USE OF AMBISONICS REPRODUCTION TECHNIQUES FOR POPULAR MUSIC – A JAZZ MUSIC CONCERT APPLICATION

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1 INTRODUCTION

Ambisonics has been used in many experimental and electroacoustic music applications, being able to deliver mesmerizing soundscapes to the audience. Applications to popular music have been quite rare and revealed very difficult to be achieved for the musicians and for the engineer/audio artist. This paper presents a case study of a practical application of Ambisonic reproduction techniques to popular music, in the particular case of a jazz concert. The author has been invited to create bespoke software that needed to operate in real time, allowing the "Ambisonic Sound Engineer" to be part of the band on stage as an additional musician.

All analysis presented in the paper are based on subjective perception as it has not been possible to produce an accurate assessment of the system at the time of the concert.

2 AMBISONICS SYSTEM FOR POPULAR MUSIC

2.1 Background

In February 2012, the author had been approached by an experimental music label in London with the proposal of producing a live performance for the upcoming album presentation of one of their jazz band.
The idea was to produce a new concept that would suit the music style of the band and present it in a form that would catch the attention of the press and the audience connected with the experimental music scene; it was also requested that the design would have a strong technical background.

The use of the surround sound technique was well received by the band that previously carried out its own experiments in the field, but using mainly binaural techniques.

One of the initial was to find a "surround sound system" adaptable to different venues, easy to hire, and affordable.
Ambisonics has been selected for its low cost, performance requirements and portability.

2.2 Overview

The presented Ambisonic software consists of a multichannel input/output system able to control the localization of each source independently and apply audio effects to each of the sound source independently.

The band played all instruments live on stage in an atypical jazz quintet configuration (Drums, Electric Bass, electric piano, Tenor Saxophone, Trumpet).
The software is modular and easy to be modified for different configurations.

All audio signals were processed through an external mixer managing all necessary processing for live music, and then were separately routed to a laptop running the Ambisonic encoding and decoding processing.
The processed signals (loudspeaker feeds) were routed back to the mixer sending them separately to the multichannel speaker system and to the audio monitoring on stage.

Moreover, two of the musicians requested to have a binaural mix of the Ambisonic feed to listen in real time in order to be "inspired" by the surround sound reproduction.

The laptop was provided with a custom designed software, built in Cycling 74 Max environment [1], which allowed to control localization of the sources, add audio effects and provide the binaural mix for the monitoring (same signal to audio engineer and musicians).
This approach allowed the "Ambisonic Sound Engineer" to be an integral part of the show and play with a system similar to a musical instrument.

The schematic below shows the basis of the system audio capture and subsequent processing for Ambisonic reproduction.

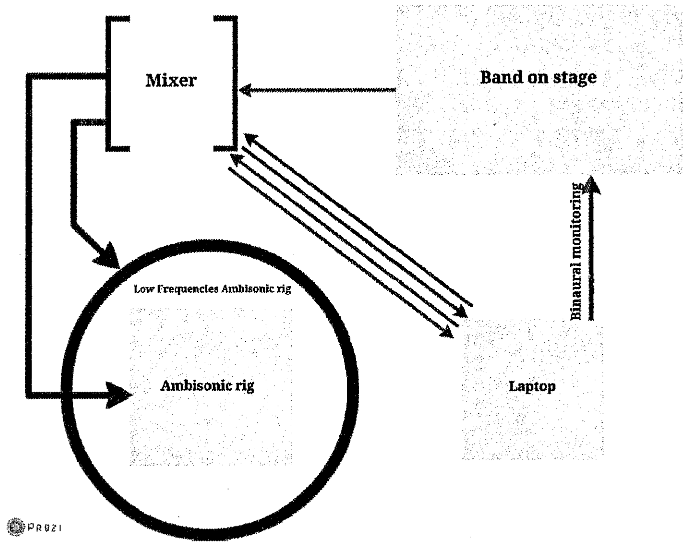

Figure 1: Schematic of the audio signal distribution

Each element of the audio processing is described in the next sections.

