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A computer-based mixing and filtering system for digital sound signals

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Research Department, Engineering Division
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Summary

This Report describes the use of digital techniques to implement the processing found in conventional analogue mixing desks of the kind used for the 'mix-downs' of multi-channel audio recordings. The various problems of doing this were investigated individually using a small, high speed computer controlled by a stored program. It is shown that while the experimental equipment did not have sufficient 'power' to perform all the processing required, a factor of three improvement would permit all the functions in one channel of a fully equipped mixing desk to be performed digitally.

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1. Introduction

Digital techniques are being used more and more in the control and processing of sound signals. As may be expected their initial use is taking place in those areas where analogue techniques can no longer satisfy requirements, for example in providing high-quality time-delay or in low-distortion recording. Digital magnetic tape recorders are being developed,[1,2,3,4,5] and it is a logical step to provide digital processing, i.e. mixing etc. to accompany these recorders.

Until recently, digital systems of this nature were implemented using general-purpose integrated circuits. However, in 1971, with the introduction of the first microprocessor, a new generation of components became available making the use of small powerful computers a practical reality.

This Report describes the use of such a computer for performing audio processing such as that found in the sound mixing desks used with multi-channel tape-recorders, working with high accuracy digital sound signals in real-time. The control of the system is by a stored program, and thus different aspects of sound mixing can be investigated by changing the program. This versatility permits the same "hardware" to carry out many different processes and this has been used to great effect to study sound companding systems,[6] error concealment methods and digital test-signal generation.

2. Conventional Analogue Mixers

Existing analogue mixers are highly developed and offer a large variety of facilities to the operator. These can be summarised under the headings; level control, spectrum shaping, monitoring, switching, control of remote reverberation devices, mute and solo systems, communications, line-up systems, level metering, channel-status indication and automation.

In general, the mixing techniques used in broadcasting differ from those used by the gramophone-record companies. In the latter, it is commercially sensible to apply a great deal of effort at the mixing stage, and so the method used is first to make a multi-channel recording using relatively simple control techniques and a very large number of microphones, adjusting the level etc. of each channel so as to record the largest possible signal on the tape. In a second operation using the replayed outputs from the multi-channel recorder, the operator attempts to achieve the required artistic effect in mono, stereo, or quad. This may take a large amount of time and various automation techniques have been designed to help at this stage.[7,8] For broadcasting it is more usual, at present, to mix directly from the multiple sources to a mono, stereo or quad output in a single operation.

It is the second mixing operation of the form used by record companies which readily lends itself to initial investigations in the use of digital methods. A digital multi-channel recorder has been built by the BBC[1], which provides digital sound data that can be processed in digital form. It was clearly not desirable to build an entire mixing desk for these investigations and so a flexible system has been designed in which a small number of control functions can be used in a number of separate experiments.

3. Processing under program control

3.1 Processing requirements

The processing facilities required in experimental digital mixing equipment are little different to those provided by most computers, i.e.

(a) the ability to carry out sequences of operations, e.g. arithmetic, logical and data-transfer sequences

and (b) the ability to choose between alternative sequences of operations at specified points, e.g. conditional tests, jumps and sub-routines.

Most computers comprise three basic parts – the central processing unit (CPU) with its associated control unit, a store containing the data and program to be executed, and input/output (I/O) devices. In the present application, the inputs are sound signals sampled at 32 kHz and digitally coded to 13 or 14 bit accuracy; thus the computer must be capable of handling words of this length. The computer performs operations serially according to the stored program and, for a real-time system, this program must execute many operations in the interval between sound samples. If certain operations require a program which extends over more than one sample interval, then special arrangements must be made to ensure that a digital word is output per sample period or the audio output will be muted. By its nature, digital sound data is continuous and there are no intervals in which 'housekeeping' can be performed and so the computer must work at high speed with powerful instructions if a useful amount of processing is to be achieved.

3.2 Choice of processor

At the required level of performance there are three approaches:

1. To construct purpose-built processors using discrete high-speed logic. This would involve a great deal of development time for both the hardware and software, but could potentially meet the requirements.

2. To construct a machine based on available micro-programmable chips or 'bit-slices'. These are vertical slices through the arithmetic of a CPU in a computer and can be assembled to form a computer of any desired word length. This is a very attractive solution as it relies mainly on the use of highly integrated devices, yet still offers the versatility of an instruction set tailored to the application, and it can combine high speed with powerful instructions.

3. To use a commercially available high-speed computer. Until recently this option would have been economically prohibitive but machines are now available whose cost is low enough to enable a valid study to be made and yet is too high for them to be used in large numbers. With such a machine the instruction set is fixed; however, one can be bought with a number of the required facilities already supplied, such as the means to load and edit programs and a range of software to aid the writing and development of programs.

The last option was clearly the most expedient solution for a Research Study and, accordingly, a feasibility study was carried out on a Plessey 'Miproc' processor.* It was decided to write, assemble and simulate a test program to see if a useful amount of processing could be done in the time available between sound samples (31.25μs).

The machine has a set of 180 instructions, most of which can be executed in a single clock cycle of 350 ns, and so a sequence of 89 instructions can be executed. The test program accepted seven input sound-signals, multiplied each by a coefficient, added four of the results in pairs to form two output signals, a third output being formed from the three remaining results. Part of each input signal contained a label to indicate to which output it should be sent.

This program was assembled using a cross-assembler program available on the 'Tymshare' time-sharing computer network. The test program was verified using a simulator which indicated that the required processing could be achieved using 86 instructions. This was thought sufficiently encouraging to proceed with the 'Miproc' processor. Assembly and simulation are explained further in Section 3.4.

3.3 Instruction set

Each instruction defines an operation to be performed during one cycle or group of cycles of the processor clock. Instructions are stored as a code word of 16 bits consisting of an 8 bit operation code an an 8 bit argument, when applicable. The processor decodes the operation code to determine the function and uses the argument as data or address information during execution.

The instruction set contains the following basic groups of operations.

(1) MOVE DATA instructions: transfer data between working registers, and between working registers and memory.

(2) ACCUMULATOR instructions: include logical, arithmetic and control (e.g. skip if accumulator zero) instructions.

(3) JUMP instructions: permits an exit from the normal program sequence, e.g. to a non-consecutive program statement or sub-routine.

(4) INPUT/OUTPUT instructions: allows data to be input and output from the accumulator.

Most instructions are executed in one clock cycle but some, e.g. jump to subroutine instructions, are multiple clock operations. This can be detrimental when program execution times need to be kept short, and can often be avoided by careful programming.

3.4 Programming methods

Computer programs are usually developed by means of flow charts, etc., which can be translated into a series of instructions. These then have to be expressed using the specific instruction set of the computer and with consideration for the hardware characteristics of the processor. Factors such as the number of working registers, methods of addressing memory and input and output arrangements have to be defined before this can be done. These operations are simplified by using mnemonics with symbols and labels to represent each instruction.

The next stage is to convert the mnemonic language, (called the source code) into the binary, or equivalent, machine instructions (called the object code) which can be executed by the processor. For simple programs this can be done manually by looking up the code or codes corresponding to each mnemonic. However, this requires the assignment of memory space to constants and intermediate results, link addresses for subroutines, etc., and is liable to human error. These processes can be done automatically using a cross-assembler, which is itself a program that translates the source code into the object code. The user can specify memory locations symbolically and use labels to mark program statements; syntax errors or logical inconsistencies will be detected by the cross-assembler program. The output of the cross-assembler is a paper tape which contains the object code in a suitable form for the processor, and also a listing of the input text indicating memory locations of instructions and data.

