Higher order Ambisonics - a future-proof 3D audio technique

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Abstract

Current approaches to three-dimensional sound reproduction build upon the discrete-speaker paradigm of ITU-R BS 775-1 5.1 and related formats, relying on additional speakers and transmission channels for each new directional cue.

Consequently, they inherit the many disadvantages of discrete formats: non-homogeneous reproduction, strong dependency on the consumer's speaker setup, and lack of up and downwards compatibility.

In contrast, Ambisonics provides a rigorous and systematic approach to 3D audio which can be scaled to any desired level of precision using higher-order spherical sampling. Its signal representation, known as B-format, is independent of the speaker setup used for listening and can be decoded to arbitrary layouts at the consumer's end, with graceful degradation all the way down to mono.

1. Introduction

Most sound engineers' prior exposure to Ambisonics is limited to first-order recordings made with the Soundfield microphone. First-order Ambisonics has undeniable limitations in terms of imaging stability and size of listening area, sometimes further deteriorated by improper decoding for speaker layouts ill-suited to the purpose (such as ITU 5.1). Consequently, it suffers from a long history of unimpressive demonstrations [1], and the trade as a whole has pretty much shelved Ambisonics as a historical dead-end.

This is unfortunate, since it can be superiour to discrete multichannel approaches in many ways and offers an alternative to the never-ending game of adding yet another round of speakers and transmission channels (thereby obsoleting most if not all of the existing workflows and investments).

Recent research in Higher-Order Ambisonics (HOA) [2, 3] based on ideas sketched by Gerzon in the 1970s [4] has shown ways to overcome the known limitations and harness the advantages of systematic spatial sampling. It may be time to re-evaluate the Ambisonic paradigm for serious use!

Since Ambisonics can be an irritating concept, the paper begins with a gentle introduction to the underlying theory, followed by the description of a new hybrid HOA recording technique. The paper concludes with a report of a contemporary HOA production to demonstrate its practical feasibility.

2. A brief run-through of Ambisonics fundamentals

2.1. The spherical sampling series

The theoretical foundation of Ambisonics is the ***Kirchhoff-Helmholtz integral***. It states that if the sound pressure and velocity on the surface of an arbitrary volume that is itself free of sound sources is known, we have complete knowledge of (and thus the ability to reproduce perfectly) the sound field in the interior.

This is a very powerful theorem, because it reduces the problem of a three-dimensional sound *field* to a two-dimensional *surface area*.

A naive implementation of the Kirchhoff-Helmholtz integral might use a shell of infinitely many microphones that surround the listener completely. Their recorded signals would then be reproduced by an infinite number of speakers in the same places, resulting in a perfect reconstruction of the sound field (in the physical sense, not just perceptually).

Clearly, the two infinities in the previous paragraph are a serious showstopper! So we need to find a way to approximate this ideal, preferrably one that converges quickly to a usable result.

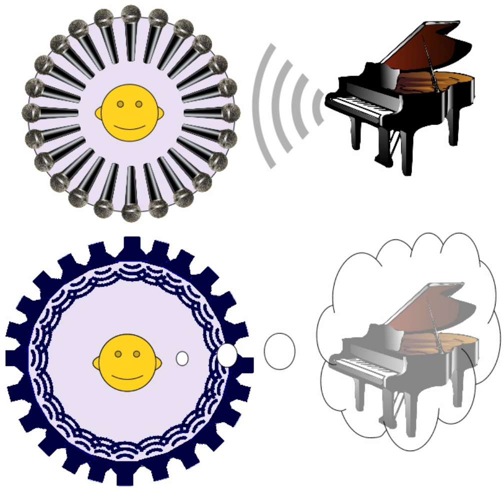

Illustration 1: A naive implementation of the Kirchhoff-Helmholtz integral.

Enter the paradigm of ***spherical sampling***: we want to find a series of sampling functions that „measure“ the sound on the sphere surface around the listener uniformly, gradually adding more detail until a satisfactory result is achieved.

