# Sound Field Analysis in Room Acoustics

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## Abstract

Current state-of-the-art methods of the analysis of room acoustics are still widely based on capturing room impulse responses (RIRs) using omnidirectional or low order directive microphones. These RIRs can be used in order to achieve simple auralization or to derive typical room acoustic criteria. The problem here lies in the very limited spatial resolution. In this case, spatial information is merged inseparably in the omnidirectional impulse response. Employing microphone arrays and appropriate sound field analysis techniques, it is possible to decompose the original sound field in situ and e.g. to compute directional RIRs. This data can be used for a highly detailed analysis of room acoustics, similar to the methods offered in room acoustic simulation tools. It is possible to analyze and especially to visualize reflections, to create reflection trace models or to modify and auralize the acoustic environment. The principles of sound field analysis in room acoustics are introduced in a first step. In a second step some practical application examples taken from the WDR (Westdeutscher Rundfunk) radiobroadcast studios are presented and discussed. The special focus of the analysis lies on computed reflection trace models and the illustrative visualization of single reflections by combining sound field decomposition techniques and spherical panorama photography.

## 1. Introduction

The analysis of room acoustics and the first instrumentally acquirable room acoustic criteria trace back to the early 20$^{th}$ century, when W.C. Sabine (*1868-†1919) proposed the reverberation time as a first physical and instrumentally acquirable quantity describing and quantifying room acoustic properties. Sabine additionally found a simple formula to estimate the reverberation time depending on the room volume and the total absorption area that is still valid and being taught at universities around the world. [1] Since then, besides of the highly relevant RT (Reverberation Time) criterion, a variety of additional instrumentally acquirable criteria like e.g. EDT Early Decay Time, STI Speech Transmission Index, BR Bass Ratio, C50 Clarity 50 or C80 Clarity 80 has been proposed in order to enable an approximate deduction of the perceptual qualities of a room [2]. Today, these criteria are usually derived from omnidirectional room impulse response measurements as depicted in ***Figure 1***. The criteria indeed allow for an approximate prediction of the overall perceptive quality but rarely serve to detect and locate any concrete origin of room acoustic problems. Even if more modern spatial criteria like IACC Interaural Cross Correlation, LE Listener Envelopment or LF Lateral Fraction involve slightly more complex microphone configurations like MS setups or dummy heads, the spatial resolution and ability to locate acoustical problems is limited. All spatial information is inseparably merged in the non-directional impulse responses. On this account, it can be very difficult to deduce the spatial direction of a single contribution to the impulse response. With experience and luck, it might be possible to guess the correct spatial direction for some first order reflections, but for higher order reflections the correct assignment is hardly possible. For small rooms and chambers this process is even more difficult, as the temporal sequence of reflection contributions is squeezed. In room acoustics, the directional assignment of early reflections can be of great interest for the rating of the acoustic properties on the one hand and for a precise optimization of the acoustic behavior on the other hand. Currently, acousticians employ CAD based room acoustic simulation software to obtain this kind of information. But the required highly detailed modeling of complex room geometries is difficult and very time-consuming in practice. Additionally, the results might not be sufficiently accurate. Sound field analysis methods based on microphone array measurements might be a very promising alternative, as will be shown in the present paper. A basic introduction and survey on the required sound field decomposition and analysis techniques is presented and some application examples from the WDR (Westdeutscher Rundfunk) radiobroadcast studios are presented in a second step.

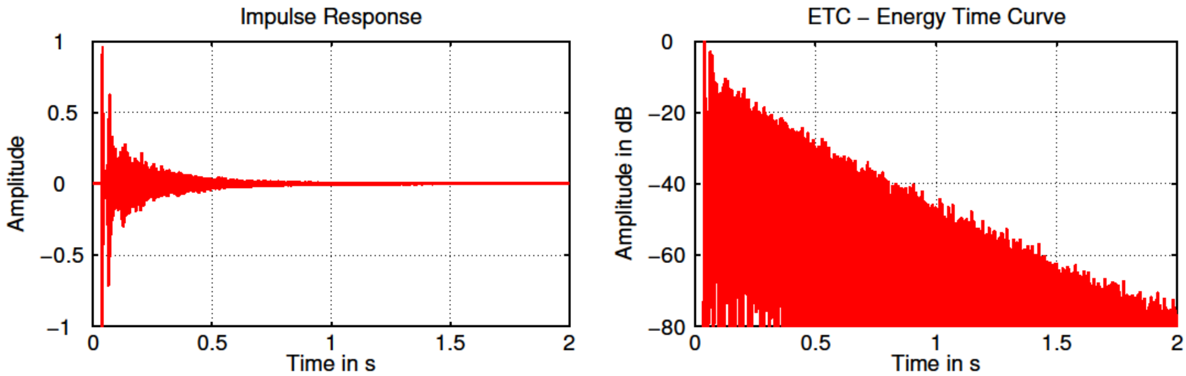



***Figure 1*** *Typical room impulse response and the corresponding Energy Time Curve. The impulse response has been captured at the large broadcast studio of the WDR in Cologne (Germany) using an omnidirectional source on stage and an omnidirectional microphone positioned in the audience area. The contributions of single reflections are visible, especially in the first segment of the impulse response. But the concrete directional allocation can only be guessed.*

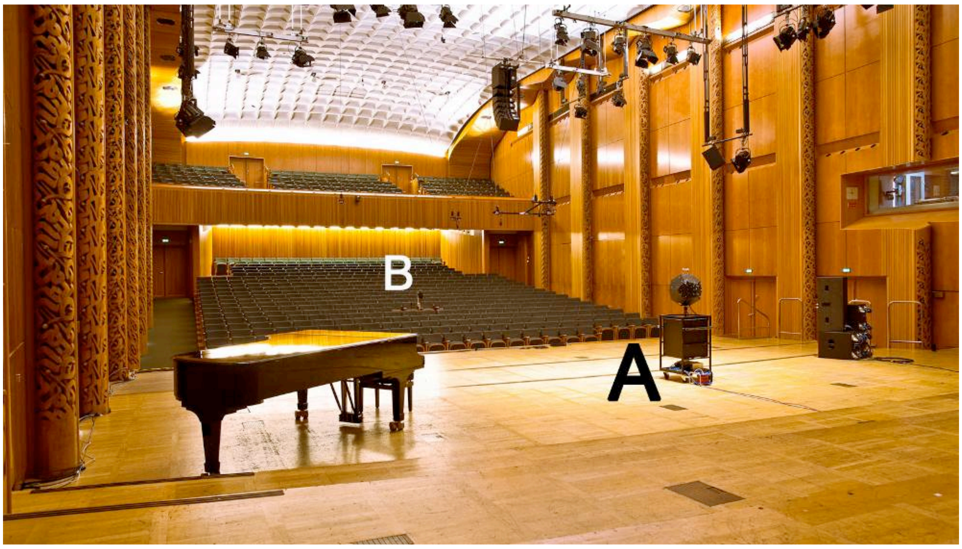



***Figure 2*** *Photography of the "Klaus von Bismarck Saal", the large broadcast studio of the WDR at the heart of Cologne, taken from a perspective on stage during the measurement session. The sound source is placed at position* ***A*** *and the receivers are placed at position* ***B*** *in the listening area.*

# 2. Sound Field Analysis with Microphone Arrays

*Sound Field Analysis* (SFA) or *Wave Field Analysis* (WFA) is a method that is closely related to the widely known *Wave Field Synthesis* (WFS) spatial audio reproduction technique [3], only turning the underlying mathematics upside down. The fundamentals of SFA are based on the Kirchoff-Helmholz-Integral (KHI), which relates the sound pressure inside a source free volume with the pressure and velocity on its surface. The definition of inside and outside is exchangeable, splitting up into an interior and an exterior problem.[4] Hence, if a sound field is determined on a real or imaginary surface only, the rest of the source free sound field can be derived mathematically. This enables some useful applications like sound field extrapolation, plane wave decomposition or modal beamforming. Sound field extrapolation can e.g. be employed to derive a driving signal for spatial audio reproduction systems or for an emulation of arbitrary (e.g. stereophonic or surround) microphone constellation. For the considerations and applications in room acoustics as presented in the following, especially the plane wave decomposition techniques are of interest.

## 2.1. Microphone Arrays

As a first step, the sound field must be captured on a real or imaginary surface. In practice, a spatially distributed array of microphones can be used for that purpose. The theory demands the continuous definition of sound field properties on the respective surface, but the array in practice only acquires values at discrete spatial sampling positions. This entails some dramatic effects in the output response as will be discussed later.

