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Application of Oversampling A/D and D/A Conversion Techniques to R-DAT

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Abstract

The rotary-head digital audio tape recorder (R-DAT), which was launched in the Japanese consumer market in March 1987, is the latest in a series of digital audio equipment which has appeared recently. Besides offering high-speed search, ease of operation and other functional advantages, it boasts recording and playback performance well above that of conventional analog consumer recording equipment.

Audio performance of the R-DAT is determined mainly by the quality of the A/D and D/A conversion circuits. This paper describes the application of an oversampling technique to both A/D and D/A conversion circuits in our first-generation R-DAT. A drastic improvement was achieved in the phase characteristics in the recording/playback processes, resulting in virtually identical recording and playback waveforms.

1. Introduction

With the rapid advances in hardware and software for digital signal processing, coupled with progress in LSI technology, a vast number of systems employing digital technology have been developed in the communications and computer equipment fields. In the audio field, too, the trend toward use of digital techniques over the past ten years has affected both professional and home audio markets. A growing array of digital audio equipment has passed through the development and manufacturing stages and now appears on the market. This includes PCM recording systems making use of videocassette decks, as well as CD (Compact Disc) players, digital mixing consoles, digital delay equipment, and multi-track PCM recorders.

Compact discs, in particular, offer a remarkable advance in sound quality and ease of use over conventional analog discs. Another advantage is longer playback time. As a result, they are now beginning to replace analog discs as a standard medium for recorded music.

Although CD players boast such outstanding qualities, they were developed for playback only, and cannot be used for recording purposes. Now, however, the R-DAT (Rotary-head Digital Audio Tape recorder) overcomes this drawback by offering a means of digital recording with the same similar ease of use as CD players. In Japan, the first generation of R-DAT recorders began to appear on the market in March 1987, and models have now been announced by nearly all major audio manufacturers. The price of R-DAT recorders is still on the high side, as they have not yet reached large-scale mass production. Within the next few years, however, they are expected to capture a large share of the audio cassette market, as the applications spread to include car decks and portable models.

Since a DAT recorder is used for recording as well as for playback, unlike a CD player it requires an A/D converter, which converts analog signals to digital signals. The authors, in designing an R-DAT recorder, paid particularly close attention to the A/D converter and to the overall A/D conversion circuitry, as well as to the D/A conversion circuit, which converts digital signals back to analog signals. DAT is not a digital copy machine.

Rather, it is an instrument for exact recording and playback of music and other audio signals. Accordingly, it must reproduce analog signals as faithfully as possible. In order to get close to this goal, the authors applied the oversampling technique to both the A/D and D/A conversion processes. This resulted in a notably improved phase characteristics to both recording and playback, with virtually identical input and output waveforms. It has thus become possible for the first time to achieve truly transparent audio reproduction through a consumer recording and playback process.

2. R-DAT System Configuration [1]

A block diagram of the R-DAT system which we designed is shown in Fig. 1. Since the main purpose of this paper is to discuss the audio signal processing system, the other aspects will be introduced here only briefly, without a detailed explanation.

The R-DAT is a complex system, but it can be divided into four main blocks. These are the audio data processing block, the digital signal processing block, the servo block, and the control block. Each of these four blocks will be described below.

2-1 Audio data processing block

The audio data processing block consists of the following three sections:

(1) Analog audio interface
(2) Digital audio interface
(3) Digital level meter

The analog audio interface performs the task of converting analog audio signals to digital audio signals, and also that of converting digital audio signals back to analog audio signals. This section contains the most distinctive features of the R-DAT we designed, namely, use of the oversampling technique in both the A/D and D/A conversion processes. This will be described in more detail below. In order to take best advantage of this technology, and to get the highest audio performance, careful attention was paid also to the analog circuitry. It was designed as a single printed circuit board, shown in Fig. 2.

The digital audio interface sends and receives audio data to and from other digital equipment. Besides audio data, it also sends and receives part of the sub code (sampling frequency, with or without emphasis, etc.). Specifications of the digital audio interface are based on IEC draft standards [2].

A digital level meter is superior to analog level meters in many ways. For example, it is accurate, has quick response, is stable, and does not affect the analog audio circuit. On the other hand, digital level meters tend to be expensive. However, we were able to design an inexpensive digital level mete featuring very flexible display capabilities, by employing a small-scale LSI along with a single-chip microprocessor. The LSI, besides performing logarithmic conversion of magnitude of digital audio data, performs peak hold until these data are read by the microprocessor. This means that the microprocessor is able to read peak data faithfully, even though the microprocessor doesn't have high processing speed.

2-2 Servo block

The servo block consists of the following three sections.

(1) Drum servo
(2) Capstan servo
(3) Reel servo

These servos switch from one operation to another depending on the system mode, for example, playback, record, or fast search.

The drum servo rotates the drum at 2,000 r.p.m. during the playback and record modes. During fast search, it controls the drum rotational speed in accordance with the tape speed, so that the relative velocity of the head and tape is the same as that during the playback mode. This enables the necessary sub code to be read correctly from the tape even during fast search.

During recording, the capstan servo controls the capstan motor, keeping the tape speed at a uniform 8.15 mm/s. During playback, by the use of the automatic track finding (ATF) signal recorded on each track, it controls the capstan speed to make certain that the head traces the track accurately.

The reel servo keeps the tape speed practically constant during fast search, to ensure that this operation is carried out stably. It also keeps the tape tension at a proper level for optimal head-to-tape contact during recording , playback and fast search.