Microphones are, in effect, spherical sampling devices. A simple omni-directional pressure transducer at the listening spot will already cover the sphere surface completely. And even though it does not yield any direcctional information at all, it represents the root of the spatial sampling series, also known as the ***zero'th order***. In the Ambisonic signal representation (commonly referred to as ***B-format***), the omnidirectional component is denoted by ***W***. It is what remains when you have to play an Ambisonic recording in mono.

The next set of signals comprises three figure-of-eight components – as would be produced by pure velocity transducers. When they are aligned in the same spot so that one points to the front, the next to the left, and the third upwards, we have again covered the sphere completely and uniformly: regardless of where a source moves to, the summed power of these three capsules will remain constant. You can easily visualize this for the 2D case by drawing two figure-of-eight polar patterns superimposed at an angle of 90° (the famous Blumlein pair): when you add the combined sensitivities of both capsules for each angle geometrically, you will get a constant value of 1 everywhere - the combined sensitivity pattern will be a circle.

Now we have directional information, and now it also becomes evident why we must not skip the zero'th order signal: it serves as a polarity reference that enables us to determine whether a source is coming from the frontal lobe of the figure-of-eight or from its polarity-inverted rear lobe.

In the B-format, the figure-of-eight components are called ***X***, ***Y***, and ***Z***. They constitute the ***first-order*** set of the series. Since they are orthogonal to one another, they must be ***linearly independent***, which is a highly desirable feature that simplifies subsequent processing a lot and ensures we do not transmit or store redundant information. In fact, we want to make sure that all sampling functions of any order that we use are linearly independent.

If we truncate our sampling series here, we have ***classical Ambisonics***: a set of only four components that already provides full 3D sound reproduction.

2.2. Panning

To position a source within a B-format representation, we imagine a vector connecting the origin of our coordinate system (which represents the listening spot) to the source. Each component is then the distance from the origin to the intersecton point of our source vector and the corresponding polar pattern. Note that the sign of the first-order components will be governed by the polarity of the lobe (see illustration 2).

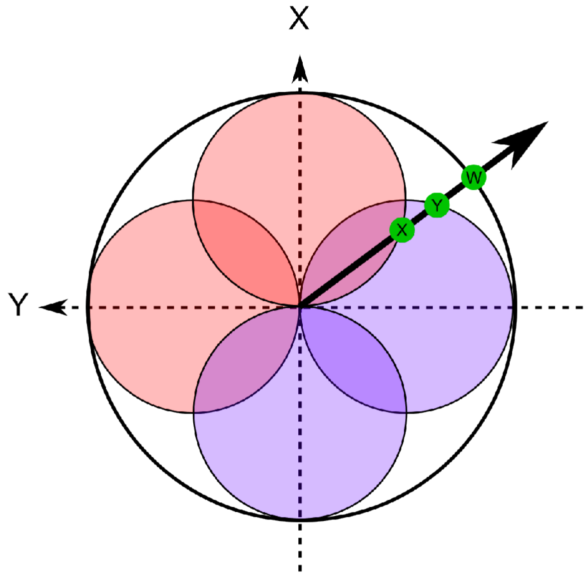

Illustration 2: Ambisonic panning. Draw a vector from the origin to the source. The component values are the distances to the intersections of the vector and the component polar patterns (green dots W, X, and Y), taking into account the sign of the corresponding lobe. In this example, Y is negative.

It is convenient to represent polar patterns in polar notation, i.e. to express radius as a function of angle. For the first-order horizontal components, we get

$$\mathbf{X} = \cos \varphi \cdot \mathbf{s}$$

$$\mathbf{Y} = \sin \varphi \cdot \mathbf{s}$$

for a source signal s coming from an angle φ (disregarding elevation for simplicity).

2.3. Decoding

For playback, we have to decode the B-format into suitable speaker feeds, just as M/S stereo (a very similar concept) has to be transformed into L and R signals prior to listening. While this might seem tedious at first sight, it is in fact a great advantage, because it decouples the production and transmission format from the consumer's speaker layout. Since most customers at present do not have the means to decode native Ambisonics at home, producers can pre-decode the material to various speaker arrangements such as rectangular quadraphonic or 5.1, and distribute those.

