VARIABLE-POLAR PATTERN SYNTHESIS FOR USE IN TWO DIMENSIONAL SOUND REPRODUCTION

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ABSTRACT

In this paper the authors present an approach for two-dimensional sound reproduction using a circular layout of speakers where the gains are obtained from a variable polar pattern. The method presented here has the ability to be variable-order whilst keeping the same key features of a base polar pattern. Comparisons are drawn between the new approach and a previous approach by the authors using variable-order, variable-decoder Ambisonics. The new method is found to not be as directional as the Ambisonics approach, yet it maintains the base polar pattern unlike with Ambisonics. Whilst both approaches have two variable parameters the new approach's parameters are independent and are therefore intuitive to an end user using such a tool as a spatialisation effect as well as technique.

1. INTRODUCTION

Various methods exist for both 2D and 3D spatial audio reproduction. These include, but not are limited to Ambisonics [1, 2], Wave Field Synthesis [3, 4, 5], Vector Based Amplitude Panning (VBAP) [6, 7] and three dimenional highly directive virtual microphone approach (3DVMS) [8, 9, 10, 11]. Systems have been built using mixtures of these techniques to overcome limitations of single technique systems. These are usually based on the mixture of Ambisonics and VBAP creating systems such as Directional Audio Coding (DirAC) [12, 13] and Vambu Sound [14, 15]. Most spatial audio work undertaken relies on examining the theory and listening tests undertaken in acoustically controlled conditions. However, Bates et al. have shown that monophonic sources can be localised well in reverberant environments. They give the best outcome expected by spatial audio techniques for a distributed audience [16], as would be experienced in a real world performance using a spatial audio system. To overcome the errors expected from non-anechoic environments and off sweet spot positioning of an audience, as well as to distribute spatial material so that it does not require a large speaker setup, binaural techniques can be used to create headphone representations of the material [17, 18].

The authors have previously presented an approach for two-dimensional sound reproduction based on Ambisonics theory [19]. Since the final output of an Ambisonics reproduction to the speakers is the same as sampling a polar pattern [1] exhibited by the audio source material, the authors propose to create a method whereby the intermediary B-Format and decoding is omitted. This method gave two new parameters for user control; variable-order and variable-decoder. Whilst the method was successful in producing a system where the directionality of a sound source and the polar pattern type could to some extent be changed, the two controls overlapped in their result considerably. This leads to confusion for end users, and in reality the two controls were not fully utilised when creating a demonstration film soundtrack of the tool. In this previous system the polar pattern types are those created from combinations of zeroth, first, second and third order circular harmonics by the pseudo-inverse matrix.

In response to the feedback on the previous tool the authors are proposing a new approach that also has two unique features to 2D sound reproduction; variable-order and variable-pattern. It will be shown that these controls do not overlap like those of the previous method, in section 2. The newly presented method, described in section 3, and variable-order, variable-decoder Ambisonics will be compared in section 4. Our implemented demonstration application is described in section 5. Finally, conclusions on this approach to two-dimensional sound reproduction will be drawn in section 6.

2. VARIABLE-ORDER, VARIABLE-DECODER AMBISONICS

In [19], the theory of variable-order, variable-decoder Ambisonics was first presented where the work was limited to the case of an octagonal arrangement of 8 speakers. A variable-order is calculated by summing a ratio of the final speaker signals for the integer orders above and below the variable-order, $\lceil M \rceil$ and $\lfloor M \rfloor$, where M is the system order. The ratio of the two orders used is $M - \lfloor M \rfloor$ for $\lceil M \rceil$ and $1 - (M - \lfloor M \rfloor)$ for $\lfloor M \rfloor$. The variable-decoder is calculated in a similar way as an interpolation between two adjacent decoder types; rV and max rE or max rE and in-phase. This creates a two way interpolation methodology. In the authors implementation in [19] for both the lower and higher integer order the two decoder types are calculated and interpolated between. This leaves speaker gains for the variable-decoder for both the lower and higher integer order which are then interpolated between to produce the final speaker gains. Throughout the interpolations used the sum of all the speaker gains remains one at all points keeping power constant. Figure 3 shows a block diagram of the interpolation process involved to create the Variable-Order, Variable-Decoder Ambisonics speaker gains.

The speaker feeds for an Ambisonics 2D reproduction for an

* This research was supported by the Engineering and Physical Sciences Research Council [grant number EP/P503426/1].

