Microprocessor Mixing and Processing of Digital Audio Signals*

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0 INTRODUCTION

Digital techniques are being used more and more in the control and processing of sound signals. As may be expected, their initial impact is taking place in those areas where analog techniques no longer satisfy requirements, for example, in providing low-distortion recording or high-quality time delay. Digital magnetic tape recorders have been developed [1], and it is a logical step to provide digital processing, that is, 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 paper describes the use of such computers in various real-time investigations in digital audio processing. Experience was gained by using a commercial high-speed computer, and although the performance of the system was somewhat limited, a number of useful investigations were carried out. From these investigations it was possible to specify the design of a modular computer specifically for digital audio processing. The features of this module are discussed in the latter part of the paper.

1 PROCESSING UNDER PROGRAM CONTROL

1.1 Processing Requirements

The requirements for processing digital audio signals are little different from those provided by most computers, for example, the ability to carry out sequences of operations, such as arithmetic, logical, and data-transfer sequences, and the ability to choose between alternative sequences of operations at specified points, such as conditional tests, jumps, and subroutines.

This second feature is necessary for more advanced processing which goes beyond "number crunching." These features can be implemented by a computer with a conventional organization of a central processing unit (CPU) with its associated control unit, a store containing the data and program to be executed and devices to input and output the digital audio data.

High-quality audio signals must be sampled at 32 kHz at least, with 13 bits or more per sample, and so the computer must be capable of handling words of this length and running its program at the real-time sampling rate. Therefore to achieve a significant amount of processing, the computer must work very fast with a powerful instruction set.

1.2 Choice of Processor

At this level of performance there are three options:

1) To construct special-purpose processors with a limited number of applications, using "discrete" high-speed logic [2]. These can be managed by a relatively slow but flexible computer to give a versatile system.

2) To construct a module based on microprogrammable chips or "bit slices" [3]. These are "vertical" slices through the logic of the CPU of a computer and can be assembled to form a computer of any desired word length. These programmable modules could themselves be supervised by a slower speed computer.

3) To use a commercially available high-speed computer. Until recently the option would have been economically prohibitive, but suitable machines are now available whose cost is low enough to permit a valid study to be made, although it is still too high for computers to be used in large numbers.

This last option was clearly the most expedient for an initial study, and accordingly a Plessey Miproc processor was used. This has a speed such that 89 instructions can be executed in the 31-μs interval between audio samples at 32

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kHz, and it has a 16-bit organization.

2 DESCRIPTION OF FIRST-GENERATION EQUIPMENT

The heart of the equipment is the Miproc processor (Fig. 1). It has the high-speed CPU with memory space for 1k words (16 bits per word) of program and 0.5k words of data. Programs are prepared on paper tape and loaded via a reader into the RAM program and data memories, a useful feature when different programs are being tested and minor changes are frequentlv made.

A high-speed multiply instruction is provided by using a hardware multiplier as a peripheral. This permits a 16 × 16 multiplication with 16-bit product to be executed in 1150 ns.

Interfaces have been added (see Fig. 1) to permit the processing of 13- or 14-bit PCM audio signals. These are in serial form with a data rate of 448 kbit/s; 10 inputs and 6 outputs are provided. Low data rate inputs are also catered for. Analog inputs are digitized to 12-bit accuracy, and by multiplexing, 16 such inputs may be connected. Further, up to 256 switch settings can be interfaced via another unit. Unlike many computer systems, all these interfaces were designed to incur minimum software overheads so that most of the processing power was conserved for signal processing operations.

Much of the early work was concerned with the digital mixing of audio signals, and the system was made directly compatible with the BBC's multichannel recorder [1] and housed in a purpose-built console with high-quality faders and switches (Fig. 2). However, because the entire system is program controlled, the layout of the desk is not critical; each switch assumes the role for which it is programmed.

As a digital mixer, 12 channels are equipped with faders and two of them may be used as input/output units with full facilities. A third unit is partially equipped and may be an input/output or grouping unit, and a fourth can be used as a "master." However, it must be emphasized that the organization and control functions are entirely determined by the stored program.

3 APPLICATIONS OF THE EQUIPMENT

3.1 Fading and Mixing

Addition and multiplication are the main arithmetic processes for the digital mixing of sound signals. With the available processing power five sources may be mixed to a stereo pair with a master fader used to adjust the final output signal level. Although the program could eliminate the possibility of digital overload by ensuring that spare bits are available before additions, etc., it was not practical to provide "headroom" for all operations, and so the desk must be operated intelligently to take full advantage of the available dynamic range, just as in conventional mixing.

3.2 Spectrum Shaping

A mixing desk in which all signal processing is achieved digitally still has to provide the facilities found in conventional analog desks. It has become standard practice to supply a large number of different "equalization" or spec-