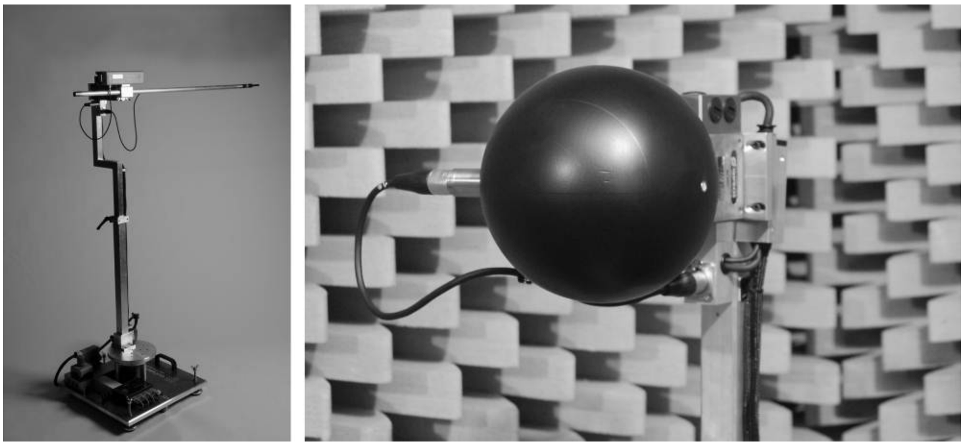


***Figure 3*** *VariSphear, a variable spherical microphone array measurement system designed and built at the Cologne University of Applied Sciences in 2009 [5]. A typical room is assumed to behave like a linear time-invariant system to great extent. Therefore room impulse responses at different microphone positions can be captured sequentially. VariSphear is a fully automated robotic scanning array system that acquires room impulse responses with a single microphone only. During a sound field capture session, the microphone is moved by two motors (azimuth and elevation) to the different positions e.g. on a large virtual sphere (left picture) or a small physically present rigid sphere body (right picture) sequentially. The advantages of a scanning array system with a single microphone are the high amount of possible spatial sampling positions and the perfectly matched audio chain for all positions. Obviously the real-time acquisition of a sound field is not possible. However, this is not a great disadvantage for the analysis of room acoustics.*

Arranging the microphones of an array on a real or imaginary spherical surface is very popular due to the rotational symmetry of the sphere and the well examined and defined spherical mathematics. Indeed, different configurations have been investigated and presented, but the practical advantages of the basic spherical geometries like e.g. simplified mathematical treatment and physical constructability still prevail and thus we restrain to spherical microphone array geometries in the following. Different array configurations can be realized, e.g. using a rigid and sound reflective spherical body with embedded microphones. This configuration enforces the sound velocity to be zero on the surface and leads to an improved mathematical handling, as conventional microphones can only measure the sound pressure and do not acquire the sound velocity. Furthermore, there are important parameters like the measurement radius or the density, the number and the arrangement of spatial sampling points. These factors have a direct impact on the valid frequency range that is resolvable by the array. [6][7][8][9]

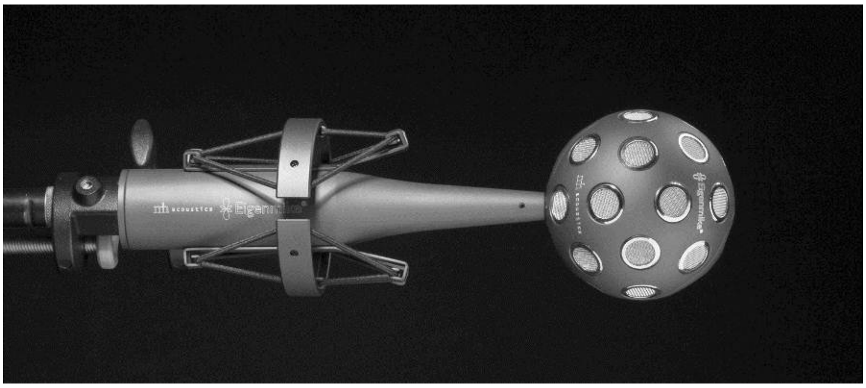


***Figure 4*** Eigenmike*, a commercially available spherical microphone array from Gary Elko and Jens Meyer (mh acoustics,USA) with a diameter of 8.4cm involving 32 sensors. The array can be used for real-time applications like spatial audio recordings or the analysis of room acoustics.*

The design of a microphone array always comes along with the process of finding the best tradeoff between different factors for the intended application. Generally, low frequencies demand a large measurement radius and high frequencies a high sensor density. Thus wideband arrays unfortunately need to meet both of these requirements at the same time. But increasing the radius with a fixed number of sensors obviously leads to a decreased sensor density. To work on a large bandwidth, the number of sensors has to be increased or different radii for different frequency bands must be involved [8].

For the SFA with spherical aperture microphone arrays, the sound field is decomposed into Spherical Harmonics (SHs) by applying a Spatial Fourier Transform (SFT). The SHs are a set of orthonormal base functions on the sphere. They are defined for different orders and modes and provide combined solutions to the angular part of the linear wave equation in a spherical coordinate system. [4] The more orders and its corresponding modes of the SH are involved, the more accurate and detailed the resolution of the sound field will be, and a higher directional selectivity of the array will be reached. Even if the high frequencies typically tend to excite more higher order SHs than low frequencies, the SHs inherently are not depending on the frequency. This initially leads to a very convenient constant directivity over the frequency in theory. Higher order SHs show more complex and delicate structures. To be able to resolve an increasingly fine structure of the corresponding function, the density of sensors must be sufficiently high. If the sensor density is not adequate, the SHs cannot be resolved with certainty. Thus the urgently required orthogonality property for the base functions of a Fourier transform is violated. The result of the spatial undersampling is the appearance of unwanted spatial aliasing artifacts in the array output signal [10][11]. As high frequencies tend to excite more higher order SH, the spatial aliasing artifacts appear much stronger at higher frequencies. Thus spatial undersampling has a major impact on the

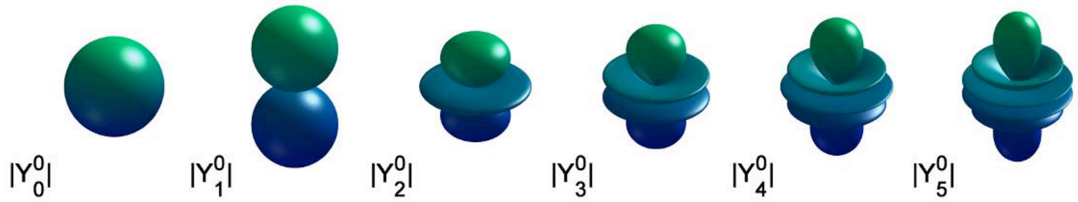



***Figure 5*** *Magnitudes of different exemplary Spherical Harmonics* $|Y_n^m|$ *for different orders n and the corresponding mode m=0.*

upper frequency range and the sensor density defines the upper stable frequency limit. Different approaches have been proposed to avoid spatial aliasing at low sensor densities and to extend and improve the array response at high frequencies in practical applications [11][12][13].

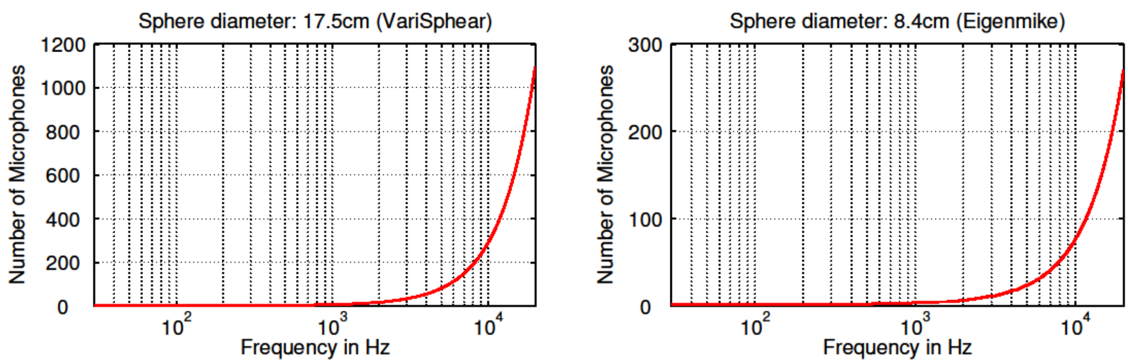



***Figure 6*** *Illustration of the (absolute) minimum required number of microphones or spatial sampling positions over the frequency for two exemplary fixed array diameters: A VariSphear rigid sphere extension with a diameter of 17.5cm (left plot) and the Eigenmike with a diameter of 8.4cm (right plot). With its 32 sensors, the Eigenmike can resolve frequencies up to a theoretical maximum of around 5-6 KHz with low contribution of spatial aliasing. In practice, the stable frequency limit is slightly higher and additionally mh-acoustics provides stabilization for the higher frequencies [12]. Generally, the quadratic increase of required microphones over the frequency is a serious challenge for the construction of broadband real-time arrays.*

The radial part of the solution to the wave equation in spherical coordinates is given by a combination of spherical Bessel and Hankel functions, depending on the respective configuration and problem set [4][8]. The radial part is compensated during the array processing by modal radial filters (**Figure 7**). The compensation is increasingly critical especially towards lower frequencies, as higher orders tend to have a rapidly decreasing energy contribution and thus need to be amplified excessively. In practice, the demanded amplification gain cannot be applied below a certain frequency due to the natural noise in the signal chain that is coming e.g. from microphones, amplifiers and converters. But also sensor positioning errors, calibration inconsistencies, temperature deviations and other factors can be understood to produce noise in the signal chain [8]. Thus the compensation can only work successfully if the signal to noise ratio is sufficiently high. In practice, the radial amplification must be limited in order to obtain a stable array output signal. This limitation reduces the directional gain and response selectivity at lower frequencies.

