A Simulated Ambient Acoustic Environment at PAMELA – A Pedestrian Accessibility and Movement Environment Laboratory, at University College, London

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1 INTRODUCTION

The Accessibility Research Group at University College, London, needed a facility to evaluate pedestrian movements across a range of paving layouts, and early in 2004 commissioned Arup for its design. The result is PAMELA (Pedestrian Accessibility Movement Environmental Laboratory), the main component of which is an adjustable platform that can be configured in a range of slopes and features to provide a controlled environment for researching street layouts, lighting, and paved surface types.

This facility, so far the only one of its type in the world, had to be safe and reliable. Unlike most test rigs at UCL, it was to be used by members of the general public, both able-bodied and disabled, and so this aspect needed to be considered carefully from the start.
Since the facility was financed by the Engineering and Physical Sciences Research Council (EPSRC), the anticipated cost meant that procurement was subject to EU bidding and procurement rules. This created a complication as meaningful tendering could only really be achieved if the platform was clearly defined. Arup undertook a concept study to help define and refine the tender information.

Arup's service included:
• project management, planning supervision and procurement management
• concept design of movable platform (structure, mechanisms, and control systems)
• specification and preparation of tender documents
• Category III checking of structure and mechanisms
• facility modifications
• lighting design
• acoustic engineering advice and sound system design
• supervision of rig manufacture.

2 BACKGROUND

Accessibility research is a relatively new science that examines all aspects of accessibility within the built environment. It works towards eliminating barriers to access, especially for those who experience physical, sensory, or cognitive barriers to their involvement in society. The UCL group, based at the Centre for Transport Studies in Bloomsbury, London, has a wide-ranging brief: what is accessibility, why is it important, what barriers to it exist, how can these be eliminated or reduced, and who do they affect?

Central to UCL's research is the shifting of demographic in the UK and elsewhere towards larger handicapped and aging populations, and the need to understand their difficulties and investigate appropriate solutions. The UK Disability Discrimination Act 1995, phased in from 1996 to 2004, has started to give disabled people a right of access to goods, facilities, services, and premises. These apply to pedestrian facilities and services, and in transport-related infrastructure: bus stations and stops, airports, and rail stations. PAMELA is a key tool for improving understanding of pavement layout and pedestrian facilities, and influencing policy.

From the outset, UCL wanted a world-class facility in terms of flexibility and range. In such an environment, where pedestrian movements can be evaluated under controlled conditions, research into surface type, layout, topography, and barriers such as vertical and horizontal gaps and obstacles, as well as ambient lighting and noise and audio information, can be carried out. The facility also needed to house a range of data acquisition systems to capture robust experimental data for analysis.
As well as pedestrians, particularly disabled and older people, the facility can benefit:
• designers, who will have (a) an improved database of pedestrian characteristics and (b) a comprehensive simulation tool to support accessibility audits of existing and proposed designs
• government and agencies, who will have better data on which to base policy that affects pedestrian activities
• owners and managers of pedestrian facilities (including public buildings) who will be able to use the simulation tool to help make strategic decisions about pedestrian routes (including escape routes) and the location of unavoidable obstacles and facilities such as seats and other resting places
• academics and consultants, to support further research in designing and implementing more inclusive pedestrian infrastructure.
• medical research through an ability to examine the impacts of medical conditions that impact the senses and physical movement within a controlled environment, supporting diagnosis and review of the performance of proposed treatments.

The lab wanted to introduce the effects of audio into the test environment, both to aid realism and to detect whether the presence of audio both as an ambient environment and foreground sound effects would affect the capabilities of people under test.

This paper explores the issues, options, development and implementation of a system based on directionally controllable loudspeaker arrays and ambisonics that can provide an excellent simulation of an audio environment within a reverberant factory-style building.

3 PROJECT CONTEXT

3.1 PAMELA Facilities Overview

At the time of systems development for the Ambient Acoustic Environment System, PAMELA was housed in an industrial warehouse style building. The plan area of the building is not rectangular. Walls are relatively low at 6.5m to the eaves, with lightweight trusses supporting a fabricated metal roof. Neither the trusses or roof have significant load-bearing capability. Sound insulation is relatively poor, making rain noise and passing traffic relatively disturbing, though the cul-de-sac character of the industrial estate and nature of other occupants makes day-time traffic relatively light and therefore tolerable.

The laboratory consists of three parts:
• a set of demountable paved platform units,
• a lighting system and
• a surround-sound audio system.

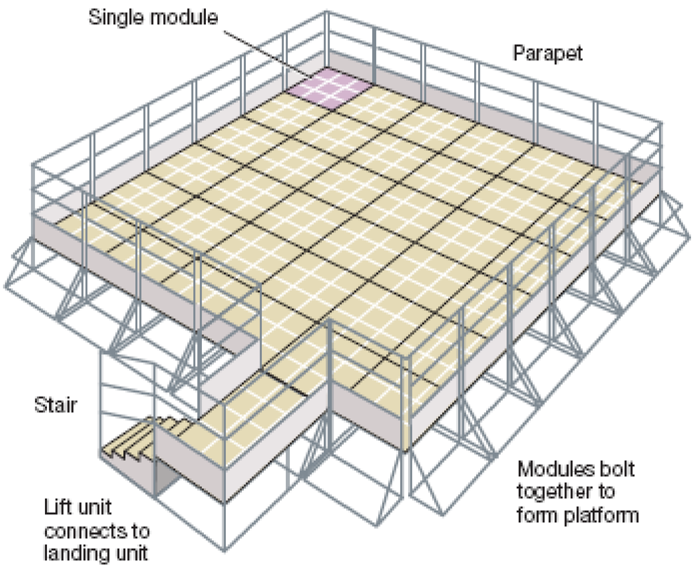
There are also several measuring systems:
• Laser Scanner Tracking system – which is used to identify the position of test subjects on the platform
• Fixation Tracking system – which observes the test subject's viewpoint with a camera, at the same time as observing their exact eye viewing positions
• Digital Video Recording system – which is generally used with a wide shot and/or handheld camera to provide an overview of test subject activities.

All systems are controlled by computers, which are, at present, independent of one another. However the data collection systems each work against a real-time clock, providing a way to synchronize these for later analysis, where necessary. The laboratory has wheelchair access and facilities for conducting interviews and other basic data collection.

The lease of the building is short term. The test platform has been extended, making space now tight within the building. These facts, plus the success and popularity of the lab, mean that either it is likely to move to a larger building, or a second parallel test lab may be constructed in a different building.

3.2 The Pedestrian Platform

The platform consists of 57 modules. These can be arranged to create different shapes of layout and topography. There are 21 passive modules, which can be set to a required height between 1m and 1.5m in 250mm intervals. There are 36 active modules, which are computer-controllable and can create step heights (±125mm) or slopes (max 1:5). These have a total range of 500mm with 0.5mm accuracy.



Each module is 1.2m square. These can be moved with respect to each other to create any desired layout for the test area, including L and U shapes. The platform surface is interchangeable (concrete, asphalt, etc.). The platform has been designed to feel like solid ground by using robust structural elements to resist horizontal linear and rotational loads as well as vertical forces. The platform includes 176 computer controlled actuators, 52 reactive lateral stabiliser columns and 84 manually adjustable legs.

On one long side of the building, the platform lighting is permanently fixed to the wall. Due to the variable shape, the lighting on the other side is built onto a large wheeled gantry that can be placed alongside the other side of the platform. The fixed lighting leads to the platform generally being located very close to (in fact directly against) one side wall, although this is not guaranteed.

The platform itself must be kept clear of any objects not directly related to the proposed test.

3.3 PAMELA Laboratory Room Acoustics

Wherever a pedestrian is walking, there is also an ambient acoustic environment. That may distract or disturb their ability to walk safely, either by soaking up some of their brain processing power, or by startling or frightening them. While it is well known that these effects occur, there has been but little detailed study of the subject.

The goal was to create an Ambient Acoustic Environment of 2 or 3 dimensional character, over the whole platform area – no matter how the platform had been set up. In addition, it was desirable to be able to include “spot effects” – for example the arrival of a bus, a baby crying next to the pedestrian under test, or a gunshot. The system had to be able to deliver the sound pressure and frequency response required to recreate the experience of standing in a car park beneath the flight path at Heathrow Airport.