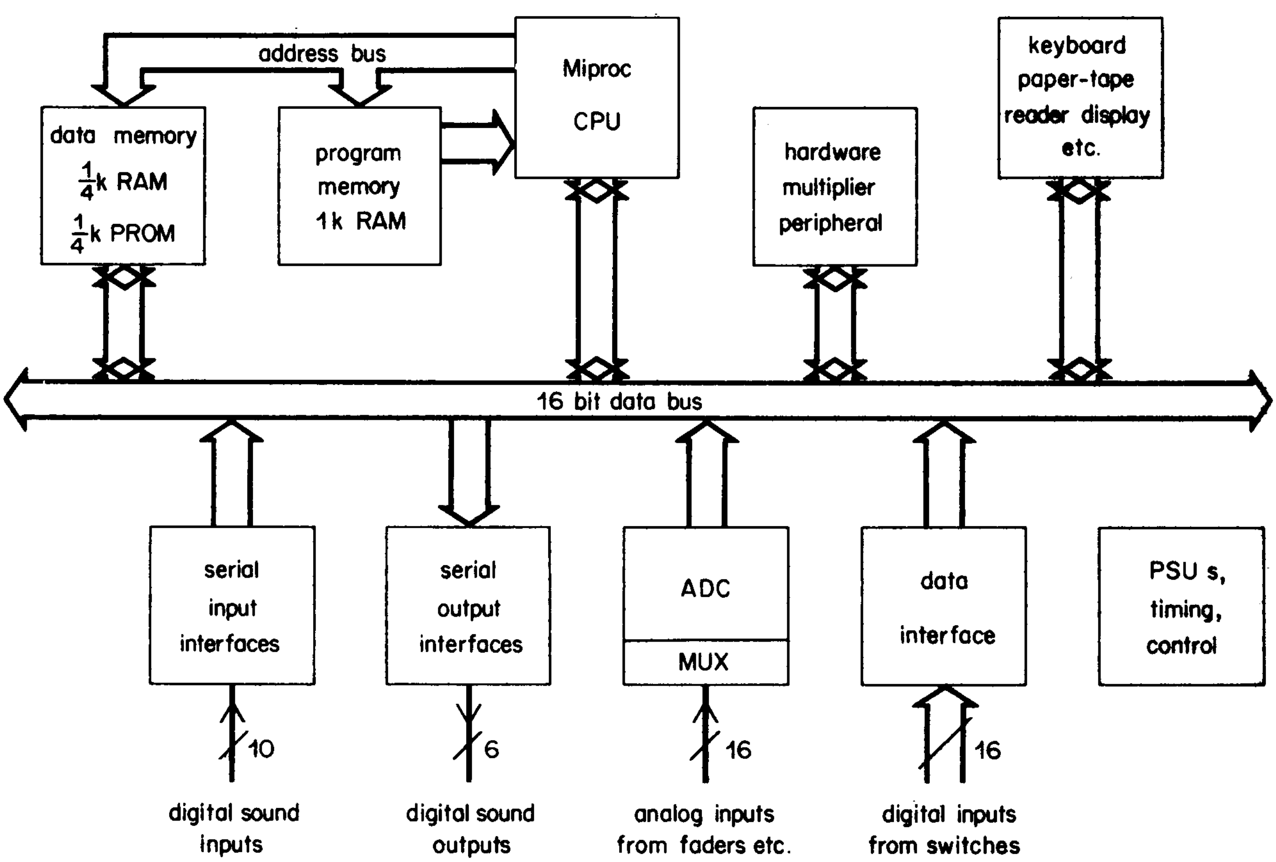


Fig. 1. Miproc system and interfaces.

trum shaping options, usually with separate control over low- and high-frequency shelving characteristics and mid-band peaking (presence) responses. The frequencies and degree of boost/cut of these characteristics are variable in steps, or sometimes continuously. Such specifications can be substantially met by a single digital filter structure in which the chosen characteristic is determined by an appropriate set of coefficients.

The transfer function for an analog filter producing these characteristics is a biquadratic function having two poles and two zeros and can be expressed as [4], [5]

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

The corresponding digital filter can be derived by means of the z transform [6], and the general form of this new transfer function is

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

where r_1, ϕ_1, and r_2, ϕ_2 represent the positions of the poles and zeros in the z plane using polar coordinates and z^{-1} is the transform operator which corresponds to a unit delay in the discrete-time sequence. This transfer function can be directly implemented using multipliers, adders, and delay elements as, for example, in Fig. 3. This structure and its derivatives, known as transpose representations [7], were chosen because they are insensitive to the errors that can occur in digital filters. These errors include the truncation of arithmetic operations in fixed-point arithmetic and overflow, which can occur at various nodes within a filter. The detailed analysis of the effects of these errors and the choice of structure on filter performance is the subject for a further study.

Rewriting,

$$H'(z) = \frac{A_0 + C_1 z^{-1} + C_2 z^{-2}}{1 + B_1 z^{-1} + B_2 z^{-2}}$$

it can be seen that the coefficients

$$\begin{aligned} C_1 &= -2r_1 \cos\phi_1 \\ C_2 &= r_1^2 \\ B_1 &= -2r_2 \cos\phi_2 \\ B_2 &= r_2^2. \end{aligned}$$

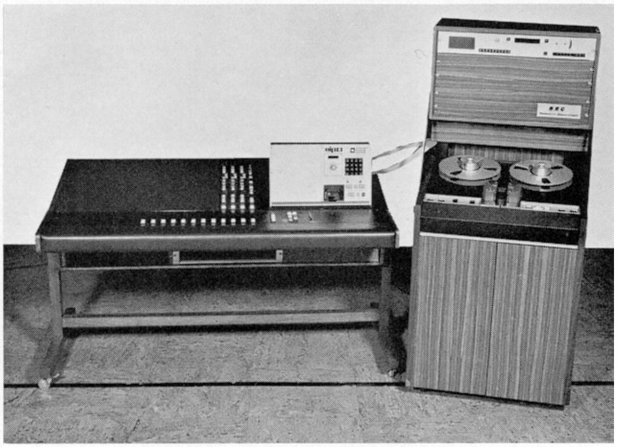


Fig. 2. Experimental digital multichannel recorder and mixer.

The coefficient A_0 controls the overall gain through the filter when C_1 and C_2 are scaled accordingly. A further scaling, arranged to be a factor of 2, is performed at the filter output to make best possible use of the available dynamic range. The five coefficients and the scaling factor can be chosen to provide all the characteristics described above.

3.3 Calculation of Digital Filter Coefficients

Consider a specification for a presence filter in which the amplitude characteristic is described by a center frequency ω_0, the boost/cut required A, and the sharpness of the peak/notch by a factor Q. So that characteristics which, for example, peak by only 1 dB can be accommodated, this Q factor will be defined as $\omega_0/\Delta\omega$, where $\Delta\omega$ is the width of the characteristic at a point where the gain has changed by Δf dB from its extreme value.

These characteristics are realized by confining the pole/zero positions in the s plane to

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

The transfer function thus simplifies to

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

It has been shown [4] that the boost/cut at ω_0 is

$$A = \sin\theta_1 / \sin\theta_2$$

and that if F is the gain corresponding to a Δf dB change from the extreme value, then

$$Q = [(1 - F)/4(F \sin^2\theta_2 - \sin^2\theta_1)]^{\frac{1}{2}}.$$

It can be seen that θ_1, θ_2 control the depth of the peak/notch and its Q, while ω_0 sets the frequency scale. Now the positions of the poles/zeros in the s plane can be calculated from

$$\left.\begin{aligned} \sin\theta_1 &= \frac{A}{2Q}\left[\frac{1 - F^2}{F^2 - A^2}\right]^{\frac{1}{2}} \\ \sin\theta_2 &= \frac{1}{2Q}\left[\frac{1 - F^2}{F^2 - A^2}\right]^{\frac{1}{2}} \end{aligned}\right\} \text{for } A^2 < F^2.$$

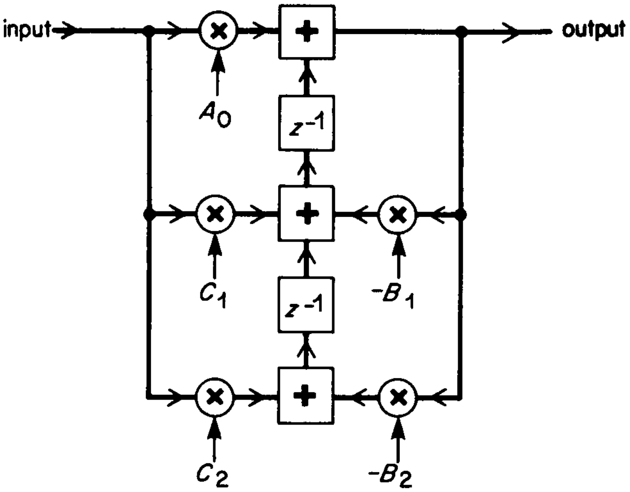


Fig. 3. Filter structure used for spectrum shaping.

For low values of Q with A^2 approaching F^2, sin θ_1, sin θ_2 can exceed unity. For this condition the poles/zeros lies on the negative real axis of the s plane, and if sin $\theta = x$ ($x > 1$), then the singularity positions are at $-[x \pm (x^2 - 1)^{\frac{1}{2}}]\omega_0$.

These singularities are then mapped into the z plane using the transform [6]

$$z^{-1} = \exp(-sT)$$

where T is the sample period. Thus the coefficients C_1, C_2, B_1, B_2, and A_0 are calculated. Fig. 4(a)–(c) shows families of digital filter characteristics obtained by this method.