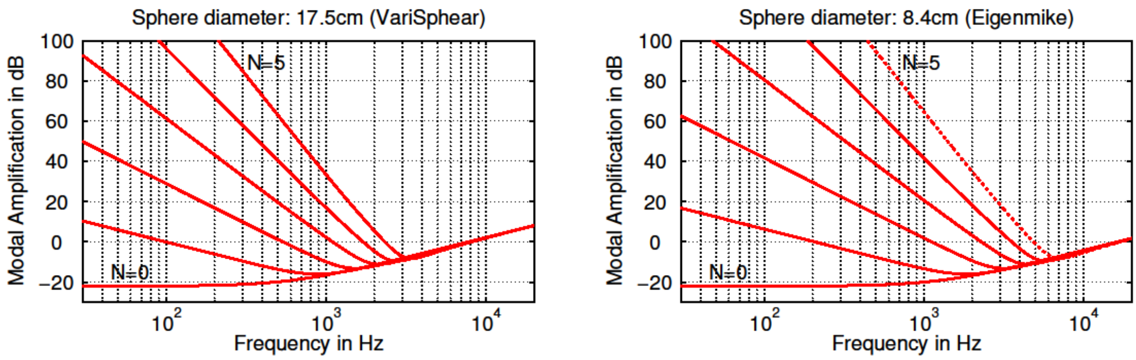



***Figure 7*** *Illustration of the required modal amplification (magnitude of the modal radial filters) up to an order of N=5 over the frequency for the two exemplary array diameters. Due to its low number (32) or density of sensors, the Eigenmike can resolve a maximum order of N=4. Therefore N=5 is illustrated as a dotted line only. The VariSphear scanning array theoretically can involve unlimited spatial sampling positions and hence resolve arbitrary orders. However, in practice the modal resolution is limited due to a limited signal to noise ratio. The modal radial filters demand excessively high amplification gain at lower frequencies. The amplification must be limited in order to keep the array response stable. This dramatically reduces the directional gain and response selectivity at lower frequencies. In general, spherical arrays tend to work best around the visible knee around* $f=(N\cdot c)/(2\pi r)$ *[5], where N denotes the highest involved mode, f the frequency, c the speed of sound and r the measurement radius.*

## 2.2. Plane Wave Decomposition

In theory, the infinite summation of all SH orders and corresponding modes equals a spatial Dirac pulse that reaches unlimited directional gain and enables perfect spatial sampling. Perfect spatial sampling means, that a plane wave impact to the array excites an infinitely small point in the spatial domain and thus the spatial selectivity (the ability to resolve different directions and different closely spaced sources) is unlimited. In practice the summation of all SH orders and corresponding modes is not possible, because the SHs cannot be detected unambiguously above certain order - the reasons for that have been discussed in the last section. Hence the summation of SHs is truncated at a certain order N.

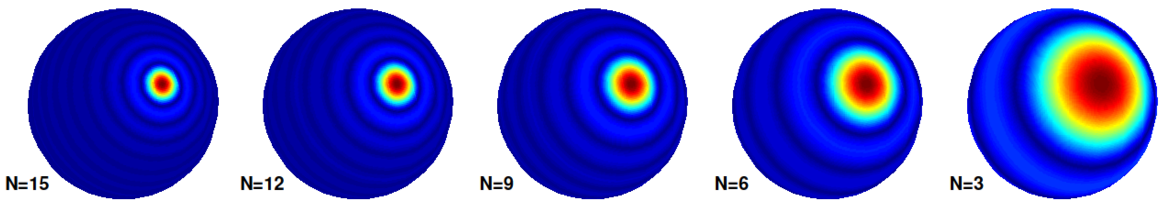



***Figure 8*** *Simulated spatial response to a plane wave impact truncated at different orders N.*

This truncation leads to limitations in the attainable directional gain and spatial resolution. A plane wave impact then leads to a spreaded point in the spatial response, similar to the phenomena described by point spread functions in optics. Here, the corresponding point spread function describes the resulting non-ideal spatial impulse response of the system that is introduced due to the truncation of involved SH orders. The main axis response is blurred and additional sidelobes arise. The ability to separate directions and sources is reduced. Furthermore, the underlying model for the plane wave decomposition of a sound field entails certain amplitude and phase inaccuracies for spherical wave fronts. In the following, this influence of wave front curvature is neglected, as the observed wave fronts in room acoustics are assumed to be sufficiently plane to produce a valid output signal with minor error contributions only.

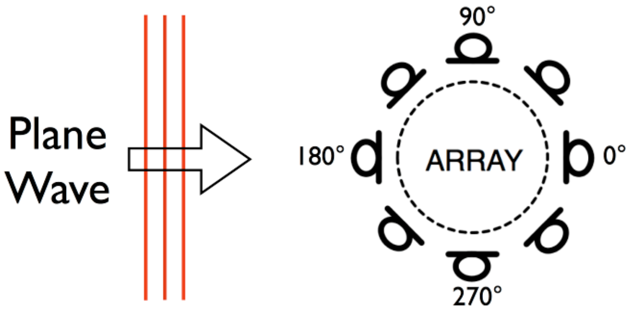



***Figure 9*** *Schematized top view of a plane wave impact to a spherical array to produce the responses depicted in* ***Figure 10*** *and* ***Figure 11****. The wave arrives at the back side of the array (180° azimuth).*

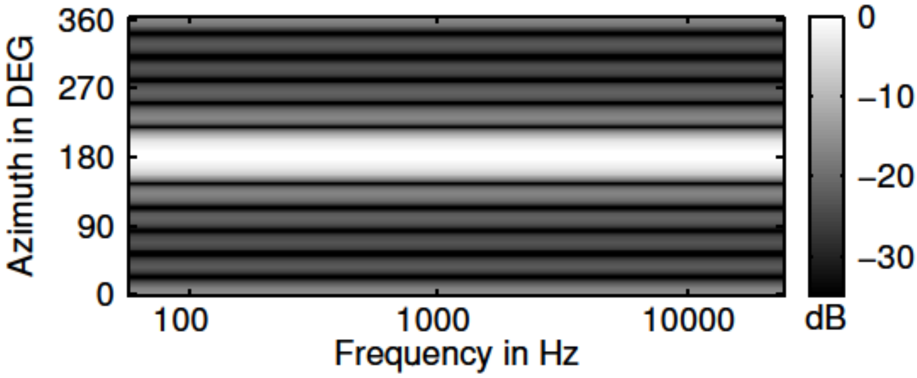



***Figure 10*** *Circular azimuthal array response magnitude over the frequency to a simulated plane wave arriving from 180° azimuth at an ideal array working at a decomposition order of N=5. The array is called ideal, even if the order is truncated, as no discrete spatial sampling is involved and the modal radial filters are allowed to avail unlimited gain to maintain the constant directivity at low frequencies. In practice, microphone arrays have only discrete sampling positions and due to the noise in the signal chain, the gain of the radial filters must be limited.*

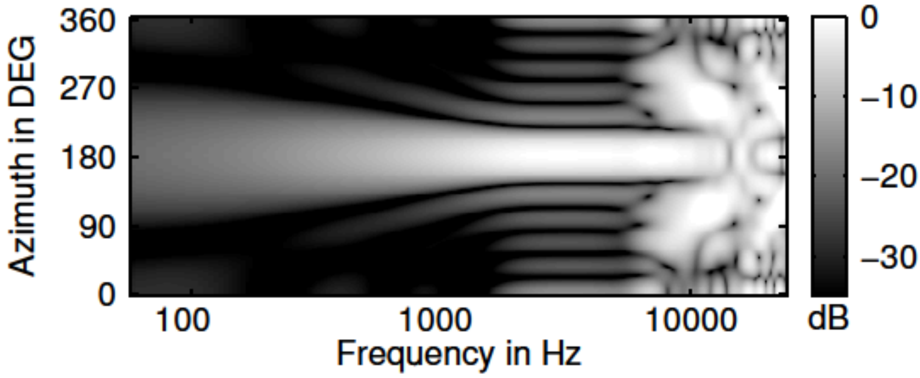



***Figure 11*** *Simulation of an array with some typical limitations in practice. The simulated array has a diameter of 17.5cm and involves 86 discrete spatial sampling positions arranged on a Lebedev quadrature grid. By theory this configuration is stable up to a frequency of approximately 4500Hz; spatial aliasing is expected to disturb the output signal for higher frequencies and can be observed in the figure. The radial filters are limited to a maximum gain of +10dB using amplitude soft-limiting [14]. The directional selectivity of the array decreases and the constant directivity property cannot be retained towards lower frequencies. The stable bandwidth of the array is reduced dramatically. To increase the bandwidth towards high frequencies, more spatial sampling positions must be involved and to retain a high directional selectivity at low frequencies, the array radius must be increased. The simulations are made using the SOFiA sound field analysis toolbox [15].*