At this point the program can either be loaded into the processor and tested directly or a 'simulator' can be used. This is a program which simulates the execution by a processor of the object code generated by the cross-assembler and can provide details of the states of the processor during the execution of a program. A simulator is most useful when a processor is not available for the initial testing of programs, but for many applications, especially real-time ones, the processor must be used for the final test.

*This feasibility study was done by F.A. Bellis.

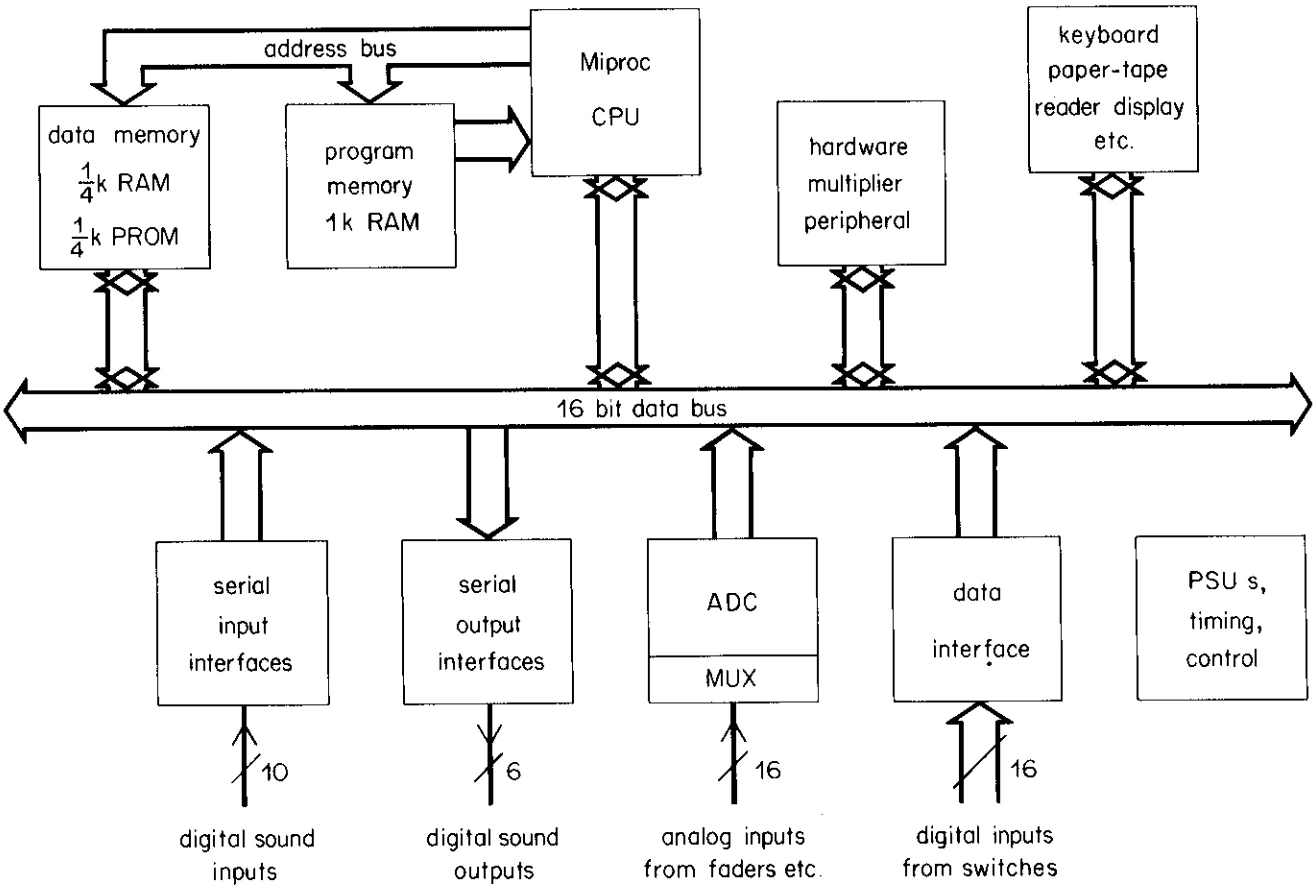

Fig. 1 - The Miproc processor and its interfaces

4. The mixing system

4.1 Number representation

The machine uses 16-bit fixed-point arithmetic, all arithmetic operations within the processor are carried out in two's complement notation, and so all numbers to be processed must be converted to this form if not already in it. For multiplication of numbers it is convenient to assume that all numbers are integers or fractions. In all multiplications the $2b$ product of two b bit numbers is rounded or truncated to a b bit result. This is easier using fractional arithmetic since overflow cannot occur. Additions of fixed point fractions can produce overflow unless precautions are taken to maintain spare bits in the more significant positions. The sound samples are linearly coded with 13 bits (though more can be accommodated) and thus the format used to represent the value of a sound sample is

15																0
S	S	S	.	X	X	X	X	X	X	X	X	X	X	X	X	X

$$\frac{1}{2} \leqslant \text{sample} \leqslant \frac{1}{2} - 2^{-13}$$

where S is the sign of the binary number in two's complement form. Coefficients representing fader settings, etc., are always positive and are linearly coded to 12 bit accuracy (giving control over a dynamic range of 72dB). The format is

15													3			0
0	.	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	Y	0	0	0

$$0 \leqslant \text{coeff.} \leqslant 1 - 2^{-12}$$

4.2 Description of equipment

The mixer system is shown in Fig. 1. The Miproc CPU has a program memory of 1k words (16 bits per word) RAM* and a data memory of ¼ kwords RAM and ¼ kwords PROM*. A paper-tape reader allows program and data to be entered quickly into the RAM stores.

A hexadecimal keyboard allows data to be entered and loaded into registers or memory (for example to modify a program already in store or to give new data to a program already running). A four digit hexadecimal display monitors the contents of registers or memories according to the settings of selector switches.

A high speed multiply instruction is provided by using a hardware multiplier as a peripheral. This reduces the execution time from 34 clock cycles for the software version to 3 clock cycles (1150ns), and is done by outputting first the multiplicand, then the multiplier, and

*RAM = Random Access Memory

*PROM = Programmable Read Only Memory.

then inputting the result to the CPU.

Interfaces have been added to permit the processing of 13- or 14-bit PCM sound signals. Ten serial inputs are provided with a data rate of 448 $kbit.s^{-1}$ and six serial outputs are provided with the same format. Analogue inputs are digitised to 12-bit accuracy at a sampling rate selectable between 1 Hz and 500 Hz, and by multiplexing the analogue inputs 16 inputs can be interfaced through a single converter. Finally, switch settings can be connected to the processor by formatting them into 16 bit words and 16 of these words are interfaced with the system via another unit. Unlike many processor systems in which as much 'housekeeping' as possible is kept under software control, all interfaces have been designed to incur the minimum software load.

The equipment is housed in a purpose-built console incorporating high quality faders and switching. Because the entire system is program controlled, the layout of the switches is not critical, each switch assuming the role for which it is programmed. However the layout does follow the general pattern of existing mixing desks to assist its use. A photograph of the equipment is shown in Fig. 2.

Twelve channels are equipped with faders, and two of them may be used as Input/Output modules with full facilities. A third module is partly equipped and can be used as an I/O module or grouping unit and the fourth is designed to be used as a master control unit. However it must be emphasised that the organisation and functions of the controls is entirely determined by the stored program.

5. Functional and design aspects of digital mixing

5.1 Effect of digital processing on mixing performance

Addition and multiplication are the main arithmetic processes needed for the mixing of digital sound signals. Errors can occur in these processes in two ways - (1) quantisation errors due to the rounding or truncation of results, and (2) limitation of dynamic range due to the use of fixed-point arithmetic and finite word-length.

