Auralisation of the Herz Jesu Church in Graz

Index

In order to present the contents of this project in a structured and detailed way, six chapters have been defined, containing each of them the necessary information for every step made.

Abstract

Chapter 1 contains an abstract in which the project is roughly explained: Firstly the main goal (auralisations of the Herz Jesu Church for different audience configurations and different source positions) is exposed. Afterwards, the single steps followed in this project : impulse response measurement and studio work are briefly explained.

Chapter 1 : *Theory*

Chapter 1 explains the theoretical background that lies beneath the *"Listening to a church"* project. Firstly, the criteria for the election of the Herz Jesu Church as a suitable building are exposed. Afterwards, the acoustical effects (diffraction,acoustic shadow...) expected in the church as well as the relevant acoustical factors (reverberation time, clarity degree ...) are described into detail. The *Theory* chapter ends up explaining the basics on the impulse response measurement method: The importance of the impulse response, the way the reverberation time can be won out of the impulse response and ,finally, how the measurement works using sine sweeps as testing signals.

Chapter 2: *Measurement*

Chapter 2 explains the measurement procedure carried out in the Herz Jesu Church.

Firstly, the measurement set up is briefly explained, continuing with a comparison between an impulse response measurement and a reverberation time (RT60) measurement, rounding off the theoretical background presented in Chapter 1. Afterwards, the characterisation of both paths, which has been carried out following the same structure, is explained into detail.

Firstly, the paths have been presented thanks to a map that shows the path as well as the location of the microphone positions. Afterwards, the expected behaviour along each path as well as the acoustical interesting microphone positions have been depicted, fleshing out the assumptions done according to the theoretical basics presented in Chapter 1.

Finally, a possible explanation for the different frequency ranges of the first microphone positions attached to each path is explained, falling back to the theoretical background explained in chapter 1.

Chapter 3 : *Studio Work*

Chapter 3 aims to explain how the auralisations and the videos were processed at the studio.
Firstly, the video processing will be explained, taking especial care about the way a map can be inserted in a video and the way to draw a line along the certain path.
After the auralised pieces are presented, the way the auralisations were carried out is described, firstly for the path attached to the source position at the altar and then for the one attached to the source position at the organ. Both audience configurations (empty and full church) are described for each path, pointing out the changes on the equalization and the loudness carried out in Samplitude in order to enlarge the acoustical effects found along each path.

Chapter 4: *Conclusions*

Chapter 4 gives an overview throughout the whole project, recalling on the one hand the single steps of the project and, on the other hand, reflecting the learning experience this project means for the ones involved in it.

Chapter 5 : *Literature*

Chapter 5 contains the bibliography used to carry out this project.

Abstract

This project aims to present with help of auralisations, how the architectural conditions of the Herz Jesu Church affect a dry music recording at different positions within the church depending on two different source positions,one at the altar and one at the organ and on two different audience configurations :an empty and a full church.

Firstly, 2 paths along the church were defined and each attached to a different source position.
Along each paths, 30 microphone positions were located in order to measure the impulse response " the acoustical fingerprint of the church " .
The impulse response measurement was carried out in an empty church in order to catch the very essential behaviour of the church.
Knowing the impulse responses along both paths, one music piece for each path was auralised, that means, convoluted with the impulse responses of the church along the corresponding path.
The auralisations were carried out in Samplitude, an audio editing software, using an effect called "Room simulator", which allows the convolution of music with impulse responses.
Changing the Room simulator parameters such as the reverberation time, the auralisation for a full church could be easily carried out.
Finally, two videos were recorded to show the paths and provide the listener with and audiovisual experience that should show how exciting it can be listening to a church.

Picture 1: Herz Jesu Church

THEORY

Introduction:

Chapter 1 *"Theory"* presents and develops a whole set of acoustical effects and parameters as well as relevant measurement concepts in order to gain a deeper view on the theoretical fundamentals of this project. Although the abstract does not mention the theoretical component of this project, it seems sensible to provide the reader with enough knowledge on room acoustics so that he can understand and judge the decisions and assertions that configure this project.

Firstly, the selection of a church instead of another great building (e.g. an opera house) is explained into detail, being followed by an explanation about the reverberation time (RT60) and the reverberation radius. In order to give a sense for common values for both parameters, they have been calculated for the Herz Jesu Church and displayed in a table, together with all relevant parameters for their calculation.

Moreover, the clarity degree C_{80} is introduced to show how an increase of the influence of reverberating or direct field can be recognized . This 3 concepts, added to the explanation about the building selection, show the so called statistical acoustic model, which is valid as soon as the reflections can not be single traced, which is the case of a church.
Next, the geometrical acoustical model will be introduced by a definition on resonant frequencies, as well as their recognition.
Acoustical shadow and diffraction pass for perfect examples to develop a deeper understanding of the geometrical model. So, the acknowledged effects in a church correspond to two completely different acoustical models that exclude themselves, showing the great complexity of the church.

The third block explains the basic knowledge about the measurement procedure: The importance of the impulse response, the way the reverberation time can be won out of the impulse response and ,finally, how the measurement works using sine sweeps as testing signals: the importance of the impulse response,the way the reverberation time can be won out of the impulse response and ,finally, how the measurement works using sine sweeps as testing signals.
Finally, the Schroeder Integration Method, which allows the calculation of all acoustical parameters, will be described for the calculation of the reverberation time. Although this block only shows the tools used to calculate acoustic parameters , it seems sensible to show how an impulse response is defined, measured and manipulated since an auralisation is based on the fact, that impulse responses are the acoustical " finger prints " of any building.

Criteria for the choice of the Herz Jesu Church

The Herz Jesu Kirche is a neo-Gothic style Roman Catholic church in the Graz area St. Leonhard. The building was built in 1881-1887 has the third highest church tower in Austria and one of the most important buildings of historicism in Styria.

Size

Upper	Ship width	13 m
	Length	43.5 m
	Overall Length	62 m
	Peak height	24 m
Lower	Width	13 m
	Length	47 m
	Peak height	6 m
Tower height		109.6 m

Picture 1.1 : Church dimensions

Churches awake always great acoustical interest due to their complexity and heterogeneity that provide many interesting spots like arches, columns, chapels or even spiral staircases or an organ where several acoustical effects can be easily acknowledged

Formerly there were no microphones or any electronics that could help the priest to amplify his voice. Therefore, he had to use the geometry of the church to avoid having to shout and try to raise the sound.

The churches were built taking care of acoustics and ornament. Many churches, like the Herz Jesu in Graz, have passageways that work as resonators.

The evolution of dark Romanesque churches to luminous Gothic cathedrals increased the acoustic problems. Large dimensions, especially the height of the nave and large reflective surfaces,as well as the vaults were the cause of an excessive reverberation and echo production. Favourable effects are the windows over large areas and the provision of side chapels. They produce an effect similar as resonators, thereby contributing to diffusion for bass sound that are more difficult to absorb. Ornaments such as lamps, statues or secondary altars, which can be generally found in the side chapels, also affect the absorption.

The reverberation existent in a church depends on numerous factors such as the absorption coefficients of materials, volume, surface and geometrical shape of this.

In the Herz Jesu Church only a few different absorption materials can be found, such as the ornaments, the wooden banks and when it is occupied by the audience.

Large volumes (>10000 m^3) for the Herz Jesu Church ,14527,86 m^3, involve a significant decrease of the RT at the high frequencies (>2000 Hz). This effect, which induces a loss of brilliance is explained by the air absorption, which becomes significant at these frequencies, in particular for great volumes where the free average path between two reflections becomes very large.

In addition, in this church the stone and the harmony of great spaces are the principal decorations, which can explain the raised RT values at the low frequencies.

Reverberation time RT60:

The reverberation time is one of the oldest and most important parameters in room acoustic since it can characterize by it´s self almost every building.

It represents the -60 dB - decay time of a signal after having being reflected against some kind of surface, which means that the signal looses 10^6 of it´s energy. Obviously, the reverberation time measurement only makes sense in a closed volume, where the signal can be reflected. Because of the presence of a diffuse field, the model used is the one that involves statistical acoustics Sabine discovered, that the reverberation time doesn´t depend on the loudness of the signal rather than on the geometrical structure of the room where it is played. Furthermore, he realized that each frequency had his own reverberation time. Due to this reason, RT60 – values are always calculated for each octave.

Reverberation times in the lower frequency range are always longer than for high frequencies due to the poor absorption coefficients most materials present for low frequencies. So, an averaged reverberation time over all octaves doesn´t reflect the characteristics of a room at all.
Nowadays, it´s often seen that only a -30dB decay of a signal is measured (-5 to -35dB) (RT30). The non-measured values are extrapolated. This procedure is necessary because having at least 60 dB SNR through the diffuse field it´s nearly impossible because there is also an SPL value >0 by silence.

The development of a formula to calculate the reverberation time was firstly carried out by Sabine, who developed the most popular formula:

$$\boxed{RT60 = 0{,}161 * \frac{Volume}{\sum S}}$$

$S = \alpha * A$ … absorbing surfaces
α … absorption coefficient

Later one, Eyring developed a more accurate formula but starting at a completely different point as Sabine:

$$\boxed{RT = 0{,}161 * \frac{V}{-A * \ln(1-\alpha)}}$$

If $\alpha < 0{,}2$, the difference between both formulas can be ignored.
Although this project doesn´t base on a reverberation time measurement, it´s importance seems obvious since suitable values for the reverberation time show how the church behaves for different frequencies.

The theoretical values for the RT60 and the measured ones obviously differ and in order to show how great the difference in fact is, the reverberation time, as well as the reverberation radius, were calculated using the formula by Sabine (stone doesn´t have absorption coefficients > 0,2 !)

Reverberation radius:

Since there are two different ways to explain the reverberation radius (Sabine and Eyring) , it has been decided to present the one developed under consideration of Sabine´s theoretical model to explain the reverberation time. Both paths lead to the same conclusion.

By free field sound propagation, the energy density of a spherical source with power P can be described this way:

$$E_{dir} = \frac{I_{dir}}{c} = \frac{P}{4\pi r^2 * c}$$

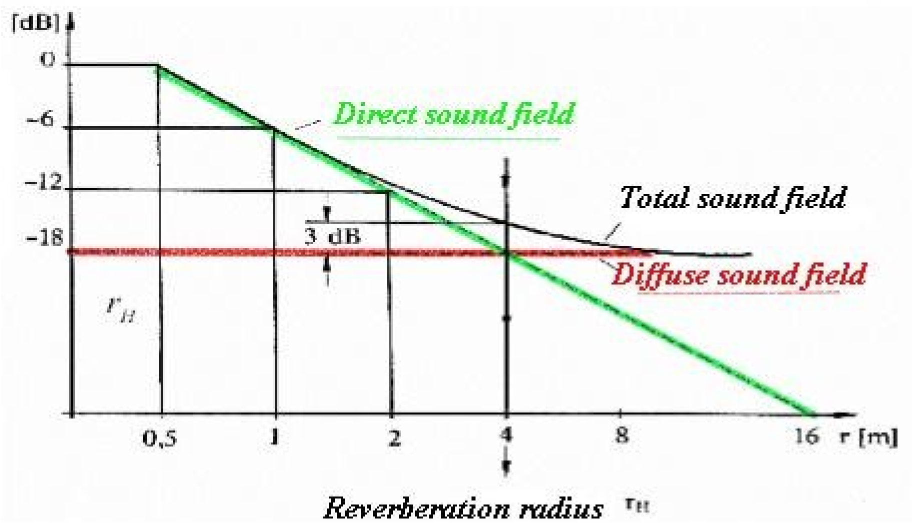
If such a source is switched on in a room, a sound field builds up till the energy density E_{st} reaches a constant value so that the power coming from the source is equal to the reflected power.
($P_{reflected} = P_{source}$)

$$E_{st} = \frac{4*P}{c*A_{ges}}$$

The reverberation radius shows the places along a circular path (the source regarded for this experiment is a spherical one !) where $E_{dir} = E_{st}$

$$\frac{P}{4\pi r_H^{2*} c} = \frac{4*P}{c*A_{ges}}$$

$$\boxed{r_H = 0{,}057 * \sqrt{\frac{V}{T}}}$$



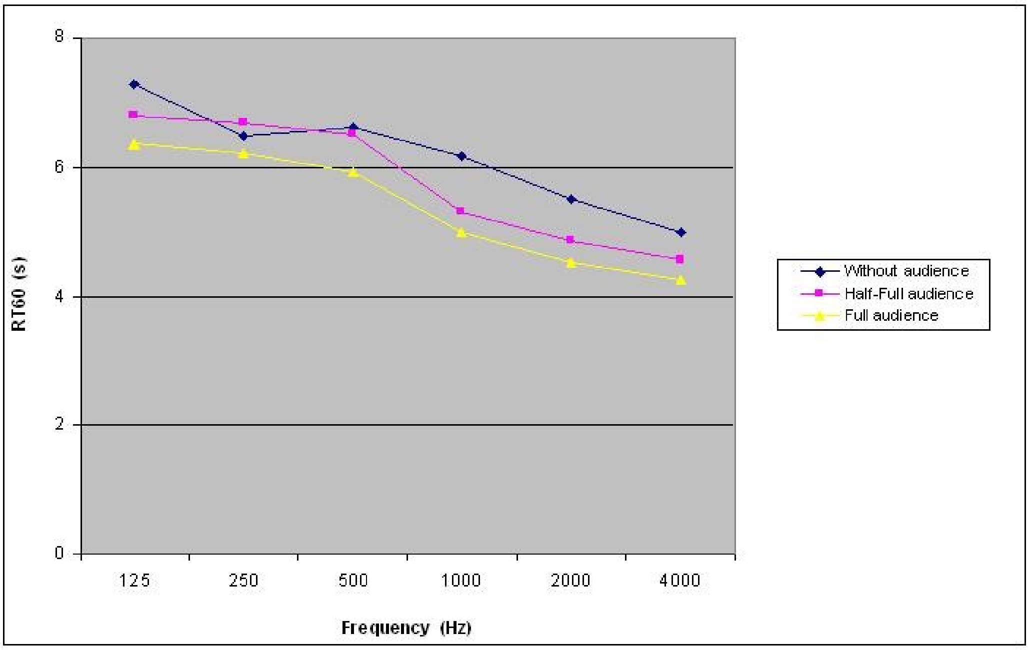
Picture 1.2: reverberation radius

Looking at the formula above, the Volume and the RT 60 play a key role in it. If the volume increases whereas RT60 remains constant, the reverberant radius will increase. The shorter the reverberation time, the smaller is the reverberation radius.

			α (f)					
Surface	**(m^2)**	**Material**	**125Hz**	**250Hz**	**500Hz**	**1000Hz**	**2000Hz**	**4000Hz**
Audience	153,4	People in wooden seat	0,34	0,39	0,51	0,44	0,56	0,56
Seats	153,4	Wood	0,04	0,05	0,06	0'06	0,08	0,08
Floor	850,43	Stone and wood cladding	0,09	0,08	0,08	0,09	0,1	0,15
Walls	2863,83	Lime and sand plaster	0,05	0,06	0,06	0,08	0,08	0,08
Ceiling	1147,279	Lime and sand plaster	0,05	0,06	0,06	0,08	0,08	0,08
Glass	383,5	Window glass	0,1	0,02	0,02	0,01	0,02	0,04

Without audience	Critical Distance[m]	2,54	2,69	2,67	2,76	2,93	3,07
	RT60[sec]	7,27	6,47	6,6	6,18	5,49	4,99
Half-Full audience	Critical Distance[m]	2,63	2,65	2,69	2,97	3,11	3,21
	RT60[sec]	6,78	6,68	6,49	5,32	4,85	4,56
Full audience	Critical Distance[m]	2,72	2,75	2,82	3,07	3,23	3,33
	RT60[sec]	6,36	6,21	5,92	4,98	4,5	4,25

Picture 1.3:Absorption coefficients for each octave band. RT60 and Critical Distance without,half-full and full audience.



