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DIGITAL SIGNAL PROCESSING IN ELECTROACOUSTIC MUSIC

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INTRODUCTION

In a previous paper[1] the author has described a real-time digital signal processor for audio applications. Since then practical experience has been gained in using the processor for the live performance of electroacoustic music. The present paper details some example applications and illustrates the use of the processor in several different contexts. The system has been designed to be flexible and expandable, and this paper illustrates the use of configurations ranging from a single processor to a moderately large multiprocessor system.

Since writing the previous paper a new version of the PHI processor module has been designed. This allows use of the faster version (160ns cycle time) of the TMS32010 digital signal processing chip which is now available, accommodates up to 2 Mbytes of parameter/delay memory (approximately 23 seconds of delay), and is compatible with the VME bus system. The faster processor executes 140 instructions per sample period, achieving an average increase in throughput of approximately 30 per cent.

EXAMPLE 1

ROGER SMALLEY - 'PULSES'

This work requires only the minimum PHI configuration shown in figure 1. (In fact this example was designed using the original version of PHI controlled by an 8 bit micro, a Research Machines 380Z). The score asks for two separate sound channels (premixed from five brass groups) to be independently ring modulated with sine waves controlled by two musicians. The signal flow is given in figure 2. Each channel uses twenty-six TMS32010 instruction periods, so even the slower version of the processor could accommodate four modulation channels if additional Analogue to Digital and Digital to Analogue converters were available. Alternatively the spare time could be used for some pre-processing of the input signal, compression or low-pass filtering for example.

The same signal transformation is used throughout the piece, so once the program has been downloaded the host system need only be concerned with passing sine generator frequency data and level data from the performers to the PHI module. Each performer has a control panel consisting of several key switches and fader controls. Each of these is assignable to any function by software control. In general, the Smalley work requires a combination of specific pitches indicated by staff notation and fluctuating pitch shapes indicated graphically. An example page from the score is shown in figure 3. The performer preselects the required starting pitch and uses one fader to control dynamics and another to create

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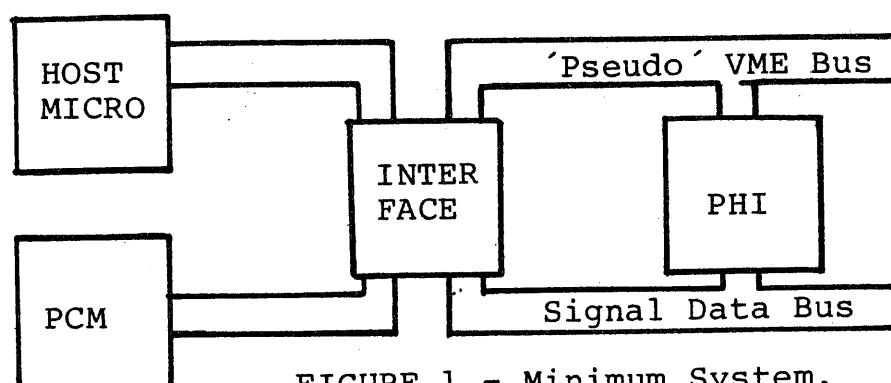


FIGURE 1 - Minimum System.

ADC1->)
GEN1->)RMOD1->DAC1

ADC2->)
GEN2->)RMOD2->DAC2

FIGURE 2 - Smalley Flow Diagram.

The image shows a sample page from a musical score titled 'Pulses' by Smalley. The score is written for a large ensemble, including strings, woodwinds, brass, and percussion. The notation is complex, featuring many notes, rests, and dynamic markings. At the top right, there is a box labeled 'GROUP 4 TRANS.' and a tempo marking '14/8'. To the right of the score, there is a circled number '23' and the text 'AMPMOD 2'. The score is divided into several systems, with the first system starting with a key signature of one sharp (F#) and a time signature of 14/8. The notation includes various musical symbols such as notes, rests, and dynamic markings like 'p' (piano) and 'f' (forte). At the bottom of the page, there is a small table with four columns and two rows. The first row contains the letters 'P+', 'P+', 'P+', and 'T+'. The second row contains the letters 'P+', 'P+', 'P+', and 'T+'. Below the table, there is a handwritten note 'Smalley 1971' and a signature 'K. P. 15'.

FIGURE 3 - Sample page from Smalley 'Pulses'.

