

A Tetrahedral Microphone Processor for Ambisonic Recording

Fons Adriaensen

Laboratorio di Acustica ed Elettroacustica, Parma, Italy
fons@kokkinizita.net

5th Linux Audio Conference
TU Berlin, 22...25 March 2007



- A short intro to Ambisonics.
- The tetrahedral microphone.
- The Tetraproc application.
- Tetrahedral microphone calibration.
- The Fourmic JACK backend.



- Ambisonics can be viewed on many levels:
 - A systematic way to analyse and represent the spatial structure of a wave field.
 - A systematic way to analyse and represent the directional information in sound.
 - A surround sound technology, including recording and reproduction systems.
- Developed 30 years ago by British mathematicians Michael Gerzon, Peter Craven, e.a.
- Firmly based on physics, maths, and psycho-acoustics.
- Full potential could not be realised at the time.
- Today used 'inside' many professional 5.1 and 7.1 applications.
- Much research (Higher Order Ambisonics) in recent years, mainly in France.

The Oxford University Tape Recording Society (1968)



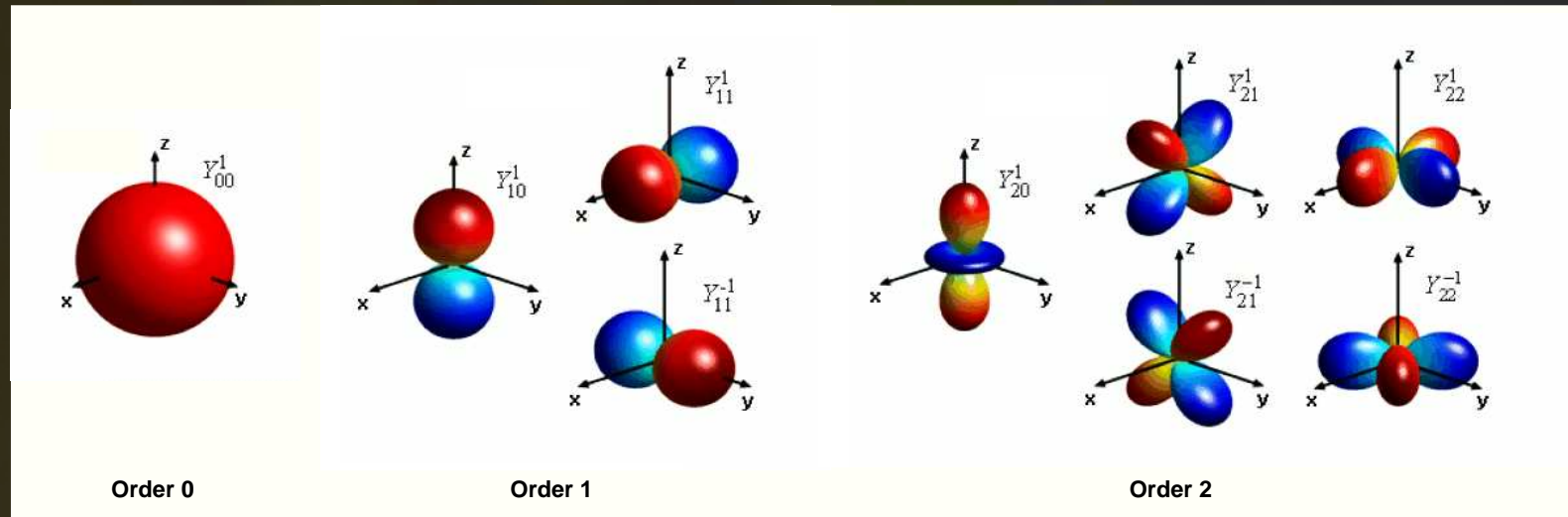
L to R: Michael Gerzon, Paul Hodges, Peter Craven, Stephen Thornton