Picture 1.4: All RT60

Clarity degree:

The clarity degree C_{80} for music is equal to the D_{50} for speech, though the integration runs through other values. The clarity degree provides information about the appreciation of succeeding tones and in how far different instruments can be acknowledged.

$$\boxed{C80 = 10 * \log \frac{W_{0...80}}{W_{80...\infty}}}$$

0 dB < C_{80} High clarity in the music.

-3dB < C_{80} < 0dB Music evolves the listener

The Energy throughout the first 80 ms is evaluated and set into proportion with the energy after 80 ms.

The clarity degree has been used to show how the church behaves towards the music at certain point of both round walks. Indeed, the feeling of being surrounded by music can be easily realized by the listener´s of the video, which is very positive.

Resonant frequency:

In smaller places there is an element that affects the acoustic quality which is the **resonances** or normal modes **.** This happens as a result of successive reflections on opposite walls. If a sound wave moves perpendicular towards a wall and is then reflected onto the opposite wall, a stationary wave will be generated. Resonant frequencies stand out very clearly due to their high energy.

$$\boxed{f_n = \frac{340\ m/s}{2} * \sqrt{\left(\frac{nx}{lx}\right)^2 + \left(\frac{ny}{ly}\right)^2 + \left(\frac{nz}{lz}\right)^2}}$$ (for a rectangular room)

lx...length [m]

ly... width [m]

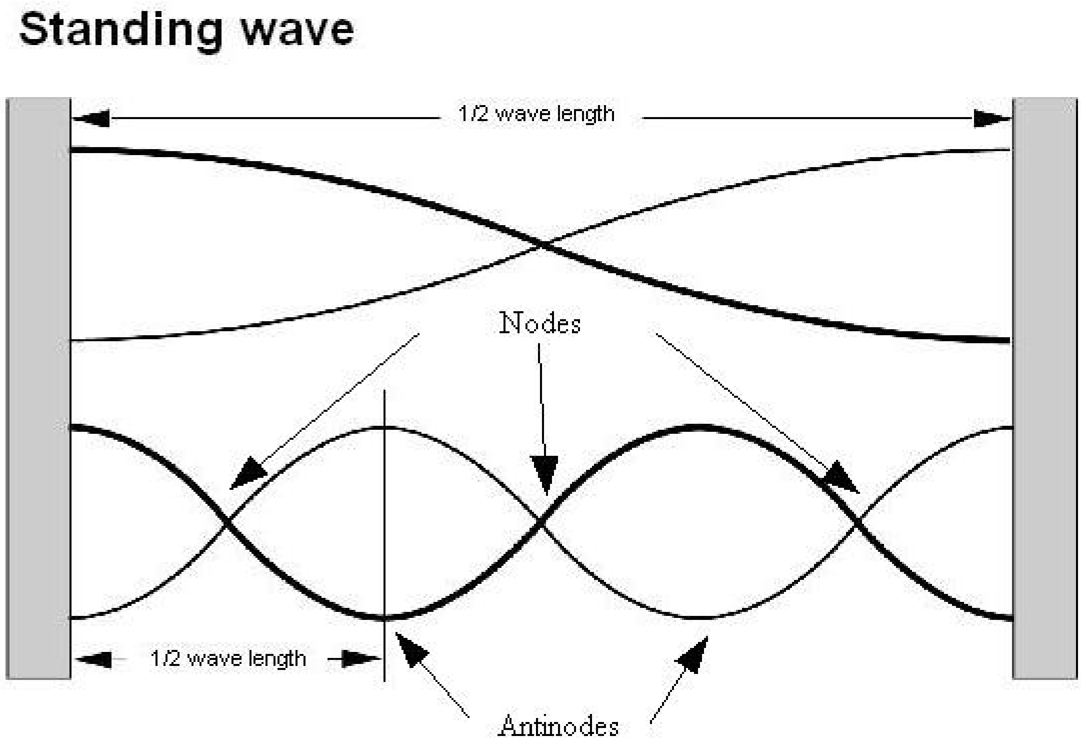
lz... height [m]

Resonant frequencies in small rooms can be found in the low frequency range. Resonant frequencies can´t be regarded as single frequencies as soon as the field turns more and more diffuse. There is a critical frequency, which depends on the reverberation time and the volume, that shows till what frequency resonant frequencies can be heard as single disturbing sounds.

$f_{g,\,H} = \frac{1000}{\sqrt[3]{V}}$ [Hz] … critical frequency for rectangular rooms with hard reflecting walls.

$f_g = 2000 * \sqrt{\frac{T}{V}}$ [Hz] … critical frequency for rectangular rooms with great absorption.

In the church, resonant frequencies can be expected in the tower since the reverberant field of the church is virtually separated from it. Due to the narrowness of the walls and the hard reflecting stone walls, $f_{g,\,H}$ would be the critical frequency.



Picture 1.5 : Resonant frequency

Acoustic Shadow:

Acoustic shadows appear as soon as an item (e.g. a column, a person...) obstructs the propagation of the sound field. An acoustic shadow can be understood as a region behind the obstructing item in which the sound pressure is lower (no direct sound can reach this region). This region can not be understood as a totally isolated region, much more like a wide region in which different sound colour changes take place.

In order to know what frequency range is affected by the obstructing item, it´s diameter must be at least 5 times bigger as the wavelength:
The shadow border can be calculated with this equation (c = 343 m/s , T = 20°) :

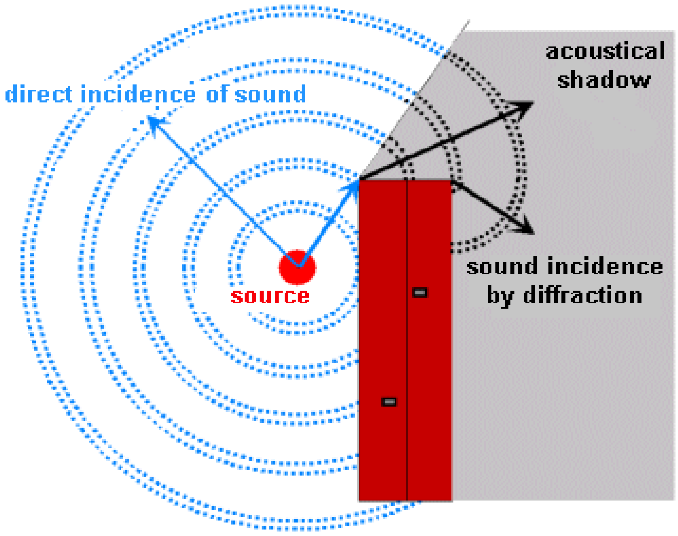
$d = 5*\lambda = 5* (c / f) ; f = 5*c / d$

Acoustic shadows can be easily showed behind the columns of a church and, in this project, there is at least one microphone position that should show this effect, which is also deeply related to diffraction.

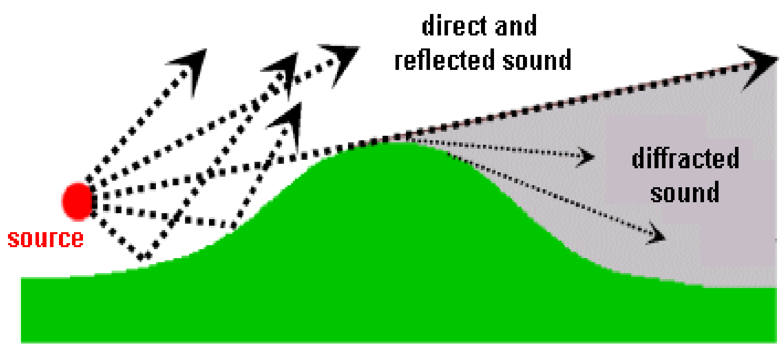
Diffraction :

If an item, e.g., a column, disturbs the propagation of the sound field, it is know that within the acoustic shadow direct signal it´s not present. Thanks to diffraction, based on the Huygens model, the acoustic shadow is also provided with energy. That´s why it´s possible to hear cars behind a building though they are on the front side .The acoustical experience is very characteristic because high frequencies are not present and the sound seems to come from far away.

It can be said that, the greater the wavelength is in comparison with the diameter of the obstacle, the less effect it has on the propagation of the wave (geometrical acoustic). If the diameter of the item as very little in comparison, the obstacle doesn´t exist for that exact frequency and the geometrical acoustic model loses it´s validity.

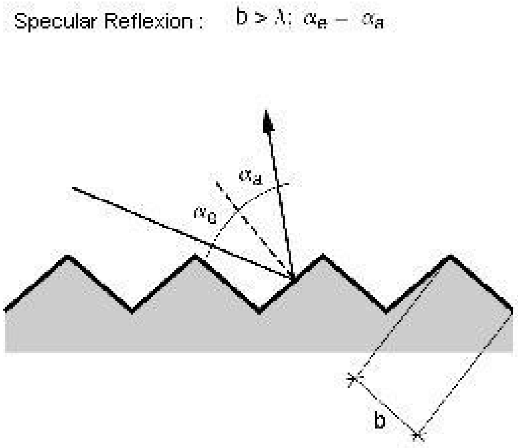


Picture 1.6 :Huygens wave propagation principle



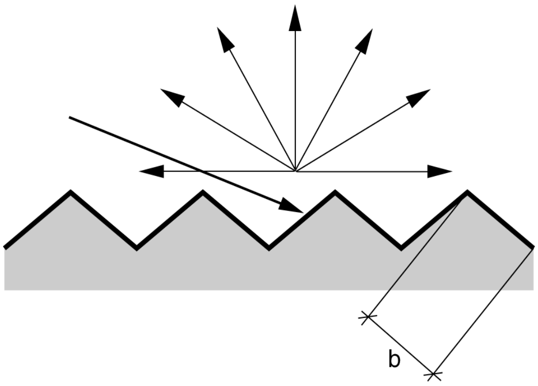
Picture 1.7 :Diffraction on an obstacule

Diffuse dispersal:
As soon as a wave collides with a surface, a certain amount of energy gets lost and the wave is then reflected. Since the energy loss depends on the frequency, it is plausible that also the way the wave is reflected depends on each frequency. Since a church gathers many hard reflecting surfaces with different angles and sizes, it seems relevant to present what kind of reflections could take place:

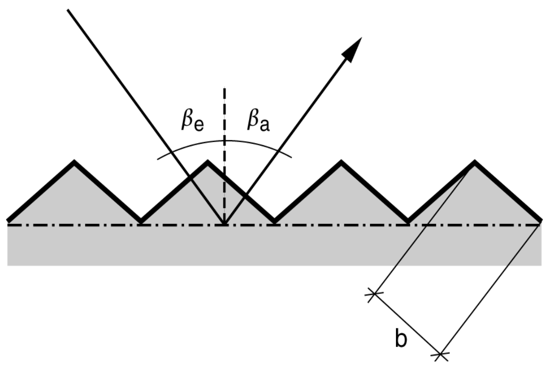


λ<b : Each sub surface reflects the wave on it´s own according to the geometric reflection lemma $\alpha_e = \alpha_a$

Diffuse Reflexion: b = λ



λ≈b: The reflected wave can be reflected in any direction, which increases the diffusivity of a sound field.



λ>b : In this case, the wall behind the sub surfaces reflects the wave according to the geometrical reflection lemma $\beta_e = \beta_a$

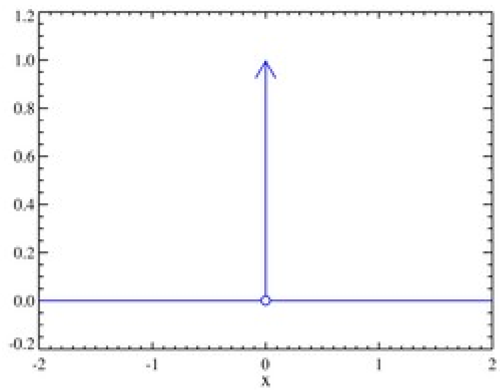
Picture 1.8 : Different dispersal types

Wave propagation can be also disturbed by items such as columns. In these cases, the effect of the item on the wave must to be explained in a different way. Therefore, is seems suitable to explain both major effects that occur as soon as an item stands on the way: acoustic shadowing and diffraction.

What is an impulse?

Dirac was the first one who ,mathematically , defined what an impulse was.
It is an infinitesimally narrow pulse in which all frequencies can be found with the same magnitude.
A Dirac impulse is generally described in this way:

$$\delta(x) = \begin{cases} \infty & \dots x = 0 \\ 0 & \dots x \neq 0 \end{cases} \quad , \quad \int_{-\infty}^{\infty} \delta(x)\, dx = 1$$

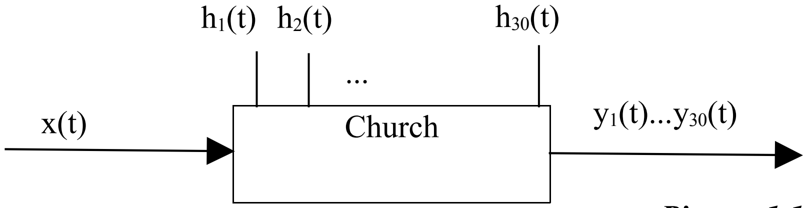


Picture 1.9 : Dirac pulse

As it can be depicted from the first formula, a Dirac pulse can not be created since a ∞- high magnitude value requires an ∞- high power in an infinitesimal little time.
The relevance of an impulse for the acoustic is the so called impulse response of a system.

