INTERNSHIP PEUTZ

Signal processing in experimental room acoustics

Bas Binnerts

University of Twente

2011-12-13

SUMMARY

This section gives a short summary of the projects of Peutz that I contributed to during my internship. As the title of this report *(Signal processing in experimental room acoustics)* implies, was the main assignment to optimize and validate three Matlab codes, used for processing measurement data. Short summaries, describing the operation of these programs are given below.

- **Diff_polar_5gr:** is a program used for the analysis of reflection measurements in the anechoic chamber. The program processes impulse response measurements into polar figures and organized text files.
- **Raci(dist):** is a code used for processing "non-scale model" room acoustic impulse response measurements. After filtering and averaging is applied, the required acoustic parameters are extracted from the signal.
- **WINMLS:** is a code similar to the Raci code, but in this case applicable to scale model measurements. Also the output format and code structure is different.

The internship started with the task to determine the acoustic transparency of a mesh. The architect of the "Staatsoper Unter den Linden" designed this mesh to visually conceal an added volume to the room, serving to increase the reverberation time. With two different experimental setups it has been shown that the transparency of the mesh was not sufficient for this purpose.

The second project involved a scale model research of the "Bühnen Köln" opera. Various structural changes were made to the scale model, while studying the changes of the acoustic field by means of impulse response measurements. However, the most time was spent to the optimisation of the WINMLS and Raci code. The time needed to process all the data of a measurement session has been reduced from more than an hour to less than 15 minutes. Extra features have been added to the programs, including conversion of the acoustic parameters to an Excel file. A detailed summary, describing all the major changes to the codes, can be found in the chapter "Matlab codes scale model analysis".

During the last weeks of my internship a large amount of measurements, concerning a scale model research of a new reverberation chamber design, was efficiently processed.

Trainee
Name: Bas Binnerts
Student number: s0143669

Company
Name: Peutz
Address: Lindenlaan 41, 6584 AC Molenhoek, Netherlands
Phone: 024-3570707
Trainee period: 1st August until the 4th of November 2011

Mentors
Margriet Lautenbach - m.lautenbach@zoetermeer.peutz.nl
Martijn Vercammen - m.vercammen@mook.peutz.nl

University of Twente
Faculty: Applied mechanics
Research group: Structural Dynamics and Acoustics
UT supervisor: prof.dr.ir. de Boer

FIGURE 1: MY OFFICE AT PEUTZ

INTRODUCTION

The first contact with Peutz was made after recommendation by Ysbrand Wijnant. I was looking for an internship in the field of acoustics, and Peutz seemed an interesting company. My application for a position at Peutz as a trainee was successful, and some of the possible assignments were discussed. At that time, Peutz was involved in scale model research of the Opernhaus in Köln, and the optimization of a new reverberation chamber design with volume diffusers instead of the panel diffusers. An exact assignment was not yet established, but there were enough learning possibilities to make my choice for an internship at Peutz definitive. My expectations of the internship were that I would learn a lot about experimental room acoustics and signal processing. Also I wanted to learn what acoustic consultancy work involved.

COMPANY PROFILE [1]

The company profile of Peutz has been complemented, serving as a short introduction to the company.

During the fifty-odd years since its founding Peutz has executed more than 10.000 commissions, from comparatively small up to very large projects, like "Schiphol Airport" (Amsterdam). During this period the company and its employees have specialized in a number of related fields. A major philosophy of Peutz is the integration of experimental and state of the art computational facilities in the consulting practice. Hence the company has had its own laboratories since its founding. The main fields of expertise are given in Figure 2*. The Peutz group now includes offices in the Netherlands, the United Kingdom, Germany, Belgium, France and Spain with about 200 staff in total.* Figure 3 *shows the locations of the various offices of Peutz.*
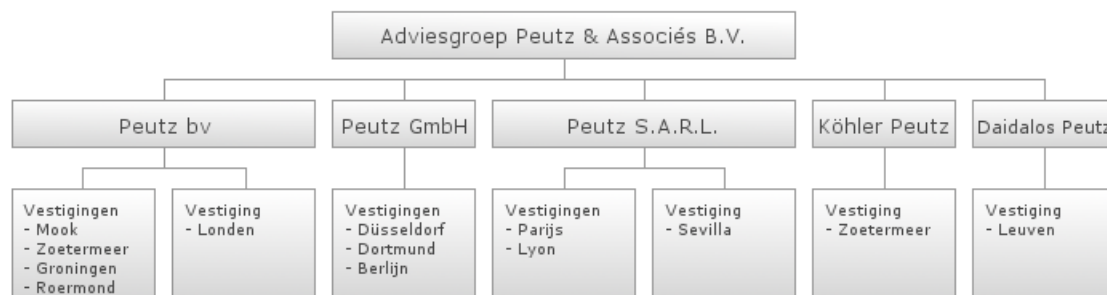

FIGURE 2: OFFICES OF PEUTZ

ROOM ACOUSTICS[1]

The internship took place within the room-acoustics department of Peutz, of which its pursuits are best described by: *"The acoustical aspects of building design, construction and installation"*

Two essential aspects of the acoustics of a room are:
- A good musical sound quality.
- Good intelligibility.

But also the room must be:
- Pleasant to perform in.
- Suitable for the coherence of the musicians.
- A good balance between orchestra and singers.
- Suitable for films and electronically amplified music or speech.
- Suitable for making recordings or for listening to recordings.

All these different aspects of a good acoustics; make the assessment of a room a complicated process. Although computers have a lot of computing power these days, they can still not solve a lot of the problems encountered in acoustics. Much of the evaluation is therefore still based on the experience of the acoustician.

FIGURE 3: DIFFERENT DEPARTMENTS OF PEUTZ[1]

As the title of this report *(Signal processing in experimental room acoustics)* implies, was the main assignment to optimize and validate three Matlab codes, used for processing measurement data. Short summaries, describing the operation of these programs are given below.

- **Diff_polar_5gr:** is a program used for the analysis of reflection measurements in the anechoic chamber. The program processes impulse response measurements into polar figures and organized text files.
- **Raci(dist):** is a code used for processing "non-scale model" room acoustic impulse response measurements. After filtering and averaging is applied, the required acoustic parameters are extracted from the signal.
- **WINMLS:** is a code similar to the Raci code, but in this case applicable to scale model measurements. Also the output format and code structure is different.

Besides programming I assisted with the scale models research of the opera in Köln and with diffusion measurements done in the anechoic chamber. The last weeks of internship it was asked to efficiently process a large amount of measurements data into a specific excel file format.

STRUCTURE REPORT

This report attempts to give a complete overview of all the themes encountered during the internship. The theory chapter discusses the theory behind important aspects of experimental room acoustics and signal processing.

The projects chapter focuses on the projects involved in the internship, concentrating on the research methodology, the measurement setup and the results. The first project involved the determination of the acoustic transparency and absorption of a mesh. Measurements were done and Matlab codes were written to process the large amount of measurement data into polar figures. The second project concerned scale model research done for the renovation of the Opernhaus in Köln. As the final analysis of the data was done by Martijn Vercammen and Margriet Lautenbach, this report does not cover the final advice for the renovation given by Peutz. The "Matlab codes scale model analysis" section deals with the changes of the revised Matlab codes. The last discussed project involves the processing of (scale model) reverberation room measurement data. The report finishes with a reflection on the internship.

INTRODUCTION TO EXPERIMENTAL ROOM ACOUSTICS[2]

It is generally known that a concert hall, theatre or a lecture room can have good or poor acoustics. To be able to evaluate the acoustic quality of rooms, many different factors of influence have been researched. However, the present level of reliable knowledge in room acoustics is not particularly advanced and many of these important factors are understood only incompletely, or not at all. This is mainly due to the complexity of a sound field in closed spaces. Room acoustics therefore is sometimes compared with the design of a musical instrument, for that it is often considered more an art then an exact science. On the other hand, it should be noted that the research done in the field of acoustic has brought some indispensable tools/methods for room acoustic evaluation.

A main task of the acoustic consultant is to examine the "objective" acoustical properties of a design or room by means of:
- Calculation
- (Scale) model investigation or measurements at location.
- Computational simulation (e.g. a ray trace models).

With the obtained information a proposal has to be submitted with suitable adjustments that will improve the acoustic quality of a room.

A challenge for the acoustician is that he is often faced with a two-fold problem. He has to apply the relation between the structural futures of a room (such as shape and materials) with the objective sound field that occurs in this space. On the other hand he has to consider the subjective part of acoustic that involves the interrelation between objective and measurable/audible sound field parameters. The first problem lies within the realm of technical reasoning. It is the second problem however that makes room acoustic different from other technical disciplines. The success or failure of a room depends on the collective judgement of users. Measuring techniques must therefore take into account the hearing response of the listener and the consumer listening capabilities. Also it should be mentioned that the sound field in a real room is so complicated that it is not open to exact mathematical treatment. Because the evaluation of the sound field is so involved, simplifications are inevitable and the total of sound field data is reduced to averages. The problem is that these simplifications and averages have to be done in accordance with the properties of human hearing such that the results resemble the subjective sensations of the users as closely as possible. This is studied in the field of psychological acoustics. As long as the relations between physiological, psychological processes in hearing and the measured/calculated acoustic field are not completely understood, the relevant relations between the objective and subjective must be investigated empirically.

In the following section of the report, some important aspects of room acoustics are discussed. The theory starts with an introduction to measurement techniques in experimental room acoustics. As the impulse response plays a prominent role, the corresponding theory is discussed extensively. Also the techniques for obtaining the impulse response are treated; with special attention given to scale modelling of the acoustic field. Finally the methods for evaluating of some of the most important room acoustic parameters are given.

A common task in modern acoustics is to define and evaluate sound field parameters that are the objective counterpart of the subjective acoustic impression of the listener. These parameters give an interrelation between the subjective world and architecture. It is the objective of this chapter to describe these quantities, how they are can be measured and what kind of equipment is required. Also the theory of impulse responses and excitation signals is dealt with.

INSTRUMENTATION

The introduction of the digital computer triggered a revolution in the field of measuring techniques. All the different kind of signal processing (Filtering, storing and evaluation of measured signals) can now be done with the help of digital equipment.

Gun shots, bursting balloons, explosives of other methods that yield a large excitation of the acoustic field in a room can be used to obtain the impulse response from the sound source to a microphone, which is the main source of information in experimental room acoustics. A disadvantage of these types of excitation signals is that they are not (exactly) reproducible. Special excitation signals can be used using loudspeakers that are reproducible (e.g. sine sweeps or MLS signals). The different types of excitation signals are discussed in more detail in the "excitation signals" section. For **loudspeakers** it is important that their transfer function has a flat spectrum for the relevant frequencies and that the excitation is as omnidirectional as possible. If the source is not omnidirectional, the measurement results will not only depend on the location, but also on the orientation of the source. Two examples of omnidirectional loudspeaker designs are described below:

Dodecahedron (12 equal areas) or icosahedrons (20 equal areas): Figure 4 shows an example of a dodecahedron speaker design. These sources have strong omnidirectional characteristics. For elevated frequencies, however, the directionality is becoming more present. Sometimes two different size speakers are used to avoid this problem. Another issue is that the acoustic centre of the speakers is frequency dependent. For higher frequencies the assumption that the speaker behaves as a points source, located in the geometrical centre becomes questionable.

Reversed horn loudspeaker: Peutz designed a special inversed horn shaped tube that is put on speakers to model a point source with omnidirectional properties. The end of the tube behaves as a point source for wavelengths that are larger than the diameter. Resonances of the tube are suppressed by using damping material. Figure 5 shows an example

FIGURE 4: DODECAHEDRON LOADSPEAKER
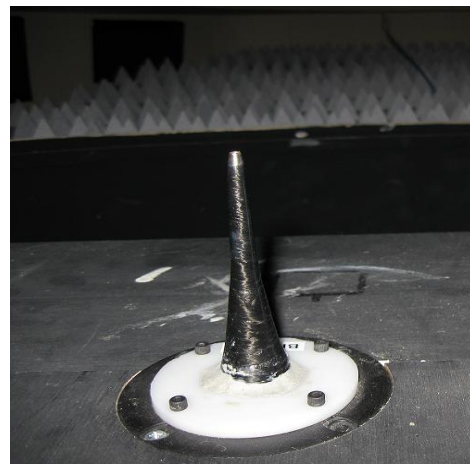
FIGURE 5: PEUTZ OMNIDIRECTIONAL SPEAKER DESIGN

The next step is to pick up the sound produced by the source in the room. Pressure sensitive **microphones** with omnidirectional characteristics are most commonly used. For certain type of measurements, however, a gradient receiver is needed. An example is a dummy head microphone that takes into account the effect of the human head on the perceived sound. The microphones are calibrated relative to each other at an important frequency (often 1Khz (with a 1:1 scale)). This is acceptable because most acoustic parameters do not depend on the absolute sound pressure. A flat frequency spectrum of the microphones is desirable however since the calibration only takes place at one frequency. If no flat frequency response microphone is available, the measured signal can be corrected by a filter (amplifying certain frequency more than others). Figure 6 shows the omnidirectional microphones used by Peutz in scale models.