Picture by P. Hodges, used with permission

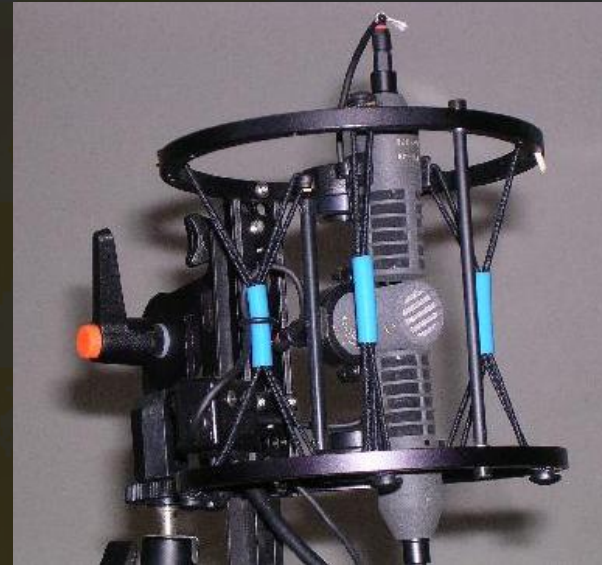
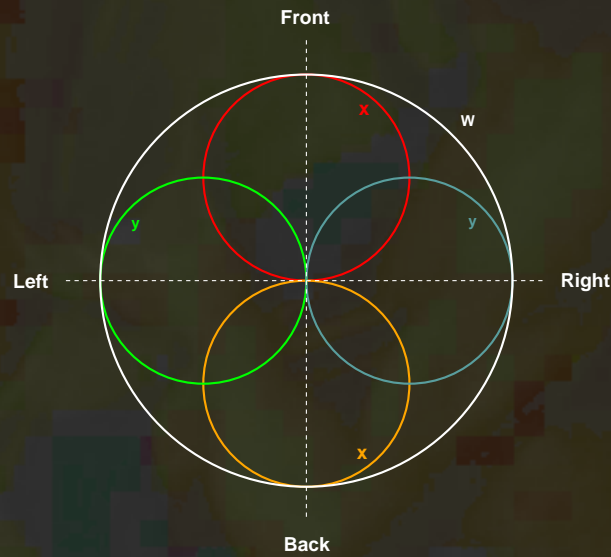


- Ambisonics encodes a sound field using a representation based on [Spherical Harmonics](#).
 - Spherical harmonics represent the 'spatial spectrum' of a sound field.
 - Similar to Fourier analysis, but on the surface of a sphere.
- Given the 'spatial spectrum', a sound field can be reconstructed.
- There are $2M + 1$ spherical harmonics of order M .
- At low and medium frequencies, only the lower orders are required.
- Good 3-D results can already be achieved using only orders 0 and 1 (4 signals).
- 2nd order horizontal only systems (5 signals) are far superior to ITU 5.1.
- Today we have working reproduction and recording systems up to 4th order, full 3-D

A short intro to Ambisonics (3)

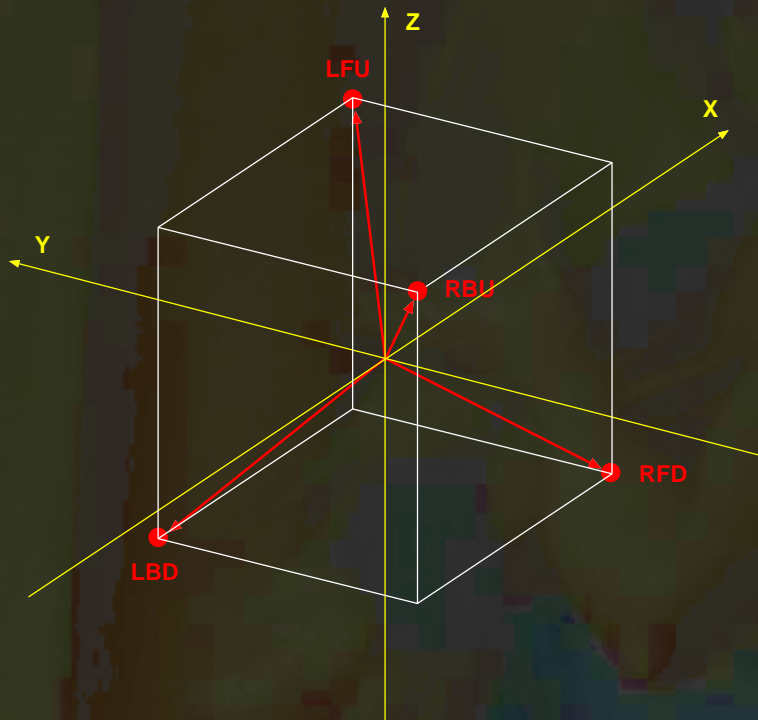


- Spherical harmonics can be seen as polar patterns.
- Low order SH correspond to familiar physical quantities.
 - Order 0: pressure (W).
 - Order 1: the three components of the velocity vector (X, Y, Z).
- A set of signals corresponding to all SH up to order M is called **B-format** of order M . This is the standard Ambisonics format.



