A Professional DAT System

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This paper is concerned with a 2 channel professional DAT system which consists of a studio DAT, an editing controller and a portable DAT. The studio DAT incorporates professional functions, such as Read After Write, recording and playback of SMPTE time code, assemble/insert editing, chase synchronization and variable speed. The editing controller is designed to perform subframe editing. The editing accuracy is +/-30us. In the portable DAT, Read After Write and record,replay capability of SMPTE time code are incorporated, and the low power consumption LSIs enable full recording of a 2-hour DAT tape with a rechargeable battery.

1.Introduction

DAT was originally developed in the consumer field by the DAT conference[1)], and because of its high performance, DAT is applicable to professional use.[2)-(5)] In order to adapt DAT to professional field, it is required that the professional DAT incorporates not only consumer use functions but also professional use functions, because the standards such as time code and digital input/output formats are different between professional and consumer equipment, and the functions such as editing and synchronization are not included in a consumer DAT. There are some important factors which must be taken into account.

The first point is recording and playback of SMPTE/EBU time code which are widely used in broadcasting and production studios. It is required that the recording format should be standardized. The time code recording format was discussed and formed in IEC SC60A/WG16 Japan National Committee. This draft standard was submitted to IEC/Paris in February of 1989 and the discussion has started. Our DAT system employs the draft standard (IEC/Japan)[6)] and maintains the input/output compatibility with other professional equipments. The compatibility with consumer equipments is also important. In broadcasting and production studios, it is required that pre-recorded consumer tapes are played back in professional players, and also it is desired to play back professional tapes on low-cost consumer DAT decks. This implies that some cross-compatibility between consumer and professional format be required. Furthermore, it is necessary that consumer tapes are played back in a professional DAT without an external adapter or increase of hardware cost. Our professional DAT system satisfies these requirements.

The second point is to be able to edit and synchronize with high accuracy. In tape to tape editing, the accuracy is sometimes required to be less than 1/10 of a frame in applications such as spot erase and alteration of notes. Because of the differences in frame rates between DAT and SMPTE/EBU time codes, there is a difficulty in frame to frame matching. In order to achieve high accuracy, there are two basic requirements: One is to determine an edit point precisely. The other is to synchronize with a master equipment precisely. The studio DAT is designed to satisfy these requirements and has achieved high accuracy of +/-30ms (about 1/1000 of a SMPTE frame period) in both editing and chase synchronization.

For the user's convenience, we take the following things into account. The studio DAT includes editing capability and synchronizing capability for audio-for-video use. Furthermore the studio DAT incorporates memory check and rehearsal using solid state memory to rehearse many times quickly. In a portable DAT, light weight mobility and long battery life are necessary along with features such as Read After Write, and SMPTE/EBU time code record,replay capability. Helical scan DAT withstands shocks and vibrations compared with stationary head taperecorder, and it is suitable for a portable equipment. The decrease in power consumption and weight is the key to achieve a portable professional DAT.

In the following sections, we will explain the techniques employed in the professional DAT system and the measurement results of the accuracy with respect to editing and chase synchronization.

2. DAT format and recording method of time code

2.1 DAT format

Fig.1 shows the tape format of DAT. A pair of plus azimuth track and the following minus azimuth track form a frame. The optional track 1 and 2 are on the lower and upper edge of a tape. The part that contains music signal is Main data. Sub data area 1 and 2 in two locations before and after the Main data contain Sub data information such as program number and DAT time code. ATF is used as a tracking signal for servo control during playback.

2.2. Recording method of time code

SMPTE/EBU time code consists of 80 bits per frame. The information to be recorded is time code, user's bits and some other flag bits. When SMPTE/EBU time code is

Fig. 1: Tape format of DAT

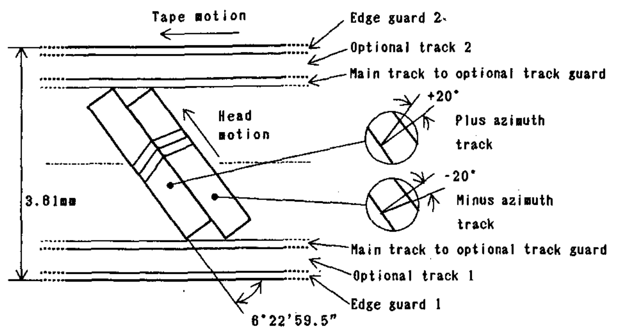

Fig. 1a: Track configuration (View on Magnetic sensitive side)

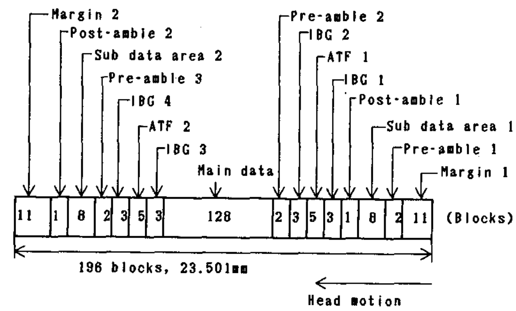

Fig.1b: Signal allocation on a track (Normal track pitch mode)

recorded on a tape, a time code with DAT frames is preferable for the following reasons.

1)In order to realize the compatibility between consumer and professional tapes, it is better to design a professional time code format close to a consumer DAT format.

2)Because the internal signal processing of a DAT is carried out on the basis of a DAT frame, it is advantageous to use a DAT frame rather than to use a SMPTE/EBU frame for editing and synchronization process.

Furthermore, it is necessary to record a timing informa-

Table 1: Ratio of professional time code and DAT frame rates

TIME CODE		RATIO (TIME CODE : DAT)
SMPTE	29.97Hz	900 : 1001
	30Hz	9 : 10
EBU		3 : 4
Film		18 : 25

tion which relates SMPTE/EBU time code frame to DAT time code frame for the following two reasons.

1)A DAT frame cannot be specified by a single SMPTE/EBU frame number, because the frame period of SMPTE/EBU is longer than that of DAT as shown in Tab.1.

