ADVANCEMENTS IN 3D AUDIO USING AMBISONICS

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ABSTRACT

This contribution reviews the basics of Ambisonics and outlines examples of applied Ambisonics using microphone and loudspeaker arrays, and headphone-based rendering. Recent achievements will be addressed in Ambisonic software, panning and decoding, and the observation and optimizing of perceptual properties. Moreover, another highly topical field of higher-order Ambisonic recording is discussed, its fundamental spatial resolution versus SNR and bandwidth problem, a new filtering approach, and spatial mastering effects.

1. INTRODUCTION

It was nearly 15 years ago, when higher-order Ambisonics became a research question at our lab. This was caused by the necessity in advant-garde music theatre, where defined sound objects should be moved on fixed trajectories around the audience at various venues. Ambisonics was considered to be a good starting point, because it is a 3D recording and playback method that is based on the representation of the sound field excitation as a decomposition into spherical harmonics. This representation facilitates spatial sound production that is independent of the playback system. The adaptation to a given playback system (loudspeakers or motion-tracked headphones) is achieved by a suitable decoder.

This contribution gives an overview of the current state-of-the-art in Ambisonics including content production using Ambisonic main microphone arrays or panning of virtual sources, spatial effects, and reproduction by loudspeakers and headphones. The software for the whole production chain is already available as a VST-plugin suite for digital audio workstations.

Ambisonics [1, 2] has a fourty-year tradition now in producing 3D audio content. Its potential nowadays to become one of the main production technologies might be due to this long tradition. Its original recording technology was conceived to be of low spatial resolution, as one could only manufacture first-order gradient transducers of high audio quality. This limitation made a perceptually stable spatial playback hard to accomplish, but the ten-to-twenty years use of higher order Ambisonics [3, 4, 5, 6] could be shown to be free of tight resolution deficits.

Still, the area around the center of practical higher-order Ambisonic systems, the *physical sweet spot* in which a sound field can be recreated phyically accurately, is limited to few centimeters at high frequencies [7]. Nevertheless, the *perceptual sweet spot* covers most of the listening area when employing Ambisonics as an amplitude-panning method [8]. In this case, the advantage of higher-order Ambisonics lies in the perceptual effect of a phantom source [9] rather than a physically accurate sound field synthesis [10]: At low frequencies, the level differences between the loudspeakers create time differences between the ears [11] which are the dominant localization cues in the horizontal plane [12].

Why the time is now to start producing in Ambisonics:

We not only have all scientific evidence that it works [13, 14], but also we finally have the higher-order recording equipment [15] and user-friendly tools:

Free VST-plugins [16] that largely simplify producing Ambisonic content were released and awarded the gold medal of the AES Student Design Competition 2014, Berlin. Moreover, a tool to prepare arbitrary loudspeaker layouts for playback is available for free: the Ambisonic Decoder Design Toolbox [17]. Last but not least, the MPEG-H standard is on its way which will support distributing higher-order Ambisonic content.

This paper first gives an overview of Ambisonics in terms of its basics, signal flow, and perceptual aspects. The subsequent sections introduce first- and higher-order main microphone arrays for Ambisonic recordings and spatial effects in the Ambisonic domain. Finally, the paper presents current decoding strategies for Ambisonic playback on arbitrary loudspeaker arrangements and headphones.
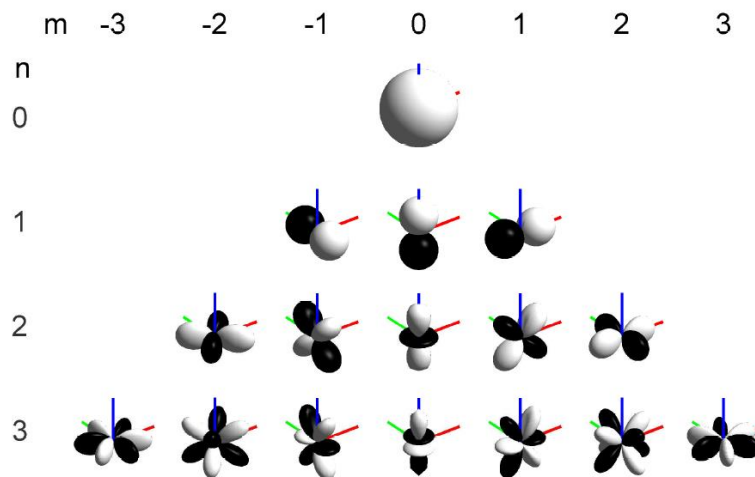

Fig. 1: First 16 spherical harmonics with order *n* and degree *m*.