- First-order B-format signals correspond to 'real' microphones.
- An array of 3 mics can be used for 1st order horizontal recording.
- Practical solution for 3-D is a *tetrahedral* microphone.

Tetrahedral Microphones (1)



- Four near-cardioid mics at the vertices of a regular tetrahedron, pointing outwards.
- The four mics should ideally be as close together as possible. Practical radius is 15...18 mm.
- The set of four signals (LBU , RFD , LBD , RBU) is known as [A-format](#).



- A to B format conversion is simple (in theory):

$$W = F_0(\omega) \times (LFU + RFD + LBD + RBU)$$

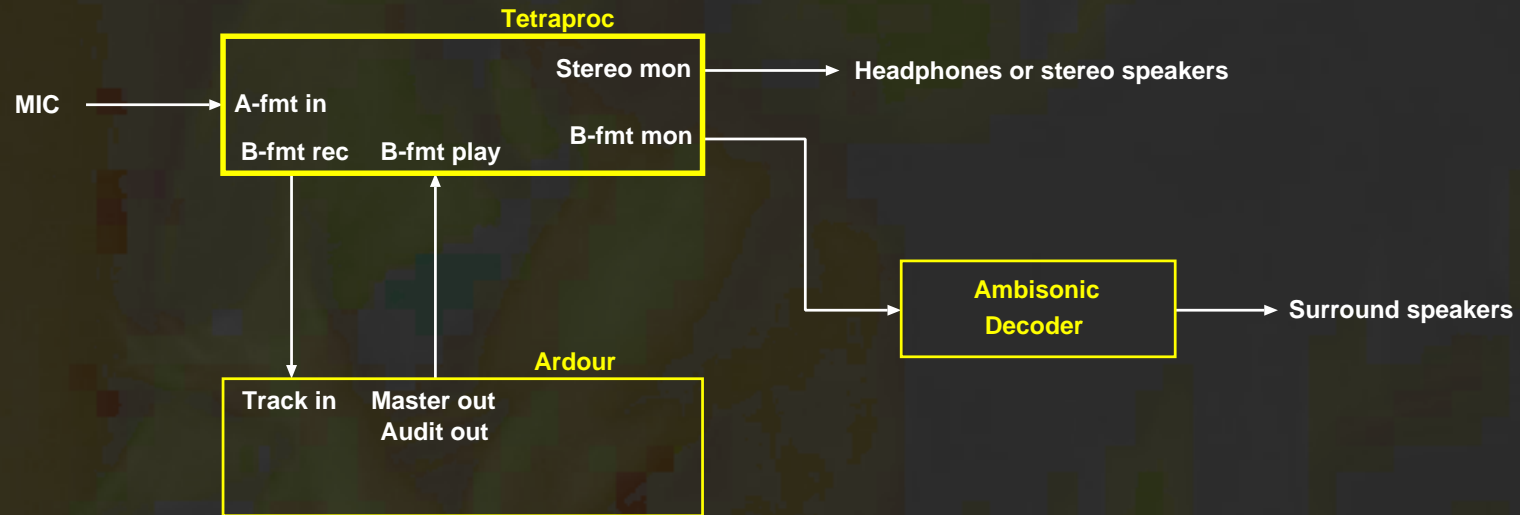
$$X = F_1(\omega) \times (LFU + RFD - LBD - RBU)$$

$$Y = F_1(\omega) \times (LFU - RFD + LBD - RBU)$$

$$Z = F_1(\omega) \times (LFU - RFD - LBD + RBU)$$

- The **post-matrix filters** $F_0(\omega)$ and $F_1(\omega)$ are necessary because the four A-format mics are not really coincident.
- The filters can't be perfect for all directions. They are always a compromise.
- In reality the matrix must be adjusted to compensate for microphone mismatch.
- A practical processor needs some auxiliary functions: high pass filters, test controls, metering, monitoring.