The measured output signal can be saved, preferably by a digital recorder. It is the most convenient if the signal is directly saved into a portable computer that can process the data into the relevant acoustic parameters. This does require however that the computer is equipped with a suitable analog-to-digital (AD) convertor.
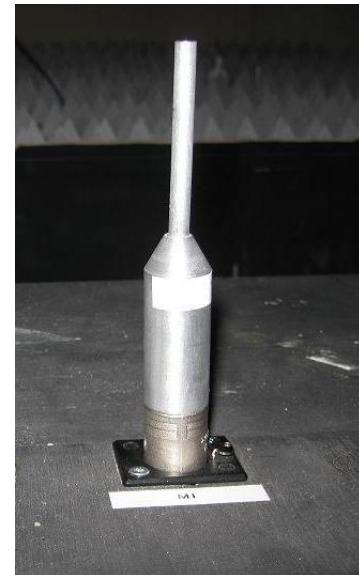
FIGURE 6: SCALE MODEL OMNIDIRECTIONAL MICROPHONES

IMPULS RESPONSE MEASUREMENTS[2]

According to system theory, all properties of a linear transmission system (Figure 7) are contained in its impulse response g(t) or transfer function (i.e. the Fourier transform of an impulse response). A system is considered linear when a multiplication of the input results in a multiplication of the output by the same factor. Because a room can be considered as an acoustical transmission function, the impulse response yields a complete description of the changes to a sound signal travelling from the source (input) to the receiver (output). Almost all of the acoustic parameters can therefore be derived from the impulse response, making its determination a fundamental task in (experimental) room acoustics. The impulse response is defined as:

"The signal that is obtained at the systems output after its excitation of a vanishingly short impulse (yet with non-vanishing energy), i.e. by a Dirac delta pulse δ(t)"
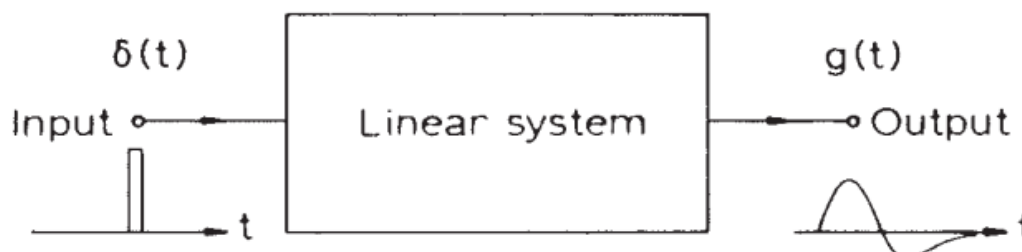
FIGURE 7: THE IMPULSE RESPONSE OF A LINEAR SYSTEM

The Dirac delta pulse has the property that its amplitude spectrum is constant for all frequencies Figure 8 shows the Fourier Transformation of a Dirac Delta pulse.
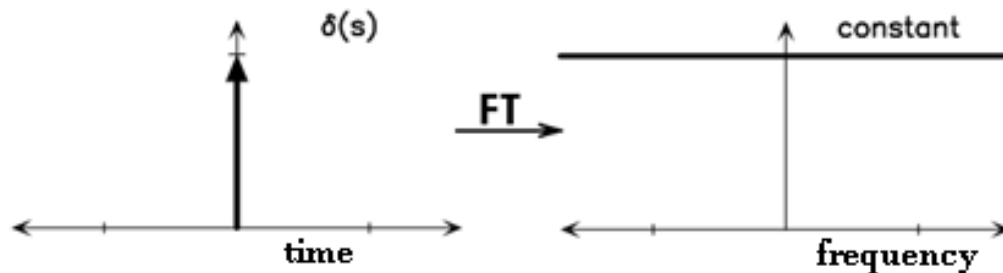
FIGURE 8: THE FOURIER TRANSFORM (FT) OF THE DIRACT DELTA PULSE

The delta function has the following representation:

$$\delta(t) = \lim_{f_0 \to \infty} \frac{1}{f_0} \int_{-\frac{f_0}{2}}^{\frac{f_0}{2}} exp(2\pi f t)df \;\to\; \mathcal{F}\big(\delta(t)\big) = S(f) = 1$$

And the fundamental properties that:

$$s(t) = \int_{-\infty}^{\infty} s(\tau)\delta(t-\tau)d\tau$$

Where s(t) can be any function of time. For s(t)=1 it follows that:

$$\int_{-\infty}^{\infty} \delta(\tau)d\tau = 1$$

Since the delta function is zero for all t≠0, it follows from the last property that the value of the delta function must be infinite at t=0. Since for practical purposes only frequencies below around 10 kHz are relevant, the Dirac delta pulse can be approximated by a short pulse of arbitrary shape, with a duration which is smaller than 20 milliseconds.

CONVOLUTION AND CROSS CORRELATION[2, 3]

Consider an impulse response g(t) measured using a the Dirac function δ(t) as excitation signal. The impulse response g(t) vanishes for t<0 because the response cannot precede its excitation. Now consider an input signal s(t) applied to the same acoustical transmission system. The measured impulse response g(t) can now be used to obtain the output signal s'(t). This is done using the convolution operation. If the impulse response of a room g(t) is known, the output signal $s'(t)$ with respect to any input signal s(t) can be obtained as:

$$s'(t) = \int_{-\infty}^{\infty} g(\tau)s(t-\tau)d\tau$$

A convolution is an integral that expresses the amount of overlap as a **time reversed** function is shifted over another. The time reversion is needed to ensure that the (time) start of the sliding signal first arrives at the start of the impulse response (while it slides from minus to plus infinity). The shorthand notation is:

$$s'(t) = s * g = \int_{-\infty}^{\infty} s(t-\tau)\,g(\tau)d\tau = \int_{-\infty}^{\infty} s(\tau)\,g(t-\tau)d\tau$$
$$where:\ s * g = g * s$$

Figure 9 graphically illustrates the convolution of two boxcar function f and g.
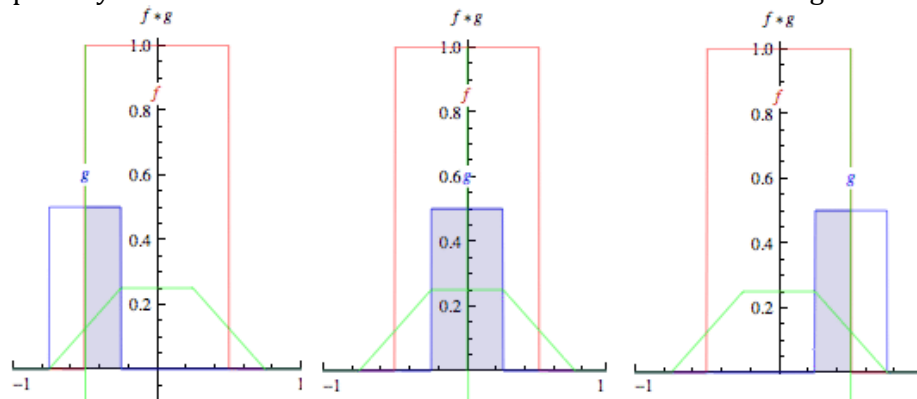
The **green** curve shows the convolution of the **blue** and **red** curves as a function of the time t on the x-axis (for which the position is indicated by the vertical green line). The filled area of the box function represents the product *f(τ)g(t-τ)* as a function of t, and thus its area (represented by the green line) precisely equals the convolution $\int_{-\infty}^{\infty} s(\tau)g(t-\tau)d\tau$.

Figure 10 shows an example that resembles the situation where a boxcar pulse g is applied to a room with an impulse response f.
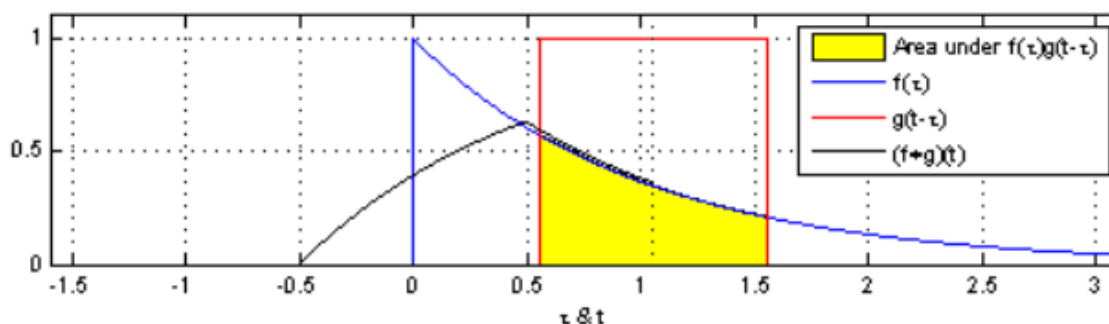
FIGURE 10: CONVOLUTION EXAMPLE

The **convolution theorem** states that the Fourier transform of the convolution of two functions is the product of their individual Fourier transforms.

$$s'(t) = s * g \Leftrightarrow S'(f) = S(f) \cdot G(f)$$

The complex spectrum of the measured impulse response is also known as the transfer function of the system. If a harmonic signal with frequency f is applied to a transmission system, its amplitude will be changed by the factor |G(f) | and its phase will be shifted by the phase angle of G(f).

In linear acoustics the convolution of the input signal at the source and the impulse response of the room, resembles the echo heard at the output (where the impulse response was measured). Another application of the convolution operation is adding artificial reverberation. The impulse response of a real room is mapped onto a digital audio signal to add the acoustic of the room to the sound. In room acoustics this technique used to create an audible impression of how a room will sound like, using the impulse responses measured in a scale model. This technique is discussed later in the auralisation section.

The inverse process of convolution is called **de-convolution**. The process is used for retrieving the impulse response of a room from a given input and measured output. This operation is done in the frequency domain, because it then only involves a division of the power spectra:

$$\boxed{S'(f) = S(f) \cdot G(f)} \rightarrow \boxed{\frac{S'(f)}{S(f)} = G(f)}$$

The transfer function can now be obtained by inverse Fourier transformation. Figure 11 visualizes this process:
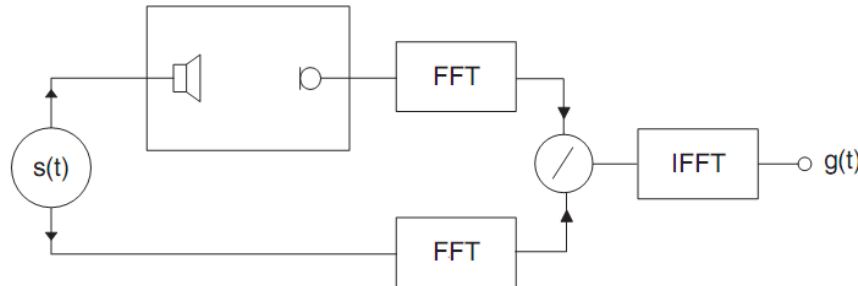
FIGURE 11: PRINCIPLE OF DUAL CHANNEL ANALYSIS
(FFT = FAST FOURIER TRANSFORM, IFFT=INVERSE FFT)

Note: If the signal s(t) is known beforehand, the spectrum S(t) must be calculated only one once.

Another practical application is the situation when you want to study the influence of amplifier on the signal for all frequencies. This can be done by sending a sine sweep to the input of the amplifier, measuring the output of the amplifier. Again division of the output spectra by the input spectra yields the transfer function G_{amp} of the amplifier. The linear distortion of the amplifier can be corrected by passing the test signal though a filter with the inverse of G_{amp} as the transfer function. This process is called Pre-emphasis and is often applied to microphones.

Discrete convolution comes into play when the involved signals are discrete measurements in time or space instead of continuous signals. The behaviour of a time-invariant discrete-time system with input signal x[n], impulse response h[n] and output signal y[n] is described by the convolution sum:

$$y[n] = h[n] * h[n] \equiv \sum_{k=-\infty}^{\infty} h[k] \cdot x[n-k]$$

Cross correlation is a very similar operation to convolution, with the difference that the second function is not time reversed. This leads to the following definition:

$$\boxed{s \star g \equiv \int_{-\infty}^{\infty} s(\tau)\, g(\tau - t) d\tau = \int_{-\infty}^{\infty} s(t - \tau)\, g(\tau) d\tau}$$
$$\text{But: } s \star g \neq g \star s$$

A cross correlation is used to study the similarity of two signals. The equation also has a useful form in the frequency domain, given by the cross correlation theorem:

$$\boxed{s(t) \star g(t) \Leftrightarrow \bar{S}(f) \cdot G(f)}$$

Autocorrelation is a special case of cross-correlation where a function is correlated with itself. The Wiener-Khinchin theorem states that:

$$\boxed{g(t) \star g(t) \Leftrightarrow \bar{G}(f) \cdot G(f) = |G(f)|^2}$$

This means that the Fourier transform of an autocorrelation function is the power spectrum, (or equivalently, the autocorrelation function is the inverse Fourier transform of the power spectrum). The autocorrelation and the theorem are often used to compute the power spectra.

Another important application of the autocorrelation function is to test the randomness (or pseudo randomness) of a signal.