2. BASICS OF AMBISONICS

Ambisonics [18, 1, 6, 3] is based on the representations of the sound field excitation in terms of orthogonal basis functions. In the three-dimensional case, these functions are the so-called *spherical harmonics*, cf. Figure 1. Their maximum order N determines the spatial resolution, the number of channels, and the minimum number density of loudspeakers required for reproduction.
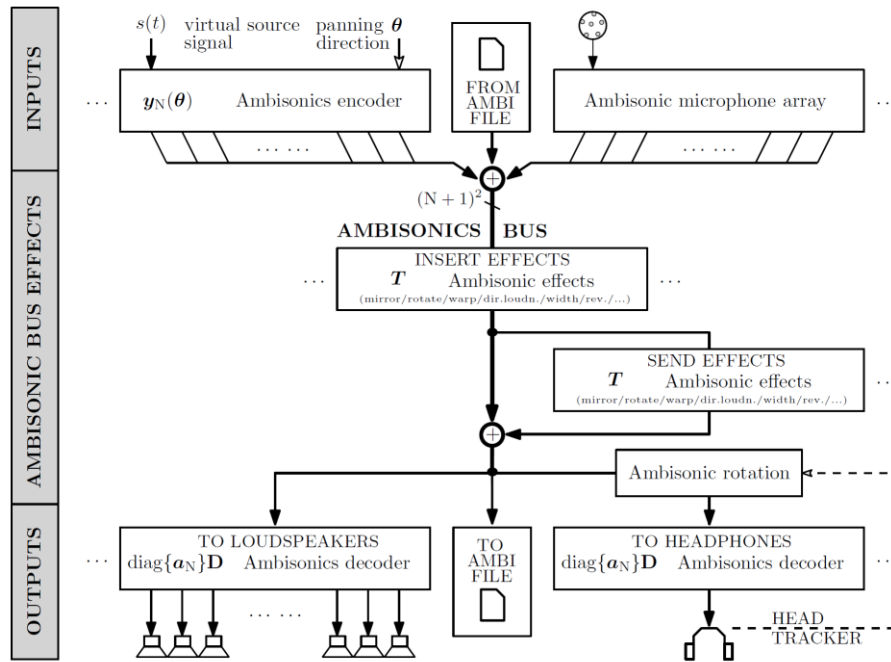
Fig. 2: Signal flow of Ambisonics including inputs (Ambisonic encoder, microphone arrays, files), Ambisonics bus with effects (insert or send), and outputs (loudspeaker signals, head-tracked headphone signals, files).

2.1. Signal Flow

The signal flow of Ambisonics is exemplarily shown in Figure 2. The spherical harmonics representation allows for the application of a bus system. The channel count $(N+1)^2$ of this bus is determined by the maximum order N and does not depend on the number of sources. For a single source panned at a direction $\boldsymbol{\theta}$, $\boldsymbol{\theta} = [\cos(\varphi)\sin(\vartheta), \sin(\varphi)\sin(\vartheta), \cos(\vartheta)]^T$ with the azimuth φ and zenith angles ϑ, the Ambisonic spectrum $\boldsymbol{y}_N(\boldsymbol{\theta})$ is calculated frequency-independently by evaluating the spherical harmonics at $\boldsymbol{\theta}$. As alternative inputs, Ambisonic microphone arrays (see section 3) or pre-produced Ambisonic files can be employed.

All inputs are summed up in the Ambisonics bus that can be modified by either multichannel insert or send effects. More details on the spatial Ambisonics effects can be found in section 4. Note that effects are not bound to the Ambisonics bus, but they can also be applied directly to the inputs.

