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# The evolution of digital processing for CD mastering

A.H. Proudfoot & R.E. Cameron
Neve Electronics
Royston, England

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# The Evolution of a Digital Signal Processing System for Compact Disc Mastering

A.H.Proudfoot
R.E.Cameron

Neve Electronics International Ltd, Melbourn, Royston, Herts, UK, SG8 6AU

**Abstract**

As the compact disc, and allied digital mediums for reproduction of high quality audio, gain increasing penetration and acceptance into the consumer market place, the significance of keeping the signal path in the digital domain during all stages of post production operation becomes an item of increasing importance.

This paper describes the development of one of a number of systems which are available to provide digital signal processing during these stages.

## 1. Introduction

During past years many records have suffered from being mastered from tapes that may be a number of generations removed from the original session tape. Often analogue transfers may have been copied from machines that were not adequately lined up, and therefore suffer from a consequential loss of quality.

In the case of compilation albums, the studios and/or record company may not have been able to find the original session tape, or determine how many generations a production master is removed from the original. All of these problems create albums, even if on CD, that are inconsistent from track to track and are substandard at best.

There is a growing awareness that the greatest degree of degradation to digital audio signals takes place during repeated conversions from A/D and D/A due to the effects of anti-aliasing and anti-imaging filters, and of the converters themselves. Quality may thus be improved by using digital signal processing, where no D/A or A/D conversions are required.

However, the myth that anything done in the digital domain is free of any undesirable side effects has been dispelled as more people become familiar with digital equipment. It is clear that, although 16-bit converters are used, the internal resolution of the equipment needs to be much greater to avoid loss of information and thus audio quality in fading and other processing.

Several manufacturers are offering consoles which address these problems, ranging by orders of magnitude in both price and performance. This paper examines the development of one such console, the Neve Digital Transfer Console (DTC) (1,2).

The application of the DTC for CD mastering and digital post production allows for all processing, including EQ, filtering, dynamics and level control to be performed within the digital domain.

The evolutionary path of this console is perhaps different from all others, being developed from technology first seen in full-blown digital consoles. This path is traced in detail in the following sections, with both the similarities and the differences between it and its stable-mates highlighted. The later sections illustrate how the DTC may be used in the studio environment to increase both the quality and the efficiency of digital mastering.

## 2. Design Goals

The DTC was designed to meet the requirements outlined in the introduction of this paper. In particular, it was to:

- Provide digital level, EQ, and dynamics control of professional quality;
- Provide accurate and 'smooth' A/D conversion for mastering from analogue masters;
- Provide similar D/A conversion for both monitoring and vinyl cutting;
- Provide facilities to automate the mastering process to SMPTE timecode using 'snapshots', accurate to within a frame;
- Allow connection to various digital devices used in mastering and post-production;
- Permit operation at either 44.1kHz or 48kHz sampling rates;
- Deal automatically with emphasis, with manual override for anomalies; and
- Achieve all of the above whilst still remaining simple to operate.

## 3. System Description

The DTC has two output channels driven from two independent mixes that accept signals from two digital sources and one analogue source, each controlled by an assignable fader. The digital inputs are provided with manual and automatic digital de-emphasis. The outputs are controlled by an assignable fader and are available in digital and analogue form. Processing functions, available on the outputs, include high-pass and low-pass filtering, equalizing and dynamics; if required digital pre-emphasis can be applied to the digital output (Figure 1).

All functions digitally implemented and, when processing digital recordings, there are no A/D or D/A conversions in the programme path.

Because the control surface of the console has no direct connection with the signal processing circuits, control assignability can enhance the DTC's range of operational modes. In general, controls affect identical functions on both Left and Right channels, but filter, equalizer and dynamics controls may be freely assigned to operate on Left, Right or both channels. Faders are also assignable in operation, but via a different system of assignment control which results in faders moving to take up the position of the desired selection, and displays changing, without affecting the signal.

The DTC can store up to 250 'snapshots' of control configurations which can then be inspected, modified and recalled for repeat performance. Stores may be stored manually or under the control of the automation system. Certain control modes allow inspection and modification of the contents of any one of the 250 stored snapshots without adjustment to the audio signal. (3).

## 4. Evolutionary Path

The design of the Digital Transfer Console owes much to the work put into Neve's previous digital products, but it is more than a cut down 48-into-32 mixer, or an expanded stereo in / stereo out transfer unit.

### 4.1 Control Surface Assignability

The control surface philosophy common to all Neve DSPs is used in the DTC, giving an elegant and functional surface which has been tailored to the new application. This has advantages in allowing the console to be set up automatically with no requirement on the operator to null pots or make any other manual adjustments.

The principle of assignability, first demonstrated on the early Neve DSPs, is used in a slightly different way. The most common usage is to allow one set of controls to operate on several different parameters in order to simplify the control panels. This is most obvious on the motorized faders, each of which adjusts not only the path fader, but also independent Left, Right, and Stereo trims. In a similar manner to the 'ACCESS' operation of larger systems, the filter, equalizer and dynamics panels may be 'assigned' to operate either on the Left, Right, or both (ganged) paths of the output bus.

The metering system also demonstrates assignability of function as well as source, with the left-hand pair able to meter either the incoming analogue signal or the sidechain control voltage of the dynamics unit. All of these operations are performed by single key-presses. It was decided at an early stage not to incorporate a VDU into the console, but to allow all functions to be easily controlled using either a single key, or a simple sequence. With logical grouping and labelling of controls, and a sensible default mode on switch-on, it is easy to operate the console from power-up, and to learn quickly how to use the special facilities it has to offer (Figures 5 - 10).

