Spatial Sound Design: From Special Effects to Spatial Effects

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Abstract

Extensive research in the field of spatial audio reproduction has led to a large variety of reproduction systems available today. Depending on the desired application one can choose between systems based on acoustic field reconstruction (e.g. wave field synthesis (WFS) or higher order ambisonics (HOA)) and systems based on stereophonic principles as well as binaural systems and their combinations.

On the other hand a large number of tools exists to modify single audio streams. The development of spatial sound design tools is a field of current research and will become important for the production of content using spatial audio systems. This paper presents a new tool for spatial sound design based on measured room impulse responses and will discuss important issues in the development of spatial sound design tools from a sound designer's point of view.

1 Introduction

Today a large variety of established spatial audio systems exists. Beside experimental setups used by the acoustic avantgarde especially in the area of electronic music [1], several systems are used in the area of movie sound reproduction. Such industry is a large motor to bring spatial audio to a large audience.

A closer look on current developments shows a strong convergence in several systems. On one hand side the stereophonic systems are extended and growing from two channels to the ITU-R BS.775 surround setup [2] to larger systems like 22.2 [3] for example. On the other hand sound field reproduction systems aiming to reconstruct a acoustic field like wave field synthesis [4] and Ambisonic [5] about to be available to the market. The convergence of all loudspeaker based reproduction systems is an current point of research and is discussed by the research community. On the other hand binaural reproduction is established especially for simulation and auralization application and psychoacoustic research. Furthermore combinations of reproduction system try to combine the advantages of each reproduction system e.g. Binaural Sky [6], Binaural Room Scanning [7], WFS with Virtual Panning Spots etc..

For a very few applications the aim of a spatial reproduction system is a *natural* reproduction of a virtual or recorded situation. In most cases the aim is to communicate artistic messages. Examples are:

- a recorded music performance transformed to a spatial reproduction by sound engineers
- a pure virtual piece of music e.g. pop music produced in a studio or pure electronic music
- a virtual piece of acoustic art e.g. radio drama
- an audio-visual artwork e.g. a movie and its corresponding sound track

In most applications not a real acoustic environment is reproduced. The spatial audio scene is a pure virtual construct. The development and realization is termed *sound design*. The sound designer tries to communicate an acoustical idea and needs to transform this abstract concept to acoustic reality in a given environment. Such a concept of sound is not necessarily describable by the use of physical models. Furthermore such an acoustic idea does not depend on a particular reproduction system.

During the last decades of audio signal processing development, several tools for the modification of single audio streams have been developed (EQ, Compressor, etc.). All these process a physical property of an audio stream instead of a perceptual property. The sound designer transforms his acoustic idea into a physical parameter to reach his goals of acoustic communication. This requires besides the artistic knowledge a large background in signal processing and the interaction of both. Such a knowledge is the key element in the know how of a sound engineer, sound designer or Tonmeister. The number of tools to modify a single audio stream is endless in each specific category.

In contrast the process of spatial sound design modifies the properties of an audio stream direction depending on its direction and/or orientation. This includes the simulation of an acoustic environment but also its direction dependent visualization and modification through the user. E.g. a direction dependent editing of the early reflections of a virtual source.

In comparison to the variety of tools that process single stream, the number and possibilities of spatial sound design tools are very limited up to now. In current systems the spatial layout of a acoustic scene becomes more important but the tools for spatial sound design are restrained and in addition mostly specific for only one reproduction system. Futhermore the effort is concentrated on the task of direct sound positioning [8] or integrating the simulation of distance [9].

For this reason the aim of this work is to visualize the virtual acoustic environment and it's corresponding sound field in a direction dependent way. Furthermore, an intuitive way to modify the sound field is a direct interaction with such a graphical representation, which is investigated throughout this work.

For the derivation of reproduction system independent spatial sound design principles the following assumptions are made:

- An acoustic field can be represented in the far field of its sources as superposition of plane-waves.
- A linear time invariant (LTI) system can be fully characterized by its impulse response (IR).

A room can be approximated as a LTI system. As a result, the sound field in a given point can be fully described by a plane-wave decomposition of its room impulse response (RIR). Furthermore based on a plane wave decomposition the sound field can be easily extrapolated to another point as well as adapted to a specific reproduction system. Without loss of generality the discussion in this paper is limited to a pure horizontal direction dependent examination.

2 Plane wave decomposition of measured room impulse responses

Several authors have discussed the topic of direction dependent room impulse analysis for auralization purposes, especially in terms of wave field synthesis reproduction. Hulsebos [10] has described the principles based on circular array measurements for WFS applications. For a detailed description of the analysis of impulse responses the reader is referred to the detailed discussion in the literature e.g. [11] [12] [13]. For the following section only the result of such a process and the basic steps are considered. Figure 1 presents a block diagram of the analysis process. The basic steps of the room impulse response analysis are:

1. Measurement of impulse responses using a microphone array and system equalization
2. Detection and removal of direct sound
3. Segmentation of the impulse responses into its early and late/diffuse part
4. Detection and extraction of reflections from the early part
5. Analyzing the remaining early part by calculating the plane wave decomposition
6. Analyzing the diffuse part by calculating the plane wave decomposition

The result of this process are high spatial resolution room impulse responses separated in late and early parts as well as discrete reflections. This representation offers the basis for the following interaction and visualisation process.

