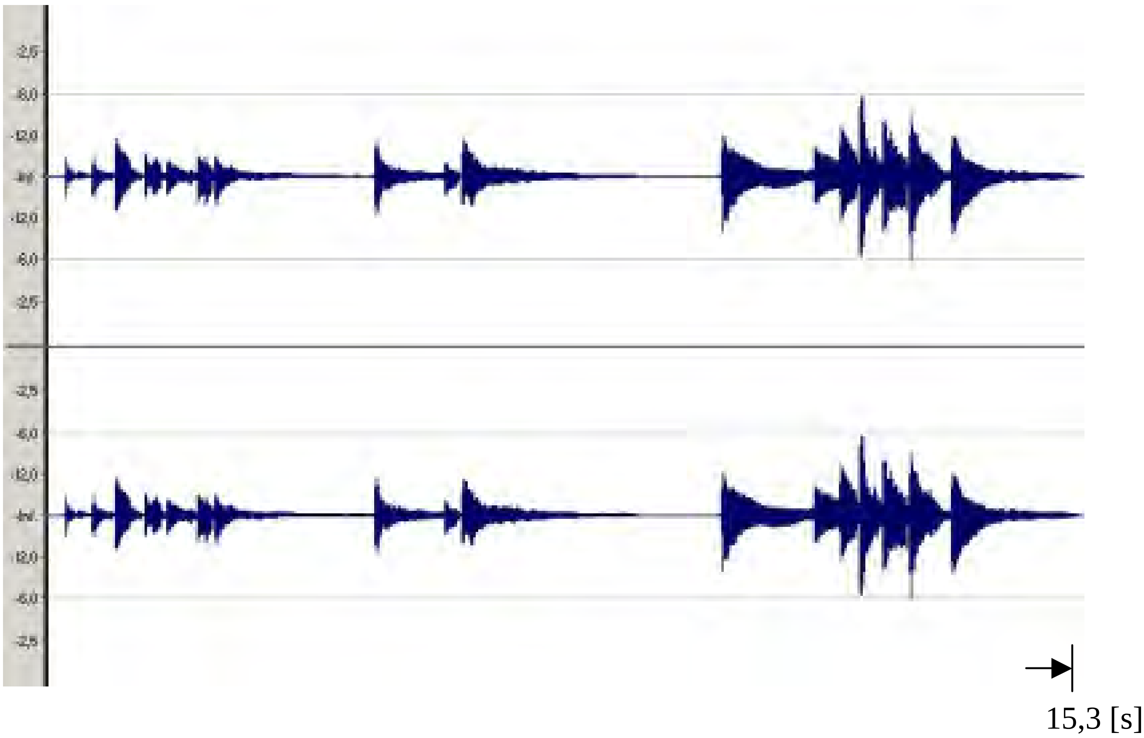

Figure A3: Guitar test motif.

Figure A4: Female chorus test motif.

Figure A5: Orchestra test motif.

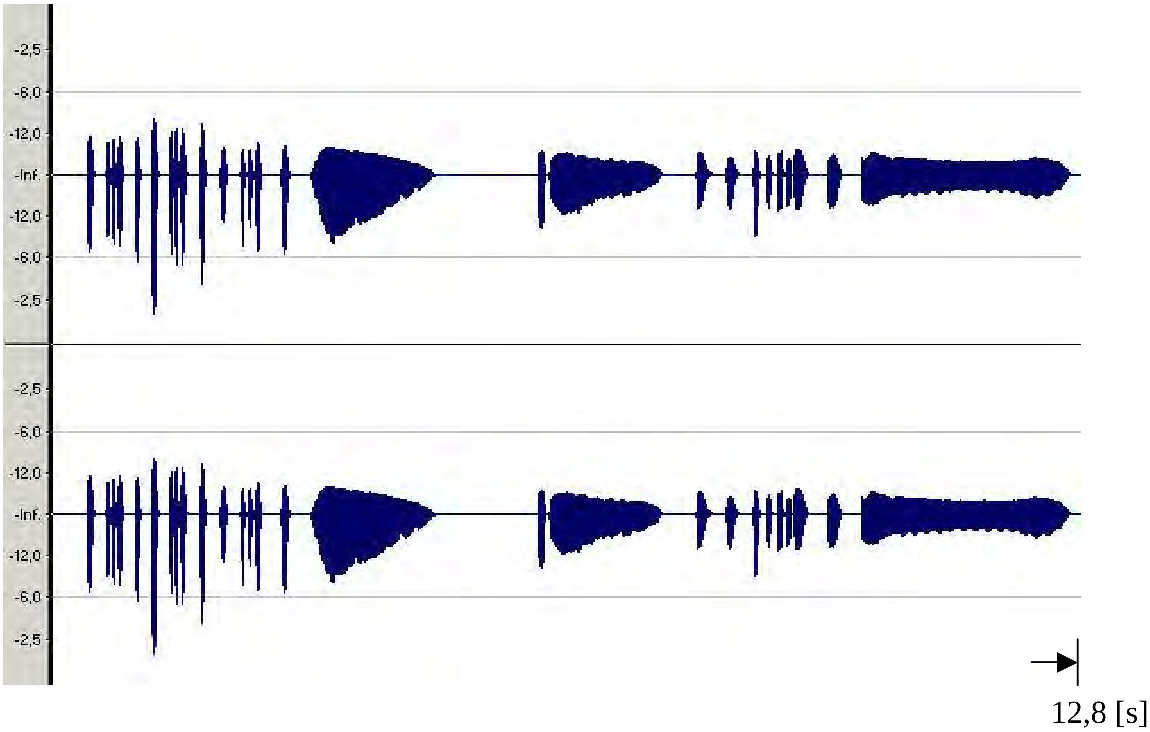

Figure A6: Trumpet test motif.

Appendix B – Listening Test Answering Sheets

Answering sheets for all test sequences are included in the following. A counter displayed on the screen showed the progress of each listening test so test persons could synchronize their answering with the sample flow continuously.

Three options presented to the test subjects in each sample case:

(1) Echo Not audible (N)

(2) Audible, but not disturbing, echo (A)

(3) Disturbing echo (D)

Name: **Test sequence: Speech**

* Space for test samples *

1: Anechoic sample
2: Strong echo mid delay
3: Echo mid delay
4: Weak echo mid delay
5: Spacious sample
6: Strong echo long delay

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D
15.	N	A	D
16.	N	A	D
17.	N	A	D
18.	N	A	D
19.	N	A	D
20.	N	A	D
21.	N	A	D
22.	N	A	D
23.	N	A	D
24.	N	A	D
25.	N	A	D
26.	N	A	D
27.	N	A	D
28.	N	A	D
29.	N	A	D
30.	N	A	D
31.	N	A	D
32.	N	A	D
33.	N	A	D
34.	N	A	D
35.	N	A	D
36.	N	A	D
37.	N	A	D
38.	N	A	D
39.	N	A	D
40.	N	A	D
41.	N	A	D
42.	N	A	D
43.	N	A	D
44.	N	A	D
45.	N	A	D
46.	N	A	D
47.	N	A	D
48.	N	A	D
49.	N	A	D
50.	N	A	D
51.	N	A	D
52.	N	A	D
53.	N	A	D
54.	N	A	D
55.	N	A	D
56.	N	A	D
57.	N	A	D
58.	N	A	D
59.	N	A	D
60.	N	A	D
61.	N	A	D
62.	N	A	D
63.	N	A	D
64.	N	A	D
65.	N	A	D
66.	N	A	D
67.	N	A	D
68.	N	A	D
69.	N	A	D
70.	N	A	D
71.	N	A	D
72.	N	A	D
73.	N	A	D
74.	N	A	D
75.	N	A	D
76.	N	A	D
77.	N	A	D
78.	N	A	D
79.	N	A	D
80.	N	A	D
81.	N	A	D
82.	N	A	D
83.	N	A	D
84.	N	A	D
85.	N	A	D
86.	N	A	D
87.	N	A	D
88.	N	A	D
89.	N	A	D
90.	N	A	D
91.	N	A	D
92.	N	A	D
93.	N	A	D
94.	N	A	D
95.	N	A	D
96.	N	A	D
97.	N	A	D
98.	N	A	D
99.	N	A	D
100.	N	A	D
101.	N	A	D
102.	N	A	D
103.	N	A	D
104.	N	A	D
105.	N	A	D
106.	N	A	D
107.	N	A	D
108.	N	A	D
109.	N	A	D
110.	N	A	D
111.	N	A	D
112.	N	A	D
113.	N	A	D
114.	N	A	D
115.	N	A	D
116.	N	A	D
117.	N	A	D

* Space for test samples *

1: Reverberant field
2: Echo in reverberant field

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D

Name: **Test sequence: Trumpet**

* Space for test samples *

1: Anechoic sample
2: Strong echo mid delay
3: Echo mid delay
4: Weak echo mid delay
5: Spacious sample
6: Strong echo long delay
7: Annoying (interference)

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D
15.	N	A	D
16.	N	A	D
17.	N	A	D
18.	N	A	D
19.	N	A	D
20.	N	A	D
21.	N	A	D
22.	N	A	D
23.	N	A	D
24.	N	A	D
25.	N	A	D
26.	N	A	D
27.	N	A	D
28.	N	A	D
29.	N	A	D
30.	N	A	D
31.	N	A	D
32.	N	A	D
33.	N	A	D
34.	N	A	D
35.	N	A	D
36.	N	A	D
37.	N	A	D
38.	N	A	D
39.	N	A	D
40.	N	A	D
41.	N	A	D
42.	N	A	D
43.	N	A	D
44.	N	A	D
45.	N	A	D
46.	N	A	D
47.	N	A	D
48.	N	A	D
49.	N	A	D
50.	N	A	D
51.	N	A	D
52.	N	A	D
53.	N	A	D
54.	N	A	D
55.	N	A	D
56.	N	A	D
57.	N	A	D
58.	N	A	D
59.	N	A	D
60.	N	A	D
61.	N	A	D
62.	N	A	D
63.	N	A	D
64.	N	A	D
65.	N	A	D
66.	N	A	D
67.	N	A	D
68.	N	A	D
69.	N	A	D
70.	N	A	D
71.	N	A	D
72.	N	A	D
73.	N	A	D
74.	N	A	D
75.	N	A	D
76.	N	A	D
77.	N	A	D
78.	N	A	D
79.	N	A	D
80.	N	A	D
81.	N	A	D
82.	N	A	D
83.	N	A	D
84.	N	A	D
85.	N	A	D
86.	N	A	D
87.	N	A	D
88.	N	A	D
89.	N	A	D
90.	N	A	D
91.	N	A	D
92.	N	A	D
93.	N	A	D
94.	N	A	D
95.	N	A	D
96.	N	A	D
97.	N	A	D
98.	N	A	D
99.	N	A	D
100.	N	A	D
101.	N	A	D
102.	N	A	D
103.	N	A	D
104.	N	A	D
105.	N	A	D
106.	N	A	D
107.	N	A	D
108.	N	A	D
109.	N	A	D
110.	N	A	D
111.	N	A	D
112.	N	A	D
113.	N	A	D
114.	N	A	D
115.	N	A	D
116.	N	A	D
117.	N	A	D
118.	N	A	D
119.	N	A	D
120.	N	A	D
121.	N	A	D

* Space for test samples *

1: Reverberant field
2: Echo in reverberant field

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D

Name: **Test sequence: Cello**

* Space for test samples *

1: Anechoic sample
2: Strong echo mid delay
3: Echo mid delay
4: Weak echo mid delay
5: Spacious sample
6: Strong echo long delay

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D
15.	N	A	D
16.	N	A	D
17.	N	A	D
18.	N	A	D
19.	N	A	D
20.	N	A	D
21.	N	A	D
22.	N	A	D
23.	N	A	D
24.	N	A	D
25.	N	A	D
26.	N	A	D
27.	N	A	D
28.	N	A	D
29.	N	A	D
30.	N	A	D
31.	N	A	D
32.	N	A	D
33.	N	A	D
34.	N	A	D
35.	N	A	D
36.	N	A	D
37.	N	A	D
38.	N	A	D
39.	N	A	D
40.	N	A	D
41.	N	A	D
42.	N	A	D
43.	N	A	D
44.	N	A	D
45.	N	A	D
46.	N	A	D
47.	N	A	D
48.	N	A	D
49.	N	A	D
50.	N	A	D
51.	N	A	D
52.	N	A	D
53.	N	A	D
54.	N	A	D
55.	N	A	D
56.	N	A	D
57.	N	A	D
58.	N	A	D
59.	N	A	D
60.	N	A	D
61.	N	A	D
62.	N	A	D
63.	N	A	D
64.	N	A	D
65.	N	A	D
66.	N	A	D
67.	N	A	D
68.	N	A	D
69.	N	A	D
70.	N	A	D
71.	N	A	D
72.	N	A	D
73.	N	A	D
74.	N	A	D
75.	N	A	D
76.	N	A	D
77.	N	A	D
78.	N	A	D
79.	N	A	D
80.	N	A	D
81.	N	A	D
82.	N	A	D
83.	N	A	D
84.	N	A	D
85.	N	A	D
86.	N	A	D
87.	N	A	D
88.	N	A	D
89.	N	A	D
90.	N	A	D
91.	N	A	D
92.	N	A	D
93.	N	A	D
94.	N	A	D
95.	N	A	D
96.	N	A	D
97.	N	A	D
98.	N	A	D
99.	N	A	D
100.	N	A	D
101.	N	A	D
102.	N	A	D
103.	N	A	D
104.	N	A	D
105.	N	A	D
106.	N	A	D
107.	N	A	D

Name: **Test sequence: Mixed**

* Space for test samples *

1: Anechoic sample
2: Strong echo
3: Medium strong echo
4: Weak echo

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D
15.	N	A	D
16.	N	A	D
17.	N	A	D
18.	N	A	D
19.	N	A	D
20.	N	A	D
21.	N	A	D
22.	N	A	D
23.	N	A	D
24.	N	A	D
25.	N	A	D
26.	N	A	D
27.	N	A	D
28.	N	A	D
29.	N	A	D
30.	N	A	D
31.	N	A	D
32.	N	A	D
33.	N	A	D

* Space for test samples *

1: Not audible echo
2: Audible echo

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D

* Space for test samples *

1: Reverberant field
2: Echo in reverberant field

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D

* Space for test samples *

1: Anechoic sample
2: Strong echo
3: Medium strong echo
4: Weak echo

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D
9.	N	A	D
10.	N	A	D
11.	N	A	D
12.	N	A	D
13.	N	A	D
14.	N	A	D
15.	N	A	D
16.	N	A	D
17.	N	A	D
18.	N	A	D
19.	N	A	D
20.	N	A	D
21.	N	A	D
22.	N	A	D
23.	N	A	D
24.	N	A	D
25.	N	A	D
26.	N	A	D
27.	N	A	D
28.	N	A	D
29.	N	A	D
30.	N	A	D
31.	N	A	D
32.	N	A	D
33.	N	A	D
34.	N	A	D
35.	N	A	D

* Space for test samples *

1: Reverberant example

* Space to start *

1.	N	A	D
2.	N	A	D
3.	N	A	D
4.	N	A	D
5.	N	A	D
6.	N	A	D
7.	N	A	D
8.	N	A	D

APPENDIX C – MATLAB M-FILES

All the criteria investigated were implemented in MatLab. The m-files that computes Dietsch and Kraak`s criterion with txt-file (*dkism.m*) and wave-file (*dkwav.m*) input are reproduced in this appendix. The m-file used to calculate the new criterion (*newcriterion.m*) is then reproduced, before an example of the listening test generation is included (*speechtest.m*). All m-files are included in Appendix F – CD.

