

## THE SPACIO-ACOUSTIC PROCESSOR

David G. Malham

Electronic Music Studio, York University.

### INTRODUCTION

The Spacio-Acoustic Processor came into being as a result of the demands of Dr. Trevor Wishart's electro-acoustic composition VOX 1. In this piece, one of the performance requirements is for four sources to perform a complex and at times very rapid 'dance' within the performance space which is defined by, typically, a square array of loudspeakers. It was felt better to do this live, rather than pre-record it in the studio and as this could only be done with the aid of computers, the present design was evolved.

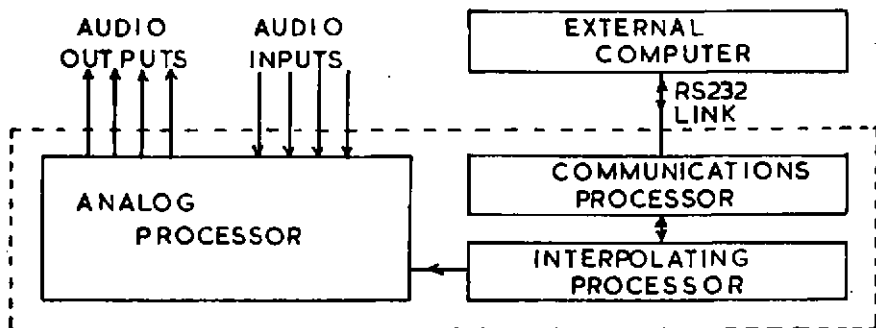
### THE AMBISONIC CROSS - BLENDER

The design was based on experience gained during the construction of the 'Ambisonic Cross-Blender' which has been described elsewhere [1] although it is not, in its current form, Ambisonic. The cross blender had four input channels, each of which fed to four separate multiplying DAC's which controlled the amount of signal that channel contributed to each of the four B - format output channels. For a four input, four output device, this resulted in a design which was very expensive to produce, since the DAC's were of the 14 bit plus sign variety. This was obviously inappropriate for a performance device which was intended for use on extended concert tours, so an alternative design approach was used.

### THE SPACIO-ACOUSTIC PROCESSOR

#### General System Configuration

Fig.1. System Block Diagram.



# Proceedings of The Institute of Acoustics

## THE SPACIO-ACOUSTIC PROCESSOR

In the diagram above it can be seen that the Spacio-Acoustic processor (or SAP 1) consists of two microprocessor units and an analog processor which does the actual controlling of sound distribution. An external computer, presently a Sinclair QL, acts as a controller. This runs a program, written by Trevor Wishart, which produces data on the start and end points of each move, the type of move (arc or line) and the time to be taken to complete the move. This data is sent over an RS 232 serial link to the actual SAP box where it is recieved by the communications microprocessor. This takes the raw end point and time information and from this it computes a set of positional increment data which is passed to the interpolating processor. This computes the moment to moment gain coefficients for the analog processor. That then varies the amount of each input channel sent to each of the four output channels so that the required moves are generated.

### Data Format.

Each move is coded into twelve bytes of data by the controlling computer and sent over the RS 232 link in the following format:

Table 1.

Byte No.	1	2	3	4
	See Table 2	X Coordinate	Y Coordinate	Z Coordinate
Byte No.	5	6	7	8
	Time		Volume	Acceleration
Byte No.	9	10	11	12
	Arc Centre X	Arc Centre Y	Arc Centre Z	Separator

The first byte is interpreted as an ASCII character which encodes the input channels involved and the type of move required as follows:

Table 2.

Input Channel	ASCII															
	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	
1	X	O	X	O	X	O	X	O	X	O	X	O	X	O	X	
2	O	X	X	O	O	X	X	O	O	X	X	O	O	X	X	
3	O	O	O	X	X	X	X	O	O	O	O	X	X	X	X	
4	O	O	O	O	O	O	O	X	X	X	X	X	X	X	X	

X Indicates that the current data is for that particular input channel.

If the character is lower case, the move is a straight line, if it is upper case it is a clockwise arc and a control code indicates an anticlockwise arc.