For playback, the final channels of the Ambisonics bus can be either decoded to loudspeakers and head-tracked headphones, or stored in a file for playback on arbitrary systems. The decoder derives the signals $s_l(t)$ for the L loudspeakers of an arrangement from the Ambisonic bus by multiplication with the order-weighted, real-valued and time-invariant decoder matrix $diag\{\boldsymbol{a}_N\}\boldsymbol{D}$. The matrix $\boldsymbol{D}$ is derived from the spherical harmonic spectra $\boldsymbol{y}_N(\boldsymbol{\theta}_l)$ of each (virtual) loudspeaker $\boldsymbol{Y}_N = [\boldsymbol{y}_N(\boldsymbol{\theta}_1), \boldsymbol{y}_N(\boldsymbol{\theta}_2), \dots, \boldsymbol{y}_N(\boldsymbol{\theta}_L)]$. More details on decoder design strategies are shown in section 5.

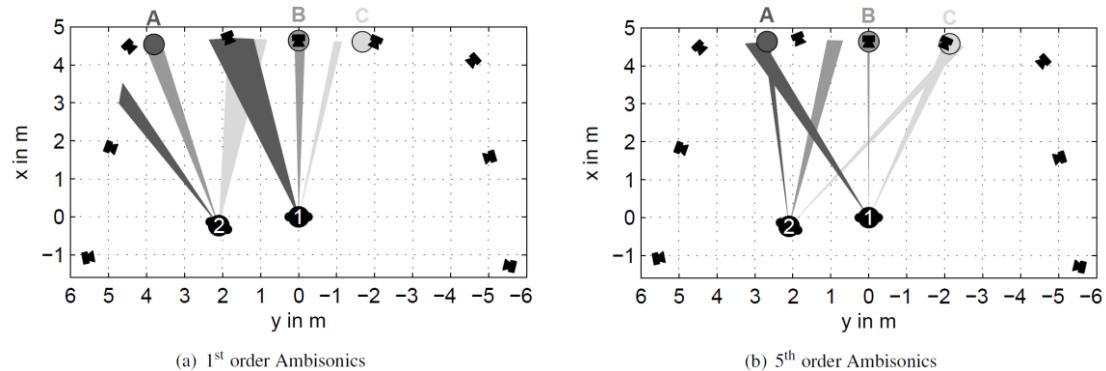(a) 1st order Ambisonics   (b) 5th order Ambisonics

Fig. 3: Confidence intervals (gray beams) of the median localized directions in a listening experiment [19] for 3 panning directions A, B, C (gray filled circles), and 2 listening positions 1, 2.

2.2. Order and Order Weighting

In comparison to vector-base amplitude panning [20] (VBAP) that does not have such a modal representation, Ambisonics is designed to enable playback with direction-independent, smooth quality. Smoothness is gained by giving up on the accurate localization achievable with a single active loudspeaker. The match between panning direction and perceived direction increases with the Ambisonics order, cf. Figure 3 [19]. This is even more evident for off-center listening positions. Besides the increased match, the variance of the experimental results decreases. Truncation of the spherical harmonic series to N yields disturbing side lobes that can be attenuated by appropriate weighting $\boldsymbol{a}_N$, cf. [6] and Figure 4. *Inphase* weighting completely suppresses side lobes at the cost of a wide main lobe. As a trade-off, the $max - \boldsymbol{r}_E$ weighting [4] maximizes the energy towards the panning direction and achieves the best localization and least coloration at all listening positions [19, 13, 21, 8]. For higher-order Ambisonic microphone arrays, different order weighting should be applied in sub-bands, cf. Section 3.2.

3. AMBISONIC MAIN MICROPHONE ARRAYS

3D audio production appears to be rather complicated, especially in cases of real audio scenes with a lot of maybe moving sources at various directions including height. Typical two-channel stereo or 5.1 main microphone recording technology will not map all the information correctly, and using many spot microphones can be challenging when all spatial and timbral aspects should be represented adequately.

In this case, Ambisonics offers a great advantage as a 3D audio production technology: There are various main microphone arrays that can be emloyed for recording spatially rich audio scenes, spatial music, or rooms with interestingly complex reverberation.

3.1. First Order

The first-order Ambisonic microphone array technology has become about forty years old [22]. It consists of four coincident cardioid microphones oriented in different directions, angled by 70.5º between any pair. The angular resolution is therefore not large, but the low number of microphones allows to employ highest quality microphones.

By now, beside the original manufacturer Soundfield/TSL Products (DSF-2, SPS422B, ST450, SPS200), also other companies such as Core Sound (TetraMic) and Oktava (4-D/MK-012) manufacture first-order Ambisonic microphone arrays.