2)The phase relation between a SMPTE/EBU time code frame and a DAT frame must be recorded and reproduced precisely.

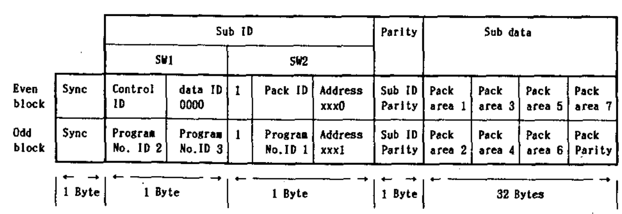

Fig. 2: Sub data block format.

The time code recording method is based upon the IEC SC60A/WG16 draft standard (Japan). SMPTE/EBU time code is converted into professional DAT time code and is recorded in the Sub data area in the form of Packs. Fig.2 shows Sub data block format. When it is played back, the professional DAT time code is converted into SMPTE/EBU time code. The professional DAT time code carries the time code with DAT frames and the above timing information with the resolution of the order of a sample.

3.Configuration of the studio DAT

3.1Signal flow of PCM

Fig.3 illustrates the block diagram of the studio DAT. The studio DAT consists of mechanism, servo control, signal processing, trim memory and digital crossfader, analog input and output, digital input and output, time code processor and system control microprocessor blocks.

In the case of editing, the playback PCM data which is pre-read from a tape with a pair of advanced heads crossfades to the input PCM data at an edit point. Both PCM data are recorded with a pair of trailing heads and the recording begins several frames before the edit point. The signal picked up by the advanced heads is amplified and waveform-shaped in RF circuit, and it is applied to the signal processing LSI-2 which decodes PCM and Sub data after 10-8 demodulation. The errors contained in PCM data are corrected by two-dimensional (C1/C2) Reed Solomon error correction strategy and the uncorrected errors are concealed by an average interpolation, then the PCM data is transferred to the trim memory-2. At the same time Sub data including the time code information, Sub-IDs and other parameters are sent from the signal processing LSI-2 to the system control microprocessor.

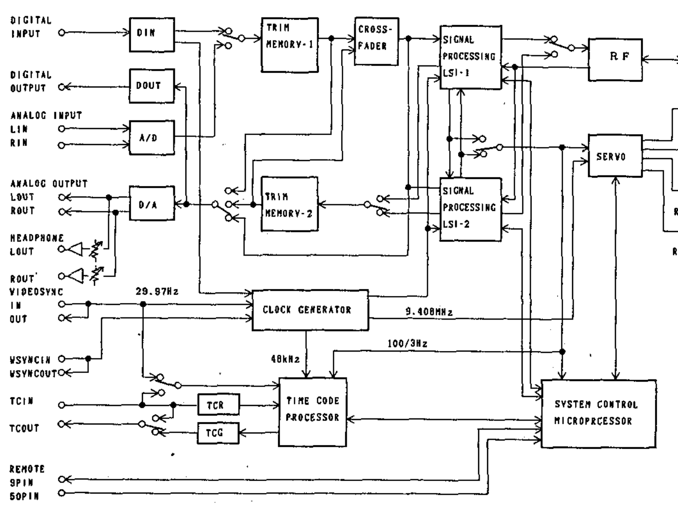

Fig.3: Block diagram of the studio DAT

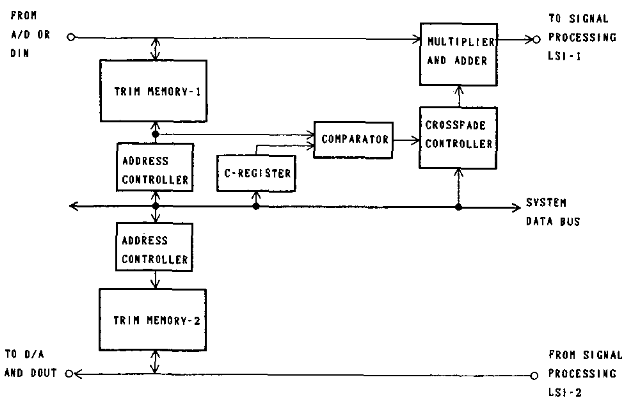

Fig.4: Block diagram of the trim memory

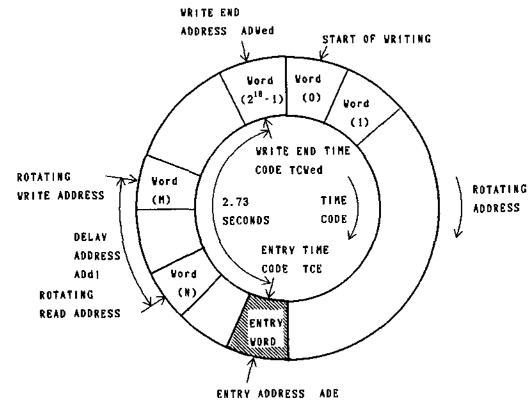

Fig.5:Trim memory and time code

An input analog signal is converted into digital form (PCM data) by a high resolution oversampling Analog to Digital (A/D) converter utilizing multistage noise shaping (MASH) modulation techniques. The PCM data is fed to the trim memory-1.

Both PCM data from the trim memories-1 and 2 are applied to the digital crossfader. The output from the digital crossfader is applied to the signal processing LSI-1 which encodes PCM and Sub data followed by 8-10 modulation; C1 and C2 parities for the Reed Solomon error correction code are attached to the PCM data. The Sub data includes the time code information, Sub-IDs and other parameters which are supplied from the system control microprocessor. The encoded signal from the signal processing LSI-1 is fed to RF circuit and then to the trailing heads.

3.2 Signal flow of time code

Time code conversion is performed by the following. In recording, the incoming SMPTE time code is demodulated in Time Code Reader (TCR). The time code read by TCR is transferred to the Time Code Processor(TCP). In the TCP, the input SMPTE time code is sampled by the DAT frame reference signal, the period of which corresponds to one frame of the PCM data. The external frame sync signal is selectable between either the input video sync signal (composite or black burst) and the input time code. The SMPTE time code sampled in the TCP is converted into the professional DAT time code which is sent to the system control microprocessor. In playback, the reproduced professional DAT time code is transferred to the TCP via the signal processing LSI-2 and the system control microprocessor. In the TCP, the professional DAT time code is converted into the SMPTE time code.