2.2.1 Band on stage

The band was a 5-piece electric jazz combo:

- Soprano saxophone
- Trumpet
- Electric piano and synth
- Electric bass
- Acoustic drums with electronic modules

The distribution of the musician on stage was typical of many jazz and rock bands with the wind instruments on the front and the other instruments on the back (keyboards on the right, bass in the middle, drums on the left).

The main issue related to the band on stage was about sound spilling to the audience creating problems to the Ambisonic reproduction due to the loud direct sound to the audience.
All instruments were captured with microphones, including all electric instruments that were connected to amps with microphones at the front.

The band rejected the use of sound separating screens, especially around the drums, as they wanted to keep a natural sound on stage similar to the one in the rehearsal studio.

This is typical to Ambisonic system design for live concerts. and it influences the decision of the processing of sources localization that needs to take into account the strong direct sound coming from the stage.

Another element to take into consideration for the band was the monitoring on stage.
For this experiment the band preferred to have standard floor monitors for the wind instruments, the drums and the bass, and binaural monitoring for the sound engineer and two of the instruments (Electric piano, Trumpet).

Unfortunately floor monitors provided additional direct sound to the audience ata different time of arrival compared to direct instruments' sound, creating additional problems to the Ambisonic reproduction.

2.2.2 Mixer

An external mixer for the sound mixing and the loudspeakers feed was used to provide a back-up element in the eventuality that the laptop and relative software would crash or have any problem related to the digital processing. A redundancy has also been created in order to create easy accessible stereo mix to be sent to the front speakers if the laptop fails or crashes.

The mixer needed to provide enough inputs to receive the signals from the stage and from the laptop processing the speaker feeds (Number of stage inputs + loudspeakers signals). It also requestedenough outputs to send the stage audio to the processing laptops and the signal to the speakers including the redundant stereo signal (Number of stage outputs + loudspeaker signals + stereo redundancy).

The following shows a simplified formula to calculate the number of analog inputs/outputs for the mixer:

$$INs = INS + SPKS$$

$$OUTs = INS + SPKS + STred$$

where:
INs = Number of mixer inputs
OUTs = Number of mixer outputs
INS = number of audio channels from instruments on stage
SPKS = number of speakers of the Ambisonic rig
STred = number of speakers for stereo redundancy

The mixer used during the concert was a digital mixing console (Digico SD9) with an expansion system (D-Rack) providing up to 32 analog inputs and outputs.

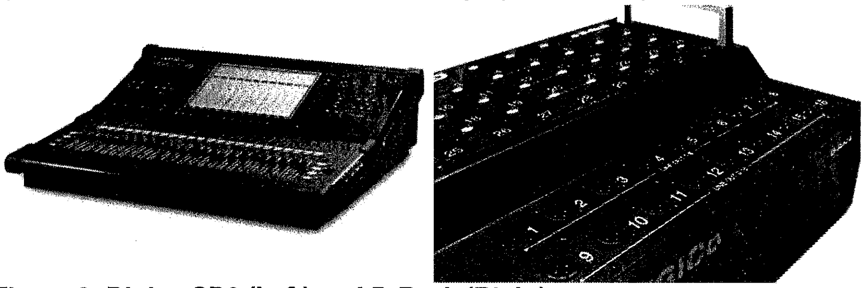

Figure 2: Digico SD9 (Left) and D-Rack (Right)

Any type of mixer, that can provide enough analog inputs and outputs, could be used for the routing purposes of the Ambisonic system.

2.2.3 Ambisonic loudspeakers rig

The loudspeaker rig is made of two main rigs with different speakers' configurations and decoding processing:

1. Mid/high frequency rig (300 Hz to 20kHz): 8 x loudspeakers (2 way powered monitors) distributed around an octahedron (inscribed in a circle of 10 m diameter), placed on tripods at about 1.2 m height form the floor (measure taken from the mid point between the two drivers)
2. Low frequency rig (up to 300 Hz): 4 x subwoofers distributed at the corners of a square of 10 m side

For the experiment the following speakers have been used and their choice have been dictated only by the availability of the hiring company.