PAMELA is within a medium-sized warehouse that has no acoustical treatment and therefore significant reverberation and acoustic character of its own. It would be necessary to suppress this, and to superimpose a simulation of the sort of space that the pedestrians were to be experiencing. This could range from streetside (nearly acoustically dry) to the interior of a cathedral (very reverberant). Initial work led to three options:

4 LOUDSPEAKER FORMAT OPTIONS

4.1 OPTION 1 - Use a Distributed Set of Compact, Full-range Loudspeakers

These would be mounted to walls and (possibly) ceiling
Advantages could be:

- Easily definable "zones" with different sounds in them, but less apparent "the sound is coming from that direction".
- Possibly (with the right loudspeakers) less aural awareness of loudspeaker position (though visual disguise might be needed).
- Lower purchase cost of loudspeakers and associated amplifiers.
- Change in sound quality with head rotation might be lower unless near them.

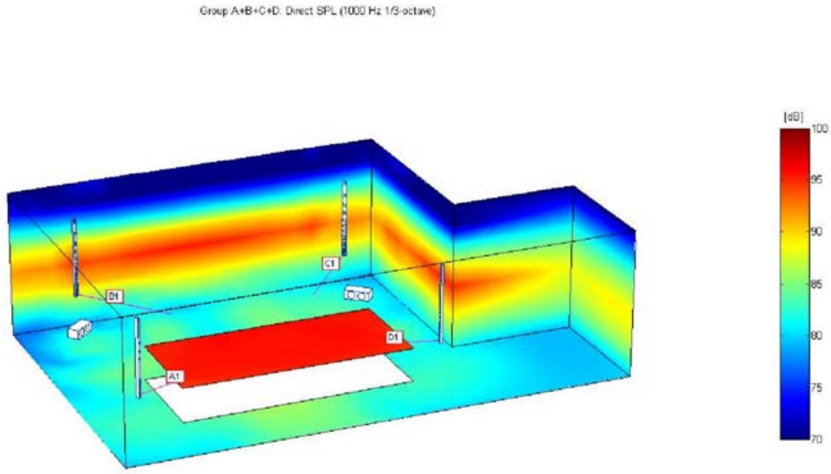
Disadvantages might be:

- Increased installation cost and time. If and when Pamela relocates, dismantling and re-erection costs would be incurred each time.
- Loudspeakers visually identifiable since they are near the listener (unless screened from view)
- Possible interaction between individual loudspeaker sound fields when near them (for example where loudspeakers are on the wall adjacent to the walkway). Therefore change of sound quality when walking along might be higher and the listeners could hear loudspeaker sound “swish” as they walked between them, especially since a lot of the typical sound from public environments is broadband noise.

Probable need for extensive acoustic treatment as the sound from each loudspeaker would be nearly hemispherical. These would excite the reverberant character of the room. Reverberation times in the PAMELA laboratory were measured, to provide information for planning any required acoustic treatment for Option 1. At 1kHz, the RT was 3 seconds. Calculations showed that the absorptive area required to bring this down to about 0.6s would be in the range of 600m^2, with half on the walls and half on the ceiling.

4.2 OPTION 2 - Use Several Phased-array Column Loudspeakers

The light text was the original set of advantages and disadvantages. The bold text provides comments from the demonstration later undertaken on site.



This option would use several phased array loudspeaker columns placed away from the walkway, near the sides or ends of the room. **(four were used for the demonstration as the practical minimum).**
The expected **advantage** of these is:

- They would not excite the room reverberant character very much due to vertically narrow beam coverage down to low frequencies, reducing any residual room acoustic effects. This might enable acoustic treatment to be reduced or even eliminated. **Comment from demonstration – reverberation almost inaudible, certainly not disturbing. No treatment required.**
- There would be little variation in sound level or spectrum as participants move toward or away from any of the loudspeakers, so not much sound change while walking about. **Comment from demonstration – it proved to be the case that variation of sound direction and loudness as listeners moved was not very noticeable, even when trying to notice it.**
- There would be less interaction between the loudspeakers themselves - therefore less likelihood of aurally disturbing effects. **Comment from demonstration – movement along or head rotation produced no disturbing aural affects.**
- Each loudspeaker could have slightly or radically different sound coming out, allowing easy simulation of spaces like railways stations where you hear a standing train in one direction, paging announcements in another and an entering train in another, all mixing together as well to create a general railway station ambience. **Comment from demonstration – this worked well using recordings made using a Soundfield microphone and ambisonic processing.**
- Mounted on wheeled stands, self-powered, remote controllable loudspeaker beams - so can be "aimed" easily for differing walkway setups, out of the way. **Comment from demonstration – from unloading the truck through to demonstration took just over an hour at the first attempt – confirming the ease of rigging.**

Disadvantages might be

- More than desirable change in sound with head rotation (though the reverberation in the room could be helpful here). **Comment from demonstration – no noticeable change in sound character with head rotation, except that the source direction for particular individual sounds remained constant.**
- Possibly a clear direction to loudspeaker sound. **Comment from demonstration – this only occurred when wanted. Otherwise the positions of loudspeakers was not distinct.**
- Higher purchase cost of loudspeakers.

4.3 OPTION 3 - Conventional 2/3-way Sound Reinforcement Loudspeakers

The final alternative was not originally seriously considered, but should be mentioned for completeness, that is the use of standard sound reinforcement loudspeakers. The expected **advantages** of this solution are:

- Intermediate cost for loudspeakers and amplification
- Beam control in the range from about 800 Hz upward would enable reverberation in the mid and higher frequencies to be controlled.
- Could be mounted on stands rather than walls, making it possible to move them easily to new premises and eliminating most of installation cost.

Disadvantages might be:

- This type of loudspeaker tends to draw attention to itself due to the varying beam control character with frequency and a higher level of intrinsic distortion of sound.
- Each loudspeaker position (practical minimum of 4 around the walkway), would require 3 loudspeakers arrayed vertically and/or horizontally with respect to each other. Their exact arrangement and physical aiming would have to be adjusted when placed in different locations around the walkway.

- Each of the arrays at the 4 locations could be expected to differ in their arrangement in order to achieve similar acoustic results. Achieving this requires a good knowledge of electro-acoustics, making it difficult for the client to undertake this rearrangement himself.
- Variation in sound level with distance could be minimised, but would require the use of external processors, manually adjusted.
- When walking toward or away from the vertical arrays, the interaction of between devices is likely to be distracting and difficult to minimise due to the mechanical constraints of this type of loudspeaker.
- The reverberation at lower mid-frequencies would rise considerably compared to the phased array solution.
- Separate external loudspeaker processing unit is essential.

4.4 Arup Soundlab Auralisation

An initial evaluation was carried out in Arup Acoustic's Soundlab. Soundlab is a 12 channel auralisation space that uses ambisonics controlled by a Max/MSP software environment.to demonstrate all kinds of acoustics issues, including room design and loudspeaker layout options. A set of example playbacks were created from recordings made on a Soundfield microphone, initially saved in ambisonics B format. Processing in MAXmsp then decodes this optimally for the playback system this is used. The playbacks simulated an urban streetside, low aircraft flyovers in a airport car park, the reverberant building interior of a major London railway station, children playing in an outdoor play area, and a racing motorcycle pass-by. For comparison, 12 channel 3D and 4 channel horizontal surround only loudspeaker formats were demonstrated. The latter represented what we expected to approximate in the final PAMELA installation. Client staff listened to the auralisations and wholeheartedly supported the approach of using the controllable line array solution.

4.5 Demonstration on Site

Following the successful demonstration of concept in Soundlab, arrangements were made for Duran Audio to bring four of their DS1608 loudspeakers to the Pamela site. These proved the concept in a one-to-one scale model – the real room. The loudspeakers are large high-powered full-range column arrays, each supported on a heavy-duty, wheeled, adjustable height stand. The Duran Intellivox DDA control software permits the platform layout, listener height and loudspeakers to be put into a computer model. This is then used to automatically compute the algorithms to control each loudspeaker driver (16 in each of the four loudspeaker systems) to give the desired coverage over the PAMELA platform. In addition, two high-powered sub-woofers provide the low frequencies and four small portable loudspeakers can be placed anywhere on the platform to provide spot effects – triggered either randomly or by the presence of a pedestrian.

