A DIGITAL MIXER/EQUALIZER FOR MASTERING UTILIZING DSP

by

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1. Abstract

This paper describes the development of a digital audio mixer/equalizer for use in a professional digital audio mastering system with four input channels and two output channels.
This mixer has as good operability as conventional analog audio mixers and has computer-controlled automatic functions. Digital signal processors (DSPs) are employed in the system to achieve digital signal equalization and level control which make it highly reliable as well as providing an excellent cost-performance ratio.
The mixer can also be connected to a limiter/compressor and the AES/EBU digital interface. It allows a wide range of applications including cutting analog and compact discs, digital audio editing, and mixing digital and analog signals such as echo signals.

2. Introduction

The use of PCM technology for production of music software is growing with the increasing popularity of compact discs (CDs) and VTRs with hi-fi sound (hi-fi video). Our commerical PCM mastering system utilizing VTRs developed in response to these trends has already been announced. 1), 2). This system includes a PCM processor, an audio editor, etc. and is widely used not only for cutting analog discs but also in mastering for CDs and hi-fi video. In this mastering process, sound processing including modification of tone and the addition of echo and cross-fades from one tune to another is required. But in the past, sound processing of digital signals has had to be performed after the signals were converted to analog; this conversion resulted in a degradation of sound quality. For these reasons, it became necessary to develop a digital mixer that can process the digital signals as they are without any degradation.
This paper outlines the development of a digital audio mixer/equalizer (DMX) for use in mastering with a 4-channel input and 2-channel output which can be used for a wide range of audio processes in CD and hi-fi video tape mastering.

3. Design concepts

The DMX was designed to have the functions required in two-channel tape and disc mastering. So that the mastering operation can be performed efficiently, the following design concepts were applied.

(1) Operations should be the same as with a conventional analog mixer, giving the same feeling of operation.

(2) Automated operation should be possible under computer control.

(3) The system should have the flexibility necessary for the connection of a limiter/compressor.

(4) In order to improve reliability and cost-performance, DSPs using LSI technology should be used in the digital signal processing section.

4. Applicability

Fig. 1 is a system block diagram of the DMX. Main applications are to be:

(1) Copying: With a PCM processor and 2 VTRs connected to the DMX, it should be possible to perform digital copying while controlling sound quality.

(2) Editing: By adding an audio editor to the system, it should be possible to combine editing with copying with sound processing.

(3) CD cutting: It should be able to perform automated audio processing, switching the equalizer pattern referring to the time code, by adding a work station to the DMX.

(4) Mixing analog and digital signals: In recording classical music, etc. when an echo effect is required, it should be possible to mix using the DMX by sending analog cue send signals to the reverberator and by inputting its return signal to the A/D converter in the PCM processor.

(5) Mixing of 4-channel digital signals: The digital input signals can be either the signals output by the PCM processor or AES/EBU format signals. Four input channels are selected from these; they are then mixed down to 2 channels and output to a device provided with a PCM processor or AES/EBU interface.

5. Composition

As shown in Fig. 1, the DMX consists of a main section and a control section which are connected using an RS-422 cable. The main section consists of equalizer, attenuator/mixer and limiter/compressor digital signal processing modules, digital signal input and output modules and the system control module which controls the entire system.

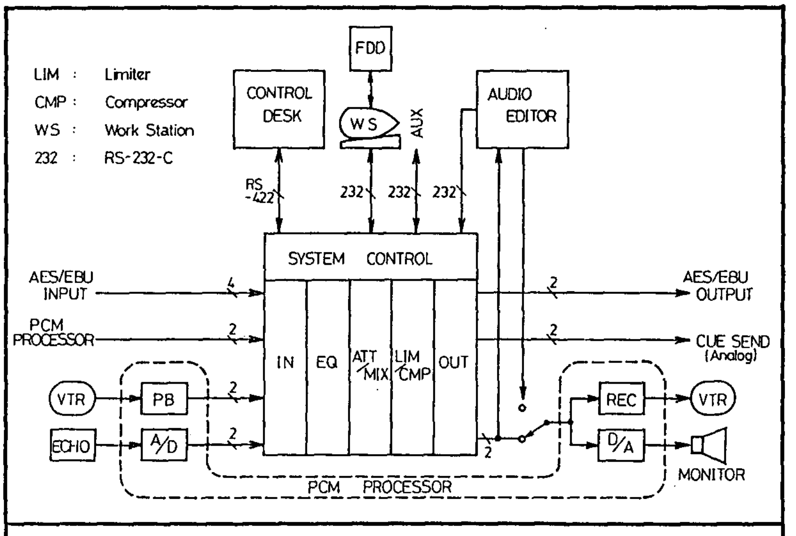

FIG. 1 SYSTEM BLOCK DIAGRAM

5-1. Equalizer module

Fig. 2 is a block diagram of the equalizer module. The PCM signal first goes through the 0 to 20 dB attenuator to prevent overflow in the equalizer module. The signal then passes through the emphasis/de-emphasis equalizer with a time constant of 50 µs/15 µs and then to an equalizer which is divided into four ranges. The specifications of this equalizer are:

