The Ambisonics C-Format for Super Stereo: an open-source decoder

Mastrorillo Alessio
Conservatorio N. Piccinni, Bari
mastrolex@gmail.com

Silvi Giuseppe
Conservatorio N. Piccinni, Bari
grammaton@me.com

Scagliola Francesco
Conservatorio N. Piccinni, Bari
francesco.scagliola@gmail.com

ABSTRACT

In this work we introduce an Ambisonics decoder, an implementation of UHJ stereo in the Envelop for Live suite (E4L). This format was developed by Michel Gerzon and is an alternative to conventional stereo and binaural decoding. We also present a sixth-order quadrature filter made with a cascade of Biquad all-pass filters, necessary to derive the UHJ stereo format from the First Order components. This decoder is developed in *gen~* codebox and is available as open-source repository.

1. INTRODUCTION

1.1 The UHJ format

Ambisonics C-format, also called UHJ from *Universal UD-4*, *Matrix H* and *System 45J*, is a development of the Ambisonics technique specially designed to be compatible with mono and stereo *(2-channel)* reproduction systems. It is stereo compatible without further decoding, while giving the listener a stereo image that is significantly wider than conventional stereo, both listening with headphones and stereo speakers. It conveys audio information from the entire Ambisonics First-Order (FOA):

. . . The front-stage material is reproduced with sharply defined images, occupying virtually the whole of the stereo stage, and some sound positions can appear marginally beyond the loudspeakers. Rear-stage sounds appear with rather less well-defined images between the stereo loudspeakers [1].

It represents an alternative to binaural, which keeps the entire sonic sphere but only works on headphones, and stereo decoding that is made by positioning virtual microphones in the sphere, thus excluding audio information coming from other directions. This format works with the four *WXYZ* components and derive four *LRTQ* components that can recreate the entire sonic sphere, where *LR* are two signals ready for stereo reproduction. For UHJ stereo applications, the first three components *WXY* are used, thus covering the entire horizontal plane.

Ambisonics UHJ can also be mixed down with classic stereo material: an example of UHJ stereo encoded material is present in many commercial album releases such as The Alan Parsons Project's *Stereotomy*, Paul McCartney's *Liverpool Oratorio*, Frank Perry's *Zodiac* and in many symphonic recordings [2].

1.2 The Envelop for Live suite

Envelop for Live (E4L) is an open-source audio production framework for spatial audio composition and performance. Envelop for Live operates within the music production environment of Ableton Live 10+ and Max for Live. Envelop for Live is designed to be a highly modular, flexible platform for artists to compose and perform spatial audio, and for developers to create new kinds of audio effects for the Ambisonics domain [3].

Since no UHJ decoder was available in this suite, the author has collaborated with the community of developers in *Envelop Sound*, by developing a brand new decoder, which has been therefore included in the suite [4]. This decoder has been used for the author 1's electroacoustic music composition *Trippin' on the edge of time* and *Emptiness of the hanging* stereo master, premiered and selected by Electric Sound, SMC, ICMC, eviMus and Sin[x]Thésis.

2. THE GERZON'S MATH

In Gerzon's *Ambisonics in Multichannel Broadcasting and Video* [1], the formula to convert B-format into UHJ has a 90° shifted *j* component, as follows:

$$\Sigma = 0.9397W + 0.1856X \tag{1}$$

$$\Delta = j(-0.3420W + 0.5099X) + 0.6555Y \tag{2}$$

while the Left and Right signals can be derived with the following equations:

$$L = 1/2(\Sigma + \Delta) \tag{3}$$

$$R = 1/2(\Sigma - \Delta) \tag{4}$$

This equation takes in the three ambisonic B-format *WXY* components and put out Σ and Δ, from which the stereo left and right signals can be derived. In Fig. 1 are shown the polar patterns of the Σ and Δ intermediate components.