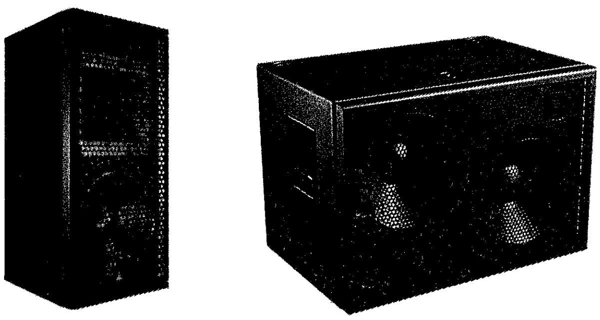

Figure 3: Meyer Sound UPJ-1p (Left, mid-high freq rig), Meyer Sound USW-1P (Right, low freq rig)

The room was a large concert venue with an average RT_{mf} = 1.8s, typically used for rock concerts.

The following Figure 4 shows a sketch of the speakers' configuration on the venue plan.

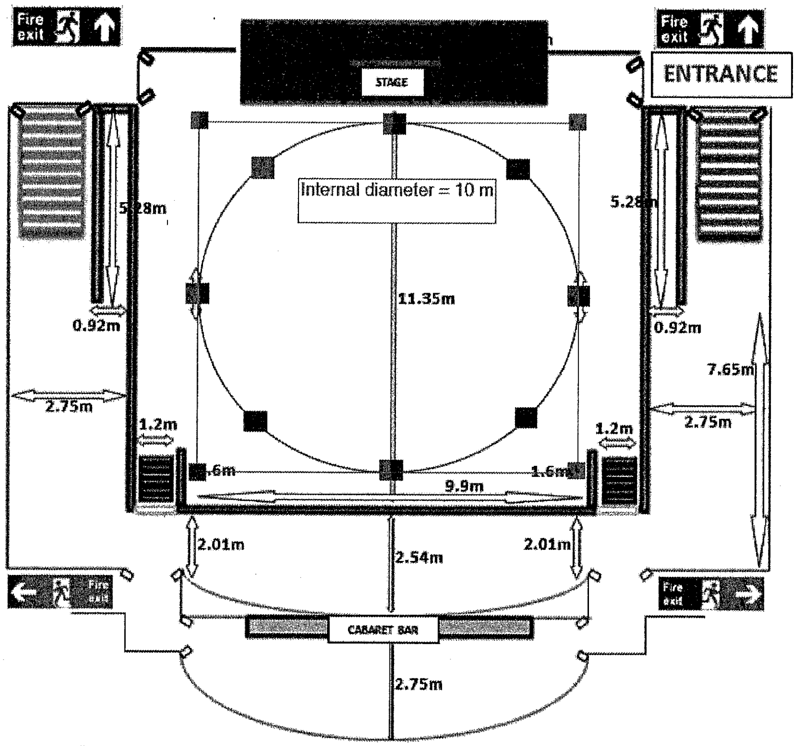

Figure 4: Loudspeakers configuration sketch on venue plan

The need of distributing the subs in a quad configuration was influenced by examples found in literature and by the intention of controlling low frequency localization [2] [3].

The diameter of the main loudspeaker rig was influenced by the dimension of the audience area (a rectangle of about 6 m x 7 m with the presence of about 35 people) in order to obtain a minimum distance between listener and speaker of about 1.5 m.

Using high order Ambisonic, the system was able to have a large sweet spot that could cover a large part of the audience area [4] [5] while the use of in-phase decoding [6](adjusted gain factors to the decoder matrix to obtain that all speakers produce equally phased signal) helped to improve localization for the external areas of the audience that were closer to the speakers.

Speakers' distance from the center has been carefully measured using a laser range finder (with an error of about +/-1 cm) in order to obtain a regular loudspeaker array.

All loudspeakers levels have also been checked with an SPL meter set in the center of the audience, with the microphone facing upwards, feeding pink noise from the main mixer to one loudspeaker at a time in order to calibrate their level within +/- 0.1 dB; the check has shown that the loudspeakers were almost perfectly level matched.

The relative level between the low frequency rig and the Mid/high frequency rig, has been checked by measuring pink noise to all speakers and matching spectrums with an SPL meter placed in the middle of the audience.

At the time of the concert set-up, it has not been possible to prepare a DRC (Digital Room Correction) software to measure Impulse response of each speaker and calculate an inverse filter for correcting disturbing early reflections [7].