These problems are reduced by using the number representation described in Section 4.1, but they are not eliminated. The rounding operation applied to a product term can introduce an error of up to ± ½ a least significant bit (LSB), which may, according to the statistics of the error, impair the signal-to-noise ratio by up to 3dB. The addition of two or more digital words may require extra bits in the result to avoid overflow and thus a choice must be made whether to provide this 'headroom' or to perform further multiplications to place the result within the original word length with its attendant error.

The restriction of dynamic range is also found in conventional analogue mixing desks, where the level of signals within the equipment is controlled by an operator by means of faders. This would be the perferred method for a digital desk also, since the maximum advantage can thereby be taken of the available dynamic range. The alternative solution, the provision of sufficient 'headroom' in the equipment, would require at least six extra bits in word-length; also, floating-point arithmetic cannot be used

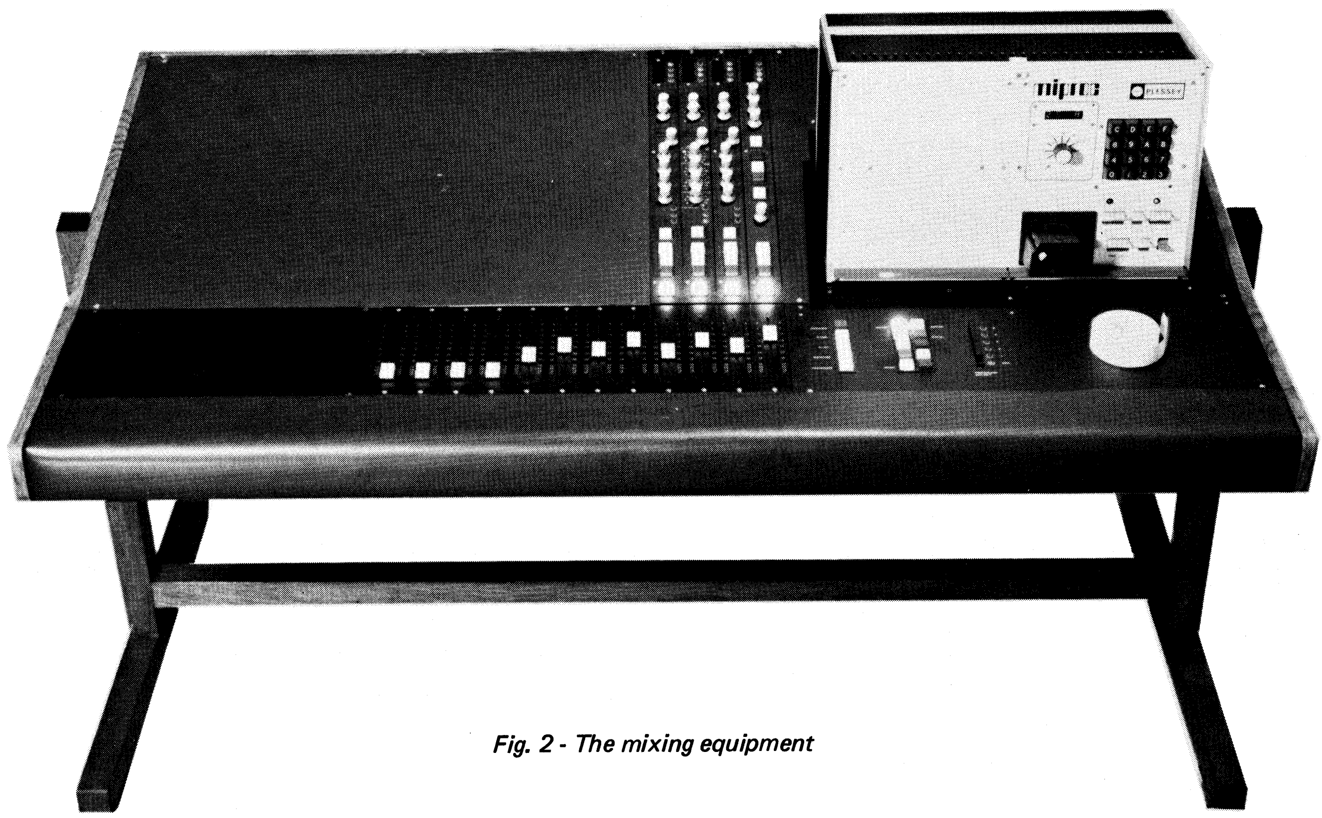

Fig. 2 - The mixing equipment

effectively at the very high processing speeds involved. here. Accordingly, the use of 16 bit arithmetic for handling 13 or 14 bit digital sound signals was found to be a satisfactory compromise for these initial investigations.

A program, similar to the one used in the feasibility study outlined in Section 2.2. was run on the equipment to test its mixing performance. This mixed five inputs to a stereo pair of outputs, and a 'master' fader was used to restore the output signal range, when necessary, to the input range. The program used 85 instructions. No undesirable effects were heard.

5.2 Spectrum shaping

A mixing desk in which all signal processing is achieved digitally would still have to provide the facilities found in conventional analogue desks. It has become standard practice to supply a large number of different 'equalisation' or spectrum-shaping options, usually with separate control over low- and high-frequency shelving characteristics and a mid-band peaking ('presence') response. The frequencies and degree of boost or cut of these responses can usually be varied either in steps or continuously. Such a specification can be substantially met by a digital filter in which the chosen characteristic is determined by a unique set of coefficients.

5.2.1 Filter design theory

Many of the design techniques used for analogue filters can be applied to digital filters. A useful method of characterising the filtering action of a system is by means of a transfer function which may be expressed as the Laplace transform of the impulse response.[9]

$$H(s) = \int_{o}^{\infty} h(t)e^{-st}dt \qquad (1)$$

The magnitude and phase of the transfer function with $s = j\omega$ are the magnitude of the steady state gain and phase shift of the system at frequency ω and thus $H(j\omega)$ is the frequency characteristic of the system.

It is convenient to represent a transfer function $H(s)$ in terms of the roots of the numerator and denominator equations giving zeros and poles respectively. The low, mid and high frequency characteristics described above can be realised by an analogue filter having a transfer function containing two poles and two zeros arranged as complex conjugates.

$$H(s) = \frac{(s - j\omega_0 e^{j\theta_1})(s + j\omega_0 e^{-j\theta_1})}{(s - j\omega_0 e^{j\theta_2})(s + j\omega_0 e^{-j\theta_2})}$$

$$= \frac{s^2 + 2\omega_o \sin\theta_1 s + \omega_o^2}{s^2 + 2\omega_o \sin\theta_2 s + \omega_o^2} \qquad (2)$$