dkism.m

```
%-------------------------------------------------------------------------------
% Written by Anders Løvstad 160603
clear all

disp(' ')
disp('**************************************************************************')
disp('*                                                                        *')
disp('*  Evaluates Dietsch and Kraak`s echo criterion based on lower order ISM *')
disp('*                                                                        *')
disp('*  Input: .txt-files of the room impulse responses.                      *')
disp('*                                                                        *')
disp('*  Output:  Results in .txt-file.                                        *')
disp('*           Plot of resulting Dietsch and Kraak levels.                  *')
disp('*                                                                        *')
disp('*  Filepath of the impulse response .txt-files must be set in            *')
disp('*  advance in the m-file.                                                *')
disp('*                                                                        *')
disp('**************************************************************************')
disp(' ')


%-------------------------------------------------------------------------------
% Setting parameters and loading .txt-files into memory

delta_t = input('Music (m) or speech (s):  ','s');
plots = input('Do you want plots of all results (y or n)?  ','s');

Fpath = 'P:\Diplom\ISM\Test\';
filenames = dir(fullfile(Fpath,'*.txt'));
nfiles = size(filenames);
filenames = {filenames.name};
resultmatrix = cell(nfiles(1,1),3);

if delta_t == 'm'
       inttime = 14/1000;
elseif delta_t == 's'
       inttime = 9/1000;
end


%-------------------------------------------------------------------------------
% Start calculation loop.

for countfiles = 1:nfiles(1,1)

       %-------------------------------------------------------------------------------
       % Load .wav-file into memory, and save filename for output .txt-file.

       importname = [Fpath,filenames{countfiles}];
       importdata(importname);
       fs = 44100;
       timewindow = 2;
       rgram = zeros(fs*timewindow,1);


       %-------------------------------------------------------------------------------
       % Summing octaveband values for the direct sound, which then is used to compute
       % a reference level, p0. Octave band levels are then added up for all later
       % arriving contributions relative the reference level. Loading the results into
```

```
        % the matrix rgram, which in turn is normalized.

        data = ans.data;
        pvalue = zeros(1,size(data,1));

        totspl =
10*log10(10^(data(1,4)/10)+10^(data(1,5)/10)+10^(data(1,6)/10)+10^(data(1,7)/10)+10^(data(1,8)
/10)+10^(data(1,9)/10)+10^(data(1,10)/10)+10^(data(1,11)/10));
        ref = 10^(-totspl/20);
        splvec = zeros(1,size(data,1));

        for i = 1:size(data,1)
                totspl = 10*log10(10^(data(i,4)/10)+10^(data(i,5)/10)+10^(data(i,6)/10)+
10^(data(i,7)/10)+10^(data(i,8)/10)+10^(data(i,9)/10)+10^(data(i,10)/10)+10^(data(i,11)/10));
                splvec(1,i) = totspl;
                pvalue(1,i) = (10^(totspl/20))*ref;
                rgram(round(data(i,3)*10^-3*fs)+1,1) = pvalue(1,i);
        end

        rgram = rgram/(max(rgram));


        %-------------------------------------------------------------------------------
        % Scaling if input is speech

        if delta_t == 's'
                rgram = rgram.^(2/3);
        end


        %-------------------------------------------------------------------------------
        % Duplicate rgram to get nominator and denominator in D&K expression. Scales the
        % nominator by multiplying by time. First value of nominator set to zero.

        nominator = rgram.';
        denominator = rgram.';

        for m = 1:length(rgram)
                nominator(1,m) = nominator(1,m).*(m/fs);
        end

        nominator(1,1) = 0;


        %-------------------------------------------------------------------------------
        % Initialize and integrate matrix element by element. First element in nominator
        % and denominator equals first in sum. Then the elements are added together
        % cumulatively, as described in D&K.

        sumirnom = zeros(1,length(rgram));
        sumirdenom = zeros(1,length(rgram));
        ek = zeros(1,length(rgram));

        sumirnom(1,1) = nominator(1,1);
        sumirdenom(1,1) = denominator(1,1);

        for j = 2:length(rgram)
                sumirnom(1,j)=sumirnom(1,j-1)+nominator(1,j);
                sumirdenom(1,j)=sumirdenom(1,j-1)+denominator(1,j);
        end


        %-------------------------------------------------------------------------------
        % To obtain EK as a function of time, values in the cumulative build-up function
        % are subtracted.

        intlength = round(inttime*fs);
        ek=sumirnom./sumirdenom;
        delta_ek = zeros(1,length(rgram));

        for k = 1:length(rgram)-intlength
                delta_ek(1,k) = ek(1,intlength+k)-ek(1,1+k);
        end


        %-------------------------------------------------------------------------------
        % As the matrix delta_ek only gives the values of the total IR after one full
```

```
        % summation (intagration time), the values before this must be added into the
        % total resulting D&K-matrix.

        ek_beg = ek(1,[1:intlength])./inttime;
        delta_ek = delta_ek./inttime;

        totek = zeros(1,length(ek_beg)+length(delta_ek));
        totek(1,[1:length(ek_beg)]) = ek_beg;
        totek(1,[length(ek_beg)+1:length(totek)]) = delta_ek;


        %--------------------------------------------------------------------------------
        % Results

        if delta_t == 'm'
                limit = find(totek>1.8);
        elseif delta_t == 's'
                limit = find(totek>1);
        end

        if limit > 0
                result = 'Echo';
        else
                result = 'Ok';
        end

        if plots == 'y'
                figure(countfiles)
                plot(totek,'b')
                axis([0 fs 0 max(totek)+0.2]);
                title(filenames{countfiles})
        end

        resultmatrix(countfiles,1) = cellstr(filenames{countfiles});
        resultmatrix(countfiles,2) = cellstr(result);
        resultmatrix(countfiles,3) = cellstr(num2str(max(totek)));
end


%--------------------------------------------------------------------------------
% Generate resultfile.

maketextfile = input('Do you want to generate a separate results .txt-file?  (y or n):  '
,'s');

if maketextfile == 'y'
        outputfilename = input('Name of generated resultfile (without extension):  ', 's');
        resultfile = [outputfilename, '.txt'];
        fid = fopen(resultfile,'a');

        for resno = 1:size(resultmatrix,1)
                fprintf(fid,[num2str(resno), '. ', '\t', resultmatrix{resno,1}, '\t',
resultmatrix{resno,2}, '\t', resultmatrix{resno,3}, '\t', '\n']);
        end

        fclose(fid);
end
```

dkwav.m

```
%---------------------------------------------------------------------------------
% Written by Anders Løvstad 160603
clear all

disp(' ')
disp('***************************************************************************')
disp('*                                                                         *')
disp('*  Evaluates Dietsch and Kraak`s echo criterion                           *')
disp('*                                                                         *')
disp('*  Input: .wav-files of the room impulse responses.                       *')
disp('*                                                                         *')
disp('*  Output:  Results in .txt-file.                                         *')
disp('*           Plot of resulting Dietsch and Kraak levels.                   *')
disp('*                                                                         *')
disp('*  Filepath of the impulse response .wav-files must be set in             *')
disp('*  advance in the m-file.                                                 *')
disp('*                                                                         *')
disp('***************************************************************************')
disp(' ')


%---------------------------------------------------------------------------------
% Setting parameters and loading selected .wav-file into memory

Fpath = 'P:\Diplom\HRTFwav\Test\';
filenames = dir(fullfile(Fpath,'*.wav'));
nfiles = size(filenames);
filenames = {filenames.name};
resultmatrix = cell(nfiles(1,1),3);

delta_t = input('Music (m) or speech (s):  ','s');
if delta_t == 'm'
        inttime = 14/1000;
elseif delta_t == 's'
        inttime = 9/1000;
end

plots = input('Do you want plots of all results (y or n)?  ','s');


%---------------------------------------------------------------------------------
% Start calculation loop.

for countfiles = 1:nfiles


        %---------------------------------------------------------------------------------
        % Load .wav-file into memory, and save filename for output .txt-file.

        [evwav,fs,bits] = wavread(filenames{countfiles});


        %---------------------------------------------------------------------------------
        % Cuts wav-file to first second to decrease comptime if necessary, adds both
        % signals of the recorded BRIR. Normalising.

        if fs >= length(evwav)
                evwavcut = zeros(fs,2);
                evwavcut([1:length(evwav)],:) = evwav([1:length(evwav)],:);
        else
                evwavcut = zeros(fs,2);
                evwavcut = evwav([1:fs],:);
        end

        evalwave = abs(evwavcut(:,1))+abs(evwavcut(:,2));
        evalwave = evalwave/(max(evalwave));


        %----------------------------------------------------------------------------------
        % Removes quantisation errors

        for nn = 1:length(evalwave)
                if evalwave(nn,1)<0.001
                        evalwave(nn,1)=0;
```

```
        end
end


%---------------------------------------------------------------------------
% Establishes a time-axis relative the directpulse and scales pressures in D&K
% if speech criterion is to be evaluated

highvalues = find(abs(evwav)>0.03);
directpulse = min(highvalues);
evalwave = evalwave(directpulse:length(evalwave));

if delta_t == 's'
        evalwave = evalwave.^(2/3);
end


%---------------------------------------------------------------------------
% Duplicate wavefile to get nominator and denominator in D&K expression, before
% the nominator is scaled as function of time. First value set to zero.

nominator = evalwave.';
denominator = evalwave.';

for m = 1:length(evalwave)
        nominator(1,m) = nominator(1,m).*(m/fs);
end

nominator(1,1) = 0;


%---------------------------------------------------------------------------
% Initialize and integrate matrix element by element. First element in nominator
% and denominator equals first in sum. Then the elements are added together
% cumulatively, as described in D&K.

sumirnom = zeros(1,length(evalwave));
sumirdenom = zeros(1,length(evalwave));
ek = zeros(1,length(evalwave));

sumirnom(1,1) = nominator(1,1);
sumirdenom(1,1) = denominator(1,1);

for j = 2:length(evalwave)
        sumirnom(1,j)=sumirnom(1,j-1)+nominator(1,j);
        sumirdenom(1,j)=sumirdenom(1,j-1)+denominator(1,j);
end


%---------------------------------------------------------------------------
% To obtain EK as a function of time, values in the cumulative build-up function
% are subtracted.

intlength = round(inttime*fs);
ek=sumirnom./sumirdenom;
delta_ek = zeros(1,length(evalwave));

for k = 1:length(evalwave)-intlength
        delta_ek(1,k) = ek(1,intlength+k)-ek(1,1+k);
end


%---------------------------------------------------------------------------
% As the matrix delta_ek only gives the values of the total IR after one full
% summation (intagration time), the values before this must be added into the
% total resulting D&K-matrix.

ek_beg = ek(1,[1:intlength])./inttime;
delta_ek = delta_ek./inttime;

totek = zeros(1,length(ek_beg)+length(delta_ek));
totek(1,[1:length(ek_beg)]) = ek_beg;
totek(1,[length(ek_beg)+1:length(totek)]) = delta_ek;
```

```
        %---------------------------------------------------------------------------------
        % Results

        if delta_t == 'm'
                limit = find(totek>1.8);
        elseif delta_t == 's'
                limit = find(totek>1);
        end

        if limit > 0
                result = 'Echo';
        else
                result = 'Ok';
        end

        if plots == 'y'
                figure(countfiles)
                plot(totek,'b')
                axis([0 fs 0 max(totek)+0.2]);
                title(filenames{countfiles})
        end

        resultmatrix(countfiles,1) = cellstr(filenames{countfiles});
        resultmatrix(countfiles,2) = cellstr(result);
        resultmatrix(countfiles,3) = cellstr(num2str(max(totek)));
end


%---------------------------------------------------------------------------------
% Generate resultfile.

maketextfile = input('Do you want to generate a separate results .txt-file?  (y or n):  '
,'s');

if maketextfile == 'y'
        outputfilename = input('Name of generated resultfile (without extension):  ', 's');
        resultfile = [outputfilename, '.txt'];
        fid = fopen(resultfile,'a');

        for resno = 1:size(resultmatrix,1)
                fprintf(fid,[num2str(resno), '. ', '\t', resultmatrix{resno,1}, '\t',
        resultmatrix{resno,2}, '\t', resultmatrix{resno,3}, '\t', '\n']);
        end

fclose(fid);
end
```