### 4.2 DSP Hardware

The relationship to the other members of the Neve DSP family is obviously more than skin deep. The heart of the DTC signal processing is the standard 28-bit processing cards used in all other Neve digital desks, with the same 16-bit converters employed on the analogue inputs and outputs. The standard cards provide an internal data format which uses 20-bit exponential notation to give acceptable headroom within the console. To limit this to the 16-bits used by the converters will result in lost information in intermediate stages, and hence reduced quality of audio. Also, it is essential to use a larger number of bits in the main ALU, in order to give the accuracy which is required in each calculation. Otherwise, the inaccuracies will result in audible distortion, such as limit-cycle noise.

Although this tried and tested heart remains the same, many changes have been made around it to provide an efficient implementation for the smaller architecture of the DTC.

### 4.3 Internal Routeing and Interfacing

The large systems, including the vinyl mastering and sound archival variants, employ a system of fibre-optics to interface the intelligent console section to the processing-intensive racks. For the DTC, this is an unnecessary facility because the communications bandwidth is small. In common with the earlier Digital Transfer Unit (DTU), the console in the DTC is merely a dumb remote-control device, with all real intelligence located in a single CPU in the rack section.

Fibre optics are also employed to link the digital racks into the analogue input/output boxes in previous systems. As the I/O need not be remote from the processing, and is of limited bandwidth, this interfacing hardware is replaced by a special parallel interface card, while the analogue and digital boards share a common rack. Suitable precautions have been taken to avoid digital breakthrough into the analogue boards, based on Neve's experience with mixed analogue and digital technology.

Figure 2 illustrates the interconnections between the various parts of the DTC system.

Being aimed at the mastering and post-production stages, the DTC does not require the extensive routeing and processing facilities of a full-size system. This overhead is replaced by a specially designed routeing board, which is capable of interconnecting all the audio paths in the DTC. A system derived from the large DSPs is used. During one phase of the wordclock, all sources (which originate from the outside world and from internal processes such as faders) are read into a block of memory called audio RAM, using time-division multiplexed (TDM) buses. During the next phase of the wordclock, the contents of particular locations of this RAM are output on several TDM buses to the processing or output sections. Each such section has its own unique timeslot on the bus, with a programmable sequencer determining which audio RAM location is sent out on which slot. This process, illustrated in Figure 3, allows any signal to be routed to any processing/output section.

### 4.4 External Interfacing

#### 4.4.1 Digital Interfaces

The limited number of paths in the DTC does not justify the use of standard 16-channel interface cards, which, coupled with the need to provide several formats, and pre- and de-emphasis, led to the development of dedicated interface boards. To avoid having to place a third DSP processor in the rack, the emphasis and metering filtering is performed on one of these cards, using four TMS3020-20 single-chip DSPs. They also generate the internal tone which is available to the operator for check purposes. In order to give the performance expected of professional audio equipment, these 16-bit devices have to be used in their double-precision mode. Although these new DSP chips are suitable for these limited tasks, it is interesting to note that they require some 12us to perform a filter section, whereas the Neve DSP cards take less than 1us. Therefore it was not felt necessary or advantageous to re-design these cards at this stage.

#### 4.4.2 Analogue Interfaces

In view of the requirement to operate at both 44.1kHz and 48kHz, several minor adjustments had to be made to the converters and analogue stages. These involved a redesign of the anti-imaging filters, to provide the stop-band of a 44.1kHz filter with the pass-band of a 48kHz one. This required a compensation circuit to remove the slight additional sinx/x droop when operating at 44.1kHz.

### 4.5 Software Evolution and Design

With all intelligence now located in a single CPU, it was necessary to combine software modules from both the console and the processing rack controllers of the larger systems. With a number of new and complex boards in the system, new tasks were written to control these cards. New software was also required to implement the automation system and to provide extensive diagnostic capabilities to aid production and servicing.

Although the DSP cards and some control modules are identical electronically to those used in larger systems, their operational characteristics and functions are somewhat different. Whereas a DSP card normally would implement EQs, filters, and dynamics units, in the DTC each card handles a complete channel. This comprises three input faders, a three-into-one mix, an equalizer, filter and dynamics unit, and an output fader. This allows further reduction of hardware, because separate fader and mix cards would be required normally.

Even the EQ, filter and dynamics sections themselves have changed to provide increased range, special characteristics, and increased performance, tailored to the needs of the post-production engineer.

This has all been accomplished in 'software', a term which may be applied to anything from timing and microcode PROMs, through assembly language routines, to high-level control algorithms. No hardware modifications were required, the cards being sufficiently flexible to be used in different systems to perform different functions.

As this serves to illustrate, software is playing an increasing role in audio products today. The advantages software based systems give include ease of update, flexibility, and reduction in complex hardware sequencing or control circuits. There are some inherent disadvantages, which have been seen in some products particularly (but not exclusively) in the early stages of their lives. For example, a software-intensive operation may appear to be slow in response to control adjustment if either the algorithm (instruction sequence) is inefficient, or the underlying processor is not sufficiently powerful. A more major problem, which many are all too aware of, is the tendency for software to wander off in some unintended direction, with the resultant operational or, even worse, audible 'crash'.