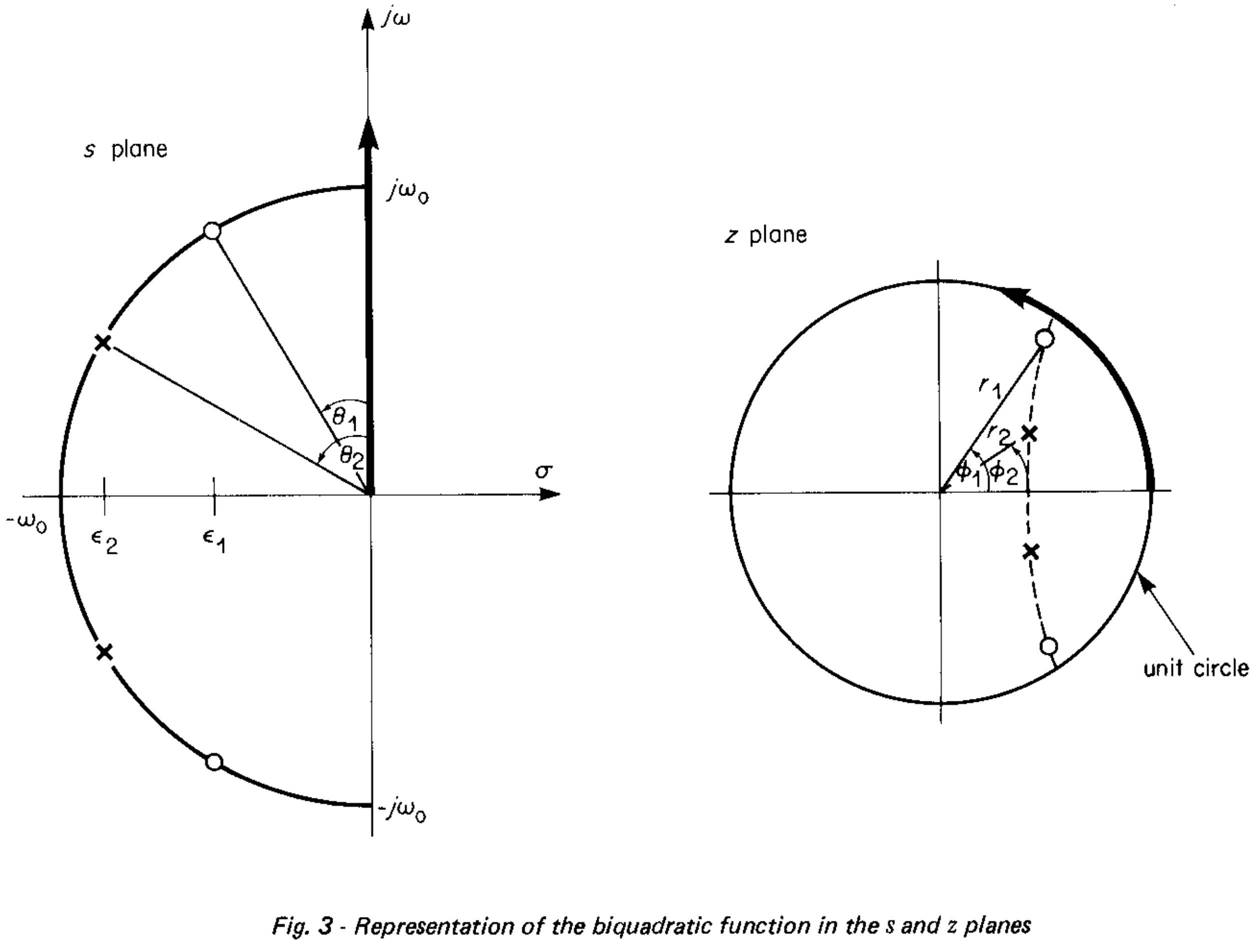

Fig. 3 - Representation of the biquadratic function in the s and z planes

This biquadratic function has its poles and zeros distributed in the complex frequency plane (s plane) on the circumference of a circle, radius ω_o at locations determined by θ_1, θ_2 as shown in Fig. 3.

Just as the Laplace transform simplifies the analysis of continuous (analogue) systems so the design of discrete sampled systems is simplified by the z transform. Thus a new transfer function can be expressed as the z transform of the impulse response.[10]

$$H'(z) = \sum_{n=o}^{\infty} h(nT)z^{-n} \tag{3}$$

where T is the sample period and the transformation from the continuous or 's' plane to the discrete or 'z' plane is obtained by the transformation.

$$z^{-1} = e^{-sT} \tag{4}$$

Using this transformation the positions of the poles and zeros in the 's' plane can be mapped into the z plane and a new transfer function constructed (see Fig. 3)

$$H'(z) = \frac{1 - 2r_1 \cos\phi_1 z^{-1} + r_1{}^2 z^{-2}}{1 - 2r_2 \cos\phi_2 z^{-1} + r_2{}^2 z^{-2}} \tag{5}$$

where r_1, ϕ_1 and r_2, ϕ_2 represent the positions of the zeros and poles in the z plane using polar co-ordinates and z^{-1} is the z transform operator which can be shown to correspond to a delay in the discrete-time sequence. For this reason it is often called the delay operator.

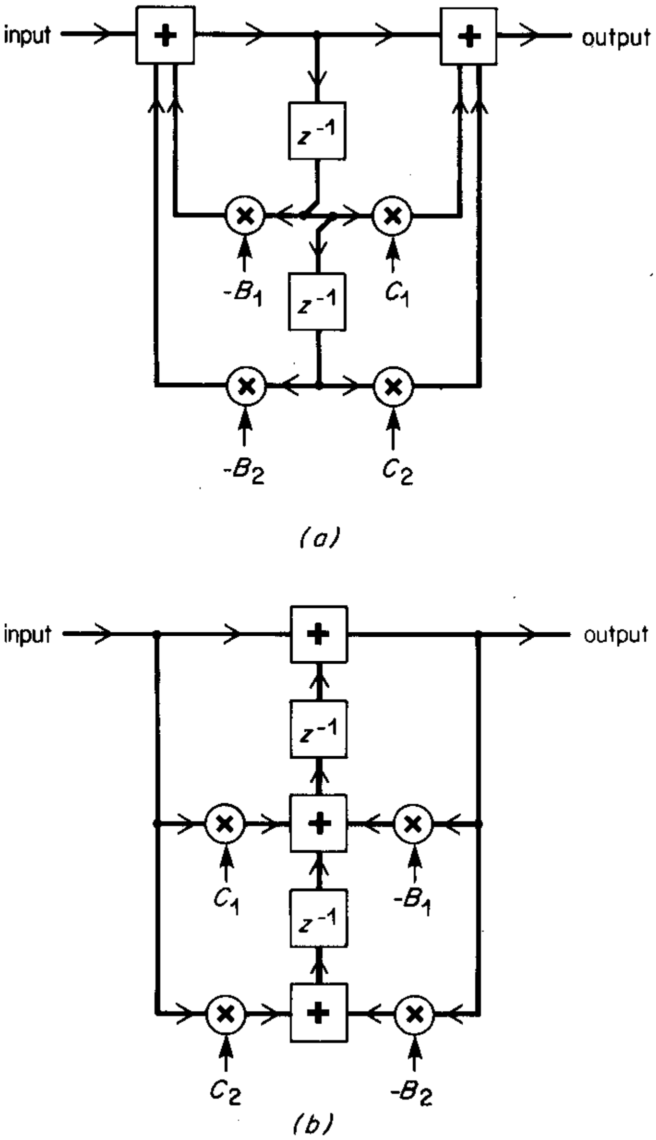

Fig. 4 - Equivalent representations of a biquadratic section digital filter
(a) canonic representation
(b) transpose system

This transfer function can now be directly implemented using adders, multipliers and delay elements. Fig. 4a shows the simplest (canonic) representation of $H'(z)$ known as the direct form,

where the coefficients are $C_1 = -2r_1 \cos\phi_1$
$C_2 = r_1{}^2$
$B_1 = -2r_2 \cos\phi_2$
and $B_2 = r_2{}^2$

C_1, C_2, B_1, B_2 can be changed to produce different filter characteristics.

5.2.2 Frequency characteristic of the filter

For digital filters the input and output signals are sequences of numbers and so a frequency characteristic must be viewed as the relationship between the continuous versions of the sampled sinusoidal input and output. Therefore the problem, here, is to relate the transfer function $H'(z)$ to the desired characteristic in a continuous system.

