DIGITAL AUDIO EDITOR

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ABSTRACT

As digital audio editing method is required to have equal or better function compared to the analog editing method, several important factors should be solved.

(1) Easy location of edit point simular to the shuttling of the analog tape recorders

(2) High editing accuracy

(3) Smooth and natural signal continuity at the edit point simular to the diagonal tape-splicing of analog tapes

In digital audio system employing VTR (video tape recorder), the electronical editor satisfied the important factors described above is reported in this paper.

1. INTRODUCTION

Editing is an essential process in professional recording.[1] In conventional analog tape recorder, editing is performed by manual splicing of the magnetic tapes. On the other hand in a digital audio system employing video cassette tape recorders,[2][3] editing is not able to do manual splicing. The editing accuracy, method of locating the edit point, and signal continuity at the edit point are not satisfied by the conventional VTR electronical editor.

In a digital audio editor,[4] as editing is done by the electronically selected data in digital-to-digital dubbing, edited tapes maintain the quality of original tape. An ideal digital audio editor should have equal or better functions compared to the analog editing.

For example,

(1) The edit point should be easily located.

(2) Editing accuracy should be higher than 1/100 sec..

(3) Discontinuity of musical signal at the edit point should be eliminated.

(4) Equal functions should be available in rehearsal and in actual editing.

In addition, a big advantage over the analog editing is that the original tapes can be used in digital editing since there is no fear of erasing or damaging the tapes.

The first digital audio editor, DEC-1000(Fig.2) was developed in MAY 1979. After field tests of more than one year, production model DAE-1100(Fig.1) was completed considering user's requirements.

Fig. 3 shows how digital editing is performed. First, decide the fade-in point on the master tape and the fade-out point on the editing tape. Then the desired musical signal is automatically dubbed from the master tape to the editing tape.

As shown in Fig. 4, the editing system consists of two VTRs (one works as player and the other, as recorder), one digital audio processor, and one digital audio editor. DAE-1100; in addition to that another VTR player can be connected in order to locate the next edit point while editing is performed on the other one. Ediitng functions are same in both players.

2. LOCATION of EDIT POINT

In a digital audio processor (PCM-1600, PCM-1610, or PCM-100), slow speed playback is not possible. The DAE-1100, therefore, incorporates a memory circuit to store digital signals in order to ensure easy edit point location by a search dial, an operation similar to that in the analog method in which the tape reels are manually shuttled to search the edit point.

By pressing the EDIT POINT button during playback of a PCM tape, the memory circuit stores the edit point information as well as the digital signals of about six seconds right before and after the edit point. However, storing the entire digital signals for six seconds require quite a large memory capacity. The original signal, therefore, is compressed to a level that the sound quality is kept sufficient for editing purpose. Stored PCM data are reduced from 32 bits ($16^{bit} \times 2^{CH}$) to 8 bits per sampling data, and converted half sampling frequency. The stored data are totally compressed into one eighth.

When the search dial is turned, the turning speed and direction are detected and the stored data are read out. Since the signal data are compressed ones, they are then expanded to a form similar to the original. The digital signal is then applied at the D/A input of a PCM processor (PCM-1600, PCM-1610, or PCM-100) and output is heard through loudspeakers. The block diagram is shown in Fig. 5.

3. EDITING ACCURACY (RESOLUTION)

The editing accuracy of a digital audio system using a VTR has to utilize the VTR frame (1/30 sec.) as a unit. The PCM music signal, however, requires much more accurate resolution than that of a VTR frame unit and, therefore, an address within a frame is required to tell at which part of the tape the PCM signals are recorded. The DAE-1100, therefore, uses SMPTE time code as the reference standard for all editing controls. Since one VTR frame contains 1,470 PCM sampling data, the edit point has to be located by detecting the corresponding frame and word through calculation of the memorized address as explained in 2. "Location of edit point."