4.6 Summary and Recommendations

The phased array solution was recommended based on:

- Ease of installation and relocation in future
- Flexibility
- Constant sound level with distance
- Ease of reconfiguration by users with modest training
- Least acoustic side effects in this application
- Elimination of the need for acoustic treatment to the room
- Proven effectiveness beyond initial expectations during the demonstration using four Duran Audio DS1608 loudspeakers with two sub-woofers

The resulting comments about advantages and disadvantages for this arrangement following site demonstration are highlighted in **bold** text in section 4.2 above.

5 AMBISONICS PLAYBACK

It was clear that the PAMELA pedestrian platform was not going to provide an idealized listening experience for each listener or for all locations on the platform. Sound arrival times from each of the four loudspeaker arrays would differ by up to 40mS at different listening positions from what was set up at the optimized central location. Ambisonics was chosen as the surround sound technique, because it provided the widest known area over which directional information can be maintained. It also would allow “spot effects” loudspeakers to be included (if desired) within the ambisonics mapping, and would permit the easy extension of the playback system to other loudspeaker configurations after production was completed.

More details of the background and implementation are given in the sections below.

6 SOUND PRODUCTION AND PLAYBACK

6.1 Overview

The requirement was for a complete sound production and playback system. This must be capable of handling and producing Ambisonic B-Format sound files. Additionally it was requested that sound production be possible while monitoring on headphones. If the spatial qualities of Ambisonics were to be represented, a binaural encoding would be required.

Ambisonics was originated in the 1970’s by Michael Gerzon, a mathematician with a keen interest in music and sound recording, and Peter Craven. It is an extension of the MS system, originally proposed by Alan Blumlein. It uses a number of directionally encoded signals to represent the sound field at a central listening position. These signals are based on the 'spherical harmonics', of which the fundamental is a sphere, the first harmonic a set of three figures-of-eight at right angles to each other, and the second and beyond by more and more exotic directional shapes with all sorts of lobes.[1]

First order Ambisonic recording requires an omni-directional microphone, and three figure-of-eight microphones, one pointing forward, another to the left and the third upwards. This configuration is synthesised in the ‘Soundfield’ microphone, which has been available for many years, by combining the outputs of a tetrahedral array of microphone capsules. The four resulting channels, labelled W, X, Y, Z, are enough to represent a complete 3-D sound-field. This group of signals is known as ‘B-Format’. For 2-D horizontal work, Z is ignored. B-Format signals can be mixed together to produce a composite B-Format output.

B-Format can be decoded to a variety of differing loudspeaker layouts, usually based on symmetrical arrays on regular polygons or polyhedra. The encoding and decoding are completely separate processes. A B-Format signal can be decoded for nearly any array of speakers, and thus the B-Format signal is a ‘universal’ transmission medium.[2]

Ambisonics suffered from the general disillusion after the commercial failure of ‘quad’, and the lack of any widely available delivery format with more than two discrete channels. The success of the home cinema has provided greater acceptance of more loudspeakers in the home, better and cheaper reproduction systems, and a multi-track delivery system in the form of DVD. It has triggered renewed interest in surround reproduction of music. Work on Ambisonics and its ability to provide a more precise surround directionality than screen-based systems like Dolby 5.1 Surround has continued, notably by David Malham at York University and Jerome Daniel at France Telecom.

In recent years there has been increasing interest in Ambisonics, and several attempts to make it more useable and widely known. Wave Field Synthesis is reputably better, but requires a large number of loudspeakers, amplifiers and extensive DSP. Ambisonics requires less DSP and is scalable in terms of number of channels and loudspeakers.

Modern portable recorders are 'solid-state', recording to memory cards. The recordings can be transferred easily to a computer via a USB interface. Ambisonic location recordings can be made using a 'Soundfield' microphone and a portable digital four-track recorder, such as the Edirol R-4. Although portable, this is a somewhat bulky package, which requires some care to set up correctly. It tends to draw attention to itself and modifies the behaviour of people in the environment being recorded. Recordings can be rendered useless by enquiries as to what is being recorded and why. Portable stereo recorders are smaller and very portable, often have built-in microphones, and can be got into record much more quickly. More covert binaural recordings can be made using small microphones in the ear canals of the recordist, as a person wearing earphones to listen to portable music players is a common sight and is taken as ordinary. Such recordings, binaural or 'normal stereo', and those available on CD sound effect libraries are not Ambisonic, and thus some means of encoding them to an Ambisonic format is required.

Nearly all audio production is now done on computers, with extra audio hardware to supply multiple inputs and outputs, and suitable software to supply multi-track editing, processing and mixing. Ambisonics is not generally supported in commercial audio production software. There are a number of one-off attempts to make this possible, but no complete solution.

Most software audio production software (ProTools, Cubase, Logic etc) is based on music or picture post-production practices, with a linear time-line. What PAMELA seemed to require was a system that more resembled those used in theatre: one enabling sound cues that could be played at will. It should also be capable of fulfilling future requirements, such as the integration of sensors to control sound. Thus a one-off solution, which allowed development at minimal extra hardware cost, was imperative. Max/MSP is a software programme that allows one to build and modify such systems. It is not really suitable for making an equivalent to ProTools for example, but it does allow the construction of 'plug-ins' for such packages.

Max, named after computer music pioneer Max Mathews, is a graphical development environment for music and multimedia, developed and distributed by Cycling 74. It has a large user-base of programmers who enhance the software with commercial and non-commercial extensions to the program. It is widely used by artists and programmers to produce interactive works. MSP, a set of audio processing objects was added in 1997. It has been continuously developed and improved since, adding 'Jitter' for video manipulation, been ported to Windows, and adding 'Pluggo' plug-in making capabilities.[3]