4.Trim memory and crossfader

4.1 Configuration

The purposes for using the trim memory are as follows.

1) Memory check and rehearsal to hear the edit played out of memory up to and away from an edit point.

2)Providing timing delays for the PCM data to precisely synchronize with an external time code.

The crossfader is used to join together successive programs without click noises. Fig.4 illustrates the block diagram for the trim memory and the crossfader. The memory size of 16 Mbits allows storage of 5.46 seconds PCM data for each of the trim memories-1 and 2. Because the write and read address space is 2^{18}, and the memory size is 4 Mbits per channel respectively, the resolution of address is a 16 bit-word (1 sample).

4.2Memory check and rehearsal

The trim memories-1 and 2 are ring memories as shown in Fig.5. In memory check and rehearsal, the memorized PCM data can be read out up to and away from the entry time code at which the content of a recorded tape is altered by recording new material. So it is very important to obtain the accurate entry address corresponding to the entry time code. The process to determine the above entry address is as follows. In normal playback, the reproduced PCM data is continually written into the trim memory-2 by the rotating write address ADW and is read out from the trim memory-2 by the rotating read address ADR which is equivalent to ADW. This means that the delay address ADdl is set zero. When an entry time code is given, address writing is stopped about 2.73 seconds after the entry. The entry time code is converted into the professional DAT time code denoted as TCE. The write end address and the corresponding write end time code are denoted as ADWed and TCWed. When TCWed and TCE are expressed in units of seconds, the entry address ADE can be calculated by ADWed, TCWed and TCE by the following equation.

$$ADE = ADWed - INT[(TCWed-TCE)/(1/48000)] \quad (4.1)$$

Because the resolution of the professional DAT time code is of the order of 1 sample and one increment of address corresponds 1 sample, the entry address ADE can be obtained with the accuracy of 1 sample.

4.3 Timing delay of PCM data in synchronization

The process for giving a timing delay is as follows. The delay address ADdl between a read address and a write address is determined according to the time difference between the reproduced time code and an external time code. Because the delay address is calculated by using the professional DAT time code with the accuracy of 1 sample, the reproduced PCM data can be delayed by 1 sample. This method is used in the synchronization process in section 5.3.

4.4Crossfader

The edit point is defined as the start of crossfade. In Fig.4, the crossfade address is calculated by using the professional DAT time code and set in C-register. The comparator generates a crossfade start flag when the read address ADR of the trim memory-1 coincides with the crossfade address in the C-register. The crossfade start flag triggers crossfade. The crossfade time ranges from 0 to 160ms. The process to determine the crossfade address is described in section

5.Edit

5.1 Edit mode

In assemble editing, all the data (Main data, Sub data and ATF) are overwritten at the same time. In insert editing, any desired data except for ATF can be edited, which means that audio data insert and Sub data (time code and Sub IDs) insert are supported. In insert editing, since recording is performed referring to the previously recorded ATF signal, a tape on which ATF signal has been already recorded must be used. The internal process of the editing and synchronization is described below.

5.2 Editing process

The master equipment performs editing and supplies the master time code to the slave equipment. In this system, the Recorder is the master and the Player is the slave. The Player chase synchronizes to the master time code supplied from the Recorder. The signal flow diagram for digital tape to tape editing using studio DATs and an editing controller is shown in Fig.6 . The PCM signal is supplied from the digital output of the Player to the digital input of the Recorder. So, the Recorder is clock-synchronized to the Player. Fig.7 shows the algorithm for editing. In step 1, the entry SMPTE time code is converted into the professional DAT time code. In step 2 and 3, the crossfade address is calculated and is set in the C-register several frames before the edit point. Let the reproduced time code corresponding to the rotating read address ADR be TCC. The crossfade address is calculated according to the time difference between the entry time code TCE and the time code TCC. When TCE and TCC are expressed in units of seconds, the crossfade address ADcf is obtained as:

$$ADcf = ADR + INT[(TCE - TCC) / (1/48000)] \quad (5.1)$$

The crossfade address ADcf can be obtained with the accuracy of a sample.

The output SMPTE time code from the Recorder must be frame-synchronized to the DAT frame. The frame-synchronization of time code means that the phase relation between the DAT frame and the SMPTE frame is reproduced as it is recorded. As it is mentioned in section 2.2, the phase relation between the DAT frame and the SMPTE frame is obtained from the timing information of the professional DAT time code. The timing of the synchronization bits of the output SMPTE time code from the Recorder is determined on the basis of the above phase relation.

5.3Synchronization process

The synchronization process is that the slave DAT synchronizes the PCM signal at the digital output terminal to the master time code with the time code difference between the edit points of the master and the slave equipments. The synchronization is performed in the Player. Fig.8 shows the synchronization method. After prerolling to points several seconds before edit points, both the Recorder and the Player run forward and go into play mode. Ideally the Player must run ahead of the Recorder by time (TEPpl TEPre), where TEPpl and TEPre are the entry time codes for the Player and the Recorder respectively. In order to explain the synchronization process, we consider a time code which is exactly in line with the PCM signal. Let TDOpl(t) be the time code in line with the PCM signal at the digital output terminal of the Player, and TCOre(t) be the master time code from the Recorder. The ideal relation between TDOpl(t) and TCOre(t) is :

$$TDOpl(t) = TCOre(t) + TEPpl - TEPre \quad (5.2)$$

Due to the limitation of tape transport tolerances, it is generally difficult to satisfy the above eq.(5.2) only by mechanical ways. An alternative is utilizing memory. In our system, the Player runs ahead of the Recorder by time (TEPpl - TEPre + TOF) where TOF is an additional offset time. Let TDTpl(t) be the time code in line with the PCM data from the signal processing LSI-2 in the Player. Then