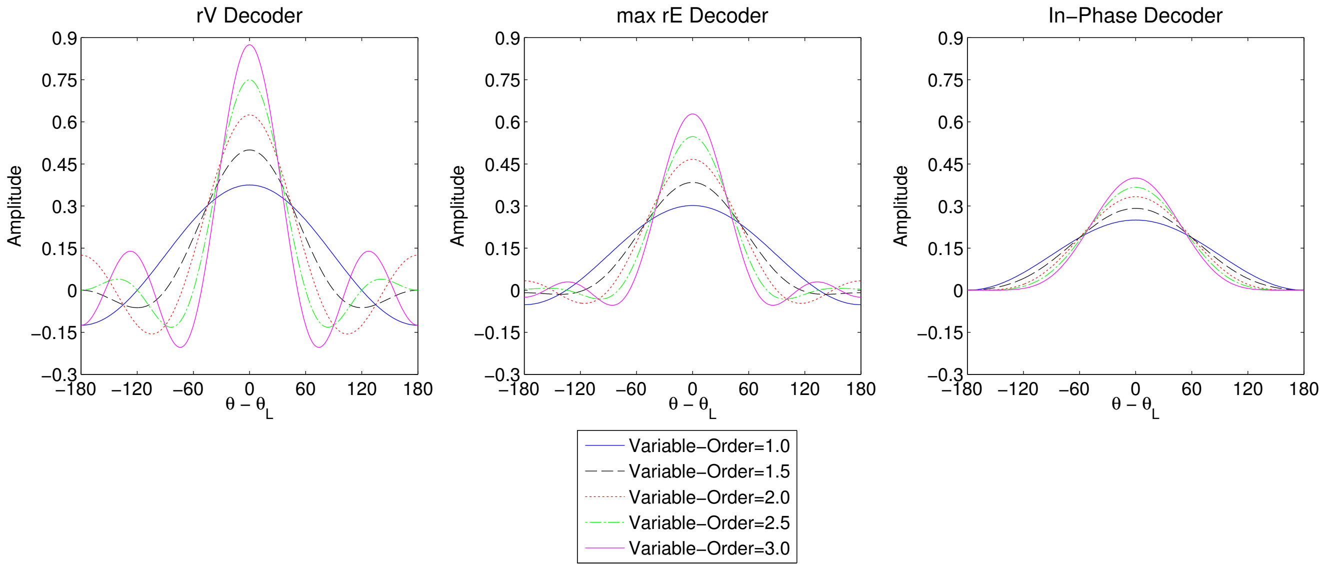

Figure 1: *Variable-Order Ambisonics shown for rV decoder (left), max rE decoder (centre) and In-Phase decoder (right) as described in Sec. 2.*

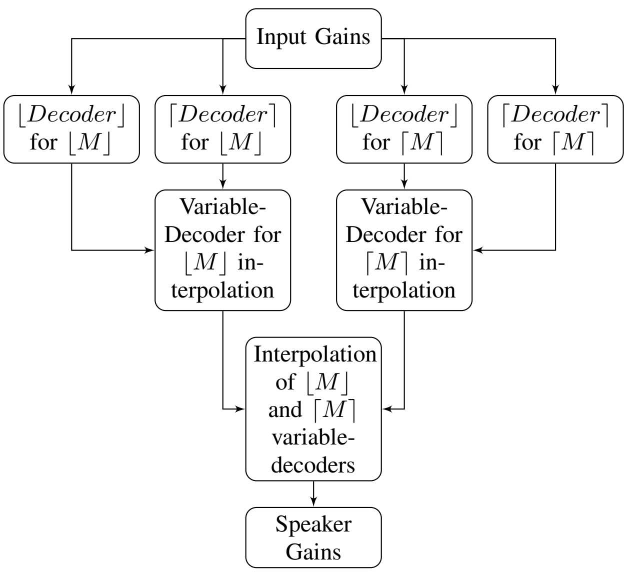

Figure 3: Block diagram of the calculation of Variable-Order, Variable-Decoder Ambisonics speaker gains.