High range (1.4 kHz - 16 kHz)

Selectable from the following three patterns:

- o Bell type: G: ± 15 dB, Q: 0.5 - 3
- o Shelving type: G: ± 15 dB
- o Pass type: 12 dB/octave

Mid-high range (600 Hz - 7 kHz)

- o Bell type: G: ± 15 dB, Q: 0.5 - 3

Mid-low range (200 Hz - 2.4 kHz)

- o Bell type: G: ± 15 dB, Q: 0.5 - 3

Low range (30 Hz - 350 Hz)

Selectable from same three patterns as high range.

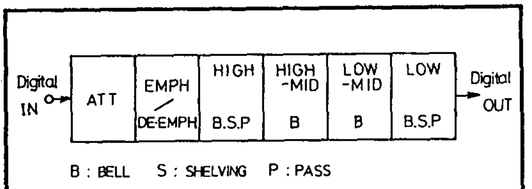

FIG. 2 EQUALIZER BLOCK

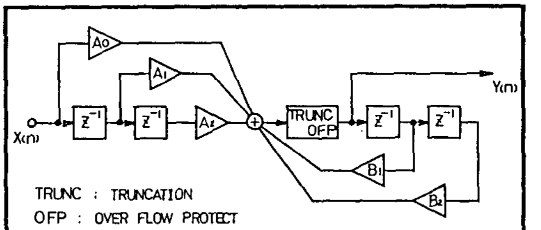

FIG. 3 2'nd order IIR Filter

Fig. 3 shows the composition of the digital filter used for equalizing in each range. The coefficient is determined by a bi-linear Z transform up to 1.4 kHz. Above 1.4 kHz, because it is required to improve the distortion characteristic, two methods, the matched Z transform[3] and method of least squares[4], are compared to see which is more applicable. Problems which appear in hardware design, the coefficient word length and register word length, are resolved as follows[5]:

Coefficient word length: in order to minimize the difference from the desired data characteristics and to obtain stability, 27 bits are used in the low and mid-low ranges and 16 bits in the high and mid-high ranges.

Register word length: in order to ensure the quality of the input signal, this is 39 bits in the low and mid-low ranges and 24 bits in the high and mid-high ranges.

It is important that the digital filters use the concepts shown above for the hardware to be made efficiently. There are two possible design methods; one is the combination of a multiplier, memory and TTL components and the other is to use a DSP. In this unit described in this paper, this second method is used because its simple hardware configuration improves reliability and makes maintenance easier; more important, because it uses a general-purpose device, it will be possible to reduce costs through mass-production.

The DSP device used in the DMX is the uPD77P20. This DSP device incorporates a programmable ROM and serial I/O ports for PCM signals; this makes it easy to increase the processing capability by pipeline processing in which a number of DSP chips are cascade-connected. Fig. 4 is a block diagram showing the equalizer module hardware. The PCM is transferred to each DSP device as a serial 32-bit signal. Respective filter coefficients are input to the DSPs from the control CPU via the control bus. DSPs can perform a command in 250ns; the high and mid-high filters each use 1 DSP and the low and mid-low filters, 2 DSPs.

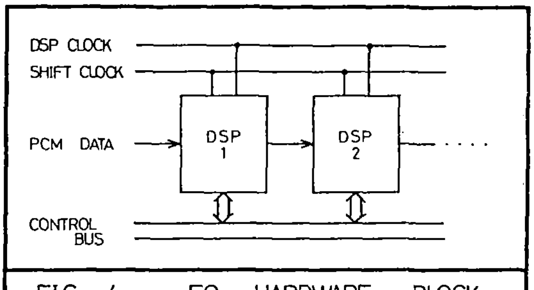

FIG. 4 EQ HARDWARE BLOCK

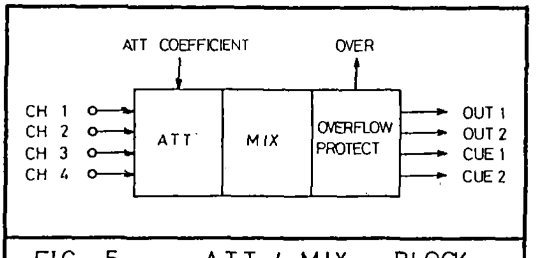

FIG. 5 ATT / MIX BLOCK

5-2. Attenuator/Mixing module

Fig. 5 is a block diagram of the attenuator/mixing (ATT/MIX) module. The four channel signals passing through the equalizer module are processed as shown below to output 2 output channels and 2 cue channels. (F and C are attenuation coefficients.)