Just as a professional approach to hardware design eliminates many problems before they arise, so a structured approach to software design helps to produce systems which are more reliable, and easier to modify in the future. With recent developments in microprocessor technology, the speed criteria should be much easier to improve, giving greater scope for running better control algorithms.

The flexibility advantages of a software-control system have been exploited in the DTC chiefly in the implementation of the automation system. As Figure 4 shows, there are no direct links between the three major parts of the system - the DSP, the console, and the automation memory. The software permits these parts to interact in a number of different ways, so providing the basic functions of the automation system. These are discussed in more detail in the next section.

## 5. Operational Features

Not only does the DTC represent an evolution of signal processing techniques and implementation for disc mastering, it is also an example of the evolution of mastering techniques themselves. By separating the control surface from the audio processing, it has become possible to control the audio automatically, with no manual intervention required. This is achieved by a system of snapshot memories.

Each snapshot contains all of the information required to configure the entire system. The settings of the controls may be copied into a snapshot, together with the current timecode, by a one-key operation. When that timecode recurs, the automation software (if enabled) copies the settings from the relevant snapshot back into the DSP and the console, thus reproducing the original effect.

Previously, as the engineer listened to the 'flat master' tape, he would begin to apply corrective or artistic changes to the material. Once he obtained the desired result, he would note down the settings and move to the next portion, only to recreate the settings he used when the master copy was made. What the DTC allows him to do is to dispense with this time-consuming process of noting down settings and resetting the console by hand. Instead, he simply creates a snapshot of all the console settings. Because these snapshots may be created and recalled at any time, without undesirable effects on the audio, changes to the treatment used are not limited to the breaks between tracks; as many as 250 snapshots may be used at any one time, with more available by recall from a floppy disc.

Of course, the engineer may not get things 'just right' first time, and may wish to change either the treatment itself or the time during which that treatment is to be used. Both these operations are supported, again using one-key operations. The timecode associated with each snapshot may be resynchronized either on the fly, or by editing it to precisely the right timecode, to frame accuracy. This facility allows changes to be brought in at precisely the right time, perhaps to catch a drum-beat, or to deal with a sudden level change on the original. Once the correct times and settings have been set up, they are applied repeatedly as often as required. No lengthy rehearsals are necessary to ensure that the final take is exactly what is required.

Unlike 'dynamic' automation systems, individual control movements are not recorded against timecode. Instead, an explicit action is required to create a snapshot of all controls. This is perfectly acceptable for EQ and dynamics changes, which rarely if ever (in mastering at least) require dynamic automation. For level changes, however, this method would lead to large numbers of closely-timed snapshots being used for even a simple fade-in or fade-out.

To overcome this problem, an 'auto fade' facility has been included. Using this system, the operator creates a snapshot, with a special marker set, at the beginning of the fade. He then performs the fade, inserting new snapshots whenever he changes the rate of movement of the fader. When the automation system recalls the first snapshot, it begins to interpolate each fader from its original position to its 'target' position in the next snapshot, during the time between the two snapshots. This process is repeated for the inserted snapshots, with no limit other than the number of snapshots available. Typically, a fade may be built up by this method using three or four snapshots, as opposed to thirty or more previously.

An example of the flexibility of software control is the Preview mode of operation. Here, the console controls may be set from a snapshot, while the DSP itself remains as it was prior to the preview operation. The controls may be adjusted and the snapshot may be modified with no effect on the DSP. This is particularly useful during automated playback as it allows the next snapshots to be viewed before they are recalled. The automation system continues to recall snapshots at the appropriate times, updating the DSP but leaving the console to preview the selected snapshot.

By giving the operator not only a completely digital processing path, with extensive frequency, dynamics, and level controls, but also an automation system for control, there is much scope for producing better quality product in the same, or less, time than with conventional equipment. Indeed, these new facilities will undoubtedly lead to advances in operational practice as new applications are found. One customer, for example, has already found that by utilizing the snapshot system to the full, he has been able to transform an otherwise unuseable tape into a satisfactory master.

Once the session is over it is essential to know what treatment the tape has been given for future reference. The DTC can generate a tracksheet of the snapshots used on a suitable printer; this tracksheet may be kept with the master as normal. The DTC also allows the snapshots to be recorded onto a 3.5" floppy disc, from which they can be recalled at any time in the future, allowing the entire automation system to be configured as it was when the disc was recorded. This also enables a project to be split over a number of sessions, leaving the equipment free for something different in the intervening periods. (4).

## 6. The DTC in the Digital Studio

As digital equipment continues to proliferate, it is becoming increasingly important to ensure that each unit can communicate with the others in the digital studio. Not only must they be able to operate at the same sampling rate, they must also have compatible interface formats. The most common sampling rates in recording, 48kHz and 44.1kHz, are supported by the DTC. In many cases, it is desirable to have one master wordclock in a studio complex, which is not always possible if self-clocking serial interfaces are used without reclocking of the data. The DTC may be selected as master or slave with respect to wordclock, with a preference to be master due to the high accuracy of its internal clock.

The AES/EBU serial data format is being adopted increasingly by manufacturers of digital equipment, and with interfaces to this standard provided on R-DAT machines this trend will undoubtedly continue. The format itself supports a number of different classes of data, and has the ability to include a large amount of information about the pedigree of the associated audio.