Impulse response:

If a system is excited with an impulse, the system will react on a certain way, changing the impulse in a certain way. The behaviour of the system can be reconstructed if the impulse response is apprehended. So called linear time invariant systems can be fully described by their impulse response.
If the system or it´s behaviour do not ever change, their response will be always the same if the input signal (impulse) remains the same. If the system is also linear a change on the input α will also cause a change α on the output: $\alpha * x(t) = \alpha * y(t)$
For this project, 30 impulse responses were measured all along the church. Buildings can be regarded as rather linear systems as shown in Picture 1.10, but their impulse response is certainly not always the same at each place. The characterisation of the Herz Jesu Church, our system, will be carried out with 30 impulse responses:



Picture 1.10 : Church as LTI system

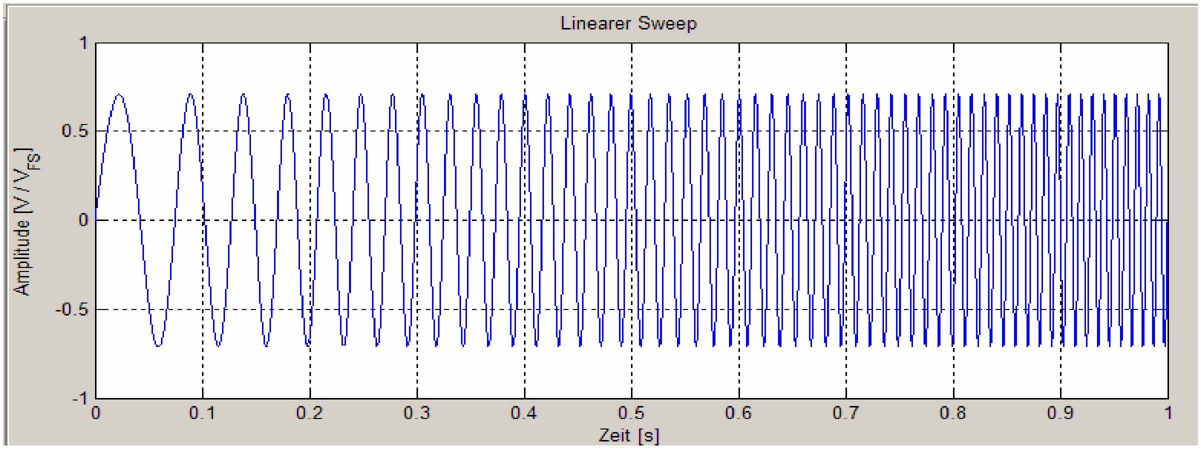
$y_k(t)$ can be also understood as the convolution of the input signal with each impulse response $h_k(t)$.
There are several ways to measure the impulse response.

Gunshots or exploding balloons try to imitate a Dirac impulse, but due to their low energy, they are not suitable for the measurement of a big building.
Another way to measure the impulse response is to excite the building with white noise- similar noise signals (MLS) or to use sweep signals.

Sine sweep:

A sine sweep is a sinusoidal test signal that sweeps through a given frequency range (0 -22 k Hz).
The main positive aspect of sine sweeps is it´s robustness against non-transient disturbances. The SNR can be improved if the measurement is carried out several times and the results are averaged, because the noise component, which represents a quantisation error, decreases after each average.
The frequency change can be linear or exponential. There are several differences between them:

Linear:



Picture 1.11 : linear sweep

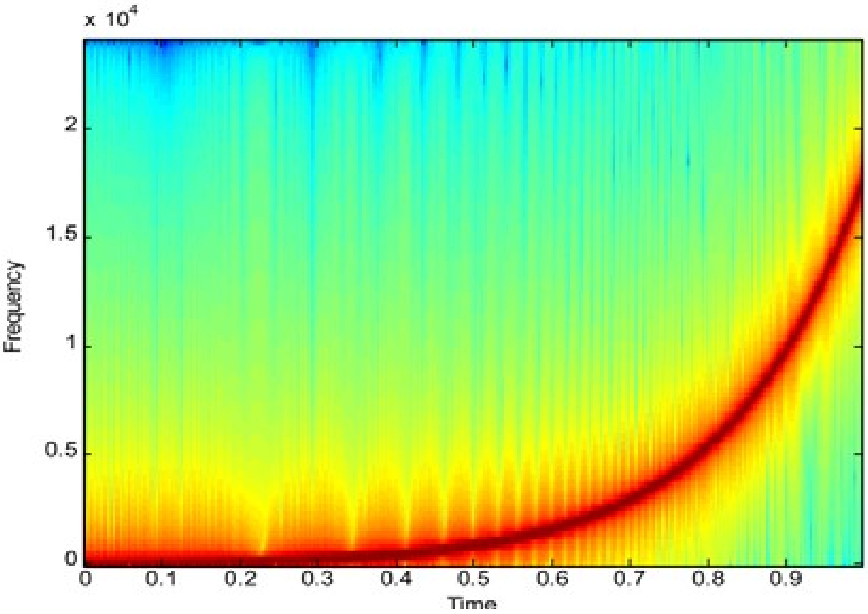
$x(t) = \sin(\varphi[t])$ ...Sine sweep
$\varphi(t) = \int \omega(t)\, dt$ … phase of the sine sweep.
The frequency varies conform to a linear function $\omega(t) = k*t + d$.
Since $\boldsymbol{\omega(0) = \omega_{start}}$, $\boldsymbol{\omega(T) = \omega_{stop}}$ **and** $\boldsymbol{\varphi(0) = 0}$

$\mathbf{d = \omega_{start}}$ $\quad$ $\mathbf{k*T+\omega_{start} = \omega_{stop}}$ $\quad$ $\mathbf{k = \omega_{stop} - \omega_{start} / T}$

$$\mathbf{x(t) = \sin\left[2*\pi\left(\frac{f_{stop} - f_{start}}{T} *t + f_{start}\right)\right]}$$

The impulse response varies much with the frequency (a difference of 40 dB can be possible). The main problem of linear sweeps is the low energy of the impulse response in the low frequency range due to the white noise spectral distribution, which is exactly the frequency range that is most problematical.

Exponential:



Picture 1.12 : exponential sweep

$\omega(t) = k^* a^{t/\tau}$

Since $\mathbf{k = \omega_{start}}$ $\quad$ $\mathbf{k^* a^{t/\tau} = \omega_{stop}}$ $\quad$ $\boldsymbol{\tau = \dfrac{T}{\log_a(w_{stop}/w_{start})}}$

$$\mathbf{x(t) = \sin\left[2\pi^* \frac{f_{start}^*\ T}{\log_a(w_{stop}/w_{start})^*\ln(a)} * \left[\left(\frac{fstop}{fstart}\right)^{t/T} - 1\right]\right]}$$

The pink frequency spectrum of the exponential sweep also leads to an impulse response which varies much within the frequency range, but the energy in the low frequency range is much higher than for linear sweeps. But, since the total energy remains the same for both signals, the energy in the high frequency range diminishes.
Moreover, exponential sine sweeps show a better noise rejection than MLS, given a signal of the same length.

Schroeder´s Integration Method

As already said, the impulse response is the spawn for all further calculations. To calculate the reverberation time out of the impulse response, the Schroeder´s method offers a solution.

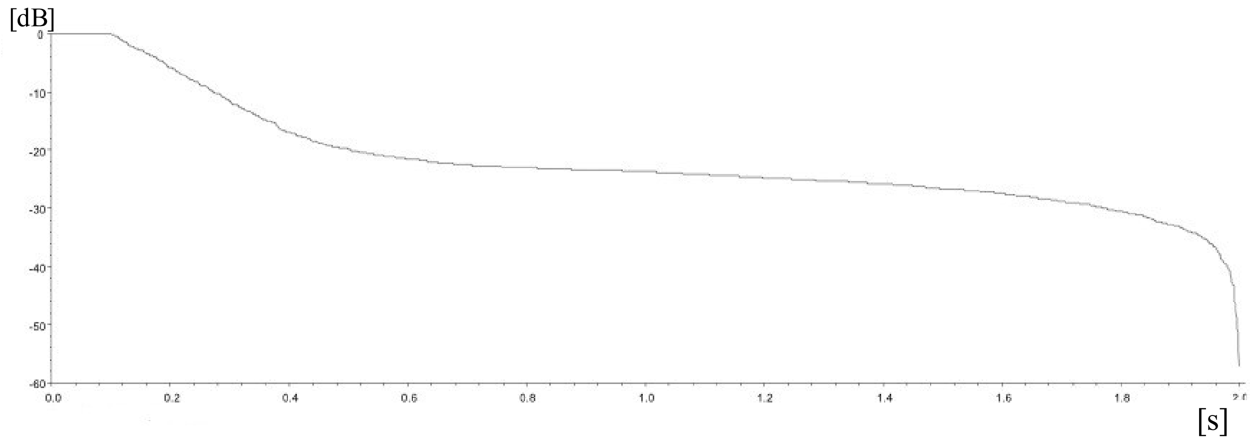
$$r^2(t) = \int_t^{\infty} h^2(\tau)\, d\tau \;\; * \text{ const}$$

$r^2(t)$... value of the square time course of the decaying noise signal.

$h^2(t)$ ... squared impulse response of the system loudspeaker room – microphone.

Since the reverberation time is based of statistical acoustics, the so called Schroeder´s frequency determines from which frequency on the sound field can be explained using the statistical acoustics model:

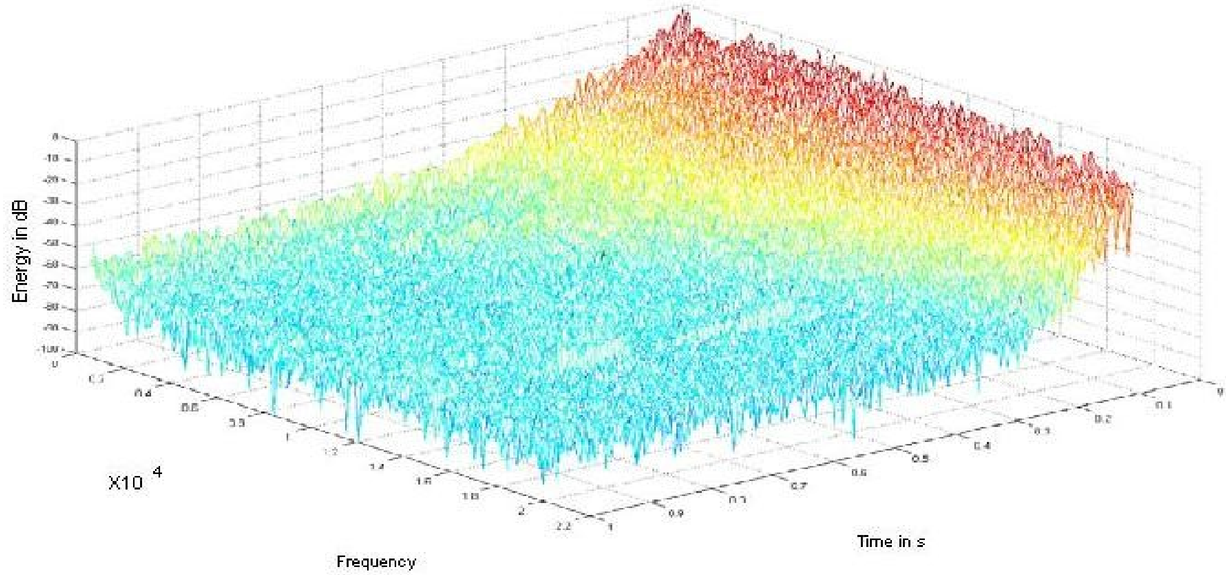
$$\boxed{f_{\text{Schroeder}} = 2000 \sqrt{\frac{T}{V}}}$$



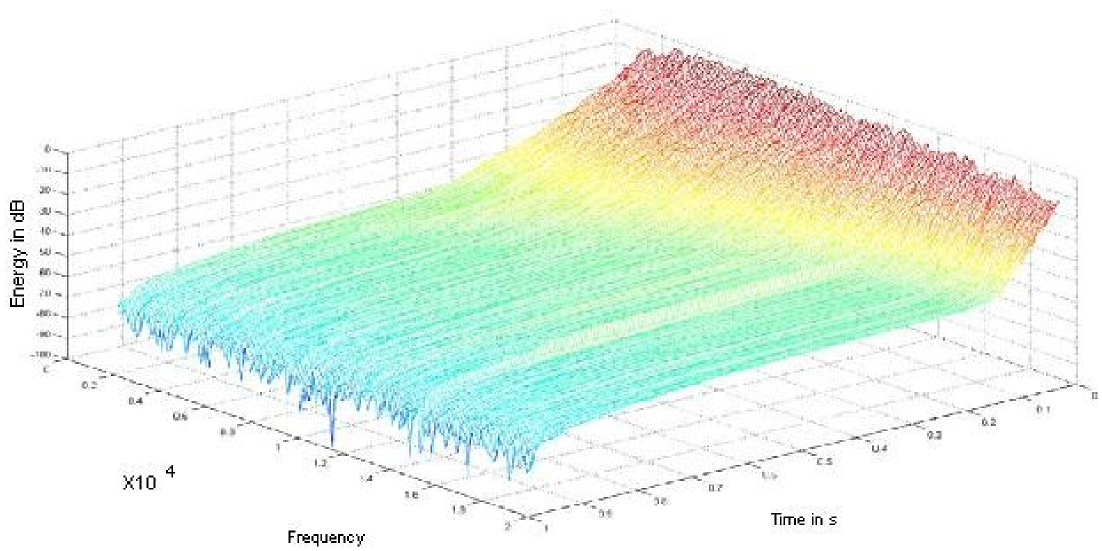
Picture 1.13 : EDC with Schroeder´s integration

This kind of measurement is free from statistical side effects but doesn´t give any information about the spectral behaviour which is so important not just for the room acoustics but also for the modelling of electro acoustic set-ups. In order to have this vital information, the calculation is carried out for each octave band.

The Schroeder integration method can also be regarded as a way to polish the early decay curve (EDC) just to have a better spectral separation at the early decay relief curves (EDR or waterfall plots).



Picture 1.14 : EDR without Schroeder



Picture 1.15 : EDR with Schroeder

Summary

In this chapter, the theoretical fundamentals on the project *" Listening to a Church "*have been described, providing the necessary knowledge to understand the decisions that were made to carry out the project in a satisfactory way. Firstly, the room acoustical basic models , statistical and geometrical model, have been presented and exemplified through the different acoustical parameters and effects, showing the relevance of the architecture on the auralisation.
Secondly, the spawn of the project, the impulse response measurement, was described into detailed so that the most elemental part of the project could be well understood: Without the impulses responses along the paths, it is not possible to carry out an auralisation.
Chapter 2 *"Measurement"* builds on Chapter 1, studying the details on the impulse response measurement procedure. Beginning with the measurement set up, it firstly falls back to Chapter 1 *"Theory"*, pointing out the differences between a reverberation time and an impulse response measurement. Afterwards, the expected and the real acoustical behaviour of the church along both paths are compared in order to put into practice the theoretical basics developed in Chapter 1 *" Theory"*.