The X, Y and Z coordinates are the end points of each move, mapped on to a 255 x 255 x 255 cube.

The time data is a 16 bit word with a resolution of 10 ms.

Volume and acceleration are currently undefined.

The arc centre coordinates are mapped onto a 510 x 510 x 510 cube. This allows the arcs to start and finish on the surfaces of the projection space with no limitations as to whether they are convex

# Proceedings of The Institute of Acoustics

## THE SPACIO-ACOUSTIC PROCESSOR

or concave. Arcs are purposely limited to  $360^\circ$  or less in order to simplify the internal programs. If more than  $360^\circ$  arc length is required it must be split up into smaller sections by the program in the control computer before being sent to the SAP.

A considerable amount of pre processing of the data is done by the program in the control computer to optimise what is actually sent over the serial link, particularly in respect of maintaining a balance between the total number of moves for each channel. This is necessary, as will be explained later, in order to avoid unbalanced storage requirements in the internal buffers of the SAP. It also has to handle the very simple protocol used on the serial link. The data is sent in packets of 32 moves - 384 bytes in all. If the SAP communications processor sends an 'M' back after receipt of a packet, then another can be sent. If it sends an 'S' then its input buffer is full and no more should be sent until another 'M' is transmitted. If it sends an 'R' then it has been reset and expects a completely new set of data. The first 12 bytes sent from the control computer after receipt of an 'R' is a special case with the X,Y and Z coordinates of each channel being sent in turn. This defines a starting position from which the other moves flow.

### Communications Processor.

Both microprocessors in the SAP use a standard board developed for this project. They are a 16 bit design based on the TMS9995 microprocessor with 14K of RAM/EPROM memory, two 16 bit I/O ports and a serial port. All the programming is done in machine code. The communications processor utilises the serial port via an interrupt driven loop which takes the data from the RS232 and places it in a 2K buffer.

The data is removed from the buffer by the main program. Any moves involving multiple channels are separated out and each channel is dealt with individually. Different procedures are necessary depending on whether it is a straight or curving movement.

### Lines.

The end point coordinates of the previous move for the channel involved are subtracted from the end point coordinates for the current move giving the total displacement occurring on each axis. These are divided by the time the move should take to complete giving the displacement per unit time. This data is calculated to 32 bit accuracy and stored in that format together with the other information in the main buffer area for that channel.

### Curves.

These require more complex treatment. As before the beginning and end point coordinates for the move are taken but they are not used directly. Instead we derive the angular displacement for each of the two points from the centre point of the arc as defined by bytes 9, 10 and 11 of the move data. The angular displacement per unit time can then be calculated by dividing by the time data. This is stored in the channel main buffer, together with other data and a flag to indicate that it is angular rather than linear data.

## THE SPACIO - ACOUSTIC PROCESSOR

### Inter - Processor Link.

Data from each of the channel main buffer area is sent to the interpolating processor on receipt of a request for data for that channel from the interpolator. The physical link between the processors consists of two 16 bit busses (input and output) and some control and status lines. The link is completely asynchronous, thus reducing any tendency for either processor to interfere with the other's main program sequences.

### The Interpolating Processor.

The main job of the interpolating processor or interpolator is to generate the time dependent coefficients for the analog processor which controls the sounds. It maintains the position information in a small buffer area. An interrupt routine, driven by a 100Hz clock, updates this information using the displacement-per-unit-time data generated by the communications processor. At the end of each move, a new set of increments is picked up from a buffer in ram. The status of this buffer is monitored by a background program. If necessary, the background program requests more data from the communications processor. The interrupt program also generates the 16 coefficients - four for each input channel - needed by the analog processor to position the sound correctly. At the start of each interrupt cycle the current coefficient values are passed to a hardware circulating store.