The autocorrelation function is given by:

$$\Phi_{gg}(\tau) \equiv \int_{-\infty}^{\infty} g(t')\, g(t' + \tau)dt' = \int_{-\infty}^{\infty} g(-t')\, g(-t' + \tau)dt'$$

The second part of the equation resembles simply the convolution of the impulse response with its time reversed replica. The autocorrelation function can experimentally be determined by exciting a room with its time-inversed impulse response. Figure 12 gives an overview of the relation between a function, its Fourier transform, its autocorrelation function and its power spectrum.

x_j (function)	$\Leftrightarrow$ DFT	X_k (transform)
$\Downarrow$		$\Downarrow$
$x_j \star x_j$ (autocorrelation)	$\Leftrightarrow$ DFT	$\lvert X_k \rvert^2$ (power spectrum)

FIGURE 12: OVERVIEW

EXCITATION SIGNALS[2]

An important question in experimental room acoustics what type of excitation signal should be used. Signal types that are particularly useful are those with flat power spectra. As discussed earlier, their autocorrelation function is a Dirac impulse, (or a regular sequence of Dirac impulse if s(t) is periodic). The frequency spectrum of such signals is of the "all-pass type":

$$\boxed{S(f) = \exp[i\varphi(f)]} \rightarrow \boxed{S(f)^{-1} = \exp[-i\varphi(f)] = \bar{S}(f)}$$

The techniques for dual channel can now be replaced by technique called "matched filtering". Combing the found relation with $S'(f) = S(f) \cdot G(f)$ gives:

$$\boxed{G(f)\ = S'(f) \cdot \bar{S}(f)} \rightarrow \boxed{g(f)\ = s'(f) * s(-f)}\ \left(\text{because } \mathcal{F}^{-1}\big(\bar{S}(f)\big) = s(-f)\right)$$

Thus the impulse response can be obtained by passing the measured signal through a filter of which *the transfer function is the time-reversed excitation signal*. This corresponds to cross correlating the sampled measured response with the original sequence to obtain the system impulse response. Some requirements of the excitation signal are:
- The signal is not reverberant itself.
- The signal has a broad enough and flat frequency spectrum.
- The signal is loud enough to produce a peak sound pressure level at least 45 dB above the background noise. (if e.g. T_{30} is to be measured).

In this section, two important types of signals/methods are distinguished:

Interrupted noise/impulse method: By directly measuring the decay after the cut-off of a continuous sound source signal (for example recording the sound of a gunshot or noise with a level recorder). When using noise, it is important that the duration of excitation is at least longer the half of the reverberation time because this ensures a steady sound field in the room. Also it is important that the frequency spectrum is "flat" (i.e. constant for the frequency range of interest). Also it is important that the measurements are averaged to obtain an acceptable measurement uncertainty (because of the randomness in the source signal).

Uninterrupted noise method: can be used in combination with special signal types to obtain higher S/N ratios. The two most used signals in this category are sine sweeps and maximum length signals (MLS). This method works as the dual channel analysis which was described earlier.

MLS signals are a form of two-level pseudorandom sequences with very good S/N properties. The signal yields an impulse under circular autocorrelation, which results in a flat frequency spectrum. A big advantage of MLS sequences is that the impulse response can be retrieved from the measured signal very efficiently (no transformation to the frequency domain is needed). To avoid time-aliasing, it is important that the period of the sequence must exceed the length of the impulse response to be measured. A period of about 6 second should be sufficient for most cases. Even though the signal is deterministic, averaging can still be applied to reduce noise from external sources. Also the system/room must be time-invariant because 'random' deviations due to temperature- or velocity gradients are not reduced by averaging.

EXAMINATION AND PROCESSING OF THE IMPULSE RESPONSE[2]

As mentioned before, the impulse responses of a room (i.e. the transmission paths of a room) are the most characteristic future describing its acoustic. An experienced acoustician can already derive quite some information from the spectrogram (the graphical representation of the impulse response). Two important characteristics are given below.

- To what extend the direct sound peak is supported by shortly delayed reflections, and how these are distributed in time. (This is an indication for the amount of colouration (frequency distortion)) and the possibility of flutter echo's)
- The presence of strong isolated peaks with long delays. (hint the danger of echoes)

The examination of impulse responses can be made much easier by removing insignificant details. Rectifying and smoothening of the impulse response can be done by means of time integration. The main disadvantage of this method however is the introduction of a certain amount of arbitrariness into the impulse response. A large time integration constant can results in the loss of important information, while a too small value the smoothening might me insufficient. A well studied mathematical tool that offers a solution is the Hilbert transformation. This transformation of a signal determines its "envelope". Consider a signal $s(t)$, the envelope is then defined as:

$$e(t) = \sqrt{[s(t)]^2 + [\check{s}(t)]^2} \qquad \check{s}(t) = \frac{1}{\pi}\int_{-\infty}^{\infty}\frac{s(t-t')}{t'}dt'$$

Where the $\check{s}(t)$ is the Hilbert transform of $s(t)$. Different techniques can be used to effectively determine the Hilbert transform of an equation. Matlab contains an effective method for fast determination of the Hilbert transform. Figure 13 shows the results of a Hilbert transformation to an impulse response.
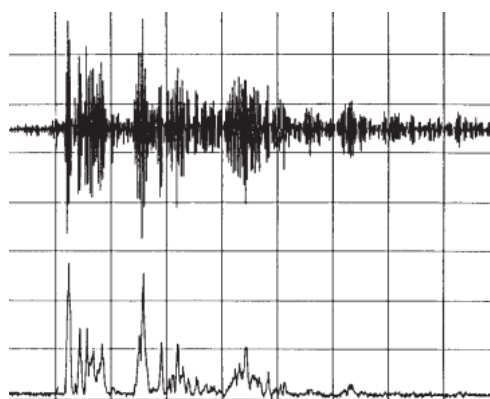

FIGURE 13: ROOM IMPULSE RESPONSE (UPPER) AND SQUARED ENVELOPE (LOWER PART)

The reflectogram can be further simplified to simulate the time integrating properties of our hearing. The obtained envelope $e(t)$ is then convolved with an exponentially decaying impulse response:

$$e'(t) = e(t) * \exp\left(-\frac{t}{\tau}\right)$$

According to Kuttruff, a reasonable choice for the time constant τ is 25 milliseconds. While just looking at a reflectogram can be very suggestive, it does not permit a safe decision to whether a particular reflection will be heard as an echo. The chapter of acoustic parameters discusses several parameters that make an objective judgement possible.

Filter types
Besides applying time integration to the impulse response its envelope, various filter types can be used. Some examples are listed below.

- The input signal is passed through a high-pass filter. This is done to protect the vulnerable (small) speakers from large (low frequency) deflections. In the scale measurement setup, the high pass filter is set to 2 kHz. This allows frequencies to be measured as low as 200 Hz (due to the scale correction).
- Low pass filters can be applied to cancel out high frequency noise and to restrict the bandwidth of a signal to satisfy the sampling theorem (and avoid aliasing).
- Octave- or third-octave can be used to extract the impulse response for a specific frequency band. These narrowband impulse responses can be used to determine the acoustic parameters for a specific frequency band.
- A (gliding) (third-) octave filter can be used to average the power spectrum of a signal to study its changes more effectively (for example to study colouration (i.e. frequency changes)).

SCALE MODELLING[2]

Often the acoustician is interested in the sound transmission of a room while it is designed. A common method to study the acoustic field of a room is by studying a smaller model (i.e. a scale model) of the room that has similar geometrical properties. The main advantage of scale modelling is that many variants of the room can be studied without much effort; from the choice of materials (absorptions/scattering) to changes in the shape of the room. Also all physical aspects of the acoustic field, including diffraction and interference, are modelled. With the modern technology, it is possible to generate and measure sound wave with a sufficient amount of accuracy to directly study the sound field in a room. Some acoustical scale modelling rules have to be considered. The quantities of the scale model are denoted by a prime.

The scale factor σ gives the relation between the corresponding lengths and can be rewritten as a scale factor for the length and time and time differences:

$$\boxed{l' = \frac{l}{\sigma}} \qquad \boxed{t' = \frac{l'}{c'} = \frac{l}{c'\sigma} = \frac{c}{c'}\frac{t}{\sigma}}$$

The scaling law for the wavelength and frequency are:

$$\boxed{\lambda' = \frac{\lambda}{\sigma}} \quad and \quad \frac{c'}{f'} = \frac{c}{f}\frac{1}{\sigma} \rightarrow \boxed{f' = \sigma\frac{c'}{c}f}$$

In the case the model is filled with air (c=c'), and the model is scaled down by (1:10), i.e. σ=10, then a frequency range of 100-5000 Hz corresponds to 1-50 kHz in the scale model.

An important aspect that needs consideration is the absorption at the walls and the attenuation of the pressure perturbations in the air. The absorption coefficient at the wall should have the same absorption coefficient at the raised frequency then the real wall at the original frequency:

$$\boxed{\alpha_i'(f') = \alpha_i(f)}$$

At Peutz this is done by selecting a specially fitted coating (paint) for the wooden model that approximately fulfils this requirement.

Because the attenuation of a medium increases with the square of the frequency, correction is particularly important in scale modelling. The correction for the scaled attenuation of the air is often done numerically after the measurement of the impulse response. At Peutz this correction is done numerically in accordance with iso9613_1.

Because often many different variants are measured and compared in scale modelling, reproducibility plays an important role. Fluctuation in temperature and humidity influence among other the attenuation of the medium in the scale model and the speed of sound. Because of this, an environmentally steady room is desirable when doing scale model measurements. At Peutz the scale model is placed in a climate controlled room. Also the temperature is constantly monitored in and outside of the model using thermocouples.

Also one should take special care with the selection of the proper equipment used in scale modelling because obtaining omnidirectional pulse excitation and reception is more difficult for higher frequencies.
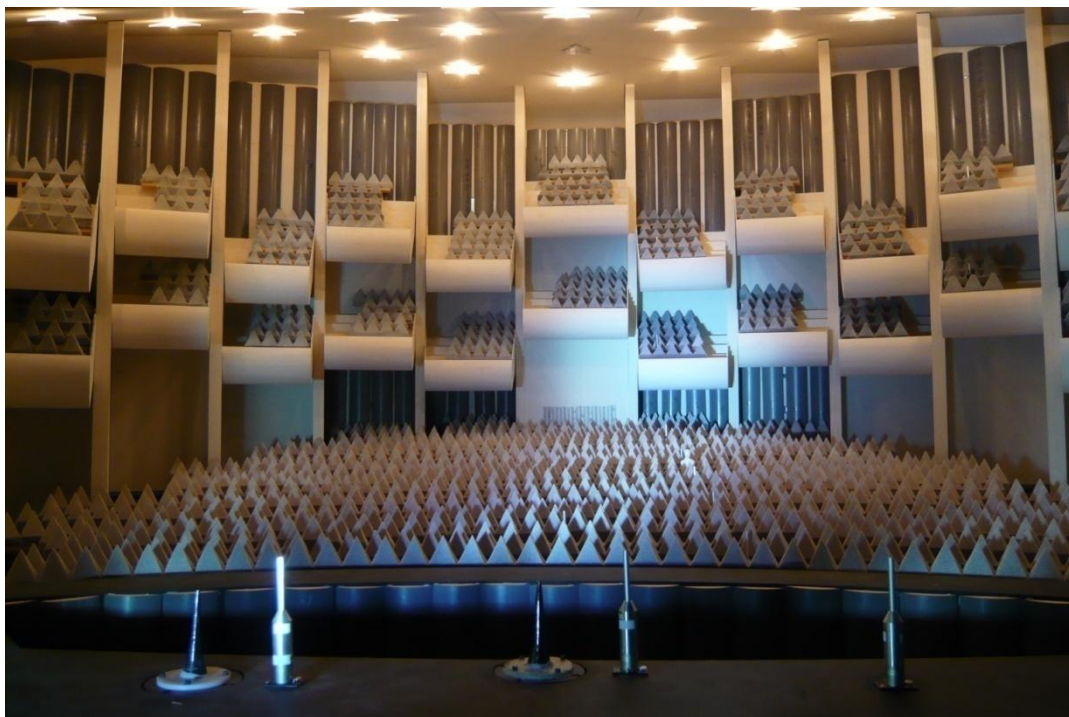

FIGURE 14: THE INSIDE OF THE OPER KÖLN SCALE MODEL

AURALISATION[2]

The term auralisation includes all the techniques that are used to create an audible impression from the design data of a (not yet existing) enclosure. Anechoic music or speech is fed to a transmission system (i.e. correlated with its impulse response) that models the changes to the sound of the real room in a similar fashion. This transmission system can be a scale model with a suitable source and receiver or a digital filter with the same impulse response as the room. The output of the room model should be binaural to transmit spatial (realistic) effects to the listener. The resulting sound can be studied to using a headphone of using two speakers in an anechoic room. Figure 15 shows this principle. Peutz uses a special head shaped microphone to capture the binaural sound in the scale model.

FIGURE 15: THE PRINCIPLE OF AURALISATION

ACOUSTIC VARIABLES

This section gives an overview of the definitions of the acoustic parameters that are used by Peutz in scale model analysis.

REVERBERATION TIME (RT30, RT60) (SECONDS)[2, 4]

Reverberation in room acoustics is the continuation of a (musical) sound in a hall after the source (e.g. an instrument) ceases to sound. The most famous concert halls have reverberation time (RT) between 1.8 and 2 seconds (at frequencies between 350 and 1400 Hz). Reverberation is not unwanted or desirable, but is rather a tool for creating musical effects. It fills the spaces between notes and it provides the fullness of a tone. It can also however reduce the distinctness of tones, or for example the intelligibility.

The reverberation time can technically be defined as:

The duration (in seconds) required for the (space averaged) sound energy density to in an enclosure to decrease by 60 dB after the source emission.