newcriterion.m

```
%------------------------------------------------------------------------------------
% Written by Anders Løvstad 160603
clear all

disp(' ')
disp('*********************************************************************')
disp('*                                                                   *')
disp('*  Echo criterion based on output .wav-files from Odeon.            *')
disp('*                                                                   *')
disp('*  Input:   Anechoic or reverberant conditions.                     *')
disp('*           Lower perceptible sound level.                          *')
disp('*                                                                   *')
disp('*  Output:  Results in .txt-file.                                   *')
disp('*           Plot of integrated response and limitcurve.             *')
disp('*                                                                   *')
disp('*  Filepath must be set in advance in the m-file. Reverberation     *')
disp('*  times must be typed into the matrix rtmatrix in line 73 for      *')
disp('*  reverberant analyses.                                            *')
disp('*                                                                   *')
disp('*********************************************************************')
disp(' ')


%------------------------------------------------------------------------------------
% Input

revoranechoic = input('Reverberant (r) or anechoic (a) conditions?  ','s');
if revoranechoic == 'a'
        signaltype = input('Speech (s) or music (m):  ','s');
end

cutwavefile = input('Do you want to cut input .wav-file length?  (y or n):  ','s');
if cutwavefile == 'y'
        cutwavelength = ('Type in length (in seconds):  ');
end


%------------------------------------------------------------------------------------
% Constants and filepath parameters.

tau = 50e-3;
rtear = 0.4;
timewindow = 3;
echo(1,1) = cellstr('Filename');

Fpath = 'P:\Diplom\HRTFwav\Test\';
filenames = dir(fullfile(Fpath,'*.wav'));
nfiles = size(filenames);
filenames = {filenames.name};


%------------------------------------------------------------------------------------
% Establish limitcurve from subjective single-echo values. Use precalculated
% limitcurve.

if revoranechoic == 'a'
        tdirect = 1116;
        if signaltype == 's'
                t = [1116 2440 3322 5527 7732 9937 18756 44100]';
                valuelimit = [0 -8.3767 -13.1764 -30.4760 -36.1704 -40.1806 -46.4757 -50]';
                load speech50
        elseif signaltype == 'm'
                t = [5527 7732 9937 18756]';
                thigh = [5527 7732 9937 18757]';
                valuelimit = [-32.8748 -42.2614 -45.1582 -46.7629]';
                valuelimithigh = [-2.8769 -7.2766 -10.1763 -23.7767; -9.8768 -14.2762 -20.1759-
-23.7767; 2.1232 -7.2766 -10.1763 -16.7767]';
                load trumpet50, load cello50, load orchestra50, load guitar50
        end
end


if revoranechoic == 'r'
```

```
        rtmatrix = [1.8175 1.6575 1.61625 1.64 1.57875 1.6525 1.62625 1.60375 1.7125 1.58875
1.4475 1.5075 1.41375 1.3925 1.52 1.47375];
        % rtmatrix = [0.92375 0.9075 0.96375 0.935 0.92 1.035 0.7025 0.665 0.70875 0.6875
% 0.70625 0.69375];
end


%-------------------------------------------------------------------------------
% Start calculation loop.

for countfiles = 1:nfiles

        %-------------------------------------------------------------------------------
        %Clear variables between each file calculation. Read files into memory.

        clear Differ maxvalue minvalue marksep dBstep;
        clear evwav evwavcut evalwave energy result dBresult;
        clear globalmaxindex globalmaxvalue sepmat timesepequal timesephor timesepvert;
        clear abovethirtyfive lastthirtyfive belowfive firstfive changecurve changesample;
        clear totcurve rtcurve earcurve evcurve difference maxind maxval;


        %-------------------------------------------------------------------------------
        % Load .wav-file into memory, and save filename for output .txt-file.

        [evwav,fs,bits] = wavread(filenames{countfiles});
        echo(countfiles+2,1) = cellstr(filenames{countfiles});

        if revoranechoic == 'r'
                rt = rtmatrix(countfiles);
        end


        %-------------------------------------------------------------------------------
        % Cut wav-file to decrease comptime if wanted, add signal from both channels
        % of the recorded BRIR. Normalise.

        if cutwavefile == 'n'
                evwavcut = zeros(length(evwav),2);
                evwavcut([1:length(evwav)],:) = evwav([1:length(evwav)],:);
        else
                evwavcut = zeros(round(cutwavelength*fs),2);
                evwavcut = evwav([1:length(evwavcut)],:);
        end

        evalwave = abs(evwavcut(:,1)).^2+abs(evwavcut(:,2)).^2;
        evalwave = evalwave/(max(evalwave));


        %-------------------------------------------------------------------------------
        % Establish a time-axis relative to the directpulse.

        highvalues = find(abs(evwav)>0.03);
        directpulse = min(highvalues);
        evalwave = evalwave(directpulse:length(evalwave));


        %-------------------------------------------------------------------------------
        % Convolve with hann-window to simulate the integration of the human ear.
        % Convert to energy, and subsequently to desibel scale. Set direct sound
        % to 0 dB.

        win = round(fs*tau);
        y = hann(win);
        energy = evalwave;
        result = conv(y,energy);
        dBresult = 10*log10(result/20e-6);
        dBresult = dBresult-max(dBresult);


        %-------------------------------------------------------------------------------
        % Find extremal points on dB-curve (local and global maxima and minima). Values
        % found when the derivative changes sign.

        [Differ] = diff(dBresult(1:length(dBresult)));
        no = 1;
        no2 = 1;
```

```
        for zerocross = 1:length(Differ)-1
                if Differ(zerocross) <= 0 & Differ(zerocross+1) > 0
                        minvalue(no,1) = zerocross+1;
                        no = no+1;
                elseif Differ(zerocross) > 0 & Differ(zerocross+1) < 0
                        maxvalue(no2,1) = zerocross+1;
                        no2 = no2+1;
                end
        end


        %-------------------------------------------------------------------------------
        % Find level step from minimum to maximum points, and separation time from last equal
        % energy level from maxima points.

        for i = 2:size(maxvalue)
                backcount = 1;

                while dBresult(maxvalue(i)) > dBresult(maxvalue(i)-backcount) &
dBresult(maxvalue(i)) ~= 0
                        backcount = backcount+1;
                end

                marksep(i-1) = maxvalue(i)-backcount;
                timesepequal(i-1) = backcount;

                if revoranechoic == 'r'
                        timesepvert(i-1) = maxvalue(i)-minvalue(i-1);
                        timesephor(i-1) = maxvalue(i)-marksep(i-1);
                        dBstep(i-1) = dBresult(maxvalue(i))-dBresult(minvalue(i-1));
                end
        end


        %-------------------------------------------------------------------------------
        % Establish limitcurve from reverberation time.

        if revoranechoic == 'r'
                [globalmaxvalue, globalmaxindex] = max(dBresult);
                rtcurve = zeros(length(dBresult),1);
                rtvalue = globalmaxvalue;
                earvalue = globalmaxvalue;


        %-------------------------------------------------------------------------------
        % An arbitrary curve with RT-gradient is established (value insignificant), and
        % then fitted to the decay of a given case by averaging the difference between
        % the arbitrary level curve and the integrated energy response on the interval
        % from the first time the energy response is -5 dB to the last time it reaches
        % a value -35 dB re directsound.

                for decay = globalmaxindex:length(dBresult)
                        rtcurve(decay,1) = rtvalue;
                        rtvalue = rtvalue-(60/((rt*fs)-globalmaxindex));
                end

                rtend = length(dBresult);
                difference(globalmaxindex:rtend) = rtcurve(globalmaxindex:rtend)-
dBresult(globalmaxindex:rtend);
                belowfive = find(dBresult(globalmaxindex:rtend) < -5);
                firstfive = belowfive(1) + globalmaxindex;
                abovethirtyfive = find(dBresult(globalmaxindex:rtend) > -35);
                lastthirtyfive = abovethirtyfive(length(abovethirtyfive))+globalmaxindex;

                meanval(countfiles) = mean(difference(firstfive:lastthirtyfive));
                rtcurve = rtcurve-meanval(countfiles);


        %-------------------------------------------------------------------------------
        % Set limitcurve to be tangential to the falling edge of the direct pulse
        % (tangential) if the edge is steeper than the reverberation time of the ear
        % (0.4 sec). If the falling edge of the integrated impulse response is steeper
        % than the RT of the ear, it will simply fall of from the maximum value sample.
        %
        % In both cases will the limitcurve fall off with RT of the ear until the
        % reverberation curve fitted above exceeds it. Limitvalue from 0 to direct sound
```

```
        % is set to the direct sound level (maximum = 0 dB as it has been normalised
        % above).

                for decay2 = globalmaxindex:(rtear*fs)
                        earcurve(decay2,1) = earvalue;
                        earvalue = earvalue-(60/((rtear*fs)-globalmaxindex));
                end

                startcount = 0;

                while dBresult(globalmaxindex+startcount) >=
earcurve(globalmaxindex+startcount) & startcount+globalmaxindex < length(earcurve)
                startcount = startcount+1;
                end

                if startcount+globalmaxindex < length(earcurve)
                        for move = globalmaxindex:(globalmaxindex+startcount)
                                sepmat(move) = dBresult(move)-earcurve(move);
                                [valuesep,indexsep] = max(sepmat);
                        end

                earcurve = earcurve+valuesep;
                changecurve = find(earcurve(globalmaxindex:length(earcurve)) <
rtcurve(globalmaxindex:length(earcurve)));
                changesample = changecurve(1)+globalmaxindex-1;
                totcurve(1:changesample) = earcurve(1:changesample);
                totcurve(changesample+1:length(rtcurve)) =
rtcurve(changesample+1:length(rtcurve));

                else
                        totcurve = rtcurve';
                        for nomove = 1:length(earcurve)
                                if earcurve(nomove) > totcurve(nomove)
                                        totcurve(nomove) = earcurve(nomove);
                                end
                        end
                end


        %--------------------------------------------------------------------------------
        % Lower limit set by thresholdvalue for perception (input).

                hearingthreshold = -50;

                for ht = 1:length(totcurve)
                        if totcurve(ht) <= hearingthreshold
                                totcurve(ht) = hearingthreshold;
                        elseif totcurve(ht) > 0
                                totcurve(ht) = 0;
                        end
                end


        %--------------------------------------------------------------------------------
        % Values compared to the limitcurve. Delay and value of the sample exceeding the
        % limitcurve the most are stored separately for output in .txt-file below.

                evcurve = dBresult-totcurve';
                [maxval maxind] = max(evcurve);

                echo(countfiles+2,2) = cellstr(num2str(maxval));
                echo(countfiles+2,3) = cellstr(num2str(maxind));
                echo(countfiles+2,4) = cellstr(num2str(hearingthreshold));


        %--------------------------------------------------------------------------------
        % Only values exceeding the limit curve are of interest, the rest set to zero.

                for evvalue = 1:length(evcurve)
                        if evcurve(evvalue) < 0
                                evcurve(evvalue) = 0;
                        end
                end
        end
```

```
        %-----------------------------------------------------------------------------
        % Plot resulting ear response for comparison with limitvalues from the first minimum
        % value.

        if revoranechoic == 'a'
                dBresult = dBresult-dBresult(maxvalue(1));
                echoval = 0;

                if echoval == 0 & length(maxvalue) > 1
                        for countanechoic = minvalue(1):length(dBresult)
                                if dBresult(countanechoic) > value(countanechoic) & echoval == 0
                                        echoval = 1;
                                        echo(countfiles+2,2) = cellstr('Echo');
                                end
                        end
                end
        end


        %-----------------------------------------------------------------------------
        % Plot resulting ear response for comparison with limitvalues.

        figure(countfiles)
        plot(dBresult,'b')
        hold on

        if revoranechoic == 'r'
                plot(totcurve,'r--')
                plot(evcurve,'g')
                plot(maxind,maxval,'ro')
                axis([0 length(dBresult) -80 max(evcurve)+5]);
        else
                if length(maxvalue) >= 2
                        plot(minvalue,dBresult(minvalue),'ro')
                        plot(marksep,dBresult(marksep),'mo')
                end
        if signaltype == 'm'
                plot(t(1):t(4)+1,value(t(1):t(4)+1),'r')
                plot(t(1):t(4)+1,valuecello(t(1):t(4)+1),'m--')
                plot(t(1):t(4)+1,valueguitar(t(1):t(4)+1),'g--')
                plot(t(1):t(4)+1,valueorc(t(1):t(4)+1),'k--')
                plot(t,valuelimit,'ro')
                plot(thigh,valuelimithigh(:,1),'mo')
                plot(thigh,valuelimithigh(:,2),'go')
                plot(thigh,valuelimithigh(:,3),'ko')
        else
                plot(value,'r')
                plot(t,valuelimit,'ro')
        end
                plot(maxvalue,dBresult(maxvalue),'go')
                axis([0 25000 -80 max(dBresult)+5]);
        end

        title(filenames{countfiles});
        ylabel('[dB] re. direct sound');
        xlabel('time [samples]');
        hold off
end


%-----------------------------------------------------------------------------
% Generate resultfile.

maketextfile = input('Do you want to generate a separate results .txt-file?  (y or n):  '
,'s');

if maketextfile == 'y'
        outputfilename = input('Name of generated resultfile (without extension):  ', 's');
        resultfile = [outputfilename, '.txt'];
        fid = fopen(resultfile,'w');

        if revoranechoic == 'r'
                echo(1,2) = cellstr('Maxamplitude    ');
                echo(1,3) = cellstr('Index    ');
                echo(1,4) = cellstr('     Lower hearing threshold');

                for resno = 1:size(echo)
```

```
                        fprintf(fid,[num2str(resno), '. ', '\t', echo{resno,1}, '\t',
echo{resno,2}, '\t', echo{resno,3}, '\t\t', echo{resno,4}, '\t', '\t', '\n']);
                end

        elseif revoranechoic == 'a'
                echo(1,2) = cellstr('Result');
                echo(1,3) = cellstr('');

                for resno = 1:size(echo)
                        fprintf(fid,[num2str(resno), '. ', '\t', echo{resno,1}, '\t',
echo{resno,2}, '\t', echo{resno,3}, '\t', '\n']);
                end
        end

        fclose(fid);
end
```

speechtest.m

```
%-------------------------------------------------------------------------------
% Written by Anders Løvstad 240303
clear all

%-------------------------------------------------------------------------------
% Listening test person choice for generation of output .txt-file to be
% connected with the corresponding test person.

disp(' ')
disp('Listening person: ')
disp('1: BS                     2: JPS '        )
disp('3: PN                     4: TG '         )
disp('5: AS                     6: PS '         )
disp('7: UK                     8: JT '         )
disp('9: GS                     10: AB '        )
disp('11: HF                    12: OAE '       )
disp('13: AL ')

tpn = input('Name: ');

if tpn == 1
        name = 'BS';
elseif tpn == 2
        name = 'JPS';
elseif tpn == 3
        name = 'PN';
elseif tpn == 4
        name = 'TG';
elseif tpn == 5
        name = 'AS';
elseif tpn == 6
        name = 'PS';
elseif tpn == 7
        name = 'UK';
elseif tpn == 8
        name = 'JT';
elseif tpn == 9
        name = 'GE';
elseif tpn == 10
        name = 'AB';
elseif tpn == 11
        name = 'HF';
elseif tpn == 12
        name = 'OAE';
elseif tpn == 13
        name = 'AL';
end


%-------------------------------------------------------------------------------
% Setting directory that contains the test wave-files.

fdir = 'F:\Users\Anders\';
seq = 'Speech';
filedir = [fdir,seq,'\'];
revdir = [fdir,seq,'rev\'];

orderfile = [seq,int2str(tpn),'.txt'];


%-------------------------------------------------------------------------------
% Read directory content into struct array. Create randomized file-order, and
% remove redundant information.

clear filenames;
clear revnames;
filenames = dir(fullfile(filedir,'*.wav'));
revnames = dir(fullfile(revdir,'*.wav'));

nfiles = size(filenames);
nrevfiles = size(revnames);

order = randperm(nfiles(1,1));
revorder = randperm(nrevfiles(1,1));
```

```
filenames = {filenames(order).name};
revnames = {revnames(revorder).name};


%--------------------------------------------------------------------------------
% Test section.

disp(' ')
disp('***********************************************')
disp('                                               ')
disp('             Part 1: Anechoic                  ')
disp('                                               ')
disp('***********************************************')
disp(' ')

pause


%--------------------------------------------------------------------------------
% Play anechoic sample and test samples clarifying the echo level span.

[a,fs,bits]=wavread('F:\Users\Anders\Speech\d_s.wav');
wavplay(a,fs,'sync')
[b,fs,bits]=wavread('F:\Users\Anders\Speech\s_s_150_plus10.wav');
wavplay(b,fs,'sync')
[c,fs,bits]=wavread('F:\Users\Anders\Speech\s_s_150_plus3.wav');
wavplay(c,fs,'sync')
[d,fs,bits]=wavread('F:\Users\Anders\Speech\s_s_150_min3.wav');
wavplay(d,fs,'sync')
[e,fs,bits]=wavread('F:\Users\Anders\Speech\s_s_18_lim.wav');
wavplay(e,fs,'sync')
[f,fs,bits]=wavread('F:\Users\Anders\Speech\s_s_400_plus15.wav');
wavplay(f,fs,'sync')

pause


%--------------------------------------------------------------------------------
% Start test section. Read wave-files into memory in the generated randomized
% order. Display counter for synchronization with answering sheet.

for i = 1:nfiles
        [file,fs,bits] = wavread(filenames{i});
        wavplay(file,fs,'sync');
        disp(int2str(i))
end


disp(' ')
disp('***********************************************')
disp('                                               ')
disp('             Part 2: Reverberant               ')
disp('                                               ')
disp('***********************************************')
disp(' ')

pause


%--------------------------------------------------------------------------------
% Play anechoic sample and test samples clarifying the echo level span.

[aa,fs,bits]=wavread('F:\Users\Anders\Speechrev\r_s_aud_tilt50_mic2');
wavplay(aa,fs,'sync')
[bb,fs,bits]=wavread('F:\Users\Anders\Speechrev\r_s_aud_tilt10_mic6');
wavplay(bb,fs,'sync')

pause


%--------------------------------------------------------------------------------
% Start test section. Read wave-files into memory in the generated randomized
% order. Display counter for synchronization with answering sheet.

for ii = 1:nrevfiles
        [revfile,fs,bits] = wavread(revnames{ii});
```

```
        wavplay(revfile,fs,'sync');
        disp(int2str(ii))
end


%-------------------------------------------------------------------------------
% Save file order, name of test person and test sequence.

fid = fopen(orderfile,'a');
fprintf(fid,['Test person: ', name, '\n', 'Sequence: ',seq, '\n\n']);

for i = 1:nfiles
        fprintf(fid,[num2str(i), '. ', '\t', int2str(tpn), '\t',filenames{i}, '\n']);
end

for ii = 1:nrevfiles
        fprintf(fid,[num2str(ii+nfiles(1,1)), '. ', '\t', int2str(tpn), '\t',revnames{ii},
'\n']);
end

fprintf(fid,['\n\n']);
fclose(fid);
```

APPENDIX D – LISTENING TESTS AND CALCULATED RESULTS

All results from the listening tests and calculations for all the corresponding situations are listed in this appendix. Anechoic results are given first, followed by the reverberant ones. Results from single echo situations where echo reflections are generated through a single reflection from the back wall of the concert hall are included in the reverberant section. Invalid results are drawn through with a dotted red line.

Results are presented in the following order:

- Speech, anechoic
- Cello, anechoic
- Guitar, anechoic
- Orchestra, anechoic
- Trumpet, anechoic

- Speech, reverberant
- Orchestra, reverberant
- Trumpet, reverberant
- Female chorus, reverberant

Figure A7 exemplifies and explains the results displayed below.