Tetraproc connections and GUI

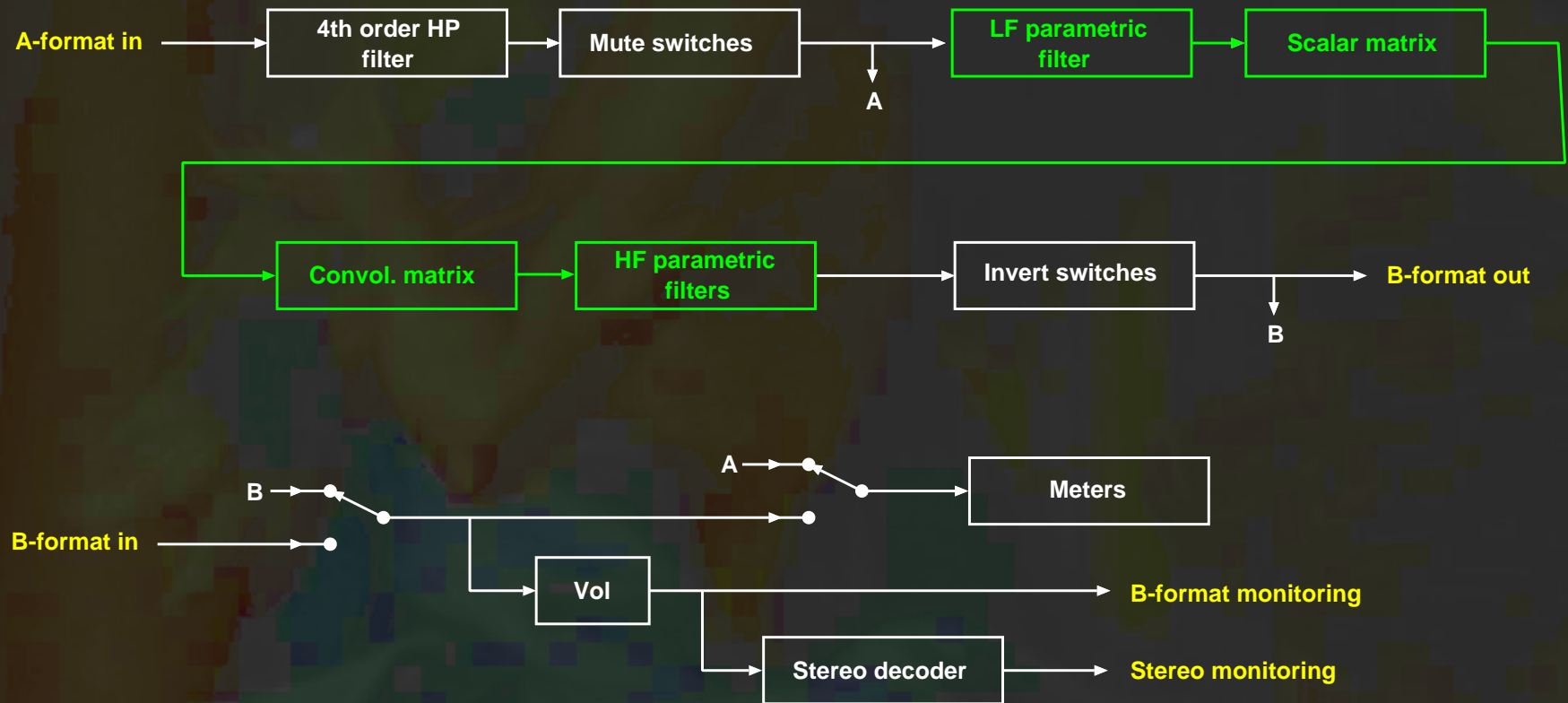


The screenshot shows the **TETRAPROC - Tetrahedral Microphone Processor - 0.0.1 [tetraproc]** window. The interface is divided into several sections:

- Recording (orange dashed box):** Includes an HPF filter with a frequency slider (10, 20, 40, 80, 160 Hz), Mute buttons for LF, RF, LB, RB, and Invert buttons for X, Y, Z. A yellow **Test** button is also present.
- Metering and monitoring (red dashed box):** Features four level meters for W, X, Y, and Z channels, each ranging from -60 to 0 dB. Below the meters are buttons for **Meters**, **Inp**, **Mon**, **Monit**, **Rec**, and **Ext**. A **Volume** slider is located at the bottom right of this section.
- Virtual stereo microphone (cyan dashed box):** Contains controls for **B**, **L**, **Azim**, **R**, **B**, **0**, **Angle**, **180**, **-90**, **Elev**, **90**, **Omni**, **Card**, and **Fig-8**. It also includes **Xtalk** and **Mono** buttons.