Different decoding techniques exist, to optimise for sweet spot performance or extended listening area [5]. Decoders should also implement shelf filtering, to maximise the velocity gradient in the LF range where the ear is sensitive to phase information, and the energy gradient in the HF range, where interaural level difference (ILD) is the dominating cue [6,7]. Higher-order decoders must provide near-field compensation to ensure correct LF reproduction [8].

2.4. Ambisonic microphones

To create B-format recordings, one could use a combination of one omnidirectional and three figure-of-eight microphones. In practice, it is impossible to arrange them to be truly coincident, and time errors are inevitable. Worse, these errors will be different for each angle of incidence, which violates the design goal to create a truly isotropic system. If only horizontal surround is desired, two figures-of-eight and an omnidirectional capsule can be stacked so as to be perfectly coincident for sounds emanating from the horizontal plane, but a certain amount of HF coloration for the reverberant field is unavoidable. Such "native" B-format arrays have been used to good effect by British label Nimbus Records.

To achieve truly isotropic 3D coverage, four cardioid capsules can be arranged in tetrahedral fashion, facing outward. This is equivalent to placing them in every other corner of a cube. Let us call them the left-front-up (LFU), right-front-down (RFD), left-back-down (LBD), and right-back-up (RBU) capsules. We can now obtain the B-format components as follows:

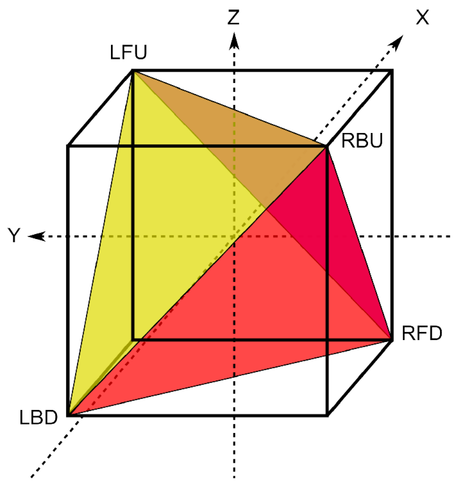

Illustration 3: Capsule arrangement in a first-order Ambisonic microphone.

For W, we simply add all four capsules:

W = LFU + RFD + LBD + RBU

For X (front minus back), we add the frontal capsules and subtract the rear ones:

X = LFU + RFD – LBD – RBU

For Y (left minus right), we add the capsules on the left and subtract the ones on the right:

Y = LFU – RFD + LBD – RBU

For Z (up minus down), we add the upper capsules and subtract the lower ones:

Z = LFU – RFD – LBD + RBU

This idealized matrix assumes that all capsules are located in the same point in space. While the displacement error is a lot smaller and more regular than for a native array, some corrective EQ is still required to repair the frequency and phase responses. The residual error is small and distributed evenly around the sphere, which meets our design goal.

Farina has described a simple approach to obtain the required filters [9]; a more comprehensive description can be found in Gerzon's original proposal [10].

3. Problems with first-order Ambisonics

The figure-of-eight pattern has very limited directivity. A source ten degrees off-axis will only be about a tenth of a dB lower than an on-axis source, and it takes 45 degrees offset for the level to fall to -3dB. In practice, this translates to blurry sources and a very small ***sweet spot***, outside of which localisation will be unstable. In fact, since the sweet spot size is a function of the wavelength, it will be considerably smaller than a human head for the most part of the spectrum. Hence, first-order material is best listened to alone, at the precise center of the speaker array – for more listeners, the experience will generally be unsatisfactory.

That classical Ambisonics works at all is for the most part due to the post-processing in our brain: we focus on correct localisation cues, and discard ambigious ones.