# 3. Array Signal Processing

## 3.1. Impulse Response Acquisition

For the analysis of room acoustics, the acquisition of room impulse responses and the subsequent offline post processing is a good choice. The main advantage over real time excitation, processing and analysis is the considerably improved achievable signal to noise ratio. In contrast to currently applied techniques in room acoustics, the microphone array based analysis usually demands a higher signal to noise ratio coupled with the need of capturing a multitude of impulse responses simultaneously (physical array) or sequentially (scanning array).

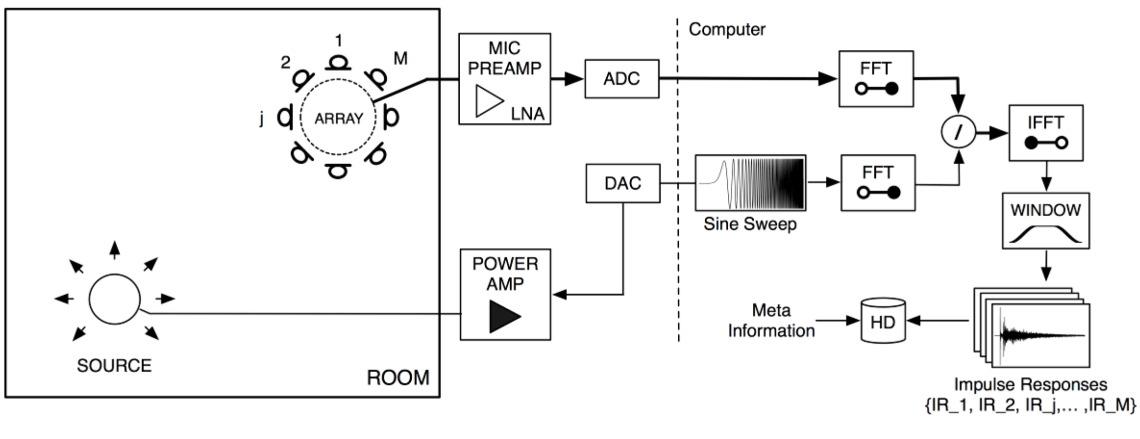



***Figure 12*** *Sequential or simultaneous acquisition of room impulse responses with (colored) sine sweep excitation signals and deconvolution in the frequency domain as proposed in [16].*

The acquisition of the sound field employing a scanning array (e.g. as depicted in **Figure 3**) can take some minutes up to several hours, depending on the demanded node density and target signal to noise ratio. In contrast, physical arrays (e.g. as depicted in **Figure 4**) can acquire the data within seconds, though at the cost of the typical constraints concerning the limited node density. The impact of spatial undersampling due to low node densities was discussed in the last sections. The concrete approach of capturing the impulse responses can e.g. be conducted as proposed in [16]; a corresponding exemplary block diagram is depicted in **Figure 12**.

### 3.2. Plane Wave Decomposition

The basic signal processing for a plane wave decomposition of the captured sound field is shortly introduced in the following. An illustrative block diagram is depicted in **Figure 13**. First, the captured array impulse responses from section 3.1 are transformed to the frequency domain by application of a usual Fourier transform from the time to the spectral domain, as the array processing is performed in the frequency domain. The spatial Fourier transform is applied, transforming from the space domain to the spherical wave spectrum domain involving the SHs (**Figure 5**) as orthonormal base functions.

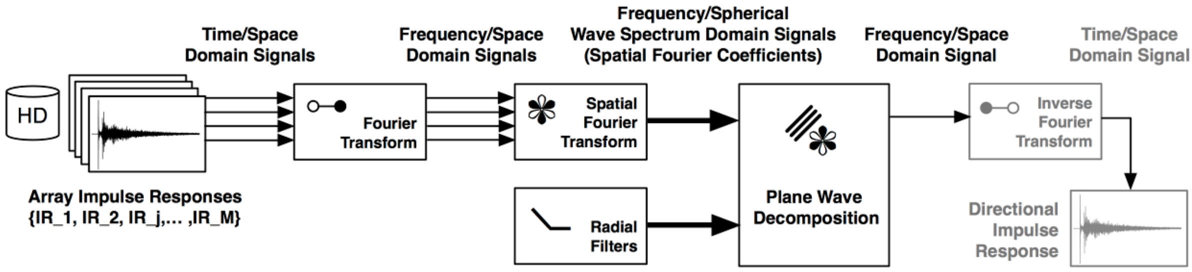



***Figure 13*** *Array processing for the plane wave decomposition*

The subsequent plane wave decomposition is performed in the spherical wave spectrum domain by an appropriate summation of spatial Fourier coefficients for the indented array steering direction and involving specific radial filters that compensate for the natural mode strength and the respective physical configuration of the microphone array. The plane wave decomposition inherently involves an inverse spatial Fourier transform and hence the resulting output is a signal in the space and frequency domain. By application of a usual inverse Fourier transform from the spectral domain to the time domain, a directional impulse response can be obtained. The directional impulse response can e.g. be useful for auralization purposes. In the present paper, the processing chain, depicted in **Figure 13**, is only used up to the plane wave decomposition, as the CAD reflection analysis and the panoramic reflection view that are presented in the following are both based on intensity distributions in the frequency/space domain.

### 3.3. CAD Reflection Analysis

The CAD based reflection analysis is an exemplary application of the presented techniques and combines a 3D CAD model of the respective location on the one hand and a geometric description of reflections that can be obtained by application of SFA methods on the other hand. The involved CAD model can be very basic and reduced in detail, as it only serves for orientation purposes and is not required for any kind of acoustic simulation algorithm. The reflection analysis is based on SFA methods, actually on the previously introduced plane wave decomposition technique. Some of the relevant processing steps were shortly discussed in the last section; important differences are the temporal segmentation (temporal windowing) of the impulse responses on the one hand and the introduction of a spherical steering grid on the other hand. The temporal segmentation serves for the exploration of the reflection incidence time and the spherical steering grid for the exploration of the incidence direction. Accordingly, the size of the time segments determines the temporal certainty of the reflection analysis and the density of the steering grid determines the directional accuracy. The number of plane wave decompositions to be performed depends on the density of the steering grid. The time-segmented multiple plane wave decomposition operation delivers a spatiotemporal intensity distribution matrix, which must be analyzed in order to detect and rate incident reflections. Reflections produce local maxima in the respective spatiotemporal intensity distribution. For this purpose, an analysis algorithm is applied to detect and rate the local maxima in the matrix. The output of the algorithm can be a simple geometric description of the reflections and contain e.g. times, levels and directions. This geometric description can subsequently be visualized in the CAD model as will be presented in section 4, **Figure 22** and **Figure 18** or be passed to room acoustics simulation tools. Similar approaches for the reflection analysis based on impulse responses have been proposed and investigated e.g. in [17][18][19].