A similar procedure is followed for the design of shelving filters. In this case the specification is described by the boost/cut A, the shelving frequency ω_b, defined at a point Δf dB from its extreme value, and a factor to determine the steepness of the characteristic Q. These characteristics are obtained by confining the pole/zero positions in the s plane to

$$\omega_p \exp(\pm j\theta), \qquad \omega_z \exp(\pm j\theta)$$

where sin $\theta = 1/(2Q)$.

The amplitude characteristic is solved at $\omega = \omega_b$ to give the required gain, and using the relationship that at zero frequency

$$A = \omega_z^2 / \omega_p^2$$

the values of ω_z and ω_p are calculated. For a given value of Q the pole/zero locations are identified and the digital filter coefficients calculated as before. Fig. 4(d)–(f) shows digital shelving filters derived in this way.

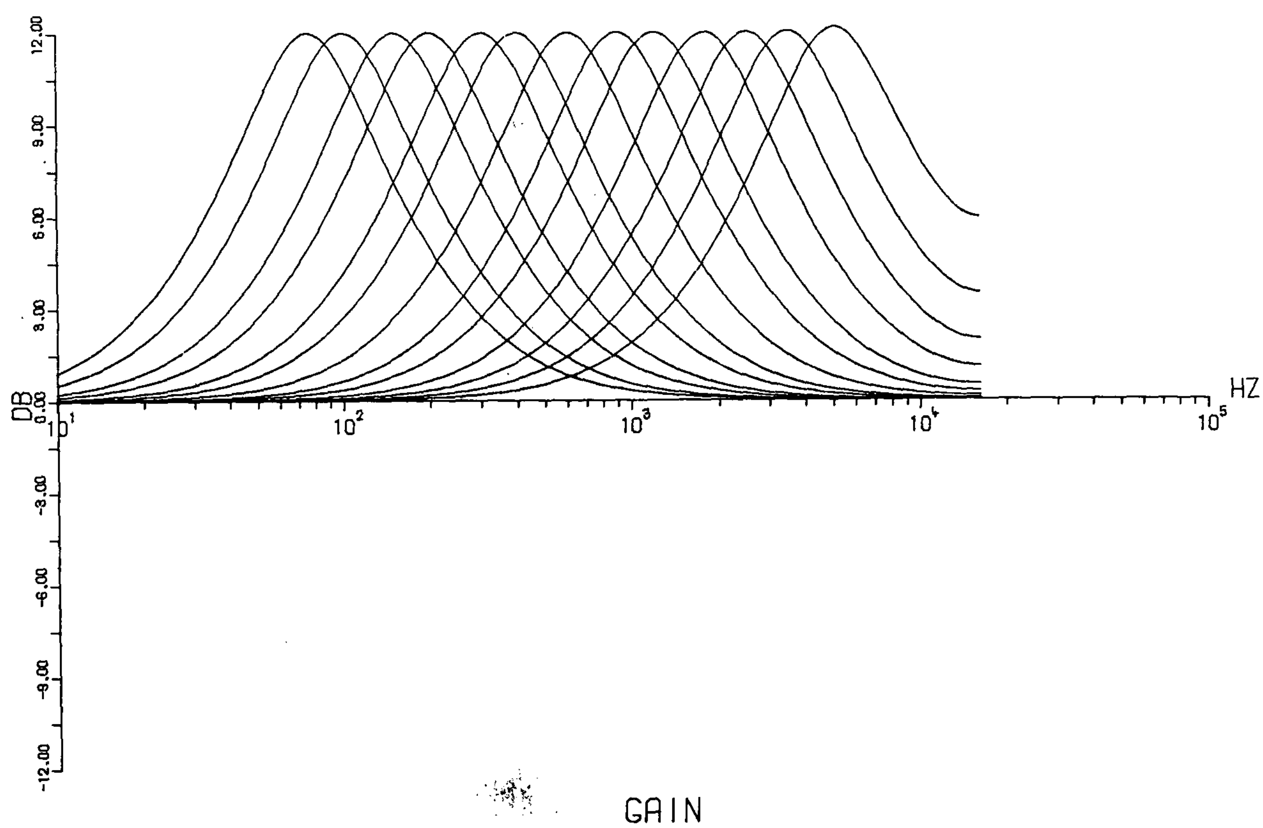
These characteristics were produced by the experimental equipment by storing a library of filter coefficients in PROM. The program monitors switch settings on the desk and select the appropriate coefficients for the characteristic required.

3.4 Automated Mixing

The process of recording desk adjustments during the mix, so that the mix can be repeated automatically at a later date, is simplified when all control settings are available in digital form. A variety of systems have been proposed [8], [9], most of which require coding of fader positions, etc., into digital words, which can be accurately stored. Thus the basis of an automated mixer is inherent in the computer system described in this paper, and a program has been written in which the digital control data are stored on a spare channel of the multichannel recorder shown in Fig. 2. Three modes of operation are possible: first the writing of control data while a mix is being made, second a read mode so that the control data can be used to control the mixing automatically, and finally an update mode so that minor adjustments can be made to data obtained in a previous mixing session. Due to the large amount of extra processing involved, only a very simple mix of four channels can be executed in the time available.

3.5 Display

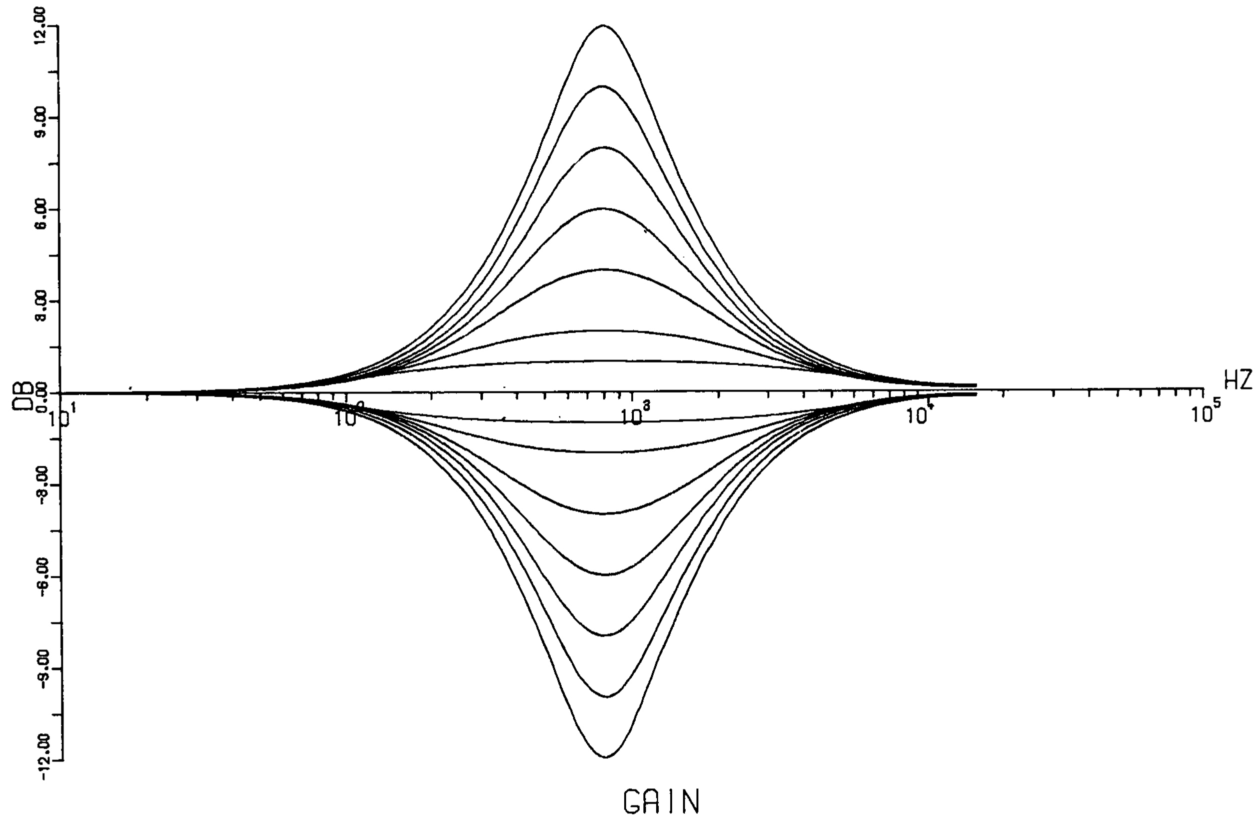
A digital mixing desk offers considerable scope for new methods of display for control settings and signal level