It is obvious that this analysis would have improved the phases and frequency responses of the speakers and reduce disturbance from room defects but could also have reduced the dimension of the sweet spot and introduced disturbing artifacts in the listening positions outside it.

2.2.4 Laptop

The laptop is the main processing element in the Ambisonic system.

It was equipped with two external soundcards connected via firewire IEEE 1394 to the computer and via digital ADAT between them, providing a total of 16 analog inputs and 20 analog outputs. Each input had an high quality preamp that could control the input gain for the signals coming from the mixer (all gains were set to the standard line input value as set by the sound card manufacturer) while each analog output level was controlled by the main output knob placed on the front panel of the sound card.

Single output control was very useful in order to set an equal level to all speakers that have been calibrated for level matching.

3 SOFTWARE

The software used for the performance has been developed in Cycling'74 MAX [1].
MAX is an object based programming environment that offers tools for DSP, data and video processing; it helps creating flexible customized software (called patch) with real-time operation.

The customized patch is divided in different modules, as shown in Figure 5 below:

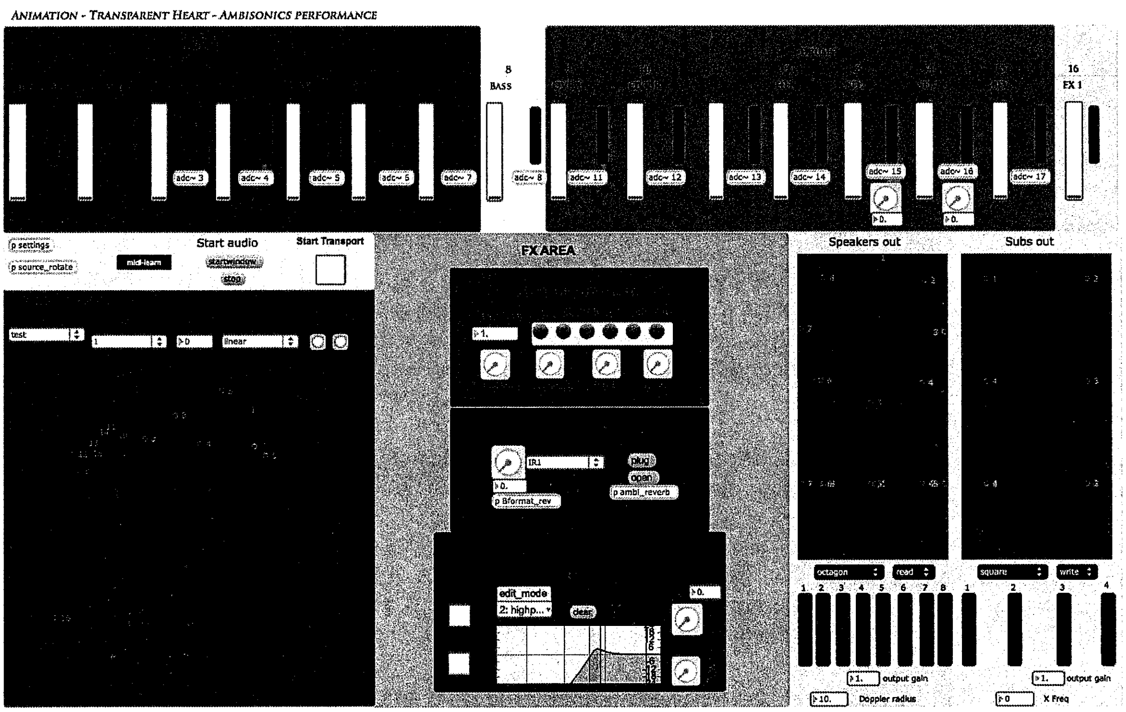

Figure 5: Custom Ambisonic software created for the case in study

All the design is based on real-time operability for source location, source rotation and audio effects (3rd order Ambisonic convolution reverb, Ambireverb, 3rd Ambisonic delay, Modulating digital delay). The graphical interface provides a visual feedback to the sound engineer with its modules controlled via midi controllers connected to the laptop.