The data are recorded in terms of frame on the video tape, and in some types of PCM code, one datum spreads over more than one frame. The edit point, therefore, is controlled through the digital input of the digital audio processor in sampling data form of PCM signal. Fig. 6 shows the relation between the edit point and edit record mode. Old data are re-recorded up to the edit point and new data are recorded from after the same point. Although a resolution level as high as the sampling cycle (22.6μ sec.) can be achieved by this method, the DAE-1100 employs a resolution level of 363μ sec. (equals to 16-word PCM signal) which is high enough to handle music signals

4. SIGNAL CONTINUITY at EDIT POINT

Direct connection of two music signals at edit point will cause discontinuity of signals and create noise as shown in Fig. 7 (a). A cross-fade, therefore, is required to produce smooth and continuous signal and still keep the high editing accuracy.

The cross-fade corresponds to the slant cut of the tapes in analog splicing. It is performed as fading out of the preceding signal and fading in of the following signal. The fade in and out time is called the cross-fade time, which can be set at 10 different steps between 1 m sec. and 99 m sec. on the DAE-1100.

A cross-faded signal, F (t_n) is made by the addition of the fade-out signal, F_1 (t_n) multiplied by a coefficient K (t_n) and the fade-in signal, $F_2(t_n)$ by [1-K(t)] in a multiplier. The coefficient K (t_n) varies with sampling point during cross-fading.

$$F(t_n) = K(t_n)\,F_1(t_n) + [1 - K(t_n)]\,F_2(t_n) \qquad (1)$$

$$0 \leqq K(t_n) \leqq 1 \qquad (2)$$

Fig. 7 (b) is shown smooth and continuous signal at the edit point through cross-fader.

Digital gain control is equiped in order to match the level difference between player and recorder. The gain offset is manually controlled with slide lever on the DAE-1100 keyboard. This lever, just like those on mixing consoles, can also be used to produce a normal fade-in or fade-out effect. Its adjustable range is from + 6 dB to minus infinity.

5. REHEARSAL and EDITING OPERATION

One of the merits in digital editing is that exactly same operation can be repeated in rehearsal and in actual editing. The editing system using the DAE-1100, as shown in Fig. 4 uses only one digital audio processor. Hence, exact synchronization between the two VTRs (player and recorder) is required. In addition, as explained in 3. "Editing accuracy", the memory circuit has to store the data necessary for the cross-fade which cannot be played back by the two VTRs during editing, plus the old data which have to be re-recorded on the editing tape.

The data to be memorized here cannot be compressed like the ones used for manual search. Therefore, the 16-bit, 44.056KHz sampling data are memorized in the same memory circuit used for manual search purpose because the search function is not used during rehearsal and actual editing. Main pair of the PCM data to be stored in the memory prior to rehearsal are those adjacent to the edit point on the editing tape with which the cross-fade as well as the switching of playback video input of the PCM processor are controlled.

The signal paths are same both in rehearsal and in actual editing. Only difference being whether the recorder is in record mode (actual editing) or not (rehearsal). Fig. 8 explains the signal path in editing.

The output of the VTR (recorder) applied to the video input of the PCM-1610, PCM-1600, or PCM-100, is decoded into PCM signals which is then applied to the D/A input for reproduction as musical signals. Reading of the data stored in the memory prior to the editing operation starts when the editing tape reaches the point eight frames before the fade-out point. When the data are applied to the D/A input, the VTR is switched from the recorder to the player and the two VTRs are synchronized so that the fade-out point of the recorder and the fade-in point of the player will coincide. When the fade-out point is reached, the synchronized PCM signal of the player is read and the cross-fade starts. After completion of the cross-fade, the synchronized signal of the player is reproduced as output sound.

In actual editing when recording on the editing tape is required, same PCM signals as those applied to the D/A input are applied to the encoder input. Since the signals for the D/A input are delayed for monitoring purpose, the input to the encoder is applied slightly earlier to realize continuous signal on the tape.

In short, continuous signal is realized by switching the signals in the following order:

Recorder's signal ⟶ Memorized signal ⟶ Synchronized signal of the player.

6. STRUCTURE of DAE-1100

The DAE-1100 consists of the keyboard and the main unit. The keyboard consists of; switch encoder, display data decoder, LED driver, search clock oscillator, and power supply block. The main unit consists of following main blocks, and block diagram is shown in Fig. 9. Controls are mostly covered by the Z-80 micro-computers.