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the indicated glissandi. The required pitches and fader ranges for each section have been prerecorded in software.

EXAMPLE 2

EDWIN ROXBURGH - 'AT THE STILL POINT OF THE TURNING WORLD....

This work was composed to use the complex tape delay system created by Barry Anderson for performances of Stockhausen's 'Solo' by the West Square Electronic Music Ensemble. A long multi-tapped delay is required with taps of 4.2, 8.75, 15, 20, 42 and 60 seconds. In addition, ring modulation and filtering are required at certain points in the score. The 60 second maximum delay requires at least three PHI boards. In fact this application (for a projected performance in 1987) has been designed for a four board system (given in figure 4) still under construction. All the functions used have, however, been tested individually.

The introduction of the VME processor allows much larger and more complex systems to be effected. This processor runs a fast real-time multi-tasking operating system called pSOS[2] and is able to handle all real-time controls in a system with several PHI modules. In very large systems the use of multiple VME processors is possible. The host microcomputer can deal with user interface, mass storage and non time critical functions.

The flow diagram given in figure 5 indicates the complexity of this application. However, the PHI system carries out all mixing and panning functions, distributing the sound over four loudspeakers as required. The only additional equipment required to perform this work are a microphone and preamp, power amplifiers and speakers. This is in marked contrast to the complexity of the conventional analogue setup usually needed.

The delay is split into three sections of 20, 22 and 18 seconds respectively over boards 2, 3 and 4. The other functions are carried out on board 1. The ring modulation function is identical to that used in the Smalley work, and the filter consists of two cascaded conventional second order bandpass sections. The algorithm used is that given by Chamberlin[3].

Perhaps the most interesting aspect of this application is inter board communication. Each PHI board has four input ports and four output ports onto the Signal Data Bus. As more inputs and/or outputs are required by each board, it is necessary to multiplex these ports. This is perfectly feasible provided the timing of writing, bus transfer and reading each channel is carefully controlled. Figure 6 shows the bus timing specification for the 'Still Point' program. The transfers are divided into two groups. The "Quad Bus" is conceptually equivalent to the mixing busses of a four group mixer. This bus uses all four output ports of all four modules and all four input ports of modules 2 to 4. All modules write to the outputs at approximately the same time and then busy themselves with other activities while the bus transfers occur, after which the input ports may be read and new data written to outputs. The other transfers use only ports 1 and 2 in

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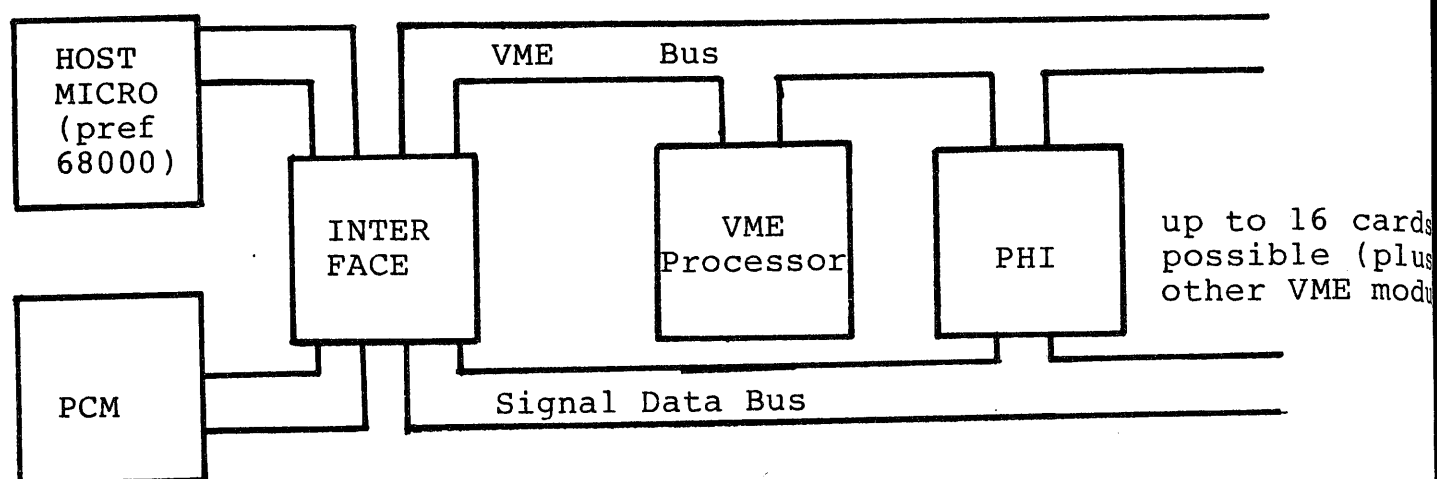


FIGURE 4 - Expandable VME-based system.