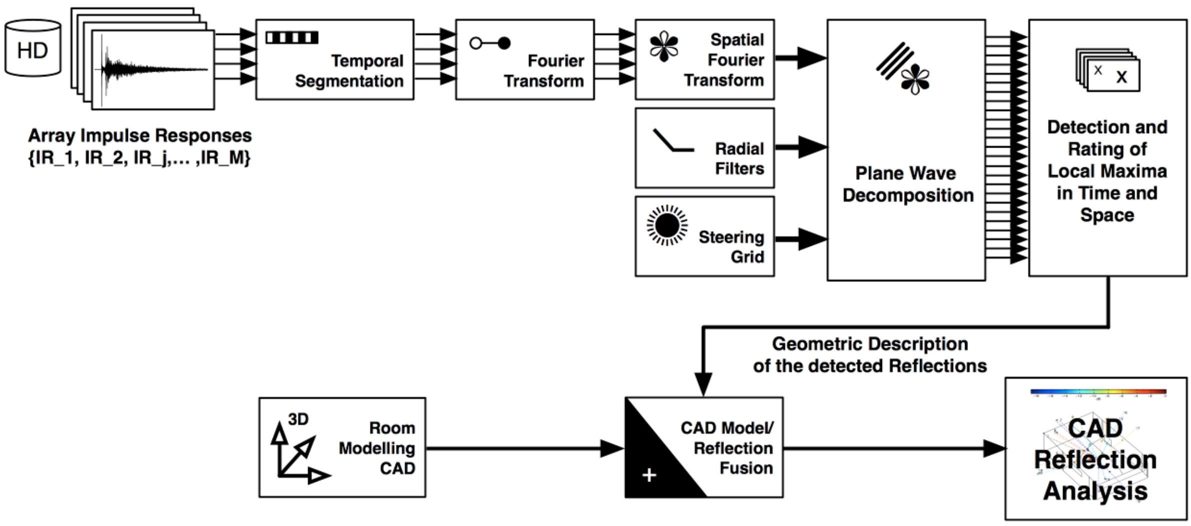



***Figure 14*** *Overview of the processing for the CAD reflection analysis*

## 3.4. Panoramic Reflection View

Another useful application for the analysis of room acoustics comes into use, when the methods of sound field analysis and spherical panorama photography are combined. This allows for a visual study of wall reflections in a studio control room or concert hall for example, leading to vivid illustrations that besides bringing along a substantive analytic quality can be quite useful for a picturesque management or business client presentation.

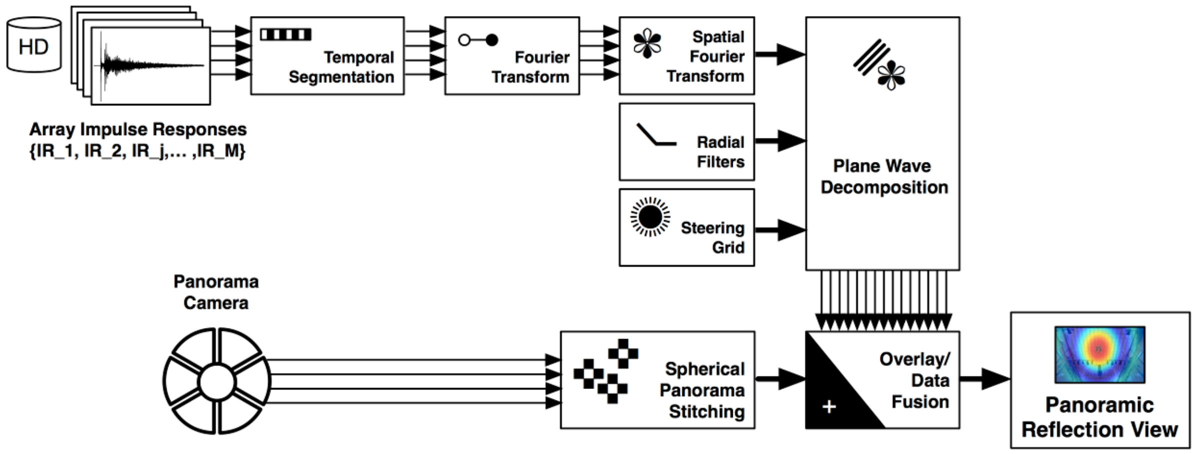



***Figure 15*** *Overview of the processing for the panoramic reflection view*

The resulting images are similar to those of an acoustic camera. But while classical acoustic cameras are mainly used to observe active noise sources (motors, air flow noise etc.), the methods presented here are applied to observe passive systems (e.g. control rooms or concert halls) based on impulse response measurements. This approach brings along a very high achievable signal to noise ratio on the one hand and enables the precise separation of the exciting source and the dedicated system response (i.e. room response) on the other hand. The respective room is excited intentionally, playing a known stimulus through a sound source and subsequently the array impulse response is captured and processed as discussed in section 3.1. Again, a plane wave decomposition of the sound field is used in this context. This technique brings along the useful property of constant directivity over the frequency (always referring to a limited stable band, depending on the array configuration in practice, compare **Figure 11**) and a predictable level distribution. In contrast to the presented SFA technique with spherical microphone arrays and full sphere panoramas, most of the acoustic cameras are based on delay-and-sum beamformers and bring along ambiguous results concerning directions and sound pressure levels. To analyze and observe the reflections in a room, microphone array impulse responses and spherical panorama images must be captured around an identical pivot point either sequentially (scanning micro-phone/camera arrays) or at the same time (real physical microphone and camera arrays or even combined arrays). The temporal segmentation as well as a spherical steering grid have already been introduced in the last section. The temporal segmentation is needed in order to observe different time intervals of the response. The spherical steering grid defines the optical image resolution of the visualization in this case, as the points of the steering grid directly define the pixels that are involved for visualization. The output of the plane wave decomposition is a spatiotemporal intensity distribution matrix once again. The absolute values of the intensities are mapped to a color scale and assigned to the corresponding angles of the steering grid for visualization. The spherical panorama typically consists of several conventional images. An important operation is the stitching [20], which consists of a combination of the single images to a seamless and continuous full sphere panorama. In the present case, the stitching is based on a geometric model instead of the typical feature point detection that is most often used. The full panorama and the spatiotemporal intensity distribution must be overlaid in a data fusion stage, which manages the angle projection and enables to cross-fade between the panorama image and the colored intensity distribution layer. Typically the resulting output shows e.g. a single frequency band and allows for a dynamic observation of the temporal progression and to tilt, pan and zoom the observed segment. Similar approaches for the combination of microphone arrays and optical cameras in order to visualize acoustic reflections have been presented in [21][22][23].

# 4. Application examples from the production studios of the WDR

During the summer of 2012, the authors conducted impulse response measurements in different production studios and control rooms of the WDR radiobroadcast studios located in Cologne, Germany [24]. The resulting compilation[1] is freely accessible to the scientific community e.g. for research and education purposes under a Creative Commons license. In the following, some examples for the application of SFA in room acoustics are presented, which are based on the respective WDR dataset containing impulse responses, CAD data and spherical panorama photographs. The full SFA processing is done using the SOFiA sound field analysis toolbox[2] for MATLAB [15] that is also freely available under a GNU GPL license.

## 4.1. Measurement setup and configuration

The measurements were conducted using a VariSphear scanning array system [5] in a rigid sphere configuration, sampling on a 110 nodes Lebedev quadrature at a diameter of 17.5 cm. The radial filters are limited to +10 dB and the underlying plane wave decomposition is resolved to the order N=5. For the following analysis, a center frequency of around 2000 Hz is observed, which lies in the optimum frequency range for this array configuration. The origin is defined to be the center of the array, representing the listener's point of view. In the broadcast studios, a high power omnidirectional wideband dodecahedron [25] on stage was used for excitation. Besides the main purpose consisting in capturing the array room impulse responses, the VariSphear system allows for capturing a CAD model of the venue by employing a laser sensor. This CAD model is used for the reflection analysis. Furthermore, the scanning array system carried a camera (Canon EOS 5D) to acquire spherical panoramas from the origin in a separate run. The panoramas each involve 98 pictures with a focal distance of 50 mm. The geometrically stitched spherical panorama sets are merged with the SFA data and enable the visualization of single acoustic reflections as discussed in section 3.4., using a proprietary viewer called GIXEL that was developed by a student project group at the Cologne University of Applied sciences in 2012. For more specific information on the measurements, e.g. concerning the locations or the involved technical equipment, the reader is referred to [24].

## 4.2. Introduction to the examples and graphical representations

### *4.2.1. Pictures and Omnidirectional Impulse Response*

In the following examples, initially some photographs are depicted to convey an overall impression of the respective location. Subsequently, extracts of corresponding omnidirectional impulse responses are shown. The omnidirectional microphone capturing the impulse responses was always exactly placed at the geometric origin. Within the plots of the impulse responses, several exemplary numbers appear and demark certain temporal regions. The

[1] Download: **http://www.audiogroup.web.fh-koeln.de**

[2] Download: **http://code.google.com/p/sofia-toolbox/**

numbers correspond to the numbers in the CAD reflection analysis view and the panoramic reflection view. Both are introduced in the following.

#### 4.2.2. *CAD Reflection Analysis*

In **Figure 18** and **Figure 22**, simplified three-dimensional CAD models of the rooms are depicted. The first dominant reflections are delineated inside the CAD models. Each reflection is represented by a numbered and colored ray, pointing from the origin towards the three dimensional spatial direction of incidence. The numbering is chosen according to the time of impact. Thus a higher number means a later impact of the corresponding reflection. The ray number one demarks the direct sound emitted from the source. Furthermore, each ray carries a small bubble climbing along the ray. The distance of the bubble to the origin is directly proportional to the time of incidence. Hence a ray with a globe that is close to the origin represents an early reflection impact. The color of the ray represents the intensity of the reflection. The dB values at the color bar refer to the level of the direct sound ray marking 0 dB. The sensitivity of the reflection detection algorithm is set very low for the given examples, thus indicating the most salient reflections only. Furthermore, only the first temporal section of the impulse response (80 and 125 ms) is analyzed, to keep the illustration clear. Involving higher detection sensitivities and analyzing a broader temporal section of the impulse response concurrently can raise hundreds of rays and result in a confusing image. A Lebedev decomposition grid involving 3074 Nodes and a FFT block size of 64 samples with a progress of 1 sample are applied for the reflection analysis. This configuration entails a location inaccuracy of approximately 0.75° and a temporal uncertainty of around ±665 µs at a sampling frequency of 48000 Hz. The source position is marked with an “X” in the CAD model; the position of the origin (listener) should be obvious.