2-3 Digital signal processing block

The roles of the digital signal processing block during recording include data interleaving, encoding of the error correction code, and data modulation. During playback it performs data demodulation, decoding of the error correction code, and data de-interleaving. It is also used for sub code processing.

2-4 Control block

The control system is made up of five separate microprocessors. Of these, the three basic chips are the system CPU, mechanism CPU, and operation CPU. The system CPU has the extremely important job of controlling the R-DAT system as a whole. It arranges the sub code during recording and search so that the search operation can take place simply and accurately. A suitable means of data protection is essential especially during fast search because of the low reliability of readout of the sub code.

The mechanism CPU controls the tape transport mechanism and each of the servos. The operation CPU scans the key switches, processes remote control data, and displays various information on the panel.

The remaining two chips, the digital I/O CPU and the meter CPU, are processors used for implementing various system functions. The digital I/O CPU sends and receives sub code via the digital audio interface LSI. The meter CPU reads data from the level meter LSI and drives the panel display.

3. Application of the Oversampling Technique

Characteristics of a digital audio system are determined by the anti-aliasing filter, sample and hold circuit, A/D converter, and D/A converter. Among these the design of the anti-aliasing filter is particularly important. The reason for this is that this filter is required to have small passband ripple; narrow transition bandwidth; and at the same time, large attenuation. If an analog filter is used, these requirements can be met only by using a high order filter. Such a filter, however, is subject to large group delay distortion near the cutoff frequency. For this reason most CD players, in order to achieve sharp cutoff characteristics along with low group delay distortion, use a digital filter in combination with an analog filter [3]. The analog filter used in such cases has gentle cutoff characteristics and excellent group delay characteristics. This approach is known as the oversampling technique. In a DAT recorder, overall group delay characteristics can be improved by applying this technique not only in the playback process but in the recording process as well.

3-1 Methodology

Figure 3 shows the processing steps of the audio signal when twofold oversampling is applied to both recording and playback.

In the recording mode, the analog signal first passes through an analog low-pass filter, where its spectrum is band restricted to 1.5 fs (fs=sampling frequency). The relatively wide transition band of this filter, between 0.5 and 1.5 fs, is acceptable here; more importantly, it must have excellent phase characteristics. After the signal is sampled and held at 2 fs, it is converted to a digital signal. The resulting digital data signal is restricted by the digital filter to 0.5 fs. Next, in decimation of the signal the sampling frequency is reduced from 2 fs to fs. The digital filter used here is a linear phase FIR filter, whose group delay characteristics are uniform at all frequencies.

The process during playback is exactly the reverse of that during recording. The sampling frequency of the digital data for playback is increased by an interpolation filter from fs to 2 fs. The characteristics of this digital filter are the same as those used on the recording side. D/A conversion takes place at 2 fs. The resulting analog signal has already had the components from 0.5 fs to 1.5 fs removed by the digital filter. Accordingly, exactly the same analog low pass filter can be used at the final playback stage as on the recording side. Using this approach, it is possible to achieve recording and playback of audio signals with practically no group delay distortion.

3-2 Implementation

Timing of the A/D conversion circuit

In actually implementing the twofold oversampling technique for A/D and D/A conversion described above, certain problems must be solved. The biggest of these is the tradeoff between the conversion time of the A/D converter and the acquisition time of the sample and hold circuit. If, for example, the sampling frequency is chosen such that 2 fs = 96 kHz, then the sampling interval will be 10.42 µs. During this time the analog signal must be sampled and converted to a digital signal. Unfortunately, nearly all audio A/D converters currently available require more than 10 µs for conversion. The A/D converter we have adopted, however, is a successive approximation A/D converter, which is able to perform the necessary conversion in only about 8 µs. Thus the allowable acquisition time is within approximately 2.4 µs, so that sample and hold is at least possible.

Another problem is that during the transition from sample to hold, the output from the sample and hold circuit is unstable.

Starting A/D conversion right at the beginning of the analog signal hold interval can result in a conversion error. Accordingly, it is necessary to design a waiting interval at the beginning of the hold interval before starting A/D conversion. There is thus a tradeoff here also, between the waiting time and the A/D conversion time.

A timing chart of the A/D conversion process, which we designed keeping these tradeoffs in mind, is shown in Fig. 4. The sample time of the sample and hold circuit is 2.44 μs, and the hold time is 7.98 μs. The A/D converter starts conversion 163 ns after the beginning of the signal hold interval. Because of limitations in operating speed of actual devices, such as those noted above, determining the timing of the A/D conversion circuit is an extremely difficult task.

Analog filter

A 7th order Butterworth filter, which has practically uniform group delay characteristics and gradual cutoff characteristics, is used for the analog low pass filter. These characteristics are shown in Fig. 5. The group delay at 20 kHz is approximately 5 μs more than that at lower frequencies. For comparison, the characteristics of an 11th order Chebyshev filter, used for normal sampling, are shown in Fig. 6. With this filter the difference in group delay at 20 kHz and that at lower frequencies is greater than 400 μs.

Digital filter

The frequency response of the digital filter we adopted is as shown in Fig. 7. The attenuation level in the stopband is more than 90 dB, while ripple in the passband is within 0.001 dB, and group delay is uniform. A linear phase FIR filter requires a large number of taps in order to achieve sharp cutoff characteristics. Increasing the number of taps, however, also increases the number of computations, so that the filter characteristics are restricted by the processing speed of the hardware. To ease this restriction, a half-band (Nyquist) FIR filter is widely used [4]. A feature of half-band FIR filters is that the passband width and stopband width are equal. Likewise, passband ripple and stopband ripple are the same. Furthermore, the filter tap coefficients are zero at every other tap, reducing the number of computations to half that of an ordinary linear phase FIR filter. As a result, using the same number of computations, a half-band FIR filter can achieve sharper cutoff characteristics than an ordinary linear phase FIR filter.