It is common to use a smaller dynamic range to determine the slope of the decay (in case of for example a high S/N ratio. For RT30, the values of 5 and 35 dB below the initial level are used. The curve is then extrapolated to the decay time corresponding to 60 dB. The decay curve from which the reverberation time is extracted is the graphical representation of the sound pressure level versus the time. The two methods for obtaining the impulse response have been discussed in the excitation signals section. As Peutz uses the integrated impulse response method in combination with MLS excitation, this process is discussed in some more detail further in this section.

The quasi-random fluctuations of the decay level can be avoided by averaging over a great number of reverberation curves. A similar result can be obtained by a technique called reverse time integration, first applied by Schroeder. He derived that the reverberation time can

accurately be derived from the curve that is obtained by applying reversed (i.e. backward) time integration to the square of the impulse response. This is mathematically done applying:

$$E(t) = \int_{t}^{t_s=\infty} p^2(\tau)d\tau = \int_{t_s=\infty}^{t} p^2(\tau)d(-\tau) = \int_{0}^{t_s=\infty} p^2(\tau)d\tau - \int_{0}^{t} p^2(\tau)d\tau$$

E: is the energy of the decay curve as a function of time
p(t) is the sound pressure and t the time.
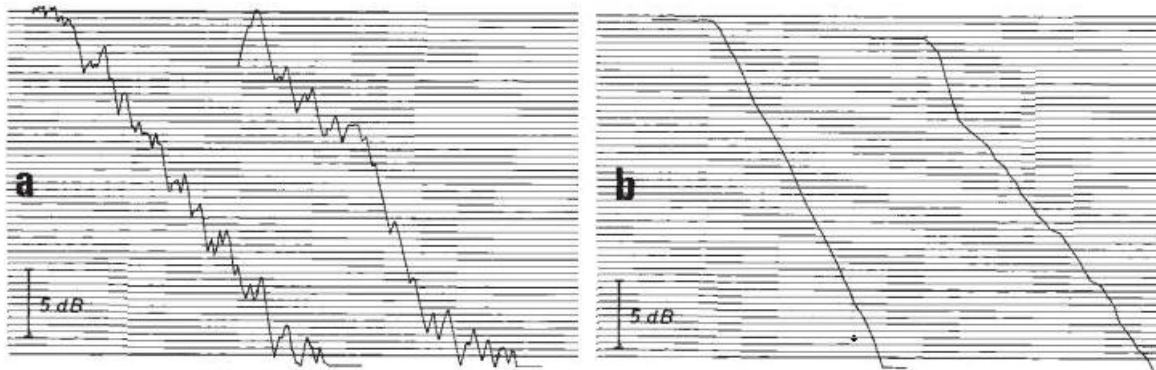
FIGURE 16: (A) SQUARED IMPULSE RESPONSE AND (B) APPLYING BACKWARD INTEGRATION

Theoretically the integrated impulse response corresponds to an infinite number of interrupted noise excitations. In practice however the measurement uncertainty is comparable to ten averages of interrupted noise excitations. No averaging is therefore needed to increase the measurement accuracy, i.e. if there are no external sound sources and the temperature and humidity field stay constant. Because there can always be external influences, some level of averaging is often still applied.

If the level of the background noise in known, the influence of background noise can be minimized by changing the starting value t_s of integral above. The value t_s should then be the time corresponding to the intersection of the horizontal line through the noise and the line through a representative part of the impulse response (on a decibel scale). The reverberation time can be derived from E(t) in a similar fashion as for the impulse response. As an example, to determine T30, a least squared fit is drawn trough the curve from 5 dB until 35 dB below the steady state level (i.e. the total level of the integrated impulse response). The slope of the line is called the decay rate d in dB per second. T30 is then simply given by:

$$T30 = 60/d$$

A more subjective measure of the reverberation time is called the early decay time (EDT). When music is played rapidly, only the early part (the first 10 dB decay after cut-off) of the reverberation remains audible between successive notes. For this reason the EDT time is therefore measured only over the first 10 dB of the decay curve. The decay is then multiplied by a factor six such that it can be compared with the reverberation time.

(FLUTTER) ECHOS AND COLOURATION[2]

Echoes describe a delayed reflection that can be heard as a discrete event. For a reflection to be heard it must arrive at least 50 ms later then the direct sound. Also the reflection has to be more prominent then neighbouring reflections. For an echo to be heard as a reflection, it requires reflection from a large surface by a path that is simpler than other reflections of the same order of delay. Figure 17 and Figure 18 illustrate what is described above.

FIGURE 17

FIGURE 18
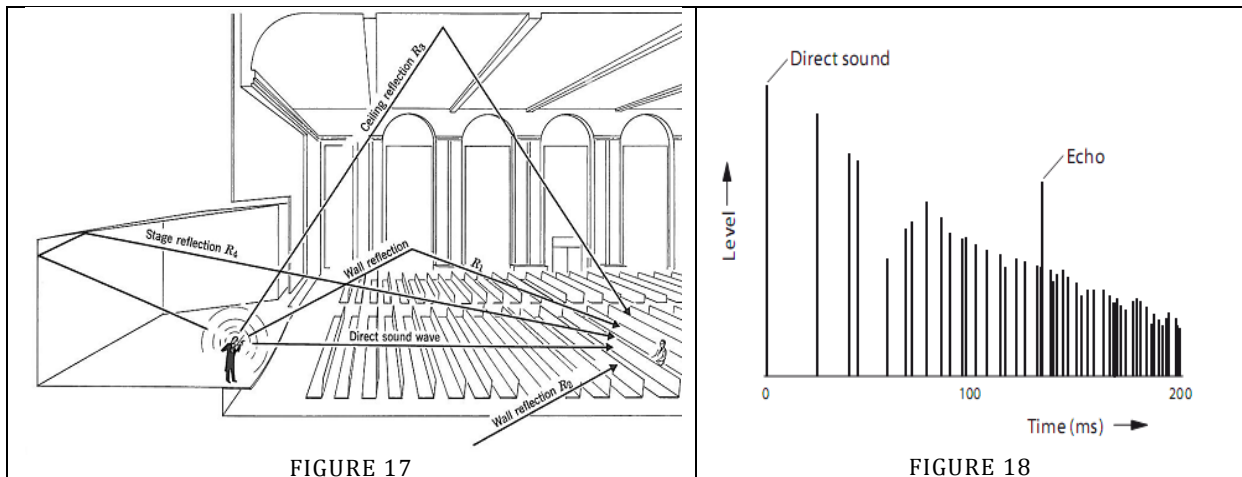
When a reflection is perceived as an echo is not trivial. Many studies have been done concerning the perceptibility of sound. Some factors of influence are the orientation, the relative loudness, the type of sound (impulse, speech or music) and the time-integration of the ear. A typical example of the complexity of the problem is that when at first an echo might not be heard, once detected, you might be unable to detect it. To be able to predict if a reflection is heard as an echo various "echo criterions have been designed"

Curing echoes in building can be a complicated task. Basically three methods are available to cure a room from echoes. Either absorption or scattering equipment can be used or the surface responsible for the echoes can be remodelled to redirect the reflection away from the source and listeners. Large areas of absorption should however be avoided because these lead to quieter sounds and possible "holes" in the (directional) sound distribution. Also too much scattering might me visually undesirable.

A special case of echo is the flutter echo. While echoes involve long path lengths, flutter echoes normally involve short path lengts, and are basically trains of period trains of reflections. A common place for a flutter echo to occur is between to parallel walls. The repeated reflections create a characteristic "twang" sound. This sound is for example head when you clap you hands in a corridor or room with flat parallel (non-absorbing) walls. More complicated repetitive paths are possible but rare. Figure 19 illustrates the principle.
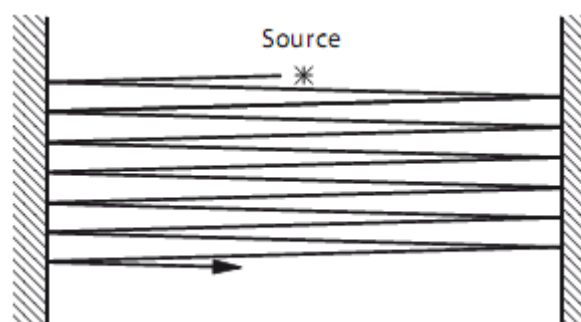

FIGURE 19: FLUTTER ECHO BETWEEN PARALLEL WALLS

Flutter echoes are often easily correct by applying absorption, or apply an inclination of only 5 degrees. When flutter echoes have short repetition time they can change the frequency spectrum of the sound as it travels through the room. This effect is called colouration of the sound.

The terms definition (Deutlichkeit) and clarity (Klarheitsmass) both describe the same musical quality. They describe the degree to which the listener can distinguish sounds in music or speech. The early time limit t_e of 50 ms is often used for speech, while a value of 80 ms is used for music. Mathematically the definition ($for\ t_e = 50\ ms$) is defined as the early to total sound energy ratio, and is defined as:

$$D_{50} = \frac{\int_0^{0.050s} p^2(t)dt}{\int_0^{\infty} p^2(t)dt}$$

The clarity is then defined as the early to late arriving sound energy ratio:

$$C_{50} = \frac{\int_0^{0.050s} p^2(t)dt}{\int_{0.050s}^{\infty} p^2(t)dt}$$

And the relation between the two is given by:

$$C_{50} = 10 \log\left(\frac{D_{50}}{1 - D_{50}}\right)\ dB$$

Criticism is give on the sudden discrete division between the early and late energy region, which is not how the ear operates. A parameter that does not have this deficiency is the centre time Ts given by:

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

It is time centre of gravity of the squared impulse response. In practice however, the centre time turned out to be highly correlated with the EDT which resulted in a loss of popularity.

EARLY AND LATE SUPPORT (ST1 ST2)[2, 4, 5]

For a good acoustics, it is important that the musicians can hear each other properly. A parameter that is designed to describe this aspect is the Support (ST), which describes the energy folded back to the musicians on stage. With the microphone and source positioned close together, two different version of the support van be derived from the measured impulse response.

The **early support** relates to the ensemble conditions, i.e. how easy other members of an orchestra can be understood. The early support is the ratio (in dB) of the early reflected energy (20-100 ms) relative to the direct sound (0-10 ms) measured at 1.0m from an omnidirectional source.

$$ST_{early} = 10 \log\left[\frac{\int_{0.020}^{0.100} p^2(t)dt}{\int_0^{0.010} p^2(t)dt}\right] dB$$

The early support does not consider influences of the direct sound, delay time or reflections from nearby surfaces. Reflecting objects should therefore not be closer than 2 meter from the measurement position.

The **late support** is the ratio of the he late reflected energy (100-1000ms) with respect to the direct sound, both measured at 1.0m from an omnidirectional source.

$$ST_{late} = 10\log\left[\frac{\int_{0.100}^{1.000} p^2(t)dt}{\int_{0}^{0.010} p^2(t)dt}\right] dB$$

The late support relates to the perceived reverberance of the room. This is the response of the room heard by the musician. What is the best support value depends per musician and their instrument. Singers generally prefer high ST late values; it should be noted however that the ratio of the two support variants is also very important. If both the early and the late support are high, the reverberance experienced by the musician might appear weak. If this balance is too low, musicians get the feeling that they are decoupled from the auditorium.

SOUND STRENGTH (G) (DB)[2, 4]

The subjective perception of Loudness is quantified by the objective parameter known as sound strength (G). A same orchestra will produce different sound levels in different halls and because listeners are sensitive to changes in loudness, it is important to be able to judge and tune this parameter:

$$G = 10\log\left[\frac{\int_{0}^{\infty} p^2(t)dt}{\int_{0}^{\infty} p_A{}^2(t)dt}\right] dB$$

p_A is the response at 10 meters from the same source in free field. The starting time should correspond to the start of the direct sound and the time chosen for infinite should be larger than the time needed for the decay curve to decrease by 30 dB. It is possible to do the 10 measurement close to the source. In this case a correction is needed however for the geometrical spreading.

According to ISO3382-1:2009, this minimum distance is 3 meter (i.e. 30 cm for a 1:10 scale model). Peutz measures the direct sound at 1 meter, resulting in a correction of:

$$20\log\left[\frac{1}{10}\right] = -20 dB$$

When the measurements are normalized to the direct sound at 10 meter, the obtained strength values are usually between 0 and +10 dB.

SPEECH INTELLIGIBILITY [6, 7]

This section deals with speech intelligibility and discusses two evaluation methods, STI and Alcons. Speech consists of vowels and consonants. Two categories of speech sounds are distinguished. **Voiced sounds** (including all vowels) are produced by air from the lungs pushed through vibrating vocal chords. This (broadband) sound is then filtered by the resonant cavities in the throat, nose and mouth giving the sound the characteristics that distinguishes our voices. Movement of the thong, jaws and lips make it possible to produce different vowel sound. **Consonants** have an impulsive character, and then can be voiced (e.g. b,d) or unvoiced (e.g. p,t).

When late reflections or long reverberation times are present, the larger energy present in vowels (with respect to consonants) often results in the masking of consonants, making it more difficult to understand words. The influence that a transmission system has on speech

intelligibility is, among other factors, dependent on the reverberation time, echoes, S/N ratio, non-linear distortions and psychoacoustics masking effects.

The absolute measurement of speech intelligibility is a very complex process. In the following section, two commonly used parameters for the evaluation of the intelligibility are discussed.