6-1) Digital signal interface

This block interfaces PCM signals between digital audio processor and editor.

6-2) PCM signal processing block

This processes bit reduction or expansion of PCM signals in search mode, cross-fade in editing, and signal level control by fader. Control signals for multiplier and so on are also generated.

6-3) Memory block

Memory block is used to locate the edit point in search mode or to cross-fade, to syncronize PCM signal at the edit point, and to delay D/A digital output behind encoder output to PCM-1610, PCM-1600, or PCM-100 in edit mode.

Memory capacity is 1 Mbits.

6-4) Time code generator and reader

DAE-1100 has one SMPTE time code generator and three readers to use as the absolute reference for editing and autolocating controls. Time code readers employing a micro-computer correct the time code data when dropouts.

6-5) Editing sequence control block

This block provides the controls as a frame unit for VTRs, signal processing block and editing timing control block. It calculates exact editing points by time code and memory address.

6-6) Editing timing control block

The DAE-1100 specifies higher editing accuracy than a frame unit. This block controls the signal processing block as a sampling frequency unit since editing sequence control block controls as a frame unit. Video switchers are also controlled at the accurate timing by the signal made in this block.

6-7) System control block

With all of the keyboard operation, this block makes the controls to provide in the other block. And display data in the keyboard is transmitted through keyboard interface.

6-8) Video signal block

Video signal which is input from players, recorder, or PCM processor is switched by the video selecter, and is output to the PCM processor to reproduce the required musical signal. Video Sync signal is separated to utilize in time code block and editing timing control block.

6-9) VTR remote control and keyboard interface

DAE-1100 is built in remote controller for SONY professional VTR BVU-200B.

Table 1. shows the specifications of DAE-1100.

7. CONCLUSION

We have described a new digital audio editor in coordination with a digital audio processor and VTRs. Compared to the analog editing, the digital audio editing system is complex and expensive. A merit of the electronical editing, however, is that edit point can be decided at ease and editing can be certified in rehearsal without fear of damaging the tape. We feel more natural sound continuity at the edit point than analog editing.

In closing, the authors would like to thank to the members of digital audio division for their advice and cooperation in developing and producing this system.

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[4] T.Ohtsuki,S.Kazami,S.Anju,M.Tanaka,A.Iga; "Electoronical Editing System for PCM Recorder" J.AES 3-5-11 (June 1979)

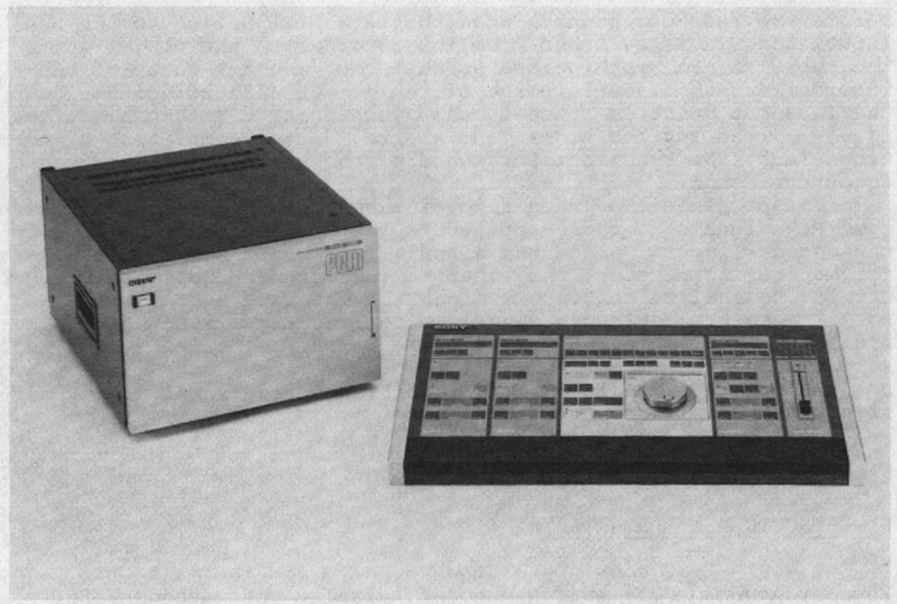

Fig. 1 SONY Digital audio editor DAE-1100(available)