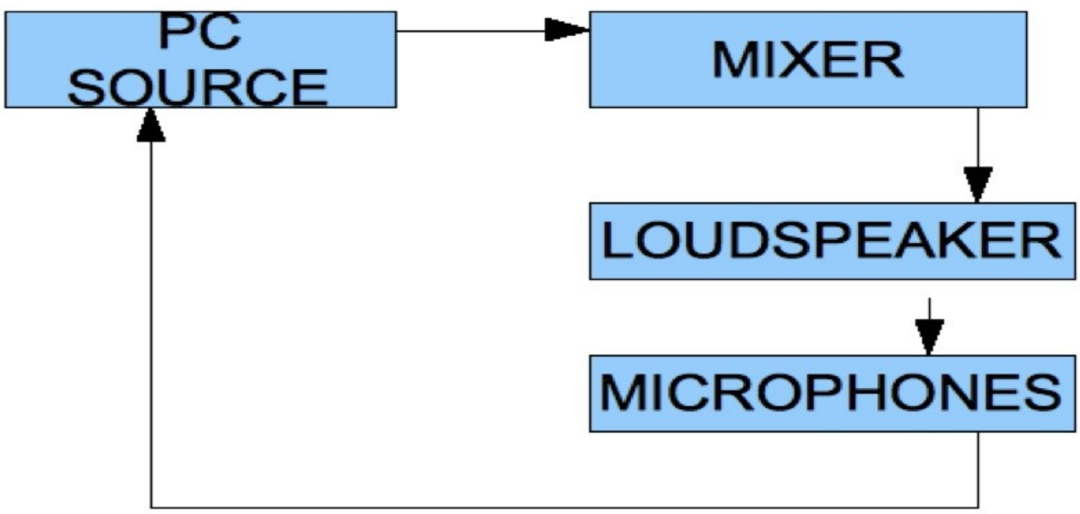
MEASUREMENT

Introduction

Chapter 2 *" Measurement "* describes step for step how both impulse response measurements were carried out: The criteria to determine the paths and the theoretical location of acoustical interesting positions are explained into detail as well as the differences between real and expected behaviour. Furthermore, the differences and similarities of RT60 and impulse response measurements were depicted in order to make a link that rounds off the theoretical basics presented in Chapter 1 *" Theory"*. In order to ease the work of other students, some tricks and suggestions have been gathered together
The measurement was carried out using WinMLS , an acoustical measurement software which is able to calculate the impulse response and,consequently, the reverberation time.
In order to visualize the sound changes along each microphone path, different impulse responses have been compared using WinMLS plots of the impulse responses, which contain a normalized data plot as well as an frequency response magnitude plot to show in how far the shape can provide a rough idea of what has been measured. After both measurements have been described, the last part of the chapter deals with the causes for the different frequency ranges at the first microphone position of each path, regarding also it´s possible consequences on the auralisation.
By the end of chapter 2, all relevant aspects concerning the measurement have been described: The measurement set up and procedure and a complete evaluation of the measured data regarding the expected acoustical effects.

Set up:

The measurement set up is shown on Picture 2.1. WinMLS,the measurement software, generated the test signal (sine sweep) which was the amplified with a mixer and the played by a loudspeaker (source). This way, the whole church was excited with the same excitation signal.
The microphones along both paths are necessary to record the reflected signal, that means, the one changed by the architectural conditions of the church. Since microphones only can "see" pressure fluctuations, this information has to be transported to the computer so that WinMLS calculates the impulse response.



Picture 2.1 :Measurement set up

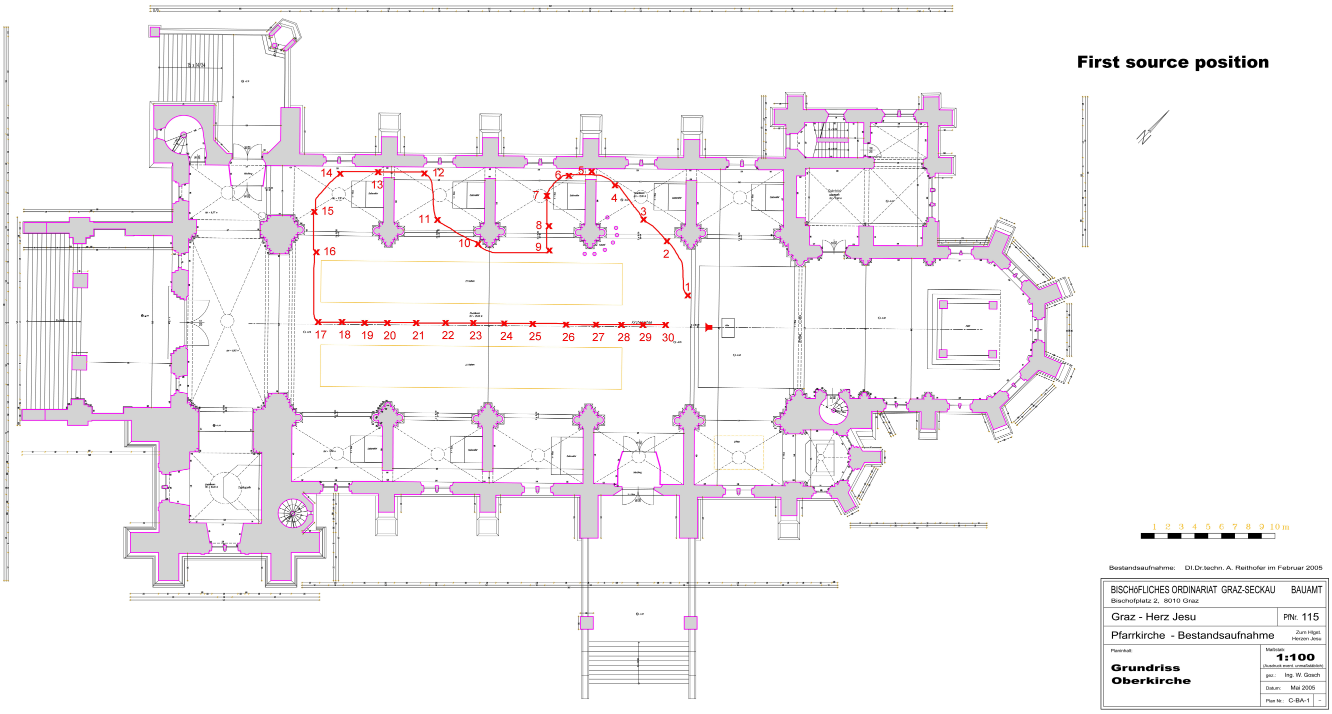
The microphones used were two omnidirectional AKG 480.
The loudspeaker is a dodecahedron loudspeaker, property of the Joanneum Research Center. This special loudspeaker type irradiates signals in all directions, so that the whole building can be excited in the same way, which is very important to gain truthful results.
We used a Firefox 400 microphone pre amplifier and a Mackie mixer in order to regulate the loudness of the test signal.
As explained in the abstract, two source positions were selected and each attached to a different path across the church. The first source position (SP1) was placed at the altar while the second source position (SP2) was placed at the organ.
The microphone position density and the distance between microphone positions were chosen in order to capture the whole acoustical behaviour of the church.
The measurement took place at night in order to avoid disturbing noises which would have spoiled the results. Moreover, the church had to be empty in order to capture the very essence of the church.
Before measuring, the microphones as well as the loudspeaker, were calibrated.

In how far does a reverberation time measurement differ from ours?

Firstly, it should be remarked that this project didn´t intend to measure the reverberation time of the church. Nevertheless, it seems sensible to point out which reverberation times can´t be regarded as valid, yet we have compared the theoretical and the measured ones.
The height of the microphones was always 1,2 meters above the ground, the same as in a common reverberation time measurement: when people seat, the ear is placed about 1,2 meter above the ground.
The first microphone position stands within the reverberation radius, which is not permitted in a reverberation time measurement because the direct energy density is bigger than the stationary energy density;
Since the reverberation time is defined as the time a signal needs to loose 10^6 of its energy density (-60 dB attenuation), the microphone needs to stand out of the reverberation radius in order to record only the reflected signal.
Furthermore, the distance between microphones should be bigger than 2m in order to avoid the cancellation of the lowest frequency that should be captured ($f = c/\lambda = 340/4 = 85$Hz) and this wasn´t always followed (positions 3,4).
Finally, microphones should be at least 1 meter far away from any reflecting surface in order to avoid cancellations.
This rule was the most broken one, then, in order to obtain interesting acoustical effects, this rule had to be broken (SP1 : position 5 ,6,10 ,12,13,14 , SP2: 3-10)
Indeed, the second source position was placed at the organ in order to obtain interesting impulse responses all along the stairway!
Summing up, it can be said that nearly all impulse responses which provided the raw material to present interesting effects break at least one of the rules that regulate a valid reverberation time measurement.
Tricks:

- Plan your schedule as precise as possible in order to avoid improvisation. As soon as the measurement starts ,the microphone positions and the logistic schedule should be clearly fixed.
- Test the measuring chain at least once before start measuring, because afterwards it´s not always easy to know if the results are suitable or no.
- Taking a look at the reverberation time after each measurement can show if the measurement has been carried out properly.
- Check the result of the measurement as soon as possible to have a chance to carry out a further measurement if the results are not suitable.

Source Position 1:



Picture 2.2 : First source position walk through

This source position is very interesting because music or at least speech signals are generated from this very point (priest, chamber music...).

Expected acoustical behaviour:

During the first 16 microphone positions, the music was expected to turn more and more indirect due to the rising percentage of indirect (that means, reflected) signal and to loose clarity and brilliance. The second half of the microphone positions should show how the music turns louder and more brilliant due to the rising percentage of direct signal as soon as the distance between source and microphone diminishes.

Interesting acoustical microphone positions

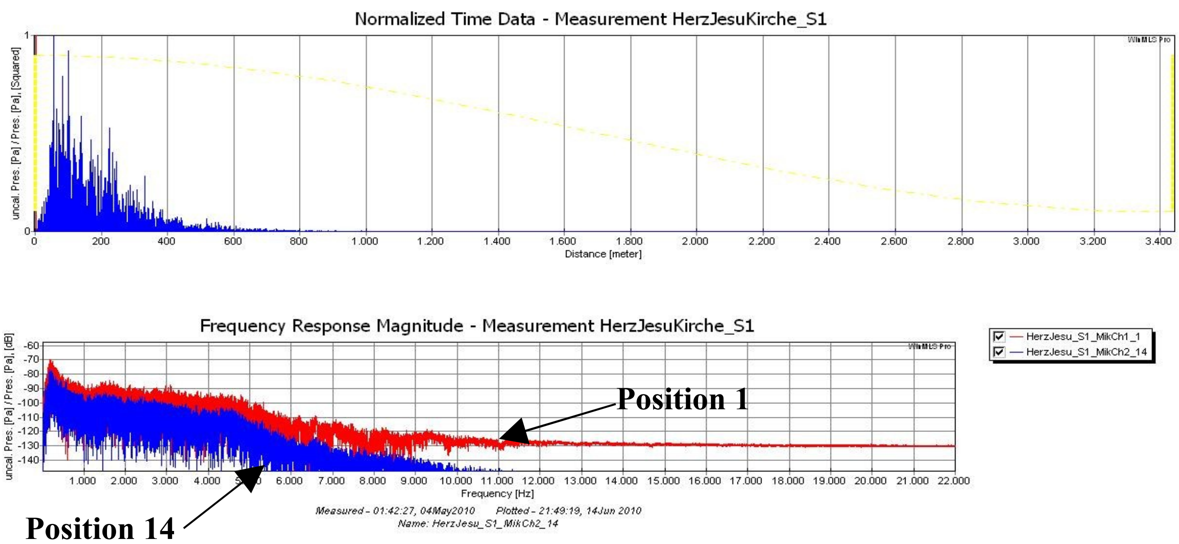
Among the 30 microphone positions, there are at least 3 acoustical interesting spots: Position 5 (Picture 2.3.1 Page 24), position 10 and position 17 (Picture 2.3.2 Page 24).
Position 5 is a small passageway , which is very narrow, that connects two chapels. It was expected to hear a very indirect sound mixed up with an unique behaviour, allowing a defined resonant frequency. Position 10 was intended to be an example for acoustic shadow and diffraction,
Position 17 is the first microphone position just in front of the source and therefore it was expected for the music to turn more brilliant and louder, because the percentage of direct signal at that spot is much higher then in the last positions.

Picture 2.3.1: Microphone Position 5 and 6

Picture 2.3.2: Microphone Position 17 and 18

Real acoustical behaviour and non-expected effects

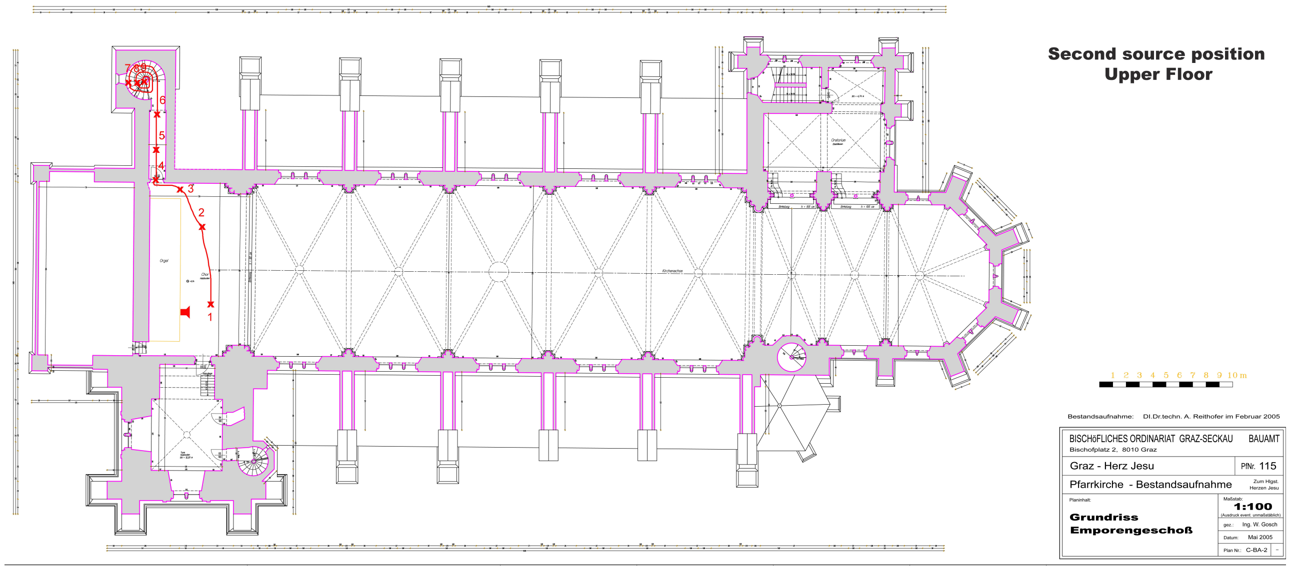
As expected, the clarity and brilliance diminished as the microphone positions were more far away from the source, but the influence the chapel had on the impulse response at the positions 3 and 4 was much bigger than expected. A possible explanation for this phenomenon could be, that the dimensions of this chapel are much lower than the ones of the church, so, regarding the chapel as a closed volume, it would be valid to assert that the test signal used (mend to be for the church) excited the chapel in a very characteristic way.
In addition to this point ,the low absorption factors of the absorbing surfaces surely played a key role in the development of this acoustical field, because otherwise it would have been difficult to explain why high frequencies are so outstanding at this point of the measurement.
As expected, point 5 presented a very particular impulse response.
Afterwards, the music turns brighter and lower, though the column presents an exception: the music is certainly louder and it´s colour changes (Diffraction, acoustical shadows).
It should be possible to read out this kind of information in the impulse response since the impulse response contains all information needed to characterize a place in the church.
Two impulse responses (at positions 1 and 14) have been compared.