The author has therefore conjectured that Ambisonic listening is an acquired skill, and that it takes training or at least habituation to fully benefit from the spatial information contained in first-order Ambisonic recordings [11]. This might provide another explanation as to why casual listeners have frequently remained unimpressed by demonstrations, to the astonishment of the presenting Ambisonics enthusiast.

Recording aesthetics, while not strictly related to the underlying reproduction paradigm, pose another problem: native Ambisonic recordings are usually (but not necessarily) created with a single soundfield-type microphone, without the use of spots. Consequently, they will sound rather distant and to some ears even „un-emotional“ compared to contemporary, very defined and present classical productions. Hence, their appeal is limited to those listeners who would also prefer the spatially correct yet difficult-to-approach renderings created by a Blumlein pair, such as advocated by Lipshitz [19].

4. Extension to higher orders

To overcome the stability problems of classical Ambisonics, we can continue our spherical sampling series, for a more precise reproduction of the sound field. We will choose *spherical harmonics* as sampling functions, because they exhibit the desired characteristics: linear independence and uniform coverage of the sphere.

4.1. Spherical harmonics

Just as a string can only vibrate at discrete partials, which form the natural harmonics series, a sphere can only vibrate in particular modes (think of a very large air balloon). The fundamental mode (or zero'th order) is rapid shrinking and extending – what a balloon would do if you suddenly changed its air pressure.

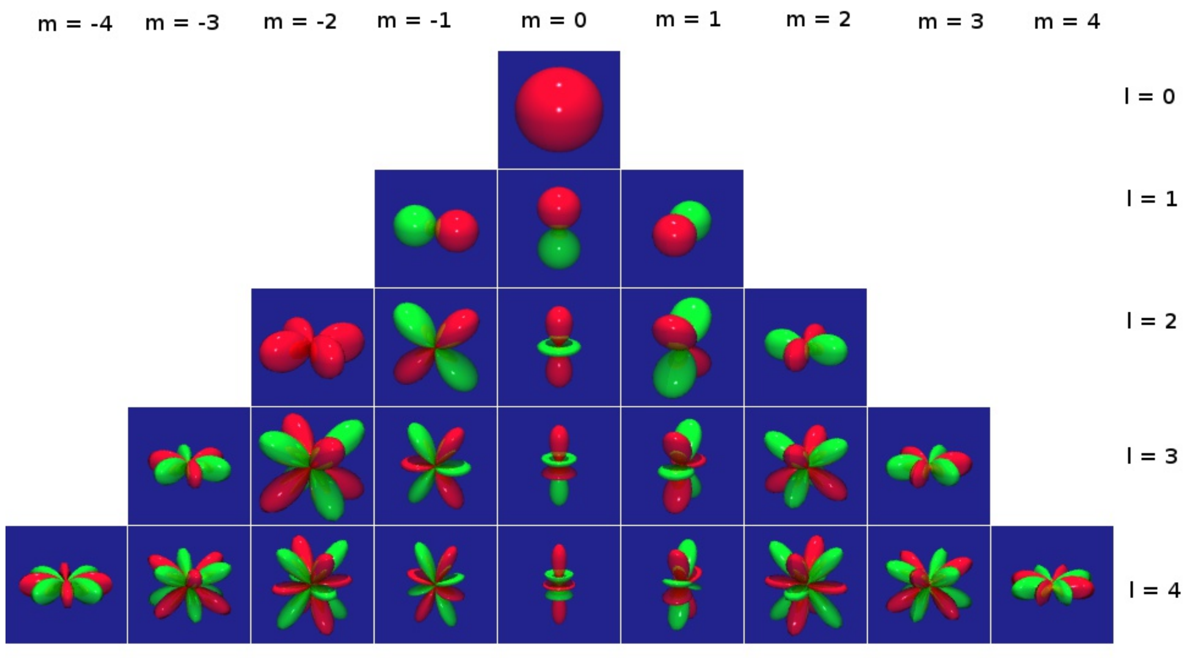

Illustration 4: Spherical harmonics of orders 0 through 4.