The **speech transmission index (STI)** (dB) is the most commonly used indicator for evaluation of the speech transmission. The STI can be extracted from the impulse response in a room and describes how well a transmission system can transfer the characteristics of a speech signal. The Method uses the modulation transfer function, defined as:

"MTF: the magnitude of the Fourier transform of the squared impulse response
divided by the total energy in the impulse response"

For the STI method, the modulation transfer function is determined for 7 octave filtered impulse responses (125 Hz to 8 kHz). The amplitudes at 14 modulation frequencies (spaced 1/3 of an octave apart) are then determined for each impulse response. Peutz uses the following modulation frequencies:

$$0.63 \quad 0.8 \quad 1 \quad 1.25 \quad 1.6 \quad 2 \quad 2.5 \quad 3.15 \quad 4 \quad 5 \quad 6.3 \quad 8 \quad 10 \ and \ 12.5 \ Hz$$

The obtained amplitudes are called the m-values. With 14 m-values for 7 octave bands, this leads to a total of 98 m-values. These m-values can be rewritten to an apparent form of the S/N ratio (apparent because the reverberation also influences m, and thus it is not the real S/N ratio).

$$\left(\frac{S}{N}\right) = 10 \log\left[\frac{m}{1-m}\right]$$

The range of (S/N) is limited to plus/minus 15 dB. For every octave band, the ordinary average is determined, leading to 7 new m-values. Next a weighted average is applied to average these new found m-values. Peutz uses the weighting values in the table below:

Frequency band	125Hz	250Hz	500Hz	1kHz	2 kHz	4kHz	8kHz
Weights	0.13	0.14	0.11	0.12	0.19	0.17	0.14

The STI value can now be extracted from the apparent S/N ratio, using:

$$STI = \frac{(S/N) + 15}{30}$$

The range of possible STI values has been experientially coupled to a scale describing the intelligibility. Figure 20 shows the STI scale.
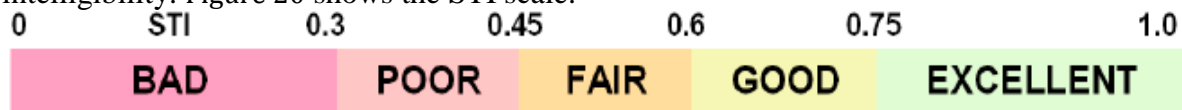
FIGURE 20: STI SCALE

ALcons (articulation loss of consonants) is the percentage of wrongly understood consonants. This is a test for which test persons are required. Peutz however designed a method to extract the ALcons data from an impulse response. There is also a mathematical relation between the STI values and ALcons that are used in the signal processing programs:

$$AL_{cons}(\%) = 170.5405 \cdot e^{-5.419 \cdot STI}$$

Other commonly used indicators for speech intelligibility are the articulation index (AI), the Clearance (C50) and derivation from STI as STIPA or RASTI.

To give some insight in what is required for a good acoustic quality; this section gives an overview of some important criteria.

- The optimum reverberation time is a compromise between:
    - Clarity (good intelligibility; requiring short reverberation times)
    - Sound intensity (requiring a high reverberant level)
    - Liveness (requiring a long reverberation time).

    The optimum also depends on the purpose of the room. (e.g.: spoken word, chamber music, opera, etc.)
- Some other common requirements for good acoustics are:
    - Uniformity of the acoustic field (good diffusion)
    - Freedom from echoes
    - Minimal background noise.
- Some other, more subjective acoustic attributes of a concert hall include:
    - Early decay time (The initial rate of decay of reverberation is apparently more perceptually important than the total reverberation time.
    - Spatial impression (dependent on contributions to the early reflections from above and especially from the sides)
- And special care should be taken to avoid:
    - (flutter) Echoes
    - Sound focusing
    - Sound shadows
    - Background noise
    - Too much reverberant sound (will result in loss of clarity)

INTRODUCTION

In this section, the projects that were contributed to during the internship are described. Although the activities partly overlapped, the sections are chronologically ordered as much as possible.

The internship started with the task to determine the acoustic transparency of a mesh. The architect of the "Staatsoper Unter den Linden" designed this mesh to visually conceal an added volume to the room, serving to increase the reverberation time.

The second project involved a scale model research of the "Bühnen Köln" opera. Various structural changes were made to the scale model, while studying the changes of the acoustic field by means of impulse response measurements. However, the most time was spent to the optimisation of the WINMLS and Raci code. A detailed summary, describing all the major changes to the codes, can be found in the chapter "Matlab codes scale model analysis".

During the last weeks of my internship a large amount of measurements, concerning a scale model research of a new reverberation chamber design, was efficiently processed.

MEASURING THE ACOUSTIC TRANSPARANCE OF THE REVERBERENT GALLERY'S VISUAL CONCEALMENT [9]

INTRODUCTION

The internship started with the analysis of the acoustic transparency of a mesh (a diagonal diamond structure). The architect of the "Staatsoper Unter den Linden" wanted to use this mesh to visually conceal a volume that created to increase the volume of the room. This volume involved space obtained by raising the roof of the opera. Figure 21 shows the original design of the mesh, as designed by the architect.

FIGURE 21: STAATSOPER UNTER DEN LINDEN

From an acoustic point of view it is very important that this mesh transmits most of the sound energy. Because the mesh seemed relatively dense, Peutz decided that the acoustic transparency should be studied. Two (1:2) scale models of typical parts of the mesh were made (1x1 meter). Figure 22 shows these two samples.

FIGURE 22: TWO STUDIED SAMPLES OF THE MESH.
LEFT: PANEL A – MESH WITH MODERATE TRANSPARANCY.
RIGHT: PANEL B –MESH WIT LIMITED TRANSPARANCY

RESEARCH METHODS[9]

As mentioned above, the most important question is if the windows are acoustically transparent. It is also important to know if the energy is either reflected or absorbed. Two methods have been used to research these properties of the mesh. In the first method, the direct passage of the sound through the panel was studied to obtain the transmission loss through the panel. In the second method, the reflection was studied for a 180 degree circle. The percentages of reflected and transmitted energy were calculated to obtain an estimation of the absorbed energy.

TRANSMISSION LOSS[9]

Both measurements were done in the anechoic chamber of Peutz. The lower boundary for echo freedom is around 500 Hz. A semi-omnidirectional speaker and omnidirectional microphone were used for the measurements.

The distance between source and microphone was 2 meter, where the panel (mesh) was centred in the middle. Different orientation angles of the panel were measured (from perpendicular, with changes of ten degrees until parallel). Also two different variants for each panel (with and without a wire web attached) were measured. The measurements were done with a MLSSA measurement system. This system uses a MLS (maximum length sequence) signal to obtain the impulse response. With the help of a Fourier transformation and proper filtering of the signal, the frequency response function (transfer function) was obtained. Next, third-octave averages were determined. In order to calculate the transmission loss caused by the panel, these values were compared with the (third octave averaged) situation without a mesh. Finally, the transmission losses were converted back to a 1:1 scale.

The results have been presented to Peutz in a dynamic Excel sheet that made plotting of the various transmission losses possible for all angles. A summary of the obtained transmission losses, (which were presented in the final report of Peutz), is given in the table below. These values are averages for the most important orientation angles.

Frequency	Panel A	Panel B
<500 Hz	0-0,5 dB	0-0,5 dB
500 Hz	1 dB	1 dB
1000 Hz	1 dB	1 dB
1250-1500 Hz	1-5 dB	1-5 dB
2000 Hz	3 dB	5 dB
2500 Hz	4 dB	5 dB
3000 Hz	5 dB	7 dB
4000 Hz	4 dB	10 dB

A 1 dB transmission loss already corresponds with a loss of 20 percent of the energy. A 3 dB loss corresponds with a loss of 50 percent, and 10 dB with a loss of 90 percent. The effect of the wire web turned out to have little influence on the transmission loss.

From these results it was concluded that only for frequencies until 1000 Hz the transmission loss was acceptable. The losses at higher frequencies are too high. The measured transmission losses can be the result of either reflection or absorption, of which absorption is the worst of the two. This is because for reflection, the energy stays in the room (as if there was a wall), while for absorption, the energy is lost.

REFLECTION LOSS[9]

Because energy that is reflected is less of a problem (the energy stays in the room), another experiment was done to study the absorption of the incoming sound wave. The same source was now positioned at 6 meters from the mesh, measuring the reflected acoustic sound pressure for every 5th degree on a half circle (at the height of the centre of the mesh), 2 meters from the mesh. This pressure (for different frequency bands) was then multiplied with a corresponding area element (assuming radial symmetry) to obtain the reflected energy. The spectral energy reflections have been compared with the incoming sound energy (that was measured at the microphone perpendicular to the mesh). Correction was needed for the extra 4 meters that the reflected energy travelled. For free field condition, the geometrical spreading of the sound pressure behaves as $1/r^2$, resulting in a correction given by:

$$gs_{cor} = 20 \log_{10} \left(\frac{L_3}{L_3 + 2 \cdot L_1} \right) = 20 \log_{10} \left(\frac{4}{4+4} \right) \approx -6dB$$

L_1 is the distance between the microphone and mesh. (2 meter)
L_2 is the distance between the source and the mesh. (6 meter)
$L_3 = L_2 - L_1$ is the distance between the source and on axis microphone. (4 meter)

Figure 23 shows the results, plotting the ratio of the reflected sound energy versus the frequency.
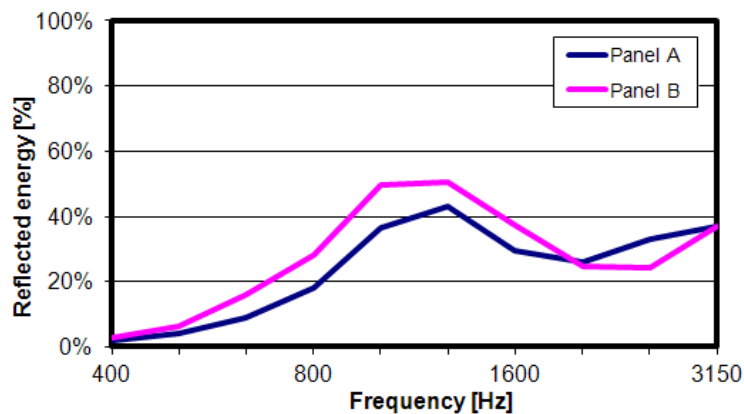

FIGURE 23: REFLECTED ENERGY RATIO OF THE TWO PANELS PLOTTED AGAINST THE FREQUENCY.

From Figure 23, it is concluded that for the higher frequencies, between 30 and 50 percent of the energy is reflected. This however explains only a part of the transmission loss that was measured, indicating that a part of the energy was absorbed. A reliable estimation of the absorption can however not be given.

Besides an analysis of the energy absorption, polar figures were created. For the polar figures the reference was not the incoming sound, but the reflection of a 100% reflecting solid plate. No correction was needed for the geometrical spreading. Figure 24 shows an example of a polar figure. Measurements close +- 90 degrees were not always accurate because wall reflections could not be removed in the time domain. However, most wall reflections were removed in the time domain using a reference measurement.
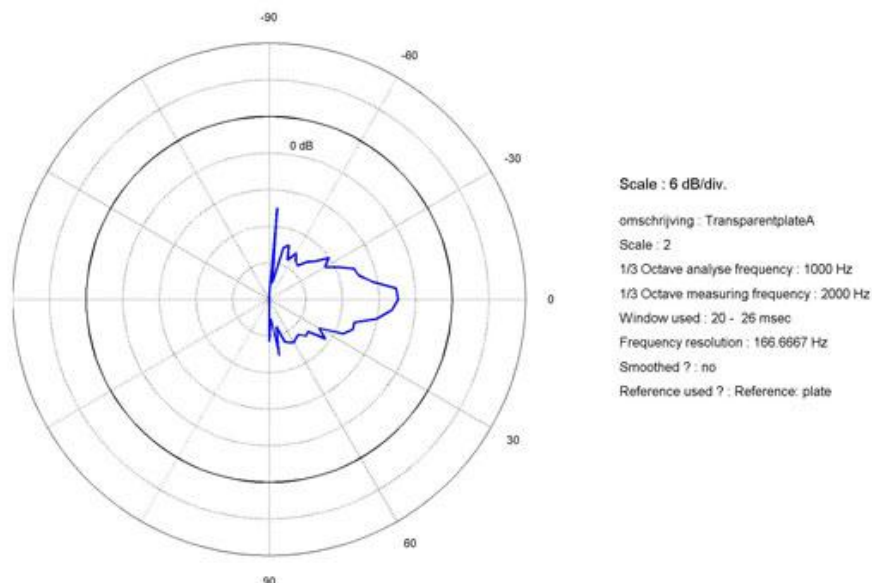
FIGURE 24: POLAR FIGURE FOR THE 1000HZ THIRD OCTAVE FREQUENCY BAND.
A 100% REFLECTING SOLID PLATE IS USED AS THE REFERENCE (THICKER CIRCLE).

From the results it is concluded that the mesh is not transparent enough for frequencies above +- 1000Hz. A more open design is therefore advised. Although a 1dB transmission loss for higher frequencies is expected to be unrealistic, the acoustic transmission can still be increased considerably by increasing the openness. It is advised to test the new design with a 1:1 sample (with an area of at least 10 square meters).

A revised version of an old Matlab code, used for processing reflection measurement data, was written during the internship.

In the current version of the program:
- o The impulse response reflection measurements are compared with a room reference measurement to remove any wall reflections still present in the measured signal.
- o Fourier transformation, filtering and third octave averaging are applied subsequently.
- o The transmission loss is determined using a reference. This reference can be a:
    - o Another panel located at the same location.
    - o The direct sound incident on the perpendicular microphone, correcting for geometrical spreading of sound energy.
- o The transmission losses with respect to the chosen reference are plotted in polar figures and are exported to a tab separated text file.