(a)

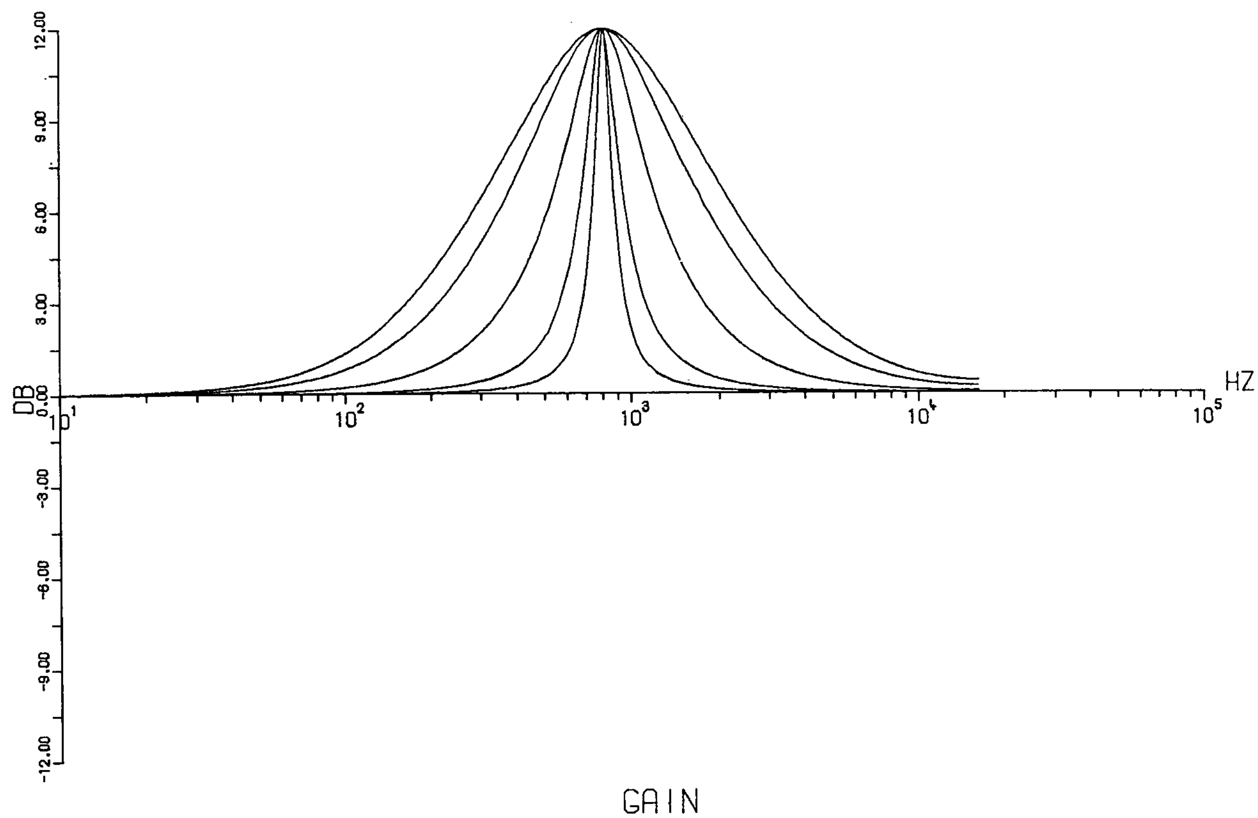
Fig. 4.(a) Family of presence filters, H = 12 dB, Q = 1.

monitoring. Assignable controls, that is, those where the single control may be used to specify a number of functions according to its "assignment," are easily implemented in the digital system. For example, a few controls can be used to set up all the spectrum shaping filters in a mixing desk. Since the status of the digital desk is at any time stored in the



(b)

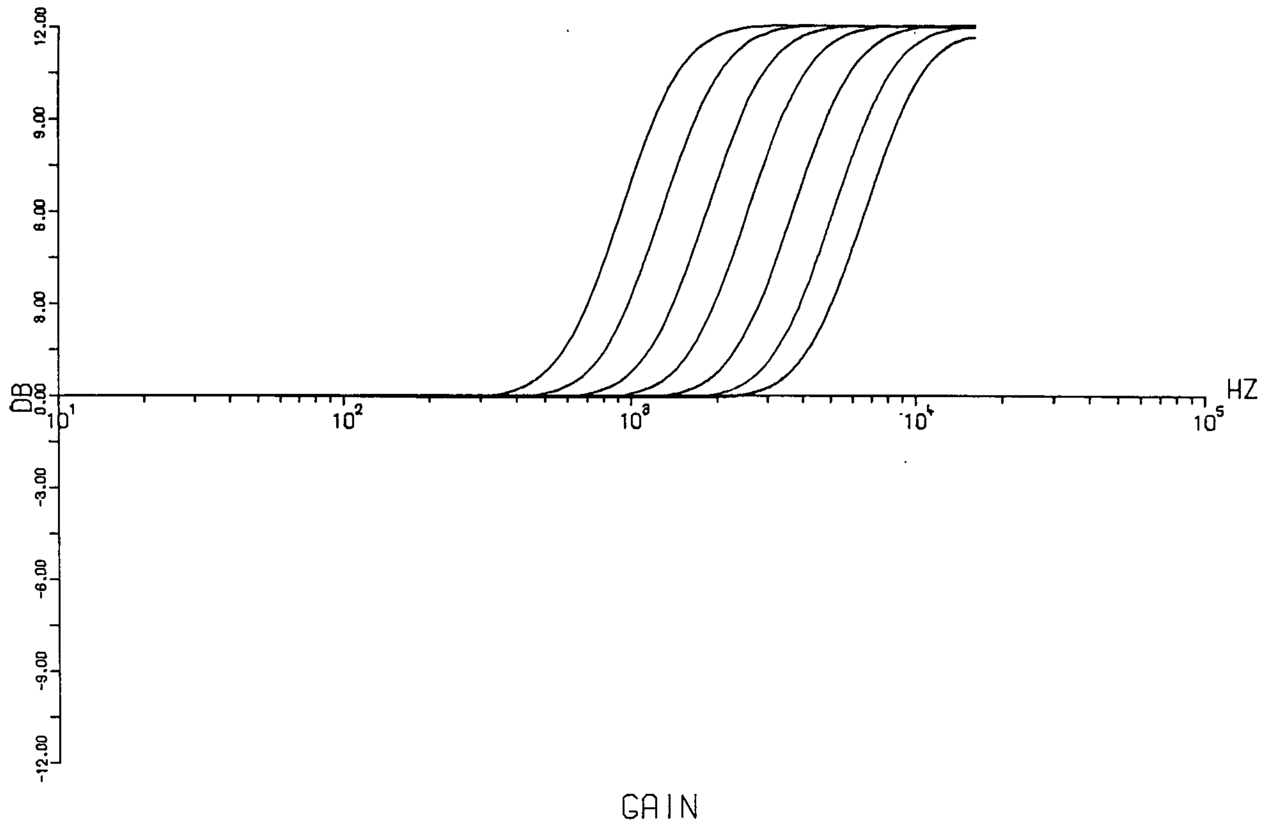
4(b) Family of presence filters, f_0 = 800 Hz, Q (0.25) variables 5.0–0.25.