Erratum: the first harmonic of the second-order set should have its front-left and rear-right lobes colored green. [Visualization by David Yerga, http://www.flickr.com/photos/yerga/3351536253/sizes/o/in/photostream/ , CC-BY-NC]

You can consider the polar pattern of the corresponding microphone to be a measure of the vibration amplitude for each part of the sphere surface.

The first harmonic oscillation is along the X, Y, or Z axes. If you were to clamp the balloon into a ring right at its equator, this it what it would do when you then slapped the top, very much like the octave flageolet on a violin is produced by obstructing the string exactly at half its length. Again, the figure-of-eight is a measure of the vibration amplitude: largest at the front and back, and zero on the sides, where the clamp is.

Now for the next, new harmonic: we clamp two rings around the balloon, for example at the zero and 90° meridians. When we excite it, the oscillation will take the shape of a clover: two opposite sections will shrink while the other two will expand, and then vice-versa. So our second-order measurement functions have the shapes of a clover - or more precisely, a *tesseral quadrupole*, whose lobes are more directional than the first-order ones. The front-back and left-right ambiguity of the second-order harmonics is resolved with the help of the corresponding first-order harmonic.

4.2. Higher-order microphones and panning

While second-order microphones have been described by Blumlein as early as 1936 [12] and further investigated by Olson in the 1940s [13], native higher order microphones are increasingly difficult to implement. Their low-frequency responses roll off with 6dB per octave per order. Moreover, since they consist of several capsules to allow for higher-order differentiation, they accumulate noise and require precise matching, which is difficult and error-prone [14].

Spherical microphone arrays with a large number of pressure sensors allow for the derivation of higher-order directional patterns, and there exists one commercially available 32-capsule microphone [15]. But all current spherical arrays battle the same fundamental physical problems (matching problems and noise amplification due to the large LF gains required) [16], and none of the existing implementations meet the quality requirements of serious music recordings.

Panning functions however need not care about such engineering constraints, which makes higher orders very practical for the artificial panning of single sources, such as spot mics. So we can happily move on to sextupoles in third order, and contraptions with ever more lobes as the order increases, gaining more directivity and enlarging the sweet spot in the process (see illustration 4). Furse has an overview of the panning equations up to third order [17].

5. A hybrid recording technique for HOA

Due to the timbral deficiencies and noise problems of existing HOA microphones and the relative ease of HOA panning, the author has explored a hybrid approach to record acoustic events: a traditional first-order Soundfield-type array is used as a main microphone, and spot microphones panned in third order provide focus and stability.

The relationship between main and spot microphones differs substantially from stereo recording techniques. Our main array's most important task is to capture the early reflections and late reverb at the listening position. Most of the direct sound is to be conveyed by the spots. We use the Soundfield where it shines: in its unique ability to capture a correct and coherent image of the room acoustics in all directions. Any localisation ambiguities of the direct sound are resolved by adding clear and robust cues from the panned spots. This approach provides us with a very convincing rendition of the room *and* precisely locatable sources over a large listening area.

With traditional stereo techniques, spots are used sparingly and only to maintain balance, because the clarity of the spatial image usually suffers as more spots are introduced. For hybrid HOA recordings, we are forced to cover each instrument or instrument group with its own microphone. Careful time-alignment of the spots is paramount if the blending is to be successful, and the positioning of the spots must be matched to the direction suggested by the main microphone, so as to avoid ambiguous localisation cues. Great care must be taken to avoid echoes when a spot picks up an instrument from the other end of the stage. In tutti passages, these echoes are masked sufficiently, but a single-instrument *sforzato* in an otherwise *pp* passage can be difficult to deal with. Consequently, directional microphones are mandatory for the spots, and mixing automation must be used to deal with any remaining echo problems.

Depending on the acoustics, it can be advisable to take room impulse responses for every single spot microphone position, so that the spots can later be convolved with some “natural” reverb individually, if they stand out too much.