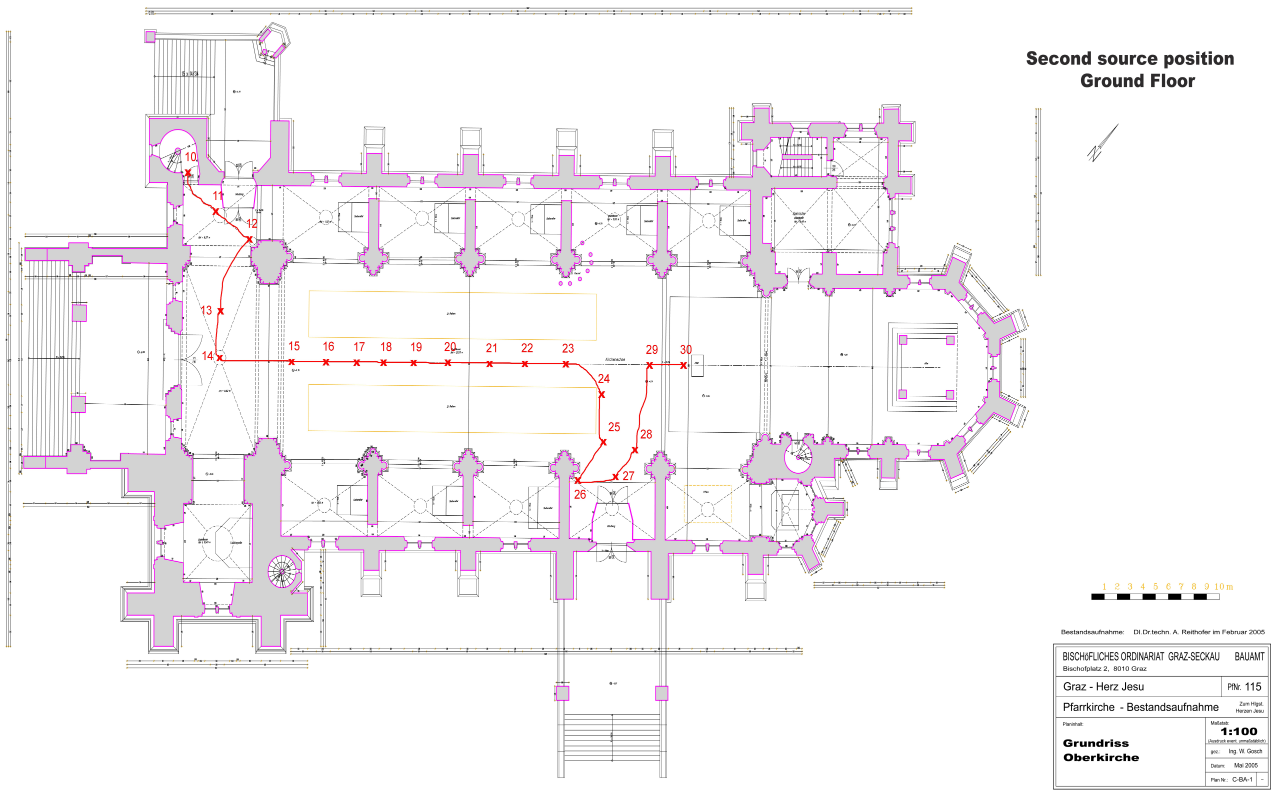
Picture 2.4: Comparison between position 1 and 14

The frequency response magnitude plot shown at picture 2.4 shows how strong each frequency of the impulse response is. It is measured in dB (normalized) .It ranges from -60 dB till -140 dB. Predictably, whereas the impulse response at microphone position 1 shows spectral elements throughout the whole frequency range (0 – 22kHz), the impulse response at microphone position 14 shows spectral elements till 11 kHz. Since position 1 was standing just in front of the source, it is plausible that this behaviour is right, because the direct signal swept through all frequencies from 0 – 22kHz!
On the way to microphone position 14, the test signal lost energy in form of air absorption and reflections. That´s why the SPL values of both impulse responses differ as the frequency increases.
After the convolution with the dry signal, the expected sound at position 14 should be less brilliant and lower than at position 1.
The assumption for microphone position 17 turned out to be wrong because the music did not turn more direct. Calling back the theoretical background explained in Chapter 1 *(“Theory”),* microphone position 17 is about 50 meters away from the source, that means, about 16 times the reverberation radius (~ 3 meters). Since the reflected signal gains in presence the larger the distance to the source is, it is not possible for the direct signal to have an overwhelming presence at microphone position 17.

Source position 2
This source position is very important, because the organ is placed there and organ concert recitals are very common in the Herz Jesu Church.



Picture 2.5.1 : Second source position Upper Floor walk through



Picture 2.5.2 :Second source position Ground Floor walk through

Expected acoustical behaviour:

The first two positions should show a loud, brilliant, clear signal which contrasts much with the one heard through the whole tower (stairs): this impulse responses should present an absence of "room information " . It was also hoped to find some resonant frequencies.

As soon as the tower is left behind, the music should turn louder and, certainly, the impact of late reflections should be greater, providing the listener with " room information ". While moving towards the altar, the impulse responses should gather more and more room information. The music should also win in brilliance.

Interesting acoustical microphone positions

Microphone position 4,5 and 6 (Picture 2.6.1) are very interesting because they are thought to show very well how quickly the direct signal influence diminishes (leaving the reverberation radius) and how the special geometrical structure of the upper floor influences the impulse response of the church.

Positions 10 (Picture 2.6.2), 11, 12 should show how the sound changes outside the tower.

Position 15 is intended to show how the music is heard if the listener is standing at the place where the audience uses to sit during a concert. Position 26 (Picture 2.6.3)should show how the music is heard behind a column if the source is placed at the organ.

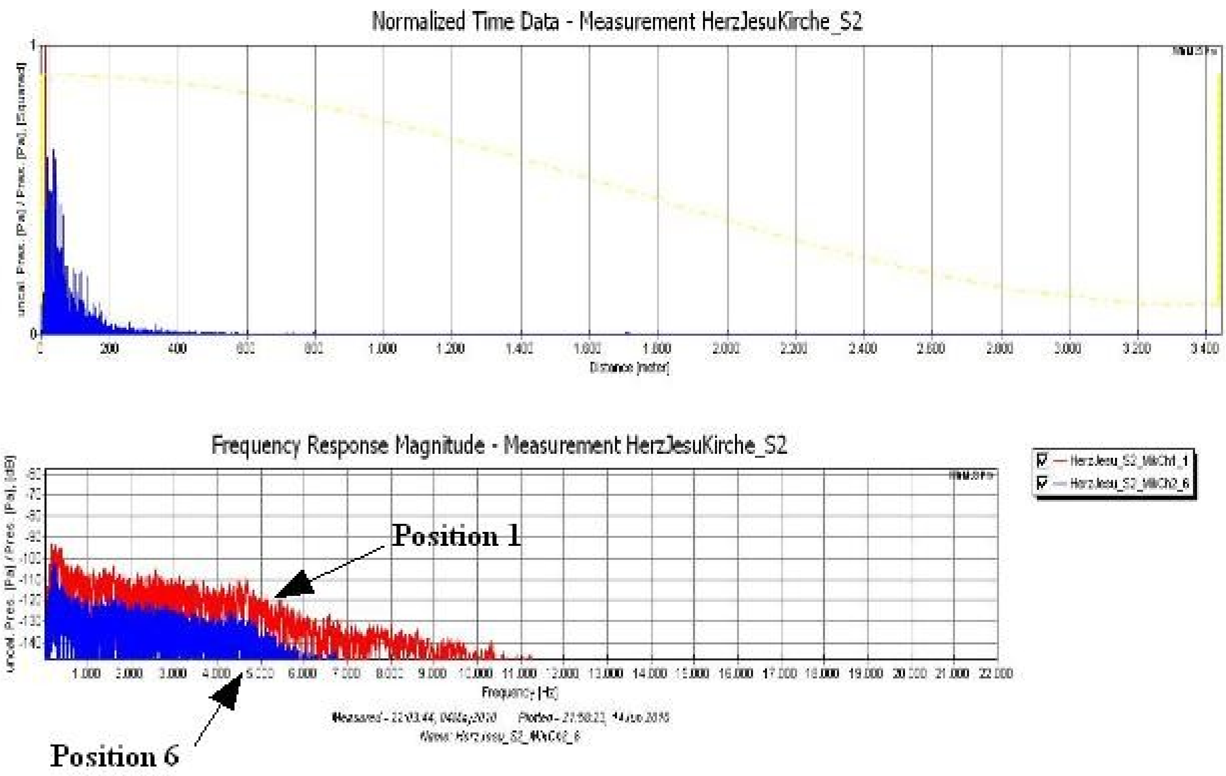
Picture 2.6.1 Microphone positions 5 and 6

Picture 2.6.2 Microphone position 10

Picture 2.6.3 Microphone position 25 and 26

Real acoustical behaviour and non expected effects

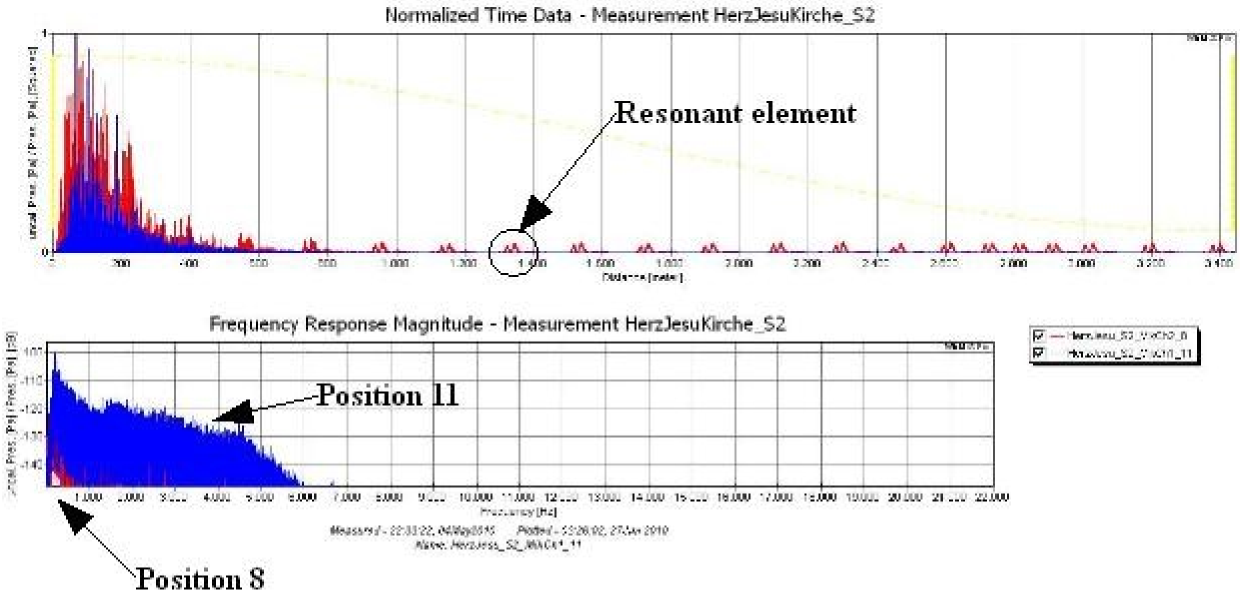
The influence of the direct sound indeed diminished, but quicker as thought because at position 3 it can be clearly heard that the loudness is much lower than a few seconds before.
As expected, late reflections didn´t have any effect on the impulse responses in the tower.
As for Source Position 1, assertions should be readable from the impulse response:



Picture 2.7: Comparison between position 1 and position 6

As expected, the SPL of the impulse response at microphone position 6 is lower than in front of the source and shows a much lower frequency range, just about 6000 Hz (Picture 2.7 Frequency Response Magnitude). That means that at the beginning of the spiral staircase, the frequency range has gone back about 2 octaves!
This dramatic loss of frequency range can´t be a consequence of dissipation because there isn´t such a significant air volume between the source and position 6.
If the sound field could be compared to a water stream, the geometrical structure of the upper floor wouldn´t be so relevant, but since a sound field can´t make a " turn " as water does, the only way to advance is through reflections. As explained in the *"Theory "*, a wave can be reflected in different ways in dependence of its wavelength and the structure of the surface where it´s reflected. Due to the many reflections between the source and position 6, a wide frequency range can be rapidly suppressed. Moreover, late reflections of the church do not have any effect on the impulse response inside the tower. Which effect will it have for the auralisation has been explained in the second great chapter of this project.
Surprisingly, resonance frequencies could be easily heard although the played notes had little duration. The narrowness of the walls and the low absorption coefficients of stone could explain this effect, which is particularly clear at position 8 : a sustained note (Organ point), gains in energy rapidly, getting annoying for the listener, which is characteristic for resonances.

The frequency response magnitude and the normalized time data plot on picture 2.8 show how resonant frequencies change the results of a measurement.



Picture 2.8:Comparison between position 8 and 11. Resonant frequencies.

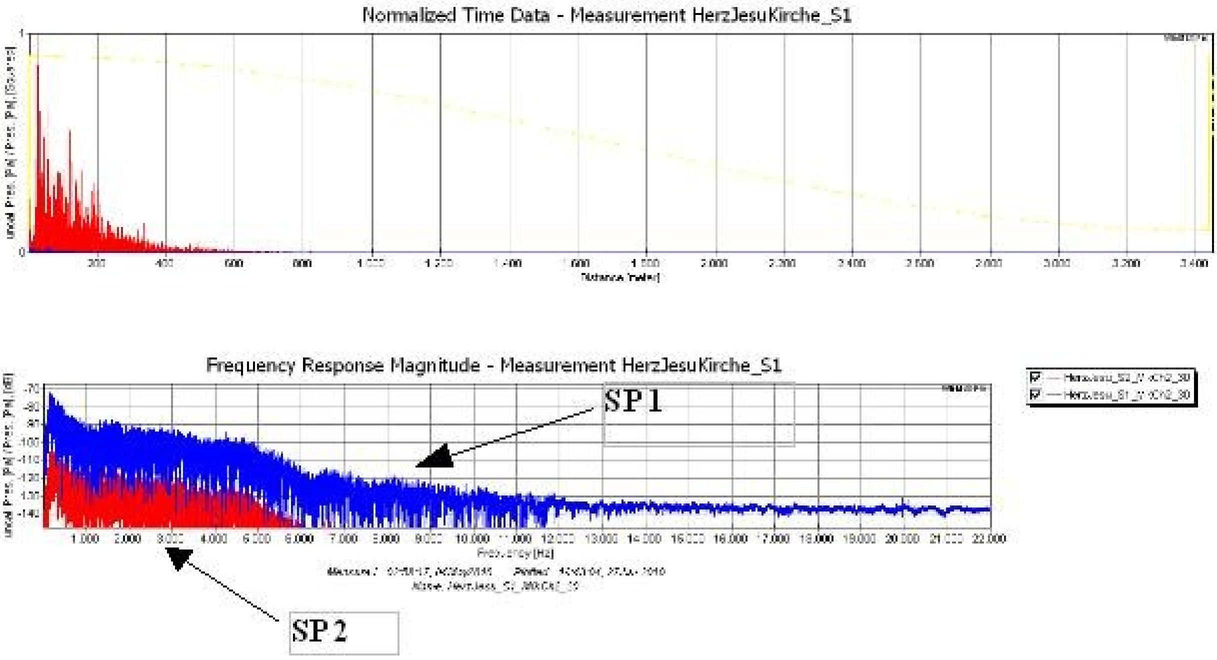
The first diagram on Picture 2.8 shows the normalized time data for two different microphone positions : 8 and 11. According to the maps shown before, microphone position 8 is placed in the tower (spiral staircase) and microphone position 11 is placed just outside the tower.
Since resonant frequencies were expected along the tower, the assumption is correct, as it can be seen at the plot.