First the old code was analysed, structured and commented. When the main operations became clear, the relevant sections were grouped and checked for errors. The code has been commented as much as possible to make future changes and understanding as easy as possible. Because it is beyond the scope of this report to go into all the details of the code, only the most important revisions are listed below.

- **Revision of code structure:** The code is now subdivided into more functions, resulting in a more comprehensive structure. Many comments have been added, contributing to the understandability and customizability of the code.
- A program was written that locates the peaks of the measured impulse responses (of the mesh and that of the reference). The sample numbers of these two peaks are compared to obtain a shift, which is used to ensure perfect overlap of the signals. The two signals are then subtracted from each other to remove any remaining wall reflections. This is plotted in a figure to check the validity of this operation. Figure 25 shows this visual verification.
- A new code was written that made it possible to compare the reflections with the incoming sound, while correcting for geometrical spreading of the sound energy.
- A new menu structure; with the option to select the direct sound as reference.
- Calibration is now disabled because the same microphone was used for all angles.
- Data is now exported to a more comprehensive and correct polar figure. The data is now also exported to "tab separated" text file.
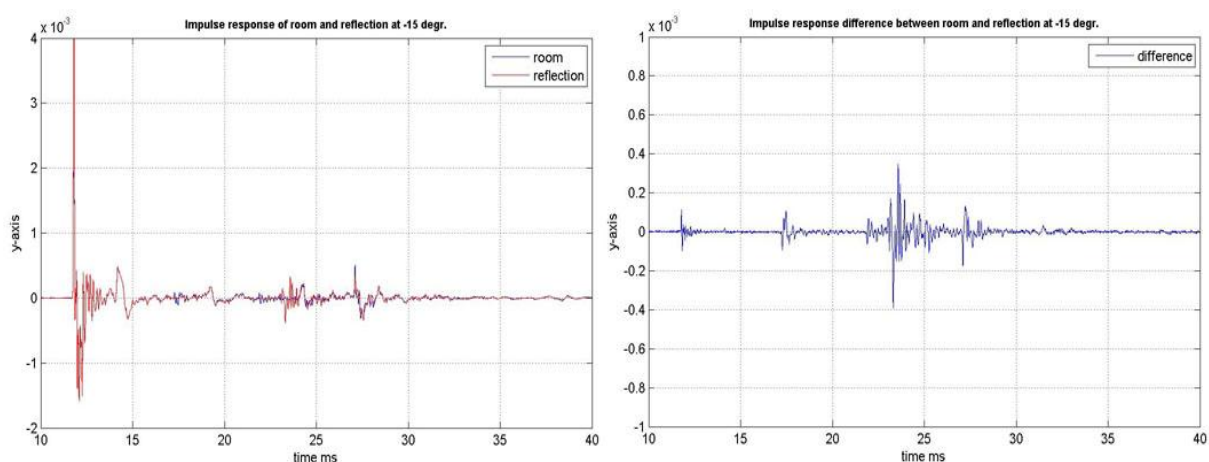

FIGURE 25: VALIDATION OF OVERLAP BETWEEN MEASUREMENTS

INTRODUCTION

The second consult that I contributed to during the internship concerned the scale model analysis of the Opernhaus in Köln. In this section a short overview is given of the different tasks that I was responsible for. Also the measurement setup is described.

For the renovation of the Opernhaus in Köln (Bühnen Köln), Peutz was responsible for the room acoustics part. From the measurements at the opera, it became clear that the acoustics was relatively "dry", and that echoes were found at certain locations. With the help of a computer model several possible "improvements" were studied. As the results were promising, a scale model was build to further study the possible improvements. Figure 26 shows the inside of the Köln scale model.

FIGURE 26: SCALE MODEL OF THE OPERNHAUS IN KÖLN

During the scale model research I was, among others, responsible for the following things:
- To test different variants of the scale model and to process these measurements with a Matlab program. Two important variants were the:
  - New proscenium walls and ceiling.
  - Added diffusers to the rear walls and ceiling.
- To improve the Matlab program that was used for processing these measurements. More information about this program is found in a later section.

In a later phase of my internship, I also contributed to the scale model research of "das Kulturpalast" in Dresden and the "Majellakerk" in Amsterdam.

The scale model has been designed as a 1:10 representation of the original design. Features smaller then 10/20 centimetres were not considered in the model, resulting in a limiting upper frequency of around 2 kHz (20 kHz in the scale model). As air was used in the model, correction for the frequency dependent attenuation was done in the Matlab code. A special paint coating (that approximated the frequency dependent absorption properties of the wall) was applied. The corrected absorption at the walls however, was not to a high level of accuracy. To increase the repeatability of the measurements, the scale model was placed in a climate controlled room. Due to this, variations of temperature (up to 1 C accurate) and air humidity were kept to a minimum. The sources (5) and microphones (20) were positioned at the same (scaled) positions as in the original measurement situation in the Opera. A dummy head was placed to study binaural effects and make auralisation possible.

The control centre of the measurement setup consisted of:

A **computer** to run a **Matlab program** that controlled the **WINMLS** software, and to save the obtained (impulse response) measurement data. **WINMLS software** that created a **MLS excitation**. A **High pass filter** that ensured the removal of low frequencies that would damage the sources. **Two amplifiers** to control the gain of the output signal. **One amplifier** to calibrate the microphones and to make microphones selection possible. A **microvolt meter** to check if the output level is correct. Various cables to connect the equipment above. Another computer to constantly monitor the temperature at certain positions in the room (using thermocouples). Figure 27 and Figure 28 show an overview of the measurement setup.

FIGURE 27: MEASUREMENT SETUP
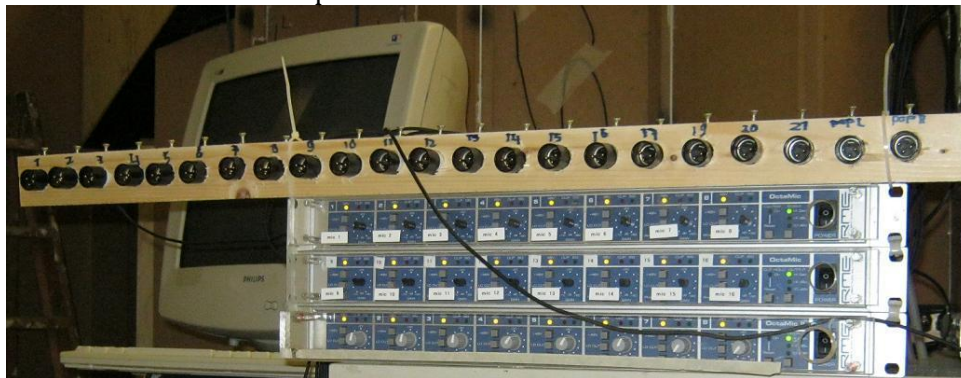
FIGURE 28: MICROPHONE AMPLIFICATION SELECTION

With this setup, the impulse responses could be calculated from all sources to all microphones. The total measurement time was around 1 hour, because all measurements (from all source to all microphones) had to be done one at time. Measuring of all microphones at a time was not possible due to hardware limitations.

Even though much effort was done to resemble the real situation as closely as possible, the approximations resulted in a simplified model. Especially because the absorption of the walls was only roughly approximating the real situation, estimation of the strength and reverberation time parameters was not really accurate. The combination with the computer model made it possible to increase the reliability of the results.

INTRODUCTION

During my internship, my main occupation concerned the validation and optimization of three Matlab codes, of which the code for processing the polar diffusion measurements (diff_polar_5gr) has already been discussed. A short description of the three codes is given below.

- **Diff_polar_5gr:** is a program used for the analysis of reflection measurements in the anechoic chamber. The program processes impulse response measurements into polar figures and organized text files.
- **Raci(dist):** is a code used for processing "non-scale model" room acoustic impulse response measurements. After filtering and averaging is applied, the required acoustic parameters are extracted from the signal.
- **WINMLS:** is a code similar to the Raci code, but in this case applicable to scale model measurements. Also the output format and code structure is different.

CODE OPTIMIZATION

As the diff_polar_5gr has already been discussed, this section focuses on changes in the Raci and WINMLS code.

Quite some time has been spent into the analysis and understanding of the Matlab codes. During this process the codes were rewritten into more comprehensive versions. Important changes with respect to the comprehensibility are:
- Unused functions and sub codes have been archived into subfolders.
- Unused codes (that seemed of no future importance) have been removed.
- The code outlining has been corrected into a more consistence and clear format.
- Different parts of the code have been reorganized into separate folders to ensure the stability of shared functions, in case of changes in the code.
- Several variables names have been renamed into Mnemonic variable names.
- Several double variables have been combined.
- Extra comments have been added and unclear comments were removed or reformulated.
- The WINMLS and Raci code now have a more similar structure.
- The three codes have been coupled with a graphical user interface, and organised in one main directory, *IRmeasurements,* with the Matlab files in the corresponding subfolder.

Besides comprehensibility, both codes have also been optimized to reduce the computation time needed for processing a set of measurements. The time needed for the data selection and variable input has also been reduced. Important changes concerning the reduction in computation time are:
- All pause statements in the code have been removed.
- All the plots that were displayed during the processing have been turned off.
- A sound is played when the program is finished, such that one can directly start to process the next dataset.
- **WINMLS only:** All the data from one measurement session can now be loaded in one time, instead of executing the program separately for every source measurement.

Besides optimising for speed, the user friendliness was improved. This has been done with the following changes:

- The large amount of unnecessary output that was printed to the command screen has been removed. Now only relevant information, showing the process of the program, is shown.
- The menu now shows the selected settings in the main menu, reducing the chance of errors.
- **WINMLS code only:** A more user friendly input window has been created, automatically entering the standard settings. This reduces the computation time, but also reduces the chance of making errors. Also Error checking is used to check if the correct "10 meter" microphones have been selected.

The programs have been verified according to NEN-EN-ISO 3382-1 (giving norms relating to room acoustic parameter acquisition). Various features have also been added to the program. The most important changes are listed below:

- Excel (.xls) output is now automatically generated, giving an organized overview of all the room acoustic parameters, ordered to parameter type, frequency and microphone-source number.
- Some bugs in the figure output have been corrected, readability and consistence between the different figures is improved.
- All the data is now automatically saved to the correct directory folder, separating the 1 kHz figures automatically.
- **Raci code only:**
  - A sliding third octave filter has been added to better study the behaviour of the early and late frequency spectrum.
  - Wrong averaging over the octave averaged frequency bands has been corrected; the arithmetical average is now calculated from the 500Hz and 1000Hz octave averages.
  - Some outdated parameters have been removed from the output.
  - The strength parameter is now also calculated for the Raci code.

SUMMARY AND FINAL RECOMMENDATION

As can be read in the previous section, quite some aspects of the code have improved. The codes have become less prone to errors caused by the user and the time needed for loading and processing a data session has decreased immensely. Where the processing of a full measurement set first took over an hour (with the user required to intervene every 15 minutes to load a new set of data), this is now required only once, with the program finished in 15 minutes.

Extra features have been added, among others the added Strength calculation in the Raci code, the excel output and the new menu structure. Last but not least a lot of effort was put into the format of the code, making future changes more convenient. Examples of the final figures and the excel output are given in Appendix A.

I would recommend combining the Raci and WINMLS code into one version. Both codes are used extensively by the company, whereas some important aspects of the code are still unclear. Doing this would show the differences between the programs and make future changes easier.

During the last few weeks of the internship, it was asked to convert a large amount of measurement data into third octave filtered reverberation times. The measurements were part of a research of Margriet Lautenbach and Martijn Vercammen. The measurements are series of impulse response measurements executed in a scale model of the reverberation of Peutz. The reverberation data should indicate if a reverberation room filled with volume diffusers instead of panel diffusers has a higher degree of diffusion. This is studied by looking at the relative standard deviation between the difference source-microphone positions.

The processing of the measurements was done with a program named Spectralizer, written by Peutz. The output was a text file that had to be converted into a specific excel format. Because the amount of measurements was enormous, together with another trainee we designed a mouse controlling Macro using the Jitbit Macro Recorder. This macro was designed to do all the typing, clicking and selection of data, reducing the chance of errors and the time needed for a measurement. For the conversion from the text files to the specific excel output, two Matlab programs were written:
- The first program automatically renamed the files into the correct format
- The second program read the "colon-space", i.e. ": " separated data, and automatically saved it into the correct format in an excel file.

With the written programs, we were able to process all the measurements in 1.5 week, were it would normally have taken roughly twice the time. Because the excel data was not yet reviewed when I was finished, it is not possible to include the results of the research in this report.

As written in the introduction, during my internship I expected to learn about experimental room acoustics, gain insight in (acoustic) consultancy work and gain experience with signal processing techniques. In the following, I will give summarize my experiences and reflect on the internship.

After a few days, getting to know the company structure and staff, I got my first real assignment. I had to design a 10 kHz microphone calibrator from a 1 kHz version. The atmosphere at Peutz was very welcoming, and it felt good to be able to contribute something to the company.

During the first month I was mainly occupied with determining the transparency and absorption of a mesh in the anechoic chamber. The transmission measurements were processed with a program called MLSSA and an Excel file which I wrote. The reflection measurements were processed with a Matlab program that I had to revise.