3 Interaction

The interaction process can be divided in a static interaction (SI) part and a dynamic interaction (DI) part. The static interaction is based on the analyzed high spatial resolution data set from an array measurement. Afterwards the high resolution data are adapted to the desired spatial resolution required for a specific reproduction system. During the reproduction process the user can manipulate the source characteristics (e.g. directivity,

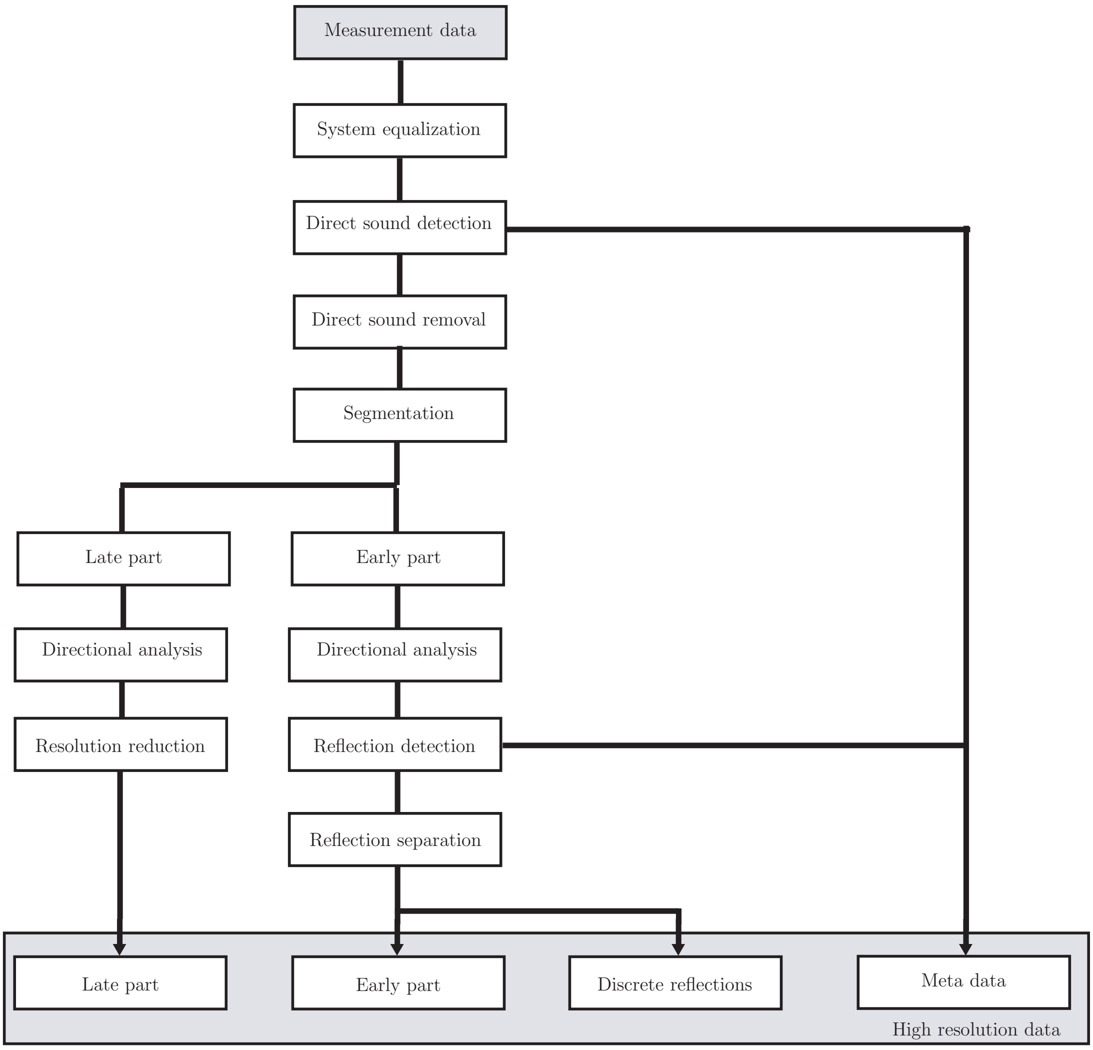


Figure 1: Block diagram of the impulse response analysis process

position, distance, etc.). This process is named dynamic interaction because it depends on the source parameters and has to be changed in real time during the reproduction. Figure 2 presents the complete process. After the previous described analysis of measured or simulated array measurements the high resolution data are available. The spatial high resolution data are used for the static interaction process and the storage of room presets because they are reproduction system independent. To enable the reproduction an adaptation to the desired systems lead to reproduction data. Depending on the desired source parameters a dynamic interaction is applied and the result is used in a convolution process resulting the required audio streams. Within this paper only the static interaction process will be discussed. The dynamic interaction is discussed in terms of source positioning and spatial audio scene creation in [8].