Percentage of listeners who heard a disturbing (D), audible (D+A) or no echo (N), respectively.

D&K ISM results, only one column in anechoic situations

Test persons

																					Reflectogram										
	Scat. Coeff.	Receiver	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	1st order	D&K value	2nd order	D&K value	3rd order	D&K value	4th order	D&K value	Wave-files	D&K value	New criterion
Backwall	10	17	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,84							N	1,07	D
		1	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	1,24							N	1,77	D
		7	D	D	D	D	D	D	A	D	D		D	A	D	80 %	100 %	0 %	D	D	N	1,71							D	2,33	D
		11	D	N	D	A	D	N	N	D	N		D	A	N	40 %	60 %	40 %	A	D	N	1,44							D	1,80	D
		13	D	N	N	A	D	N	A	A	N		A	A	N	10 %	60 %	40 %	A	D	N	1,08							N	1,22	N
		14	D	N	N	N	A	N	A	A	N		D	N	N	10 %	40 %	60 %	N	D	N	0,83							N	0,96	N
		15	D	N	N	N	N	N	N	D	N		A	N	N	10 %	20 %	80 %	N	D	N	0,54							N	0,55	N

Disturbance thresholds for listening test evaluation

D&K wave-file results

Figure A7: Example of result display.

Speech, anechoic

	Delay	Separation	Noise floor	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New Criterion
Direct						N	N	N	N	N	N	N	N	A	N		N	0 %	10 %	90 %	N	A					
Single	18			15,0		N	N	N	A	A	A	D	N	N	N		N	10 %	40 %	60 %	N	D	D	1,52	D	1,44	N
				10,0		N	N	N	D	N	D	D	A	N	N		N	30 %	40 %	60 %	N	D	D	1,37	D	1,26	N
				6,0		N	A	N	D	D	D	D	N	N	N		N	40 %	50 %	50 %	A	D	D	1,23	D	1,11	N
				3,0		N	D	N	A	N	D	D	N	N	N		N	30 %	40 %	60 %	N	D	D	1,12	D	1,00	N
				0,0		N	A	N	D	D	D	D	D	N	N		N	50 %	60 %	40 %	D	D	D	1,00	N	0,88	N
				-3,0		N	A	N	D	N	D	D	N	D	N		N	40 %	50 %	50 %	A	D	N	0,89	N	0,77	N
				-6,0		N	A	N	D	N	D	A	N	N	A		N	20 %	50 %	50 %	A	D	N	0,77	N	0,66	N
				-10,0		N	N	N	N	N	N	A	N	N	N		N	0 %	10 %	90 %	N	A	N	0,63	N	0,53	N
	30			0,0		N	D	N	D	D	D	D	N	D	N		A	60 %	60 %	40 %	D	D	D	1,60	D	1,40	D
				-5,0		N	D	N	D	D	D	D	A	D	A		N	60 %	80 %	20 %	D	D	D	1,35	D	1,16	D
				-8,0		N	A	N	D	A	A	A	N	N	A		N	10 %	60 %	40 %	A	D	D	1,17	N	0,99	N
				-11,0		N	A	N	N	D	A	D	N	N	N		N	20 %	40 %	60 %	N	D	D	1,00	N	0,83	N
				-14,0		N	N	N	N	N	N	A	N	N	A		N	0 %	20 %	80 %	N	A	N	0,85	N	0,69	N
	50			-9,8		N	D	N	D	D	D	D	D	D	A		D	70 %	80 %	20 %	D	D	D	1,78	D	1,48	D
				-13,8		N	D	N	D	A	A	D	A	D	A		D	40 %	80 %	20 %	A	D	D	1,43	D	1,16	N
				-16,8		N	A	N	A	N	A	D	A	D	A		A	20 %	70 %	30 %	A	D	D	1,20	N	0,95	N
				-19,8		N	A	A	N	N	N	A	A	A	D		D	10 %	60 %	40 %	A	D	D	1,00	N	0,77	N
				-22,8		N	N	N	N	N	N	A	N	N	A		N	0 %	20 %	80 %	N	A	N	0,82	N	0,61	N
	100			-27,1		D	D	D	A	A	A	D	A	D	D		D	60 %	100 %	0 %	D	D	D	1,23	N	0,84	D
				-30,1		A	A	D	N	A	A	A	A	A	D		A	20 %	90 %	10 %	A	D	D	1,00	N	0,62	N
				-33,1		A	A	D	N	N	N	A	N	A	D		A	20 %	60 %	40 %	A	D	N	0,81	N	0,48	N
				-36,1		N	N	A	N	N	A	A	N	A	A		A	0 %	50 %	50 %	A	A	N	0,65	N	0,34	N
				-40,1		N	N	A	N	N	N	N	N	N	A		N	0 %	20 %	80 %	N	A	N	0,49	N	0,22	N
	150			-25,8		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	2,02	D	1,42	D
				-29,8		A	D	D	A	A	D	D	D	A	D		D	60 %	100 %	0 %	D	D	D	1,54	N	0,97	D
				-32,8		A	D	D	A	A	D	A	D	A	D		D	50 %	100 %	0 %	D	D	D	1,24	N	0,74	D
				-35,8		A	A	D	N	A	A	A	A	A	D		D	20 %	90 %	10 %	A	D	D	1,00	N	0,53	N
				-38,8		N	N	D	A	N	A	A	N	A	D		A	20 %	60 %	40 %	A	D	N	0,81	N	0,38	N
	200			-33,3		A	D	D	A	A	D	A	D	A	D		D	50 %	100 %	0 %	D	D	D	1,54	N	0,89	D
				-36,3		A	D	D	A	A	D	A	D	A	D		D	50 %	100 %	0 %	D	D	D	1,24	N	0,60	D
				-39,8		N	A	A	A	N	A	A	A	A	D		A	10 %	80 %	20 %	A	D	D	1,00	N	0,46	N
				-42,8		N	N	A	N	N	N	A	N	A	A		N	0 %	40 %	60 %	N	A	N	0,80	N	0,32	N
				-45,8		N	N	N	N	N	N	N	N	A	A		N	0 %	20 %	80 %	N	A	N	0,64	N	0,20	N
	400			-34,1		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	3,02	D	1,74	D
				-39,1		A	D	D	A	A	D	D	D	A	D		D	50 %	100 %	0 %	D	D	D	2,11	D	1,00	D
				-43,1		A	D	D	A	A	D	A	D	A	D		D	50 %	100 %	0 %	D	D	D	1,57	N	0,62	D
				-46,1		A	N	D	A	A	D	A	N	A	D		A	30 %	80 %	20 %	A	D	D	1,26	N	0,38	N
				-49,1		N	N	N	N	N	N	N	N	N	A		N	0 %	10 %	90 %	N	A	D	1,00	N	0,25	N
Multiple	18	4		-4,0		N	D	N	D	A	D	D	N	N	N		N	40 %	50 %	50 %	A	D	D	1,68	D	1,45	N
				-9,0		N	A	N	D	N	D	D	A	N	N		N	30 %	50 %	50 %	A	D	D	1,47	D	1,23	N
				-14,0		N	N	N	D	N	A	A	N	N	N		N	10 %	30 %	70 %	N	D	D	1,24	N	0,99	N
				-19,0		N	N	N	N	N	N	A	N	N	N		N	0 %	10 %	90 %	N	A	D	1,01	N	0,77	N
				-24,0		N	N	N	N	N	A	N	N	N	N		N	0 %	10 %	90 %	N	A	N	0,79	N	0,57	N
		8		-6,5		N	D	N	D	N	D	D	N	A	N		N	40 %	50 %	50 %	A	D	D	1,58	D	1,34	N
				-11,5		N	A	N	A	N	D	D	N	N	A		N	20 %	50 %	50 %	A	D	D	1,36	D	1,11	N
				-16,5		N	N	N	D	N	A	A	N	N	N		N	10 %	30 %	70 %	N	D	D	1,12	N	0,88	N
				-26,5		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,69	N	0,47	N
		10		-7,6		N	D	N	D	N	D	D	N	N	N		N	40 %	40 %	60 %	N	D	D	1,53	D	1,29	N
				-12,6		N	N	N	D	N	A	D	A	N	N		N	20 %	40 %	60 %	N	D	D	1,30	D	1,06	N
				-17,6		N	N	N	N	N	A	N	N	N	N		N	0 %	10 %	90 %	N	A	D	1,07	N	0,83	N
				-27,6		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,65	N	0,43	N
		25		-18,7		A	D	N	D	N	A	D	D	A	D		D	50 %	80 %	20 %	D	D	N	0,90	N	0,77	N
				-21,7		A	D	A	A	N	A	A	A	A	A		D	10 %	90 %	10 %	A	D	N	0,80	N	0,66	N
				-25,7		N	N	N	N	N	N	A	N	A	A		D	0 %	30 %	70 %	N	A	N	0,67	N	0,52	N
				-28,7		N	N	N	N	N	N	A	N	N	A		A	0 %	20 %	80 %	N	A	N	0,58	N	0,40	N
				-33,7		N	N	N	N	N	N	N	N	N	A		N	0 %	10 %	90 %	N	A	N	0,44	N	0,27	N
		50		-28,1		A	D	D	A	D	D	D	D	D	D		D	80 %	100 %	0 %	D	D	D	1,05	N	0,76	D
				-31,1		A	D	D	A	A	A	A	A	D	D		D	40 %	100 %	0 %	A	D	N	0,89	N	0,59	N
				-34,1		A	A	D	N	N	A	A	A	A	D		D	20 %	80 %	20 %	A	D	N	0,75	N	0,46	N
				-37,1		A	A	D	N	N	A	A	N	A	A		A	10 %	70 %	30 %	A	D	N	0,62	N	0,32	N
				-40,1		N	N	N	A	N	N	A	N	N	A		N	0 %	30 %	70 %	N	A	N	0,51	N	0,25	N
	100	4		-25,8		D	D	D	A	D	D	D	D	D	D		D	90 %	100 %	0 %	D	D	D	3,38	D	2,34	D
				-30,8		D	D	A	A	A	A	D	A	D	A		D	40 %	100 %	0 %	A	D	D	2,54	D	1,59	D
				-33,8		A	A	A	A	A	A	A	A	A	A		D	0 %	100 %	0 %	A	A	D	2,12	D	1,25	N
				-37,8		N	A	N	N	N	N	A	N	A	N		A	0 %	30 %	70 %	N	A	D	1,64	N	0,82	N
				-40,8		N	N	A	N	N	N	A	N	N	N		A	0 %	20 %	80 %	N	A	D	1,34	N	0,60	N

Delay	Separation	Noise floor	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New Criterion
	8		-28,3		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	2,15	D	1,43	D
			-31,3		D	D	D	A	A	D	D	D	A	D		D	70 %	100 %	0 %	D	D	D	1,77	D	1,05	D
			-34,3		A	D	D	A	A	A	A	A	A	A		D	20 %	100 %	0 %	A	D	D	1,45	N	0,82	N
			-38,3		N	N	N	N	N	A	A	N	A	A		A	0 %	40 %	60 %	N	A	D	1,11	N	0,55	N
			-41,3		N	N	A	N	N	N	A	N	N	N		N	0 %	20 %	80 %	N	A	N	0,90	N	0,39	N
			-46,3		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,64	N	0,20	N
	10		-28,6		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	N	1,11	N	0,72	D
			-31,6		D	D	D	A	A	D	D	D	D	D		D	80 %	100 %	0 %	D	D	N	0,90	N	0,55	D
			-34,6		D	D	D	A	A	D	A	A	A	D		D	50 %	100 %	0 %	D	D	N	0,73	N	0,43	N
			-38,6		N	N	A	N	N	A	A	N	N	D		D	10 %	40 %	60 %	N	D	N	0,55	N	0,29	N
			-41,6		N	N	A	N	N	N	A	N	N	D		N	10 %	30 %	70 %	N	D	N	0,46	N	0,20	N
			-46,6		N	N	N	N	N	N	N	N	N	A		N	0 %	10 %	90 %	N	A	N	0,33	N	0,10	N
	25		-28,3		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	1,22	N	0,88	D
			-33,3		D	D	D	A	A	D	D	D	A	D		D	70 %	100 %	0 %	D	D	N	0,94	N	0,61	N
			-38,3		A	A	A	A	N	N	A	N	A	D		D	10 %	70 %	30 %	A	D	N	0,71	N	0,37	N
			-43,3		N	N	N	N	N	N	N	N	N	N		A	0 %	0 %	100 %	N	N	N	0,51	N	0,22	N
			-48,3		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,37	N	0,10	N
	50		-35,7		A	D	D	A	A	D	A	D	D	D		D	60 %	100 %	0 %	D	D	D	1,12	N	0,65	D
			-40,7		A	D	D	A	A	A	A	A	A	D		D	30 %	100 %	0 %	A	D	N	0,82	N	0,38	N
			-45,7		N	N	N	N	N	N	N	N	N	D		N	10 %	10 %	90 %	N	D	N	0,59	N	0,20	N
			-50,7		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,42	N	0,08	N
200	4		-34,8		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	3,90	D	2,20	D
			-39,8		A	D	D	D	A	A	A	D	A	D		D	50 %	100 %	0 %	D	D	D	2,81	D	1,35	D
			-44,8		N	A	A	N	N	N	A	N	A	A		A	0 %	50 %	50 %	A	A	D	1,99	N	0,77	N
			-48,8		N	N	N	N	N	N	A	N	N	N		N	0 %	10 %	90 %	N	A	D	1,50	N	0,38	N
			-54,8		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,97	N	0,08	N
	8		-35,0		D	D	D	D	D	D	A	D	A	D		D	80 %	100 %	0 %	D	D	D	2,72	D	1,47	D
			-40,0		A	A	D	A	N	A	A	D	A	D		D	30 %	90 %	10 %	A	D	D	1,93	N	0,89	D
			-45,0		N	N	A	N	N	A	A	N	A	A		N	0 %	50 %	50 %	A	A	D	1,35	N	0,50	N
			-49,0		N	N	A	N	N	N	N	N	N	A		N	0 %	20 %	80 %	N	A	D	1,01	N	0,26	N
			-55,0		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,65	N	0,08	N
	10		-35,2		A	D	D	A	D	D	D	D	A	D		D	70 %	100 %	0 %	D	D	D	1,40	N	0,76	D
			-40,2		A	D	D	A	A	A	A	D	A	D		D	40 %	100 %	0 %	A	D	N	0,97	N	0,45	D
			-45,2		N	N	D	N	A	A	A	A	A	A		D	10 %	70 %	30 %	A	D	N	0,67	N	0,23	N
			-49,2		N	N	A	N	N	N	A	N	A	A		N	0 %	40 %	60 %	N	A	N	0,50	N	0,13	N
			-55,2		N	N	N	N	N	N	N	N	N	N		N	0 %	0 %	100 %	N	N	N	0,33	N	0,08	N
	25		-31,1		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	1,87	D	1,20	D
			-36,1		A	D	D	A	A	D	D	D	A	D		D	60 %	100 %	0 %	D	D	D	1,33	N	0,79	D
			-41,1		A	D	D	A	A	D	A	D	A	D		D	50 %	100 %	0 %	D	D	N	0,98	N	0,44	N
			-44,1		N	N	D	N	N	A	A	N	A	D		A	20 %	50 %	50 %	A	D	N	0,81	N	0,34	N
			-48,1		N	N	A	N	N	N	N	N	N	A		N	0 %	20 %	80 %	N	A	N	0,62	N	0,17	N
	50		-32,5		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	D	1,97	D	1,31	D
			-37,5		D	D	D	A	A	D	A	D	A	D		D	60 %	100 %	0 %	D	D	D	1,48	N	0,77	D
			-42,5		A	A	D	A	A	D	A	D	A	D		D	40 %	100 %	0 %	A	D	D	1,08	N	0,49	N
			-45,5		A	A	D	N	A	D	A	A	A	D		A	30 %	90 %	10 %	A	D	N	0,89	N	0,36	N
			-49,5		N	N	N	N	N	N	N	N	N	A		N	0 %	10 %	90 %	N	A	N	0,68	N	0,15	N