#### 4.2.3. *Panoramic Reflection View*

The exemplary images (**Figure 19** and **Figure 23**) show an overlay of the spherical panoramas and sound field analysis data. The plane wave decomposition is resolved for around 65,000 spatial directions (1° on the full globe) and the resulting intensity distribution is mapped and overlaid to the panorama. The clearly visible squares in the SFA visualization ascribe to the 1° grid. The optical grid density can be raised, but the underlying plane wave decomposition for a full spatiotemporal decomposition is computationally demanding, and moreover, a raised optical resolution is obviously not associated with any gain of additional information in this case. The depicted sound field data is normalized to the maximum value that arises in each individual image and does not correspond to a common level reference. The image color scales can alternatively be referenced to a common maximum (typically the direct sound), which is possibly even more meaningful for the practical analysis and rating of the reflections. But the level adapted coloration of reflections turns out to be less vivid in the case of presenting single static exemplary images as depicted here. However, the intention is rather to give a general insight to the prospects of combined processing of panorama images and sound field analysis than to conduct a dedicated analysis of the acoustic properties.

### 4.3. WDR Large Broadcast Studio, "*Klaus von Bismarck Saal*"

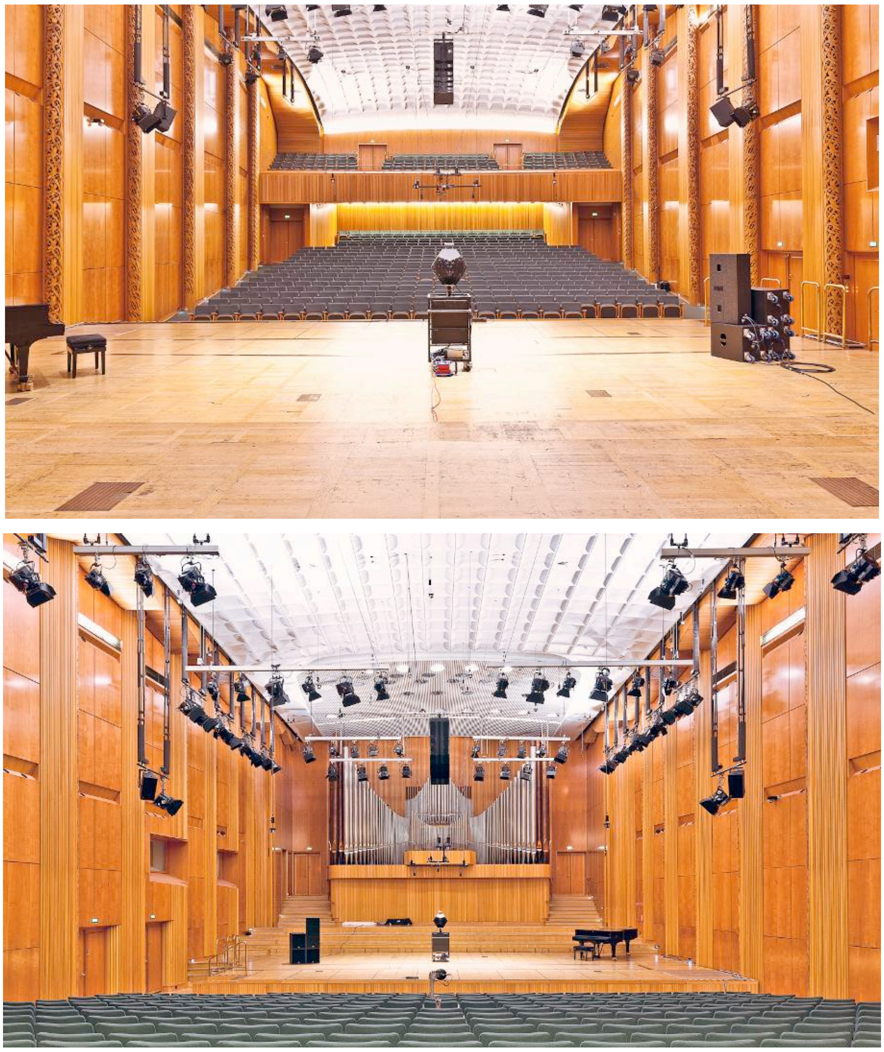


***Figure 16*** *Impressions of the large broadcast studio*

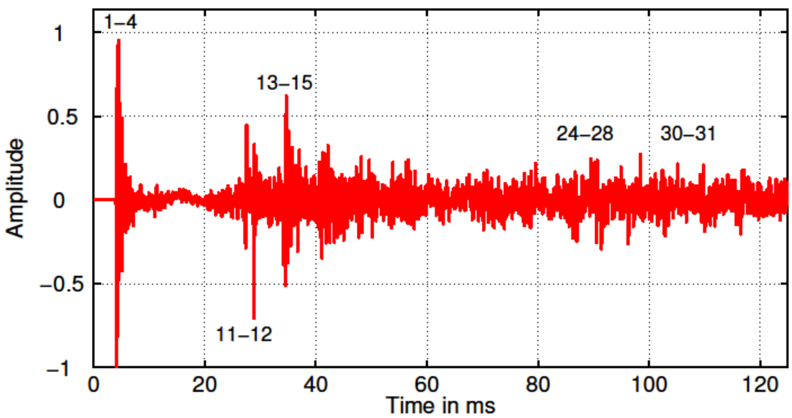



***Figure 17*** *The first 125ms of the omnidirectional impulse response of the large broadcast studio; the indicated numbers correspond to the numbers used in the CAD reflection analysis and the panoramic reflection view that are presented subsequently.*