3.1 Taxonomy of Room Impulse Response Visualization

The visualization of impulse responses provides the basis for a direct interaction through the user. For this reason a taxonomy of visualization types is given in this section. The taxonomy is based on diagram types. For each diagram type the possible and common

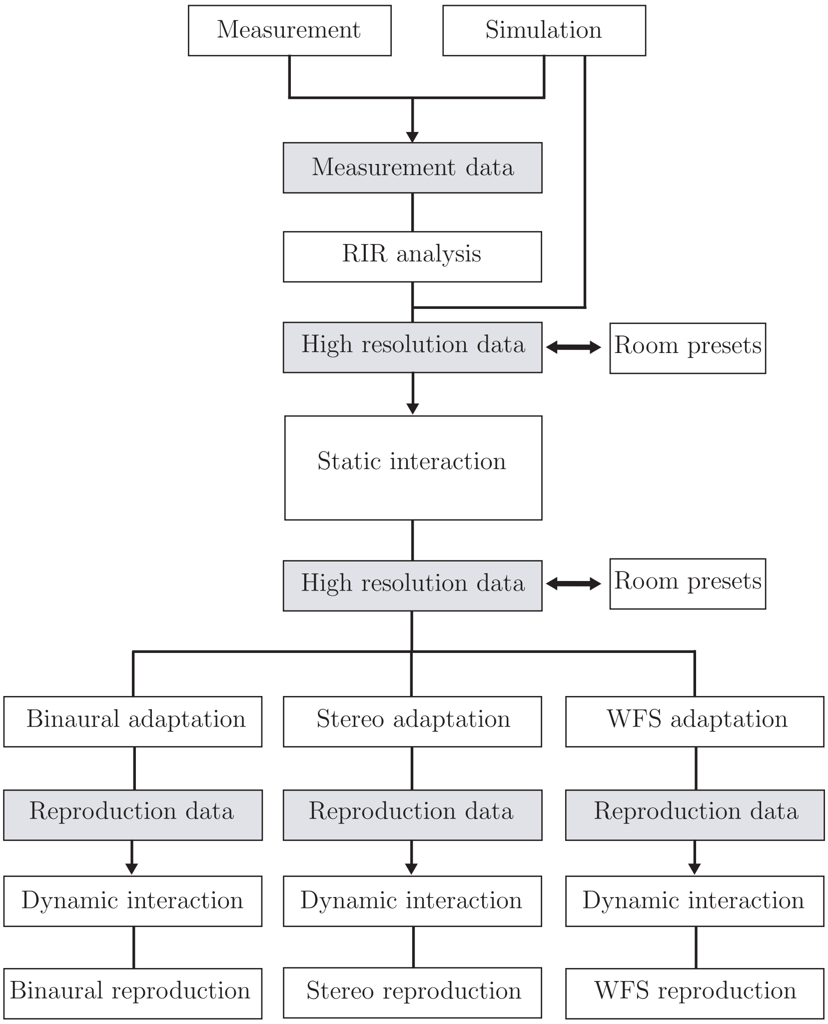


Figure 2: Block diagram of the complete interaction process.

axis are discussed. Furthermore the classification goes from single impulse responses to multiple or direction dependent visualizations.

3.1.1 Visualizations of Single Impulse Responses

A simple way to visualize an impulse response is to plot the impulse response directly in the time domain in a two-axis diagram as given in figure 3(a). Such a representation is very useful if insignificant details are removed beforehand because the fine structure of a impulse response will not be perceived directly. One can think about several models to rectify and smooth the impulse response.

A well defined method is the calculation of an envelope based on the Hilbert transform $\mathcal{H}\{h(t)\}^2$ of the impulse response $h(t)$ [14].

The next extension is the application of a psychoacoustic model to the impulse response in order to visualize only the preceptual relevant parts. The first application of a psychoacoustic model to measured room impulse responses has been proposed by Lokki et al. in [15] and Merimaa et al. in[16]. Schuitmann et al. in [17] used the processing of psychoacoustic models to denoise impulse responses. Such perceptual irrelevant noise in the impulse response is one possible reason for the spatial fluctuations of measurements of spaciousness reported in [18] and other parameters calculated from the impulse response.

Another two-axis representation of the RIR is the calculation of the energy decay curve (EDC) also named as Schroeder integral [19]. If the amplitude is plotted on a logarithmic scale the curve is strongly correlated to the perception of the decay of a room. The representation of a single impulse response in the frequency domain is not very meaningful because the temporal structure is the most important property.

Table 1 presents an overview of visualizations for single impulse responses using two-axis diagrams. A useful step is the extension to a third axis for the visualization as given in

No	Diagram	axis a	$a(b)$
1	Amplitude	time	$s(t)$
2	Energy curve	time	$s(t)^2$
3	Envelope	time	$\sqrt{h(t)^2 + \mathcal{H}\{h(t)\}^2}$
4	Energy decay curve	time	$\int_t^\infty h^2(\tau)d\tau$
5	Frequency curve	frequency	$S(\omega)$

Table 1: Axis types for single impulse response two-axis visualizations.

figure 3(b). The third axis can be used to extend the time information with a frequency information, which can be acquired by applying a short-time Fourier transform (STFT) to the representations 1 to 4 in table 1 or using other filter banks. As stated earlier Lokki proposed in [15] a visualization method for RIR based on the waterfall diagrams with a gammatone filter bank and an integration window in order to approximate the process of the inner ear. Table 2 presents an overview of the different combinations of axis used in the three-axis diagrams. The visualization of such diagrams can be 3D as given in figure 3(c). An alternative approach is to use a color coding as shown in figure 3(d), which can be advantageous if the third dimension should be used for spatial informations.