Cello, anechoic

	Delay	Separation	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wav-files	D&K value	New Criterion
Direct				N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N					
Single	30		24,4	N	N	N	N	N			N	N	N	A	A		0 %	20 %	80 %	N	A	D	2,01	D	1,96	N
			20,4	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	D	1,94	D	1,85	N
			14,4	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,77	N	1,63	N
			7,4	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,50	N	1,31	N
			3,4	D	N	A	N	D			N	N	N	N	N		20 %	30 %	70 %	N	D	N	1,28	N	1,06	N
	50		15,0	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	D	3,03	D	2,82	N
			6,0	A	A	N	D	D			N	N	N	N	A		20 %	50 %	50 %	A	D	D	2,38	D	2,04	N
			0,0	D	N	A	N	D			N	N	N	A	A		20 %	50 %	50 %	A	D	N	1,79	N	1,42	N
			-6,0	N	N	N	A	N			N	N	A	A	A		0 %	40 %	60 %	N	A	N	1,19	N	0,89	N
			-15,0	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	0,54	N	0,37	N
	100		5,5	D	D	D	N	D			D	N	D	D	D		80 %	80 %	20 %	D	D	D	4,67	D	3,97	D
			0,5	D	N	D	D	D			D	A	D	D	D		80 %	90 %	10 %	D	D	D	3,68	D	2,94	D
			-3,5	D	N	A	A	D			A	N	D	D	A		40 %	80 %	20 %	A	D	D	2,86	D	2,19	D
			-6,5	A	A	N	N	N			A	N	A	D	N		10 %	50 %	50 %	A	D	D	2,29	N	1,70	D
			-9,5	A	N	N	N	A			A	N	A	N	N		0 %	40 %	60 %	N	A	N	1,79	N	1,29	D
	150		-3,9	D	D	A	D	D			A	N	D	D	A		60 %	90 %	10 %	D	D	D	4,17	D	3,18	D
			-7,9	A	N	A	A	D			A	N	N	A	A		10 %	70 %	30 %	A	D	D	3,08	D	2,25	D
			-10,9	A	A	N	N	N			A	N	A	A	A		0 %	60 %	40 %	A	A	D	2,38	N	1,69	D
			-13,9	A	N	N	N	N			A	N	A	A	N		0 %	40 %	60 %	N	A	D	1,80	N	1,25	D
			-16,9	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,34	N	0,92	D
	200		-6,8	D	D	D	A	D			D	A	D	D	A		70 %	100 %	0 %	D	D	D	4,48	D	3,31	D
			-10,8	A	A	A	A	D			A	N	A	D	N		20 %	80 %	20 %	A	D	D	3,20	D	2,28	D
			-13,8	A	D	N	N	A			A	N	A	D	A		20 %	70 %	30 %	A	D	D	2,42	N	1,69	D
			-16,8	A	N	N	D	N			A	N	A	A	A		10 %	60 %	40 %	A	D	D	1,80	N	1,23	D
			-19,8	A	N	N	N	N			N	N	A	A	N		0 %	30 %	70 %	N	A	N	1,33	N	0,89	D
	400		-17,4	A	D	D	D	D			D	D	A	D	A		70 %	100 %	0 %	D	D	D	3,40	D	2,31	D
			-20,4	A	D	D	D	A			D	D	D	A	A		60 %	100 %	0 %	D	D	D	2,49	N	1,66	D
			-23,4	A	A	A	D	A			D	A	A	D	A		30 %	100 %	0 %	A	D	D	1,81	N	1,18	D
			-26,4	A	N	A	A	N			A	A	A	A	A		0 %	80 %	20 %	A	A	N	1,31	N	0,82	D
			-29,4	N	A	A	D	A			A	A	A	A	A		10 %	90 %	10 %	A	D	N	0,94	N	0,56	D
Multiple	18	4	10,8	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	1,44	N	1,38	N
			5,8	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	1,34	N	1,25	N
			0,8	A	N	N	N	N			N	N	N	N	A		0 %	20 %	80 %	N	A	N	1,21	N	1,08	N
			-4,2	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,02	N	0,87	N
		13	6,4	D	N	D	N	D			N	N	N	A	N		30 %	40 %	60 %	N	D	N	1,41	N	1,22	N
			1,4	N	N	D	N	D			N	N	N	A	N		20 %	30 %	70 %	N	D	N	1,23	N	1,00	N
			-3,6	N	N	N	N	N			N	N	N	N	A		0 %	10 %	90 %	N	A	N	1,00	N	0,80	N
			-8,6	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,82	N	0,63	N
		15	5,7	D	N	A	N	N			N	N	N	A	N		10 %	30 %	70 %	N	D	N	0,85	N	0,72	N
			0,7	D	N	D	A	D			N	A	N	A	N		30 %	60 %	40 %	A	D	N	0,67	N	0,54	N
			-4,3	N	N	D	N	N			N	N	N	A	N		10 %	20 %	80 %	N	D	N	0,52	N	0,48	N
			-9,3	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,45	N	0,39	N
		25	2,7	D	N	D	N	D			N	N	N	N	N		30 %	30 %	70 %	N	D	N	0,87	N	0,84	N
			-2,3	A	N	A	N	D			N	A	N	A	N		10 %	50 %	50 %	A	D	N	0,82	N	0,76	N
			-7,3	A	N	N	N	N			N	N	N	D	N		10 %	20 %	80 %	N	D	N	0,72	N	0,63	N
			-12,3	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,58	N	0,47	N
		50	1,0	D	D	D	A	A			N	A	A	N	N		30 %	70 %	30 %	A	D	N	1,62	N	1,52	D
			-2,0	D	N	D	D	D			N	A	A	D	D		60 %	80 %	20 %	D	D	N	1,54	N	1,40	D
			-7,0	A	N	D	N	D			N	A	N	N	N		20 %	40 %	60 %	N	D	N	1,33	N	1,14	D
			-12,0	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	1,06	N	0,85	D
			-17,0	N	N	N	N	N			N	N	N	N	A		0 %	10 %	90 %	N	A	N	0,77	N	0,57	D
	100	4	-6,1	D	D	D	A	D			D	A	D	D	N		70 %	90 %	10 %	D	D	D	4,44	D	3,68	D
			-9,1	A	D	N	N	D			A	A	A	D	A		30 %	80 %	20 %	A	D	D	3,81	D	3,05	D
			-13,1	N	N	D	N	D			A	N	N	D	A		30 %	50 %	50 %	A	D	D	2,96	D	2,26	D
			-16,1	A	A	N	N	N			N	N	N	N	A		0 %	30 %	70 %	N	A	D	2,38	N	1,75	D
			-21,1	N	N	N	N	N			N	N	A	D	N		10 %	20 %	80 %	N	D	N	1,55	N	1,08	D
		13	-6,9	A	D	D	A	D			A	A	D	D	D		60 %	100 %	0 %	D	D	D	3,61	D	2,68	D
			-9,9	A	D	N	A	D			N	A	A	D	D		40 %	80 %	20 %	A	D	D	2,97	D	2,11	D
			-13,9	A	A	N	A	N			N	A	A	A	N		0 %	60 %	40 %	A	A	D	2,19	N	1,48	D
			-16,9	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,69	N	1,11	D
			-21,9	A	N	N	N	N			N	N	A	A	N		0 %	30 %	70 %	N	A	N	1,05	N	0,68	D
		15	-7,1	A	A	D	A	D			A	A	D	D	A		40 %	100 %	0 %	A	D	D	2,19	N	1,61	D
			-10,1	D	D	N	N	D			A	N	A	A	A		30 %	70 %	30 %	A	D	N	1,70	N	1,22	D
			-14,1	A	N	N	N	N			N	N	N	D	N		10 %	20 %	80 %	N	D	N	1,18	N	0,82	D
			-17,1	N	N	N	N	N			A	N	N	N	N		0 %	10 %	90 %	N	A	N	0,88	N	0,60	D
			-22,1	N	N	N	N	N			N	N	N	A	A		0 %	20 %	80 %	N	A	N	0,52	N	0,38	D

Delay	Separation	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wav-files	D&K value	New Criterion
	25	-7,8	A	D	D	N	D			N	A	A	D	D		50 %	80 %	20 %	D	D	D	2,07	N	1,52	D
		-10,8	A	D	A	N	A			A	A	A	A	A		10 %	90 %	10 %	A	D	N	1,60	N	1,14	D
		-14,8	A	D	N	N	A			A	N	N	A	N		10 %	50 %	50 %	A	D	N	1,10	N	0,80	D
		-17,8	N	N	N	N	N			N	N	A	A	A		0 %	30 %	70 %	N	A	N	1,02	N	0,74	D
		-22,8	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,58	N	0,41	D
	50	-4,5	D	D	D	D	D			A	A	D	D	D		80 %	100 %	0 %	D	D	D	2,67	D	2,02	D
		-9,5	D	D	D	A	D			A	N	D	D	N		60 %	80 %	20 %	D	D	N	1,79	N	1,53	D
		-12,5	A	A	A	A	A			A	A	D	D	A		20 %	100 %	0 %	A	D	N	1,58	N	1,29	D
		-16,5	N	D	A	N	N			N	N	A	D	N		20 %	40 %	60 %	N	D	N	1,26	N	0,97	D
		-19,5	N	N	N	A	N			A	N	A	A	N		0 %	40 %	60 %	N	A	N	1,02	N	0,75	D
		-24,5	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,67	N	0,46	D
200	4	-9,3	D	D	D	D	D			D	A	D	D	D		90 %	100 %	0 %	D	D	D	7,39	D	5,89	D
		-12,3	A	D	D	A	A			D	A	D	D	D		60 %	100 %	0 %	D	D	D	6,14	D	4,72	D
		-15,3	A	D	A	A	A			D	A	A	D	A		30 %	100 %	0 %	A	D	D	4,96	D	3,68	D
		-19,3	A	D	A	A	A			A	N	A	D	A		20 %	90 %	10 %	A	D	D	3,58	D	2,55	D
		-22,3	A	A	D	A	N			A	A	A	A	A		10 %	90 %	10 %	A	D	D	2,73	D	1,87	D
		-27,3	N	N	N	N	N			N	N	A	N	N		0 %	10 %	90 %	N	A	N	1,67	N	1,07	D
	13	-9,7	D	D	D	A	D			D	D	D	D	D		90 %	100 %	0 %	D	D	D	5,84	D	4,18	D
		-12,7	D	D	D	D	D			D	A	A	D	A		70 %	100 %	0 %	D	D	D	4,67	D	3,22	D
		-15,7	A	A	D	D	N			A	A	D	D	A		40 %	90 %	10 %	A	D	D	3,64	D	2,43	D
		-19,7	N	A	D	A	A			A	N	A	A	A		10 %	80 %	20 %	A	D	D	2,53	N	1,63	D
		-22,7	A	A	N	A	N			A	N	N	A	A		0 %	60 %	40 %	A	A	D	1,89	N	1,18	D
		-27,7	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,12	N	0,68	D
	15	-9,8	D	D	D	D	A			D	A	D	D	A		70 %	100 %	0 %	D	D	D	3,49	D	2,51	D
		-12,8	D	D	D	D	D			D	A	D	D	A		80 %	100 %	0 %	D	D	D	2,66	D	1,87	D
		-15,8	A	A	A	A	A			D	A	A	A	A		10 %	100 %	0 %	A	D	D	1,99	N	1,37	D
		-19,8	N	A	A	A	A			A	N	A	D	A		10 %	80 %	20 %	A	D	N	1,33	N	0,89	D
		-22,8	A	A	N	N	A			A	N	A	A	N		0 %	60 %	40 %	A	A	N	0,97	N	0,63	D
		-27,8	N	N	N	A	N			N	N	A	A	A		0 %	40 %	60 %	N	A	N	0,56	N	0,36	D
	25	-10,2	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	D	3,37	D	3,38	D
		-13,2	D	D	D	D	D			D	A	D	D	D		90 %	100 %	0 %	D	D	D	2,56	D	2,57	D
		-16,2	A	D	A	N	A			D	N	D	A	A		30 %	80 %	20 %	A	D	D	1,92	D	1,91	D
		-20,2	A	A	A	N	A			A	N	A	A	A		0 %	80 %	20 %	A	A	N	1,27	N	1,26	D
		-23,2	N	A	N	A	N			A	N	A	A	A		0 %	60 %	40 %	A	A	N	0,92	N	0,91	D
	50	-10,2	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	D	3,09	D	3,09	D
		-13,2	A	D	A	D	N			D	A	D	D	D		60 %	90 %	10 %	D	D	D	2,33	D	2,33	D
		-16,2	D	D	N	A	A			D	A	D	D	A		50 %	90 %	10 %	D	D	N	1,75	N	1,75	D
		-20,2	A	A	A	D	N			A	A	A	D	A		20 %	90 %	10 %	A	D	N	1,32	N	1,31	D
		-23,2	A	A	A	A	N			A	N	A	A	A		0 %	80 %	20 %	A	A	N	1,02	N	1,00	D
		-26,2	N	N	A	D	N			A	N	A	A	A		10 %	60 %	40 %	A	D	N	0,85	N	0,83	D

Guitar, anechoic

	Delay	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New Criterion
Direct			N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N					
Single	30	28,4	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	D	2,066	D	2,06	N
		19,4	A	N	N	N	N			N	N	N	N	N		0 %	10 %	90 %	N	A	D	1,94	D	1,85	N
		13,4	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	1,77	N	1,63	N
		7,4	N	N	A	N	A			A	N	N	N	N		0 %	30 %	70 %	N	A	N	1,50	N	1,31	N
		3,4	D	N	A	N	A			A	N	N	D	N		20 %	50 %	50 %	A	D	N	1,28	N	1,06	N
		-1,6	A	N	D	N	D			N	N	N	A	N		20 %	40 %	60 %	N	D	N	0,97	N	0,76	N
		-6,6	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	0,68	N	0,51	N
	50	6,0	D	A	A	N	A			D	N	D	D	N		40 %	70 %	30 %	A	D	D	2,38	D	2,04	N
		3,0	A	N	A	N	D			A	N	D	D	N		30 %	60 %	40 %	A	D	D	2,09	N	1,73	N
		0,0	D	A	A	N	A			A	N	A	D	N		20 %	70 %	30 %	A	D	N	1,79	N	1,42	N
		-3,0	A	A	N	N	N			N	N	A	D	N		10 %	40 %	60 %	N	D	N	1,48	N	1,14	N
		-6,0	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	1,19	N	0,89	N
	100	-3,5	D	D	A	N	D			D	A	D	D	A		60 %	90 %	10 %	D	D	D	2,86	D	2,19	D
		-6,5	D	D	N	N	D			D	N	D	D	A		60 %	70 %	30 %	D	D	D	2,29	N	1,70	D
		-9,5	A	A	N	N	A			A	A	D	A	N		10 %	70 %	30 %	A	D	N	1,79	N	1,29	D
		-12,5	A	N	N	N	N			N	N	A	A	N		0 %	30 %	70 %	N	A	N	1,37	N	0,96	D
		-15,5	A	N	N	N	N			N	N	A	N	N		0 %	20 %	80 %	N	A	N	1,03	N	0,71	D
	150	-7,9	D	D	D	N	D			D	D	D	D	A		80 %	90 %	10 %	D	D	D	3,08	D	2,25	D
		-10,9	D	D	N	N	D			D	D	D	D	A		70 %	80 %	20 %	D	D	D	2,38	N	1,69	D
		-13,9	D	D	N	N	D			A	A	A	A	N		30 %	70 %	30 %	A	D	D	1,80	N	1,25	D
		-16,9	N	D	A	N	A			N	A	A	A	A		10 %	70 %	30 %	A	D	N	1,34	N	0,92	D
		-19,9	A	A	N	N	N			A	A	A	N	N		0 %	50 %	50 %	A	A	N	0,98	N	0,66	D
	200	-10,8	D	D	N	N	D			D	D	D	D	D		80 %	80 %	20 %	D	D	D	3,20	D	2,28	D
		-13,8	D	D	N	N	D			D	D	D	D	D		80 %	80 %	20 %	D	D	D	2,42	N	1,69	D
		-16,8	D	D	N	N	D			D	A	D	A	A		50 %	80 %	20 %	D	D	D	1,80	N	1,23	D
		-19,8	A	A	N	N	D			A	N	A	N	A		10 %	60 %	40 %	A	D	N	1,33	N	0,89	D
		-22,8	A	A	N	N	A			N	N	N	N	N		0 %	30 %	70 %	N	A	N	0,97	N	0,63	D
	400	-17,4	D	D	A	N	D			D	A	D	D	D		70 %	90 %	10 %	D	D	D	3,40	D	2,31	D
		-20,4	D	D	N	N	D			D	D	D	D	A		70 %	80 %	20 %	D	D	D	2,49	N	1,66	D
		-23,4	D	A	A	N	A			A	D	A	A	N		20 %	80 %	20 %	A	D	D	1,81	N	1,18	D
		-26,4	A	A	N	N	A			A	A	A	N	A		0 %	70 %	30 %	A	A	N	1,31	N	0,82	D
		-29,4	A	A	N	N	N			A	N	A	N	N		0 %	40 %	60 %	N	A	N	0,94	N	0,56	D