The DTC implementation of the AES interface takes account only of the emphasis bit, which is used to switch in the (digital) de-emphasis filters automatically. There is no lock-out of consumer data streams, allowing connection of consumer DAT equipment, given a suitably stable wordclock. On the output side, it sets the Professional and Stereo bits, and adds the emphasis bit if pre-emphasis has been selected by the operator.

Besides adopting the AES/EBU standard, the DTC also provides the Sony PCM 1600/1610/1630 serial interface. Being the CD mastering standard, this format is provided not only by CD mastering systems, but also by some external effects units. By providing both these standards, which may be mixed provided the sampling frequency is the same, the DTC may be interfaced to much of the equipment which is presently in use. An optional Mitsubishi X80/X86 interface is available, to allow direct connection to these recorders.

With the advent of disc-based recording and editing systems, increased flexibility and accuracy is available in editing and compiling material. The latest generation of these systems have surpassed the traditional 16-bit limit in I/O stages and recording medium, with 18 bits being offered on the recently announced Digital Audio Research system. By providing 20-bit transfers on its AES interfaces, with the dynamic range extended internally by the use of a floating-point format, the DTC is ready to make use of this and further advances in conversion and storage dynamic range as they become available.

The move to digital has, with present technology, brought with it the need to force-cool audio equipment. To many people, it is not acceptable to have fans running in the control/mastering room, and so there is a trend to provide separate equipment rooms for noisy tape machines and digital racks. This in turn makes it desirable to have control surfaces which may be remote from the main racks, and which require no fan assisted cooling themselves. As has been shown in previous sections, the DTC console meets these requirements, as the rack may be located up to 30 metres away from the console.

## 7. Conclusion

There is an increasing tendency for CD mastering to take place on an international basis. The track running order may need to be changed, or additional tracks may need to be inserted into an album subsequent to the original mastering session. Subtle changes may need to be made to a particular section within an album without affecting the whole.

All of the settings of the DTC, including level changes, crossfades, processing and setup parameters can be programmed and recalled with total accuracy. Stereo matching of processor settings is absolute.

All of these console settings may be stored on a floppy disc, and any subsequent replay from any other DTC, possibly in another part of the world, will be exactly the same as the original master. This gives the producer and record company freedom to utilize the talent and equipment in studios across the world, without encountering the problems outlined above.

The DTC represents an example of the very best in digital technology available today, encompassing all of the requirements for signal processing in one integrated system, coupled with the ability to communicate easily with most professional digital formats in current use.

The DTC has now established itself as a proven system for its application in the most critical area in digital post production, and to date there are some 14 systems in daily commercial use in England, the USA, France, and other parts of the world.

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2. 'Digital Transfer Console Neve Specification' Issue 2.2.

3. 'Digital Transfer Console Operator's Manual' Issue 1.

4. Proudfoot, A. 'The Neve Digital Transfer Console' Pro Sound News, October 1987.

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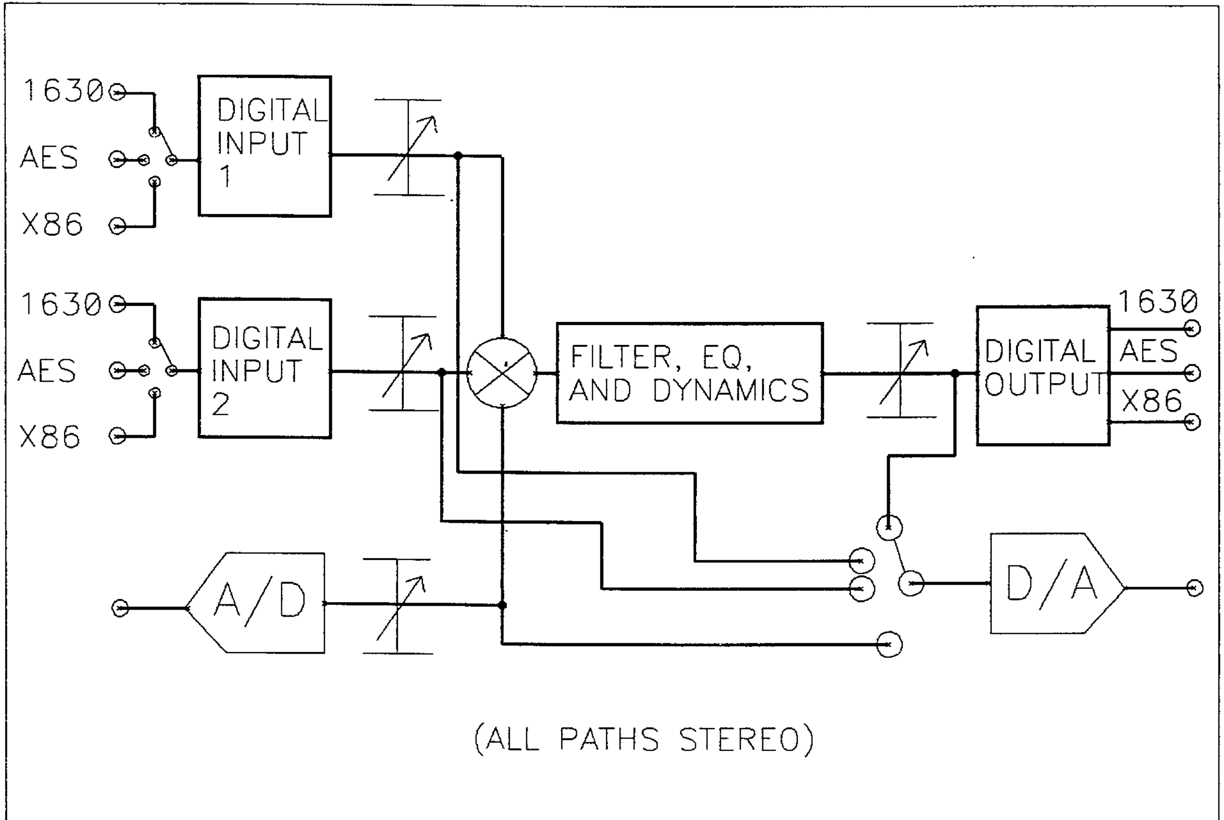