Annotations with arrows point from the text labels to their respective sections in the GUI.

Tetraproc audio processing (1)





- LF parametric EQ.
 - Settings derived from measured response.
 - Optionally subjective adjustment.
- Scalar A-B matrix.
 - Also corrects gain and directivity mismatch.
 - Used only if no impulse responses are available.
- 4 by 4 convolution matrix, combines:
 - A to B-format conversion.
 - Individual microphone frequency response correction.
 - Frequency-dependent gain and directivity correction.
 - Optionally also the post-matrix parametric EQ.
- Post-matrix parametric EQ.
 - Required for correct encoding at high frequencies.
 - Function can be absorbed into convolution matrix.



- Hardware A-B processors are factory aligned for one microphone serial number.
- The same is required for a software implementation.
- The calibration requires some measurements. For each of the four A-format mics:
 - Eight impulse responses at 45 degree intervals in the horizontal plane. These are used to compute the A-B matrix (scalar or convolution), and the post-matrix equalisation.
 - (Optional) On-axis, far-field impulse response, used for individual mic equalisation.
 - (Optional) On-axis, near-field impulse or frequency response, used for low frequency equalisation.



- For an absolutely minimal calibration, the impulse responses can be replaced by 4×8 level measurements, using bandlimited noise in the 500 Hz range.
- Measurements can be done in a large room, or in open air, using e.g. Aliko.
- An omnidirectional reference microphone is required.
- A separate program is used to derive matched processing parameters, and to create a configuration file for Tetraproc.



- Calibrating a tetrahedral mic is **not** a simple procedure...
 - There are many options and variations.
 - Very few people have any real experience with this problem.
 - Performing acoustic measurements correctly can be tricky.
- The methods used are based on procedures recommended by Soundfield microphone designer Richard Lee, together with new possibilities offered by IR measurement software.
- More work will be required to make the calibration as easy as possible for a non-expert user.



- MF and HF equalisation using far-field, on-axis impulse response.
 - Uses only the first 5. . . 20 milliseconds of the IR, no LF equalisation.
 - Kirkeby inversion of IR, with manual correction.
 - Result used in convolution matrix.
- LF equalisation based on near-field measurement using a small source.
 - Corrected for near-field effects using known distance and directivity.
 - Used to set the LF parametric equaliser.
- These procedures do not compensate for directivity mismatch.

Calibration - Gain and directivity mismatch (1)



- Small errors in sensitivity and directivity of the A-format mics have a big impact on the first order B-format signals.
- Eight measurements in the horizontal plane allow to compute a cross section of the polar diagram for each of the four A-format microphones.
- P and V can be computed from 2 values, but using all 8 is much more accurate.
- The real V is $\sqrt{3/2}$ times the value in the horizontal plane.



Given P and V for each mic, a matched A-B transformation matrix can be computed. If (x, y, z) are the direction cosines, the outputs of the four mics can be expressed as:

$$\begin{aligned}LF(x, y, z) &= P_{LF} + V_{LF} \times (x + y + z) \\RF(x, y, z) &= P_{RF} + V_{RF} \times (x - y - z) \\LB(x, y, z) &= P_{LB} + V_{LB} \times (-x + y - z) \\RB(x, y, z) &= P_{RB} + V_{RB} \times (-x - y + z)\end{aligned}$$

The required B-format signals are:

$$\begin{aligned}W(x, y, z) &= 1 \\X(x, y, z) &= x \\Y(x, y, z) &= y \\Z(x, y, z) &= z\end{aligned}$$

Expressing (W, X, Y, Z) as linear combinations of (LF, RF, LB, RB) now only requires solving sets of linear equations.



- A complex matter. Result is a compromise and must be evaluated subjectively.
- Automatic derivation of the post-matrix filters using the measured P, V values over the entire frequency range is a research project...
 - The maths get complicated when delays are taken into account.
 - A simple first order microphone model is no longer valid.
 - Complex diffraction effects make the analysis almost impossible.



- Good results seem to be achieved by equalising the cardinal directions.
 - Process the IR using the computed mic EQ and A-B matrix.
 - Adjust parametric EQ using displayed frequency and phase plots.
 - Or use an automated IR inversion procedure, with manual correction.
- Alternatively, equalise the diffuse field response (average power over all directions). This requires more measurements.