integer order [1, 2, 20] for N speakers are:

$$[L_1 \dots L_N] = [Y_{(0,0)}(\theta) \dots Y_{(M,n)}(\theta)] \begin{bmatrix} Y_{(0,0)}(\theta_1) & \dots & Y_{(M,n)}(\theta_1) \\ \vdots & \ddots & \vdots \\ Y_{(0,0)}(\theta_N) & \dots & Y_{(M,n)}(\theta_N) \end{bmatrix}^{\dagger} \tag{1}$$

where θ is the azimuth angle of the sound source and $\theta_1 \dots \theta_N$ is the angle of each corresponding numbered speaker. The circular harmonic of an order m for $m = 0 \dots M$ and a degree n for the terms $\mp m$ is given as:

$$Y_{(m,n)} = \begin{cases} \cos(n\theta) & n \geq 0 \\ \sin(n\theta) & n < 0 \end{cases} \tag{2}$$

The minimum speakers for an order M is $N \geq 2M + 2$ [21, 22]. This is just one method that can be used to decode an Ambisonics signal and is referred to as mode-matching.

Figure 1 shows various orders for three decoder types rV, max rE and in-phase (also known as controlled opposites). The definition and affect of rV and rE is discussed in detail in section 4. Higher orders give a higher directionality, increasing the gain in the desired direction and reducing the non-primary lobes in gain. However, it is difficult to see how the polar patterns merge for the rV and max rE decoders due to the increase in rear lobes as the order increases. In fact, for the non-integer orders the negative rear lobes have a lower amplitude level than even $\lfloor M \rfloor$ and so are not a linear transition between the integer orders. This is non-problematic for the in-phase decoder, as it has no rear lobes. Figure 2 shows the variable-decoder implementation for orders 2, 2.5 and 3.0. The variable-decoder has a somewhat similar outcome as the variable-order, whilst having a different technical description. The variable-order increases the gain and narrows the main lobe whilst consequently increasing the count of the rear lobes and reducing their gains. The variable-decoder reduces the gain of the rear lobes whilst having the negative effect of reducing the gain

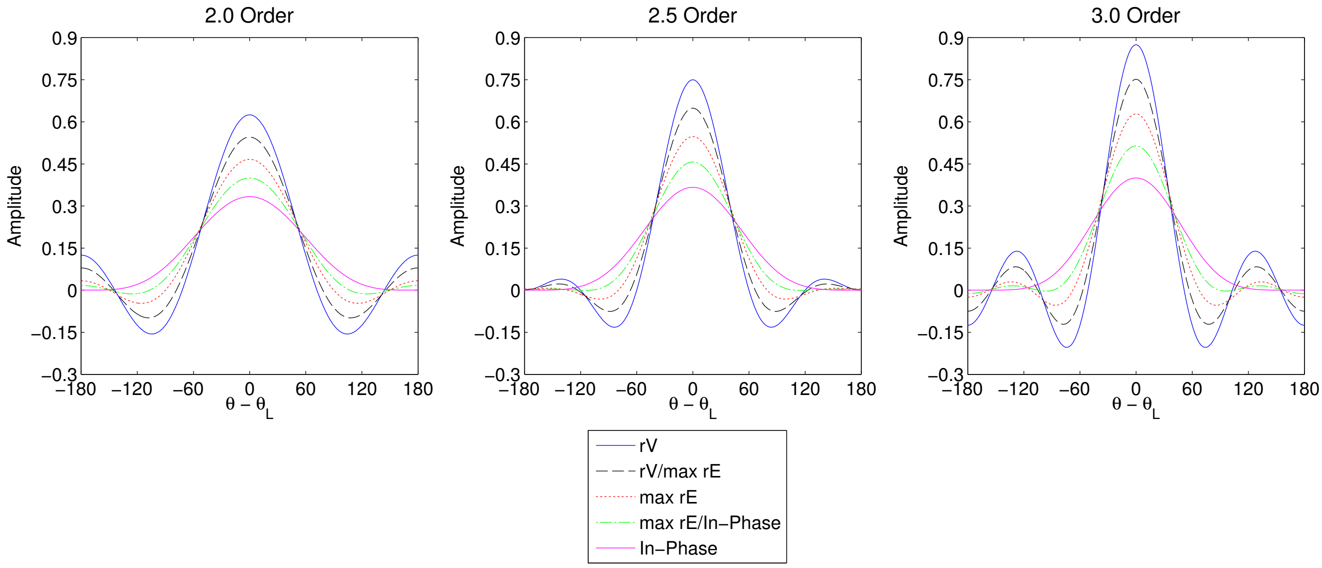

Figure 2: *Variable-Decoder Ambisonics shown for 2.0 order (left), 2.5 order (centre) and 3.0 order (right) as described in Sec. 2.*

and width of the main lobe. A comparison can be drawn to illustrate this between the right hand graph of figure 2 for 3.0 order variable-decoder and the left hand graph of figure 1 showing the variable-order using an rV decoder demonstrating these subtle differences.