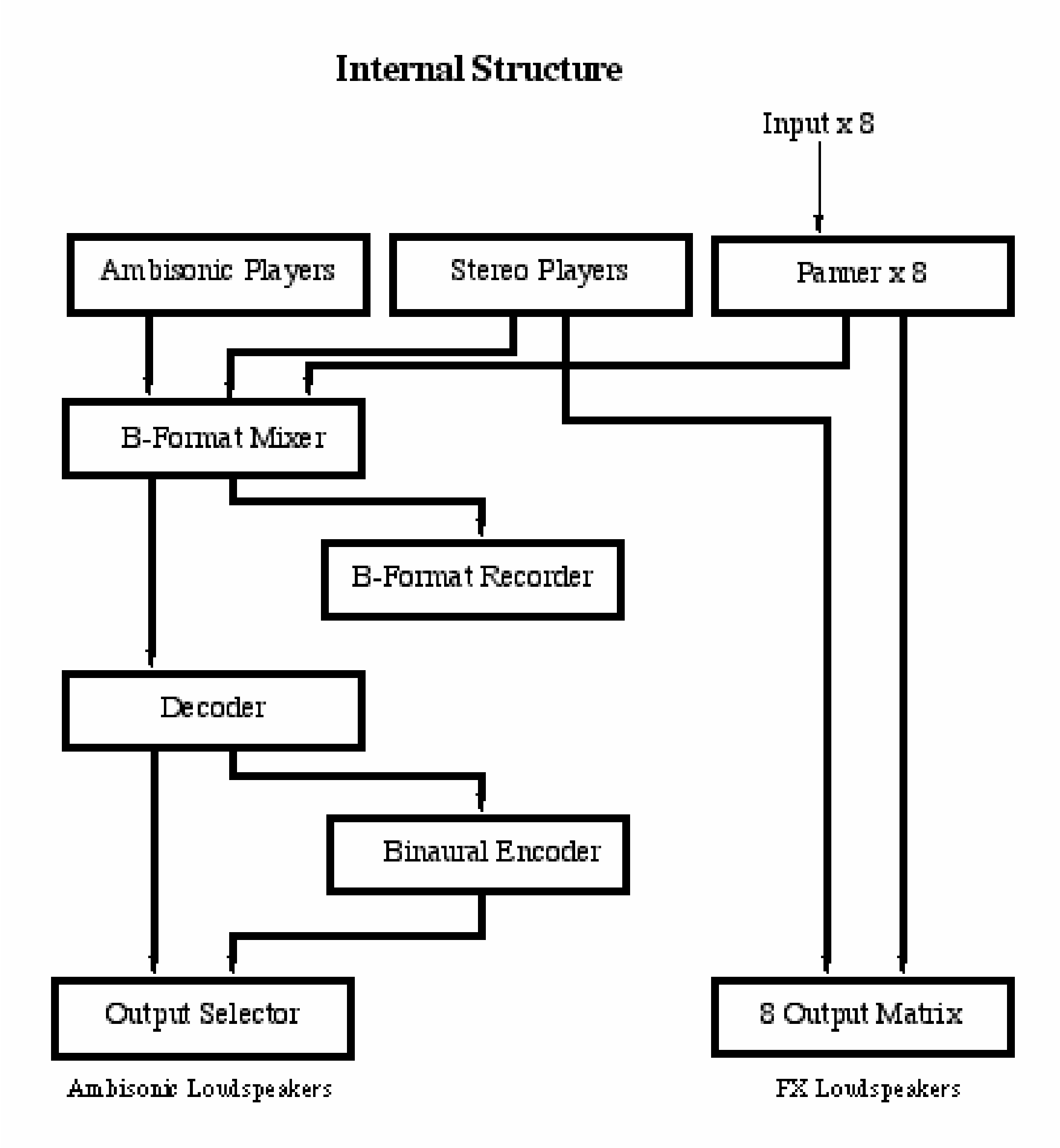
Max 'patches' are constructed from a library of relatively low-level building blocks, connected together with 'patchcords'. Some other software resembles Max in appearance, e.g. Soundweb Designer, Reaktor, Plogue Bidule. This can make for a rather steep learning curve, although it does allow you to do exactly what you need, and to make it as efficient as possible without necessitating raw code writing. Once you have a suitable audio hardware interface you can build nearly anything, without PC boards and metal and panel work. A variety of graphic tools including knobs, sliders, and labels are included. Midi control is easy to incorporate, opening up the use of the many reasonably priced Midi controllers available. Switches, potentiometers and varying analogue voltages can be interfaced, and computers can be networked, so that control data can be passed from one application to another. It could even be arranged to supply a print out of what happened when. The system can do one thing one day, and something completely different the next, and a patch can be easily transferred to another machine. The complementary minus is that a Max patch is never truly finished.

Thus the best solution seemed a computer with a multi-channel audio interface, an appropriate commercial audio software package with Ambisonic plug-ins written in Max/MSP for audio production, and a specially written Max patch for playback and system development. Fortunately the author had been working on this with Max for several years, so did not have to start from scratch.

The output of commonly available multi-channel audio interfaces, such as the MOTU 828 Mk2 is balanced and can be directly connected to the amplifiers local to each of the four Duran DS1608 loudspeakers, or the smaller self-powered loudspeakers supplied for more localised sound. Connection to a computer is via a standard a Firewire or USB2 port.

After some research, it seemed that Cubase SX, or its bigger brother Nuendo, was the only commercial software with a suitable multi-channel architecture for Ambisonic production. It supplies a multi-track capability, with considerable equalisation, dynamics and delay-based processing, mixing and automation. It can handle and produce inter-leaved or discrete multi-channel files, and uses the VST plug-in format, which Max/MSP is optimised to produce.

6.2 The Max/MSP patch.



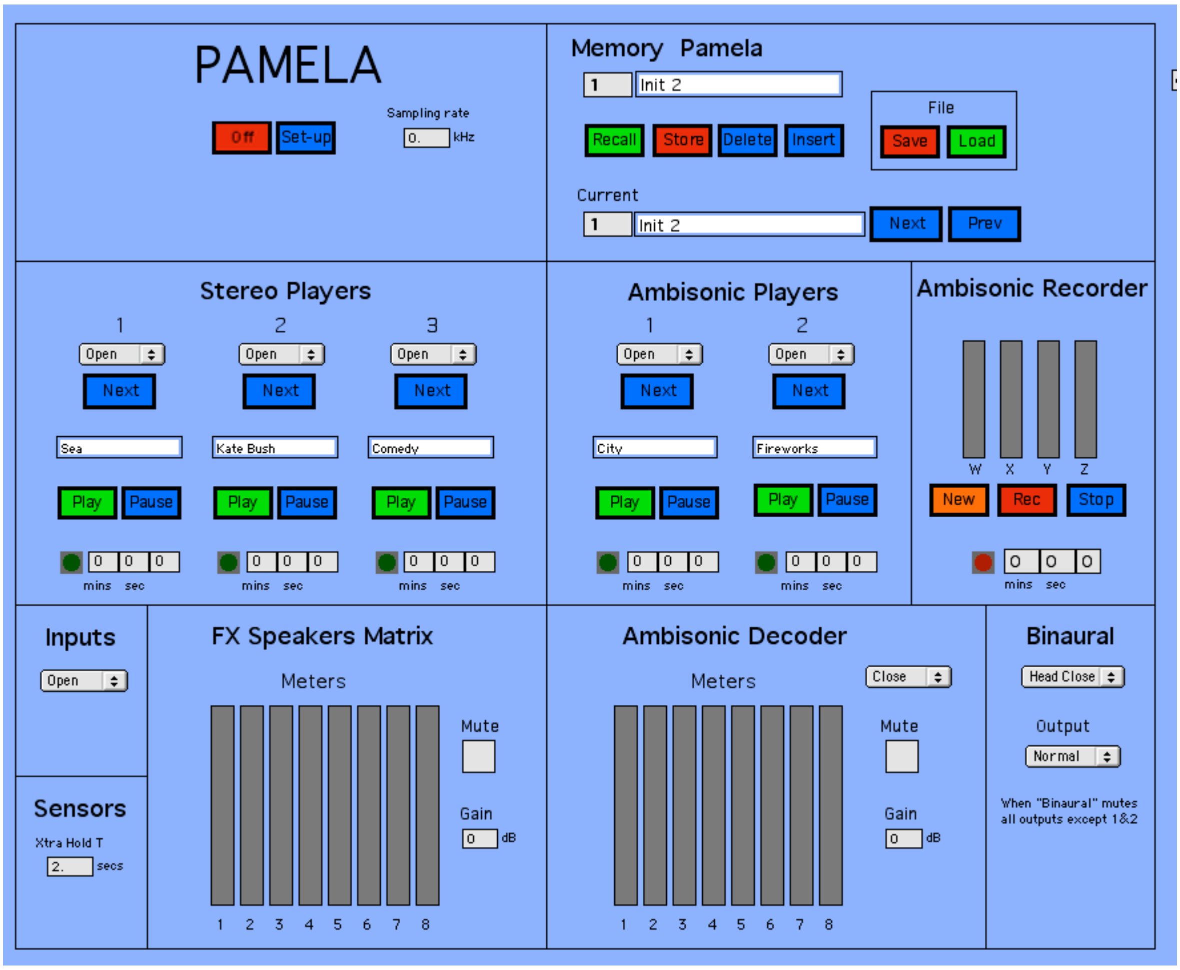
There are two Ambisonic File Players, which send their outputs to the B-Format Mixer and then to an Ambisonic Decoder, to transform the B-Format signal into loudspeaker feed signals. It was envisaged that these would supply the ambient acoustic environments for PAMELA. These background recordings could be looped. The use of two independent loops of different lengths means that the join in the loops are covered up, and that the whole sequence never repeats exactly, leading to greater variety.

There are three additional Stereo File Players. The output of these can be Ambisonically encoded and mixed to the Ambisonic Decoder. Simultaneously they can be sent to an 8-channel output matrix. These could provide ‘spot’ sound cues, triggered at will, or by the output of sensors, with the ability to control the apparent source location via a mixture of Ambisonics and level-based panning.