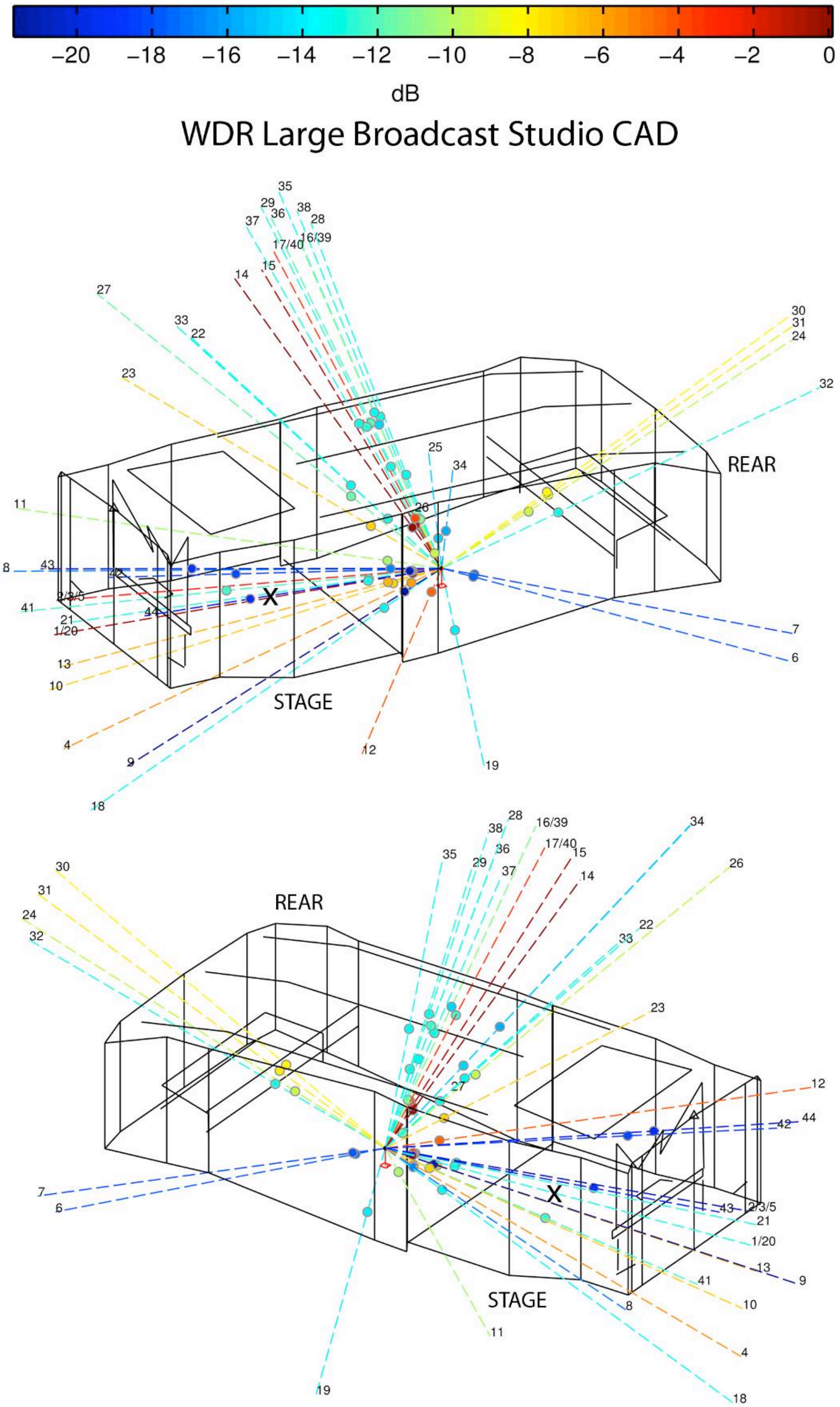



*Figure 18 CAD reflection analysis of the large broadcast studio*

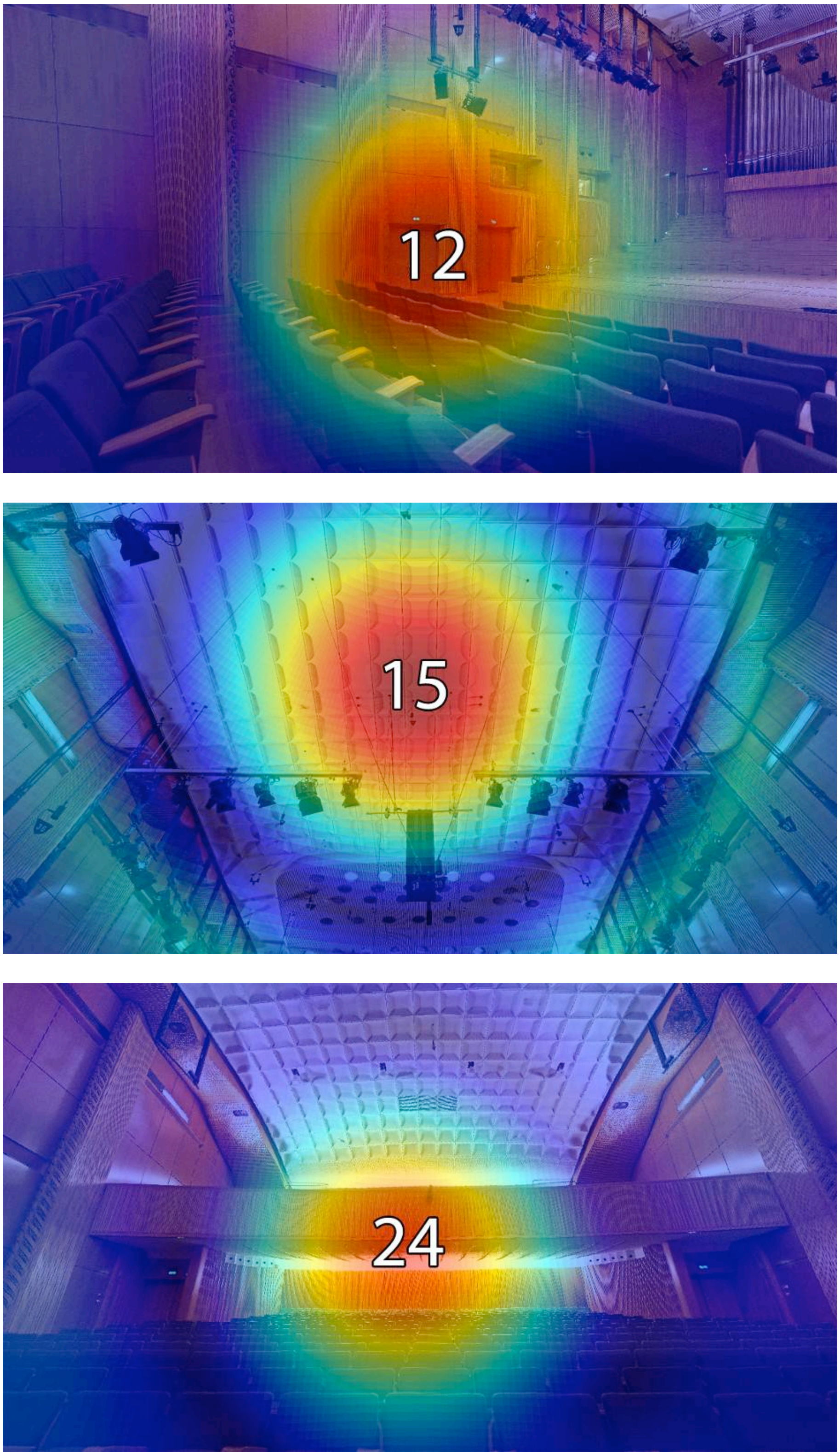



***Figure 19*** *Exemplary visualization of reflections in the large broadcast studio.*

### 4.4. WDR Small Broadcast Studio (SBS)

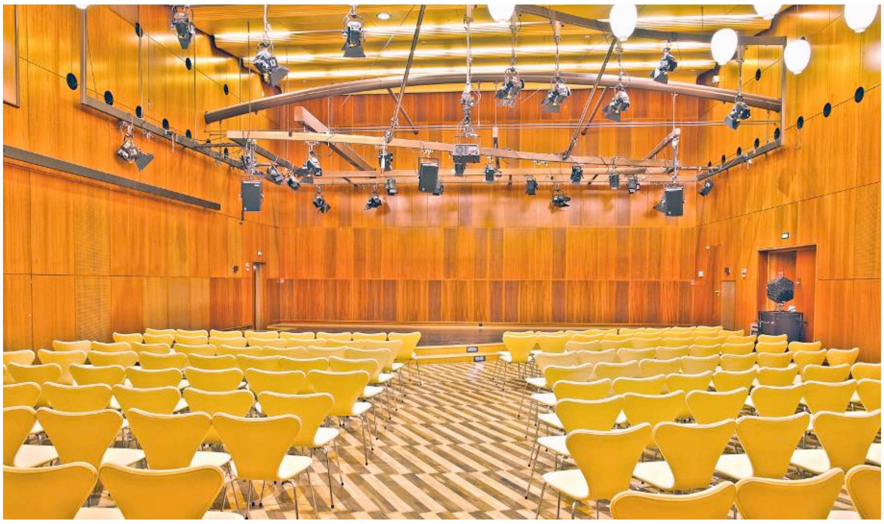


***Figure 20*** *Impressions of the small broadcast studio*

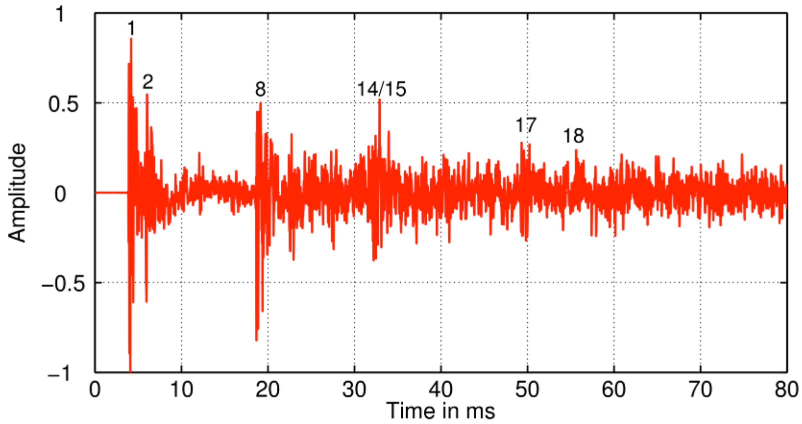



***Figure 21*** *First 80ms of the omnidirectional impulse response captured at the small broadcast studio; the indicated numbers correspond to the numbers used in the CAD reflection analysis and the panoramic reflection view that are presented subsequently.*