Orchestra, anechoic

	Delay	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New Criterion
Direct			N	N	A	A	N			N	N	N	N	N		0 %	20 %	80 %	N	A					
Single	30	28,4	A	N	N	N	A			N	N	N	N	N		0 %	20 %	80 %	N	A	D	2,066	D	2,06	N
		19,4	A	N	N	N	A			N	N	N	N	N		0 %	20 %	80 %	N	A	D	1,94	D	1,85	N
		13,4	A	N	N	N	N			N	N	N	N	N		0 %	10 %	90 %	N	A	N	1,77	N	1,63	N
		7,4	A	N	N	N	N			N	N	N	A	N		0 %	20 %	80 %	N	A	N	1,50	N	1,31	N
		-1,6	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	0,97	N	0,76	N
	50	15,0	A	N	N	N	A			N	N	N	A	N		0 %	30 %	70 %	N	A	D	3,03	D	2,82	N
		6,0	N	N	N	A	N			N	N	N	N	N		0 %	10 %	90 %	N	A	D	2,38	D	2,04	N
		0,0	A	N	N	N	N			N	N	N	N	N		0 %	10 %	90 %	N	A	N	1,79	N	1,42	N
		-6,0	A	N	N	N	N			N	N	N	N	N		0 %	10 %	90 %	N	A	N	1,19	N	0,89	N
	100	10,5	D	N	A	A	D			D	A	A	D	D		50 %	90 %	10 %	D	D	D	5,50	D	4,93	D
		5,5	D	D	N	A	D			D	N	D	D	D		70 %	80 %	20 %	D	D	D	4,67	D	3,97	D
		0,5	A	N	A	N	N			N	N	A	A	A		0 %	50 %	50 %	A	A	D	3,68	D	2,94	D
		-3,5	D	N	A	D	N			A	N	N	D	N		30 %	50 %	50 %	A	D	D	2,86	D	2,19	D
		-6,5	N	N	A	N	N			N	N	N	A	N		0 %	20 %	80 %	N	A	D	2,29	N	1,70	D
		-9,5	A	N	N	N	N			N	N	N	A	N		0 %	20 %	80 %	N	A	N	1,79	N	1,29	D
	150	1,1	D	D	D	D	D			D	A	D	D	A		80 %	100 %	0 %	D	D	D	5,70	D	4,60	D
		-3,9	D	D	D	D	N			D	N	D	D	A		70 %	80 %	20 %	D	D	D	4,17	D	3,18	D
		-7,9	A	N	N	D	N			A	N	A	A	A		10 %	60 %	40 %	A	D	D	3,08	D	2,25	D
		-10,9	D	N	N	D	A			A	N	A	A	N		20 %	60 %	40 %	A	D	D	2,38	N	1,69	D
		-13,9	A	N	N	N	N			N	N	A	A	N		0 %	30 %	70 %	N	A	D	1,80	N	1,25	D
	200	-1,8	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	D	6,41	D	4,99	D
		-6,8	D	D	D	D	N			D	N	D	D	A		70 %	80 %	20 %	D	D	D	4,48	D	3,31	D
		-10,8	D	A	A	D	A			A	N	D	D	A		40 %	90 %	10 %	A	D	D	3,20	D	2,28	D
		-13,8	D	N	N	D	A			N	N	A	A	N		20 %	50 %	50 %	A	D	D	2,42	N	1,69	D
		-16,8	D	N	N	N	A			N	N	A	A	A		10 %	50 %	50 %	A	D	D	1,80	N	1,23	D
	400	-8,4	D	D	D	D	D			D	A	D	D	D		90 %	100 %	0 %	D	D	D	7,87	D	5,73	D
		-13,4	D	D	A	D	D			A	A	A	D	A		50 %	100 %	0 %	D	D	D	5,03	D	3,52	D
		-17,4	A	A	N	D	A			A	A	A	A	N		10 %	80 %	20 %	A	D	D	3,40	D	2,31	D
		-20,4	A	A	A	D	N			A	N	A	A	N		10 %	70 %	30 %	A	D	D	2,49	N	1,66	D
		-23,4	A	A	N	A	A			A	N	A	A	N		0 %	70 %	30 %	A	A	D	1,81	N	1,18	D

Trumpet, anechoic

	Delay	Separation	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New criterion
Direct				N	N	N	N	N	N	N	N	N		N	N	N	0 %	0 %	100 %	N	N					
Single	30		28,4	D	N	A	A	D	N	A	A	N		A	A	D	10 %	70 %	30 %	A	D	D	2,066	D	2,06	N
			19,4	D	N	N	N	D	A	D	D	N		A	A	D	30 %	60 %	40 %	A	D	D	1,94	D	1,85	N
			13,4	D	N	N	N	D	A	D	A	N		N	A	D	20 %	50 %	50 %	A	D	N	1,77	N	1,63	N
			7,4	D	N	N	A	D	N	A	D	N		N	N	D	20 %	40 %	60 %	N	D	N	1,50	N	1,31	N
			-1,6	D	N	D	N	A	N	A	A	N		N	N	N	10 %	40 %	60 %	N	D	N	0,97	N	0,76	N
	50		-6,0	D	N	D	N	N	N	N	D	N		N	N	N	20 %	20 %	80 %	N	D	N	1,19	N	0,89	N
			-10,0	D	N	N	N	A	N	D	A	N		N	A	A	10 %	40 %	60 %	N	D	N	0,86	N	0,62	N
			-15,0	A	N	N	A	D	N	N	A	N		A	N	D	10 %	40 %	60 %	N	D	N	0,54	N	0,37	N
			-20,0	A	N	N	A	A	N	N	A	N		A	N	A	0 %	40 %	60 %	N	A	N	0,32	N	0,22	N
			-25,0	N	N	N	N	A	N	N	A	N		A	N	D	0 %	30 %	70 %	N	A	N	0,19	N	0,12	N
			-30,0	A	N	N	N	A	N	N	A	N		N	N	N	0 %	20 %	80 %	N	A	N	0,11	N	0,07	N
	100		-19,5	D	A	D	D	D	D	D	D	D		D	A	N	80 %	100 %	0 %	D	D	N	0,68	N	0,46	D
			-24,5	D	N	A	D	D	D	D	D	N		A	A	D	50 %	80 %	20 %	D	D	N	0,40	N	0,26	D
			-29,5	D	D	D	D	A	D	D	A	D		A	N	D	60 %	90 %	10 %	D	D	N	0,23	N	0,14	D
			-34,5	D	A	A	A	D	A	A	A	N		A	A	D	10 %	90 %	10 %	A	D	N	0,13	N	0,07	N
			-39,5	D	A	A	D	N	N	A	A	A		A	N	D	10 %	70 %	30 %	A	D	N	0,07	N	0,04	N
	150		-23,9	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,64	N	0,42	D
			-28,9	D	D	A	D	D	D	A	A	D		D	D	D	70 %	100 %	0 %	D	D	N	0,37	N	0,22	D
			-33,9	D	A	A	D	D	D	D	D	A		A	D	D	60 %	100 %	0 %	D	D	N	0,21	N	0,12	D
			-38,9	N	A	D	D	D	A	A	A	A		D	D	D	50 %	100 %	0 %	D	D	N	0,12	N	0,06	D
			-43,9	D	A	D	D	N	A	A	D	D		A	A	A	40 %	90 %	10 %	A	D	N	0,07	N	0,04	N
	200		-26,8	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,62	N	0,39	D
			-31,8	D	D	D	D	D	D	D	D	D		A	A	D	80 %	100 %	0 %	D	D	N	0,36	N	0,21	D
			-36,8	D	D	D	D	D	D	D	D	D		A	A	D	80 %	100 %	0 %	D	D	N	0,20	N	0,10	D
			-41,8	D	A	D	D	A	D	A	D	D		D	A	D	60 %	100 %	0 %	D	D	N	0,12	N	0,05	D
			-46,8	D	A	D	D	A	A	N	A	A		A	A	A	20 %	90 %	10 %	A	D	N	0,04	N	0,07	N
	400		-29,4	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,94	N	0,56	D
			-33,4	D	D	D	D	D	D	D	D	D		D	A	D	90 %	100 %	0 %	D	D	N	0,60	N	0,34	D
			-38,4	D	D	D	D	D	D	A	D	D		A	A	D	70 %	100 %	0 %	D	D	N	0,34	N	0,17	D
			-43,4	D	D	D	D	A	A	A	D	D		A	A	D	50 %	100 %	0 %	D	D	N	0,19	N	0,09	D
			-48,4	D	A	A	N	N	A	N	A	A		A	N	D	0 %	60 %	40 %	A	A	N	0,11	N	0,04	N
Multiple	18	4	10,8	D	N	N	N	D	N	N	A	N		N	A	N	10 %	30 %	70 %	N	D	N	1,44	N	1,38	N
			5,8	D	N	D	N	N	N	A	A	N		N	N	N	10 %	30 %	70 %	N	D	N	1,34	N	1,25	N
			0,8	D	N	N	N	N	N	N	D	N		N	N	N	10 %	10 %	90 %	N	D	N	1,21	N	1,08	N
			-4,2	D	N	N	N	N	N	N	A	N		N	N	N	0 %	10 %	90 %	N	A	N	1,02	N	0,87	N
		13	-3,6	D	N	D	A	N	N	N	D	N		A	N	N	20 %	40 %	60 %	N	D	N	1,00	N	0,80	N
			-8,6	D	N	N	N	N	N	N	D	N		A	A	N	10 %	30 %	70 %	N	D	N	0,82	N	0,63	N
			-13,6	D	N	N	N	A	N	N	A	N		A	N	D	0 %	30 %	70 %	N	A	N	0,63	N	0,45	N
			-18,6	N	N	N	N	N	N	N	A	N		N	N	A	0 %	10 %	90 %	N	A	N	0,44	N	0,30	N
			-23,6	N	N	A	A	N	N	N	A	N		N	N	N	0 %	30 %	70 %	N	A	N	0,29	N	0,18	N
		15	-4,3	D	N	N	N	A	N	A	D	N		A	N	N	10 %	40 %	60 %	N	D	N	0,52	N	0,48	N
			-9,3	D	N	N	A	N	N	A	A	N		A	N	A	0 %	40 %	60 %	N	A	N	0,45	N	0,39	N
			-14,3	D	N	N	A	N	N	A	A	N		N	N	D	0 %	30 %	70 %	N	A	N	0,36	N	0,28	N
			-19,3	N	N	N	A	D	N	A	A	N		N	N	D	10 %	40 %	60 %	N	D	N	0,25	N	0,19	N
			-24,3	A	N	N	N	N	N	N	A	N		N	N	N	0 %	10 %	90 %	N	A	N	0,17	N	0,11	N
		25	-12,3	D	N	N	A	N	D	A	D	N		A	N	D	20 %	50 %	50 %	A	D	N	0,58	N	0,47	N
			-17,3	D	N	N	D	N	A	A	D	N		N	N	D	20 %	40 %	60 %	N	D	N	0,43	N	0,32	N
			-22,3	A	N	N	D	A	A	A	A	N		N	N	D	10 %	50 %	50 %	A	D	N	0,29	N	0,20	N
			-27,3	D	N	A	D	N	A	N	A	N		N	N	D	10 %	40 %	60 %	N	D	N	0,18	N	0,12	N
			-32,3	D	N	N	D	N	N	N	A	N		N	A	A	10 %	30 %	70 %	N	D	N	0,11	N	0,06	N
			-37,3	A	N	N	A	N	N	A	A	N		N	N	N	0 %	30 %	70 %	N	A	N	0,06	N	0,04	N
		50	-32,0	D	N	A	D	A	D	D	A	D		A	A	D	40 %	90 %	10 %	A	D	N	0,19	N	0,12	D
			-35,0	D	A	A	D	A	N	N	A	A		A	N	D	10 %	70 %	30 %	A	D	N	0,14	N	0,08	N
			-38,0	D	A	D	D	A	A	N	A	A		N	A	D	20 %	80 %	20 %	A	D	N	0,10	N	0,05	N
			-42,0	D	A	D	D	A	A	N	A	A		N	A	D	20 %	80 %	20 %	A	D	N	0,07	N	0,04	N
			-47,0	D	N	N	D	N	A	N	A	N		A	A	N	10 %	50 %	50 %	A	D	N	0,04	N	0,04	N
	100	4	-33,1	D	A	D	D	D	D	D	A	A		A	A	D	50 %	100 %	0 %	D	D	N	0,46	N	0,27	D
			-36,1	D	A	D	D	N	A	A	A	D		D	A	D	40 %	90 %	10 %	A	D	N	0,33	N	0,18	N
			-41,1	A	A	A	D	N	A	N	A	D		A	A	D	20 %	80 %	20 %	A	D	N	0,19	N	0,09	N
			-46,1	A	A	D	D	A	A	N	A	A		A	A	D	20 %	90 %	10 %	A	D	N	0,11	N	0,04	N
			-51,1	N	N	N	A	N	A	N	A	A		N	N	N	0 %	40 %	60 %	N	A	N	0,06	N	0,04	N
		13	-33,9	D	N	A	D	D	N	D	D	A		D	A	D	50 %	80 %	20 %	D	D	N	0,32	N	0,18	D
			-36,9	D	A	A	D	D	A	D	A	A		D	A	D	40 %	100 %	0 %	A	D	N	0,23	N	0,12	N
			-41,9	D	A	A	D	N	N	A	A	D		A	A	D	20 %	80 %	20 %	A	D	N	0,13	N	0,06	N
			-46,9	D	N	A	D	A	A	N	A	A		A	A	A	10 %	80 %	20 %	A	D	N	0,08	N	0,04	N
			-51,9	D	N	A	A	N	N	N	A	A		A	N	N	0 %	50 %	50 %	A	A	N	0,04	N	0,04	N