Control of eight analogue inputs, from CD players, microphones etc., is provided. A mono signal can be routed to individual matrix outputs, or encoded to B-Format with a software panner and mixed to the Ambisonic Decoder.

A Binaural encoder is included to enable listening on headphones when preparing material for the system when away from the PAMELA test laboratory.

A B-Format recorder is included to allow recording of the mixed Ambisonic signals passing through the system. This, for example, would allow archival recordings to be made of specific experiements, and time locked to a parallel visual recording.



This is the main operations page, from which others are accessed. The aim was to enable basic operation in as uncluttered a manner as possible, considering the number of operations being controlled. There are separate pages for each player, where preparation of 'cues' for each player is possible. There is a page for setting up the external inputs, and one for the decoder.

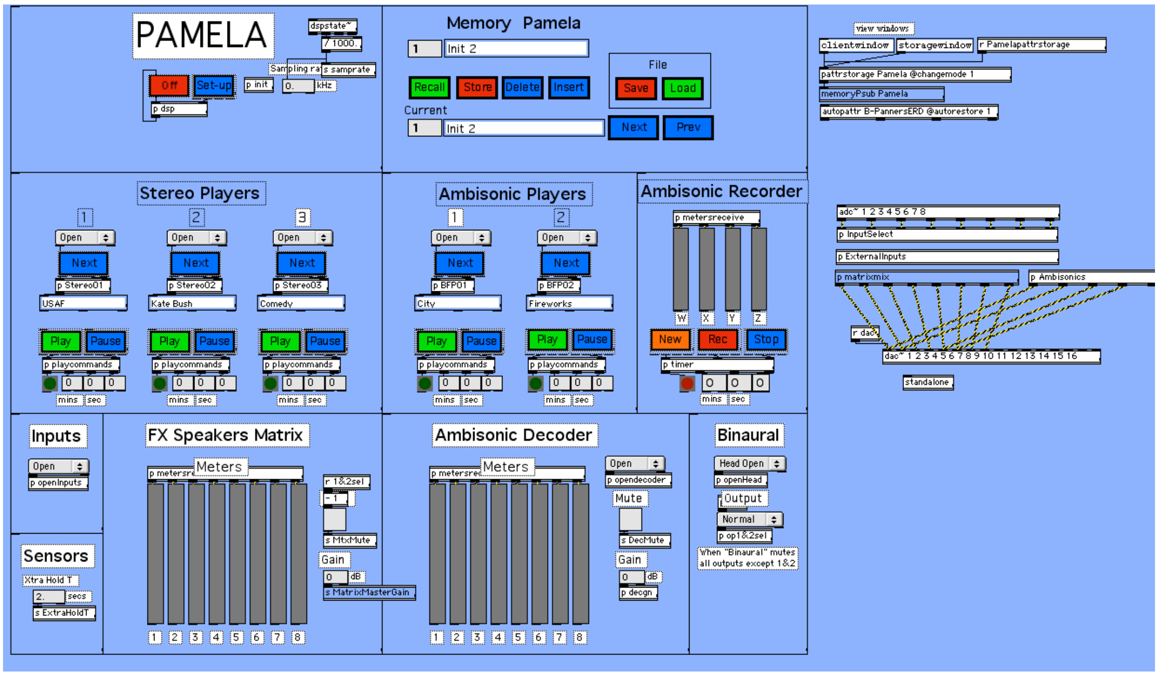
There are several independent Memories within the patch, each with their own Save/Load facility. The one on this page is for the PAMELA system, including the Ambisonic Decoder and Inputs. Memory Location 1 is recalled automatically when the programme is loaded. Each of the File Players has its own Memory. All Memories have the same basic functions: storing the variable control parameters at a numbered location, selecting and recalling those configurations, inserting and deleting a numbered location, and the ability to 'Save' and to 'Load' the entire contents of the Memory to disc.

Basic 'Play' and 'Pause' controls for the players are provided on the main page, plus a display of the name of the current cue and the elapsed or remaining time within it. Other Cues can be selected with the 'Next' button. A flashing green LED indicates that the player is 'playing'.

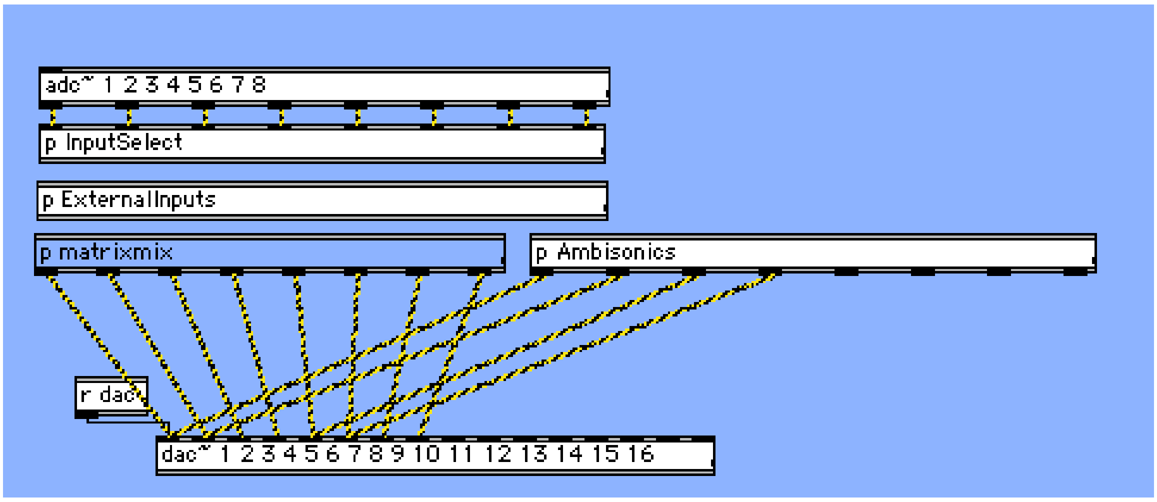
The 'Open/Close' menus open and close more detailed player windows, where programming each player is done.

All these players use 'interleaved' sound files. Stereo interleaved files are common and easily prepared. 4-channel Ambisonic files are not so common, and may have to be specially prepared using Cubase SX.

Shown below are some screen grabs of parts of the Max patch, demonstrating the basic appearance and operation of Max.

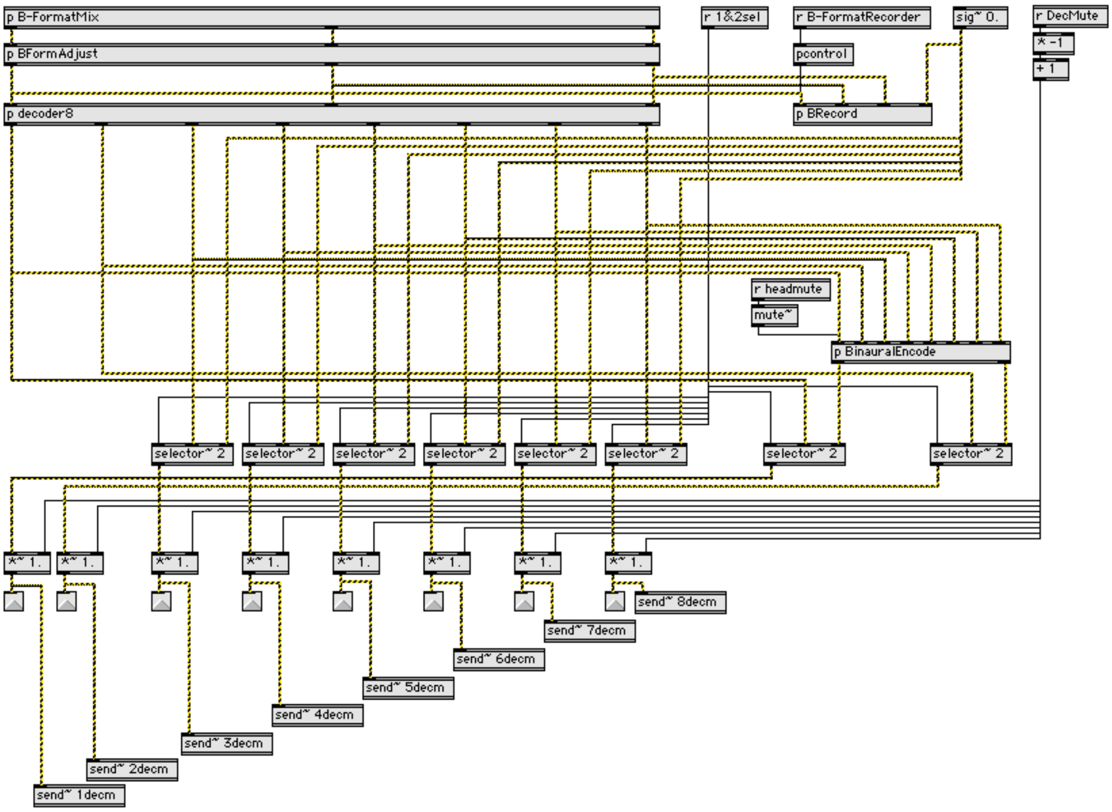


This is the Main Page in 'edit' mode, revealing several objects and their connecting 'patch-cords'. The "schematic" is hidden when the patch is put into 'operate' mode. Inputs to objects are the little black boxes at their top, and outputs are similar boxes at the bottom.