In this period I also started to assist with the scale model research, testing different variants and processing data. I gained experience with the measurement setup and the important aspects that need consideration in experimental room acoustics scale modelling.

Because I felt the need for a more puzzling assignment, I pointed this out to my supervisor (Margriet Lautenbach). This way I became responsible for the revision of the Raci and WINMLS code. Because of my position I was then able to:
- Study the programs used for processing the impulse response measurements.
- Do these measurements with the WINMLS code.
- Read about the literature in remaining spare time

This made it possible to learn a lot about experimental room acoustics and the underlying concepts. Because the program was used extensively during the revision, many ideas developed that would make the program more user-friendly. The time needed to process a measurement session was reduced to a minimum (reducing the time needed from more than an hour to less than 15 minutes). Because at the time of my internship, there was no one who knew the exact operation of the Raci program, it was sometimes difficult to understand what was happening in the code. This resulted in many small literature studies. Because the supervision was often limited to once a week, it was not possible to understand, and properly describe all of the aspects of the code within the short period of my internship. Whenever I got stuck, it was possible to find something interesting and educational to do until the return of my supervisors.

In conclusion it can be said that all my expectations of the internship have been fulfilled. I gained insight in signal analysis and hand on experience with acoustic measurements techniques. From staff members I learned what acoustic consultancy work involves.

With the final codes handed over to Peutz, I feel that I have really contributed something to the company. I was able to apply the knowledge obtained over the past years into practice. Although more supervision from Peutz would have resulted in a little more productivity, it did allow me to work independently, and think of some creative solutions for the program.

Although I spent most of my time working alone, I enjoyed collaborating with colleagues at Peutz. Working on a small part of a large project, reminded me of group work done during my study. A big difference however, was that the consequence of my work was not only a judgement, but also influenced the course of the project. I found this responsibility highly motivating.

APPENDIX A: FINAL FIGURES AND THE EXCEL OUTPUT

WINMLS

Place   : no matching description available in varianten.txt
Place   : no matching description available in varianten.txt
Source pos. : 2
Mic. pos.   : 1
Filter   : 1000 Hz
Scale   : 10

Absorption corr. 1   : 0.154 dB/m @ (10000 - 1000) Hz
Absorption corr. 2   : 0.148 dB/m @ (10000 - 1000) Hz
Time to first arrival : 9 ms
Stationairy SPL 1   : 93.8 dB for KB08
Stationairy SPL 2   : 94.5 dB for KB18
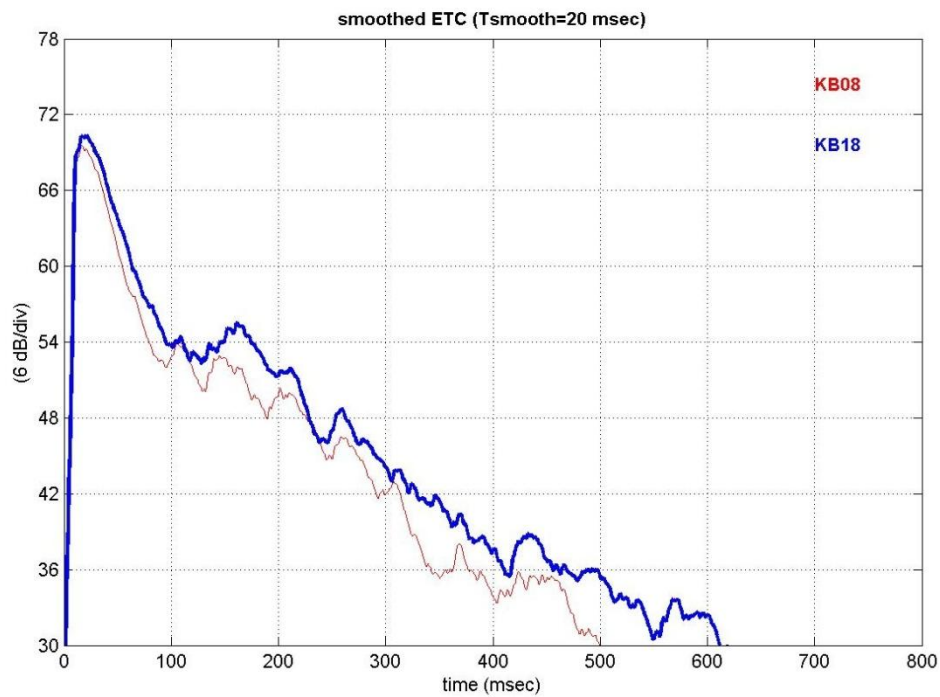
Parameters: KB08

CL80 : 15.7 dB

D50   : 0.97

ST1   : -15.5 dB

G   : 0 dB [ref. no 1m file]

G5-80: 0 dB [ref. no 1m file]

RT60 : 1.2 sec

ALcons 4.6 %

Air conditions:

Temp   : 20 deg. Celsius

Rel. Humid : 50 %

Air Press.   : 101.325 kPa

Parameters: KB18

CL80 : 14.6 dB

D50   : 0.96

ST1   : -13.8 dB

G   : 0 dB [ref. no 1m file]

G5-80: 0 dB [ref. no 1m file]

RT60 : 1.3 sec

ALcons 4.7 %

Air conditions:

Temp   : 21 deg. Celsius

Rel. Humid : 50 %

Air Press.   : 101.325 kPa

Programma versie 2011I 2ch, 24-10-2011 , 9:53:41, Peutz bv (c)

FIGURE 29: WINMLS - COMPARISON BETWEEN TWO VARIANTS

Place : test version 1
File : TEST01-101.wmb
Source pos. : 1
Mic. pos. : 1
Filter : 1000 Hz

Scale : 10
Absorption corr. : 0.154 dB/m @ (10000 - 1000) Hz
Time to first arrival : 9 ms
Stationairy SPL : 97 dB

ETC

Parameters:

CL80 : 11.4 dB

D50 : 0.93

ST1 : -12.1 dB

G : 24.6 dB [ref. TEST01-101.wmb]

G5-80: 22.4 dB [ref. TEST01-101.wmb]

RT60 : 2.3 sec

ALcons 6.1 %

Air conditions:

Temp : 20 deg. Celsius

Rel. Humid: 50 %

Air Press. : 101.325 kPa

smoothed ETC (Tsmooth=20 msec)

cumulative ETC
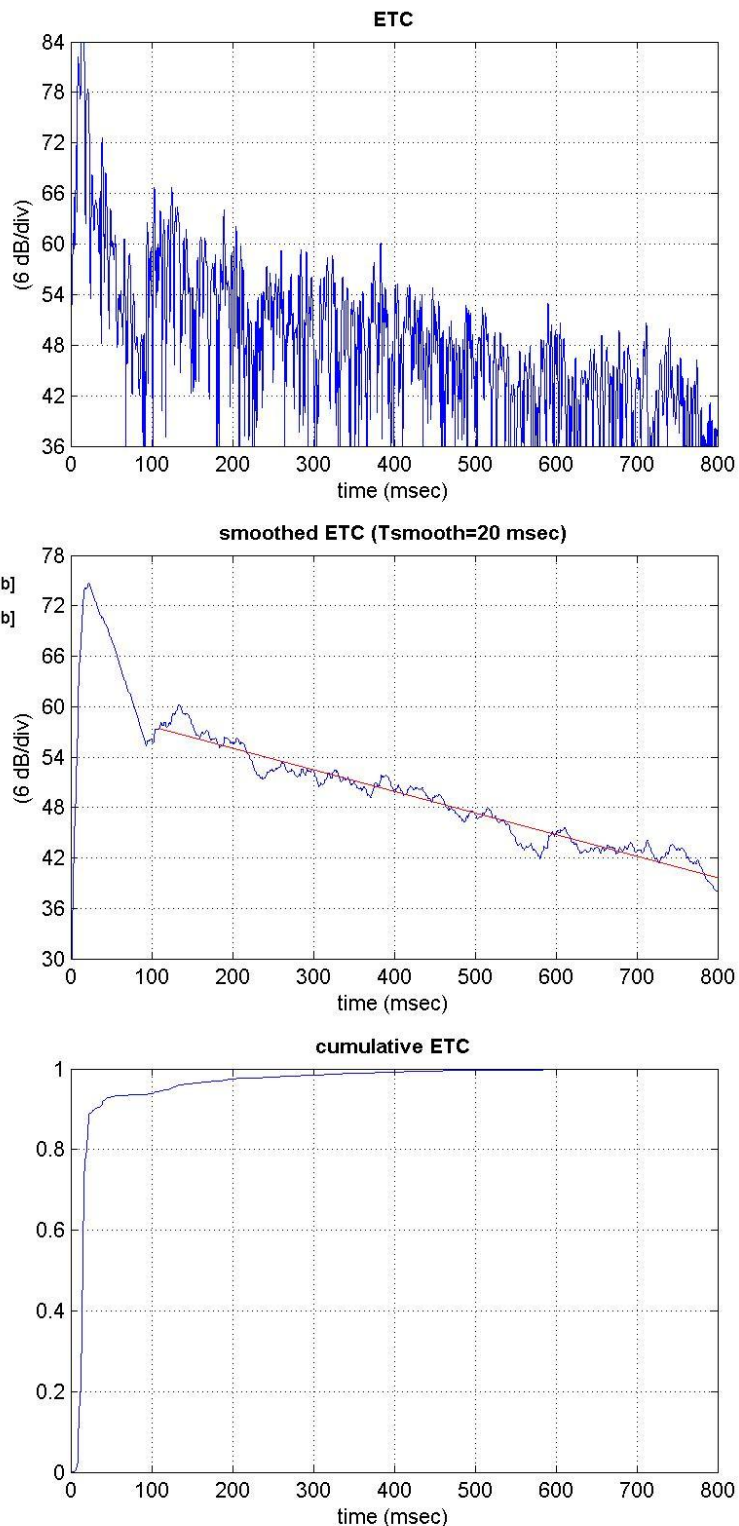
Program version 2011J 2ch, 28-11-2011 , 14:54:27, Peutz bv (c)

FIGURE 30: WINMLS - SINGLE VARIANT PLOT FOR 1KHZ

Location | enter name of oper

File | KOUG-104.TIM

Sample frequency | 36036 Hz

Frequency band | 1000 Hz

Source distance | 7.62 m

Time to first arrival | 22 ms

Parameters:

S/N	22.4 dB
Dir/Rev	-1.5 dB
C_{50}	0.6 dB
C_{80}	2.3 dB
C_{5-80}	-0.2 dB
D_{50}	53 %
D_{80}	63 %
T_{Center}	88 ms
$Echograd_{Musik}$	0.7
$Echograd_{Sprache}$	0.7
EDT_{Sch}	1.53 s
$EDT_{ETC10dB}$	1.41 s
$T_{30\ ETCLSQ}$	1.44 s
$T_{30\ ETCsm.}$ ----	1.45 s
ST1	-1.2 dB
ST2	1.5 dB
G	0 dB
G_{5-80}	0 dB
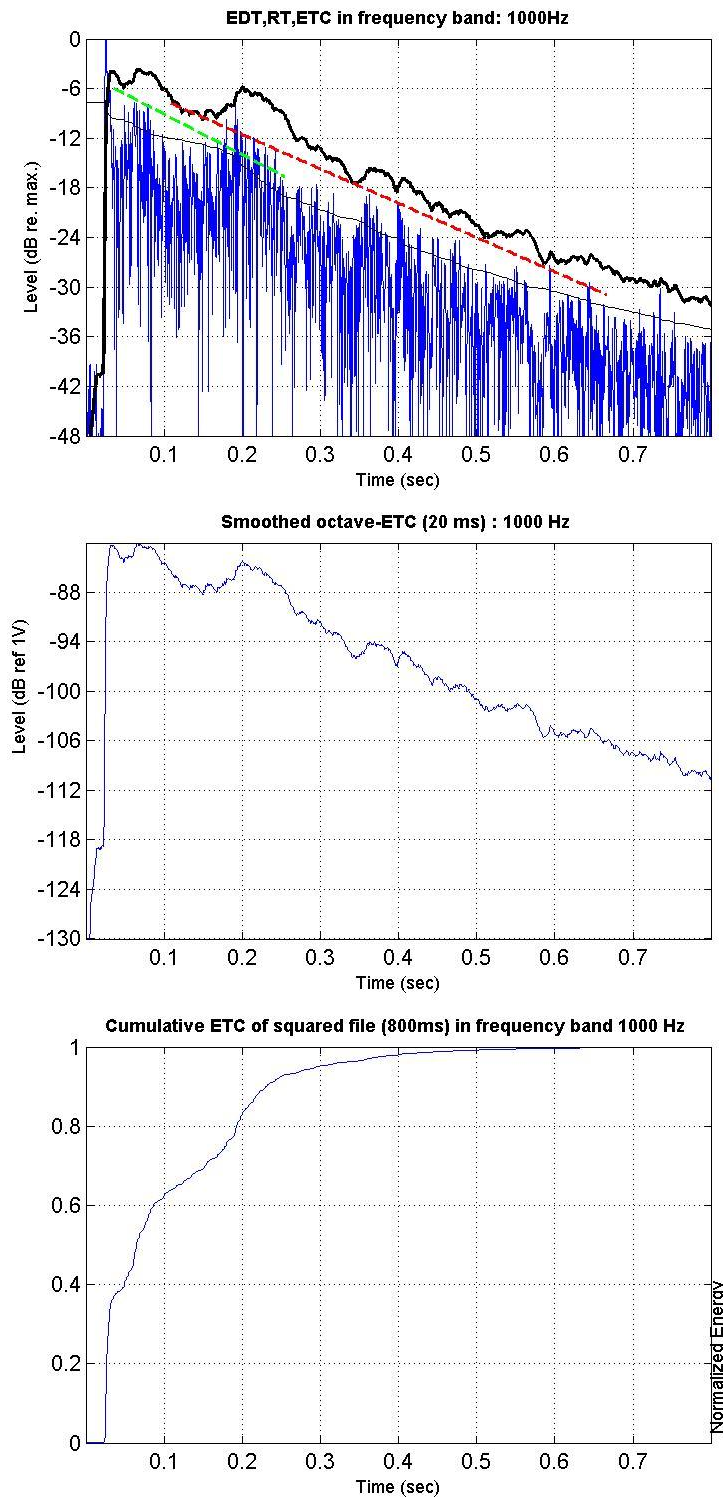
(c) 2001-2010 Peutz