The frequency characteristic of a continuous system is obtained by evaluating the transfer function $H(s)$ on the $j\omega$ axis. The same procedure can be carried out in the z plane where the $j\omega$ axis maps into the unit circle $\exp(-j\omega T)$. The frequency characteristic of the digital filter is therefore periodic at intervals of sampling frequency. If the digital filter is derived by mapping the poles and zeros of the required continuous filter then the repeating poles and zeros implied by Equation (4) will affect this required characteristic as shown in Fig. 5.

Using the transformations $z = e^{j\omega T}$ and $z^{-1} = e^{-j\omega T}$ and expressing the transfer function in its real and imaginary components on the unit circle.

$$H'(z) = H'(e^{j\omega T}) = A(\omega) + jB(\omega)$$

$$H'(z^{-1}) = H'(e^{-j\omega T}) = A(\omega) - jB(\omega)$$

and

$$H'(z)H'(z^{-1}) = A^2(\omega) + B^2(\omega)$$
$$= (\text{amplitude characteristic})^2$$
$$= |H'(e^{j\omega T})|^2$$

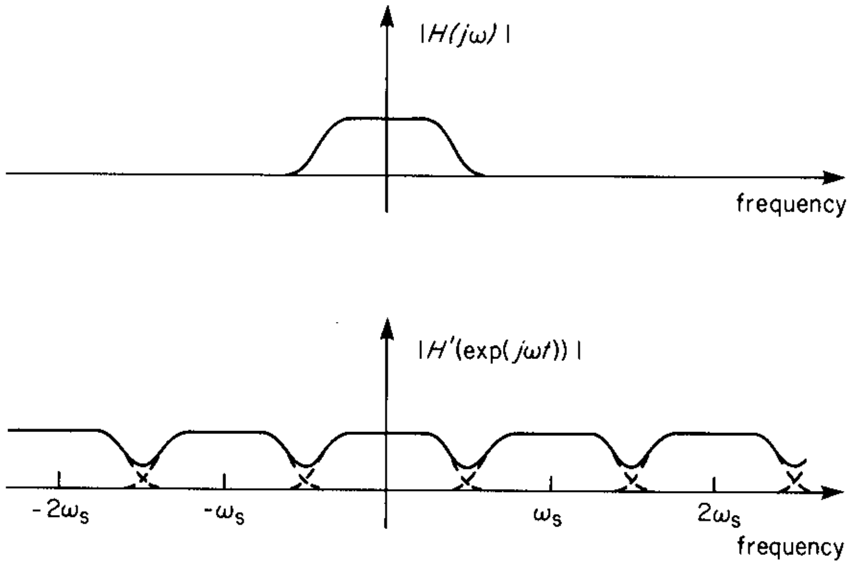

Fig. 5.
Comparison of frequency characteristics of a bandlimited filter and its digital equivalent derived using the z transform.

The corresponding phase characteristic is given by

$$\phi'(\omega) = \tan^{-1}\frac{B(\omega)}{A(\omega)} = \tan^{-1}\left[-j\frac{H(z^{-1})-H(z)}{H(z^{-1})+H(z)}\right] \quad (7)$$

For the biquadratic section under consideration, the amplitude characteristic is given by

$$\left|H'(e^{j\omega T})\right|^2 = \frac{1+C_1^{\,2}+C_2^{\,2}+2C_1(1+C_2)\cos\omega T+2C_2\cos 2\omega T}{1+B_1^{\,2}+B_2^{\,2}+2B_1(1+B_2)\cos\omega T+2B_2\cos 2\omega T} \quad (8)$$

5.2.3 Errors in the digital filter

There are three main types of error associated with digital filters:

(i) Quantisation of the coefficients - analogous to a component in an analogue filter which does not have exactly the required value. This will alter the pole and zero positions and thereby alter the response from that of the ideal filter.

(ii) Quantisation of the products and sums within the the filter - round-off errors may be introduced at various nodes of the filter and if overflow can occur due to the limited dynamic range, this may result in severe distortion and oscillations. Certain filter configurations are less sensitive to these errors than others with the same transfer function. A filter derived by the transpose system[11] is the least sensitive and can be simply generated from the canonic representation, according to a set of rules repeated from the reference in the Appendix, Section 9.4. Using these rules the filter structure of Fig. 4b can be derived.

(iii) Aliasing - the amplitude characteristic of a digital filter is repeated at intervals of sampling frequency and if the continuous amplitude characteristic falls off slowly in the region of half sampling frequency,

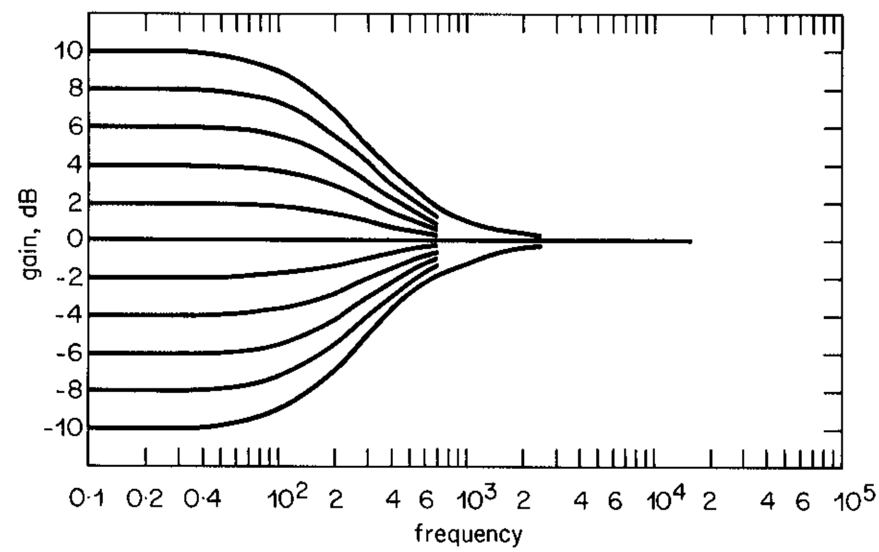

Fig. 6.
Family of low frequency shelving filters (100 Hz)

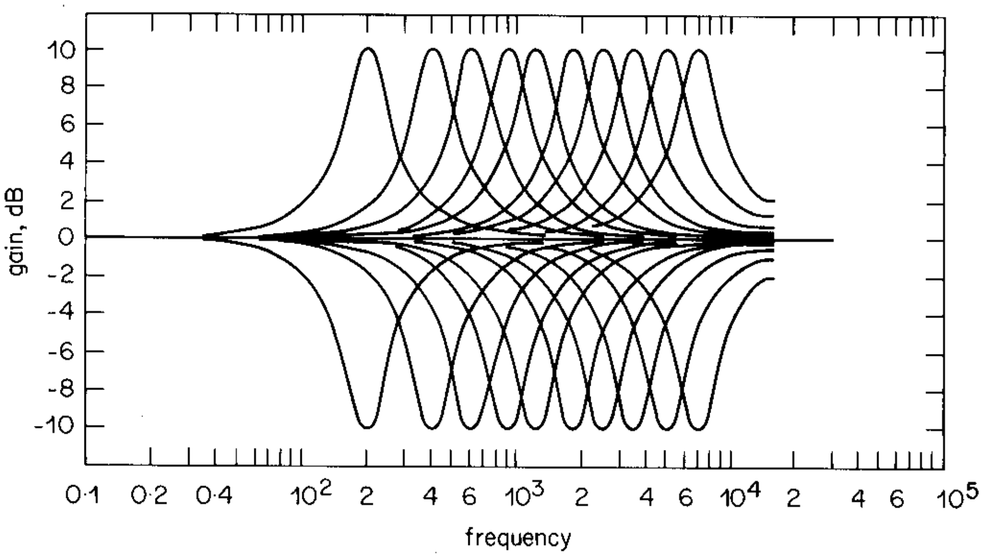

Fig. 7. Family of presence characteristics (gain = ± 10 dB)

then the effects of aliasing will be apparent in the amplitude characteristic of the corresponding digital filter. Optimisation techniques can be used to reduce the errors between this obtained characteristic and the desired characteristic.[12]

5.2.4 Programming the filter

The processing required to instrument the spectrum-shaping filter described above is five multiplications (including an overall gain term), three additions and two delays or stores. A 'library' of filter coefficients is held in the data memory and the program interprets switch settings to establish the response, and then the coefficients that are required. With the equipment used it was possible to provide for lift and cut of a low frequency 100 Hz shelf and a high frequency 7.5 kHz shelf in 2 dB steps to a maximum of ± 10 dB. Further, 'presence' controls arranged at musical half octave intervals between 200 Hz and 7 kHz were provided, and again these were variable in 2 dB steps up to ± 10 dB. This required the storage of 240 coefficients and a program of 44 instructions. The characteristics obtained are shown in Figs. 6, 7 and 8.