Surprisingly, the difference between being in the tower or intermediately out of it is not as big as expected, though after position 12, the difference can be better spotted. This unexpected behaviour of the church could be related to the fact that, at positions 10 and 11 (at both sides of the tower door), the music is very quiet and the loudness is not high enough to perceive the changes.
Nevertheless, the next impulse responses gather more and more room information, so the sound turns brighter and more powerful.

As expected, the impulse response at position 26 (behind the column) differs much from the previous ones: the music is not as powerful as it was and the loss of high frequencies is also remarkable.

Importance of the reverberation radius

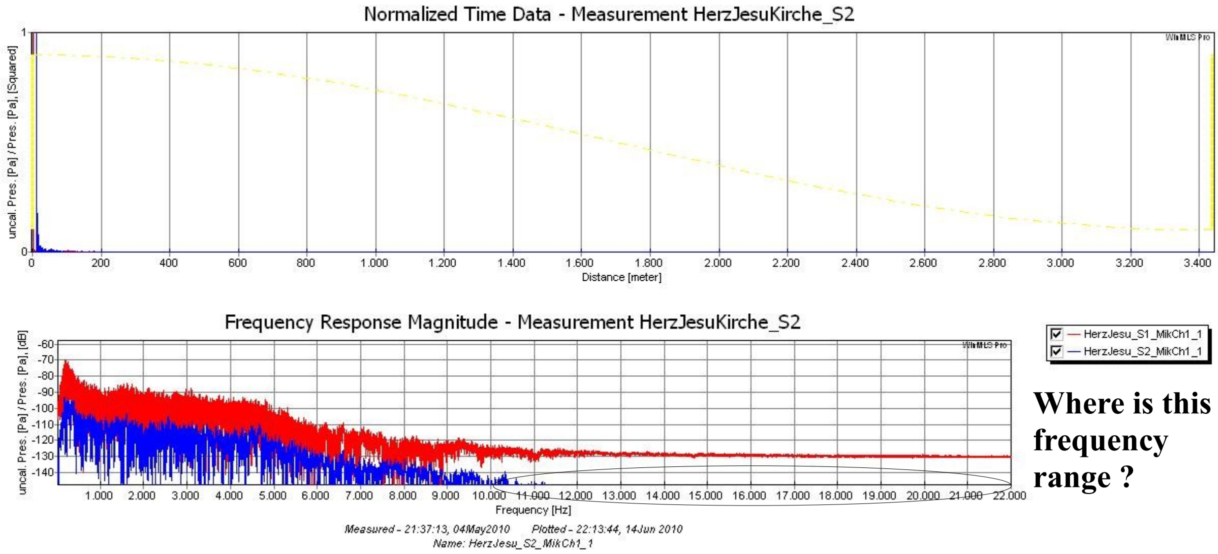
In order to show how important the reverberation radius can be, microphone position 30 has been plotted for both source positions.



Picture 2.9 : Importance of the reverberation radius

It can be seen on the normalized time data plot contained in picture 2.9 that an impulse response within the reverberation radius shows very little information about the church, whereas the one outside the reverberation radius provides much information about the church . Taking a look at the blue curve (impulse response attached to source position 2) shown at the frequency response magnitude plot on Picture 2.9 ,it is almost like the frequency range of the impulse response measured at position 6 , just before the spiral staircase begins. The fact, that the frequency range suppressed by the whole church is almost equal to the one suppressed along a little, geometrically awkward way is certainly a good example of how complicated estimations can be made about the acoustical behaviour of a church or, generally, about a big building.

What could went wrong ? - Different frequency ranges for both first microphone positions



Where is this frequency range ?

Picture 2.10:Frequency range comparison at microphone position 1 for each source position

It was expected to have very similar frequency response magnitudes just in front of the source, but at least 2 octaves can´t be showed at the frequency response magnitude plot of source position 2 (Picture 2.10, blue curve) . The difference of about 30 dB between both signals at the start is certainly a prove for a wrong done loudspeaker calibration. Because the SPL value that was given to WinMLS was acceptable (80 – 85 dB), the distance towards the loudspeaker was much to little. Therefore, the loudspeaker didn´t irradiate enough power in the higher spectral range.

On one hand, the microphones could have been wrong aligned. They pointed towards the church and not towards the loudspeaker (source),so,consequently, the recorded signal at microphone position 1 (SP 2) had to much "room information", that means, it was much to distorted by the architecture of the church. As it has been explained at Chapter 1 *(" Theory ")*, higher frequencies can be better absorbed than low frequencies so it seems plausible that "high" frequencies (>10000Hz) are not present.

On the other hand, the normalized data plot shows quite similar shapes for both impulse responses. That means, that both responses provide little information about the church, which is typical for microphone positions within the reverberation radius. If the microphone at source position 2 would have pointed at the altar, the building information seen in the impulse response should stand out in the plot. Consequently, a wrongly carried out loudspeaker calibration seems to be the main problem source at this source position. A possible consequence for the auralisation could be, that the convoluted signal is much to low to perceive and changes in the music.

Nevertheless, the impulse response measurement seems to work, because notorious differences can be at least seen. Thanks to the auralisation, this differences can be heard.

In order to make these effects stand out more clearly and show the behaviour of the church in a more clear way, the music, as well as the impulse responses, have to be processed.

Music signal processing can be easily carried out with so called audio editing software's (Samplitude, Cubase, Logic, Sequoia, Pro Tools...) and impulse response processing is commonly carried out in Matlab .The report on the fulfilment of these tasks can be found at the second main chapter of this project : **" Studio work ".**

Summary

Along Chapter 2 *“Measurement”* , there have been 2 major points: the analysis of the microphone positions and the evaluation of the measured impulse responses: Each microphone path has been firstly described regarding especially to the criteria lying beneath the determination of the microphone positions and afterwards, the measured data was compared to the expected behaviour.
The comparison between real and expected behaviour has been always done using plots of the impulse response to prove the differences mentioned in the text.
The last bit of this chapter intends to show why the measurement for the second source position (organ) went wrong and the possible causes.
Now, the raw material to start an auralisation is available: The impulse response measurements along both paths have been successfully carried out and the results have been recorded by WinMLS; the measurement software.
Within chapter 3 *“ Studio Work “* , the auralisation process for an empty and a full church ,as well as the video processing, will be described, pointing out the differences between natural and processed auralisation. Consequently, the effects used on the auralisation in order to make the acoustical effects stand out more clearly will be single discussed.

STUDIO WORK

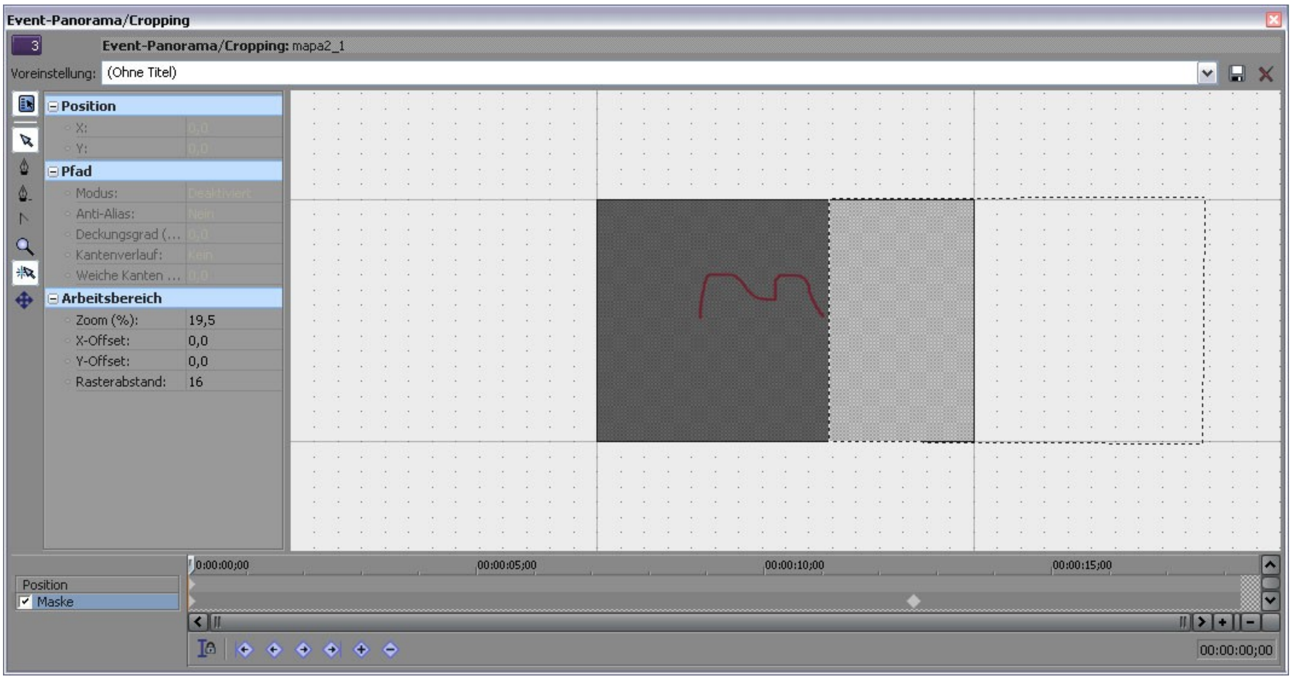
Introduction

The impulse response measurements have delivered two sets of 30 impulse responses along the Herz Jesu Church for two different paths, each attached to a different source positions: One at the altar and another one at the organ. How the impulse response measurement has been carried out and further detailed information about the results of the measurement can be found in Chapter 2 *“ Measurement “.*
Chapter 3 (*“Studio Work”*) is intended to describe the audio processing in Samplitude and the video processing in Vegas in a precise way. Auralisations can be done using any audio editing software (Sequoia, Pro Tools, Logic, Cubase, Samplitude...) thanks to an effect called *Room simulator*, which allows the convolution of a signal with an impulse response in a very easy way. This project consists of 4 auralisations: 2 for each path, simulating an empty and a full church.
Each auralisation has been attached to a video in order to provide an audiovisual stimulus. Samplitude was the chosen software since it´s *Room simulator* shows a good quality. The auralisation itself was carried out at the IBK studio of the TU Graz. The auralised signals (“ dry “ signals) were chosen regarding to their little information about the recording room so that the impulse responses from the Herz Jesu Church do not get mixed with the ones of the recording room, spoiling the auralisation.
A trio for violin, cello and piano was chosen for the round walk attached to the source position at the altar and the *Fantasie in g-moll 542BWV* from J.S.Bach was the piece chosen for the one attached to the source position at the organ.
The question, if some kind of filtering is necessary to remark the acoustical effects, has led to two opposite concepts: natural auralisation and forced auralisation.
In other words: the auralisation without any kind of signal processing and an auralisation which has been edited using filters (Low pass, High pass filters...) in order to remark certain effects.
The auralisations carried out for this project are forced auralisations and therefore all kind of changes done to the natural auralisation have been reported and explained.
The explanations always aim to show the criteria lying beneath the election of the filters and their critical frequencies, so that the insertion of the filters can be well justified.
By the end of Chapter 4, the audio and video processing have been explained, taking special care of explaining and justifying the diverse filtering methods used.

Photoshop+Vegas
The software used to video edition is Vegas Pro 9.
This program has been used to attach the length of the audio with the length of the video.
The first step is to draw the walkthrough in the map with Photoshop. A new layer has to be created above the map and with the brush function, the walkthrough can be drawn (here in red). The drawn line is saved with .png extension. The project should be in "film and video" format.

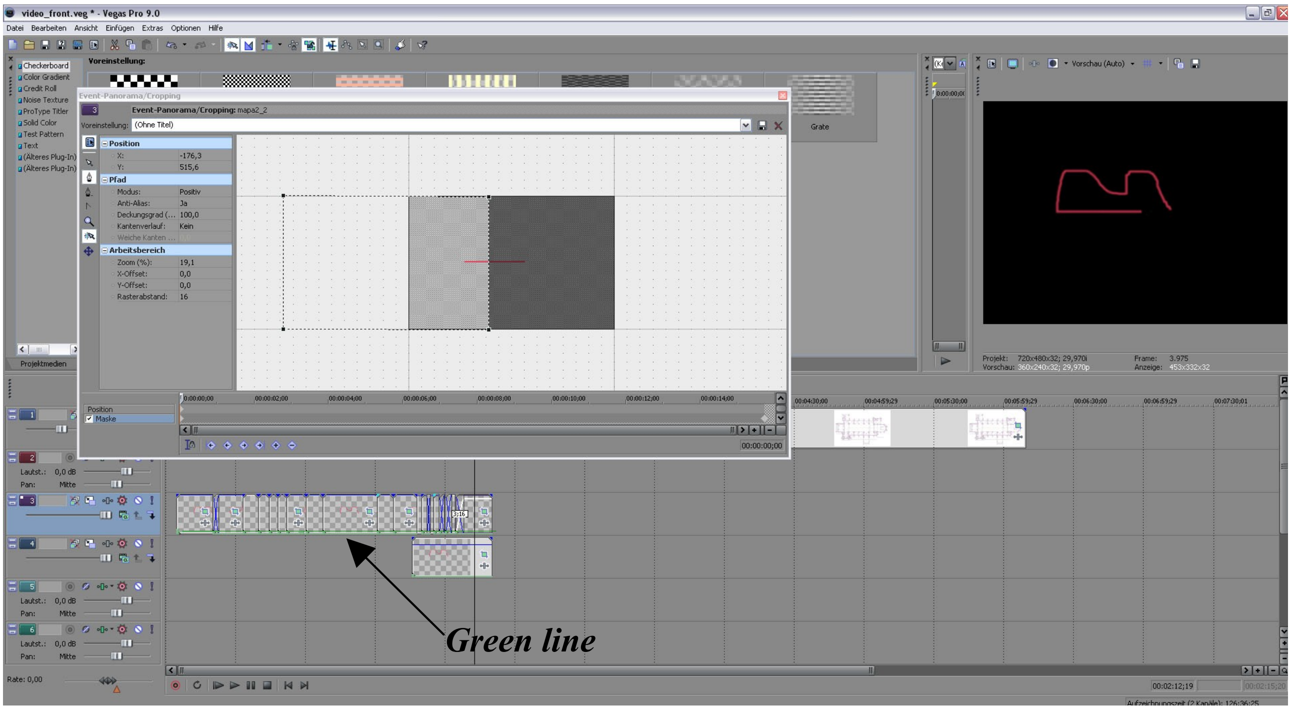
With the map.jpg and the draw.png we can start working with Vegas Pro 9. The lower part shows the work area with multiple tracks. The video is loaded in the first one , the .png file in the second one and in the last one the .jpg file. To synchronize the drawn line with the map , a coincidence with the appearance output has to be achieved, making clic in the event Panorama/Cropping(right part of the images) of our files and right clic--> "Outputverhältnis angleichen ".

The next step is creating a mask in the direction the drawn line shall move over the map. To create a mask ,"Panorama/Cropping" should be selected in the .png file and the "Maske" option should be chosen. With the tool "Verankerungswerkzeug" four points are drawn, making a square in the right of the line so the movement will be from right to left (Picture 3.1.1) .