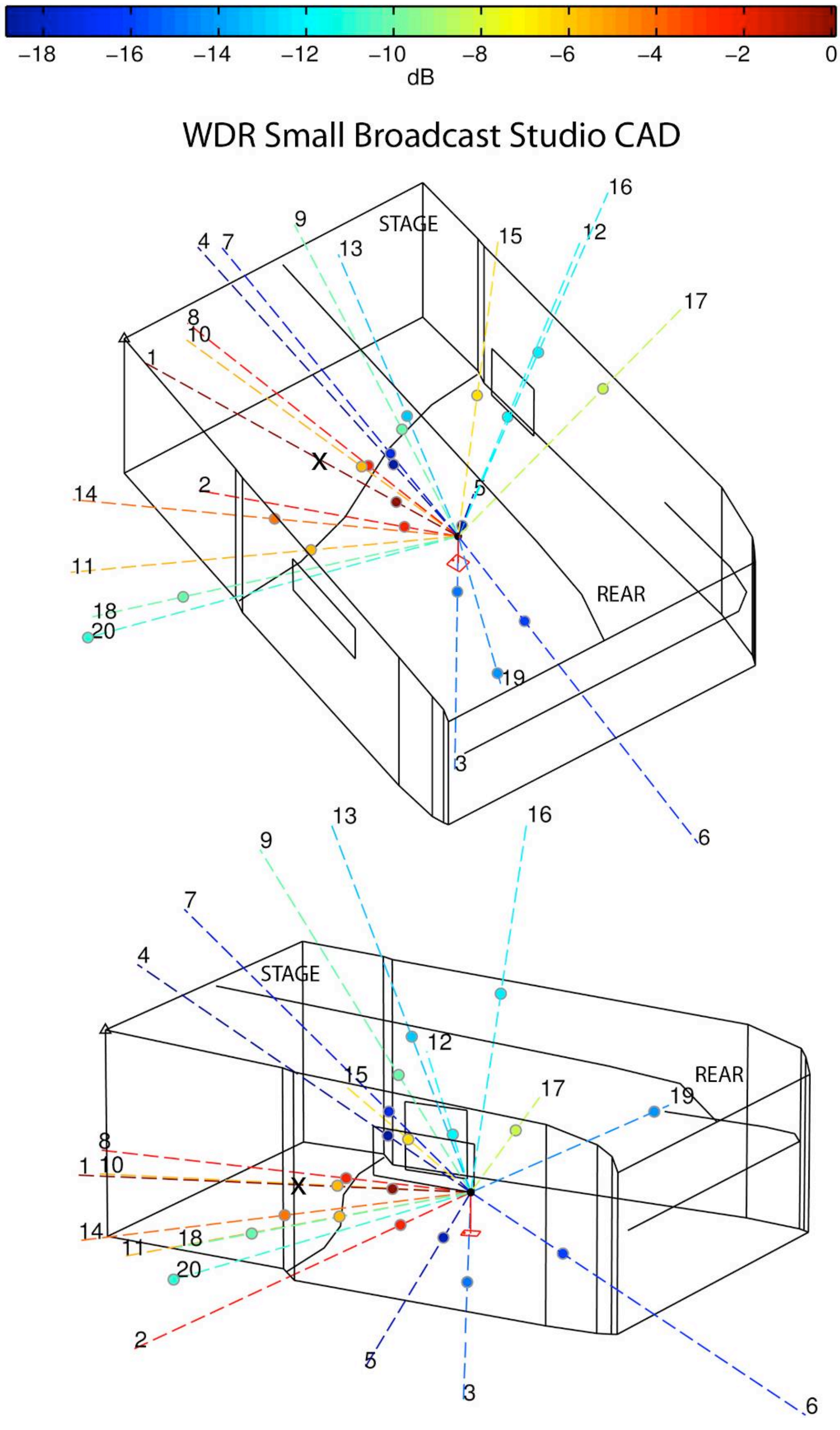
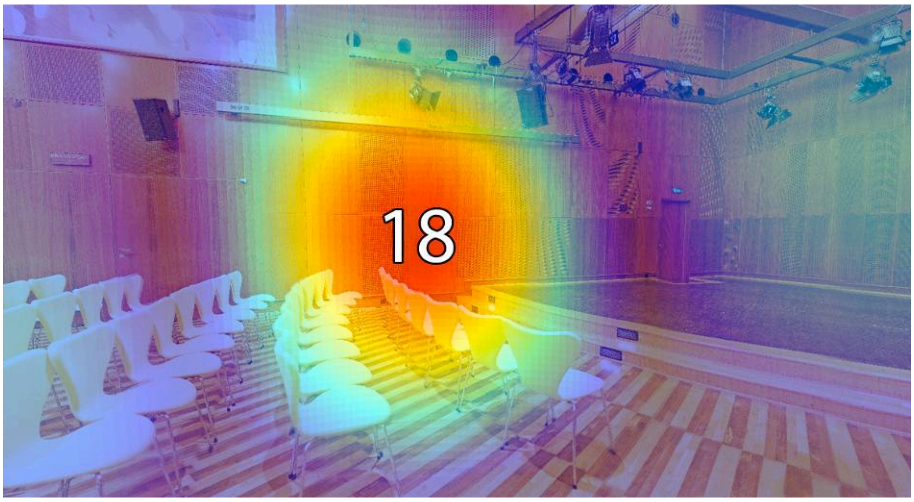



***Figure 22*** *CAD reflection analysis of the small broadcast studio (SBS).*



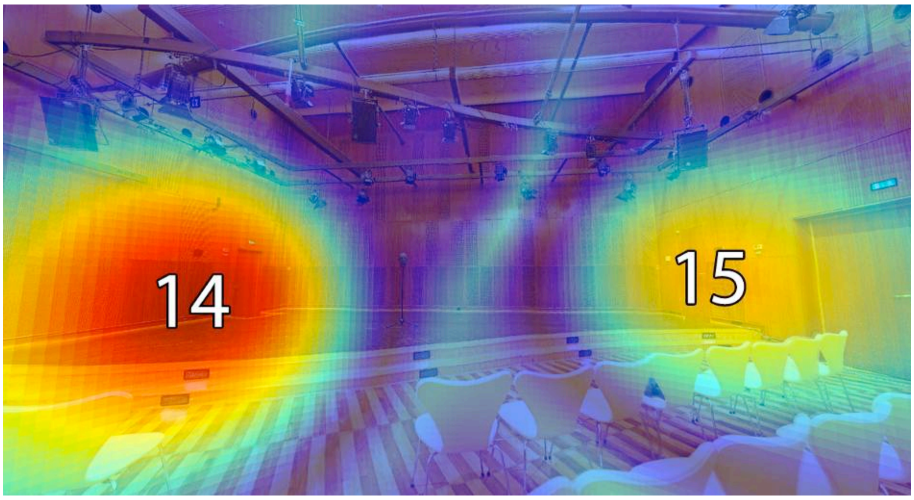




***Figure 23*** *Exemplary visualization of some reflections in the SBS.*

## 5. Further Applications of Sound Field Analysis in Room Acoustics

Besides employing the presented analysis methods for the analysis and optimization of passive room acoustics, it can also serve to adapt and optimize sound reinforcement systems. Another promising application for the SFA methods in room acoustics is the modification of the captured sound field by directional weighting and filtering in the spherical wave spectrum domain in order to emulate constructional changes of the room. Successively, the auralization (e.g. using binaural synthesis) of the modified sound field most likely enables a very plausible prediction of the influence of projected constructional changes on the acoustic properties in a room. These techniques are currently being subject of investigation. Furthermore, the methods of sound field decomposition generally entail a high degree of additional information content that possibly might be useful for the analysis and prediction of room acoustic qualities. The substantially improved spatial resolution could be employed for the definition of new instrumentally acquirable criteria to estimate the perceptual quality of a room or to extent and improve the existing criteria that are e.g. listed in section 1. Different investigations concerning this topic have been presented e.g. in [26] or [27].

## 6. Conclusions and Outlook

A basic introduction to the methods of sound field analysis using microphone arrays and some examples for applications in the field of room acoustics have been presented. Currently, these methods can still be considered as slightly advanced concerning the practical application. The technique as a whole is considerably more complex than typical state-of-the-art techniques used in room acoustics today. Currently no commercial hard- and software is available for that purpose. But it can be expected to be only a matter of time for these methods to find the way into the tool chain of acousticians. Especially if the SFA methods and room acoustics simulation tools are melted, which indeed is an obvious next step, the advantages and possibilities for the analysis of room acoustics can be considered as excellent.

## 7. Acknowledgements

We would like to thank Markus Haßler and Bendikt Bitzenhofer from the WDR radio production department for their outstanding hospitality and support. Further we like to thank the GIXEL project group of the Cologne University of Applied Sciences that besides Philipp Stade (co-author) consists of Julian Achatzi, Klaus Hesse-Camozzi, Karsten Stein, Leonie Kirk and Ramona Haas, for the design and implementation of the spherical viewer in the scope of a student project at the Cologne University of Applied Sciences.

We particularly thank Christoph Pörschmann, professor for acoustics and audio signal processing at the Cologne University of Applied Sciences and head of the research group for his valuable contributions and ideas, and for making all of the presented work possible.

## 8. Funding

The research activities are funded by the Federal Ministry of Education and Research in Germany under the support codes: 1707X08-ASAR and 17009X11-MARA. We appreciate the support.

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