The differences between variable-order and variable-decoder for the end user can appear to some extent, to do the same thing and can in fact counteract one another, which makes their use unintuitive. It is also counter-intuitive that the shape of the polar pattern changes with order, in regard to rear lobes.

3. VARIABLE SOURCE RADIATION PATTERN

[23] gives two formulations for calculating higher order microphone polar patterns. The first is given for calculating cardioid patterns in the form of $G = (0.5 + 0.5\cos\theta)cos^{(M-1)}\theta$ for the M^{th} order, which is expanded for any base polar pattern in equation 3 below. We define a base polar pattern as that created as a mixture of zeroth (A) and first (B) order components to calculate a gain G at horizontal angular position θ.

$$G = (A + B\cos(\theta))\cos^{(M-1)}(\theta) \tag{3}$$

where A is the zeroth order (omnidirectional) component and B is the first order (figure of eight) component. $A + B = 1$ must remain constant for all positions of θ.

The second equation for a higher order pattern is given as the product of two or more first order microphone patterns:

$$G = (A_1 + B_1\cos(\theta))(A_2 + B_2\cos(\theta))\cdots(A_M + B_M\cos(\theta)) \tag{4}$$

where $A_{1...M}$ and $B_{1...M}$ are the zeroth and first order terms for each order order creating the higher order polar pattern up to the system order M. To keep controls to a minimum we can limit the possible polar patterns so that $[A_1, B_1] = [A_2, B_2] = [A_M, B_M]$. By using this identity we can use a variable order for M below:

$$G = \begin{cases} (A + B\cos(\theta)^M & M \text{ is odd} \\ -(|A + B\cos(\theta)|^M) & M \text{ is even} \end{cases} \tag{5}$$

Figure 4 shows the differences for calculating omnidirectional, cardioid and figure-of-eight polar patterns using equation 3 for method A and equation 5 for method B. It can be seen that for method A in the top row of the figure that for the omnidirectional above order 1, the pattern changes to a figure-of-eight pattern of order $M - 1$. When looking at higher order cardioid for method A, we see that rear lobes are formed on the cardioid pattern. Finally the figure of eight pattern for method A behaves as we want, as the order increases the angular distance between the -3dB points also decreases, giving a tighter polar pattern around the maxima and minima points. In the second row of the figure we see the results of method B, equation 5. Firstly the omnidirectional pattern remains omnidirectional at all orders. The cardioid pattern for B does not develop rear lobes, but becomes a beam like pattern. Finally the figure-of-eight pattern for method B behaves like that of method A, as we expect; a tighter figure of eight with greater side rejection. From these findings the authors choose to continue to only use method B as it produces the most useful higher order polar patterns.

When the user is presented with a choice of polar pattern there are several reasons to chose the various types. A cardioid, like the in-phase Ambisonics decoding, can be chosen because it has no rear lobes and therefore there is less chance of wrongful perception of the source directivity away from the centre of the speaker array. Hyper-cardioids can be chosen for the smallest angular separation between the -3dB points of the polar pattern. The hyper-cardioid however has the largest rear lobe and so has the possibility of non-central listeners localising the source in the opposite direction as was intended. Typically for Ambisonics systems the hyper-cardioid has more accurately reproduced low frequencies in terms of correct direction perception. An omni-directional can be chosen so that the sound source has no directivity, this is the same as making the source of zeroth order. The sub-cardioid can be used to have a sound enveloping around a user with some directional aspect to the source remaining. Depending on listener distance from the central position there could be a chance of the direction of the sound source being incorrectly perceived. Figure-of-eight polar