Delay	Separation	Amp. re direct	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram	D&K value	Wave-files	D&K value	New criterion
	15	-34,1	D	A	D	D	D	A	A	D	D		A	D	D	60 %	100 %	0 %	D	D	N	0,17	N	0,10	D
		-37,1	D	A	A	D	N	A	A	D	A		A	D	D	30 %	90 %	10 %	A	D	N	0,12	N	0,06	N
		-42,1	D	N	N	D	N	D	N	A	D		A	A	D	30 %	60 %	40 %	A	D	N	0,07	N	0,04	N
		-47,1	D	N	A	D	A	N	N	A	A		A	A	A	10 %	70 %	30 %	A	D	N	0,04	N	0,04	N
		-52,1	A	N	N	A	A	N	N	A	A		N	N	N	0 %	40 %	60 %	N	A	N	0,02	N	0,04	N
	25	-30,8	D	A	D	D	D	D	D	D	N		A	D	D	70 %	90 %	10 %	D	D	N	0,273	N	0,17	D
		-34,8	D	A	D	D	D	D	A	D	A		D	A	D	60 %	100 %	0 %	D	D	N	0,18	N	0,10	D
		-37,8	D	A	D	D	N	D	A	A	D		A	A	D	40 %	90 %	10 %	A	D	N	0,13	N	0,07	D
		-42,8	D	N	A	D	N	A	A	A	D		A	A	D	20 %	80 %	20 %	A	D	N	0,08	N	0,04	N
		-47,8	A	N	A	D	N	D	N	A	A		A	N	A	20 %	60 %	40 %	A	D	N	0,04	N	0,04	N
	50	-32,5	D	D	D	D	D	A	D	D	A		D	D	D	80 %	100 %	0 %	D	D	N	0,31	N	0,18	D
		-36,5	D	D	D	D	D	D	A	D	D		D	A	D	80 %	100 %	0 %	D	D	N	0,20	N	0,11	D
		-39,5	D	A	D	D	D	D	A	D	A		D	A	D	60 %	100 %	0 %	D	D	N	0,14	N	0,07	D
		-42,5	D	A	A	D	N	A	N	A	D		A	D	D	30 %	80 %	20 %	A	D	N	0,10	N	0,05	N
		-45,5	D	A	D	D	N	A	N	A	A		A	A	D	20 %	80 %	20 %	A	D	N	0,07	N	0,04	N
200	4	-39,3	D	D	D	D	D	D	D	A	D		A	A	D	70 %	100 %	0 %	D	D	N	0,46	N	0,23	D
		-42,3	D	A	D	D	D	D	D	D	D		D	A	D	80 %	100 %	0 %	D	D	N	0,33	N	0,15	D
		-45,3	D	D	D	D	D	A	A	A	D		D	A	D	60 %	100 %	0 %	D	D	N	0,23	N	0,09	D
		-48,3	D	D	D	D	A	D	A	D	D		A	A	D	60 %	100 %	0 %	D	D	N	0,17	N	0,52	N
		-52,3	A	A	A	D	A	A	N	A	D		A	N	D	20 %	80 %	20 %	A	D	N	0,11	N	0,04	N
	13	-35,7	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,48	N	0,26	D
		-39,7	D	D	D	D	A	D	D	D	D		D	A	D	80 %	100 %	0 %	D	D	N	0,31	N	0,16	D
		-42,7	D	D	A	D	D	A	A	D	D		D	A	D	60 %	100 %	0 %	D	D	N	0,22	N	0,10	D
		-47,7	D	A	D	D	A	A	A	D	D		A	A	D	40 %	100 %	0 %	A	D	N	0,13	N	0,40	N
		-52,7	A	A	N	D	A	A	A	A	D		A	N	A	20 %	80 %	20 %	A	D	N	0,07	N	0,35	N
	15	-35,8	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,25	N	0,14	D
		-39,8	D	D	D	D	D	D	A	D	D		D	A	D	80 %	100 %	0 %	D	D	N	0,16	N	0,08	D
		-42,8	D	D	D	D	A	D	D	A	D		A	A	D	60 %	100 %	0 %	D	D	N	0,11	N	0,05	D
		-47,8	D	D	D	D	A	A	A	D	A		A	A	D	40 %	100 %	0 %	A	D	N	0,07	N	0,04	N
		-52,8	A	A	A	D	A	A	N	A	A		A	N	D	10 %	80 %	20 %	A	D	N	0,04	N	0,04	N
	25	-36,2	D	D	D	D	N	D	D	D	D		D	D	D	90 %	90 %	10 %	D	D	N	0,26	N	0,23	D
		-40,2	D	D	D	D	D	A	D	D	D		A	D	D	80 %	100 %	0 %	D	D	N	0,17	N	0,14	D
		-43,2	D	D	D	D	A	A	A	D	D		A	D	D	60 %	100 %	0 %	D	D	N	0,12	N	0,09	D
		-46,2	A	D	D	D	D	A	A	D	D		A	D	D	70 %	100 %	0 %	D	D	N	0,09	N	0,60	N
		-49,2	D	A	A	D	N	A	A	D	D		A	A	D	30 %	90 %	10 %	A	D	N	0,06	N	0,04	N
	50	-36,2	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,28	N	0,15	D
		-40,2	D	D	D	D	D	D	A	D	D		A	D	D	80 %	100 %	0 %	D	D	N	0,18	N	0,08	D
		-43,2	D	D	A	D	D	D	A	D	D		A	D	D	70 %	100 %	0 %	D	D	N	0,13	N	0,06	N
		-46,2	A	D	D	D	N	A	A	A	D		A	A	D	40 %	90 %	10 %	A	D	N	0,09	N	0,04	N
		-49,2	A	A	A	D	N	A	A	D	D		A	A	D	30 %	90 %	10 %	A	D	N	0,07	N	0,04	N

Speech, reverberant

	Scat. Coeff.	Rec. No.	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram 1st order	D&K value	2nd order	D&K value	3rd order	D&K value	4th order	D&K value	Wave-files	D&K value	New Criterion
Reverberation	10	1		A	D	D	D	A	A	D	A	A	D		D	50 %	100 %	0 %	D	D	N	0,64	N	0,94	D	1,19	D	1,19	N	0,89	D
		2		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	N	0,54	N	0,85	D	1,00	D	1,12	N	0,75	D
		3		D	D	D	D	D	D	D	A	D	D		D	90 %	100 %	0 %	D	D	N	0,41	N	0,66	N	0,86	N	0,86	N	0,68	D
		4		A	D	D	D	D	D	D	A	D	D		D	80 %	100 %	0 %	D	D	N	0,40	N	0,58	N	0,77	N	0,93	N	0,62	D
		5		D	D	D	D	D	D	D	D	D	D		D	100 %	100 %	0 %	D	D	N	0,44	N	0,59	N	0,64	N	0,65	N	0,64	D
		6		A	D	D	D	D	D	D	D	D	D		D	90 %	100 %	0 %	D	D	N	0,41	N	0,67	N	0,70	N	0,70	N	0,60	D
	50	1		N	N	A	N	N	N	A	N	A	A		N	0 %	40 %	60 %	N	A	N	0,61	N	0,88	D	1,09	D	1,09	N	0,77	N
		2		N	N	A	N	N	N	A	N	N	N		N	0 %	20 %	80 %	N	A	N	0,51	N	0,83	D	1,01	D	1,01	N	0,66	N
		3		N	N	D	A	N	A	D	N	A	A		A	20 %	60 %	40 %	A	D	N	0,65	N	0,87	N	0,87	N	0,87	N	0,75	N
		4		N	N	A	N	N	N	A	N	N	A		N	0 %	30 %	70 %	N	A	N	0,43	N	0,63	N	0,69	N	0,70	N	0,72	N
		5		N	A	D	A	D	N	D	A	D	A		D	40 %	80 %	20 %	A	D	N	0,49	N	0,65	N	0,71	N	0,70	N	0,72	N
		6		A	A	D	D	A	D	D	A	A	D		A	50 %	100 %	0 %	D	D	N	0,45	N	0,68	D	1,19	N	0,71	N	0,65	D

Orchestra, reverberant

	Scat. Coeff.	Rec. No.	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	Reflectogram 1st order	D&K value	2nd order	D&K value	3rd order	D&K value	4th order	D&K value	Wave-files	D&K value	New criterion
Backwall	10	17	A	D	N	N	A			A	N	A	A	A		10 %	70 %	30 %	A	D	N	0,84							N	1,07	D
		1	D	D	A	D	A			A	N	A	D	N		40 %	80 %	20 %	A	D	N	1,24							N	1,77	D
		7	N	A	A	N	N			N	N	N	N	N		0 %	20 %	80 %	N	A	N	1,71							D	2,33	D
		15	A	N	N	N	N			N	N	A	N	N		0 %	20 %	80 %	N	A	N	0,54							N	0,55	N
Reverberation	10	17	A	D	N	N	A			N	N	A	N	N		10 %	40 %	60 %	N	D	N	0,43	N	1,02	N	1,04	N	1,02	N	1,00	N
		1	D	A	A	N	N			A	N	A	D	N		20 %	60 %	40 %	A	D	N	0,78	N	1,23	N	1,29	N	1,37	N	0,97	N
		7	N	A	N	N	N			N	N	N	D	N		10 %	20 %	80 %	N	D	N	0,72	N	0,90	N	0,93	N	0,93	N	0,75	N
		15	A	N	N	D	N			N	N	N	D	N		20 %	30 %	70 %	N	D	N	0,72	N	0,89	N	0,92	N	0,92	N	0,85	N
	70	17	A	N	N	A	N			N	N	A	A	N		0 %	40 %	60 %	N	A	N	0,43	N	0,61	N	0,73	N	0,73	N	1,49	N
		1	A	D	A	A	N			N	N	N	A	N		10 %	50 %	50 %	A	D	N	0,80	N	1,07	N	1,11	N	1,11	N	1,59	N
		7	N	N	N	N	N			N	N	A	D	N		10 %	20 %	80 %	N	D	N	0,67	N	1,00	N	1,03	N	1,03	N	1,03	N
		15	N	N	N	N	N			N	N	A	A	N		0 %	20 %	80 %	N	A	N	0,69	N	0,86	N	0,89	N	0,89	N	0,90	N

Trumpet, reverberant

																					Reflectogram										
	Scat. Coeff.	Receiver	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	1st order	D&K value	2nd order	D&K value	3rd order	D&K value	4th order	D&K value	Wave-files	D&K value	New criterion
Backwall	10	17	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,84							N	1,07	D
		1	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	1,24							N	1,77	D
		7	D	D	D	D	D	D	A	D	D		D	A	D	80 %	100 %	0 %	D	D	N	1,71							D	2,33	D
		11	D	N	D	A	D	N	N	D	N		D	A	N	40 %	60 %	40 %	A	D	N	1,44							D	1,80	D
		13	D	N	N	A	D	N	A	A	N		A	A	N	10 %	60 %	40 %	A	D	N	1,08							N	1,22	N
		14	D	N	N	N	A	N	A	A	N		D	N	N	10 %	40 %	60 %	N	D	N	0,83							N	0,96	N
		15	D	N	N	N	N	N	N	D	N		A	N	N	10 %	20 %	80 %	N	D	N	0,54							N	0,55	N
	70	17	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,49							D	2,28	D
		1	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	0,75							D	3,27	D
		7	D	D	D	D	D	D	D	D	D		D	D	D	100 %	100 %	0 %	D	D	N	1,09							D	3,55	D
		11	D	D	D	D	D	A	D	D	N		D	A	A	70 %	90 %	10 %	D	D	N	0,96							D	2,70	D
		13	D	A	N	A	D	N	D	D	N		A	N	N	30 %	60 %	40 %	A	D	N	0,74							D	1,98	D
		14	D	N	D	N	N	D	D	D	N		N	N	D	40 %	40 %	60 %	N	D	N	0,58							N	1,59	N
		15	D	N	N	N	N	N	N	A	N		N	N	N	0 %	10 %	90 %	N	A	N	0,38							N	1,01	N
Reverberation	10	17	A	D	D	D	D	D	A	D	D		A	D	D	80 %	100 %	0 %	D	D	N	0,43	N	1,02	N	1,036	N	1,02	N	1,00	D
		1	D	D	A	D	N	D	A	D	D		A	A	N	50 %	90 %	10 %	D	D	N	0,776	N	1,23	N	1,294	N	1,368	N	0,97	D
		4	D	D	D	D	N	N	A	A	D		A	N	A	40 %	70 %	30 %	A	D	N	0,728	N	1,03	N	1,064	N	1,137	N	0,65	N
		7	N	D	D	D	N	N	N	A	N		A	N	N	30 %	50 %	50 %	A	D	N	0,721	N	0,90	N	0,93	N	0,93	N	0,75	N
		11	D	D	D	D	D	A	N	D	D		D	A	D	70 %	90 %	10 %	D	D	N	0,631	N	0,94	N	0,97	N	0,97	N	0,94	D
		13	N	D	N	D	N	N	N	A	N		N	N	N	20 %	30 %	70 %	N	D	N	0,619	N	0,87	N	0,891	N	0,891	N	0,88	D
		16	D	N	D	D	D	D	A	D	D		A	A	D	60 %	90 %	10 %	D	D	N	0,416	N	0,64	N	0,712	N	0,712	N	0,75	D
	70	17	N	D	A	D	N	A	N	A	N		N	A	N	20 %	60 %	40 %	A	D	N	0,433	N	0,61	N	0,73	N	0,73	N	1,49	D
		1	D	N	N	D	D	A	A	A	N		N	A	N	20 %	60 %	40 %	A	D	N	0,804	N	1,07	N	1,11	N	1,11	N	1,59	N
		4	N	A	N	D	N	N	N	A	N		A	N	N	10 %	40 %	60 %	N	D	N	0,573	N	0,89	N	0,89	N	0,89	N	0,97	N
		7	N	A	N	D	D	N	N	A	N		A	A	N	20 %	60 %	40 %	A	D	N	0,667	N	1,00	N	1,03	N	1,03	N	1,03	N
		10	D	A	D	D	N	A	N	A	N		N	N	N	20 %	50 %	50 %	A	D	N	0,688	N	1,00	N	1,03	N	1,03	N	1,05	D
		13	D	N	N	D	A	N	A	A	N		A	N	N	10 %	50 %	50 %	A	D	N	0,653	N	0,89	N	0,92	N	0,92	N	0,93	N
		16	A	N	A	D	D	A	N	A	N		A	N	N	20 %	60 %	40 %	A	D	N	0,498	N	0,71	N	0,75	N	0,75	N	0,78	D

Female chorus, reverberant

																					Reflectogram										
	Scat. Coeff.	Receiver	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	AL	D	D+A	N	50 %	10 %	1st order	D&K value	2nd order	D&K value	3rd order	D&K value	4th order	D&K value	Wave-files	D&K value	New criterion
Backwall	10	17	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	N	0,84							N	1,07	D
		1	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	N	1,24							N	1,77	D
		7	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	N	1,71							D	2,33	D
		9	D	N	D	D	N			D	A	A	D	N		50 %	70 %	30 %	D	D	N	1,66							D	2,20	D
		11	D	N	N	D	N			N	N	A	A	N		20 %	40 %	60 %	N	D	N	1,44							D	1,80	D
		13	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	1,08							N	1,22	N
	70	17	D	D	D	D	D			D	D	D	D	A		90 %	100 %	0 %	D	D	N	0,49							D	2,28	D
		1	D	D	D	D	D			D	D	D	D	D		100 %	100 %	0 %	D	D	N	0,75							D	3,27	D
		7	D	D	D	D	A			D	D	D	D	D		90 %	100 %	0 %	D	D	N	1,09							D	3,55	D
		9	D	D	A	D	A			D	D	A	A	N		50 %	90 %	10 %	D	D	N	1,08							D	3,21	D
		11	D	D	A	D	A			A	A	A	A	D		40 %	100 %	0 %	A	D	N	0,96							D	2,70	D
		13	D	N	A	A	N			N	N	N	A	N		10 %	40 %	60 %	N	D	N	0,74							D	1,98	D
		15	N	N	N	N	N			N	N	N	A	N		0 %	10 %	90 %	N	A	N	0,38							N	1,01	N
Reverberation	10	17	D	D	D	D	D			D	A	D	D	A		80 %	100 %	0 %	D	D	N	0,43	N	1,02	N	1,036	N	1,02	N	1,00	D
		1	N	N	D	A	A			A	N	A	A	N		10 %	60 %	40 %	A	D	N	0,776	N	1,23	N	1,294	N	1,368	N	0,97	D
		7	N	A	N	N	N			A	A	N	A	N		0 %	40 %	60 %	N	A	N	0,721	N	0,90	N	0,93	N	0,93	N	0,75	N
		10	D	D	D	D	D			D	D	D	D	A		90 %	100 %	0 %	D	D	N	0,638	N	0,95	N	0,979	N	0,979	N	0,81	D
		11	D	D	D	D	A			D	D	D	D	A		80 %	100 %	0 %	D	D	N	0,631	N	0,94	N	0,97	N	0,97	N	0,94	D
		13	D	D	D	D	A			D	D	D	D	A		80 %	100 %	0 %	D	D	N	0,619	N	0,87	N	0,891	N	0,891	N	0,88	D
		16	D	D	D	D	D			D	D	D	A	D		90 %	100 %	0 %	D	D	N	0,416	N	0,64	N	0,712	N	0,712	N	0,75	D
	70	17	D	N	D	D	D			D	N	A	D	A		60 %	80 %	20 %	D	D	N	0,433	N	0,61	N	0,73	N	0,73	N	1,49	D
		1	A	N	D	N	A			N	N	N	A	N		10 %	40 %	60 %	N	D	N	0,804	N	1,07	N	1,11	N	1,11	N	1,59	N
		7	N	N	N	A	N			N	N	N	N	A		0 %	20 %	80 %	N	A	N	0,667	N	1,00	N	1,03	N	1,03	N	1,03	N
		10	D	D	D	D	A			D	D	A	D	A		70 %	100 %	0 %	D	D	N	0,688	N	1,00	N	1,03	N	1,03	N	1,05	D
		11	A	N	A	N	N			A	N	A	N	N		0 %	40 %	60 %	N	A	N	0,678	N	0,99	N	1,02	N	1,02	N	0,99	N
		13	N	N	N	N	N			N	N	N	N	N		0 %	0 %	100 %	N	N	N	0,653	N	0,89	N	0,92	N	0,92	N	0,93	N
		16	A	D	D	D	D			D	A	D	A	D		70 %	100 %	0 %	D	D	N	0,498	N	0,71	N	0,75	N	0,75	N	0,78	D