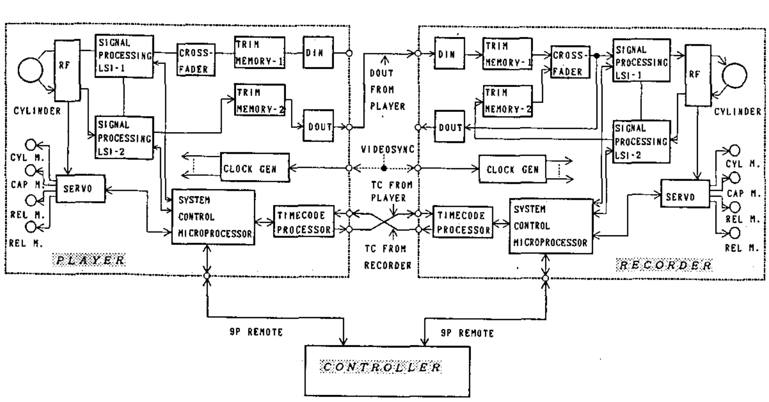

Fig.6: Signal flow in editing by an editing controller

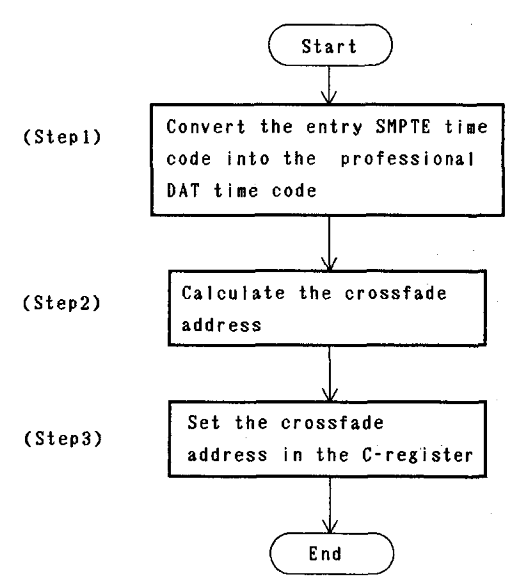

Fig.7: Flowchart for edit

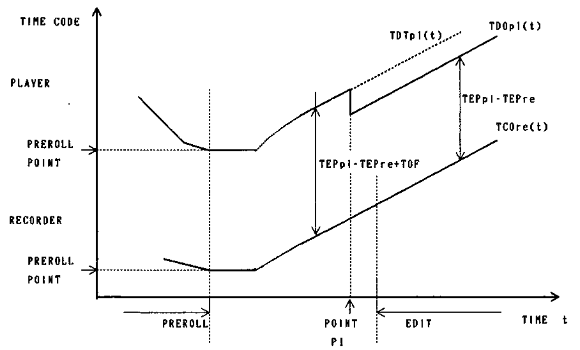

Fig.8: Synchronization between Recorder and Player

TDTpl(t) is expressed by the following equation.

TDTpl(t) = TCOre (t) + TEPpl - TEPre + TOF (5.3)

The PCM data is delayed in the trim memory-2 by TDL at point P1. Furthermore the time delay of the DOUT block must be considered. Let the time delay of the DOUT block be TTD, then TDOpl(t) is expressed as:

TDOpl(t) = TDTpl(t) - TDL - TTD (5.4)

And if

TDL = TOF - TTD (5.5)

then eq.(5.4) is equivalent to eq.(5.2). The editing accuracy depends on the accuracy of TDL. So TDL is precisely adjusted by the trim memory so as to satisfy eq.(5.5). When TOF and TTD are expressed in units of seconds, the delay address ADdl is calculated as:

ADdl = INT [(TOF - TTD) / (1/48000)] (5.6)

The address ADdl is calculated with the accuracy of 1 sample.

The above TOF is expressed as the sum of an integral multiple of DAT frames and the timing information. In order to simplify the calculation of ADdl, a frame synchronization technique is used. In this technique the DAT frame reference signal of the Player is frame-synchronized to the incoming master SMPTE time code. The counter of the signal processing LSI-2 which generates the DAT frame reference signal is reset several seconds before the edit point on the basis of the timing information. By this frame

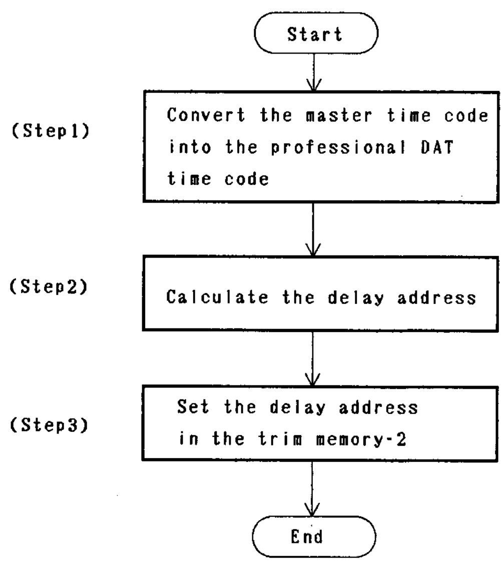

Fig.9: Flowchart for synchronization

synchronization, TOF is expressed as an integral multiple of DAT frames. This makes the calculation of ADdl easy.

Fig.9 shows the algorithm for the Player process. In step 1, the master time code is converted into the professional DAT time code. In step 2, the delay address ADdl is calculated and in step 3 the delay address ADdl is set in the trim memory-2 as offset.

5.4Deck to deck editing

In deck to deck editing, a 9pin cable connects the Recorder and the Player. The Recorder becomes a master DAT. The internal process is basically the same as the editing by the controller except that the commands are sent from the Recorder instead of the editing controller.