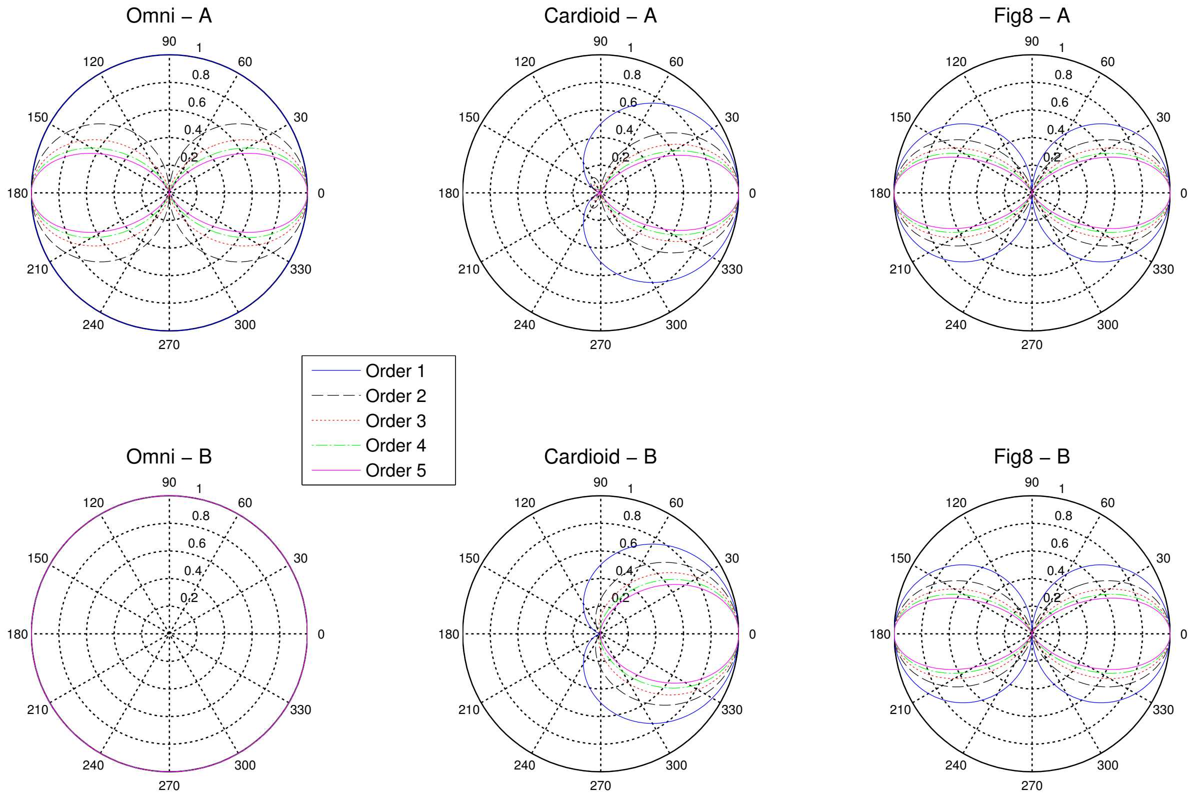

Figure 4: *Comparison of pattern methods A 3 and B 5 for omnidirectional (left), cardioid (centre) and figure of eight (right) as discussed in Sec. 3.*

patterns are not possible in the system which will be discussed later in the paper. An objective measure for the perceived source width of a sound source can be found in [24, 25, 26] and psychoacoustic based discussion on source localisation and perception in [27].

The gain applied to the L^{th} speaker in a circular array is given as:

$$G_L = \begin{cases} (A + B\cos(\theta - \theta_L))^M & M \text{ is odd} \\ -(|A + B\cos(\theta - \theta_L)|^M) & M \text{ is even} \end{cases} \tag{6}$$

To maintain a constant level whilst varying the order and/or polar pattern, like in the Ambisonics method, a factor C is needed to scale the speaker gains:

$$C = \frac{1}{\sum_{L=1}^{N} G_L} \tag{7}$$

where the maximum order is based on N number of speakers being used, $M = (N-2)/2$. Note that this will give a variable order and using the $\lfloor\rfloor$ function will give the highest integer order.

The produced gains for $\theta - \theta_L$ are shown in figure 5. The omnidirectional speaker gains remain constant irrespective of the variable-order as is predicted. The sub-cardioid reproduction increases directivity with variable-order whilst the rear side of the pattern is reduced in gain. The cardioid pattern has a constant zero point at the anti-pole, where the directivity and gain of the single positive lobe increases with order. The hyper-cardioid pattern has a single negative lobe at the anti-pole of the main positive lobe. With an increase in order the directivity and gain of the main lobe increases whilst the negative lobe decreases in gain. Since this method uses a base pattern of which the order can be changed variably, a user of a system can see the change in polar pattern easily. The figure-of-eight polar pattern poses a problem. Due to the equal gain of opposite polarities at anti-poles, the gains tend to infinity due to the cancellation when calculating C in equation 7. This also creates a problem since a speaker's signal would have a maximum above 1. For this reason the base polar pattern should be limited so that $0 \leq A \lesssim 0.75$ in equation 6.