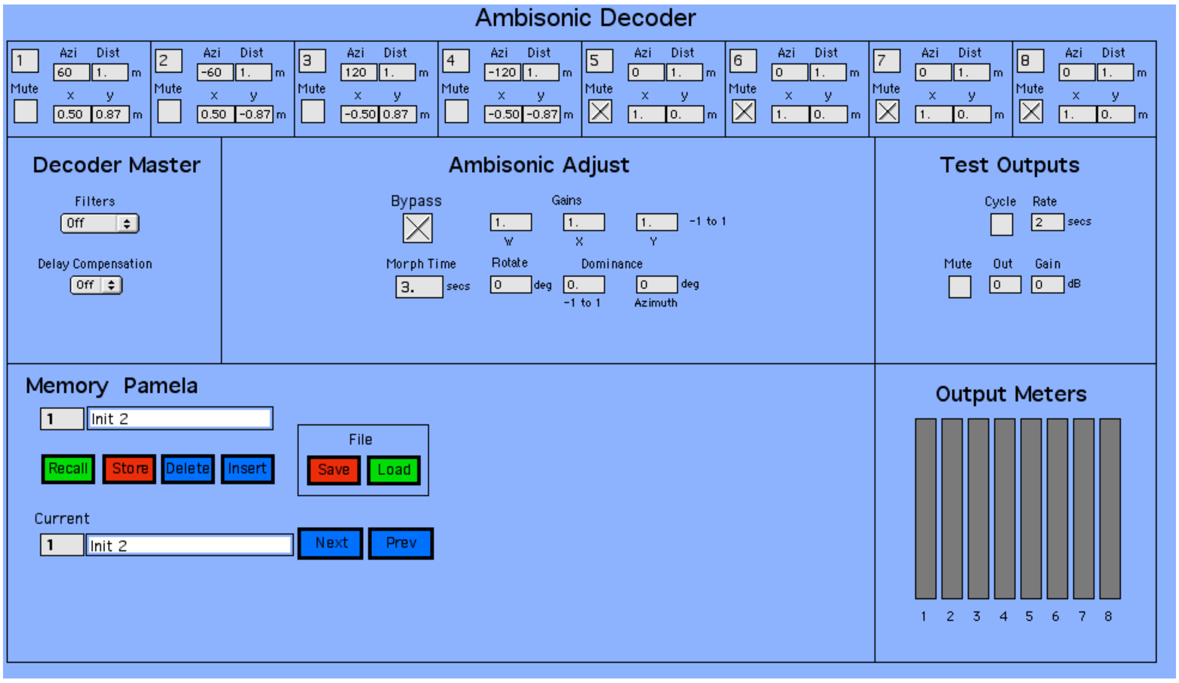


This is an expanded view of the main DSP objects on the right-hand side of the main patch, showing some of the main building blocks.

The object called 'adc~' is the 8-channel analogue to digital converter of the MOTU 828 audio interface. The object 'dac~' is the digital to analogue output of the same interface. The numbers after the name of the object specify the number of channels. The other objects, whose name starts with 'p', are sub-patches. These contain other Max objects, assembled into a working patch which is then encapsulated into an object with its own inputs and outputs. Yellow 'patch-cords' indicate that they are carrying audio signals. The symbol '~' indicates an audio processing object.



This is a screen grab of the contents of the sub-patch object called ‘p Ambisonics’. It in turn has sub-patches. The eight outputs of this patch, which are connected to the ‘dac~’ outputs in the parent patch, are the square objects with a triangle in them, above the eight ‘send~’ objects at the bottom of the patch. These latter send the audio signal to a similar ‘receive~’ object which drives the meters on the decoder and main pages, without using ‘patch-cords’. The objects whose names begin with ‘r’ are receiving control signals from elsewhere in the main patch. These control signals use black patch cords.



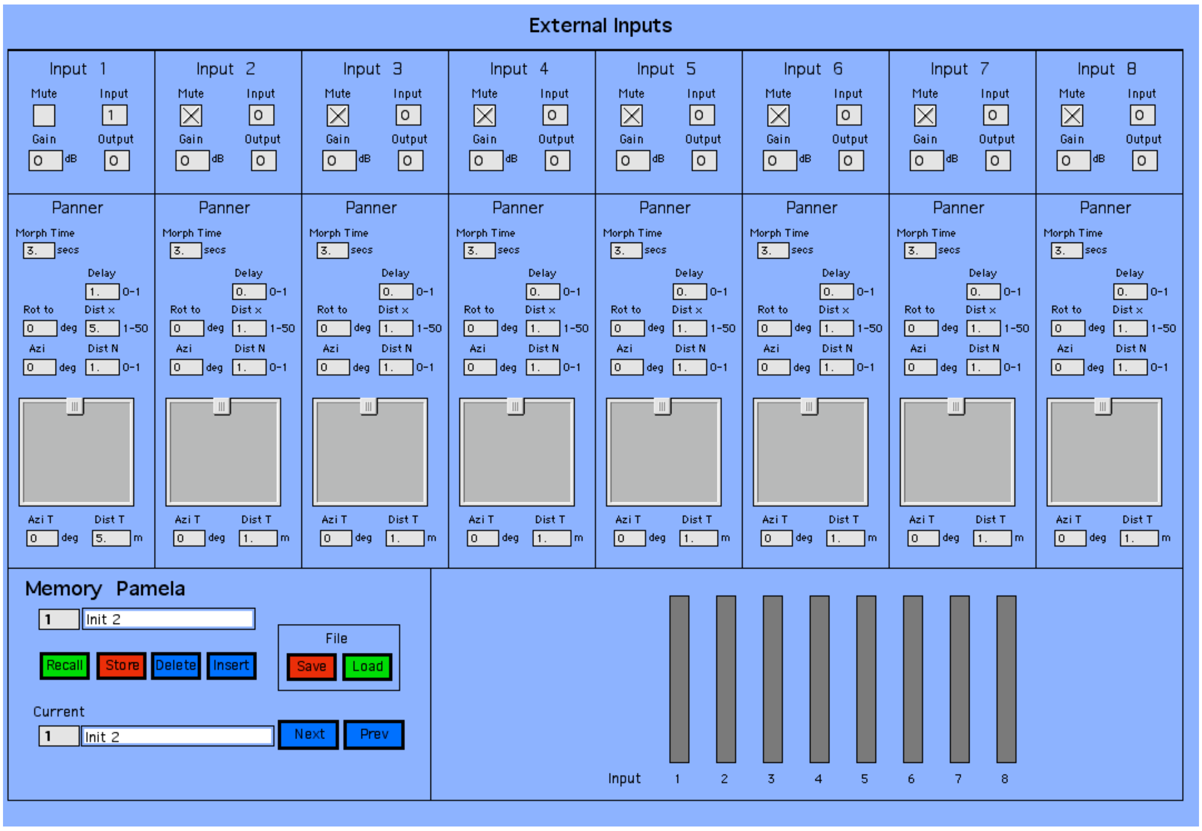
This is the Ambisonic Decoder Page. The decoder should be set up to match the layout of the loudspeakers. The position of each speaker is set up here.