The placement of the main microphone can be challenging: raising the array to ensure uniform coverage is not an option, since it would induce erroneous z-axis information that literally places the musicians six feet under. Instead, we rely on the spots to balance the instruments and put the main array on the best seat in the audience, at ear height. Unfortunately, this position makes it very susceptible to audience noise in live recording situations. While it is possible to raise the array and then rotate the resulting B-format around the y-axis until the musicians are back to zero elevation, applause would seem to come from straight down, which can only be corrected if music and crowd sounds never overlap.

While not really complicated as such, HOA sessions use many more microphones and tracks than stereo recordings, and require greater care in post-production. For music with a frontal-only sound stage in the western tradition, the extra effort might not be justified, as the surround effect is limited to the reverberant field, and immersive ambience can be faked rather convincingly by more economical means.

HOA recordings will only realise their full potential as the sound sources leave the frontal stage, as in many contemporary compositions, a select few historical works (Gabrieli's polychoral compositions come to mind), and in film sound. But as a future-proof production format that can easily scale up to produce 7.1 or Auro-3D pre-rendered content, it might be worth considering even for front- (or screen-)centric material.

6. Ambisonics compared to discrete-speaker approaches

Discrete-speaker techniques produce single stimuli, which can be either physical or phantom sources. The latter are created by pair-wise or, in the case of VBAP, triangular loudspeaker panning [18]. This makes them very resilient to speaker placement errors, which will result in in nothing worse than a corresponding displacement of the reproduced sound. On the downside, the timbral quality and perceived sharpness of sources will vary, depending on whether they are reproduced by a physical source, a phantom source on the line between two speakers, or a phantom source in a triangle of three speakers.

In stereo, we can also exploit timing differences of correlated signals between the two speakers. They translate to interaural time difference (ITD) cues, although severely blurred by crosstalk, since the left speaker also reaches the right ear and vice-versa. For lateral signals, time delays between front and rear speakers do not generally result in meaningful ITD cues.

This implies that *artificial* ITD cues fed directly to the ears via a speaker pair are only usable for left/right localisation, and only as long as we can assume the listener's head orientation to be frontal.

It also means that surround microphone arrays with non-coincident front and rear transducers such as OCT or INA do not produce „front-to-back A/B stereo“. Surround arrays can only rely on intensity difference for lateral localisation. Front-to-back microphone distance is only used to control the amount of correlation and the gradient of direct to reverberant sound.

Generally, we find that most tricks we can play with stereo to provide a rich set of left/right localisation stimuli cannot easily be extended to surround, and that the only technique that will always work regardless of head orientation is coincident stereo. This need not be an undue limitation: in a 1986 JAES article, Lipshitz argues its superiority in terms of spatial fidelity and considers the airier quality of spaced techniques to be an artefact rather than a reproduction of the original space [19].

Ambisonics aspires to reproduce the physical sound field, rather than provide disjoint stimuli. In this respect it is quite similar to wave field synthesis. In practice, it means that a number of speakers will contribute to the reconstruction of any single virtual source. The advantage is that source sharpness and timbre are perfectly constant, whether on speakers or between. On the other hand, large speaker placement errors can severely compromise the field reconstruction.

A common misconception states that, since all the B-format components are coincident, Ambisonic systems cannot convey time information, and hence lack ITD cues altogether. This is incorrect. In real sound fields, it is the distance between the listener's ears that causes ITD. The same thing happens when the listener's head is inserted into the (hopefully correctly) reconstructed sound field. Consequently, the resulting *natural* ITD cues will work regardless of head orientation: if the listener is facing sideways, s/he will perceive correct ITD cues for the lateral quadrants of the sound stage. Incidentally, Lipshitz remarks that even in intensity-only stereo systems, level differences are ultimately translated into ITD cues [19].

Discrete-speaker systems with economically reasonable (i.e. low) channels counts show deficiencies in localisation precision and/or stability. ITU 5.1 setups sacrifice lateral and rear imaging for optimal frontal reproduction. Efforts to fill the deficient areas with additional discrete signals (6.1, 7.1) have not been widely adopted for cost and bandwidth reasons.