The equations below show relations known to exist among the different parameters of an FIR filter based on equiripple design [5]. In these equations,

δp: passband ripple
δs: stopband ripple
ΔF: transition band width (Hz)
F : sampling frequency (Hz)
N : number of taps in the FIR filter

$$N \cong D_\infty(\delta p, \delta s) \times F/\Delta F \tag{1}$$

$$D_\infty(\delta p, \delta s) = [\, a_1(\log_{10}\delta p)^2 + a_2(\log_{10}\delta p) + a_3 \,] \times \log_{10}\delta s + [\, a_4(\log_{10}\delta p)^2 + a_5(\log_{10}\delta p) + a_6 \,] \tag{2}$$

Here $a_1 = 0.005309$, $a_2 = 0.07114$, $a_3 = -0.4761$, $a_4 = -0.00266$, $a_5 = -0.5941$, and $a_6 = -0.4278$.

In the case of a half-band filter, since $\delta s = \delta p$, attenuation A (dB) ($A = 20\log_{10}(1/\delta s)$) can be used in Equation (1) with the following result.

$$N \cong (k_1A^3+k_2A^2+k_3A+k_4) \times F/\Delta F \tag{3}$$

Here $k_1 = -6.64 \times 10^{-7}$, $k_2 = 1.71 \times 10^{-4}$, $k_3 = 5.35 \times 10^{-2}$, and $k_4 = -4.28 \times 10^{-1}$.

These relations are shown in Fig. 8. Assuming F=96 kHz, ΔF=4.5 kHz, and attenuation of 90 dB, then N is estimated at 113 or more. In the present design an FIR filter with 121 taps was used.

3.3 Results

To investigate the effectiveness of the oversampling technique, measurements were made of frequency response and time response.

Frequency response

When twofold oversampling is applied to both A/D and D/A conversion, the frequency response is as in Fig. 9. Within the audio band, group delay is nearly uniform, with only a 10 μs difference in group delay at 20 kHz from that at lower frequencies. By comparison, when twofold oversampling is applied to D/A conversion only, the frequency response is as in Fig. 10. On the A/D side, only an 11th order analog Chebyshev filter is used, whose characteristics were noted earlier in Fig. 6. This example shows the improvement in group delay characteristics even when oversampling is applied only to D/A conversion.

Time response

Time response waveforms are shown in Figs. 11 through 14 for tone-burst signals (of 15 kHz and 20 kHz), square wave signals (1 kHz) and music signals. In each of these pairs of figures, (a) shows the results when oversampling is applied to both A/D and D/A conversion, while (b) is for oversampling applied to the D/A side only. For all of these signals, superior reproduction is evident when oversampling is applied to both A/D and D/A conversion. The spectra of tone-burst and square wave signals contain components outside the audio band, so the input waveforms are not reproduced exactly. In the case of music signals, however, which do not include components outside the audio band, the waveform is reproduced virtually unchanged, as seen in Fig. 14-(a).

4. Conclusion

Application of oversampling to the A/D and D/A conversion circuits of an R-DAT recorder has resulted in excellent group delay characteristics along with outstanding waveform reproduction. The specifications of this R-DAT are shown in Table 1, and the recorder itself is pictured in Fig. 15.

A/D converters currently available require longer conversion time. Accordingly design for the timing operation of the A/D converter and sample and hold circuit is a critical factor, when oversampling is employed on the A/D side. A future development task will be to realize an even higher oversampling rate by devising a faster A/D converter and faster sample and hold circuit.

The authors hope that the techniques reported here will lead to increased user satisfaction with the sound quality of R-DAT.

5. Acknowledgements

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The authors would also like to thank Mr. M. Takeda of Nippon Precision Circuits LTD. for his useful suggestions.

6. References

[1] The DAT Conference, DIGITAL AUDIO TAPERECORDER SYSTEM, June, 1987.

[2] IEC ,"Draft Standard for a digital audio interface," TC84/WG11(Tokyo/Iwashita) 1, Nov., 1986.

[3] M. Takeda and M. Takahashi,"Digital Filter CMOS LSI for Compact Disc Player," AES TOKYO CONFERENCE, June, 1985.

[4] R. E. Crochiere and L. R. Rabiner, Multirate Digital Signal Processing. Englewood Cliffs, NJ; Prentice-Hall, 1983.

[5] L. R. Rabiner and B. G. Gold, Theory and Application of Digital Signal Processing. Englewood Cliffs, NJ; Prentice-Hall, 1975.

No. of quantization bits		16 bits linear
Sampling frequency		48kHz (REC/PLAY) 44.1kHz (PLAY) 32kHz (PLAY)
Frequency response	(REC/PLAY)	3 ~ 22kHz ±0.5dB
Signal-to-noise ratio	(REC/PLAY)	95dB (A-weighted)
Dynamic range	(REC/PLAY)	94dB
Total harmonic distortion	(REC/PLAY)	0.003% (at 1kHz)
Channel separation	(REC/PLAY)	100dB (at 1kHz) 90dB (at 10kHz)