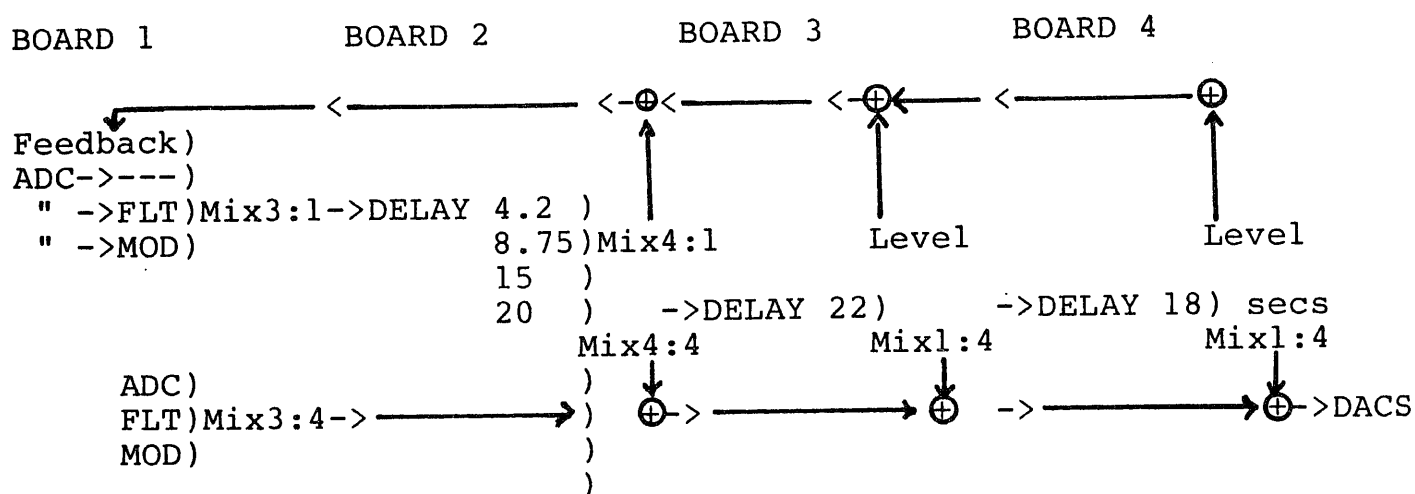


FIGURE 5 - Roxburgh Flow Diagram.

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each instance and are carried out similarly during the other half of the cycle. Port 1 links the delays and port 2 provides a feedback bus for repeated delays. This example was worked out by hand, but software to resolve bus conflicts automatically is under development.

The other interesting point about this application is the use of the VME processor as a control signal processor. A ramping function has been included that interprets data consisting of a target value and a ramp duration. The VME processor then generates the appropriate ramp. Figure 7 gives an example page from the score. The control functions for the levels of the six delays are shown graphically. These are prerecorded as a set of ramp functions, greatly reducing the quantity of stored data. Various other types of control processing are also possible.

Other functions (ring modulator and filter levels, generator and filter frequencies and overall delay level) are controlled by hand during the performance. The host also generates a timing reference for the oboist.

EXAMPLE 3

LAWRENCE CASSERLEY - 'META-L-IMITATIONS

The first two examples are instances of works conceived for older analogue systems that can be implemented by PHI. The advantages are, in general, that superior sound quality and a simpler, more reliable performance system can be achieved. The third example is different in that it was conceived with digital signal processing in mind, and at least some of the functions would be difficult, if not impossible, to achieve with conventional equipment.

'Meta-L-Imitations', for amplified metal and computer, is not a single piece, but a structured environment for improvisation. As such it requires that both the performer and the electronic transformations be able to modify their behaviour according to context. This necessitates a processor that can be reconfigured virtually instantly in an unpredictable sequence.