Picture 3.1.1: Mask to get a movement right-left

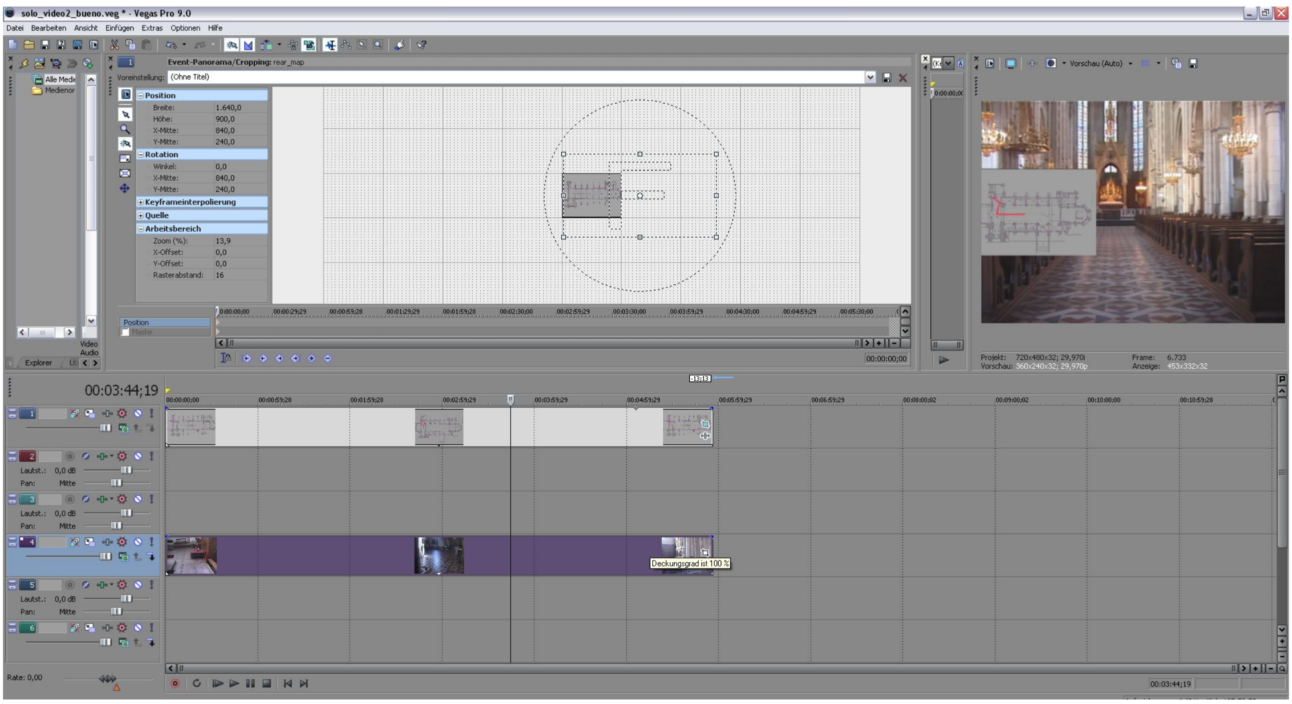
If the movement should go in the opposite direction ,the mask should be made on the left (Picture 3.1.2).



Picture 3.1.2: Mask to get a movement left-right

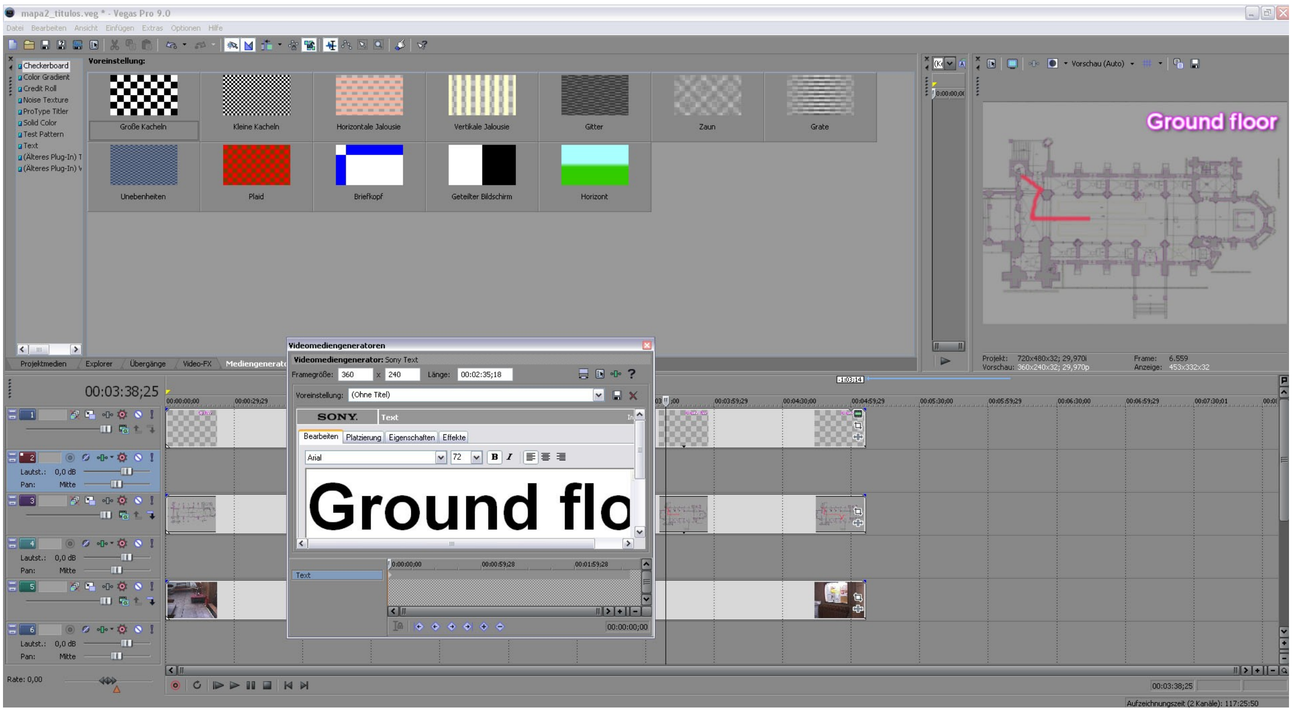
If the reproduction velocity has to be changed--> "Hüllkurve einfügen/entfernen", "Geschwindigkeit" .A green line should appear on the track.Moving the green line, the velocity can be diminished or increased.

To load an image in a video (Picture 3.2) ,the image track should be loaded in a track above the video.
==> "Event – Panorama / Cropping " and move the "F" towards the position where the image should be placed.



Picture 3.2 :Insert an image in a video

If the video has to be seen behind the map, the opacity of the map should be diminished moving the white line on top of the map.
To add comments ==> " Mediengeneratoren"

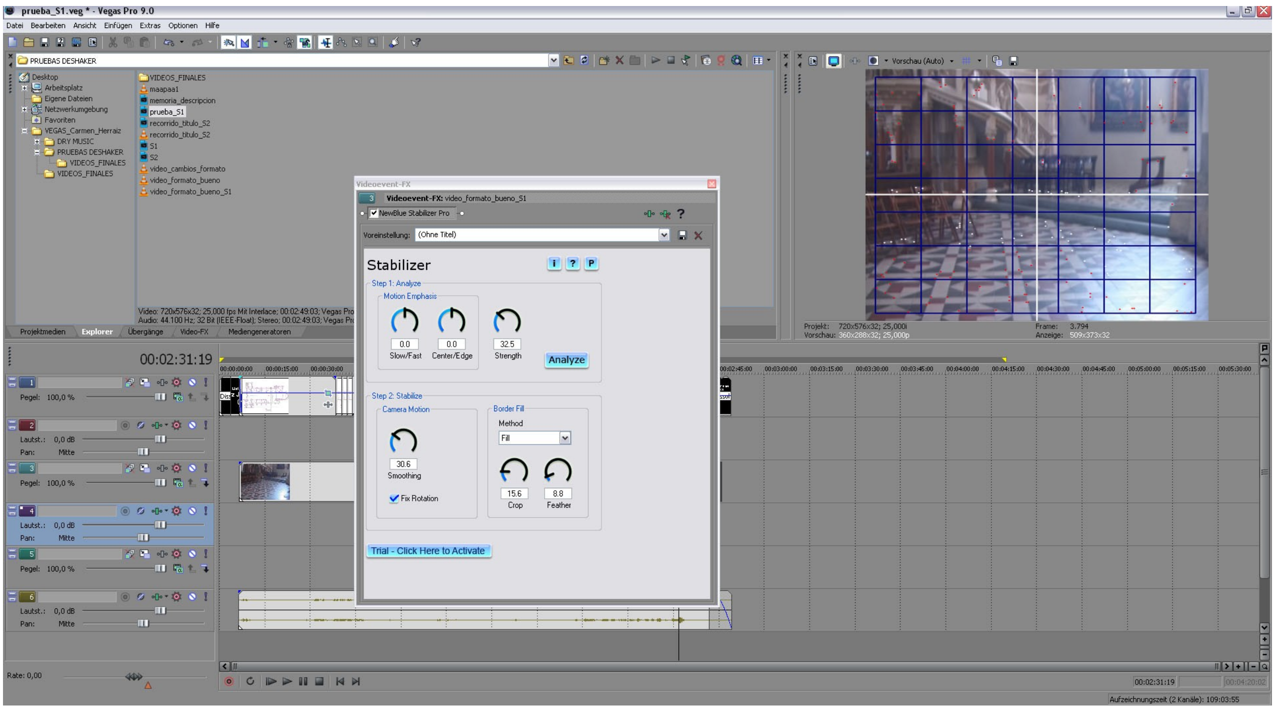


Picture 3.3 : Add comments

At last, all layers must to be rendered together to one track ==>"Datei","Render als..." and save in the desired format.

Tricks:

Any tripod or steady-cam has been used to record the videos, therefore a stabilizer has been used (Picture 3.4). “Deshaker” is a Plug-in for Vegas that stabilize an image with a filter looking for differences between a picture and the following image pixel to pixel. The filter modifies each of the images based on pixel changes.



Picture 3.4 : Deshaker to stabilize the image

The resolution of the videos has been changed to 25fps to reproduce less vibration due to the camera movement.
When used, the deshaker should be only applied to the video. Afterwards, the map cad be attached and rendered. Otherwise, the map won´t be at the right place after rendering the video.

Music:

For the first source position, a trio (piano, cello, violin) was chosen as dry music signal. The dryness of the signal is from vital importance since the auralisation of the church has to show the acoustical behaviour of the church. The piece was recorded at the IBK – Studio at the TU Graz, a rather dry studio. It was recorded by audio engineering students and Mr. Stevcic.
For the second source position, *Fantasie in g-moll 542BWV* from J.S.Bach was chosen.
The recording was done in a 50 m^2 large rehearsal room in the KUG. The audio engineer that carried out this recording is called Ferdinand Fuhrmann. The student who performed the recording is called Christiane Puh. Since the behaviour of the music in dependence with the impulse responses was unknown, 2 ORTF- microphones were placed in order to allow different microphone combinations while the signal processing. The first set up was about 3 meters away from the organ and the second one about 6 meters. The music was recorded using a laptop, a 24-channel ADAT interface and 2 8-channel microphone pre amplifiers.
The most adequate microphone combination to be convoluted with the impulse responses was the first ORTF microphone (2 X Neumann KM140), due to it´s dryness and clarity.
The auralisation was carried out in Samplitude using a Room simulator, which convolutes a given impulse response with the original signal. As shown in Chapter 1 *" Theory"*, a LTI system can be fully described by its impulse response, so, if only the wet signal is left left in the Room simulator, the heard signal corresponds to the one that should be heard at the same place of the church.
In order to convert the impulses responses into WAVE- files, a little Matlab program had to be written in order to convert the impulse responses with the desired sampling frequency and resolution.

Matlab source Code:

Impuls.m:

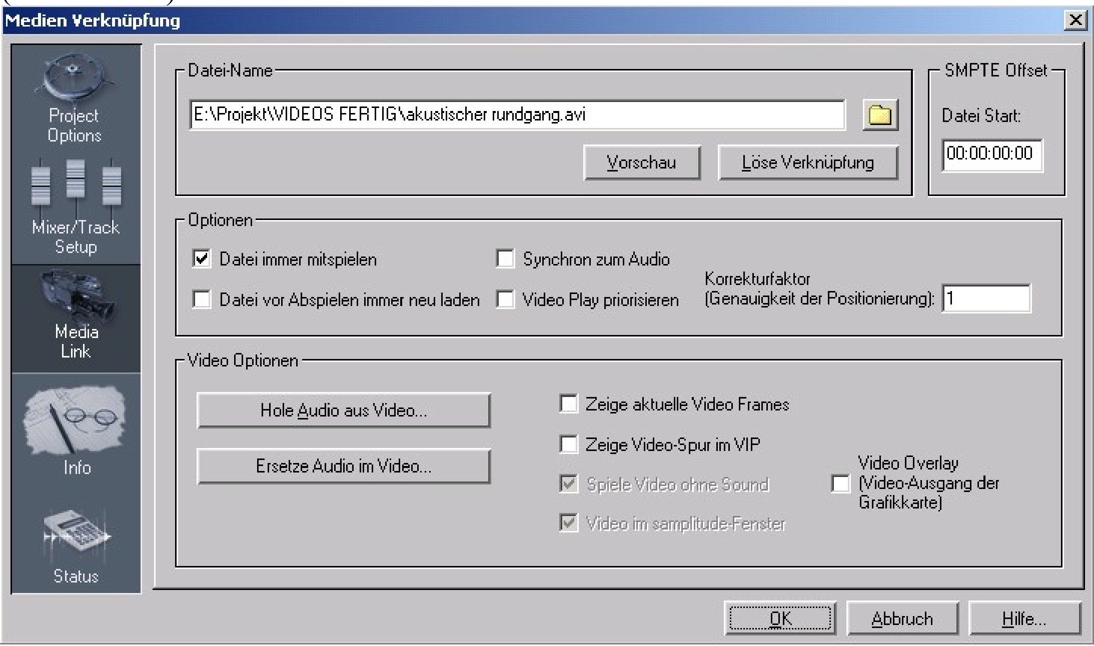
```
[y, Fs, Format] = loadimp('HerzJesu_S1_MickCh1_1.wmb');
                                        %loading the impulse response

y=y.*10./2^15;                          %calibration on to the maximum level

wavwrite(y,44100,24,'HerzJesu_S1_MickCh1_1 .wav');
                                        %delivery of the impulse response
```