APPENDIX E – CORRELATION ANALYSIS OF LISTENING TEST RESULTS

A correlation analysis was performed to validate the listening test results. Test results are discussed in Section 6.1. Results are reproduced in the following order:

- Speech, anechoic
- Speech, reverberant
- Cello, anechoic

- Trumpet, anechoic
- Trumpet, reverberant
- Guitar, anechoic

- Orchestra, anechoic
- Orchestra, reverberant

- Female chorus, anechoic
- Female chorus, reverberant

Results from single echo situations where echo reflection are generated through a single reflection from the back wall of the concert hall are included in the calculations of the anechoic sections, which deviates from the division used in previous appendix. This is done to address correlation coefficients of anechoic and reverberant situations separately.

Speech, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS													
JPS			0,73	0,79	0,46	0,68	0,59	0,45	0,79	0,69	0,71		0,66
PN		0,73		0,59	0,67	0,71	0,77	0,66	0,78	0,70	0,55		0,68
TG		0,79	0,59		0,21	0,53	0,50	0,28	0,71	0,62	0,83		0,56
AS		0,46	0,67	0,21		0,62	0,75	0,72	0,57	0,48	0,21		0,52
PS		0,68	0,71	0,53	0,62		0,69	0,62	0,72	0,65	0,45		0,63
UK		0,59	0,77	0,50	0,75	0,69		0,71	0,70	0,54	0,41		0,63
JT		0,45	0,66	0,28	0,72	0,62	0,71		0,53	0,56	0,25		0,53
GE		0,79	0,78	0,71	0,57	0,72	0,70	0,53		0,66	0,68		0,68
AB		0,69	0,70	0,62	0,48	0,65	0,54	0,56	0,66		0,61		0,61
HF		0,71	0,55	0,83	0,21	0,45	0,41	0,25	0,68	0,61			0,52
OAE													
Average:		**0,86**	**0,89**	**0,76**	**0,70**	**0,83**	**0,83**	**0,71**	**0,90**	**0,81**	**0,72**		$\boxed{\textbf{0,80}}$

Speech, reverberant

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS													
JPS			0,85	0,60	0,82	0,74	0,85	0,60	0,80	0,65	0,81		0,75
PN		0,85		0,75	0,91	0,90	0,79	0,75	0,88	0,79	0,86		0,83
TG		0,60	0,75		0,91	0,75	0,75	1,00	0,70	0,77	0,75		0,78
AS		0,82	0,91	0,91		0,80	0,91	0,91	0,82	0,75	0,91		0,86
PS		0,74	0,90	0,75	0,80		0,69	0,75	0,88	0,91	0,72		0,79
UK		0,85	0,79	0,75	0,91	0,69		0,75	0,75	0,66	0,86		0,78
JT		0,60	0,75	1,00	0,91	0,75	0,75		0,70	0,77	0,75		0,78
GE		0,80	0,88	0,70	0,82	0,88	0,75	0,70		0,79	0,77		0,79
AB		0,65	0,79	0,77	0,75	0,91	0,66	0,77	0,79		0,69		0,75
HF		0,81	0,86	0,75	0,91	0,72	0,86	0,75	0,77	0,69			0,79
OAE													
Average:		**0,88**	**0,95**	**0,86**	**0,96**	**0,91**	**0,90**	**0,86**	**0,91**	**0,86**	**0,90**		$\boxed{\textbf{0,90}}$

Cello, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,53	0,58	0,44	0,69			0,50	0,48	0,54	0,42	0,40	0,51
JPS	0,53		0,46	0,47	0,49			0,70	0,57	0,70	0,57	0,55	0,56
PN	0,58	0,46		0,52	0,68			0,51	0,61	0,52	0,45	0,45	0,53
TG	0,44	0,47	0,52		0,37			0,62	0,63	0,64	0,49	0,59	0,53
AS	0,69	0,49	0,68	0,37				0,43	0,47	0,41	0,42	0,43	0,49
PS													
UK													
JT	0,50	0,70	0,51	0,62	0,43				0,58	0,76	0,57	0,57	0,58
GE	0,48	0,57	0,61	0,63	0,47			0,58		0,57	0,45	0,54	0,55
AB	0,54	0,70	0,52	0,64	0,41			0,76	0,57		0,65	0,61	0,60
HF	0,42	0,57	0,45	0,49	0,42			0,57	0,45	0,65		0,50	0,48
OAE	0,40	0,55	0,45	0,59	0,43			0,57	0,54	0,61	0,50		0,52
Average:	**0,67**	**0,73**	**0,78**	**0,78**	**0,70**			**0,82**	**0,77**	**0,83**	**0,73**	**0,75**	$\boxed{\textbf{0,76}}$

Trumpet, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS													
JPS			0,73	0,64	0,48	0,68	0,53	0,46	0,78		0,62	0,67	0,62
PN		0,73		0,59	0,35	0,62	0,42	0,43	0,64		0,53	0,57	0,54
TG		0,64	0,59		0,29	0,64	0,34	0,26	0,73		0,53	0,58	0,51
AS		0,48	0,35	0,29		0,40	0,56	0,39	0,25		0,51	0,51	0,41
PS		0,68	0,62	0,64	0,40		0,58	0,44	0,68		0,54	0,59	0,57
UK		0,53	0,42	0,34	0,56	0,58		0,50	0,38		0,48	0,51	0,48
JT		0,46	0,43	0,26	0,39	0,44	0,50		0,31		0,46	0,46	0,41
GE		0,78	0,64	0,73	0,25	0,68	0,38	0,31			0,58	0,60	0,55
AB													
HF		0,62	0,53	0,53	0,51	0,54	0,48	0,46	0,58			0,58	0,54
OAE		0,67	0,57	0,58	0,51	0,59	0,51	0,46	0,60		0,58		0,56
Average:		**0,88**	**0,79**	**0,75**	**0,63**	**0,82**	**0,70**	**0,59**	**0,80**		**0,77**	**0,80**	**0,75**

Trumpet, reverberant

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		-0,31	0,34		0,26	0,45	0,65	0,42	0,54		0,20	0,11	0,29
JPS	-0,31		0,36		-0,50	-0,04	-0,22	0,21	0,33		0,10	0,17	0,01
PN	0,34	0,36			0,01	0,44	0,09	0,49	0,61		0,32	0,19	0,32
TG													
AS	0,26	-0,50	0,01			0,36	0,21	0,38	0,21		0,37	0,55	0,20
PS	0,45	-0,04	0,44		0,36		0,43	0,79	0,59		0,06	0,70	0,42
UK	0,65	-0,22	0,09		0,21	0,43		0,41	0,56		0,07	0,36	0,29
JT	0,42	0,21	0,49		0,38	0,79	0,41		0,85		0,53	0,69	0,53
GE	0,54	0,33	0,61		0,21	0,59	0,56	0,85			0,55	0,51	0,53
AB													
HF	0,20	0,10	0,32		0,37	0,06	0,07	0,53	0,55			0,15	0,26
OAE	0,11	0,17	0,19		0,55	0,70	0,36	0,69	0,51		0,15		0,38
Average:	**0,45**	**0,28**	**0,65**		**0,45**	**0,75**	**0,47**	**0,92**	**0,91**		**0,54**	**0,75**	**0,61**

Guitar, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,67	0,21	-0,05	0,68			0,79	0,55	0,71	0,72	0,47	0,53
JPS	0,67		0,02	0,01	0,69			0,76	0,73	0,76	0,54	0,75	0,55
PN	0,21	0,02		-0,12	0,37			0,17	0,07	0,08	0,38	0,01	0,13
TG	-0,05	0,01	-0,12		-0,22			0,01	-0,14	-0,01	-0,21	-0,12	-0,09
AS	0,68	0,69	0,37	-0,22				0,75	0,54	0,65	0,61	0,66	0,52
PS													
UK													
JT	0,79	0,76	0,17	0,01	0,75				0,63	0,83	0,64	0,68	0,59
GE	0,55	0,73	0,07	-0,14	0,54			0,63		0,61	0,42	0,65	0,45
AB	0,71	0,76	0,08	-0,01	0,65			0,83	0,61		0,68	0,63	0,55
HF	0,72	0,54	0,38	-0,21	0,61			0,64	0,42	0,68		0,45	0,45
OAE	0,47	0,75	0,01	-0,12	0,66			0,68	0,65	0,63	0,45		0,46
Average:	**0,84**	**0,86**	**0,31**	**-0,12**	**0,85**			**0,91**	**0,75**	**0,87**	**0,79**	**0,76**	**0,68**

Orchestra, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,49	0,39	0,61	0,61			0,68	0,37	0,71	0,79	0,58	0,58
JPS	0,49		0,63	0,55	0,52			0,77	0,46	0,74	0,66	0,57	0,60
PN	0,39	0,63		0,56	0,25			0,66	0,51	0,58	0,63	0,53	0,53
TG	0,61	0,55	0,56		0,38			0,72	0,42	0,64	0,68	0,36	0,55
AS	0,61	0,52	0,25	0,38				0,59	0,68	0,54	0,56	0,65	0,53
PS													
UK													
JT	0,68	0,77	0,66	0,72	0,59				0,56	0,82	0,81	0,76	0,71
GE	0,37	0,46	0,51	0,42	0,68			0,56		0,44	0,46	0,59	0,50
AB	0,71	0,74	0,58	0,64	0,54			0,82	0,44		0,75	0,74	0,66
HF	0,79	0,66	0,63	0,68	0,56			0,81	0,46	0,75		0,67	0,66
OAE	0,58	0,57	0,53	0,36	0,65			0,76	0,59	0,74	0,67		0,60
Average:	**0,78**	**0,81**	**0,72**	**0,76**	**0,71**			**0,93**	**0,66**	**0,88**	**0,89**	**0,79**	$\boxed{\textbf{0,79}}$

Orchestra, reverberant

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,34	0,65	0,27	0,14			0,71	-0,43	0,10	-0,07	-0,43	0,14
JPS	0,34		0,52	-0,21	0,57			0,11	-0,34	-0,23	-0,49	-0,34	-0,01
PN	0,65	0,52		0,00	-0,22			0,65	-0,22	-0,15	0,10	-0,22	0,13
TG	0,27	-0,21	0,00		-0,27			-0,27	-0,27	-0,55	0,13	-0,27	-0,16
AS	0,14	0,57	-0,22	-0,27				-0,14	-0,14	0,29	-0,75	-0,14	-0,07
PS													
UK													
JT	0,71	0,11	0,65	-0,27	-0,14				-0,14	0,29	0,34	-0,14	0,16
GE	-0,43	-0,34	-0,22	-0,27	-0,14			-0,14		0,29	-0,20	1,00	-0,05
AB	0,10	-0,23	-0,15	-0,55	0,29			0,29	0,29		-0,32	0,29	0,00
HF	-0,07	-0,49	0,10	0,13	-0,75			0,34	-0,20	-0,32		-0,20	-0,14
OAE	-0,43	-0,34	-0,22	-0,27	-0,14			-0,14	1,00	0,29	-0,20		-0,05
Average:	**0,94**	**0,50**	**0,80**	**0,20**	**0,10**			**0,73**	**-0,52**	**-0,07**	**0,05**	**-0,52**	$\boxed{\textbf{0,22}}$

Female chorus, anechoic

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,54	0,68	0,93	0,49			0,61	0,59	0,66	0,65	0,44	0,62
JPS	0,54		0,68	0,66	0,90			0,79	0,92	0,82	0,58	0,82	0,74
PN	0,68	0,68		0,70	0,75			0,90	0,85	0,83	0,91	0,68	0,78
TG	0,93	0,66	0,70		0,59			0,75	0,72	0,80	0,71	0,54	0,71
AS	0,49	0,90	0,75	0,59				0,77	0,89	0,88	0,69	0,82	0,75
PS													
UK													
JT	0,61	0,79	0,90	0,75	0,77				0,96	0,85	0,82	0,62	0,79
GE	0,59	0,92	0,85	0,72	0,89			0,96		0,90	0,76	0,72	0,81
AB	0,66	0,82	0,83	0,80	0,88			0,85	0,90		0,85	0,80	0,82
HF	0,65	0,58	0,91	0,71	0,69			0,82	0,76	0,85		0,63	0,72
OAE	0,44	0,82	0,68	0,54	0,82			0,62	0,72	0,80	0,63		0,68
Snitt	**0,73**	**0,89**	**0,90**	**0,83**	**0,90**			**0,92**	**0,95**	**0,96**	**0,85**	**0,82**	**0,88**

Female chorus, reverberant

	BS	JPS	PN	TG	AS	PS	UK	JT	GE	AB	HF	OAE	
BS		0,67	0,76	0,77	0,71			0,81	0,68	0,75	0,79	0,57	0,72
JPS	0,67		0,50	0,74	0,55			0,79	0,93	0,79	0,65	0,69	0,70
PN	0,76	0,50		0,69	0,81			0,69	0,44	0,73	0,73	0,45	0,64
TG	0,77	0,74	0,69		0,78			0,86	0,67	0,82	0,75	0,83	0,77
AS	0,71	0,55	0,81	0,78				0,73	0,41	0,74	0,67	0,69	0,68
PS													
UK													
JT	0,81	0,79	0,69	0,86	0,73				0,73	0,88	0,79	0,66	0,77
GE	0,68	0,93	0,44	0,67	0,41			0,73		0,69	0,65	0,57	0,64
AB	0,75	0,79	0,73	0,82	0,74			0,88	0,69		0,62	0,69	0,75
HF	0,79	0,65	0,73	0,75	0,67			0,79	0,65	0,62		0,36	0,67
OAE	0,57	0,69	0,45	0,83	0,69			0,66	0,57	0,69	0,36		0,61
Snitt	**0,88**	**0,86**	**0,80**	**0,92**	**0,83**			**0,93**	**0,80**	**0,91**	**0,83**	**0,76**	**0,85**

APPENDIX F – CD

See the included *readme.txt* file on the CD for information about its content.