Table1 Typical specifications of the R-DAT

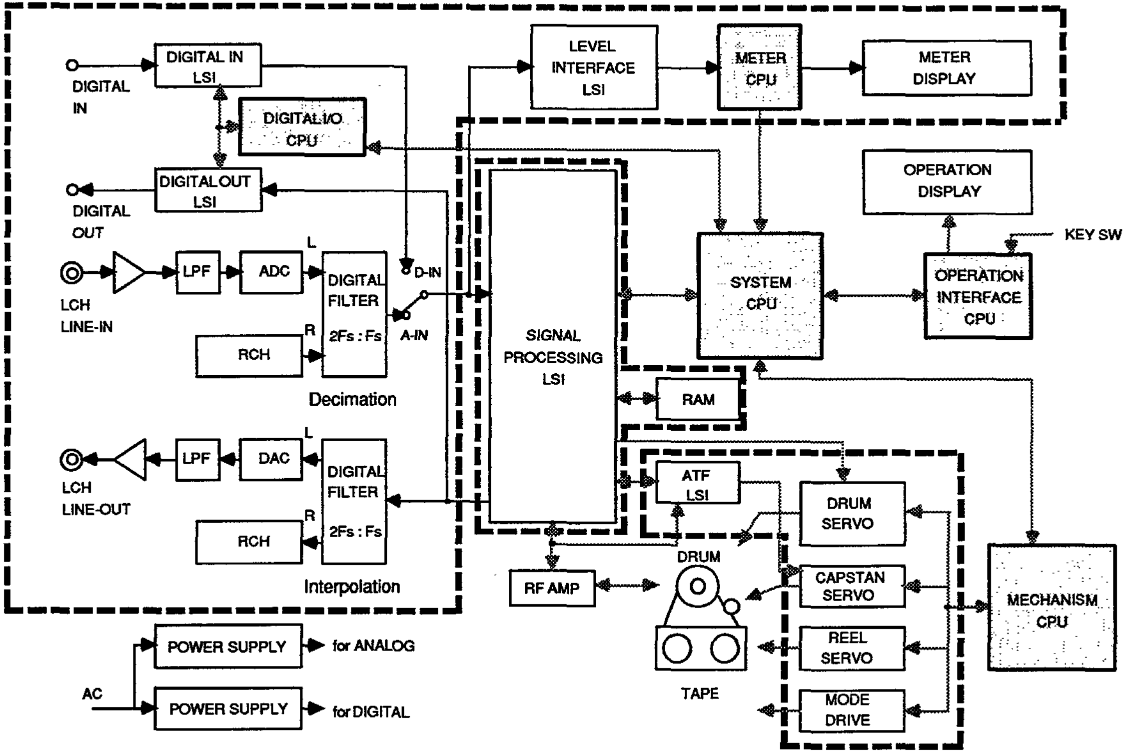

Fig. 1 Block diagram of the R-DAT

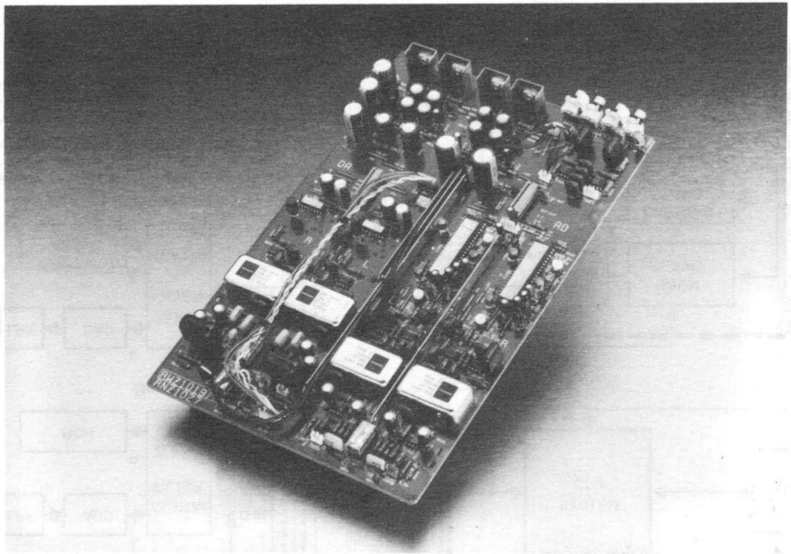

Fig. 2 Analog audio circuitry, including LPF, S/H, A/D and D/A conversion circuits