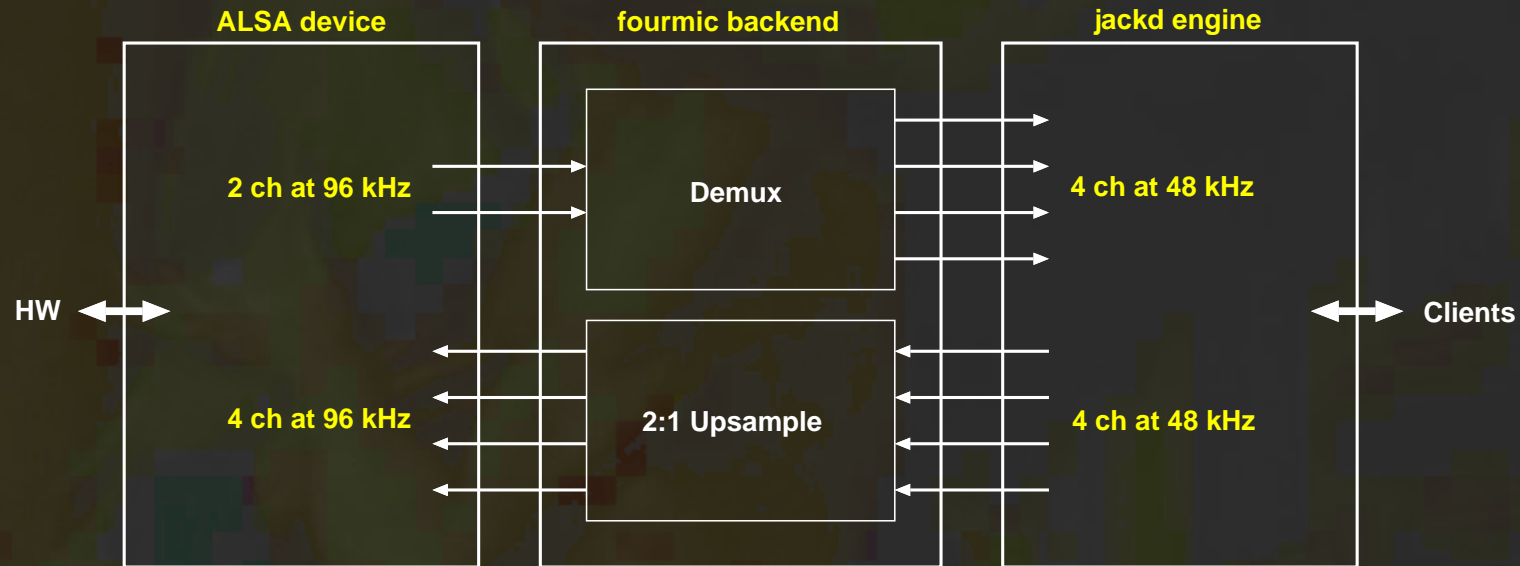


- If no IR measurements are available: use a 'standard' filter.
- Theoretical filters depend on the radius and the directivity.
- There is very little available literature, and no agreement.
- The following are all quite different:
 - The theoretical diffuse field response.
 - The theoretical cardinal direction response.
 - The filter implementation proposed by Gerzon.
 - The filters actually used in HW processors.
- More experimentation will be required.



- Using A-format mics requires capture channels with precisely matched gain controls.
- Core Sound's **4Mic** interface multiplexes 4 channels on a stereo SPDIF stream at twice the sample frequency.
- Where to do the demultiplexing ?
 - In the ALSA drivers.
 - In the user space ALSA library.
 - In an ALSA library plugin.
 - In the application.
 - ...
 - In the **JACK** backend.

The Fourmic backend (2)



- The **fourmic** backend plugin provides:
 - Demultiplexing of the 2-channel stream to the four A-format signals.
 - Upsampling of up to four playback channels for monitoring.
- Demuxing introduces an ambiguity. Automatic resolution using the 4Mic test signal may be possible.



Lots of work ...

- Manual and documentation.
- Improve the calibration procedure.
- Make some recordings.
- ...



Many thanks to Richard Lee, and to the members of the surround sound mailing list, in particular:

Aaron Heller, Angelo Farina, Dave Malham, David McGriffy, Eric Benjamin, Len Moskowitz and Paul Doornbusch.