$$\begin{pmatrix} \text{OUT } 1 \\ \text{OUT } 2 \\ \text{CUE } 1 \\ \text{CUE } 2 \end{pmatrix} = \begin{pmatrix} F11 & F12 & F13 & F14 \\ F21 & F22 & F23 & F24 \\ C11 & C12 & C13 & C14 \\ C21 & C22 & C23 & C24 \end{pmatrix} \begin{pmatrix} \text{CH } 1 \\ \text{CH } 2 \\ \text{CH } 3 \\ \text{CH } 4 \end{pmatrix}$$

After calculation, the overflow of the output signal is clipped at a maximum value; in this device, soft-clipping is possible by the addition of the limiter/compressor.
The hardware uses the TMS32010 DSP device. This DSP device makes possible 16-bit parallel input and output and is suitable for processing multiple inputs and outputs as shown in Fig. 5.

5-3. Control desk module

This provides the man-machine interface; so that operations are the same as with conventional analog mixer/equalizers, volume controls, faders, switches and level meters are in the same positions as in conventional units. All devices except the level meters are under the control of the CPU and communications with the main section use an RS-422 serial cable.

5-4. System control module

Fig. 6 is a block diagram of the system control module. This gives commands to two 8-bit CPUs, one of which performs the basic processing, changing the filter coefficients, attenuation coefficients and processing modes in response to commands from the control desk. The signal processing mode information is sent from this to the display section of the control desk.
The other CPU controls the whole system by referring to the time codes. The processing required for CD cutting is under the control of the personal computer used as a work station which is connected via an RS-232C interface. This unit automates equalization pattern changes and auto fades in and out. All relevant data is recorded with time codes on a floppy disk by the work station.

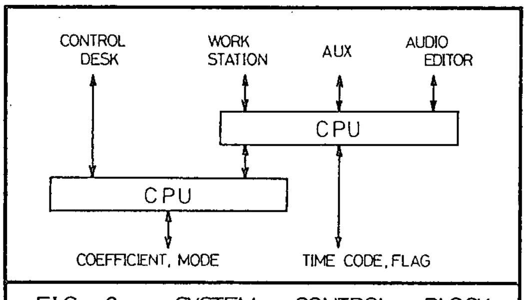

FIG. 6 SYSTEM CONTROL BLOCK

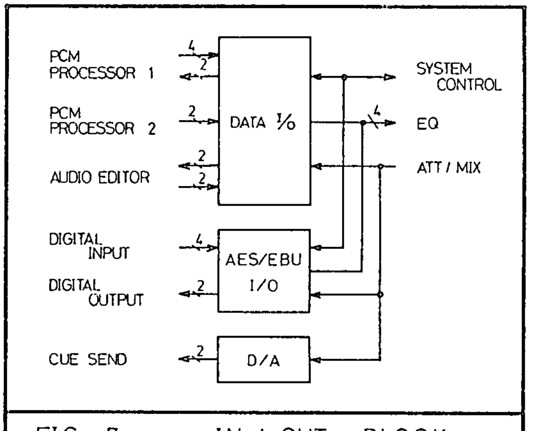

FIG. 7 IN / OUT BLOCK

5-5. Input/output module

Fig. 7 is a block diagram of the input/output (IN/OUT) module. The DATA I/O module has 6 input channels and 2 PCM output channels; these output channels can be switched to pass through the editor when editing. Time codes used in automatic recording and input/output emphasis flags can also be processed.
The AES/EBU I/O module has 4 input channels and 2 output channels for PCM signals; flag input and output are also possible.
A 2-channel D/A module is provided for the cue send signal, making it possible to provide signals to a conventional analog signal processing unit such as a reverberator.

6. Conclusion

This paper has described a mixer/equalizer for mastering digital audio signals as used in CD and hi-fi video systems. The basic design concept is that operations should be similar to those of a conventional analog mixer with automated functions available from a work station; this has resulted in a unit with improved operability. The hardware uses two different DSP devices to improve reliability and cost-performance. The development of this unit will facilitate the production of high quality digital software.
The unit is now being field tested and we are determined to utilize the opinions of users to improve its operability and make it more useful as a mastering system.

References

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(5) Masao Kasuga, "AN IMPLEMENTATION OF A VARIABLE ATTENUATION EQUALIZER WITH DIGITAL FILTERS". Journal of Acoustical Society of Japan, Vol. 39, NO. 4, 1983.