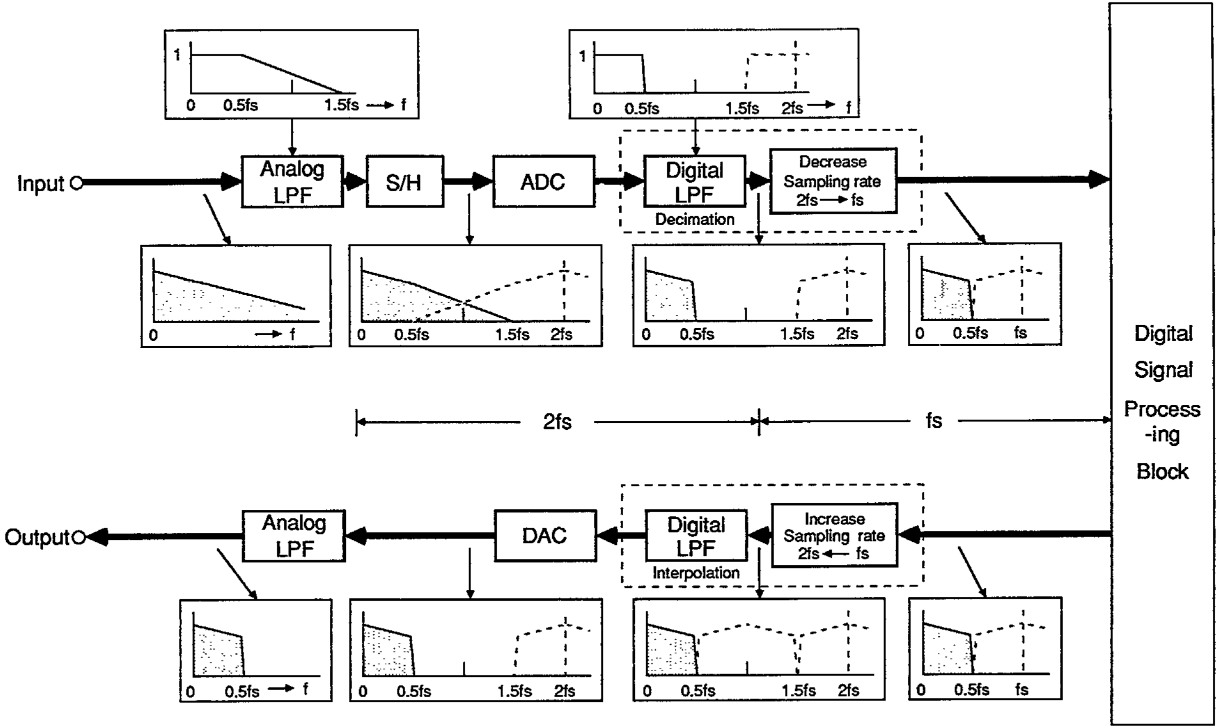

Fig. 3 Signal flow of audio signal processing

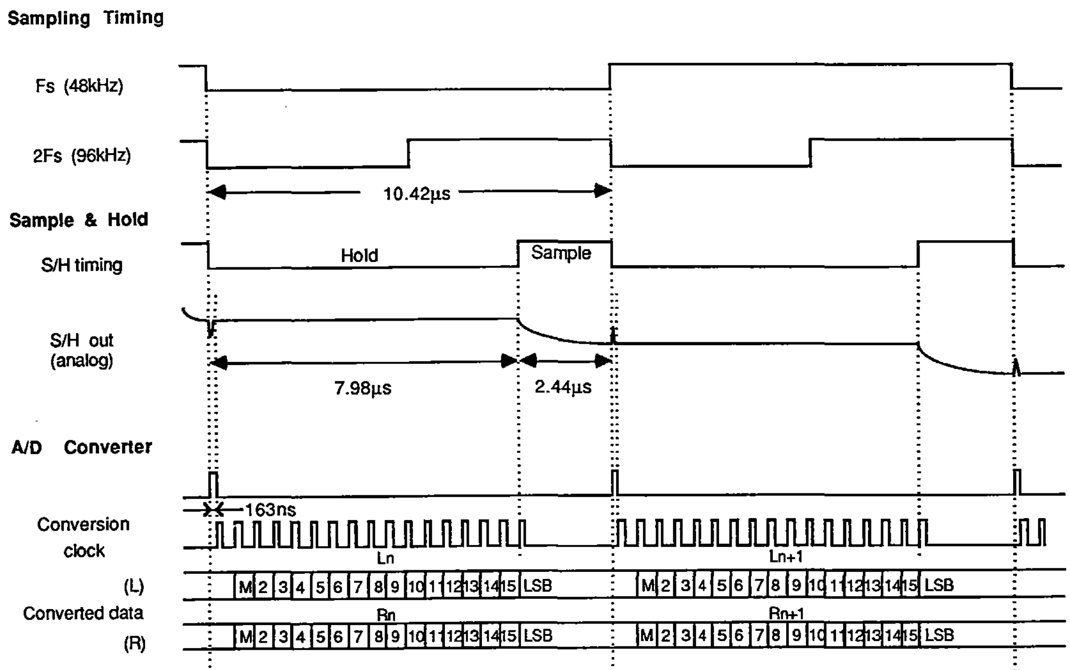

Fig. 4 Timing chart of the A/D conversion

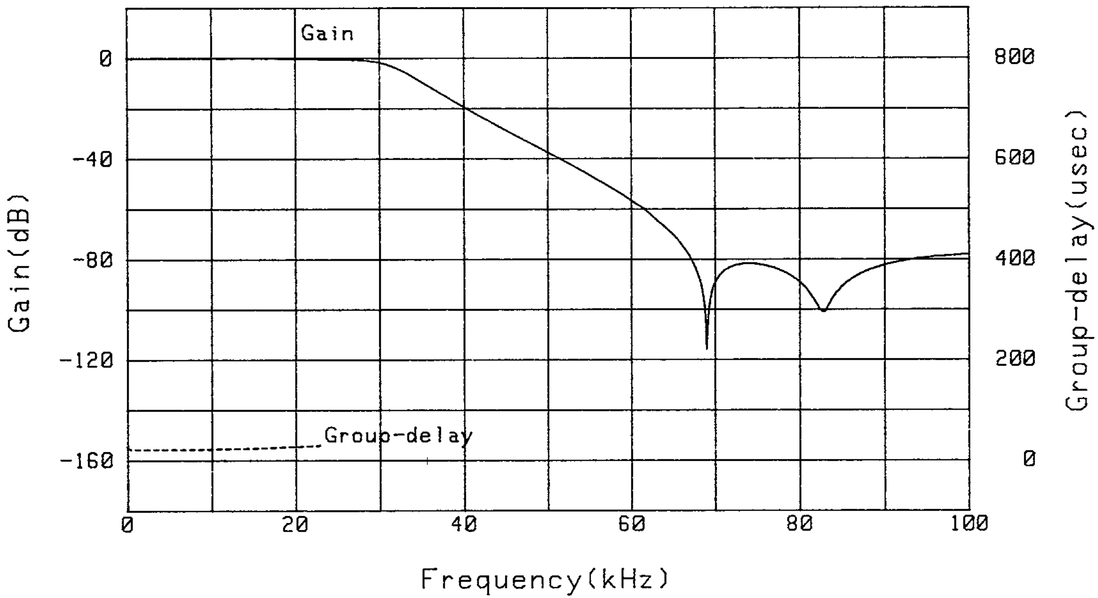

Fig. 5 Frequency response of a 7th order Butterworth filter