No.	Diagram	axis a	axis b	axis c
1	Energy decay relief	time	inverse energy integral	frequency
2	STFT	time	magnitude	frequency

Table 2: Axis types for single impulse response three-axis visualizations.

3.1.2 Visualization of Multiple Impulse Response

In case a sound field is analyzed in a direction dependent way, the visualization techniques given in the previous section are combined to represent the field and its directional components. Lokki proposed in [15] an extension with a direction dependent visualization using small arrows in each frequency bin to indicate the direction of incidence of the energy components. Figure 4 presents an example. The third dimension can also be used to integrate directional informations. Such representations are useful in scenarios for 2D array

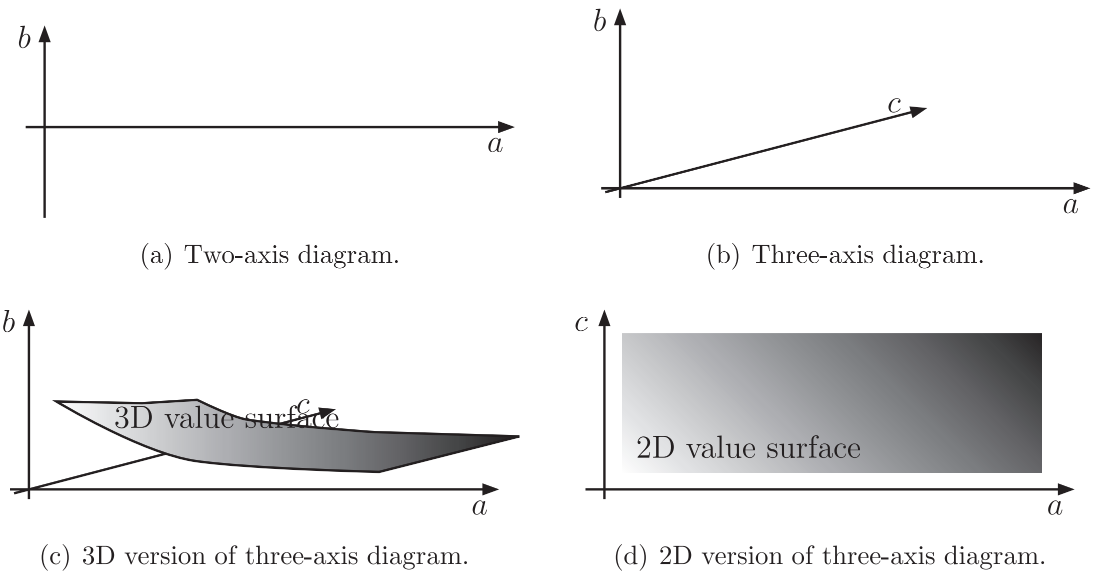


Figure 3: Diagram types for impulse response visualization using axis types given in table 2

systems if the presented RIRs are the result of a plane wave decomposition. All diagram types given in table 1 can be extended by a third axis representing the angle of incidence for the impulse response. To make such a diagram more intuitive it is useful to arrange the RIRs corresponding to their direction of incidence as presented in figure 5(a). An alternative way of including directional informations is the spatial arrangement of three-axis diagrams representing the angle of incidence for the different RIRs. Figure 5(b).

In case of an interactive visualization it is advantageous to think about volume data representations. In this representation each point in a 3D space has a value. The user can navigate through such data by placing slices in the dataset, which are comparable to the representation given in figure 3(d). Figure 6 gives an example of such a diagram. This visualisation was successfully applied in the analysis of room acoustic measurements applied on a room equipped with an acoustic enhancement system in [20].

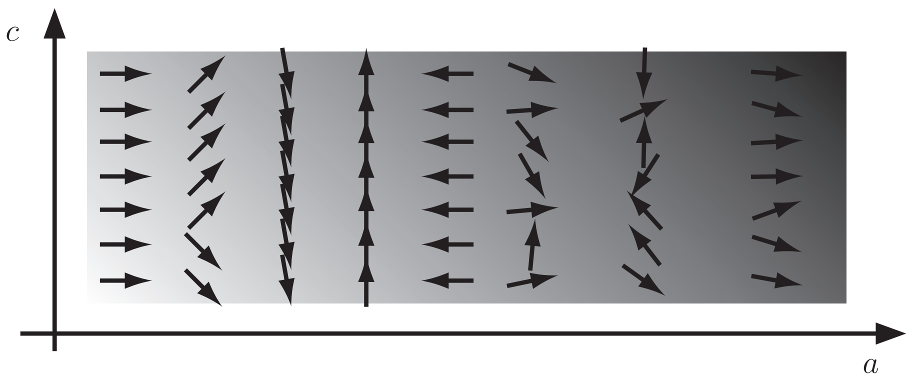


Figure 4: Extended three-axis diagram.