6.Chase synchronization

6.1Chase mode

Two types of chase synchronization, mode 1 and mode 2 are supported. In mode 1, the slave DAT locks to a master time code supplied from a master Video Cassette Recorder (VCR) or DAT at one point after which the slave DAT runs in play mode. In mode 2, the slave DAT chases to the master time code continuously. If the master equipment runs in fast forward/fast rewind/play modes, the slave DAT in mode 2 automatically changes it's speed and runs in fast forward/fast rewind/play modes. The slave DAT in mode 2 also chases to the master VCR in shuttle mode.

6.2Chase mode 1

Fig.10 shows the process of chase synchronization mode 1. The DAT chases to the master time code in fast forward mode. The professional DAT time code converted from the master SMPTE time code is compared with the reproduced professional DAT time code, then the difference between the two is calculated. When the reproduced time code approaches to the master time code, the DAT runs in cue/review and 1.2 times speed modes. As soon as the DAT overtakes the master time code at point Q1, it runs in play mode. The time code TDTpl(t) in line with the PCM data from the signal processing LSI-2 is ahead of the master time code by time TOF. At point Q2, TDTpl(t) is expressed as :

TDTpl(t) = TCOmr(t) + TOF (6.1)

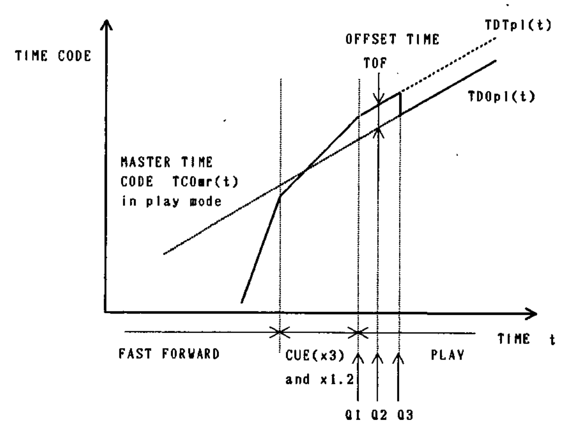

Fig. 10: Process of chase synchronization

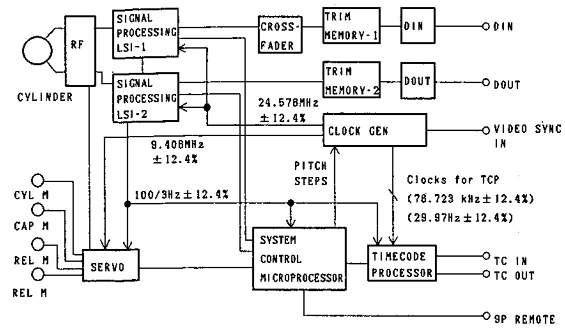

Fig. 11: Block diagram of variable pitch control

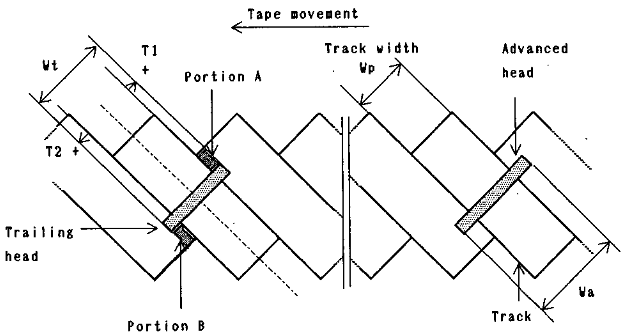

Fig. 12: Relation between tracks and heads

where TCOmr(t) is the master time code. After point Q3, the PCM data is delayed in the trim memory-2 by time (TOF - TTD), then we obtain :

TDOpl(t) = TCOmr(t)   (6.2)

The theoretical limitation for the accuracy is of the order of a sample.

6.3Chase mode 2

In chase mode 2, when the master equipment changes it's speed, the DAT runs in cue, review and fast forward/rewind mode to chase to the master time code. Although the tape speed variation steps are not the same between a VCR and a DAT, the slave DAT chases to the master time code on average. When the master equipment is in play mode, the slave DAT is also set in play mode, and chases to the master time code by variable pitch control if the time difference between the master and the slave equipments exceeds 0.4ms. The minimum step of the variable pitch is +/-0.1% so there are no audible wow and flutter in reproduced sound.

7.Variable pitch control

The range of variable pitch control is +/-12.4%. The block diagram for playback is shown in Fig.11. The clock generator supplies the master clock of 24.576 MHz to the signal processing LSIs and the clock of 9.408 MHz to the servo block which utilizes 9.408 MHz as the count clock for the Frequency Generator and the Pulse Generator of drive motors and ATF sync detection. The sampling clock Fs and the DAT frame reference signal 100/3 Hz are generated in the signal processing LSI-2. The clock generator also supplies to the TCP the clocks of 76.723 kHz for the bit clock and the internal video frame synchronization signal. In the variable pitch mode, it is necessary to change 9.408 MHz and the TCP clocks in proportion to the change of the master clock. The system control microprocessor sends the pitch information to the clock generator.

8.Mechanism and servo control

8.1Four heads cylinder assembly and mechanism

Both accuracy and reliability are raised by the mechanism which employs an independent Direct Drive motor for each of the capstan, the 4 heads cylinder and the two reels. In editing, the advanced heads are used for pre-reading and the trailing heads are used for after recording. Fig.12 illustrates the relation between tracks and heads. At an edit-in point, if T2 is greater than 0, the width of the track immediately before the start of the edit-in track will become narrow by the portion A and if T2 is less than 0, unerased portion will remain. At an edit-out point, if T1 is greater than 0, the track immediately after the edit-out track will become narrow and if T1 is less than 0, unerased portion will remain. In order to solve this problem the width of the trailing heads is designed to be a little narrower than that of the advanced heads. Although this will slightly decrease the offset margin of the trailing heads during Read After Write, appropriate and precise dimensions of the heads give satisfactory results. The fundamental mechanisms are basically the same in both studio and portable DATs. Except for the cassette loading structure, the main difference is that the emergency eject of a cassette