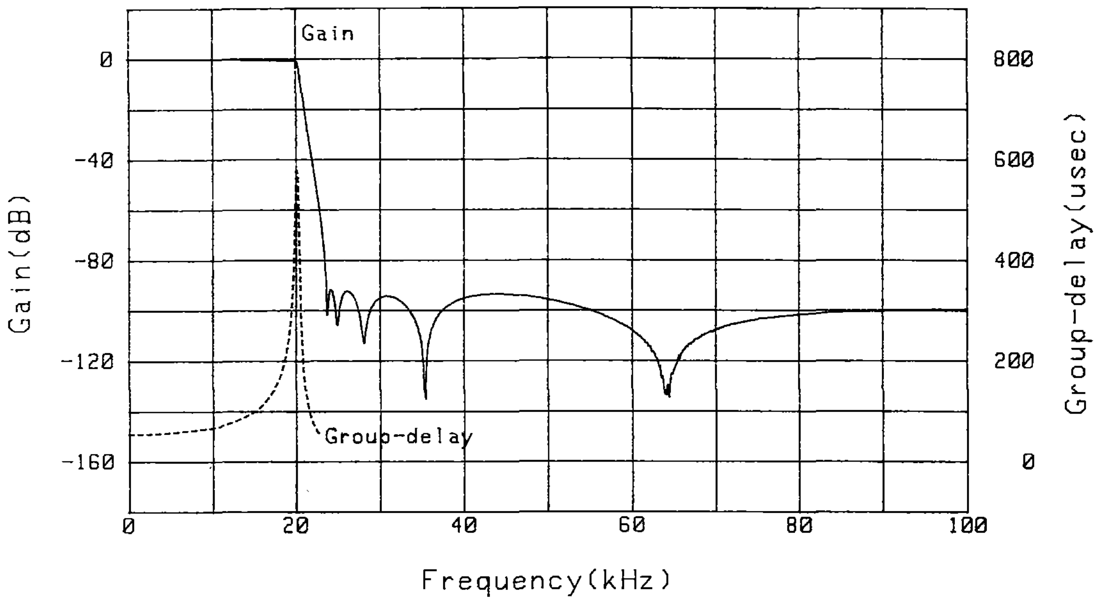

Fig. 6 Frequency response of an 11th order Chebyshev filter

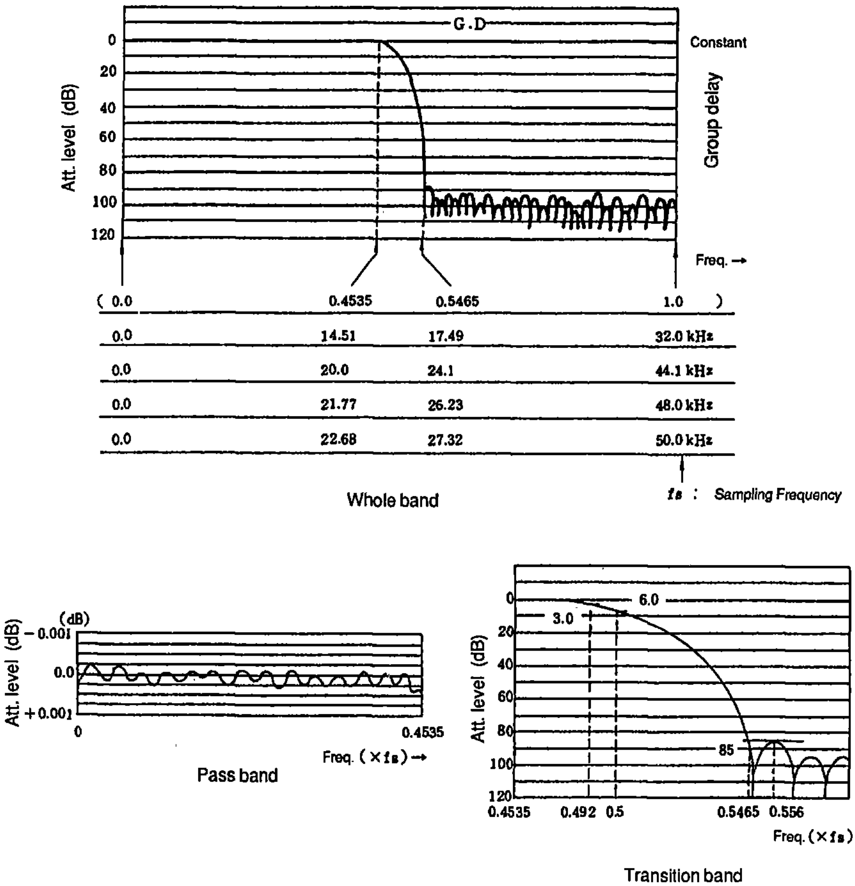

Fig. 7 Frequency respone of a Digital Filter

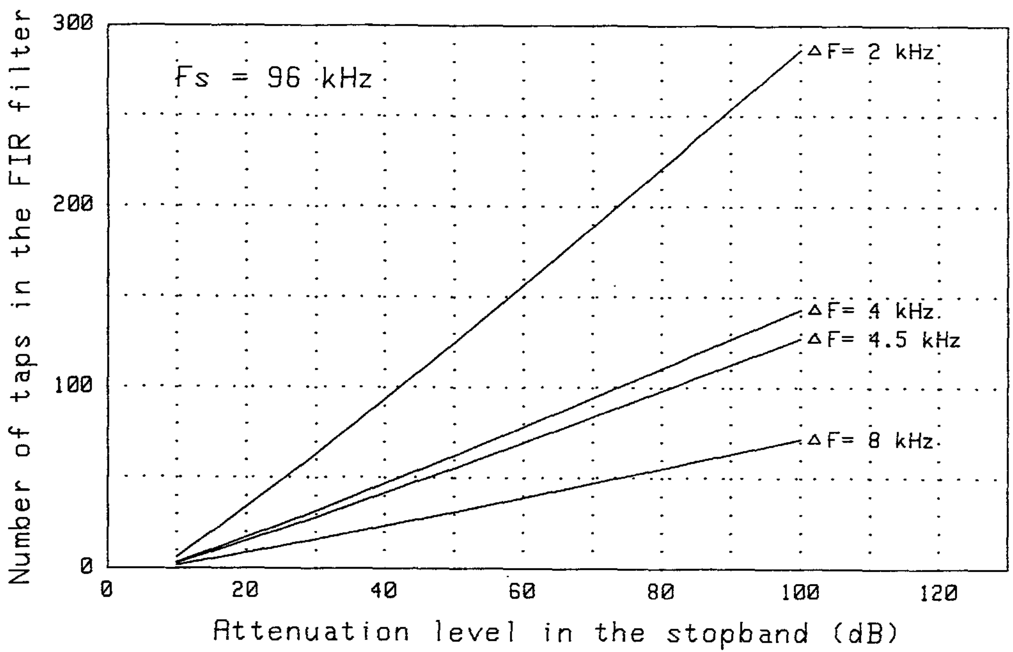

Fig. 8 Design relations for half-band FIR filters