The high level structure of the work closely mirrors that of the PHI system (figure 4). The host microcomputer, a Microbox III running the Tripos III multi-tasking operating system, in interaction with the performer, selects the structure to be performed and instructs the VME processor which, if any, transformations are required and which control functions. The VME processor downloads the required program and deals with all timing requirements, while the PHI processor does the real-time sound transformations. It would be difficult to envisage an analogue system successfully handling such a structure.

Apart from the more conventional transformations, similar to those already discussed, this piece makes use of some particular possibilities of the PHI processor. One of these is the implementation of complex and dynamically varying delay

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BOARDS

1/1->INPUT PROCESSING
INPUT MIX->1/1
DELAY LINK->2/1,3/1
FEEDBACK LINK->4/2,3/2,2/2

2/1,3/1,4/1->DELAY
3/2,2/2,1/2->FEEDBACK MIX

OUTPUT MIX OUT->1/1,2/1,3/1,4/1
1/2,2/2,3/2,4/2
1/3,2/3,3/3,4/3
1/4,2/4,3/4,4/4

2/1,3/1,4/1
2/2,3/2,4/2
2/3,3/3,4/3
2/4,3/4,4/4->OUTPUT MIX IN

BUS TRANSFERS

ADC->1/1

1/1->2/1
2/1->3/1
3/1->4/1
4/2->3/2
3/2->2/2
2/2->1/2

1/1,2/1,3/1->2/1,3/1,4/1
1/2,2/2,3/2->2/2,3/2,4/2
1/3,2/3,3/3->2/3,3/3,4/3
1/4,2/4,3/4->2/4,3/4,4/4
4/1->DAC1
4/2->DAC2
4/3->DAC3
4/4->DAC4

FIGURE 6 - Table of Bus Links For Roxburgh.

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structures. It is possible to implement up to nine delay taps on each PHI module, and these can vary in delay time from one sample to 23 seconds. Alternatively several independent delays can be set up, and longer delays can be implemented as they were in example 2 above. More importantly all parameters (delay times, output levels, feedback levels) can be changed in a predictable way during performance.

Other transformation structures use arrays of filters and combinations of delays and pitch transposition, filtering and/or modulation in varying configurations that could only be achieved by using a large quantity of analogue processors with consequent difficulties of control and repatching.

EXAMPLE 4

LAWRENCE CASSERLEY - 'PANHARMONIC'

The final example is quite different in that it is not a signal processing function but a signal generating function. This work was composed to introduce concerts by the ensemble, 'Tube Sculpture'. This ensemble performs on a specially built, very large set of panpipes. The tuning of the pipes is based on an ancient scale of five equal intervals per octave. The lowest pipes are tuned to these intervals and the higher pipes are tuned to odd harmonics of the five fundamentals up to the nineteenth harmonic. The result is a very rich, but essentially harmonious, scale with a great variety of intervals. 'PanHarmonic' introduces the audience to the harmonic world of the panpipes. The PHI system is used to implement twenty-five generators, which are programmed to produce pitches from the set described above. Thus not only is the PHI system quite capable of performing sound generation as well as processing functions, but, unlike many currently available systems, the user is not restricted to any one tuning system.

Like several of my other works, 'PanHarmonic' is already running as the audience enters. At this stage the pitches and durations of the twenty-five generators are controlled by a set of probabilities which tend to make them move step-wise within one of the five harmonic series. At an appropriate moment the performer begins to intervene by controlling the various probability weightings from a set of faders. Thus the performer is in direct control not of the individual notes but of the higher level shape and direction of the piece, gradually having more and more influence on the progress of the music until, at the climax, he takes precise control of the shift to a new music that introduces the other performers. The precise logical structure of the control system is beyond the scope of this paper, but the example serves to illustrate another approach to using the PHI system.

CONCLUSION

The four examples given illustrate the flexibility of the PHI system. The author, while designing the system for his own use as

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a performing musician, has endeavoured to avoid making assumptions as to how the system will be used so that many different approaches may be accommodated. While one system can never be expected to cope with all possible situations, systems such as the one illustrated should be capable of dealing with a wide variety of different, and indeed totally unanticipated, demands. The four examples given show quite different uses of live performance electronics. Not only is the PHI system capable of meeting all these requirements with ease, but it would be perfectly feasible to programme all of them in one concert, something the author would prefer not to contemplate with analogue equipment.