IR-eval. Date:18-Nov-2011

FIGURE 31: RACI - 1KHZ OUTPUT

Location   enter name of oper

File   KOUG-104.TIM

Sample frequency 36036 Hz

Frequency band   20 to 18019 Hz

Source distance   7.62 m

Time to first arrival 22 ms

Parameters:

AL_{cons} Peutz 88   6.7 %

AL_{cons} (STI)   6.1 %
STI   0.61

ø(500-2000 Hz):

S/N ø   24.6 dB
Dir/Rev   -1.2 dB
C_{50} ø   2 dB
C_{80} ø   4 dB
C_{5-80} ø   1.8 dB
D_{50} ø   62 %
D_{80} ø   72 %
T_{Center} ø   70 ms
$Echograd_{Musik}$ ø   0.8
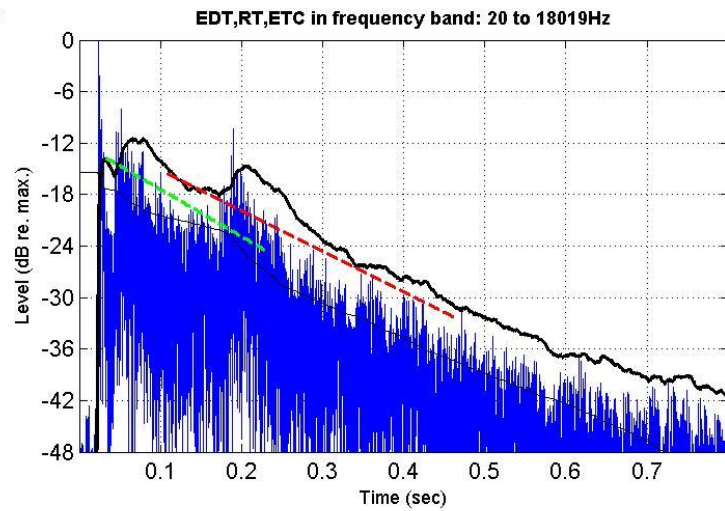$Echograd_{Sprache}$ ø   0.8
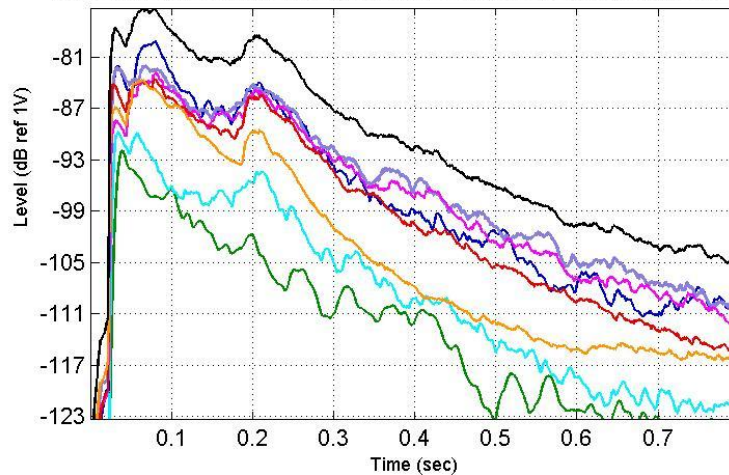EDT_{Sch} ø   1.3 s
$EDT_{ETC10dB}$ ø   1.2 s
$T_{30\ ETCLSQ}$ ø   1.2 s
$T_{30\ ETCsm.}$ ø   1.2 s
ST1 ø   1 dB
ST2 ø   2.6 dB

EDT,RT,ETC in frequency band: 20 to 18019Hz

Sm.ETC: yel.63Hz, grn 125Hz, turq.250Hz, bl.500Hz, vio.1kHz, pnk 2kHz, red 4kHz, orang.8kHz, blck.allp.

Cumul.ETC: grn 125Hz, turq.250Hz, bl.500Hz, vio.1kHz, pnk 2kHz, red 4kHz, orang.8kHz, blck.allp.

Oct.	125	250	500	1k	2k	4k -Hz
C_{50}(dB)	4.1	3.5	2	0.6	-1.7	2.2
C_{80}(dB)	7	4.3	4	2.3	0.8	3.9
D_{50}(%)	72	69	62	53	40	63
T_c(ms)	51	60	70	88	106	64
Ek_M	0.6	0.7	0.8	0.7	0.8	0.8
Ek_S	0.6	0.7	0.8	0.7	0.8	0.8
EDT_{sm}(s)	0.97	2.39	2.59	4.14	18.87	3.63
EDT_{10}(s)	0.98	1.23	1.24	1.41	1.45	1.26
EDT_{Sc}(s)	1.1	1.43	1.3	1.53	1.42	1.52
T_{30sm}(s)	1.36	1.19	1.16	1.45	1.36	1.18

BR =   1

(c) 2001-2010 Peutz
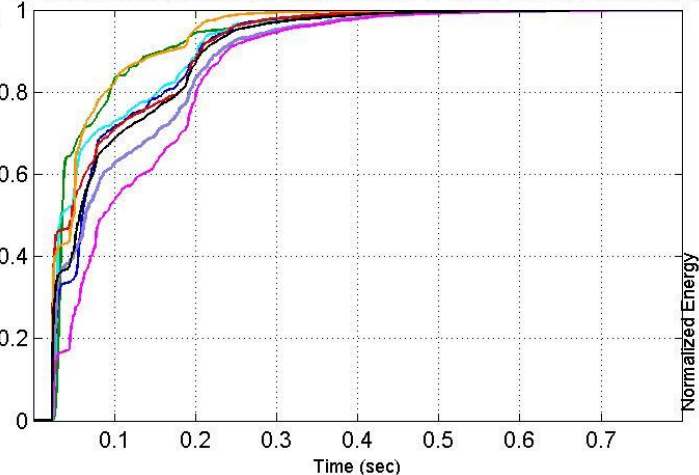
IR-eval. Date:18-Nov-2011

FIGURE 32: RACI – ALLOCTAVES PLOT

Location   enter name of oper

File   KOUG-104.TIM

Sample frequency   36036 Hz

Frequency band   20 to 18019 Hz

Source distance   7.62 m

Time to first arrival   22 ms

Parameters:

AL_{cons} Peutz 88   6.7 %

AL_{cons} (STI)   6.1 %

STI   0.61

ø(500-1000 Hz) - ISO3382-1: 2009

S/N ø	23.5 dB
Dir/Rev ø	-1.4 dB
C_{50} ø	1.3 dB
C_{80} ø	3.2 dB
C_{5^-80} ø	1 dB
D_{50} ø	58 %
D_{80} ø	68 %
T_{Center} ø	79 ms
$Echograd_{Musik}$ ø	0.8
$Echograd_{Sprache}$ ø	0.8
EDT_{Sch} ø	1.4 s
$EDT_{ETC10dB}$ ø	1.3 s
$T_{30\ ETCSch}$ ø	1.3 s
$T_{30\ ETCsm.}$ ø ----	1.3 s
ST1 ø	-0.1 dB
ST2 ø	2.1 dB

(c) 2001-2010 Peutz

IR-eval. Date:18-Nov-2011

Frequency response direct sound (blue) and total (grey)

MTF: grn 125Hz, turq.250Hz, bl.500Hz, vio.1kHz, pnk 2kHz, red 4kHz, orang.8kHz, blck.allp.

Echograd Sprache: grn 125Hz, turq.250Hz, bl.500Hz, vio.1kHz, pnk 2kHz, red 4kHz, orang.8kHz, blck.allp.

RTsm (__) EDTSch (--) EDT10 (.-) by freq.
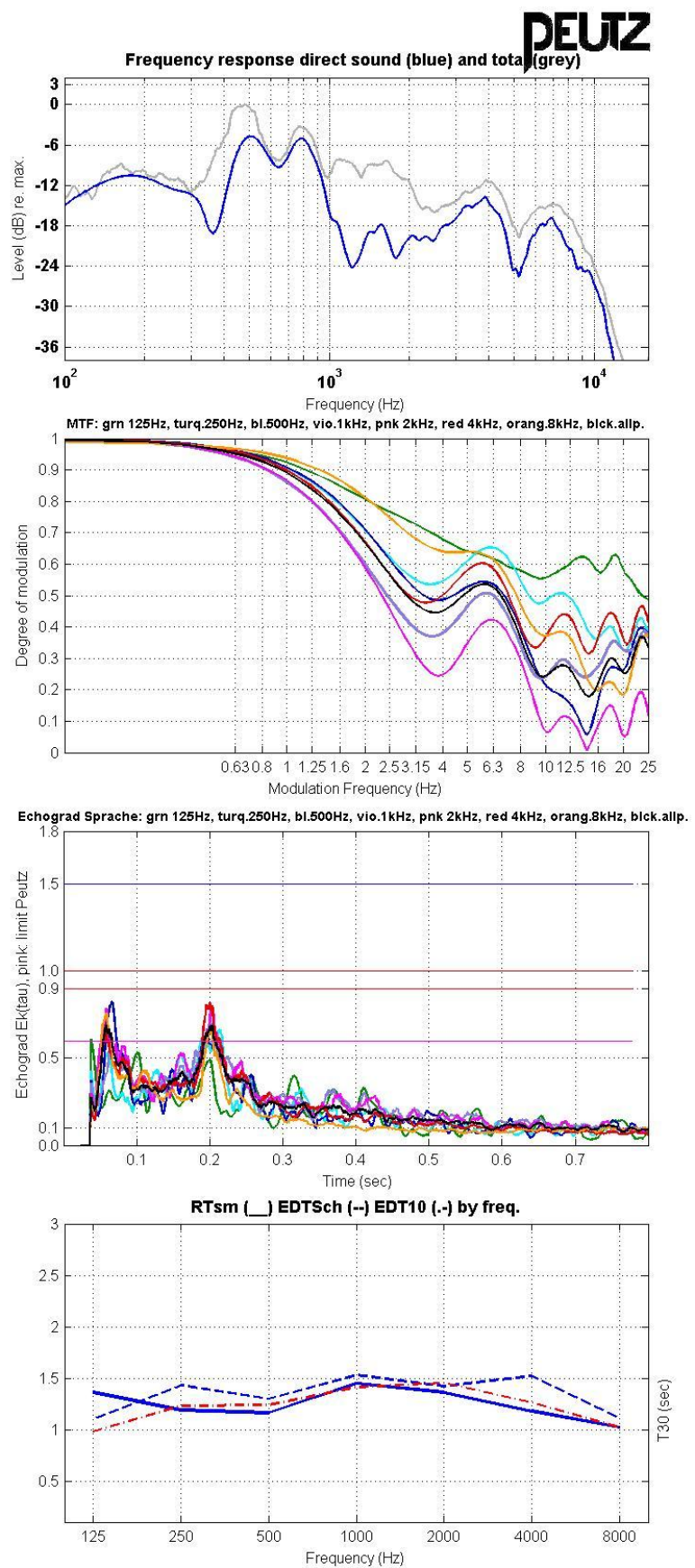
FIGURE 33: RACI - THIRD PLOT OUTPUT

1. *Website of Peutz Netherlands*. Available from:
    http://www.peutz.nl/.
2. Kuttroff, H., *Room acoustics*2009.
3. Mathworld. *Convolution*. Available from:
    http://mathworld.wolfram.com/Convolution.html.
4. NEN-EN-ISO, *Acoustics - measurement of room acoustic parameters - performance spaces (NEN-EN-ISO 3382-1)*, 2009.
5. *Acoustics for choir and orchestra*. 2011; Available from:
    http://www.akutek.info/research_files/stage_acoustics.htm.
6. Becker, F. *a do it yourselfers guide to computing the speech transmission index*. 2010; Available from:
    http://www.synaudcon.com/site/articles/a-do-it-yourselfers-guide-to-computing-the-speech-transmission-index/.
7. werff, J.v.d., *Speech intelligibility* 2004.
8. *Auditorium Acoustics* 1999; Available from:
    https://ccrma.stanford.edu/CCRMA/Courses/152/auditorium_acoustics.html.
9. M.Lautenbach_M.Vercammen, *Staatsoper Unter den Linden - Untersuchung der akustischen Transparenz der visuellen Abschirmung der Nachhallgalerie*, 2011.
10. M.Lautenbach_M.Vercammen, *Buhnen Koln - Entwerfungsplanung Raumakistik, Simulationen und Modelluntersuchungen*, 2011.