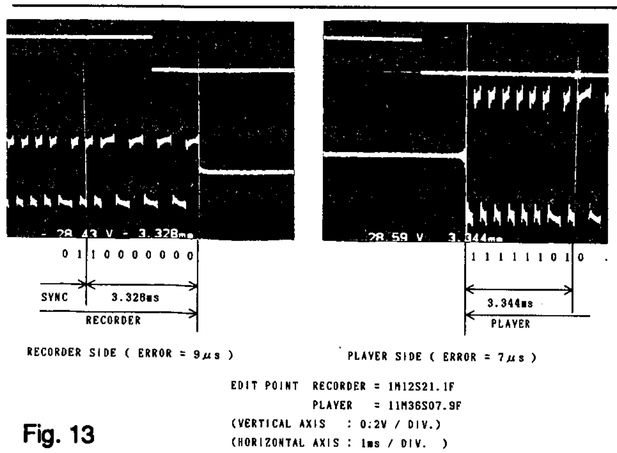

Fig. 13

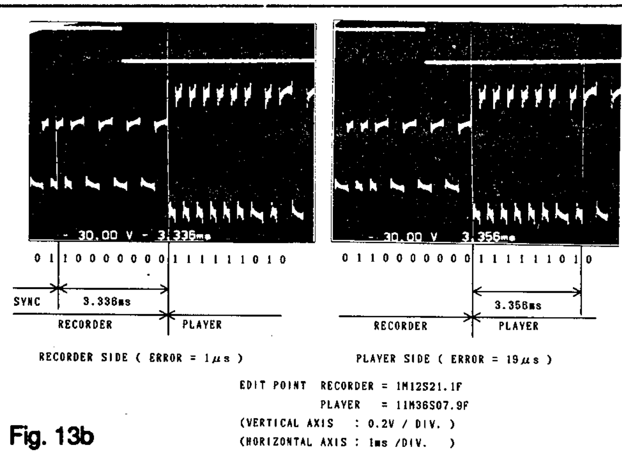

Fig. 13b

is employed in the portable DAT. When a cassette cannot be ejected by accident, it can be taken out manually.

8.2Servo control in high speed search

High speed search is performed by the following. In our system, Constant Relative Velocity (CRV) technique is used for high speed search where the tape speed and the cylinder rotation speed is controlled to keep the tape to head relative velocity constant. When the DAT is in fast forward and rewind mode, the tape to head relative velocity changes according to the tape speed, and the bit rate into the playback PLL, which is included in RF block in Fig.3, deviates around the normal rate of 9.408MHz. In order to read Sub data, the bit rate deviation must be within the lock range of the playback PLL. The servo LSI changes the tape speed and the cylinder rotation speed at 16 steps up to 200 times normal speed (x12.5,x25,x37.5,...). Furthermore the residual bit rate deviation detected by the signal processing LSI-2 is applied to servo LSI for the correction of tape speed. Consequently the bit rate deviation is suppressed within +/-3%, which realizes smooth and fast response in tape speed change at any speed including 200 times normal playback speed. Servo control is also basically the same in both studio and portable DATs.

9.A/D converter

Low power consumption is very important in a portable DAT where analog circuits usually dissipate more than half of the total power. The key device to reduce power dissipation is the A/D converter. Our studio and portable DATs employ the MASH A/D converters developed under the guidance of Nippon Telegraph and Telephone corporation. Because the MASH A/D converter is a 64 times oversampling CMOS LSI including a digital filter, only a 3rd order anti-aliasing analog filter is necessary. As a result, the power consumption of the A/D conversion block including anti-aliasing analog filters is 650 mW per dual channels. This is about 1/3 of that in the case of the combination of conventional successive approximation A/D converters, digital filters, and higher order analog filters. The MASH A/D converter and other power saving techniques in the portable DAT allow up to 2 hours of continuous recording with a commonly used NP1A rechargeable battery.

10.Other features

10.1External clock synchronization

The studio DAT is designed to accommodate an NTSC composite video signal or black burst signal for locking the internal clock to a "House Sync" reference. The synchronization to a word sync clock is also provided. From the above House sync or word clock, the master clocks are generated in the equipments. This makes possible the synchronized operation among audio and video equipments.

10.2Recording and Rehearsal of Start-ID, Program Number and Skip-ID

Start-ID,Program Number and Skip-ID are used for searching programs like a Compact Disc player. They can be rehearsed and recorded. They are also erasable and rewritable.

11.Measurement results of the accuracy in edit and synchronization

11.1Measuring method

The accuracy of editing and chase synchronization is very important. The editing accuracy has to be measured of the order of a sample period (20.83us). For this purpose we have devised a measuring method with high accuracy. The points are as follows.

1)The time code is recorded in both Sub data area and Main data area. Both time codes must be frequency-synchronized. Two types of tapes, Tape-1 and Tape-2 are prepared. In Tape-1, the time code in Sub data area is in line with the PCM data in Main data area. In Tape-2,the time code recorded in Main data area is advanced by the time corresponding to the time delay in a D/A converter,a digital filter and a low pass analog filter to read the time code in line with the PCM data at the analog output terminal.

2)In editing, Tape-1 is used. Signals with different levels are recorded for a Recorder tape and a Player tape respectively. The cut edit point is identified by the amplitude change at edit points between a Recorder tape and a Player tape. The accuracy within a frame is obtained by measuring the time difference between the cut edit point and the start of a SMPTE frame which is identical to the end of the synchronization bits of the SMPTE time code. By this method the editing accuracy of the order of several micro seconds can be measured.

3)In chase synchronization, Tape-2 is used. The time difference between the master time code and the time code at the analog output of the slave DAT is measured by comparing the two time codes by using a microcomputer. The detecting resolution is within +/-10us.