5.3 Automated mixing

Digital methods have already added significant improvements to analogue mixing-desks, particularly in automation. Automation is useful in, for example, a remix operation, where signals previously recorded on tape are to be mixed to a final, mono, stereo or quad output. During the remix, digital data specifying fader settings etc. are coded, formatted and recorded on an auxiliary track of the recorder so that on replay, the reverse process can be performed where the recorded data now controls the mix. In a simple system this may result in data rates of the order of 10 kbit.s^{-1}, which can be recorded within a normal audio bandwidth. However the period between successive recordings of the same parameter, or update time, could be as much as 100 ms and on an analogue recorder there may be crosstalk problems between the high level digital signal and the adjacent low level analogue tracks.[7] Further, when updating the information with new mixes there is a cumulative delay when reading the data from one track, modifying it, and recording it again, which could produce noticeable asynchronism between the audio tracks and their control data.

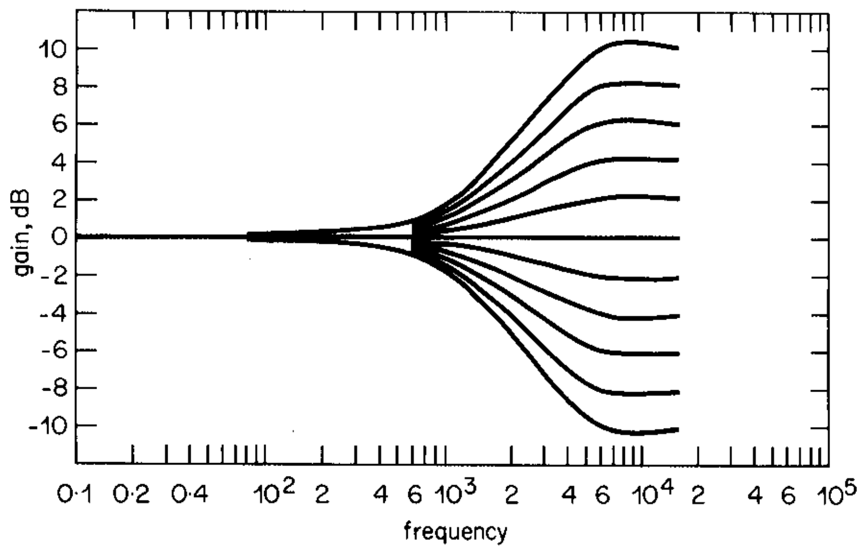

Fig. 8. Family of high frequency shelving filters (7.5kHz)

In a system in which the signal processing is also done digitally, many of these problems are greatly simplified. Firstly, the status of switch settings, fader positions, etc., are already conveniently available in digital form, i.e. they have already been formatted into digital words so that the processor can handle them. These status words can be accessed at any time by a separate computer and stored on a suitable medium under software control. Similarly, on replay, the stored settings can be routed to the processor, instead of the settings derived directly from the switches, faders etc. Thus the basis of an automatic mixer is inherent to the system.

The functions performed by the separate 'automation' computer could be similar to one proposal[8] which stores only changes in settings, and stores them along with a time code, labelling and error correcting code on a separate storage medium. On replay, this separate store can be interrogated in advance, the data checked for errors and loaded temporarily into buffer storage. The data can then be output at the precise time indicated by the timecode so that the mix is controlled without any cumulative delay.

A simple version of the system was programmed on the equipment described in Section 4.3. Data describing fader settings, etc., were labelled and stored on a spare (control) track of a multichannel digital recorder. Three modes of operation were available - a 'write' mode in which the coefficients etc. were recorded while the digital mixing operations were done; a 'read' mode, where the stored coefficients now controlled the mixing of the other tracks; and an 'update' mode in which minor changes could be made to data previously written on the track. This updated mix could either be recorded on a second track allowing comparison with the first attempt in the 'read' mode, or simply made to overwrite the first mix. Error correction was provided by the normal facilities of the recorder.

As the experimental equipment had to execute the mixing operations as well as provide the automation routines, a more complex system involving time codes, for example, could not be realised with the limited processing power available. However in a system where a dedicated processor is available, all the facilities described above could be provided; this is discussed further in Section 6.

5.4 Display of desk status and signal level

With the complexity of modern mixing desks it is essential to have a clear display of the status of switch settings etc. This has usually been achieved by 'single access' controls in which one action makes one change and the control itself provides the indication of that change. However, in the present context, all desk status information is available as stored digital information, and therefore it is possible to use a processor to format the information so that it can be displayed in a different way. This Report will not attempt to support one method or another, but simply point out some of the options with this approach.

Firstly, information can be displayed collectively in a highly ordered way on, for instance a video display unit (VDU), or other display device. Also, since the display is divorced from the control it is possible to set up the initial conditions of the entire desk from a program which has been stored on a separate medium. In reverse, the status of the desk can be recorded on to that separate medium so that for example a second user of the desk can work without losing the settings made by the first.

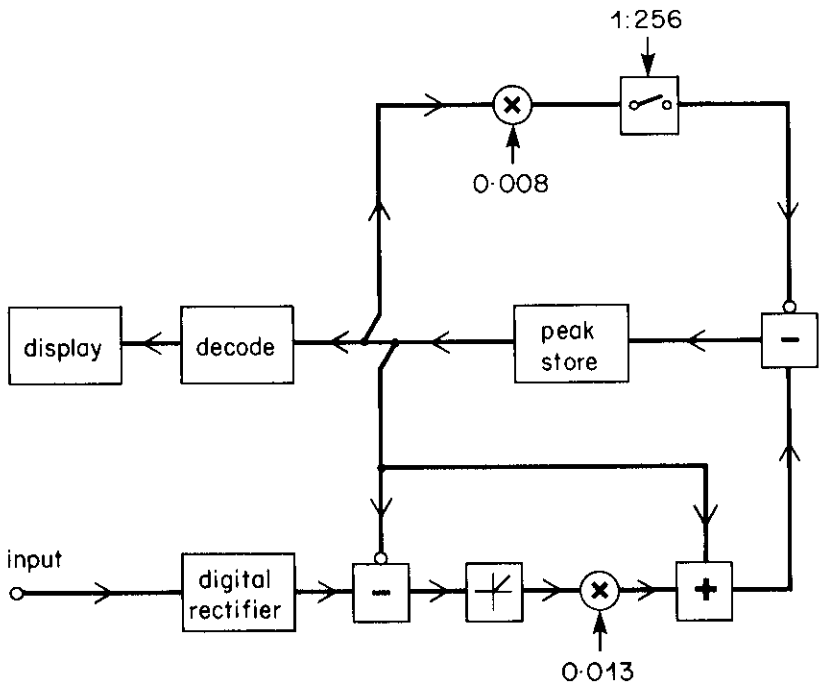

Fig. 9.
Block diagram of PPM program

Display of signal levels, conventionally done by mechanical meters or electronic displays can be made entirely program-controlled. The experimental equipment was programmed as a digital filter which reproduced exactly the ballistics of a peak-programme meter (PPM) and drove a bar-graph type display directly. Here, the hardware associated with these devices had been replaced by software and hence, the display could quickly be changed to a VU-type or genuine peak-reading meter without changing components etc. The PPM program required about 35 instructions and a few data words, and is shown in block diagram form in Fig. 9*.