The software use extensively the following tool box, developed by MAX users and provided as open source by the authors:

- ICST Tools [8]: the main Ambisonic encoder/decoder and sources' location control
- HISS tools [9]: convolution and IR measurements tools
- Ambi~ [10]: Ambisonic encoder/decoder tool used as soundfield rotation in some FXs

The ICST Tools are a very powerful set of Ambisonic tools that uses Furse-Malham and N3D/S3D coefficient for encoding/decoding but do not provide NFC (Near Field Compensation).

For the purpose of this application, Furse-Malham coefficients have been used, as they are better suited for Ambisonic coding/decoding up to 3rd order.

The software modules are:

- Mixing
- Source Localization
- FX
- Speakers configuration
- Binaural mix

3.1.1 Mixing

This module is a simple line mixer where all inputs are routed to the relative outputs, without mixing the sources, and it is used mainly for controlling input level of each source, output level of FX return and FX mix level.

It also contains a routine that control the number of input channels and adjust the Ambisonic encoder inputs accordingly.

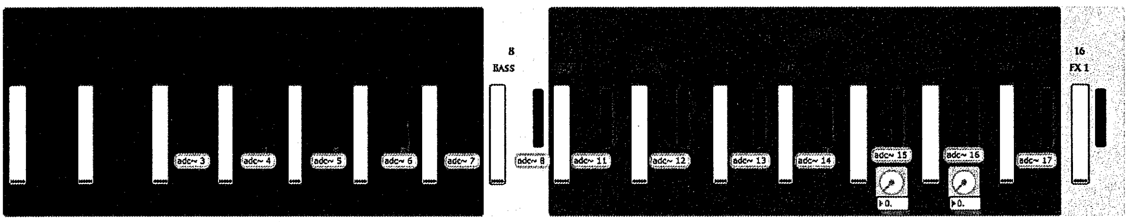

Figure 6: Mixing module

3.1.2 Source localization

This module is the main interface for controlling source location in the Ambisonic field.

It has a "radar" type graphical interface where all sources are represented as small circles.
The module can save snapshot of the sources location and recall them easily from a drop-down menu.
It also provides the possibilities of recording and recall custom trajectories and simple rotations around the central axis, together with control over velocity of the movement and the possibility to adjust the location manually.

It was used during the performance to create movements in the sound field and as a main performance tool for the sound engineer.

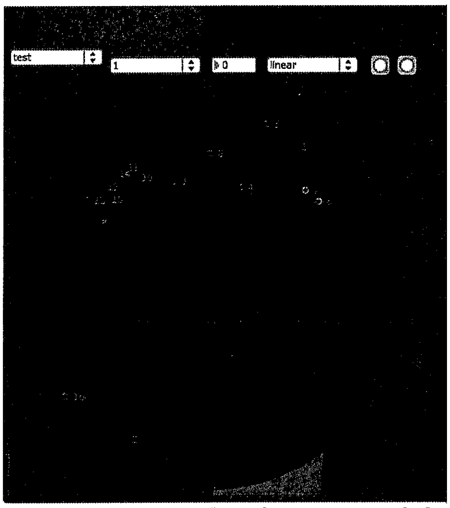

Figure 7: Source localization module

3.1.3 FX

These modules contain all the audio effects included in the software and provide a graphical control interface for the main parameters that can be tweaked in real time.

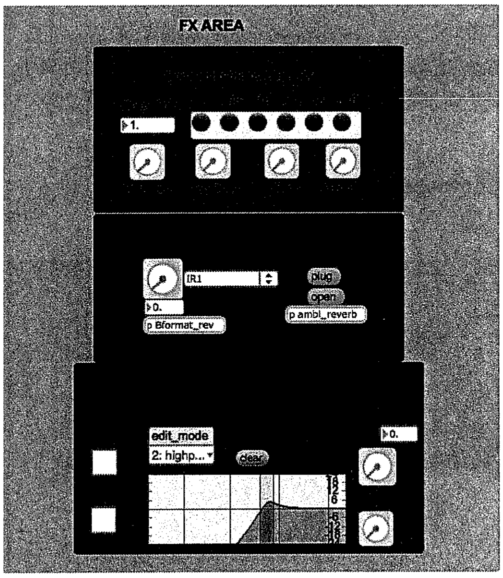

Figure 8: FX module

There are different types of sound effects in the module:

- 3rd Order Ambisonic convolution reverb
- Ambireverb
- 3rd Order Ambisonic delay
- Modulating digital delay

The **3rd order Ambisonic convolution reverb** is based on real-time partitioned convolution using 3rd order Impulse Responses generated in Acoustic modeling software simulating different rooms and source/receiver locations.
It provides control over the % of effects applied to the direct signal and the impulse response used;

IRs are manually loaded through a menu in the software graphical interface or via an external controller.