### Analog Processor.

In order to reduce costs the analog processor is a multiplexed design which needs only four DAC's

FIG 2a

ANALOG PROCESSOR Block Diagram

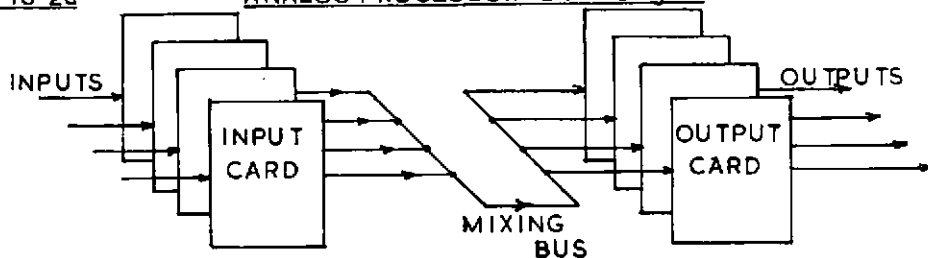
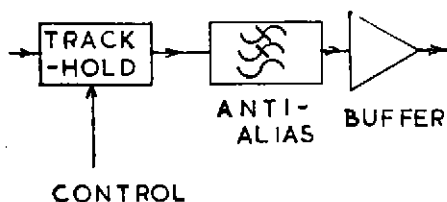
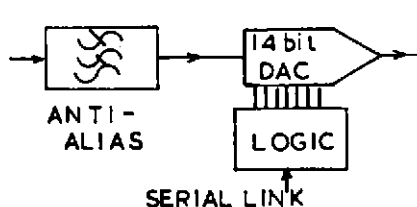


FIG 2b INPUT CARD

FIG 2c OUTPUT CARD



# Proceedings of The Institute of Acoustics

## THE SPACIO - ACOUSTIC PROCESSOR

As can be seen from fig 2 b, each input card has its own multiplying DAC which controls the amplitude of its contribution to a fast mixing bus. The data for this is sent from the interpolator's recirculating store via a synchronous serial link with a burst data rate of 8 Mb/sec. These coefficients are changed four times in each 20  $\mu$ s sample period, once for each output. Thus the mixing bus has on it each output sample in turn at a 200 KHz rate. The sample/hold circuits on the output channels are controlled so that they only pick off the appropriate sample.

### Hardware Design.

As far as possible, the digital circuiting is separated from the analog processor. Both microprocessor cards and the recirculating store are housed in one metal enclosure while the analog processor is enclosed in another. The DAC data links and the control signals for the output sample/holds are the only logic signals which pass between the two and these are well above audible frequencies so they cause few problems. As was indicated above, the microprocessor cards use the TMS 9995 microprocessor. This is a very fast - 12 MHz clock rate - 16 bit device with an 8 bit external data bus. It has a mini-computer like instruction set and a memory to memory architecture which makes context switching very easy. This feature is used to considerable effect, especially in the communications processor. Apart from the 9995's virtually all the logic devices used are CMOS types from either the 74 HC or 4000 series resulting in very low power consumption and hence, hopefully, higher reliability.

The analog processor is built mostly of TDA 1034 low noise IC's with CMOS 14 bit multiplying DAC's as the controlling elements. The anti-aliasing filters use low cost FM multiplex filter modules which are retuned to get a 1dB point of 20 KHz then going to better than 60dB's down after 25 KHz. Since the unit was intended to be used mostly in live concerts, balancing transformers were used on both inputs and outputs to maximise interference rejection and isolation.

### FUTURE DEVELOPMENTS

Currently only a horizontal pair wise panned version is fully implemented. However, only two simple modifications, one to hardware and the other to software, are needed to implement a full horizontal Ambisonic control scheme. We intend to do this immediately after the SAP returns from it's current tour with Electric Phoenix who are using it to perform VOX 1. Very shortly after this we shall go to full sphere operations with Ambisonic 'B' format output. This will require more extensive rewriting of the programs but as we will then have the capability to produce any non frequency dependent Ambisonic manipulation under full computer control we regard this as highly desirable.

### REFERENCES

# **Proceedings of The Institute of Acoustics**

## **THE SPACIO-ACOUSTIC PROCESSOR**

1 D.G. Malham "Digitally Programmable Soundfield Controller"  
Studio Sound Vol 26 No.2 p.75 (1984)