In productions, the four channels of a first-order Ambisonic microphone deliver a well-balanced representation of ambience. However, the resolution is often not sufficient for direct sound, hence strategies such as mixing with spot microphones or the enhancement of the spatial resolution are often favorable.
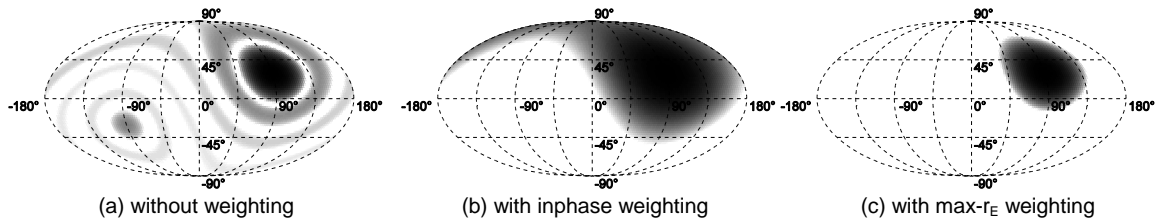
| (a) without weighting | (b) with inphase weighting | (c) with max-r_E weighting |

Fig. 4: Mollweide projection of the spherical gain distribution (25dB dynamic range) for the 5th-order source at 90° azimuth and 30° elevation for different weightings.

3.2. Higher Order

There is one higher-order Ambisonic microphone array commercially available at the moment, and it allows to capture stand-alone spatial recordings of high timbral and spatial definition: the Eigenmike from mhacoustics [15, 23], which achieves up to fourth-order resolution.

Research consistently indicates the achievable perceived spatial quality of such arrays [24, 25], and there have been quite impressive 3D Audio demonstrations from researchers in Paris [26, 27], Parma [28], and Graz[1,2] [16]. In these demonstrations, processing gradually reduces the spatial resolution towards low frequencies, see [29, 30, 31]. To avoid disturbing side lobe localization at all frequencies, different $max - r_E$ weighting is required for each sub-band [32, 33].

In contrast to electronically steered beams of similarly frequency-dependent directivity, higher-order Ambisonic recording does not seem to entail the necessity of diffuse-field or free-field compensation.

3.3. Spatial Resolution Enhancement

To enhance the spatial definition of first-order Ambisonic recordings, two prominent approaches deliver useful results. Directional audio coding [34] (DirAC) reassigns a higher directional resolution to each frequency band of the recording by the intensity vector, whenever the intensity indicates a non-diffuse sound. Similarly, high angular resolution plane wave expansion [35] (HARPEX) assigns two plane wave directions to each frequency band. These approaches successfully enhance the information present in the recording, but they cannot exactly map more than one (DirAC) or two (HARPEX) non-diffuse simultaneous sources within one frequency band.

Despite the resolution of higher-order recordings is already much better in presenting direct sounds, further enhancement is achievable. There is a higher-order version of DirAC [36]. Moreover, an elegant super-resolution sound field imaging technique was presented by Epain [37]. It is based on subspace pre-processing that enables separation of direct from diffuse sound. Compressive sensing is applied to enhance the direct sounds.

These enhancement approaches are based on decomposition of the recording into virtual source objects. Such objects contain a sound signal and spatialization parameters such as direction and diffuseness.

4. SPATIAL AMBISONIC EFFECTS

When using object-based formats, the application of spatial effects is simple, because it is accomplished by changing the parameters of the object. However, a comprehensive set of spatial effects available as VST-plugins [16] can still be applied to the Ambisonics bus, which can contain several virtual source objects.

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[1] http://www.matthiaskronlachner.com/wp-content/uploads/2013/01/AES_2014_Kronlachner.pdf
[2] http://www.matthiaskronlachner.com/wp-content/uploads/2013/01/kronlachner_aes_studentdesigncompetion_2014.pdf

4.1. Basic Effects

The Ambisonic representation as decomposition into basis functions allows for some basic spatial effects. Inverting (phase-reversing) certain channels *mirrors* the virtual sound scene along the Cartesian coordinate axes (front/back, up/down, left/right). *Rotation* requires to run the Ambisonic signals through a matrix[3].

Reordering and scaling of the Ambisonic channels can be employed in order to *convert* into different Ambisonics formats [38]. Typical non-spatial effects, such as equalization or delay, are applied to all Ambisonic channels as multichannel filters.