The effect also gives control over a graphical filter that equalize the wet signal.

The reverb is applied to the whole channels of the Ambisonic encoding.
Movement of the sources creates a surprising realistic effect but, when listening to a single source rotation, it gives the sensation of a rotating room more than a rotating source.
This issue is still not resolved and it is still under study.

When listening to more than one source mixed together, the "rotating room" sensation disappears as each audio components contributes to create different virtual reflections that give the sensation of a very diffusive room.

The following Figure 9 shows the programming diagram.

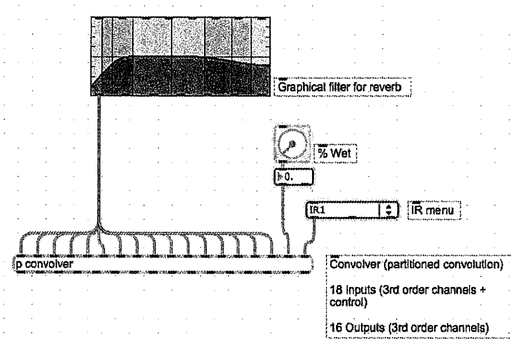

Figure 9: 3rd Ambisonic convolution reverb – programming diagram

Ambireverb is a vst plug-in developed by Dr. Bruce Wiggins as an experimental algorithmic reverb for 1st order Ambisonic.
It has been used in the performance a special sound FX on some elements of the sounds (mainly snare drum and keyboard).

The **3rd Order Ambisonic delay** is based on a typical "single delay" algorithm with the addition of 3rd order Ambisonic soundfield rotation (horizontal only).

The idea for the software has been originally discussed on the sursound list [11].

It creates a repeating echo that can decay after a certain amount of time or can be continuous, according to the amount of feedback controlled by the user.

The delayed image has a location opposite to the source image and oscillates from this position around the listeners according the values set by a rotating controller (min value +5/-5, max values +45/-45) that correspond to angles of rotation (in degrees) around the central axis of the Ambisonic system.

It provides a sensation of a very large space (larger than the venue itself) at moderate values of rotation while can be used a special effect for minimum or maximum values of rotation.
Setting the value of a number box (in Hz) can control the speed of the rotation.

A graphical filter controls a parametric equalizer applied to the wet signal; it is a very useful tool especially when used at high feedback values when the sound is overloaded.

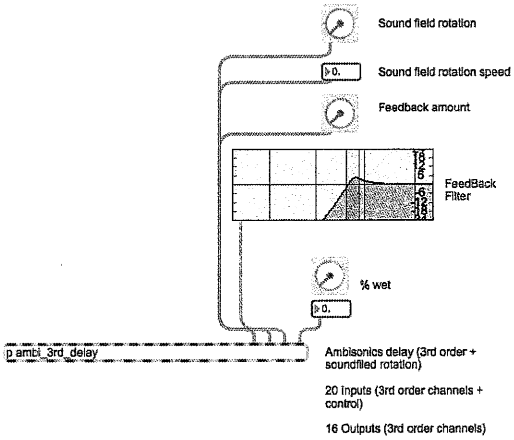

Figure 10: 3^{rd} order ambisonic delay – programming diagram

The **modulated digital delay** is a typical modulated delay (mono) that can be found in different audio application and DAW (Digital Audio Workstation) and it was applied mainly for special effects on the drums.

It has its own dedicated input in the Ambisonic encoder giving the possibility of treating its output as a separate source in the Ambisonic soundfield, applying rotation, movement and trajectories as every source.