4. COMPARISON

In this section we present comparisons between the previously presented variable-order, variable-decoder Ambisonics method and the method described in this paper of variable source radiation pattern.

It can be seen in the difference between the speaker gains shown in figures 1, 2 and 5 that the Ambisonics based method gives a higher degree of directionality due to higher gain at the main lobe position. However, it does also introduce more rear lobes of both negative and positive gain, as where the new method keeps the amount of rear lobes constant through the change of variable-order. The controls of altering a base polar pattern and variable-order are intuitive to an end user and have a clear distinc-

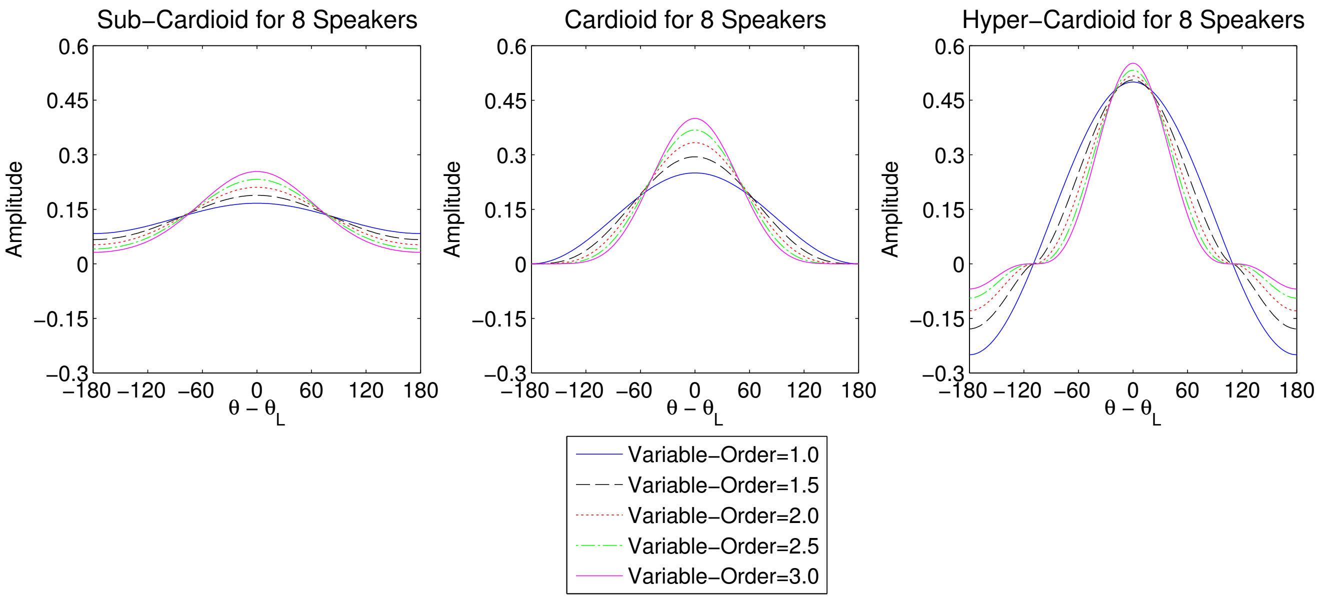

Figure 5: *Variable orders 1.0 to 3.0 for sub-cardioid, cardioid and hyper-cardioid polar patterns.*

tion when looking at the plots of altering one or the other of the parameters. With the Ambisonics method this is not the case and the two variable controls both alter the same attributes of the polar pattern, whilst doing it in different ways.