4.2. Advanced Effects

Warping [39,40] moves all sources of a sound scene away or towards the poles or the equator, cf. Figure 5. This is often useful for the adjustment of main microphone recordings. Similarly, *directional loudness modification* [40] can be applied to increase or decrease the level of arbitrary spatial areas. This effect allows for the level adjustment of single sound sources within the main microphone recording.

The control of salience can also be achieved by *widening* of selected sources. The Ambisonic widening algorithm presented in [41] is an efficient way of doing so by filtering without introducing much coloration. It provides a larger sweet spot compared to the 2-channel stereo version [42] using the same loudspeaker spacing, and it can also be adjusted to create early reflections.

The simplest way of creating convincing 3D *reverberation* is the convolution of the source signal with measured Ambisonic impulse responses of well-sounding locations. There is already of number of such impulse response databases available, such as the OpenAIR lib[4].
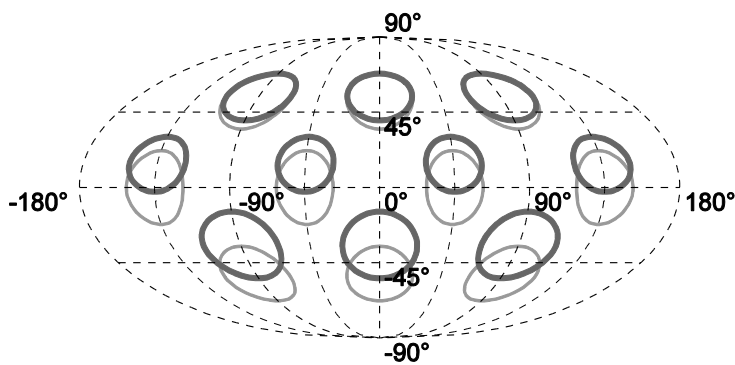

Fig. 5
Mollweide projection of 10 sources warped from their original location (light gray circles) towards the north pole (dark gray circles).

5. DECODING

Ambisonics can be played back on various systems, including both loudspeaker arrangements and headphones. This can be done by a suitable decoder matrix. In section 5.3, practical applications recently presented to a large experienced audience during the 3rd International Conference on Spatial Audience in Graz, Austria are discussed.

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[3] http://ambisonics.iem.at/xchange/format/docs/spherical-harmonics-rotation
[4] freely available at http://www.openairlib.net

5.1 Decoding to Loudspeakers

The traditional strategies of calculating decoders for loudspeaker playback are *sampling* and *mode-matching* [43], using the transposition of Y_N or its right-inverse $Y_N^T(Y_N Y_N^T)^{-1}$. However, such strategies are only suitable for regular loudspeaker arrangements covering the full sphere. For arbitrary arrangements, these strategies fail and yield large localization errors, as well as strong loudness and source width fluctuation [44, 8].

Alternative strategies were presented recently: Non-linear optimization [45, 46] can be employed to optimize perceptual relevant parameters [47, 48, 13, 21]. The *energy-preserving* strategy [49] was proposed for hemispherical loudspeaker arrangements. It reduces the set of basis functions and employs singular value decomposition.

AllRAD [50] is the most flexible strategy. It decodes Ambisonics to an optimal virtual *t*-design loudspeaker arrangement by sampling. This results in a decoder matrix that is mode-matching and energy-preserving at the same time. Signals of these virtual loudspeakers are mapped to the real loudspeakers using VBAP. Figure 6 shows the VBAP triangles for an AURO 13.1 arrangement. A stabilization of sources close to the border of the loudspeaker arrangement is achieved by the additional imaginary loudspeaker below the floor.
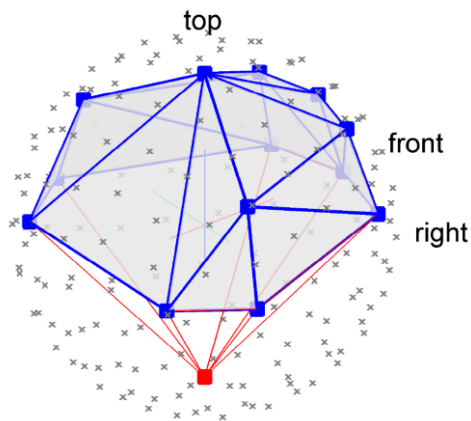

Fig.6
Exemplary VBAP triangulation of AllRAD for AURO 13.1 loudspeaker arrangement (blue squares), imaginary floor loudspeaker (red square), and 180 virtual *t*-design loudspeakers (gray crosses).