FIGURE 1 – DTC BLOCK DIAGRAM

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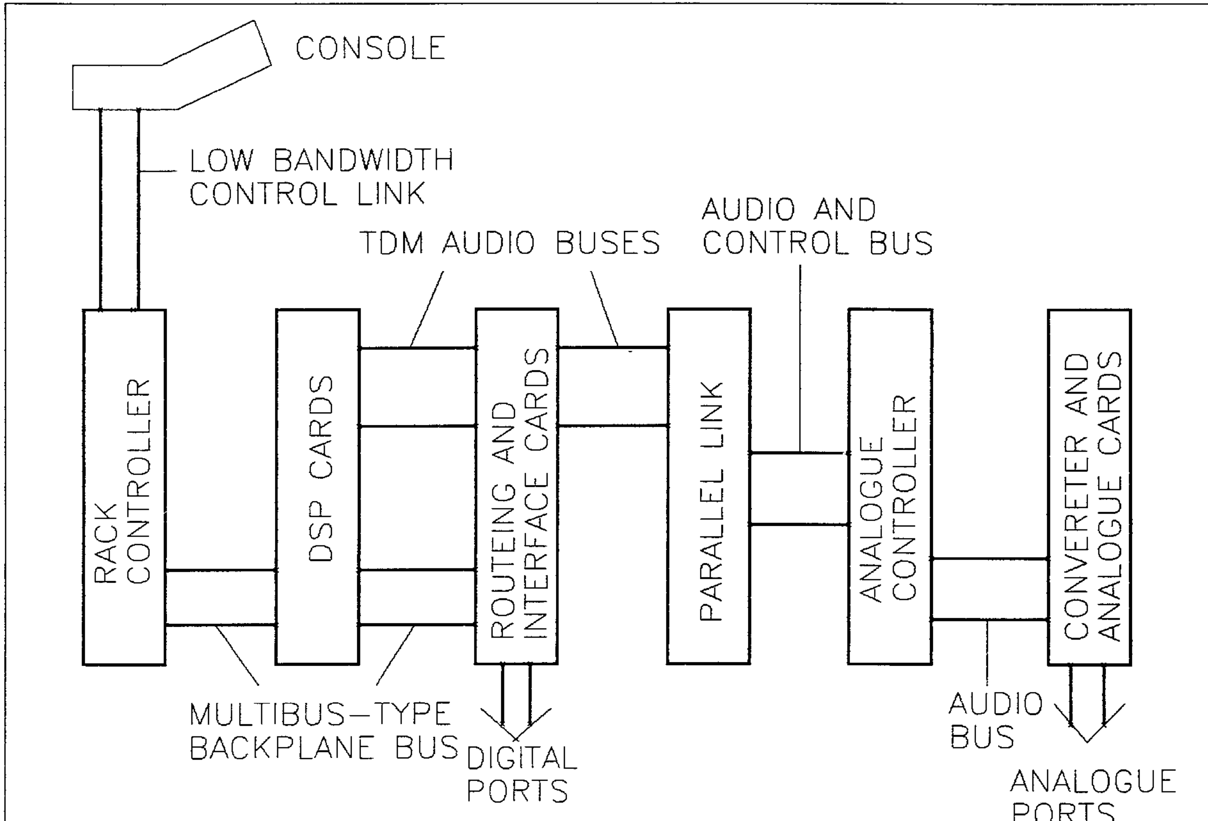