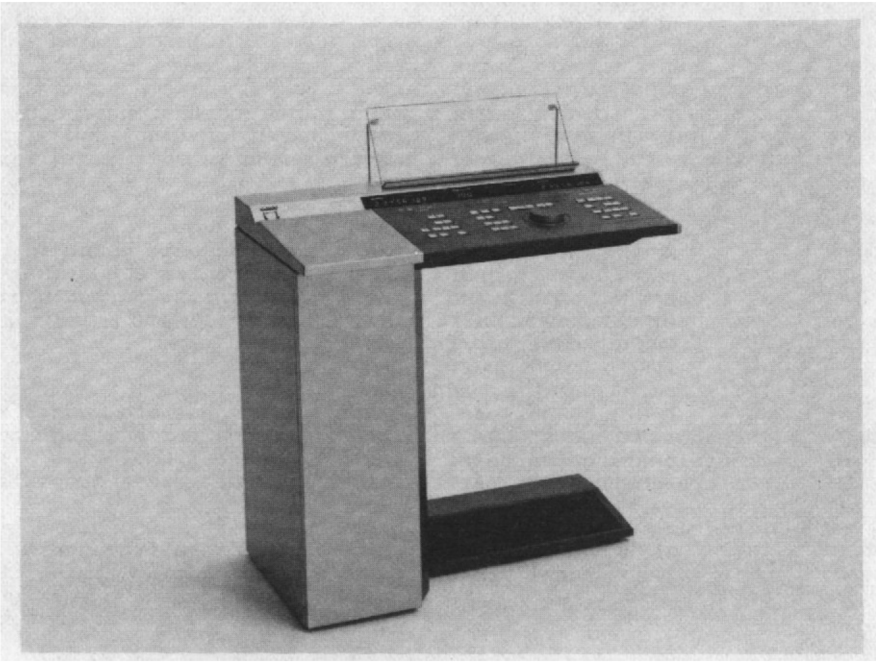

Fig. 2 SONY Digital audio editor DEC-1000

Fig. 3

EDITING METHOD

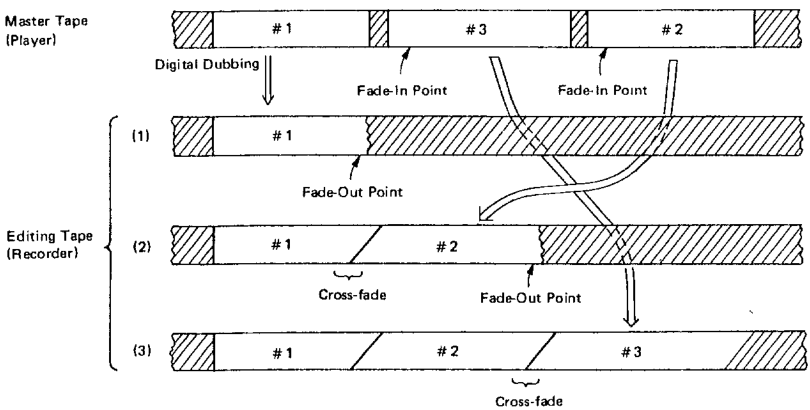

Fig. 4 DIGITAL AUDIO EDITING SYSTEM

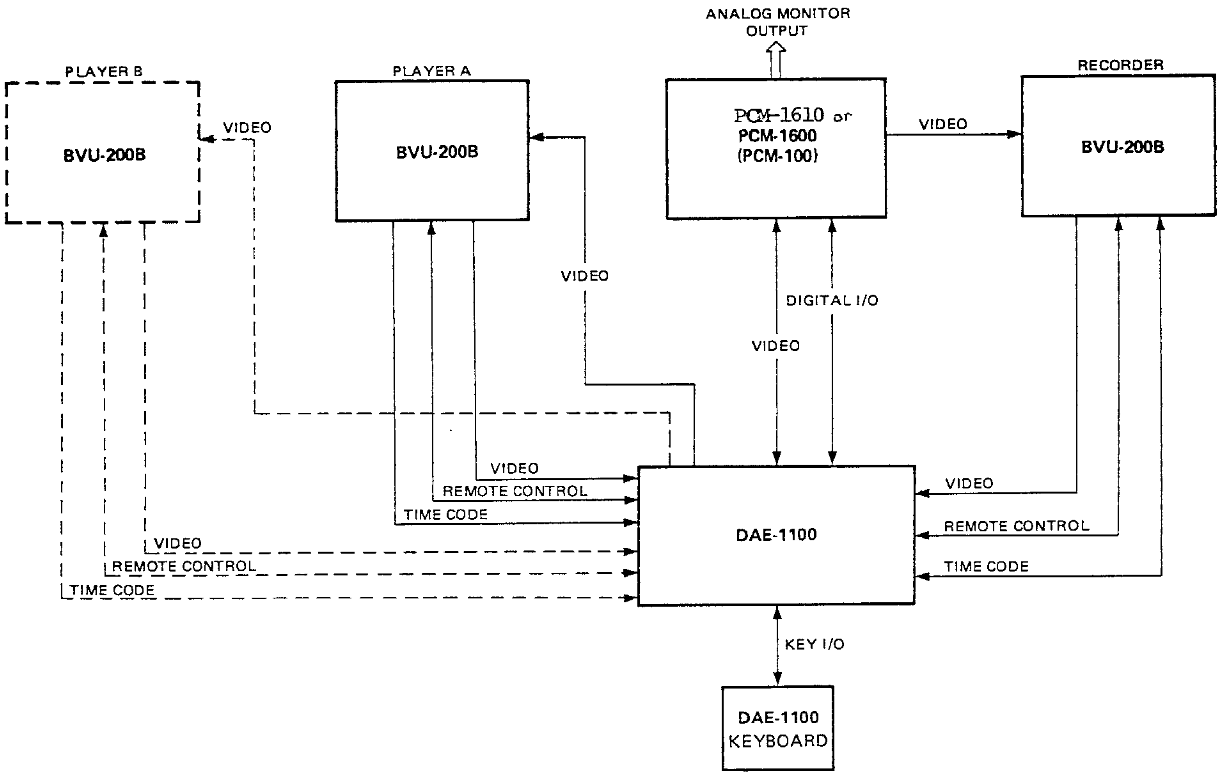

Fig. 5 **SEARCH BLOCK**

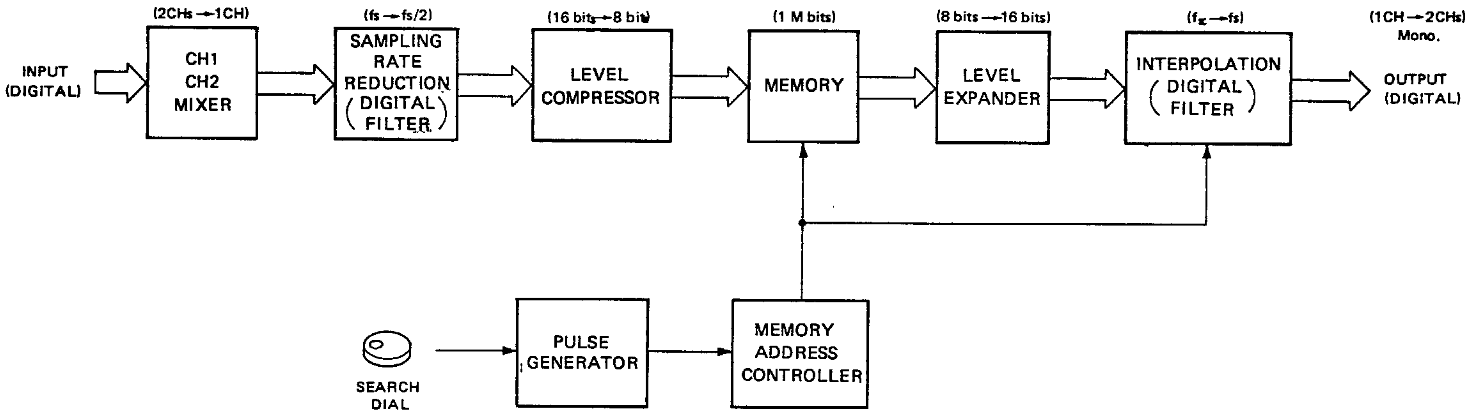

Fig. 6 HIGH ACCURATE EDITING METHOD IN VIDEO SIGNAL

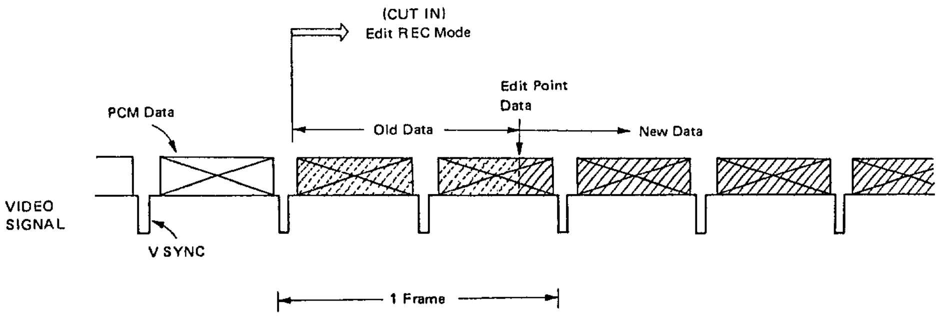

Fig. 7 SIGNAL WAVEFORM AT EDIT POINT