(c)

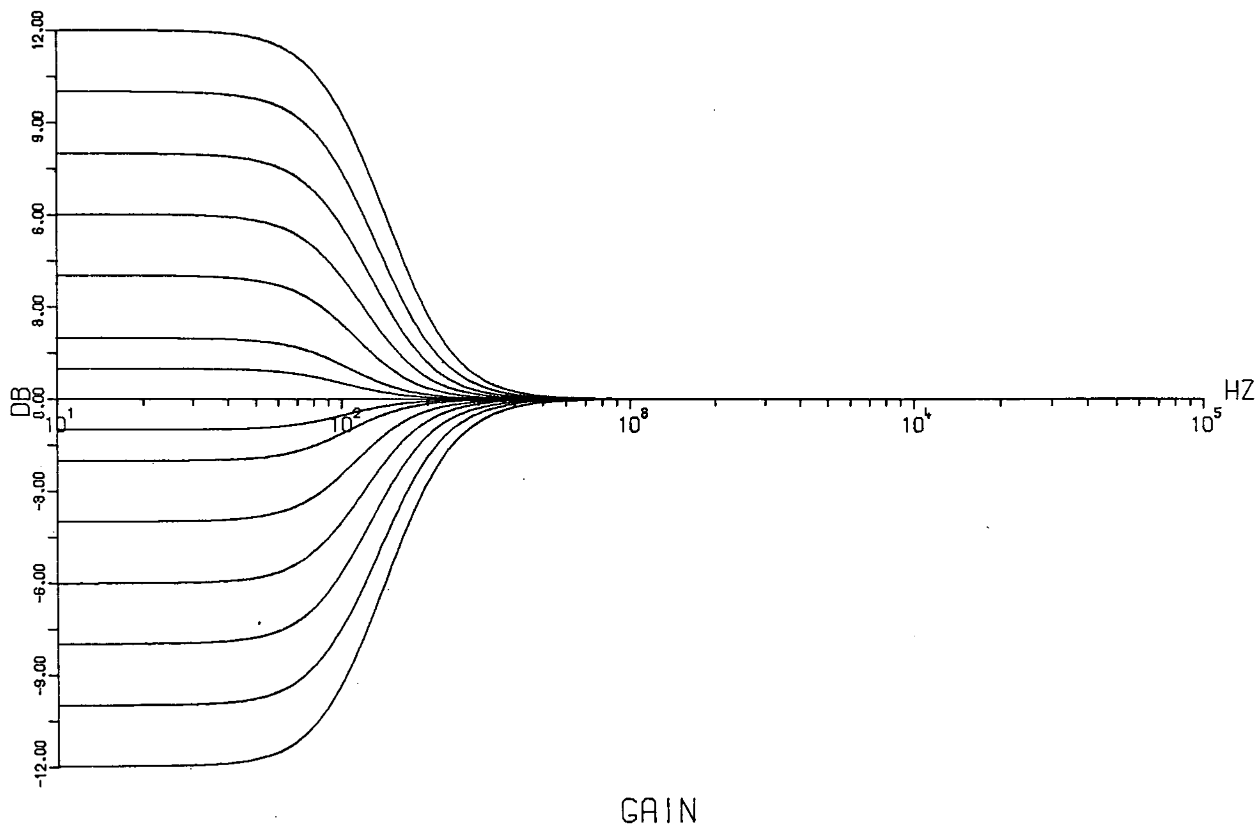
4(c) Family of presence filters, f_0 = 800 Hz, H = 12 dB.

computer's memory, it is also possible to output these data to a nonvolatile store. This can then be used to set up the desk at a known starting point or ensure that the settings are the same as in a previous session.



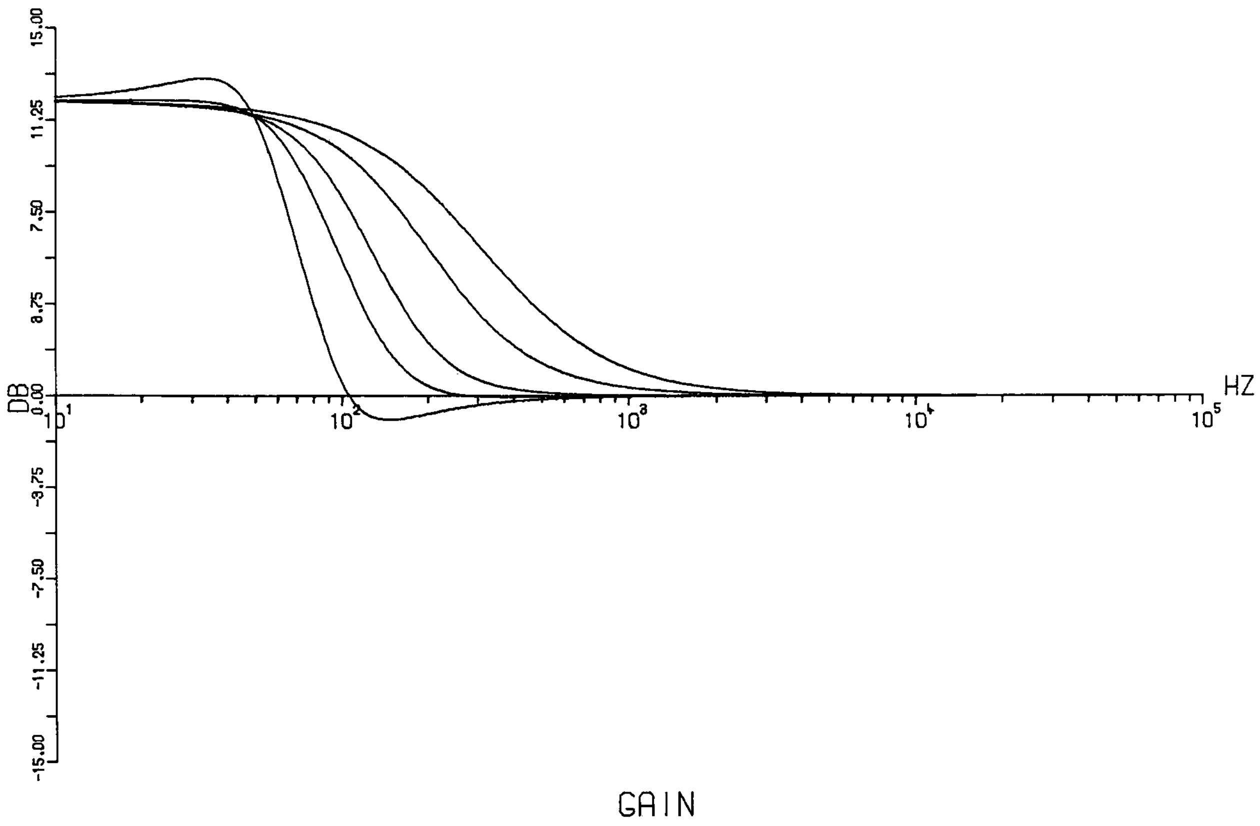
(d)