FIGURE 2 –
SYSTEM INTER-CONNECTIONS

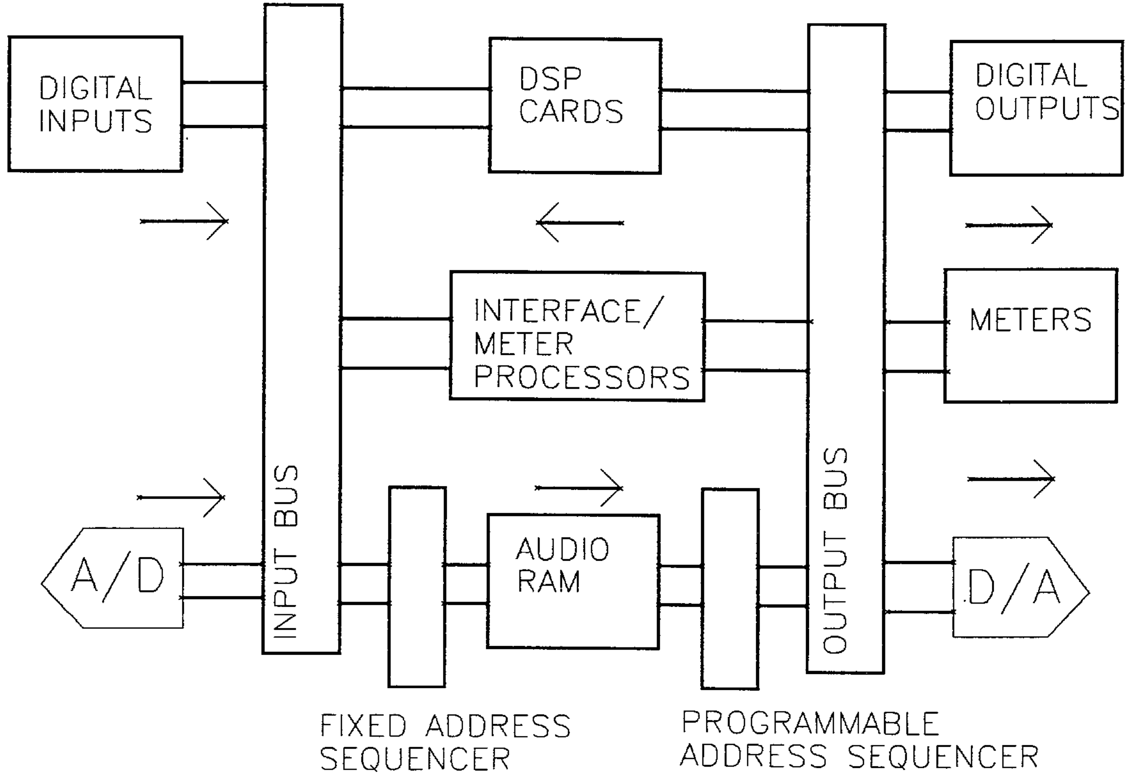


FIGURE 3 – AUDIO SIGNAL ROUTEING

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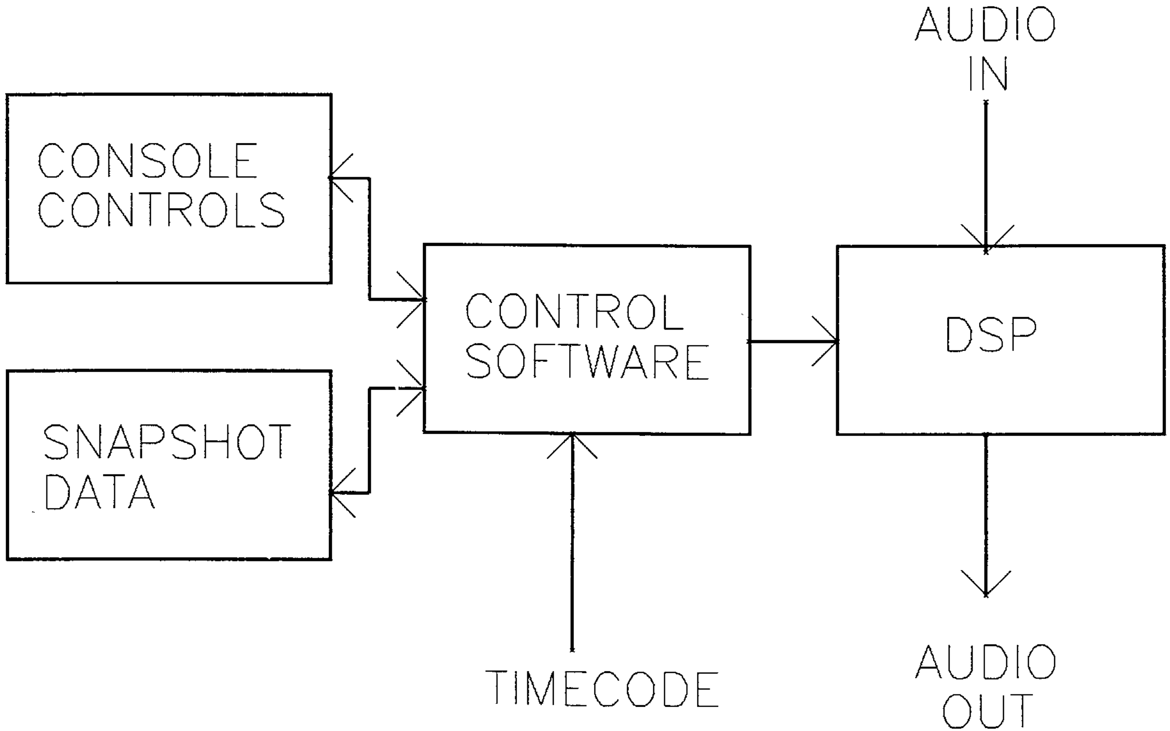


FIGURE 4 — SYSTEM BLOCKS

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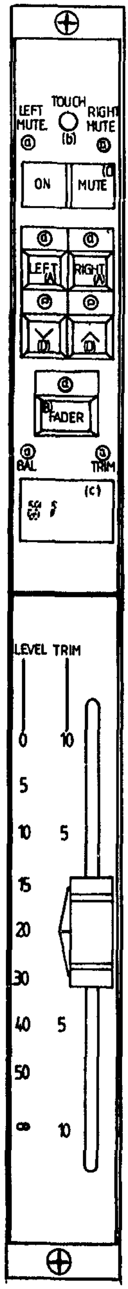