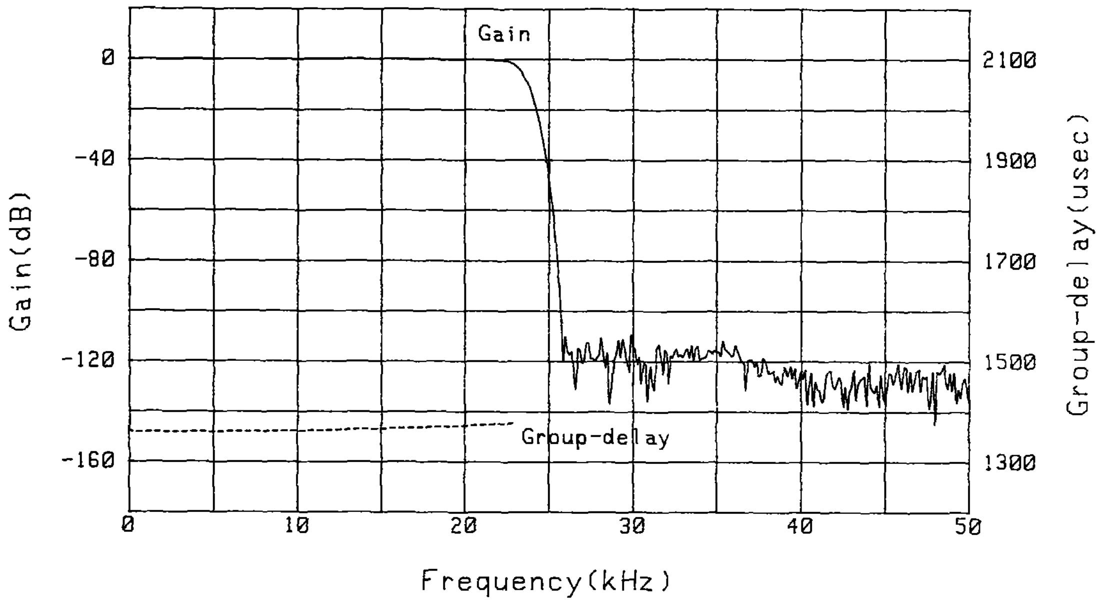

Fig. 9 Total frequency response of R-DAT with A/D and D/A oversampling

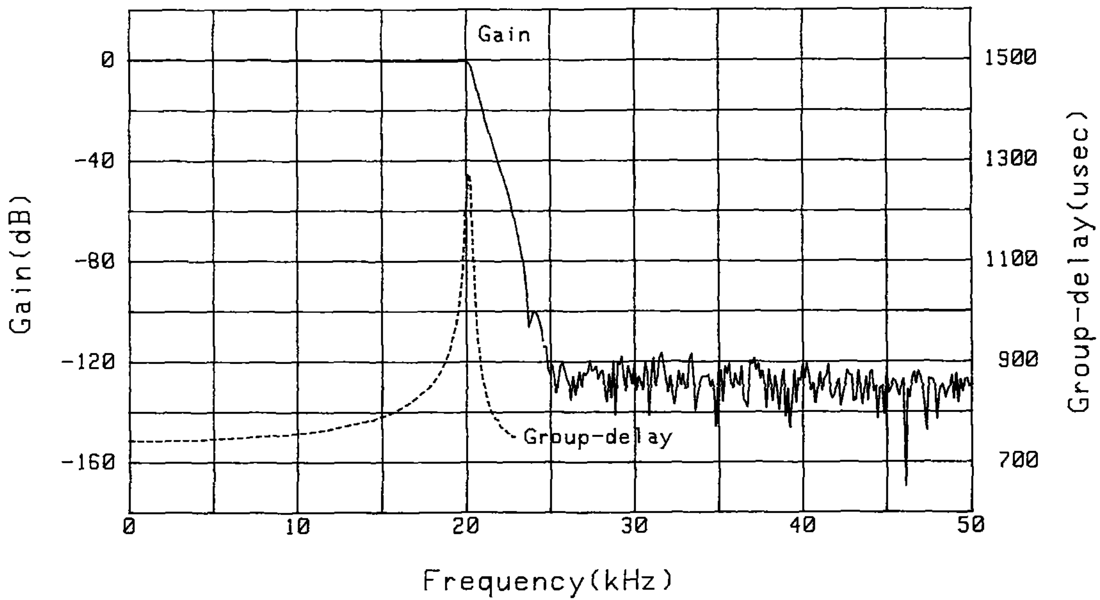

Fig. 10 Total frequency response of R-DAT with D/A oversampling only