4(d) Family of high-frequency shelving filters, $H = 12$ dB, $Q = 0.707$.



(e)

4(e) Family of low-frequency shelving filters, $F_1 = 100$ Hz, $Q_P = 0.707$.



(f)

4(f) Family of low-frequency shelving filters, $H = 12$ dB, Q varies.

It will probably remain of interest to know of the signal level at various points in a digital mixing system, and this can be conveniently displayed after signal processing, such as in Fig. 5. This structure has been programmed and reproduces the ballistics of a Peak Programme Meter (PPM) and drives a bargraph display directly. Since the processing is controlled by software, it can be changed to a VU type or genuine peak reading meter without difficulty. This structure has also been used to implement the dynamic characteristics of a limiter/compressor where further processing on the "averaged" signal produces a control signal to be used to control the gain through the device. A number of advantages accrue from this digital approach, since there is no difficulty in matching the characteristics for a stereo limiter/compressor, and a short digital delay can be provided for critical applications.

The presentation of desk status to the operator would probably incorporate further processing to format the large amount of information in an easily perceived way on a device such as a color cathode ray tube. However, this aspect cannot be described as signal processing and will not be discussed further here.

3.6 Companding

Companding is a technique for improving the signal-to-noise ratio in audio recording and transmission systems. It can be used in PCM systems to make more efficient use of the available number of quantizing levels, for example, to reduce the bit rate required for transmitting a signal. These techniques [10], [11] have been investigated as a possible means for making use, for high-quality sound signals, of the 2048-kbit/s level in the digital transmission hierarchy in Europe.

Several companding techniques have been proposed, and thus there is the problem of assessing the relative merits of each of these systems. The equipment was programmed to execute six of these systems, with immediate switching between them to facilitate comparison [12]. Fig. 6 shows the arrangement used for the comparison tests. Pre- and deemphasis is used to reduce the audibility of program-modulated noise, and the delay-line limiter ensures that the dynamic range at the input to the ADC is not exceeded. In applications such as companding it is useful to know how impairments may accumulate if the same process is repeated several times in cascade. The digital recorder stores the processed audio data so that they may be reprocessed several times. By these means the effect of up to eight codecs in tandem can be assessed, a useful feature when the impairments introduced by one codec are very small. As a result of these tests, a decision was made to use a near-instantaneous companding system known as NICAM-3 for future BBC sound program links.

3.7 Signal Generation

The equipment has also been programmed to generate, digitally, musical test signals. Sine wave data are stored in read-only memory, and the program specifies frequency, attack, and decay times and internote time delays. By these means a highly repeatable precise signal source is available, which can be used for testing or lining up other equipment.

Another technique for synthesizing audio signals is to reprogram the filter section described in Section 3.2 as an oscillator. Fig. 7 shows a block diagram of such an oscillator where the coefficient B_1 selects the required frequency f according to the equation

$$B_1 = 2 \cos 2\pi fT$$

where T is the sampling period, that is, the delay corresponding to the operator z^{-1}.

The program specifies starting conditions at each node and the coefficients for the required frequency. The building block can be used as the basis of a completely digital music synthesizer [13].

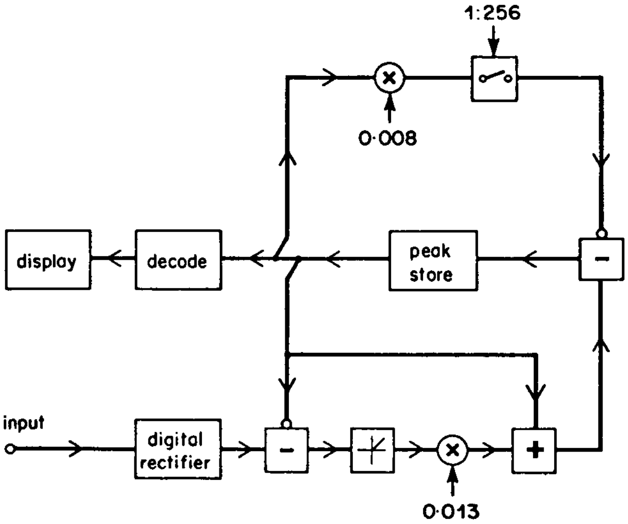


Fig. 5. Block diagram representation of a computer program for a digital signal level indicator.

4 MORE ADVANCED PROCESSORS

Although the experience of using the Miproc has been very encouraging, it is clear that in many areas more processing power is required. For example, the total processing required in a single fully equipped channel of a digital mixer would require more than three Miprocs. Therefore the options of Section 1.2 have been reassessed with a view to providing a processor specifically for audio data processing, with much more processing power.

Factors influencing the choice included the desire for a cheap self-contained unit which might be used in various configurations in large numbers. The choice had to be very versatile, implying that it should be software controlled. To construct a range of special-purpose processors as in the first option would not satisfy the requirements of cost and versatility, and so the design was based on the use of microprogrammable "bit slices." There is a speed penalty for this general approach, but the system is still at least five times as powerful as the equipment illustrated in Fig. 1.

5 DESCRIPTION OF THE COPAS SYSTEM

A computer for processing audio signals (COPAS) was devised comprising two separate computers: a high-speed microprogrammable processor based on bit slices and a slower processor based on a standard microprocessor chip. The two computers are closely linked and aid each other to work at their most effective levels (Fig. 8).

The high-speed processor used 4-bit slices to make a 16-bit machine. The processor has access to a 1k × 16-bit random access data memory which can be expanded to 64k × 16 using the 16-bit addressing capability of the computer. By these means up to 2 seconds of real-time digital audio can be accessed, indicated as bulk storage in Fig. 8. A hardware multiplier is used to provide either truncated 16-bit or full 32-bit products, and this component currently determines the cycle time, that is, the time taken to execute one instruction, at 163 ns. With a faster multiplier this can be improved to 135 ns. Interfaces have been added for two digital audio inputs and 16 outputs, and these too are expandable.