The *j* component is obtained by quadrature. A sinusoid can be expressed as the sum of a sine function (phase zero) and a cosine function (phase π/2). If the sine part is the *in-phase* component, the cosine part can be called the *phase-quadrature* component [5]. In general, phase-quadrature means '90° out of phase' [6]. The mathematical procedure to compute the phase quadrature is known as Hilbert Transform [7].

When computing such quadrature, the output will be decomposed in two components, with a relative phase distance of 90° between the two.

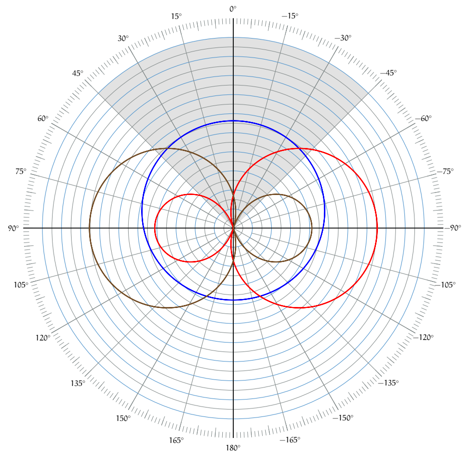

Figure 1. Σ in blue in the centre, Δ and $-\Delta$ in red and brown in the sides.

3. IMPLEMENTING THE NEW DECODER AND QUADRATURE FILTER

The decoder is written in *gen~* codebox which is an extension of the MaxMSP patching environment, allowing the code to compute at the sample-rate frequency, *'per-sample'*, for minimum latency and accurate timing. The coefficients in this *gen~* algorithm are calculated with *double-precision*.

The filter for the signal quadrature is made by a cascade of three Biquad all-pass filters with equation:

$$y[n] = b0w[n] + b1w[n-1] + b2w[n-2] \quad (5)$$

$$w[n] = x[n] - a1w[n-1] - a2w[n-2] \quad (6)$$

for the real part and three more for the imaginary part, as in Fig. 2. This sixth-order filter can be configured as desired, and the order can scale to higher values by adding more Biquads to the cascade.

The E4L suite internally uses AmbiX format with ACN channel order and SN3D weighting, thus a $-3db$ gain is applied to the W component to convert the content to FuMa format, as required by Eq. 1 and 2.

The signal goes in two filters as in Fig. 2. The coefficients for these filters are calculated considering the values of control frequency, quality factor and sample-rate. The three *WXY* components are filtered to make the *Wr, Xr* and *Yr* real parts and the *Wi* and *Xi* imaginary parts. In the Eq. 1 and 2 the real parts are used in both Σ and Δ, while the imaginary part is only used in the *j* arguments in Δ.

3.1 The search for the control frequency and quality factor values

The signal generated by 31 sinusoidal oscillators with amplitude 1, with frequencies subdivided by third of octave from 20Hz to 20000Hz, has been connected to the same amount of filters for quadrature. The control frequency and quality factor in each filter are searched observing the output of the complex equation:

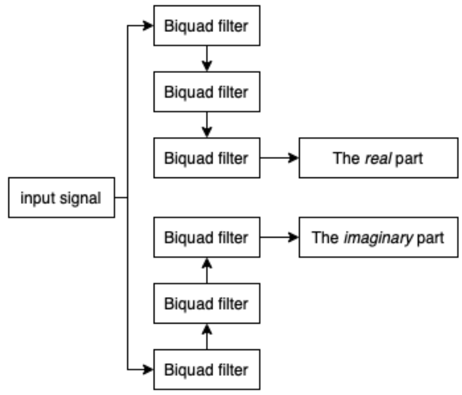

Figure 2. The quadrature filter

$$Re(x)^2 + Im(x)^2 = 1 \quad (7)$$

where $Re(x)$ and $Im(x)$ are the real and imaginary part of the signal x. The values discovered are shown in Tab. 1.

frequency Hz	**Q factor**
for the *real* part	
374.1	0.1093
666.8	0.4210
17551	0.5750
for the *imaginary* part	
35.61	0.2571
3723	0.3464
6786	0.1200

Table 1. The values for control frequency and Q factor

The output values of the quadrature filters, tested with the Eq. 7 are shown in Tab. 2. The values of Min and Max are the minimum and maximum values in the output oscillation, where a smaller deviation from 1 implies smaller error in quadrature. A greater value, as at 10Hz or above 16000Hz, means that the phase shift between the imaginary and real part is lower than 90°, thus reducing the stereo width in the lowest and highest frequencies.