*Patent No. 7901509

6. Proposed instrumentation of a digital processor suitable for a mixing desk

From the previous Sections, it is clear that the amount of digital processing involved in a mixing desk requires more than one processor. A fully equipped channel of a desk will typically require:

1. A fader with panning left, right, front and back.
2. Four filters of the type described in Section 5.
3. Assignment of the channel to various groups.
4. Provision of cue and monitor signals and output for reverberation units.

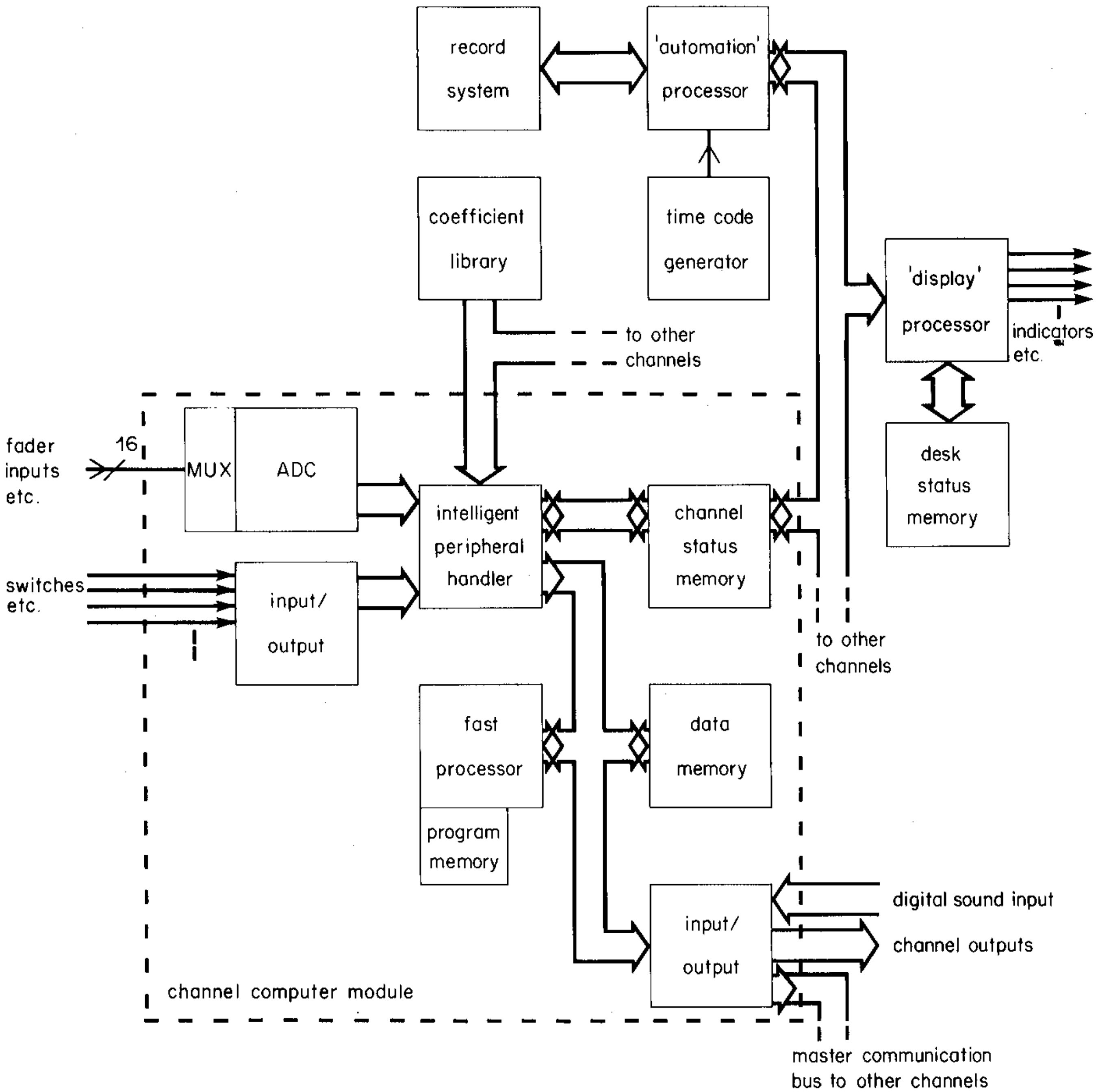

Fig. 10 - A multicomputer mixing desk

5. Miscellaneous switching for muting, solo etc.
6. Metering and displays.

Some of these functions, for example, muting and solo functions, can be performed with switches and simple hardware. Other functions, for example, the evaluation of coefficients which combine the actions of faders, pan-pots, and grouping information, can be achieved without the need for high speed processing. This can be achieved using one of the many microprocessors now available. Finally, there is the high speed processing which occurs in the audio signal path and includes the fading, using the pre-processed coefficients, filtering, and the generation of the metering data. This could be achieved using special purpose digital circuitry, but it is more likely that the use of a flexible high speed computer would be the most versatile and effective solution.

From the experience gained in the reported work, an estimate can be made of the number of (Miproc) instructions that would be required to implement these high speed functions. This gives a total of approximately 260 instructions and hence exceeds the capability of a single Miproc computer by a factor of three.

It is now possible, however, to design a purpose built processor which can work at very high speed.[13,14] An alternative system based on bit-slices as outlined in Section 3.2 which could meet the workload in the application of a mixing desk is shown in Fig. 10. This is a multi-computer system since each processor would have its own program and control circuitry with communication between computers by common system busses.

The high-speed data-processing is separated from the slow-speed operations, such as monitoring switch closures selecting coefficients etc., so that the full performance of the fast processors can be utilised. Thus, each channel computer-module would provide equalisation, panning, and the full range of monitoring signals, output signals to reverberation units, and cue signals which are required. Additionally, pre-processed PPM outputs and control signals could be provided. Each of the fast processors is identical whether it be in a channel or master location and thus modularity is achieved.

All peripherals are interfaced via the second processor. This relieves the high speed processor of housekeeping tasks and monitors switches, contrôls the multiplexing, conversion and calculation of fader settings, sets up status information and loads appropriate data, coefficients etc. into the memory of the high speed processor. The processor would be programmed separately to cope with changed switch assignments or data preparation, and again this ensures the flexibility of the system.

All the data representing the status of an individual channel can be stored in a separate memory. These memories can then be collectively accessed so that the information can be processed into a form where it can be displayed, or the data can be stored to permit automated mixdowns. The channel status memories could also be filled with data, either from the automation processor, or from a desk status memory which could load the entire desk to predetermined settings.