Each instruction for the computer is a 48-bit binary word. Since the machine is microprogrammed, the instruction can

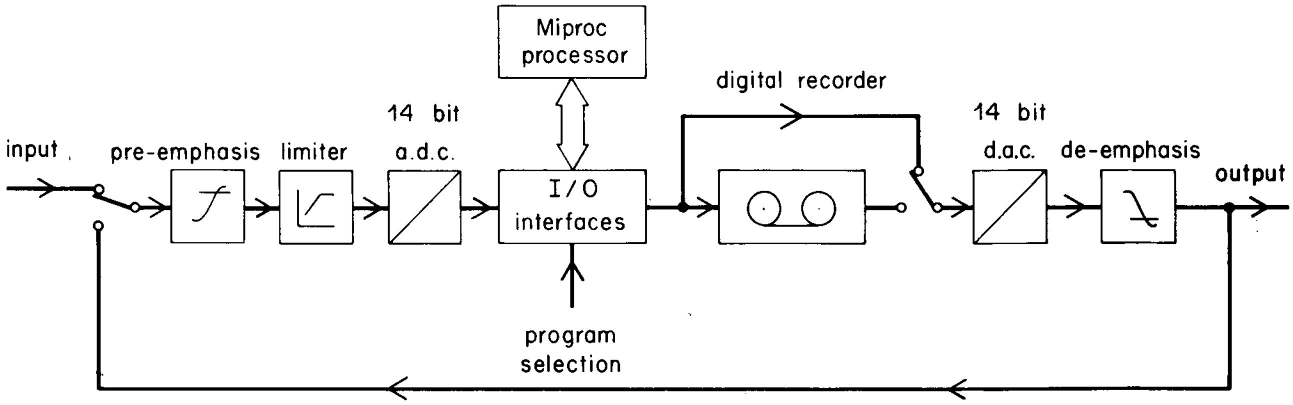


Fig. 6. Arrangement used for comparing companding systems.

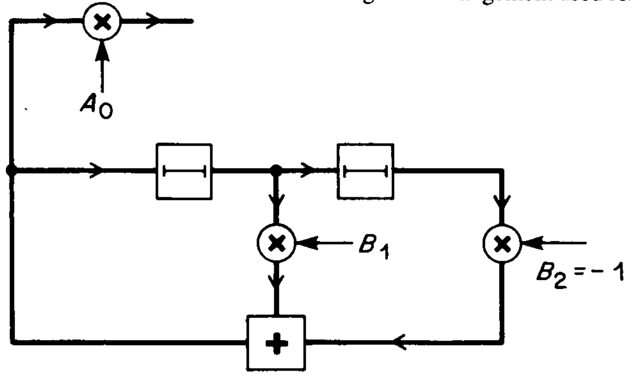


Fig. 7. Digital sine-wave oscillator.

be regarded as consisting of several groups of bits, where each group directly controls a specific function within the computer. Thus there are groups of bits, or fields, that control the arithmetic or logical function within the CPU, memory addressing modes, input/output, the location of the next instruction, the choice of conditional tests performed on previous results, etc. Altogether 13 fields completely define the operation during a machine cycle and thus give enormous flexibility to the programmer. Up to 512 instructions can be stored in the 512 × 48-bit program memory. RAM is used for this purpose, adding flexibility at the program development stage. The combination of powerful instructions with high speed of execution thus makes it ideal for real-time processing applications, and for audio signals sampled at 32 kHz almost 200 instructions are executed between samples.

The microprocessor support system serves two main functions. First it performs real-time processing for slowly changing events, such as monitoring and interpretation of switch settings and generation of suitable displays. It also executes all calculations for which the processing speed does not cause a problem, such as the selection of coefficients for a filter. Second it contains the operating system for the entire system, interfaces the high-speed circuits with the user/programmer, and supplies a number of utilities.

Consider first the real-time processing aspect of this support system. An example of its use might be to convert a control fader that has a linear law to one that has a logarithmic law. The result of the calculation is passed from the microprocessor to the high-speed system without time penalty by taking advantage of one machine cycle, the primary purpose of which is to lock the program in the high-speed processor to the audio sampling pulses. Similarly, results generated by the high-speed processor can be passed back, during the same interval, to the microprocessor for processing and display. By these means the high-speed processor reserves all its processing power for the real-time audio processing.

The operating system for the entire machine resides in 4 kbytes of PROM in the microprocessor system. The scope of this operating system falls into two categories, interfacing with peripherals and aids to program development.

The system currently interfaces with a VDU via an RS 232C link operating at 1200 band, though it is intended that the IEEE 488 standard bus will be eventually used. All system commands are supplied from the keyboard of this terminal. The operating system recognizes the commands and generates all the control signals necessary to drive a paper tape reader and also a digital cassette recorder. Programs prepared on paper tape can be loaded into the COPAS program memory and, if desired, subsequently stored on digital data cassette for easier handling. The operating system has facilities for writing, reading, and locating the programs on the tape. The tape can also be advanced and rewound using commands from the terminal. Other features of the operating system permit programs to be developed more easily. Instructions can be altered, deleted, and inserted using a simple editor, and individual instructions can be examined in detail using a disassembler. This is of great use since the 48-bit binary instruction conveys little meaning in its original form, even to an experienced programmer. Hardware contained on the board used in conjunction with a software trace routine permits single stepping through a program, with the contents of registers, memory addresses, etc., displayed on the VDU at each step. If required, data can be displayed graphically on the VDU using a plot routine, which generates axes and scales the data to suit.