3.1.4 Speaker configuration

This module offers control over the speakers' configuration and monitoring of the output levels. Custom configurations can be loaded, saved and recalled from a menu.
It also allows to modify the cross over frequency between the two loudspeakers' rigs.
Embedded in the code there is a binaural module based on MIT KEMAR IRs [12] that provides binaural monitoring to the audio engineer and the musicians.

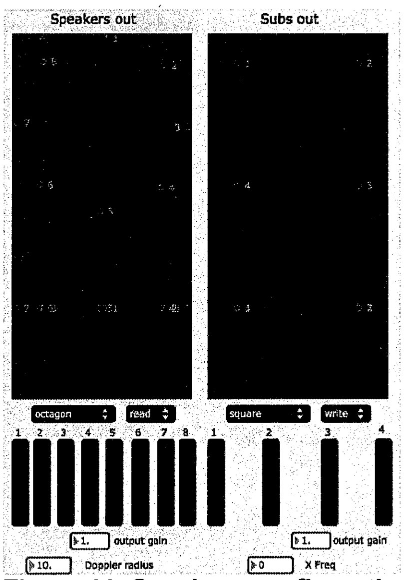

Figure 11: Speakers configuration module

4 LISTENING TEST

Initial tests of the software have been performed by using the binaural module or by auditioning on a 6 loudspeakers hexagon rig in the author's home studio showing a good source localization and enveloping audio FX.
Audio inputs were provided by the band's recording sessions of the album.

Before the main concert a series of rehearsals have been followed by listening tests with the band, the sound engineering group and different persons invited on the basis of their audio knowledge and experience with Ambisonic systems.

The first round of listening tests were performed using a simple rotating pulsing noise to understand the localization quality of the system and study the sound effects' processing quality.
The main result, shared by all participants, was that the addition of the 3rd order ambisonic reverb was not reducing localization, even if it was still providing the rotating room sensation described earlier, and that the 3rd order Ambisonic delay was creating a pleasant sensation of spaciousness.

The initial trial of the system with the band had shown confusing sound images of the main sources especially when moving and rotating the rhythmic section (Bass and drums) by losing the sense of rhythm as well; this issue has been confirmed by all the people participating to the rehearsals and has been previously individuated during the initial tests.

After maintaining the rhythmic session in the same locations as they were on stage, the confusing images disappeared and the movements of the wind instruments and keyboard have been clearly audible.

This has allowed the possibility of creating an additional "tension" to the performance where the movement of the sources was increasing the sensation of spaciousness of the sound.
The addition of the effects has also improved the localization of the sources (the 3rd Ambisonic reverb), even in areas at the border of the audience, and the sensation of movement and spaciousness (the 3rd order Ambisonic delay) experienced by all parties.

Use of extreme values in both effects (maximum % wet, extreme feedback and image source rotation) was discarded as considered quite unpleasant by the listeners.

One main unresolved issue was the loudness of the drums that influenced the sound pressure level of the whole system.
Considering the distance (about 4.5 m) between the first row of the audience and the drum kit, it was impossible to reduce the direct sound of the drums and a solution was found by reducing the level of the instrument in the Ambisonic field but not reducing its applied effects levels, maintain the beneficial sensations of spaciousness and the sound ensemble.

During these initial sessions, members of the band have been off stage to listen to the sound of the Ambisonic system expressing positive interest and providing interesting insights on the use of the software itself in a live concert situation.

The main rehearsal (1 hour before the main concert) has been very well received by the listeners with enthusiastic comments and ideas for future developments.

The main concert has been reviewed and very well received by several music blogs and music magazines in UK.

5 CONCLUSIONS AND FURTHER DEVELOPMENT

This paper presents a case study for the application of Ambisonic surround sound techniques for a live act of jazz music in a large venue in London (UK).
It describes the audio system and the bespoke software created for the performance together with presenting listening test results in a real case scenario.

Further development are required to address the rotating room sensation for a rotating source using the 3rd order Ambisonic reverb and to include the implementation of a DRC (Digital Room Correction) system in the software.

6 AKNOWLEDGEMENT

I am indebted to Dave Malham, of the department of music of the University of York, for explaining me the basic working principles of a general audio effect designed for Ambisonic signals.

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