Zieglmeier and Theile have shown that auxiliary side-center speakers fed with synthesized signals resolve the lateral localisation problems of 5.1 systems without necessitating extra transmission channels [20]. The question is how such signals are to be obtained, as it requires knowledge of the temporal relationship between frontal and surround channels. One would expect significantly different results depending on the mixing technique or type of microphone array used.

Ambisonic systems also display improved stability when using slightly more than the theoretical minimum number of speakers (for example, a hexagon out-performs square arrangements for first-order playback [7]). With Ambisonic source material, it is trivial to derive as many speaker signals as desired, without surprises.

The results of Zieglmeier and Theile underline the importance of a systematic and predictable approach to localisation and show that the number of transmission channels can (and should) be decoupled from the number of speakers used for reproduction for best results.

Illustration 5: The baroque gallery at Herrenhäuser Gärten, Hannover. The primary main microphone is located between the grand piano and the percussion set in the foreground. The secondary one is at the other end of the hall in roughly the same position. On the floor in the middle, the wind-up music boxes can be seen. The balconies at both ends of the hall and the lighting towers (whose cover flaps you can see protruding from the left wall) are used for solo and duet pieces. Photo by Florian Faber, used with permission.

7. Project Report

In June 2010, the Kunstfestspiele Herrenhausen in Hannover, Germany, commissioned British composer Rebecca Saunders to produce a rendition of her spatial chamber work "Chroma" at the gallery of the Herrenhäuser Gärten, to be performed by musikFabrik Köln.

Chroma consists of a number of chamber music groups, soloists and "objets trouvées" such as wind-up music boxes distributed widely across the venue, and the spectators are invited to move freely between the musicians as the music unfolds. Some solo instruments and duets were positioned on balconies or lighting towers, and during the final part of the composition, the performers were to move outdoors one by one.

Naturally, a composition that employs full surround sound including height and even incorporates two totally distinct natural acoustic settings is a perfect proving ground for any 3D audio technique. The author had gained some confidence in his toolchain during two earlier HOA projects [21, 22], and with the help of a number of supporters (see section 8) decided to take up the challenge.

To capture at least a small part of the „walk-around" feel of the composition, we deployed two tetrahedral main microphones, one of which followed the musicians outside near the end of the piece. So we were able to select between two distinct auditory perspectives for the mix and captured the transition of reverberant to free-field sound. For higher-order

„sharpening“, we used 40 spot microphones dispersed over considerable distances. Thanks to remote-controlled MADI microphone preamps, the entire venue could be covered without hum or noise problems, despite cable runs of several hundreds of meters. The MADI data was split in the recording room and written to two independent Linux audio workstations, each using mirrored disk arrays.

The spot tracks were equalised and cleaned up on a conventional stereo monitoring system, and a rough mix with preliminary automation was created on a second-order hexagonal Ambisonic rig in the author's home studio (horizontal only). Thus prepared, the session was taken to IEM Graz for a full 3D third-order post-production on the CUBE system, a hemispherical rig of 24 Tannoy 1200 monitors [23]. The software toolchain is described in [24].

The resulting mix has since been played back at MUMUTH Graz during the dafx10 conference [25] over another hemispherical rig of 29 Kling&Freitag CA 1001 complemented with two subwoofer stacks, and on the SPIRAL system at the University of Huddersfield, which features three stacked octagons of Genelec 8240As plus a zenith speaker and four 7270A subwoofers [26]. Neither of those renderings required any changes to the mix – as long as a suitable decoder is available, the 16-channel third order master can be played as-is.

The author is currently preparing a 5.0 rendering for home listening. While the basic fold-down could be (and in fact is) created automatically from the original master, the mix has retained the natural dynamic range of well over 50dB, since it was intended for listening in acoustically controlled environments. To become suitable for domestic enjoyment, some manual compression will be required.

Readers wishing to evaluate the recording method on a sufficiently equipped system are welcome to contact the author to arrange for a demonstration. The master can be made available for research purposes upon request.

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