Using these facilities, the user/programmer can develop,

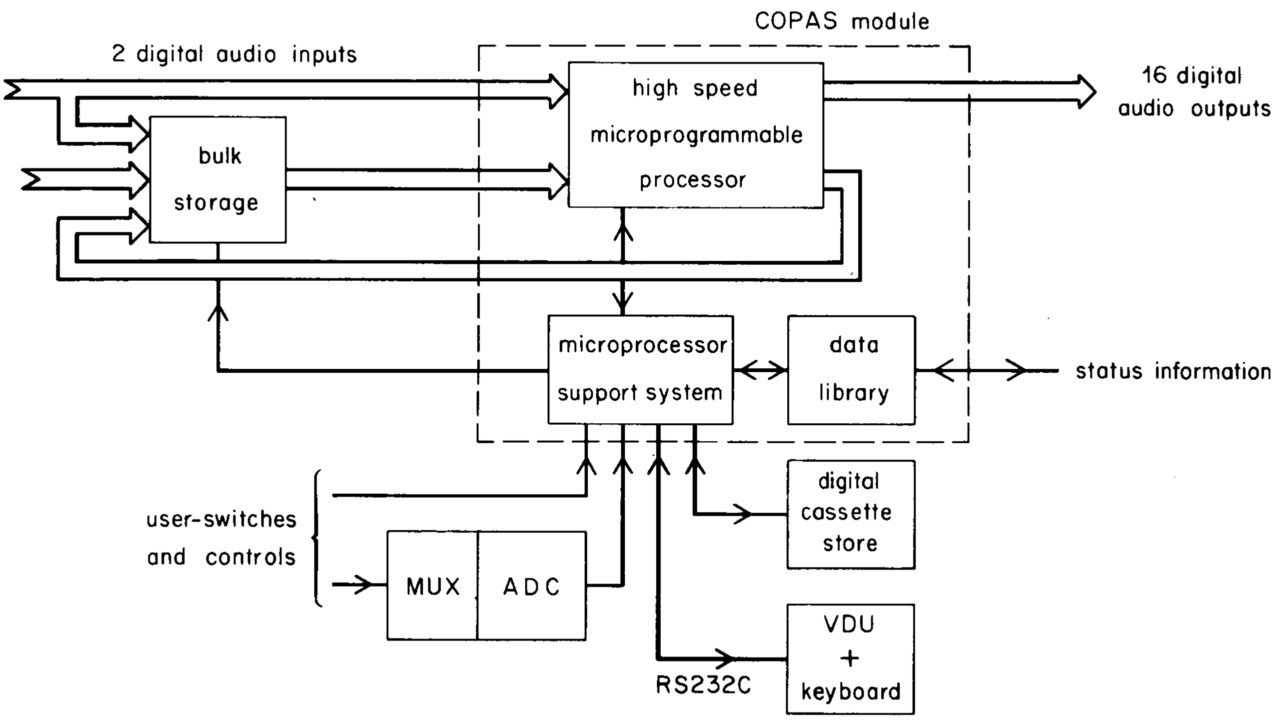


Fig. 8. General form of signal processing module.

edit, and run a wide range of programs quickly and interactively. In general two programs have to be written, one for each of the two computers on the board. Although the two computers are very closely linked, the built-in software resolves all problems associated with the communication between them. The programmer can call subroutines which supply or extract data from the high-speed processor without interrupting the real-time processing.

The module (shown inside pecked lines in Fig. 8 and as a whole in Fig. 9) is assembled on a single printed circuit board measuring 370 mm by 270 mm and contains 90 integrated circuits with a power consumption of 40 W. However, about 40% of this circuitry is accounted for by the development aids necessary for the prototype. The module contains all input/output interfaces and a crystal oscillator, from which all timing, including sampling rates, is derived. The module is therefore totally self-contained and can be used as a stand-alone system [Fig. 9(a)] for experiments in the same way as the Miproc system. Alternatively it can be combined with other modules to form a larger multiprocessor system. In this case each module [Fig. 9(b)] is microprogrammed for different roles within the system, and the microprocessor support system controls the "bus arbitration," that is, the passage of data between modules, utilizing the IEEE 488 bus.

Hardware testing has been simplified by incorporating sockets in the module which can be connected directly to a logic analyzer. By these means 75 key points in the processor can be monitored while the system is exercised using routines in the operating system.

Generating programs for COPAS has been simplified by writing a cross assembler which runs on a time-shared computer bureau. All COPAS instructions are described using a set of specially developed mnemonics. The assembler, which is written in Super FORTRAN, converts these mnemonics into the binary object code. This code is output on paper tape in a form suitable for loading directly into COPAS using its own reader. Another output of the assembler is the listings which can be annotated with comments and make up a clear record of the program.

Fig. 10 shows COPAS as part of a system for signal processing experiments consisting of a high-quality 16-bit ADC and DAC, paper tape reader, COPAS unit with the digital cassette below, and the terminal through which the system is controlled.

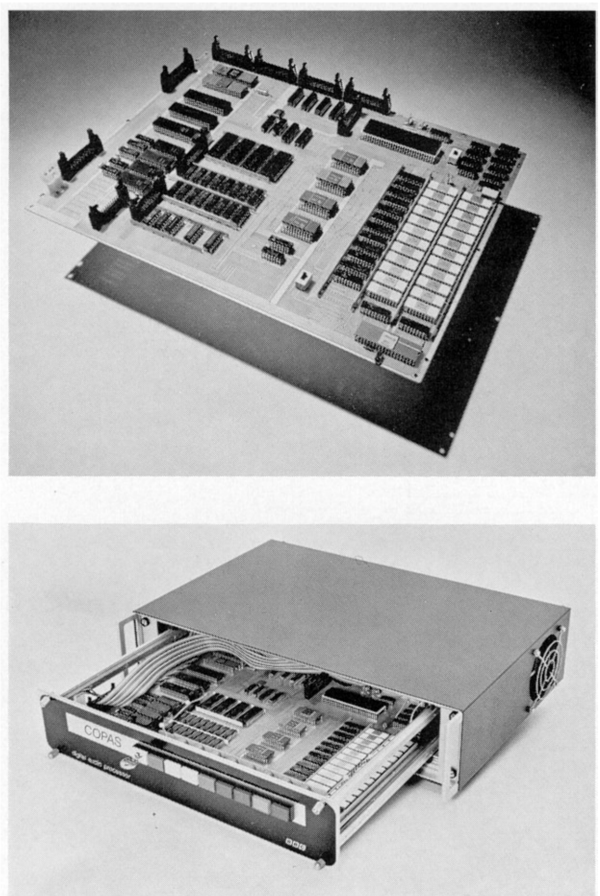
6 APPLICATIONS OF COPAS

The system shown in Fig. 10 was first used to perform some of the operations described in Section 3. This not only served as a useful way of commissioning and testing the equipment, but it also gave a yardstick by which the relative processing power of COPAS and the Miproc-based equipment could be checked. For example, a biquadratic section filter of the type described in Section 3.2 is executed in 18 instructions, that is, in less than 3 μs, a fivefold improvement.

This speed is maintained when the coefficients are changed, since this operation is managed by the support processor and incurs no software overhead in the high-speed processor.

The extra processing power means that a 64-stage transversal filter can be operated at a 32-kHz sampling rate. This is made possible, in part, by the two memory address registers which permit addressing operations to occur in parallel with arithmetic operations in the CPU. Again, coefficients can be dynamically controlled via the microprocessor support system.

Other processes that have been programmed permit COPAS to be used as a limiter/compressor/gain expander. A range of software has been developed to permit high-



(b)

Fig. 9. (a) COPAS module.
(b) COPAS as a stand-alone computer.

Fig. 10. COPAS development station.

precision working and hence evaluate subjectively the consequences of insufficient accuracy in arithmetic operations by controlling the level of truncation or rounding.

Inevitably there are many other applications, and among those that could be implemented using a single COPAS module are a range of special effects, signal synthesis, companding systems, spectral analysis, etc. Linked with bulk storage as indicated in Fig. 8, this can be extended to echo, artificial reverberation, and other effects.

7 CONCLUSIONS

A mixing desk based on a digital computer has been constructed and used in a wide range of audio signal processing investigations. The success of these experiments has led to the design of a high-speed processing module which can extend the approach into more complex processing problems.

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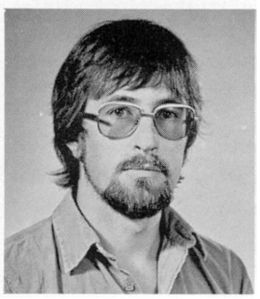

Guy McNally was born in Swansea, Wales. He studied electronics engineering at the University of Southampton, England, where he graduated in 1971. Since that time he has worked at the BBC Research Department at Tadworth, Surrey, in various activities in broadcasting including UHF directional coupler design, and projects in digital video and digital audio.

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