11.2Results

11.2.1Editing accuracy

The results for the assemble editing by the controller are photographed in Fig.13 (a) and (b), where the joint portion for memory check and edit is shown. This case is SMPTE (

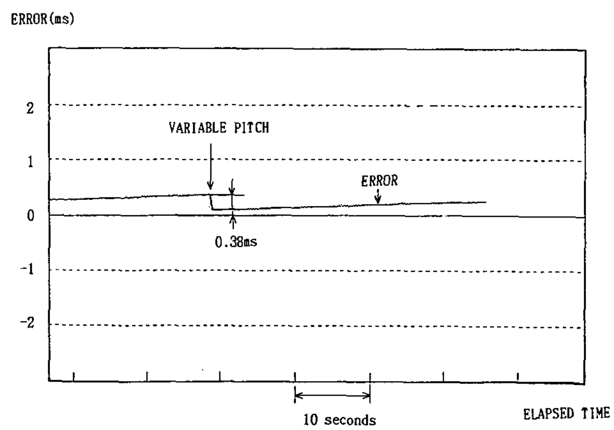

Fig. 14: Accuracy of chase synchronization mode 2 (measured)

Fig. 15: External view of the professional DAT system

29.97 Hz) time code. In each photograph, the lower part is the analog output signal equivalent to SMPTE time code. In Fig.13(a),(b) the Recorder and the Player sides show the edit point information within a frame in the Recorder and the Player respectively. The time code signal with the amplitude of about 0.4 V is the signal from the Recorder and the time code with the amplitude of about 0.8 V is the signal from the Player. The edit point is recognized by the amplitude change. The entry points are 1 minute 12 seconds 21.1 frames for the Recorder and 11 minutes 36 seconds 7.9 frames for the Player. The time code supplied from the analog output of the Recorder is memorized in a storage oscilloscope. Because the time code is a biphase mark signal, hours, minutes, seconds and frames are easily read from the screen. The sub frame accuracy is measured by magnifying the horizontal scale as shown in Fig.13(a) and (b).

In the case of memory check shown in Fig.13(a), the time information within a frame reads 3.328ms on the Recorder side and the error is calculated by subtracting 3.328ms from 3.337ms (1/10 of a 29.97Hz SMPTE frame) which results in 9 us. The error on the Player side is 7 us. The errors in editing are obtained from Fig.13(b). Thus from Fig.13 we can easily obtain the editing accuracy. Tab.2 shows an example of the accuracy measured. Many measurements give similar results and the editing accuracy is within +/-30us. In the case of deck to deck editing the accuracy is also within +/-30us. The error of +/-30us is caused by not only the conversion between the SMPTE time code and the professional DAT time code but also the small timing errors in the hardware. So, the accuracy can be improved in the future.

Tab. 2:Accuracy of assemble edit
ENTRY POINT
RECORDER = 1M12S21 . 1F
PLAYER = 11M36S07 . 9F

ITEM	RECORDER	PLAYER
	ERROR	ERROR
MEMORY CHECK	9μs	7μs
MEMORY REHEARSAL	9μs	7μs
TAPE REHEARSAL	3μs	19μs
EDIT	1μs	19μs
TAPE REVIEW	1μs	19μs

Tab.3 Accuracy of chase synchronization

ITEM	ACCURACY
CHASE MODE1	Within 30μs
CHASE MODE2	Within 0.4ms

11.2.2Chase synchronization

The results for chase synchronization mode 1 is shown in Tab.3. The accuracy, which means the error between the master time code and the output PCM data from the slave DAT, is within +/-30us at the lock-in point in chase mode 1. When both the master equipment and the slave DAT are connected to a video sync signal, the accuracy is within +/-30us as long as the equipments are in play mode. However if the video sync signal is not connected to each equipment, the error increases because of a sight difference between the two master clocks for the master equipment and the slave DAT.

The accuracy for chase mode 2 is measured when the time code is supplied from the master equipment to the DAT without a video sync signal. Because of the differences of the master clocks of the two equipments, the error increases gradually. Fig.14 is a example of the measurement results. The vertical axis is the error which is obtained by subtracting the master time code from the output time code of the slave DAT. When the error becomes greater than 0.4ms, the slave DAT changes it's tape speed by 0.1% and then the error decreases. The error is within 0.38ms when the master equipment is in play mode.

12.Summary of the features

Tab.4 shows the specifications for the studio DAT, the portable DAT, and the editing controller. In Fig.15 the external view of our professional DAT system is shown. The main features of the studio DAT and the portable DAT are as follows.

1) Studio DAT

*Read After Write and Write After Read

*Recording and playback of SMPTE time code

*Internal Time Code Reader/Generator

*Digital I/O with AES/EBU

*Memory rehearsal for both Recorder and Player (Rehearsal time: 5.46 seconds)

*Assemble and insert editing by Deck to Deck or by Controller

*Chase synchronization

*Clock synchronization

*Variable pitch control

*Digital crossfade

*Auto locate with frame accuracy

*Cueing function (3 times and 15 times normal playback speed)

*Recording and rehearsal of Sub-ID (Start-ID, Skip-ID and Program number)