Figure 5 - Fader Module

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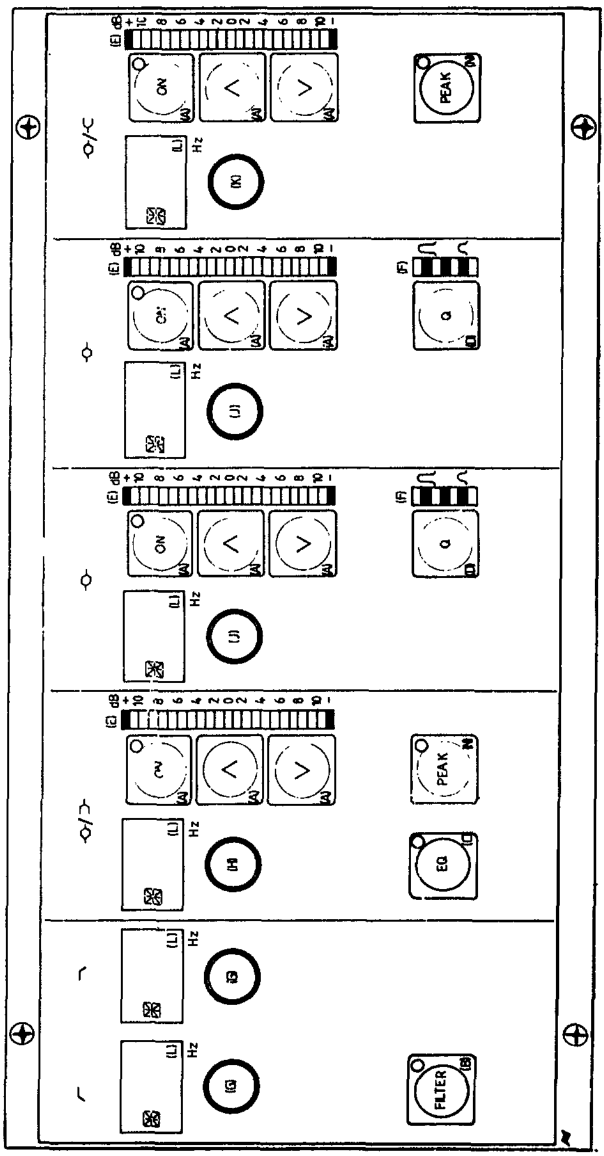