Table 1: *Comparison of variable-ambisonics and variable source pattern using rV and rE vectors to represent low and high frequency directional cues presented in Sec. 4 for order 2.5.*

		rv	rE
Variable-Ambisonics	rV Decoder	1	0.91
	Max rE Decoder	0.89	0.91
	In-Phase Decoder	0.71	0.83
Variable -Polar	Hyper-Cardioid	1.1	0.85
	Cardioid	0.71	0.83
	Sub-Cardioid	0.38	0.62

A comparison between the two methods can be drawn using the rV and rE Gerzon vectors [1, 2, 20, 22, 28] that indicate how well low frequencies (rV vector) and high frequencies (rE vector) can be localised. The vectors are given as:

$$rV = \sqrt{V_x^2 + V_y^2} \tag{8}$$

where the individual rV directional components are given as:

$$V_x = \frac{\sum_{L=1}^{N} G_L \cos(\theta_L)}{\sum_{L=1}^{N} G_L}$$

$$V_y = \frac{\sum_{L=1}^{N} G_L \sin(\theta_L)}{\sum_{L=1}^{N} G_L} \tag{9}$$

and rE is calculated as:

$$rE = \sqrt{E_x^2 + E_y^2} \tag{10}$$

where the vector components are:

$$E_x = \frac{\sum_{L=1}^{N} G_L^2 \cos(\theta_L)}{\sum_{L=1}^{N} G_L^2}$$

$$E_y = \frac{\sum_{L=1}^{N} G_L^2 \sin(\theta_L)}{\sum_{L=1}^{N} G_L^2} \tag{11}$$

Table 1 shows this comparison using 2.5 order. The top row shows the results for the Ambisonics method and the bottom row the results for the variable source approach. The in-phase decoder and cardioid pattern both produce the same polar pattern and so have the same rV and rE values. High frequencies are expected to be localised better than the lower frequencies. The rV decoder and hyper-cardioid are similar in that they both have rear negative lobe(s). The hyper-cardioid has an rV above 1.1 which is unseen in Ambisonics unless the decoding is done for an order that the speakers cannot be replayed on and in that case is an error. The rV decoder however has better high frequency localisation than the hyper cardioid. The sub-cardioid, as one might expect, would expect to have poor localisation for both high and even more so for low frequencies.

5. REAL-TIME APPLICATION

A demonstration application was built using Max/MSP that is controlled and fed audio by a digital audio workstation (DAW). Audio is sent from each track using outputs via Jack OS X audio router as monaural sound sources. Control data is sent from a VST (Virtual Studio Technology) plugin on each audio track using the OSC (Open Sound Control) protocol [29] using a similar idea as [30]. The audio plug-in does not process the audio in any way as its only use is to communicate OSC commands in this system environment.

The VST presents controls for the user the source Azimuth, Pattern, Order and Speakers. The Azimuth control is ranged [-180 180] anti-clockwise. The Pattern control varies the base polar pattern. The Order control alters the variable-order of the sound source. This control's range is altered by the Speakers control as described in section 3. Therefore it can be set as a relative maximum order, especially if the audio mixture is going to be played back over different loudspeaker configurations. The Speaker control has the range [4 12] in integers to represent the amount of speakers in the reproduction array. Finally the VST has 20 programs. These programs are presets to change the audio track the VST is altering in the application. When changing program the other controls remain the same. On an implementation level, a clear signal is sent to the current audio track and the control positions sent to the new audio track so the user does not need to manually alter a parameter so that it is sent to the new object.

The Max/MSP application presents the user with minimal controls since they are for the most part received from the VSTs within the DAW project. The user can turn audio processing on/off, select the sound source's graph to be plotted, view the number of speakers being used and see output meters for the 12 possible loudspeakers. Of most interest to a user is the graphs that are plotted. This is a plot of the polar pattern being used by the chosen sound source. The positive lobe is shown in red and the negative in blue within the applications display window. This is plotted on top of up to 12 black circles representing the loudspeaker positions. This gives the user visual feedback of how the controls of the VST are affecting the sound source reproduction. The graph to make things clear is normalised, meaning equation 7 is ignored for plotting purposes to avoid confusion to the user.

6. CONCLUSION

An approach to 2D sound reproduction has been presented based on concepts from Ambisonics. The new method does not use B-Format or another sound field representation format, but instead calculates the gains based on the speaker positions. Creating the speaker feeds in this way reduces the amount of audio calculations performed and removes errors caused by non-matching normalisation methods between the encoding and decoding stages.

By using the speaker array as a variable polar pattern of variable order the user can use spatialisation effects to alter a sound source directionality as well as width. This is opposite to how sounds are usually recorded where a microphone with a given polar pattern is then replayed as a single location.

Spatial audio reproduction generally relies on all material being of the same order and being reproduced by the same decoder. This means that each sound source has the same directional characteristics. By removing the static decoder and fixed order of the system a user can creatively choose how to reproduce sound sources, giving each a different directional attribute.

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