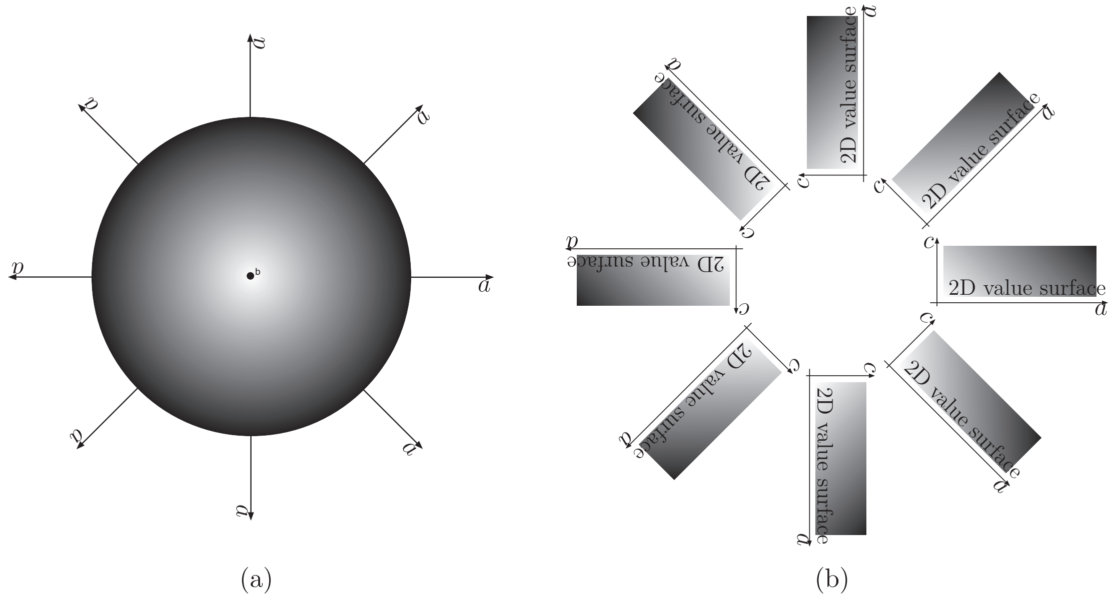


Figure 5: (a) Three-axis multi impulse response representation with correct alignment corresponding to the angle of incidence. (b) Three-axis multi impulse response representation with spatial arrangement of impulse responses corresponding to their angle of incidence.

3.2 Taxonomy of Static Interaction

In this section the static interaction will be discussed. First, basic principles will be introduced. Afterwarts such principles will be applied to single impulse responses and direction dependent multiple impulse responses.

3.3 Shaping Surfaces

An intuitive way for the user to modify a graphical representation is a direct modification of visualizations. The problem using this approach for an impulse response is the fine structure of such data which have to be kept to ensure a high auralization quality and the relatively coarse modification through the user. The modification has to be in another resolution than the data for the processing. Within this work parametric surfaces are used for this purpose as first proposed in [21]. The definition of such surfaces can be made

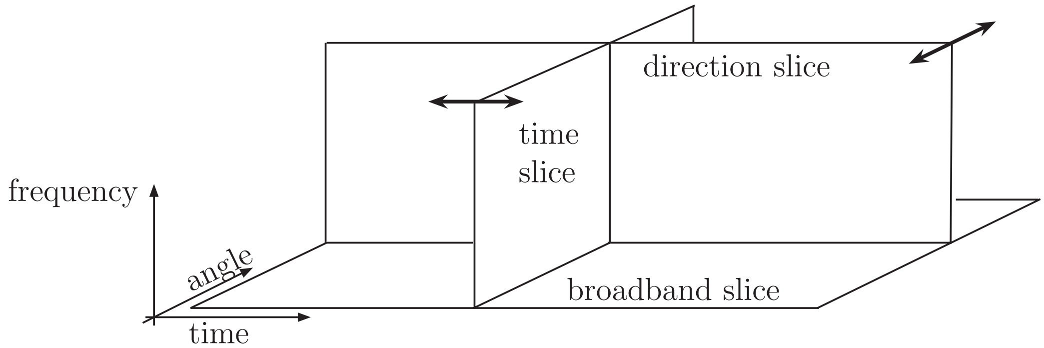


Figure 6: Example of a three-axis volume data representation with two slices.

using a limited number of control points, while the values of the resulting matrix can be sampled in any required resolution. To modify a data representation a parametric surface is generated above the selected data visualization. The user can modify the control points of the parametric surface in order to shape a modification matrix generated by sampling the parametric surface. The following steps are carried out during the interaction process:

1. Visualization of the data
2. Generation of a parametric surface, spatial separated from the data representation using a appropriate resolution of control points
3. Modification of the parametric surface according to the desired modification of the visualized data
4. Sampling of the parametric surface in a resolution according to the data
5. Processing the the data set using the sampled parameter matrix

This principle is used to realize modification of single impulse response (e.g.time variant filtering) as well as direction dependent modifications of parameter. Several mathematic descriptions exist for parametric surface generation. One with the most widespread acceptance and popularity are Non-Uniform-Rational-Bi-Splines (NURBS), specified by its control net, weights and knot vectors. These can be evaluated and drawn according to algorithms presented in [22]. A block diagram of the processing can be found in figure 7, in which $\{\cdot\}$ represents a arbitrate transformation and $\{\cdot\}^{-1}$ the inverse transformation of the data to modify.