References

- [1] L. Casserley, 'Software design for an 'economical' real-time digital audio signal processor', Proc.I.O.A. Vol17 Part 3 (1985).
- [2] Software Components Group, Inc, Santa Clara, Cal (1982).
- [3] H. Chamberlin, 'Musical Applications of Microprocessors', Hayden Book Co, New Jersey (1980).

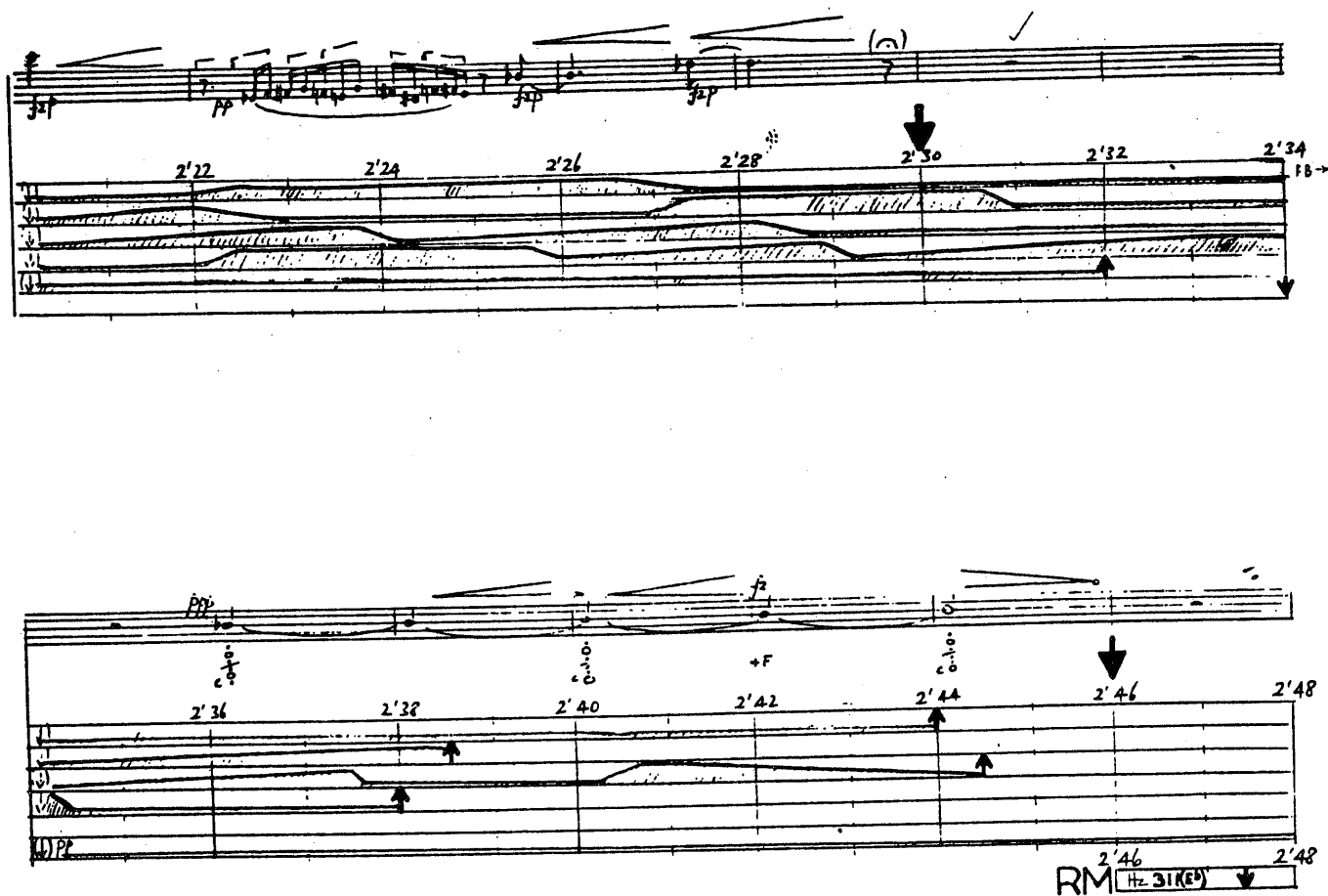


FIGURE 7 - Sample page from Roxburg 'At the Still Point of the Turning World....'