The 'Filters' menu controls shelf filters in the decoder. The 'Off' option disables the filters. 'Large' and 'Small' are not precisely defined. Gerzon felt that filters were needed to match differences in the way direction is perceived at high and low frequencies, particularly when the speakers are close to the listener. This decoder is implemented when the 'Filters' menu is set to 'Small'. Gerzon's original decoding equations lead to loudspeaker signals that are out of phase with those of the opposite loudspeaker. This would mean that a listener far from the correct speaker would hear the sound as originating from the opposite, nearer, speaker, especially in layouts where the speakers are widely spaced, and large sections of the audience are outside the 'sweet-spot'. Richard Furse and David Malham suggested modifications to the equations, which ensure that the signals remain substantially 'in-phase', and that the sweet-spot is larger at the expense of some directional accuracy. This 'in-phase' decoder is implemented when the 'Filters' menu is set to 'Large', and is suitable for larger spaces.

The 'Delay Compensation' menu turns the automatic delay compensation for speakers that are not equidistant from the centre on or off.

The 'Ambisonic Adjust' Panel allows some manipulation of the mixed Ambisonic soundfield. The gains of the W, X and Y signals can be adjusted between –1 and +1 to enable changes in directionality and reversal of the width and depth image. The image can be rotated, and the 'Dominance' control allows a 'zooming' into the image at an Azimuth angle. Zero (0) is the normal state, 1 is maximum zoom, and –1 is maximum zoom in the opposite direction.

The Memory Panel is a duplicate of that on the Main Page, as this memory is for the main settings of the system, which will not change very often.



This page controls the eight analogue inputs. There are eight channels, each represented by a vertical strip. Each 'Panner' controls distance as well as direction. The output of each Panner is a B-Format Ambisonic signal.

The 'Input' number box chooses the source of the input to the channel.

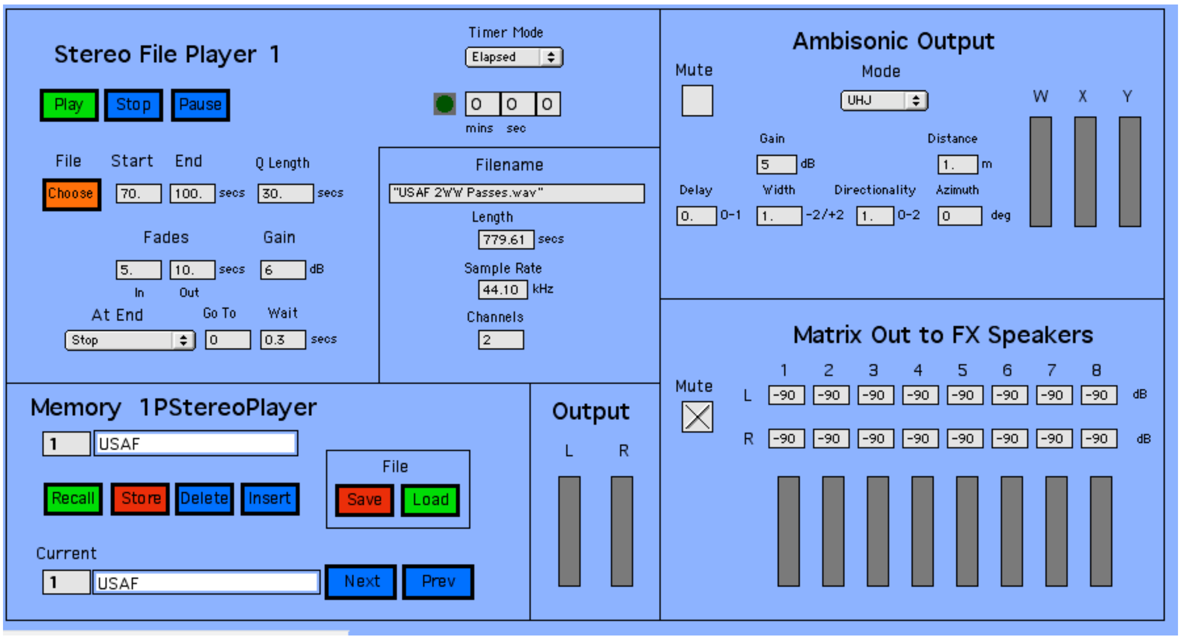
The 'Output' number box controls the output routing of the channel. Number values 1 to 8 route the channel directly to the relevant channels of the output matrix. When the number is zero (0), the output is routed to the Ambisonic Decoder through the 'Panning' control below.

The 'Azi' number box has a range of –360 to +360 degrees. Positive is in an anti-clockwise direction in line with Ambisonic convention. If settings are stored in a number of Memory Locations, along with a 'Morph Time', changes between one state and another will result in linear motion of the source between the positions over the morph time period.

The 'Rot to' number box is included to allow sources to travel in a circle around the centre, while maintaining the same distance from the centre. Values can be positive or negative. Positive values result in an anti-clockwise motion, negative values in a clockwise motion. The combination of the rotation angle and the 'Azi' setting is displayed in the 'Panning' display and the 'Azi T' box.

The 'Delay' box has a range of between 0 and 1. Sound sources that move radially towards or away from the centre listening position should display Doppler effect pitch variation. This has been implemented using a variably tapped delay line. The delay time is related to the simulated distance of the source, reflecting the finite speed of sound. At large loudspeaker distances, this delay will be added to the acoustic delay due to the distance of the speaker, and possibly adversely affect the timing of events. The Doppler pitch shift may also be disturbing for musical sounds, leading to 'out of tune' effects. Thus the delay is continuously variable between 0 and 1 for each Panner, 1 being the theoretically correct value.

The eight meters in the bottom panel show the level of the signal at the input. This level should be adjusted on the input level control of the audio interface.



Memory locations are used to store a 'cue'. The cue includes details of the file to be played, its 'in' and 'out' points, its playback level, what happens at the end of the cue, and its output routing.

The 'Play', 'Stop', and 'Pause' buttons control the transport of the player.

The File 'Choose' button opens a standard computer window where the file to be played can be chosen. By changing the numbers in 'Start' and 'End' boxes, the file can be 'top and tail' edited. The number in the 'Q length' box indicates the new length of the cue. The 'Fade' boxes below allow the cue to fade in and out.

The 'At End' menu chooses what occurs when the cue finishes. The options are Stop, Next, Next Wait Play, Go To, Go To Wait Play, Random, Random Wait Play, Loop and Wait Loop.