3.2 Setting the filter

The implementation of the Biquad filters and the Gerzon's equation is straightforward, stable and light in computational cost. The filter setting, on the other hand, took many working hours to be configured. The use of 31 simultaneous oscillators and filters is expensive in terms of memory and CPU time, exponentially increasing when testing the filter response at higher sample-rates. It is anyway necessary, since observing the filter behavior at different frequencies at the same time is essential to set it correctly. The control frequency and quality factor of every single filter has been set while observing all the 31 filters output, starting with the lower frequencies first.

The set of values shown in Tab. 1 have been discovered with this procedure, and are the most precise and stable out of hundreds of different attempts.

Frequency Hz	Min	Max
10	1.021	0.981
20	1.013	0.989
25	1.008	0.993
31.5	1.004	0.995
40	1.002	0.999
50	1.003	0.996
63	1.007	0.997
80	1.010	0.990
100	1.002	0.996
125	1.016	0.984
160	1.003	0.990
200	1.021	0.996
250	1.022	0.982
315	1.008	0.996
400	1.006	0.987
500	1.001	0.994
630	1.006	0.996
800	1.015	0.971
1000	1.016	0.993
1250	1.001	0.998
1600	1.013	0.998
2000	1.005	0.990
2500	1.014	0.998
3150	1.001	0.997
4000	1.011	0.986
5000	1.008	0.979
6300	1.007	0.990
8000	1.012	0.985
10000	1.025	0.964
12500	1.011	0.992
16000	1.148	0.901
20000	1.244	0.719

Table 2. Result of the test with Eq. 7

4. CONCLUSION

This decoder has been used in the decoding of Ambisonics recordings with sample-rate up to 192kHz, returning great sonic details and an enveloping sound. When decoding Ambisonics material to 2-channel mixing, the choice on the decoder depends on the final reproduction system, that can be stereo speakers or headphones. With the UHJ format, this choice can be overcome, with the advantage of preserving the entire horizontal plane. Head rotation can be implemented, along with controls for balancing the omnidirectional W and directional XY components. The inclusion of this decoder in the E4L suite could be the opportunity for further development of the quadrature filter, with a higher-order and a wider frequency response.

The Ambisonics UHJ has proved to be a strong tool in music composition. The production and execution of author 1's *Trippin' on the edge of time* and *Emptiness of the hanging* has been the opportunity to successfully evaluate the entire environment, from the production step to the concert hall.

Acknowledgments

I would like to thank my supervisor Giuseppe Silvi for letting me discover the Ambisonics world, and Francesco Scagliola for the support in implementing this technique in music composition.

5. REFERENCES

[1] M. Gerzon, "Ambisonics in multichannel broadcasting and video," *Journal of the Audio Engineering Society*, p. 31, 1983.

[2] "Surround Discography website," www.surrounddiscography.com, accessed: 2022-06-12.

[3] "Envelop for Live webpage," https://www.envelop.us/software, accessed: 2021-01-09.

[4] "UHJ decoder for E4L," https://github.com/EnvelopSound/EnvelopForLive/blob/master/patchers/decoder/e4l.decoder.uhj.maxpat, accessed: 2021-11-08.

[5] M. Gast, *Wireless Networks: The Definitive Guide*. O'Reilly, 2005.

[6] Julius O. Smith III, *Mathematics of the Discrete Fourier Transform (DFT), with Audio Applications - Second Edition*. W3K Publishing, 2007.

[7] C. B. Gold, A. V. Oppenheim, "Theory and implementation of the discrete Hilbert Transform," 1969.