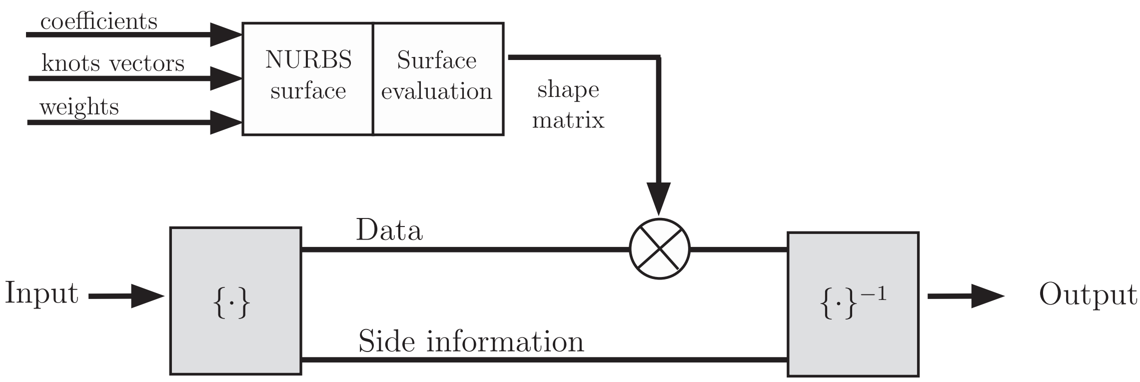


Figure 7: Block diagram for the concept of surface shaping

3.4 Interaction with Single Impulse Responses

The basis for the direct interaction with impulse responses is the interaction with a single impulse response. Depending on the axis representation according to section **??** one can think about the following main editing capabilities:

- Direct modification of the envelope applied to level or energy
- Time variant filtering

- Decorrelation of the late part
- Time stretching

These modifications are named basic layer modifications. While the latter two are out of the scope of this paper the first two will be discussed in the following.

3.4.1 Inverse Energy Decay Curve

The editing of the energy decay curve (EDC) was originally described in [23]. The EDC of a RIR plotted with an logarithmic amplitude scale is highly correlated with the perception of the decay of a diffuse sound field. Detailed fluctuations as shown in a time plot of a RIR are not perceived in detail. The modification of the envelope of measured RIR is a well known tool in sound design. The drawback of such a modification is that the envelope defined by the user is not highly correlated to the perception. For this reason a direct modification of the energy decay curve is proposed for sound design application. After the definition of the desired EDC by the user an envelope can be generated to modify the impulse response in a way that the new curve is achieved.

3.4.2 Time Variant Filtering

In case of time variant filtering the transformation in figure 7 is realiazed as a short time Fourier transform (STFT). The STFT of a discrete signal $STFT\{p(n)\}$ is a basis for the time variant modification of signals, if the parameter of the transformation are chosen appropriate. The relevant parameters are the analysis window, size of the discrete Fourier transform (DFT-size), hop-size and synthesis window. The following requirements should be met by the processing. If the user conducts no modification the processing should be transparent. In the context of time-variant filtering the user modifies the magnitude spectrum. To meet this requirements first an appropriate analysis window and the hop-size should be chosen. Second, due to modifications of the magnitude spectrum no time aliasing should occur. For this reason the size of the DFT for each frame should be $\geq (N_a + N_b) - 1$ while N_a and N_b are the number of windowed samples of the RIR and the sampled surface respectively. The time-variant complex discrete spectrum $P(s,k)$ of the signal $p(n)$ can be separated into a magnitude spectrum $A(s,k)$ and phase spectrum $\Theta_A(s,k)$ as:

$$A(s,k) = 20log_{10}(|P(s,k)|) \tag{1}$$

$$\Theta_A(s,k) = \angle P(s,k), \tag{2}$$

where s indicates the temporal samples and k are the frequency bins. While the phase spectrum is kept, the user modifies a normalized NURBS-surface, which is evaluated in the same resolution as the magnitude data $A(s,k)$. It is assumed that the NURBS representation is given in logarithmic scale with 0dB in its default setting. The evaluated and sampled NURBS-surface $S(u,v)$ is used for shaping the magnitude values of the room impulse response which results in the new magnitude spectrum $B(s,k)$.

$$B(s,k) = 10^{\left(\frac{A(s,k)+S(u,v)}{20}\right)} \tag{3}$$

with

$$u = [0,\ {}^1\!/\!_s,\ {}^2\!/\!_s,\ \dots\ 1] \qquad (4)$$
$$v = [0,\ {}^1\!/\!_k,\ {}^2\!/\!_k,\ \dots\ 1]$$

In this way arbitrary time-variant modifications are combined with the advantage of an intuitive interaction for the user. The processing is closely related to the cross synthesis of two sounds described by Serra [24], while the room impulse response represents one and the evaluated NURBS-surface represents the other sound. The complex spectra $P'(s,k)$ for the shaped room impulse response are made up of the phase spectra $\Theta(s,k)$ and the modified magnitude spectra $B(s,k)$.

$$P'(s,k) = B(s,k)\cos(\Theta(s,k)) + iB(s,k)\sin(\Theta(s,k)) \qquad (5)$$

Now the inverse STFT of $P'(s,k)$ is computed while neglecting the synthesis window because the phase response is kept. The NURBS-surface for the modification and the result as an STFT representation and energy decay relief (EDR) plot is shown in figures 8 and 9. While the STFT provides a good view on the processing of the discrete impulse response, the EDR plot delivers a very good estimation of the perception of the room effect after the convolution process. Note the fact that the modification results only in a time-variant filtering of the impulse response whereas the time structure is preserved.