Useful decoder strategies are implemented in the freely available decoder design toolbox [17].

5.2 Decoding to Headphones

Headphone playback is practical for mobile applications, however convincing binaural playback requires careful creation of the ear signals. Basically, the spatial impression can be created by convolution of a source signal with the corresponding head-related impulse responses (HRIR) or binaural room impulse responses (BRIR). Using the loudspeaker feeds as source signals can recreate the ear signals of arbitrary loudspeaker arrays in a room.

Improved localization and plausibility is achieved by head-tracking that follows the head movements of the listener [51]. The incorporation of head movements requires sophisticated interpolation of the HRIRs/BRIRs [52].

By contrast, using Ambisonics for headphone playback [5, 53, 54] provides a simple way to involve head movements using rotation matrices, cf. Section 4.1.

5.3 Practical Demonstration and Fine-Tuning

During the ICSA'15 the flexibility of Ambisonics in 3D rendering was demonstrated. Comprehensive 3D audio material was played back via various local and standarized loudspeaker setups i.e. 29.2, 25.2, 22.2, 9.1, and 5.1, as well as 2.0 - in binaural format - for headphones reproduction. Therefore, conference entrants were able to directly compare the same musical content on different rendering setups. As above stated, the *AllRAD* decoding procedure was applied to convincingly demonstrate the modularity and scalability of the Ambisonics approach.

In Figure 7, the standardized 9.1 setup (left), and the local developed mobile Ambisonics Unit (25.2, right) are depicted as representatives for the used loudspeaker layouts. The AURO 3D 9.1 setup complements the standardized 5.1 setup (cf. ITU-R BS.775-1) with 4 additional loudspeakers (upper rig) positioned above the Left, Right, and Surround Left/Right speakers. In both cases, the *AllRAD* Decoder for 5th order Ambisonics, using *t*-Design with virtual Loudspeakers and VBAP to render these loudspeaker feeds on the real loudspeaker setup, was used.
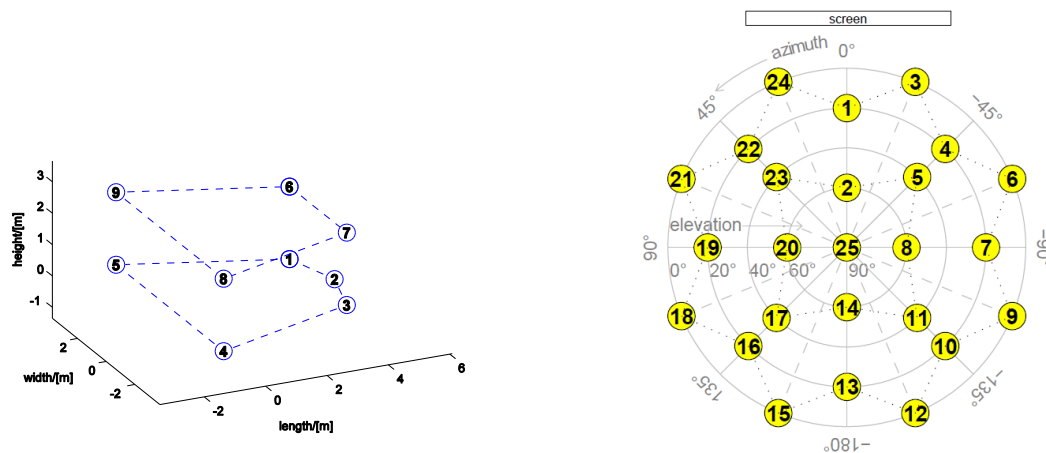

Fig. 7: Loudspeaker Layouts for AURO 3D 9.1 Setup (left), and mAmbA (mobile Ambisonics Unit, 25.2) (right). At the ICSA'15 the 9.1 Setup was built with 9 Lambda Labs CX-1A, and 1x Lambda Labs MF-15A loudspeakers. The mAmbA consists of 25 Genelec 8020, and 2 Genelec 7050 loudspeakers.