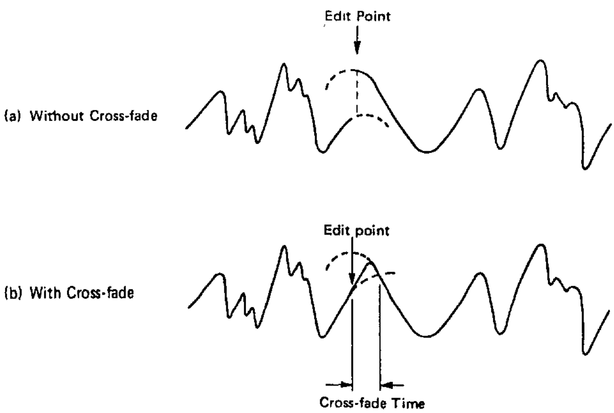

Fig. 8 **SIGNAL PATH IN REHEARSAL OR EDIT MODE**

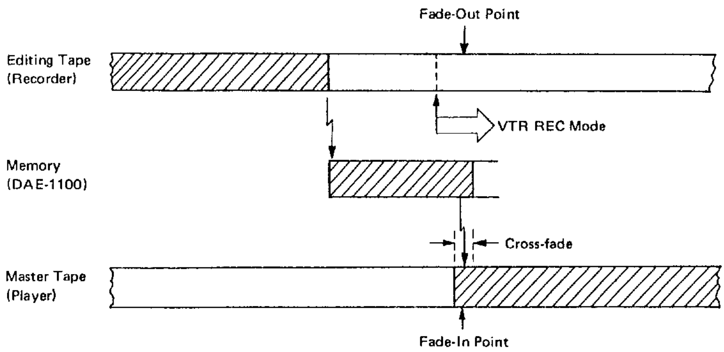

Fig. 9 BLOCK DIAGRAM OF MAIN UNIT

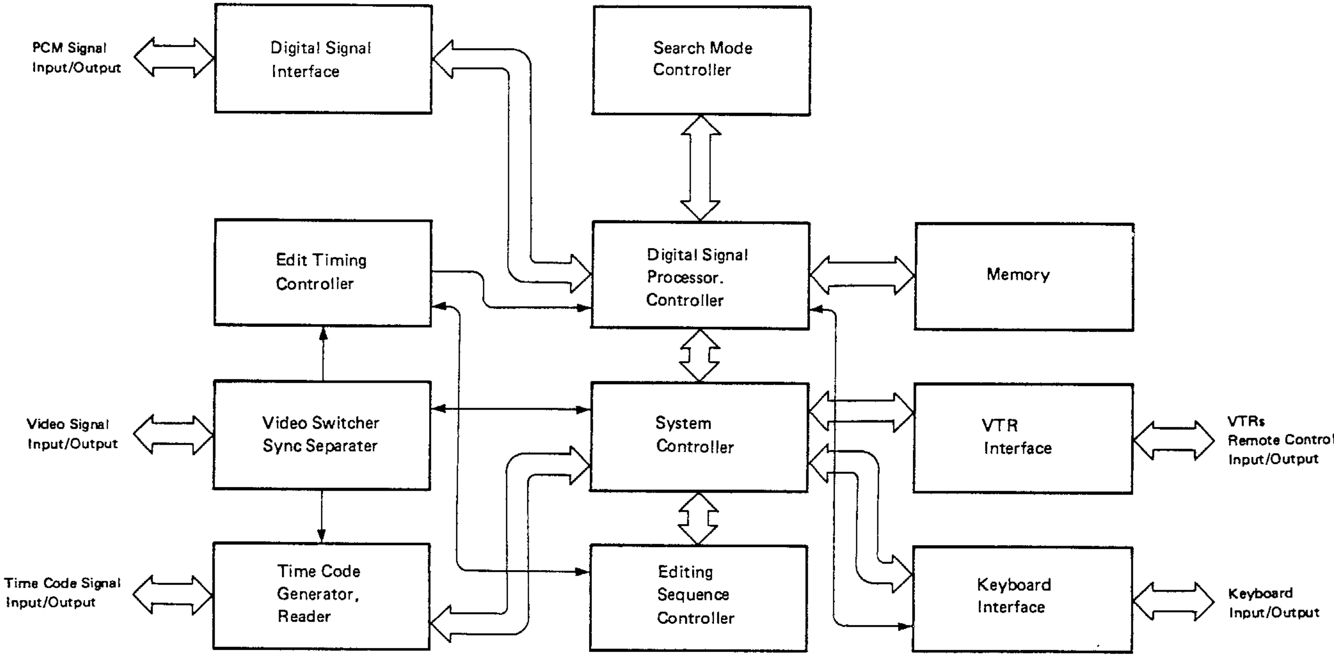

Table 1. Specifications of DAE-1100

Digital input/output	Parallel input/output(for PCM-1610, PCM-1600) 16 bits,2 channels,TTL level 2's complement code Serial input/output(for PCM-100) 75 ohms,unbalanced 1.4 Mbits/sec./CH
Video input/output	Composite video (NTSC) 0.7 Vp-p(data level: 60 IRE) for PCM-1600 and PCM-1610 1 Vp-p for PCM-100 75 ohms, unblanced
Time code input	0 dB, 600 ohms, balanced 0 dB, 10 Kohms, unblanced SMPTE time code
Time code output	0 dB, 600 ohms, balanced 0 dB, 100 ohms, unbalanced SMPTE time code
Remote input/output	TTL level
Tape time counter	00 hour 00 min. 00 sec. 00 frame to 23 hour 59 min. 59 sec. 29 frame
Editing accuracy	363 μsec equivalent to 16 words with the PCM-1610, PCM-1600/ PCM-100
Search mode memory time	5.95 sec.
Cross-fade time	1,2,4,7,10,15,30,50,70,99 msec
Fade level control	+6 dB - $-\infty$
Edit point time offset	Max. 59 sec. 29 frames
Time offset resolution	363 μsec
Power requirements	100-120 V or 220-240 V ac
Power consumption	160 watts
Dimensions	Main unit:428x276x556 mm(W/H/D) Keyboard: 722x 85x385 mm(W/H/D)
Weight	Main unit:Approx. 27 kg Keyboard: Approx. 12 kg