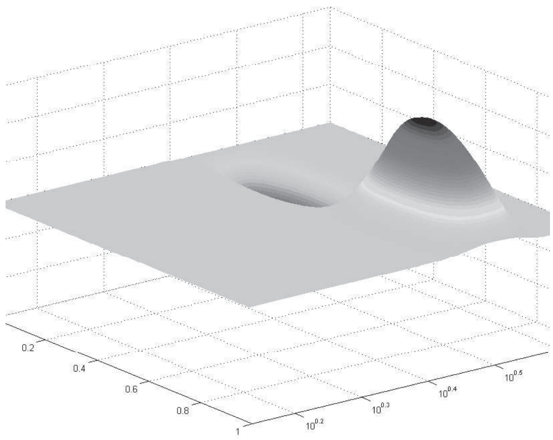


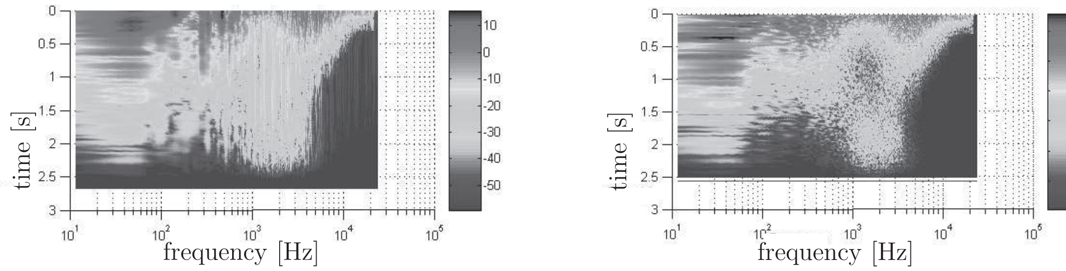
Figure 8: NURBS-surface for the modification of the room impulse response.

3.5 Interaction with Direction Dependent Impulse Responses

The interaction with direction dependent impulse responses can be structured in 3 layers, denoted as basic layer, acoustic layer and spatial layer.

3.5.1 Basic Layer

All interactions which follow the principles given in the previous sections are defined as basic layer interactions. The extension for a spatial sound design is a time and spatial



(a) Energy decay relief plot

(b) Spectrogram

Figure 9: Different representations of the shaped room impulse response.

angle selection process, corresponding to the select and modify paradigm. The following steps are performed in the interaction loop:

1. Choose the desired high resolution data
2. Select a time window
3. Select an angle window
4. Visualize the resulting impulse response
5. Modify the selection with tool given in section 3.4
6. Process the underlying data under consideration the time and angle window

This is the first step toward a spatial sound design because the parameter and properties are now treated in a direction depend way. Based on this the following two layers can be treated as macros on this basic layer interaction.

3.5.2 Acoustic Layer

Several quantities can be derived from room impulse responses. The most relevant are specified in the ISO-3382 [25]. Such quantities are related to specific perceptual attributes. In the acoustic layer these quantities are modified by the user. This is achieved by using the modification of the basic layer with dedicated macros. For example to modify the lateral fraction a direction dependent envelope can be applied modify the energy of the early sound field separated in the lateral and omni directional component.

3.5.3 Spatial Layer

In the basic layer the users selects a range of impulse responses in terms of angle and time. These selection is treated like a single impulse response, using the interaction methods for single impulse responses discussed in section 3.4. For some parameter a direct manipulation in a direction dependent way is useful. Such an editing is classified as spatial layer interaction. A solution for the fast editing of direction dependent parameters is the application of a spatial envelope. This is the use of the shaping surface concept in a spatial arrangement. In case of 2D reproduction a disk shaped surface can be used representing the desired value as height, the direction as angle and a second parameter e.g. the time in the radius. A possible application can be found in figure 10.

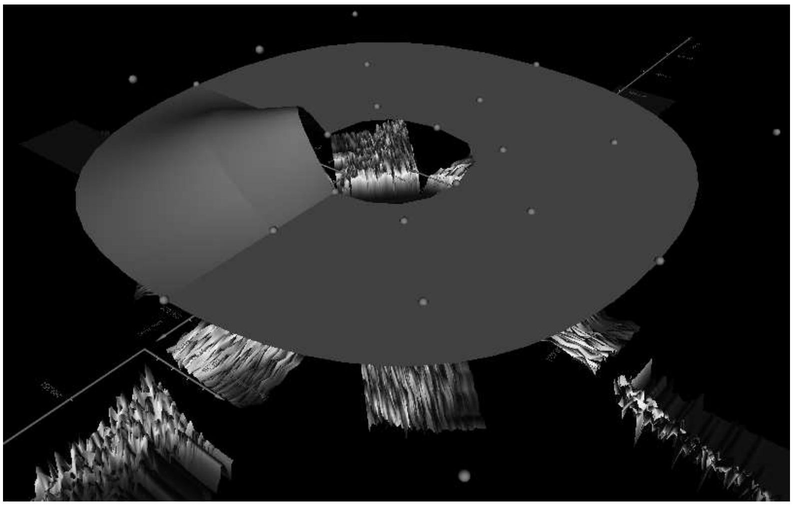


Figure 10: Spatial envelope based on NURBS applied to a three-axis multi impulse representation.