7. Conclusions

A mixing desk based on a digital computer has been constructed and the processing requirements were examined with special attention to filtering. The relative simplicity of the apparatus restricted the investigations somewhat, but the experience gained suggested that the total processing requirements of a single channel in a mix-down desk could be done by a programmed computer with approximately three times the processing power of the experimental equipment. In a practical desk, the high speed processes must be divorced from the 'housekeeping' by using a separate, slower, back-up computer so that the full performance of the high speed processor can be realised. Components are available now, notably 'bit-slice' microprocessors and single-chip microcomputers, which can adequately fit each of these requirements, and thus a versatile unit for use in each channel of a mixing desk could be constructed.

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9. Appendix

9.1 Mapping transfer functions from the 's' plane into the 'z' plane.

The general form of the transfer function of the biquadratic section can be written

$$H(s) = \frac{s^2 + 2\epsilon_1 s + \omega_{01}^2}{s^2 + 2\epsilon_2 s + \omega_{02}^2} \qquad (9)$$

Consider the particular transfer characteristic represented in Fig. 3.

$$H(s) = \frac{s^2 + 2\omega_0 \sin\theta_1 s + \omega_0^2}{s^2 + 2\omega_0 \sin\theta_2 s + \omega_0^2} \qquad (2)$$

This has singularities as follows:

Poles at $s_p = -\omega_0 \sin\theta_2 \pm j\omega_0 \cos\theta_2 = \sigma_p \pm j\omega_p$

Zeros at $z_p = -\omega_0 \sin\theta_1 \pm j\omega_0 \cos\theta_1 = \sigma_z \pm j\omega_z$

These singularities can be mapped into positions in the 'z' plane by the transform

$$z = \exp(sT) = \exp(\sigma T).\ \exp(j\omega T)$$

i.e. $r = |z| = \exp(\sigma T)$

$\phi = \arg(z) = \omega T$

Thus we can write the positions of the poles in the z plane as

$$z_p = \exp(\sigma_p T).\ \exp(\pm j\omega_p T) = r_2 \exp(\pm j\phi_2)$$

and $z_z = r_1 \exp(\pm j\phi_1)$

Hence the z transfer function can be constructed

$$H'(z) = \frac{(1 - r_1\exp(j\phi_1)z^{-1})(1 - r_1\exp(-j\phi_1)z^{-1})}{(1 - r_2\exp(j\phi_2)z^{-1})(1 - r_2\exp(-j\phi_2)z^{-1})}$$

where z^{-1} is the delay operator

$$= \frac{1 - 2r_1\cos\phi_1 z^{-1} + r_1^2 z^{-2}}{1 - 2r_2\cos\phi_2 z^{-1} + r_2^2 z^{-2}}$$

$$= \frac{1 + C_1 z^{-1} + C_2 z^{-2}}{1 + B_1 z^{-1} + B_2 z^{-2}}$$

where $C_1 = -2r_1\cos\phi_1$

$C_2 = r_1^2$

$B_1 = -2r_2\cos\phi_2$

$B_2 = r_2^2$

This process can be carried out starting with any general biquadratic function of s and the resulting z transfer function will be of the form shown above.

9.2 Calculation of the coefficients for shelving filters

The amplitude characteristic for a typical filter in this family is shown in Fig. 6. Such a characteristic can be produced by the filter having the general transfer function of equ. (9) but in which the response is critically damped. This requires double poles and zeros on the negative real axis of the s plane.[9] The transfer function thus reduces to

$$H(s) = \frac{(s+\sigma_1)^2}{(s+\sigma_2)^2} \qquad \text{where } \sigma_1 > \sigma_2 \qquad (11)$$

We wish to calculate the positions of the poles and zeros from a knowledge of the amplification or attenuation of certain frequencies, G_0 dB and a frequency, ω_0, which is chosen at the point where the gain or cut $|G_0|$ is reduced by half.

The amplitude characteristic is

$$|H(j\omega)|^2 = \frac{(\sigma_1^2 - \omega^2)^2 + 4\omega^2\sigma_1^2}{(\sigma_2^2 - \omega^2)^2 + 4\omega^2\sigma_2^2}$$

At z. f. $|H(j\omega)| = (\sigma_1/\sigma_2)^2$ and $\omega_0 = (\sigma_1\sigma_2)^{1/2}$

From which $\sigma_1^4 = \omega_0^4/A_0$ and $\sigma_2^4 = \omega_0^4 . A_0$

where $20\log_{10}A_0 = G_0$

The double pole at $s = -\sigma_2$ and the double zero at $s = -\sigma_1$ can now be mapped into the z plane and the transfer function $H'(z)$ constructed and the coefficients C_1, C_2, B_1, B_2 evaluated.

It will be appreciated that as the shelving frequency approaches half sampling frequency, the technique of mapping into the z plane will give a digital filter which is only an approximation to the required continuous filter, as mentioned in the text. However for shelving frequencies up to 10kHz the aliasing produced is not too severe, though a better approximation can be obtained using optimization techniques[12] based on this method.

9.3 Calculation of the coefficients for presence filters

The amplitude characteristic of a typical filter in this family is shown in Fig. 7. The characteristic can again be produced by a filter having the general transfer function of equ. (9) but with the simplification that the gain at both z.f. and h.f. is unity. Hence,

$$\omega_{01} = \omega_{02} = \omega_0$$

Let the poles and zeros positions be at

$$\pm j\omega_0 \exp(\pm j\theta_1),\ \pm j\omega_0 \exp(\pm j\theta_2),$$

as shown in Fig. 3. θ_1, θ_2 control the depth of the peak or notch and its Q factor while ω_0 sets the frequency scale. Rewriting,

$$H(s) = \frac{s^2 + 2\omega_0 \sin\theta_1 s + \omega_0^2}{s^2 + 2\omega_0 \sin\theta_2 s + \omega_0^2}$$

The amplitude characteristic is,

$$|H(j\omega)|^2 = \frac{(\omega_0^2 - \omega^2)^2 + 4\omega_0^2\omega^2 \sin^2\theta_1}{(\omega_0^2 - \omega^2)^2 + 4\omega_0^2\omega^2 \sin^2\theta_2}$$

At the turning point given by $\frac{d|H(j\omega)|^2}{d\omega} = 0$ a solution is $\omega = \omega_0$. The depth of the peak or notch at ω_0 is

$$A^2 = |H(j\omega)|^2_{\omega_0} = \frac{\sin^2\theta_1}{\sin^2\theta_2} \qquad (12)$$

For a notch $\theta_1 < \theta_2$ and the 3 dB frequencies are the solutions of $|H(j\omega)|^2 = 0.5$. If these frequencies are $\omega_0 \pm \Delta\omega/2$ then it can be shown that

$$\Delta\omega = \omega_0 (2\epsilon)^{1/2} \text{ where } \epsilon = 2(\sin^2\theta_2 - 2\sin^2\theta_1)$$

$$Q = \frac{\omega o}{\Delta\omega} = (2\epsilon)^{-1/2} \qquad (13)$$

Equations (12) and (13) together give

$$\left.\begin{aligned} \sin\theta_1 &= A\left[2Q(1-2A^2)^{1/2}\right]^{-1} \\ &\text{and} \\ \sin\theta_2 &= \left[2Q(1-2A^2)^{1/2}\right]^{-1} \end{aligned}\right\} A^2 < 0.5$$

For low values of Q with A^2 approaching 0.5, $\sin\theta_1$, $\sin\theta_2$ can exceed unity. For this condition the poles lie on the negative real axis of the s plane and if $\sin\theta = x$, $(x > 1)$, then the singularity positions are at $-(x \pm (x^2 - 1)^{1/2})\,\omega_0$.

9.4 Rules for generating the transpose configuration[11]

(1) Reverse all the directions of the branches in the network.

(2) Change directions of all delays and multipliers — keeping their values constant.

(3) Replace branch nodes by summation nodes and vice-versa.

CJS/AM

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