*Remote control via RS-422 protocol

2)Portable DAT

Tab. 4(a): Specifications of Studio DAT

Tab.4 (a) Specifications of Studio DAT

ITEM	SPECIFICATIONS
TECHNICAL SPECIFICATIONS	
FREQUENCY RESPONSE	within ±0.5 dB,20Hz to 20kHz
DYNAMIC RANGE	Better than 90 dB
TOTAL HARMONIC DISTORTION	Less than 0.006% (1kHz) at maximum output level
CROSSTALK	Better than 80 dB (8kHz)
EMPHASIS	T1 = 50μs, T2 = 15μs
WOW AND FLUTTER	Beyond measurement limits
CHANNEL LEVEL DEVIATION	Two channels within 0.2 dB (1 kHz)
DIGITAL SIGNAL FORMAT	
SAMPLING FREQUENCY	48kHz
QUANTIZATION	16-bit linear
ERROR CORRECTION	2 dimensional Reed-Solomon Code
NO. OF CHANNELS	2
MODULATION	8-10 conversion
CYLINDER DIAMETER	30mm
CYLINDER ROTATION SPEED	2000 rpm
TAPE SPEED	8.15 mm/s
TRACK PITCH	13.6μm
MECHANICAL SPECIFICATIONS	
FF/REWIND TIME	Within 50 seconds for a 2-hour DAT cassette
SEARCH SPEED	Up to 200 times normal playback speed
CUEING SPEED	3 times / 15 times normal playback speed
VARIABLE SPEED	Up to ±12.4% of normal playback speed
ELECTRICAL SPECIFICATIONS	
INPUT	
LINE IN	Balanced XLR-5 Nominal level: +2dBm/-22dBm (-2VU) Input imp : 10kΩ/600Ω
TIME CODE IN	Balanced XLR-3 SMPTE 29.97Hz, 80 bit time code format Level : 0.5~10Vp-p.
DIGITAL IN	AES/EBU format (CCIR-647)
VIDEO SYNC IN	NTSC format Level : greater than 0.3Vp-p
WORD SYNC IN	BNC connector,TTL compatible
OUTPUT	
LINE OUT	Balanced XLR-5 Nominal level: +2dBm/-22dBm (-2VU)
TIME CODE	Balanced XLR-3 SMPTE 29.97Hz, 80 bit time code format Level : 2.0~2.8Vp-p.
DIGITAL OUT	AES/EBU format (CCIR-647)
VIDEO SYNC OUT	NTSC format Level : greater than 0.3Vp-p
WORD SYNC OUT	BNC connector,TTL compatible
SERIAL REMOTE	9p connector,RS-422
POWER SUPPLY	100VAC,50/60Hz,65W
DIMENSIONS	435(W) x 177(H) x 427(D) mm
WEIGHT	20kg

Tab. 4(b): Specifications of Portable DAT

Tab.4(b) Specifications of Portable DAT

ITEM	SPECIFICATIONS
TECHNICAL SPECIFICATIONS	
FREQUENCY RESPONSE	within ±0.5 dB,20Hz to 20kHz
DYNAMIC RANGE	Better than 90 dB
TOTAL HARMONIC DISTORTION	Less than 0.007% (1kHz) at maximum output level
CROSSTALK	Better than 80 dB (8kHz)
EMPHASIS	T1 = 50μs, T2 = 15μs
WOW AND FLUTTER	Beyond measurement limits
CHANNEL LEVEL DEVIATION	Two channels within 0.2 dB (1 kHz)
DIGITAL SIGNAL FORMAT	
SAMPLING FREQUENCY	48kHz
QUANTIZATION	16-bit linear
ERROR CORRECTION	2 dimensional Reed-Solomon Code
NO. OF CHANNELS	2
MODULATION	8-10 conversion
CYLINDER DIAMETER	30mm
CYLINDER ROTATION SPEED	2000 rpm
TAPE SPEED	8.15 mm/s
TRACK PITCH	13.6μm
MECHANICAL SPECIFICATIONS	
FF/REWIND TIME	Within 50 seconds for a 2-hour DAT cassette
SEARCH SPEED	Up to 200 times normal playback speed
CUEING SPEED	more than 3 times playback speed
ELECTRICAL SPECIFICATIONS	
INPUT	
MIC/LINE IN	Balanced XLR-3 ×2 Nominal level: -66/-56/-46dBs/-22dBm Input imp : 10kΩ/600Ω DC48V/12V for condenser microphones and external pre-amplifiers
OUTPUT	
LINE OUT	Unbalanced Nominal level: -22dBm (-2VU)
HEADPHONE	Unbalanced Power : 20mW + 20mW
TIME CODE	SMPTE 29.97Hz, 80 bit time code format
POWER SUPPLY	100VAC,50/60Hz,15W
DIMENSIONS	257(W) x 75(H) x 188(D) mm
WEIGHT	3.5kg

Tab. 4(c): Specifications of Editing controller

Tab.4(c) Specifications of Editing controller

ITEM	SPECIFICATIONS
SERIAL REMOTE	9p,RS-422
POWER SUPPLY	100VAC,50/60Hz,8W
DIMENSIONS	300(W) x 132.5(H) x 330(D) mm
WEIGHT	5.5kg

*Read After Write and Write After Read

*Recording and playback of SMPTE time code

*Recording and rehearsal of Sub-ID (Start-ID, Skip- ID and Program number)

*Cueing function

13. Conclusion

In this paper, a prototype DAT system with professional functions is shown. This professional DAT system employing IEC draft format of the time code offers compatibility of time code with other professional equipments and capability to play back pre-recorded DAT consumer tapes. The combination of implementing IEC draft format and solid state memories has achieved high accuracy of 30us in editing and chase synchronization in mode 1. This means that our system meets the stringent demands for high quality editing and chase synchronization in audio production, audio video broadcasting and other applications. These advantages will open up a wide variety of applications including advanced audio-video shots, software exchange among consumer and professional equipments. The portable DAT offers compactness and high quality sound recording, and improves efficiency in electronic news gathering. All these features implies that the introduction of professional DAT brings great benefit to a broad range of users in the professional audio and video field.

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References

1) The DAT Conference Standard; Digital Audio Tape Recorder System,
June. 1987.

2) R.Lagadec; R-DAT AND PROFESSIONAL AUDIO,
Preprint, the 83rd AES Convention, Oct. 1987.

3) H.Yamazaki, T.Ketori,T.Morita,S.Okazawa,H.Nojima and Y.Abe.;
TIME CODE IN SUB DATA AREA ON R-DAT,
Preprint, the 84th AES Convention, Mar. 1988.

4) T.Ketori, T.Morita, S.Okazawa, Y.Abe and M.Cohen;
A PROFESSIONAL R-DAT RECORDER,
Preprint, the 85th AES Convention, Nov. 1988.

5)N.Miyajima and M.Tsukatani; TIME CODE RECORDING FOR PROFESSIONAL DAT RECORDERS,
Minutes, the AES Digital Audio Recorder Study Group,
Nov. 1988.

6)IEC Technical Committee SC60A Draft Standard; DIGITAL AUDIO TAPE (DAT) CASSETTE SYSTEM FOR PROFESSIONAL USE, Feb.1989.