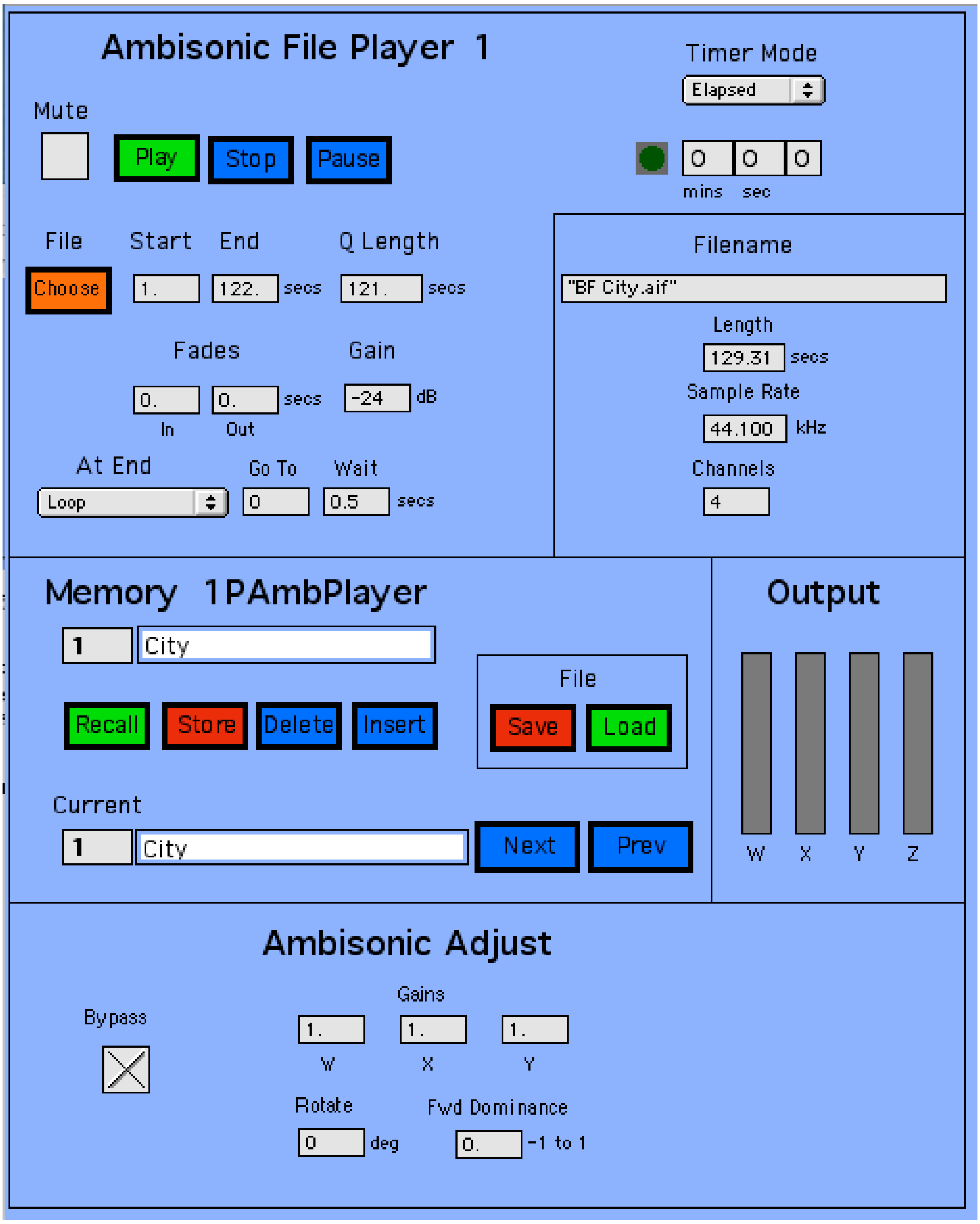
The output of the player can be sent to both the Ambisonic Decoder and the eight-output Matrix. There is an independent Mute button for each output to enable sending to only one of these options.

The 'Ambisonic Output' section includes a stereo-to-Ambisonic encoder. The 'Mode' menu chooses between various kinds of stereo input: Mono, Stereo, UHJ, MS or Binaural. Generally a Stereo or UHJ mode should be chosen. The UHJ mode extracts more spatial information than the Stereo mode, and can be used to decode Dolby Stereo or ProLogic material.

Binaural recordings are made with 'dummy heads' or microphones placed in or near the ear canals of the recordist. This is a common format, used by many hobby recordists. If this mode is chosen, the Distance of the source should be set to around 0.08, and the Width to 1, or slightly less..

'Directionality' controls the balance between the X and Y components and the W component.

The 'Azimuth' box controls the direction of the image in degrees. The 'Matrix Out to FX Speakers' panel controls the amount of each channel sent to each of the eight matrix outputs.



Two of these are provided. They can play any interleaved sound file of up to four channels, but the spatial results won't make much sense unless it is a Ambisonic B-Format encoded file. The Ambisonic Player can playback files recorded with the Ambisonic Recorder.

The general operation is identical to that of the Stereo File Player. The output is sent to the Ambisonic Decoder. There is no output to the FX Speaker Matrix.

The ‘Ambisonic Adjust’ Panel allows some manipulation of the Ambisonic signal.

6.3 Binaural

The ‘Output’ menu on the main page determines whether the output of the Ambisonic decoder is to be ‘Normal’, i.e. heard through the loudspeaker system or ‘Binaural’, heard on headphones. The ‘Binaural’ setting mutes all outputs except 1 and 2 (usually the headphone monitoring output), including the matrix outputs. On loading the programme, the ‘Normal’ condition is automatically selected.

Most binaural encoding convolves a sound with a set of Head-Related-Transfer-Function impulse responses. Computers can now do this in near real time on several streams of audio, although it is still processor intensive. HRTFs come in sets, to cover different source directions. Moving a source requires interpolation between two or more members of these sets. This increases the amount of processing. The use of virtual loudspeakers allows a simpler solution; binaural decoding the loudspeaker outputs of an Ambisonic encoder. These don’t move.

Max doesn’t do convolution very well. Readily available sets of HRTFs don’t seem to sound good enough for quality audio work. Instead a simple scheme based on a physical model of Interaural Level and Time Differences, and pinna and torso comb filtering was adopted, based on the work of Richard Duda at UPIC.[4] The model used is reasonably effective, not highly accurate, but as good as the convolution-based Spat~ from Ircam, the only one the author has been able to compare it to. It was included, as requested, to allow preparation work to be done without a loudspeaker system.

6.4 Sensors

Pedestrian triggered actuation of spot effects has been demonstrated using pressure pads and infra-red proximity detectors. Since PAMELA includes a camera-based 3D tracking system, it is hoped that triggering of spot effects can eventually be linked to that.

7 THE PLUG-INS

The plug-ins supplied for Cubase SX were all Pluggo type software plug-ins. These are built with Max/MSP “Pluggo Realtime”, a free downloadable programme. This must be installed on the computer to use the plug-ins. They use many of the same processes as the main Max patch, are similar in operation, and their control interfaces appear similar within the host application.

7.1 AmbDecoder4

A B-Format four speaker decoder, which should be inserted in a master four-channel output. Here the directions and distance of the speakers from the central listening channel can be entered.

An ‘Output’ menu determines the output mode of the decoder:

- ‘Mute’ mutes the output;
- ‘Decoder’ for output to the four loudspeakers;
- ‘Binaural’ for headphone listening;
- ‘B-Format’ for recording the mix to an interleaved four-channel B-Format signal.

7.2 AmbPanSB

This is a stereo-to-B-Format encoder, for insertion into channels containing two-channel 'stereo' material. The output is a B-Format Ambisonic signal.

There are the following options:

- Mono – the two channels are summed to mono;
- Stereo – the two channels are treated as two separate signals panned independently;
- UHJ – this contains an Ambisonic UHJ decoder. This format was used to encode Ambisonic surround material into two channels. It can also be used for Dolby Stereo encoded material, or to extract spatial attributes from normal stereo recordings;
- MS- the Mid-Side recording format widely used for film and video recording;
- Binaural – binaural recordings are obtained from a 'dummy head', or from a pair of 'in-ear' microphones.

The direction and width of the image can be altered. All parameters can be automated using Cubase's automation system.

7.3 AmbPanMB

This is a 'panner' for mono signals. It controls distance as well as direction. The output is a B-Format Ambisonic signal.

7.4 AmbB4mat

This is a four-channel plug-in that allows some manipulation of a B-Format recording

8 REFERENCES

1. Nice pictures of spherical harmonics can be found at: http://www.uniovi.es/qcg/harmonics/harmonics.html 24/10/06

2. Wiggins, B. (2004), An Investigation into the Real-time Manipulation and Control of Three-dimensional Sound Fields, PhD thesis, University of Derby, Derby. http://sparg.derby.ac.uk/SPARG/PDFs/BWPhDThesis.pdf 24/10/06
This is a good resume of surround sound and Ambisonics.

3. http://www.cycling74.com
Cycling 74 website. Can download software for free one month trial period. Numerous links to huge variety of applications and users.

4. C. P. Brown and R. O. Duda, "An efficient HRTF model for 3-D sound," in *WASPAA '97* (1997 IEEE ASSP Workshop on Applications of Signal Processing to Audio and Acoustics, Mohonk Mountain House, New Paltz, NY, Oct. 1997). http://interface.cipic.ucdavis.edu/PAPERS/Brown1997(Efficient3dHRTFModels).pdf 24/10/06
This describes the physical model used for the binaural encoding.