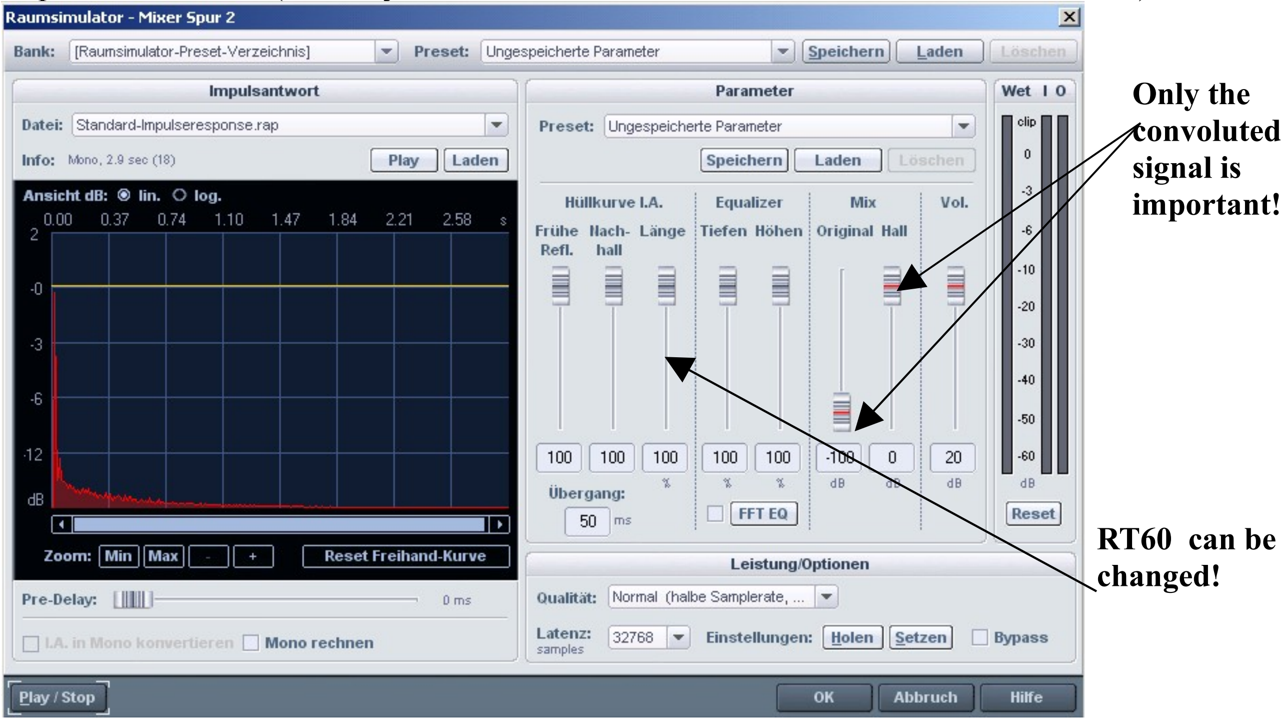
Since MATLAB only admits amplitudes between +1 and -1 in the files that have to be converted into WAVE files, the division with 2^15 was necessary to satisfy this condition with 16 bit resolution. 10 is an empirical value in order to achieve a big level adjustment, which was firstly used to adjust both impulse response sets, but the ones attached to the second source position were much to low, so the value was changed to 10^6, winning about 30 dB more loudness.

To provide each part of the song with the according impulse response, the soundtrack was cut into 30 parts (in Samplitude: objects).After having saved the project, the video was linked to the music using the " MediaLink "- option in Samplitude (Optionen-> Projekt Optionen -> Medialink) (Picture 3.5)



Picture 3.5 : Attaching a video to the soundtrack

Each object was provided with a Room simulator (Picture 3.6), in which the according impulse response was loaded (Auf Objekt clicken -> Echtzeiteffekte -> Raumsimulator-> laden).



Picture 3.6 : Room simulator

The object themselves were linked together with an exponential Fade In and a logarithmic Fade Out. The fading types were selected after having tested other Cross fading combinations which didn´t enable such a smooth change between the objects than the Cross fading option chosen.
Afterwards, the interesting acoustical effects of the church were pointed out thanks to the usage of filters or dynamic changes. Predictably, each soundtrack was processed in a different way.

Trick: How does a Room simulator work ?:
As it can be seen in Picture 1, Samplitude´s room simulator can calculate the reverberation time out of the impulse response. This is possible thanks to Schroeder´s equation:

$$<s^2(t)> \sim \int_t^{\infty} r^2(\tau)\, d\tau$$

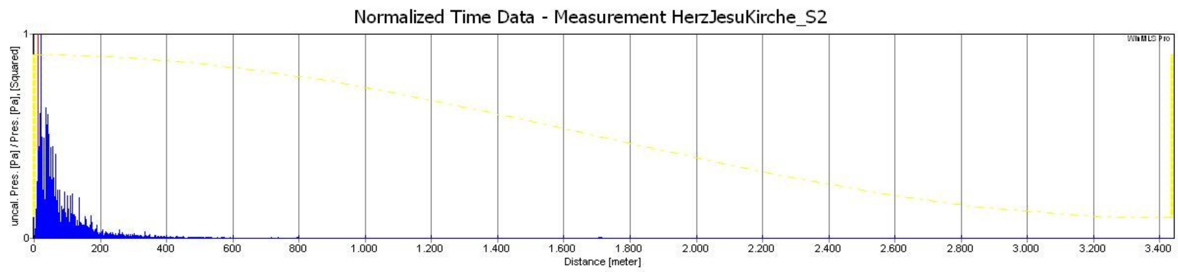
$<s^2(t)>$...mean value of the square time course of the decaying noise signal.

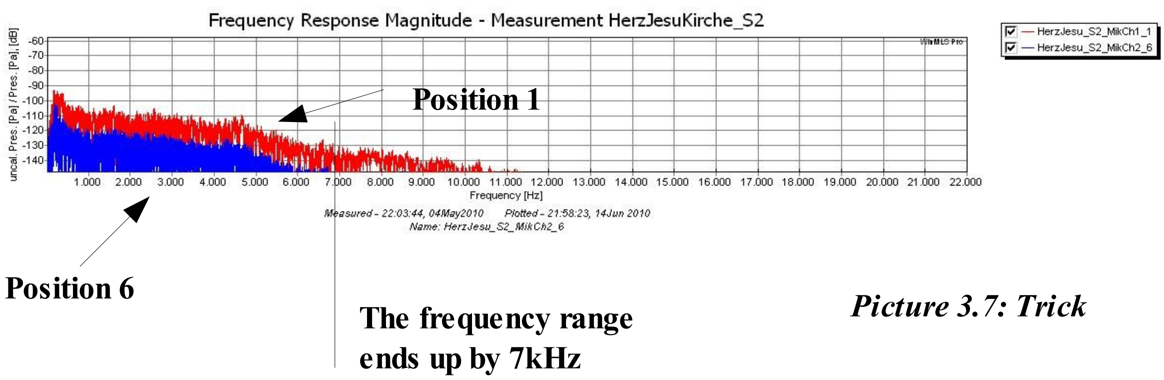
$r^2(t)$ ... squared impulse response of the system loudspeaker room – microphone.

Consequently, a room simulator´s precision is given by the implementation of this formula.
Before starting the auralisation, the reverberation times given by Samplitude should be compared with the ones calculated by WinMLS, or, if it´s possible, with the results of a RT60 measurement.
If they differ much, the reverberation time must be changed in the Room simulator.

Signal processing in Samplitude:
Samplitude is provided with a huge offer on equalization tools that can be very useful to remark the acoustical effects in the church.
The equalization tools used for these project are restricted to low – and high passes. After having decided what kind of filter helps most to show a certain effect, the impulse response magnitude plot can be very useful to know what frequency ranges play a key role or not.

Example:





Picture 3.7: Trick

If the loss of room information has to be shown e.g. inside the tower (Picture 3.7, blue curve), it wouldn´t make sense to put a low pass filter at 6800 or 6500 Hz! It would be sensible to start trying low passes which maximum frequency is about 5000Hz or lower.
Summing up, the impulse response can be very helpful in order to have a rough idea of how the music should be processed.

Trick:
Do never use Off line – Effects! This kind of effects change the sound of the file stored in the CPU, that means, the original one! To gain more security against these problems, work always with a copy of the original track and make sure Samplitude only works on the copy!

Empty church – Auralisation and audio effects.

Source position 1 (Altar):

Picture 3.8. First source position

At the very beginning (microphone positions 1 – 2), a low cut by 100 Hz was needed in order to remark the clarity that should be heard within the reverberation radius (near the source).The sound was at first very loud and brilliant, but very low frequencies were much to present.
At microphone position 5 (the door that connects two chapels), the loudness was increased 0,5 dB and a low pass at 3140Hz was added. These changes point out the loss of brightness of the sound due to the narrow walls and the much to big presence of late reflections, that kill the clarity at the point. Mathematically, the C80 parameter describes the clarity in a room:
$C80 = 10*\log (W_{0...80} / W_{80...\infty})$. If later reflections describe in a great way the impulse response at position 5, $W_{80...\infty}$ is much bigger than $W_{0...80}$. High clarity of music is guaranteed if $C80 > 0dB$ and a surround feeling is guaranteed if $-3db < C80 < 0$ dB, that means that early energy should be at least the same as late energy, which shows why the clarity at point 5 is so poor whereas the feeling of being surrounded by music is very good.
As already said, the microphone position behind the column (position 10) is a good example to show the effects of diffraction and acoustic shadow.
Due to the acoustic shadows, higher frequencies can not be heard behind the column because they are reflected (stone shows very poor absorption coefficients) whereas due to diffraction, low frequencies are bend all across the column. In order to show these effects more clearly, the loudness was increased about +0,6 dB and a low pass was set by 5kHz.
At microphone position 17, the loudness had to be set down about – 0,44 dB because otherwise the sound impression was rather bad as soon as the cello started "pushing".

Source Position 2 (Organ):

Picture 3.9 : Second source position

Because the first half of this round walk takes place in a place which is completely isolated from the church, the energy of late reflection ($W_{0...80}$) is very poor in relationship with the early energy($W_{80..\infty}$). Therefore, the clarity of the music increases during this period of time whereas position 5 at source position 1-round walk shows the opposite effect. Consequently, the surround feeling loss is dramatic. Indeed, the listener has the impression of being in a box while the music is being played outside in the box. This effect was backed up with a low pass at 3000-3800 Hz, depending on the microphone position.

Nevertheless, the sound in the tower turned naturally brighter due to the presence of diffuses field generated outside the tower (in the church), the difference between outside and inside the tower was not that big.. In order to increase this difference, at positions 11 -13, low frequencies (100-150 Hz) were slowly pushed up as well as mid – high frequencies (frequencies higher than 5000 Hz). Also the general sound level was increased (+1,2 – 2,2 dB)

At position 14, just under the organ, the loudness was increased about +0,8 dB to show how different it is to be on the side or in the centre of the church although the source is on a higher plane. At position 15, the first position at which the brilliance of the sound is greater, frequencies between 4300 Hz and 4800 Hz and frequencies beyond 10 kHz were pushed up about 1 +dB to remark the increase of the brilliance.
At position 26, behind a column, it was pursued to show how a column works if the source position is placed in a height. The loudness loss was dramatical and, to remark it, the loudness was set -1dB down. The effects of acoustic shadowing were remarked with a low pass by 4440 Hz

Full church – Auralisation and audio effects.

The simulation of a full church or,generally, a different behaviour of the church on the whole can be easily carried out using the Room simulator. Since the absorbing surface is now much bigger, the " typical " reverberant impression of the church won´t be that big.
Human bodies can absorb medium and high frequencies very well, so that the musical impression also differs from an empty church. Moreover, the intensity and the spectral characteristics of first reflections, so important to characterize the sound of a church, will also be different.
That´s why not only the reverberation time was changed, but also the early reflections, the reverberation and the equalization of the room simulator.
The first to do this simulation was to calculate the reverberation time assuming that all benches were occupied. The RT60 – loss was about 15%. In order to achieve a greater effect on the listener, the reverberation time was set down about 25% in both videos.

Source Position 1:
Throughout the whole video, the reverberation time was kept constant(≈ 6 sec), but the effect of first reflections was varied: The first 2 positions, which are very near to the source, do not show a big difference ,but, nevertheless, the relevance of first reflections was diminished about 10%. The equalization variation (low – high frequencies) was minimal since the direct signal, which dominates within the critical distance, remains the same.
Consequently, the impact of first reflections was diminished at positions near to to the walls or in the chapel to show that the audience absorbs much of the direct signal (especially high- and middle – ranged frequencies). In order to show the frequency dependant absorption of the audience, the impact of high frequencies was minimized in contrast to the impact of low frequencies that, although being also diminished, remain very present, especially in the chapels.
As soon as the videos moves towards the altar, the influence of the audience decreases very little: at the rear of the church (position 18), the greatest influence of the audience should acknowledged since the whole audience has worked as an absorbing surface.
At positions 28 – 30, very near to the source, the impact of the audience is minimal, almost equal to the empty church.

Source Position 2:
The effect of the audience is very little up at the organ because the direct signal predominates over all other components. Therefore, very little was changed on the simulator.
Throughout the spiral staircase the reverberation ("Nachhall") and the early reflections- proportion were changed, because the reverberating field in the church has little effect on the tower, which is virtually isolated from the church (there ´s only a little door connecting both volumes)
As soon as the church is entered, the behaviour was simulated in the same way it was simulated for the first source position, though the early reflections and the reverberation were not cut off so dramatically since the organ, placed on a higher plane of the church, provides the audience with many reflections coming from the top of the church, which remained the same.

Summary

Chapter 3 *"Studio Work"* presents the second major chapter of the *" Listening to a church "* project. Firstly, it includes an explanation about how the audiovisual material was processed in Samplitude and Vegas and secondly a report on the signal processing used to remark the acoustical effects on the soundtrack, which also includes the changes on the room simulator in order to simulate a full church.

Conclusions :

The *"Listening to a church"* project reflects the influence of architecture on sound with help of auralisations along two paths , each attached to a different source positions: one at the altar and one at the organ. It also regards 2 different audience configurations: an empty and a full church. Thanks to a multi sensual (audiovisual) experience, the listener can feel like listening to music in the church.

One of the most exciting facts on this project is the confluence of artistic and scientific elements, as shown on this report:
The necessary theoretical background shown on Chapter 1 *" Theory"* to carry out an auralisation can only been obtained out of a scientific approach that involves general aspects on room acoustics and acoustical measurements. Otherwise, it would be impossible to develop a suitable procedure to gain the impulse responses along both defined paths. Indeed, the impulse response as true acoustical fingerprint of any building can only be understood thanks to a higher technical knowledge.
The fundamentals on the contents that Chapter 2 *" Measurement "* presents are definitely scientific since the positive and negative elements of any measurement method can only be understood or explained falling back to scientific knowledge. Nevertheless, science and art melt as soon as the acoustical interesting microphone positions along both paths are presented, carving out a deeper knowledge on how certain effects sound like. At the same time, the comparison between the real and the expected behaviour of the church along both paths turned out to be very helpful to flesh out the bones of the hypotheses done on the expected acoustical behaviour. This way, the theoretical background could be proved on an artistic way (listening to the music) and on a scientific way (interpreting the impulse response plots).
The auralisation process described in Chapter 3 *" Studio work"* constitutes the main artistic contribution to the *"Listening to a church"* project: the technical possibilities offered by Samplitude and Vegas (software used to process sound and video respectively) obey the artistic purpose of creating 4 different auralisations, which show all acoustical effects found along the microphone paths. The original recordings and auralised pieces had to be listened many times, so that the creativity could flow easily in order to insert the right effects at the right position, providing relevant technical information as well as a delightful audiovisual experience for any listener.
At this point, we would like to thank Mag. Keil, the priest of the Herz Jesu church, for his trust and help during the measurement time and Dr. Graber for the support given along the whole project.

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