4 Prototype System

The described processing was realized in a graphical user interface (GUI) base on MATLAB® . The user interface is structured in separate modules, these working independent from each other. Furthermore a dedicated storage format for multi-trace impulse responses based on XML was included. The current version of the GUI consists of 3 modules: *Simulation Module*, *Visual Soundscape Editor* and *WFS Adaptation Module*. These modules will be described briefly in the following sections.

4.1 Simulation Module

For the ideal simulation of a plane wave decomposition a mirror image source model (MISM) extended by a diffuse sound model is used. The user can specify the absorption coefficients of the wall of a rectangle room in octave bands. A first order directivity can be applied to the direct sound included in the MISM. Based on the specification of the MISM a diffuse field is simulated based on decorrelated and shaped noise. The user can modify the envelope of the different parts of the impulse responses. The resulting impulse responses can be calculated in arbitrary spatial resolution representing a ideal plane wave decomposition. This data can be exported as high resolution data and further processed by other modules.

4.2 Visual Soundscape Editor

Based on simulated ideal plane wave decomposition or post processed array measurements the Visual Soundscape Editor provides the static interaction described in the previous sections. Figure 11 presents a screen shot of the user interface. An overview of the PWD is given in the left middle section. The user can select an angle and time section in the upper-left area. The resulting impulse response for this section and the applied window is show in the upper-right section of the GUI. Different interaction models are implemented and can be accessed in the lower-right area of the GUI. The screen-shot shows a graphical envelope editor.

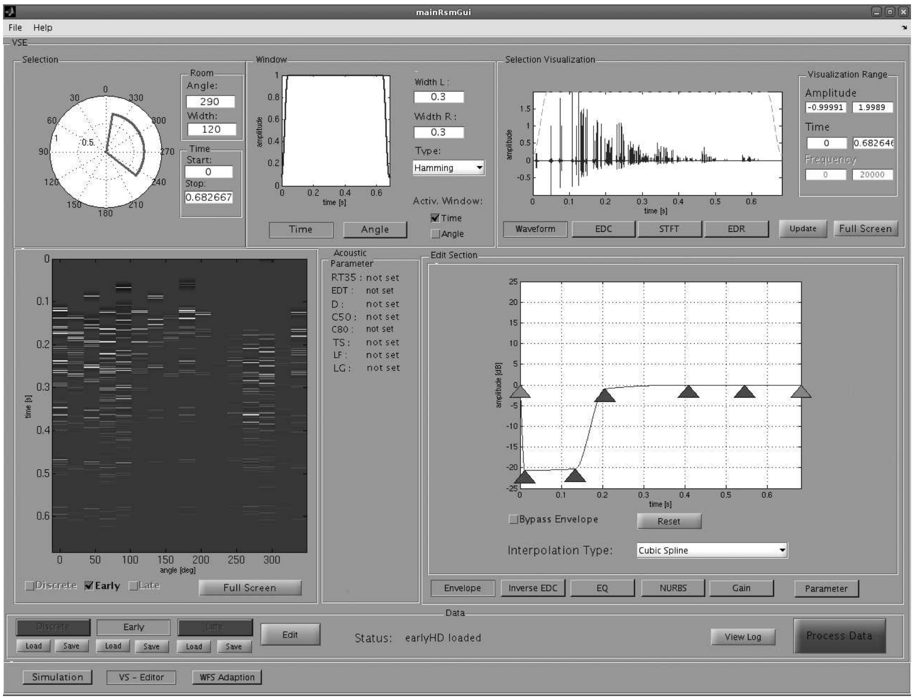


Figure 11: Screenshot of the Visual Soundscape Editor.

4.3 WFS-Adaptation Module

After the editor is applied to the high spatial resolution data an adaptation to the dedicated reproduction system is required. This process generates the reproduction data which are to be used in the dynamic interaction. The adaptation of plane wave decompositions to different reproduction systems is described in detail in [26]. Within the prototype system an adaptation to WFS reproduction is realized. The user can specify the desired reproduction scheme which includes the virtual source configuration for the reproduction of the spatial informations. The reproduction data are generated and exported to a dedicated WFS-reproduction system.

5 Summery and Further Work

This paper presented a novel system for direct interaction with measured room impulse responses. The visualization and interaction possibilities based on single room impulse responses are classified and extended in order to develop new methods for spatial sound design. The system is realized as a prototype system and will be extended to the adaptation to different reproduction system. Furthermore novel interaction possibilities will be included in the near future. The integration of new hardware user interfaces elements is also part of the further work.

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