To support a 5.1 setup also with height-information, the loudspeaker feeds of the upper rig are filtered with peak-filters and added to the lower rig in the mix-down process. The characteristics of the peak-filters are adjusted to the (Blauert's) directional bands.

In order to support a very large audience (at the ICSA: an auditorium for 200 people), the perceptual sweet spot is optimized/extend by adjusting the compensation delays of the 29.2 loudspeaker arrangement in the Ligeti-Hall on the defined conture of a rectangular (cf. Figure 8, yellow rectangle) within the listening area instead of a singular sweet spot, at the center of the loudspeaker arrangement.

The compensation delay Δ_i of each loudspeaker LS_i is derived as follows:

(1) Search the minimum distance $d_{min,i}$ between each LS_i and the given contour of the yellow rectangle (cf. Figure 8).

(2) Determine the maximum of all distances $d_{min,i}$ i.e.: $md_{min} = \max(d_{min,i})$

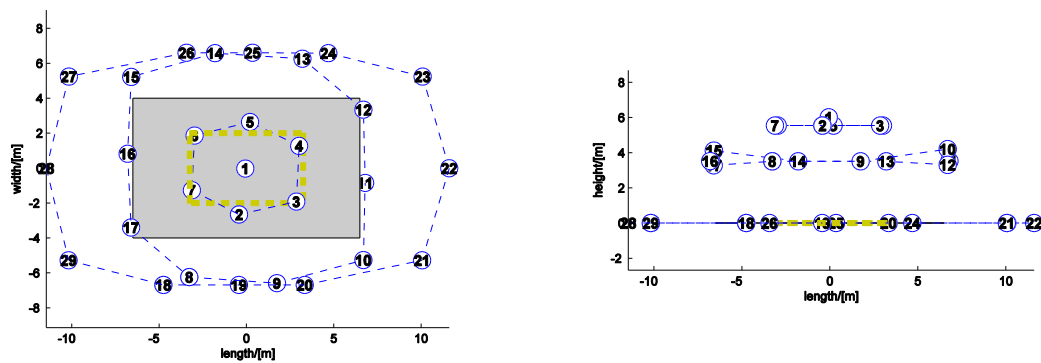(3) Calculate compensation delays: $\Delta_i = md_{min} - d_{min,i}$

Fig. 8: Variable, permanent loudspeaker layout from Ligeti Hall (KUG) with loudspeaker numbers given. Left: horizontal view; Right: vertical view. The gray rectangle depicts the area of the audience. The lowest rig is adjusted lower than the mean height of the ears of the sitting audience (approx. 1m above the audience floor). The yellow lines (rectangle) indicate the optimization contour to derive the individual compensation delays for the loudspeakers.

The permanent loudspeaker setup consists of 29 Kling & Freitag (K&F) CA1001-SP and 2 K&F SW 118E-SP. In order to provide a fair comparison at the ICSA'15, this system was driven using a 5th order Ambisonics decoder using the *AllRAD* approach.

The playback of natural 3D recordings (cf. Section 3.1, and 3.2 respectively) over hemispherical loudspeaker arrangements will cause order-dependet small deviations in regard to the reproduced vertical directions of the sound sources. This drawbacks can be solved by applying order/frequency depended warping effects (cf. Section 4.2) in the decoding stage (see also [55]).

6. CONCLUSION AND OUTLOOK

The separation of encoding/recording and playback in Ambisonics allows for a flexible production: Playback via loudspeakers is suitable for large audiences, whereas headphone playback is practical for mobile applications. The number of storage/transmission channels determines the spatial resolution and is not depending on the number of sources. These sources can either be panned virtual sources, e.g. spot microphone signals, or recordings from Ambisonic main microphone arrays. Spatial effects can be applied in the Ambisonics domain, e.g. widening or loudness adjustment of single sources within main microphone recordings. All necessary software components are freely available as VST-plugins[5].

The implementation of a dynamic compression that does not influence the spatial image is still an open question. The omni-directional component of Ambisonics includes all signals independent of their direction and thus can be used as a side-chain to control the compression of all channels equally.

The impact of format conversation, and down-mixing strategies has to be studied in perceptual test. Furthermore, automatic sound scene optimization (focussing on timbre and sound source directions) in regard to adapted natural recordings should be treated.

[5] http://www.matthiaskronlachner.com/?p=2015

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