Figure 6 - Filter / Equalizer Module

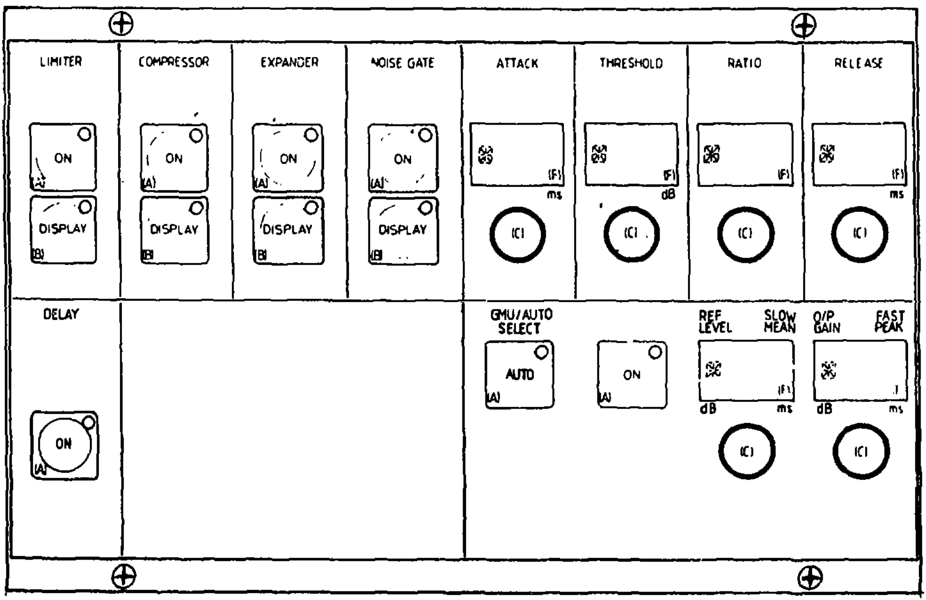


Figure 7 - Dynamics Module

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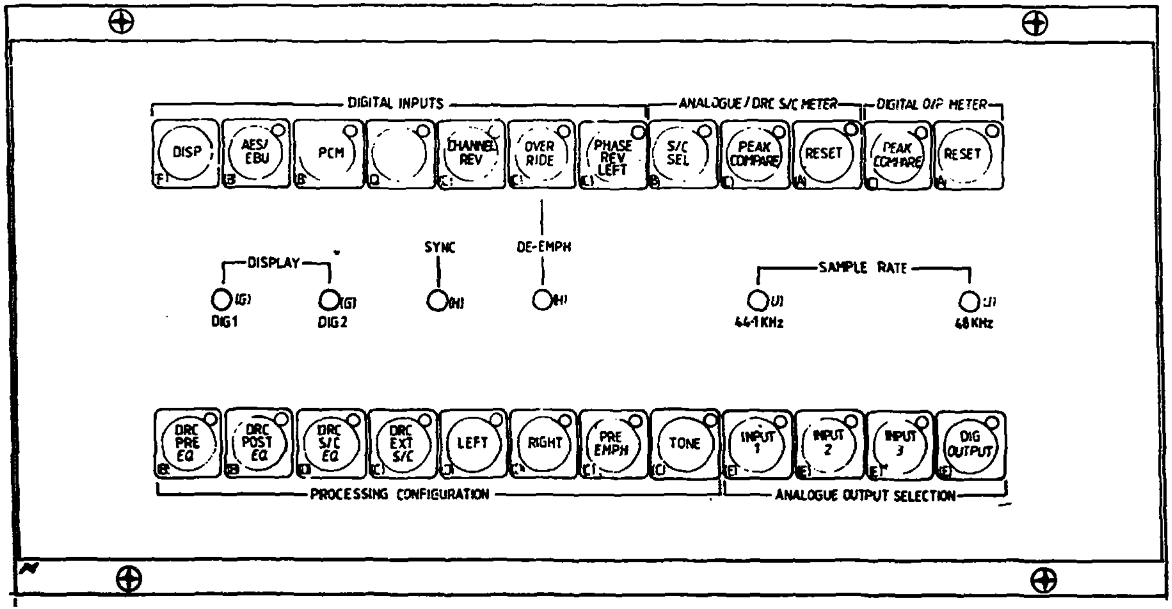


Figure 8 - Configuration Control Module

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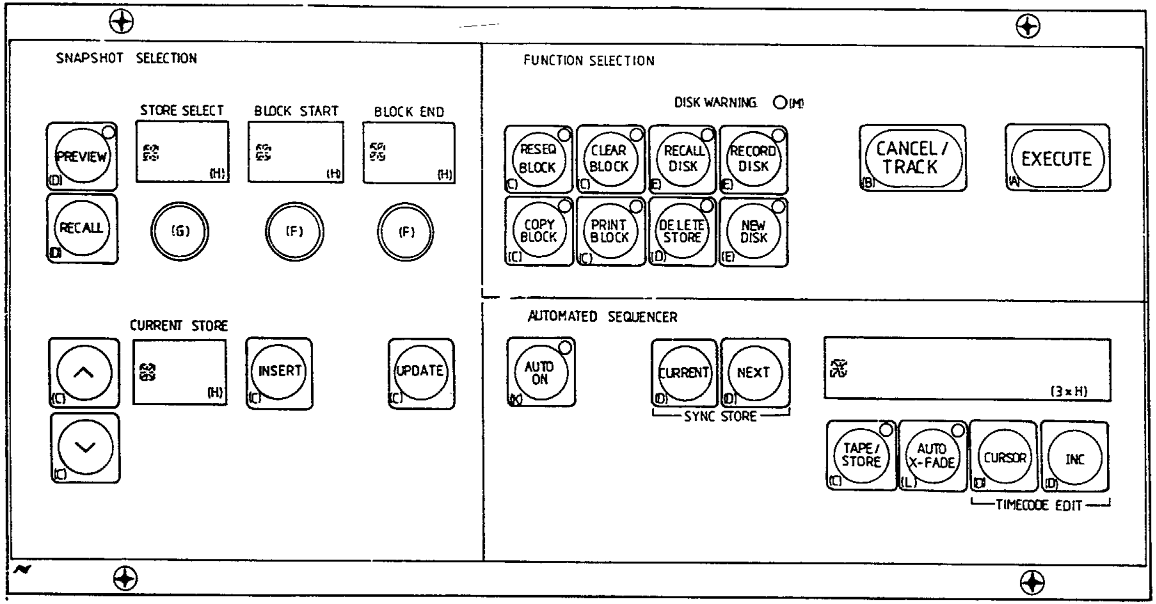


Figure 9 - Snapshot Management Module

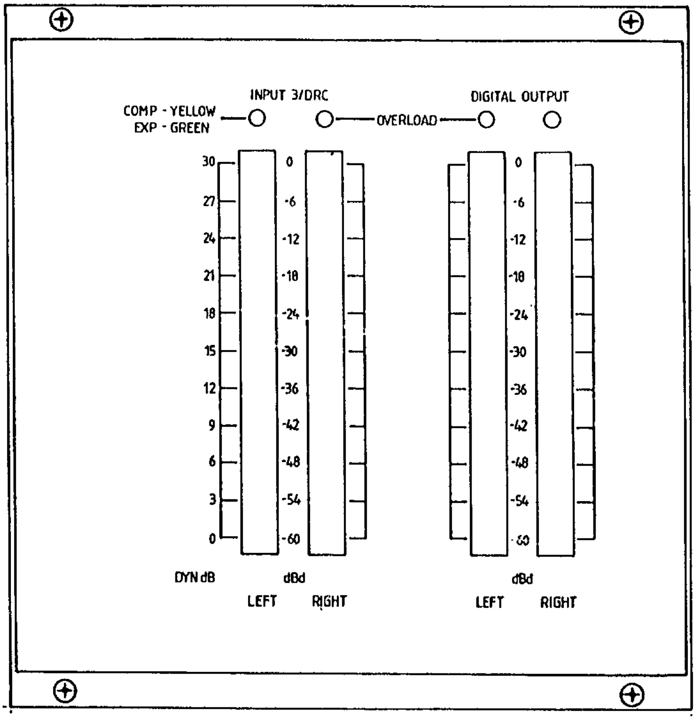


Figure 10 - Meter Module