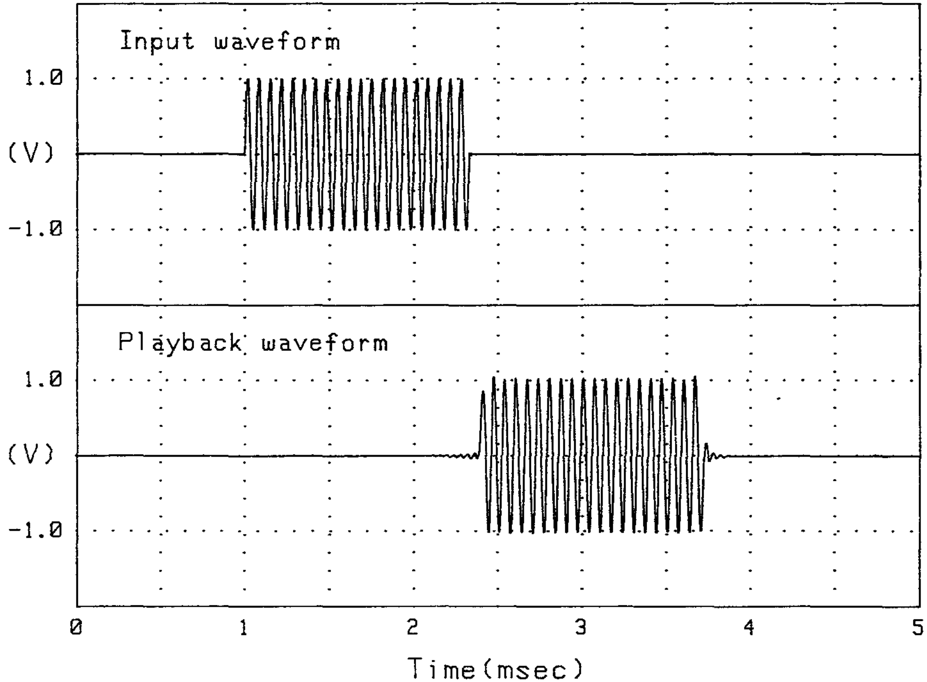

Fig. 11-(a) Playback waveform with A/D and D/A oversampling
(Input signal : 15kHz tone-burst signal)

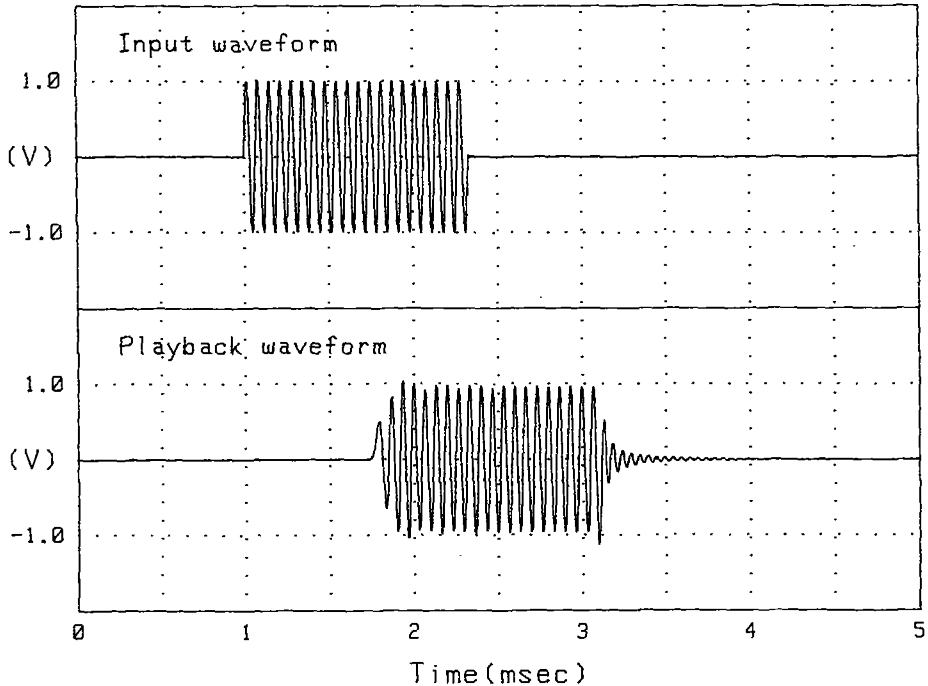

Fig. 11-(b) Playback waveform with D/A oversampling only
(Input signal : 15kHz tone-burst signal)

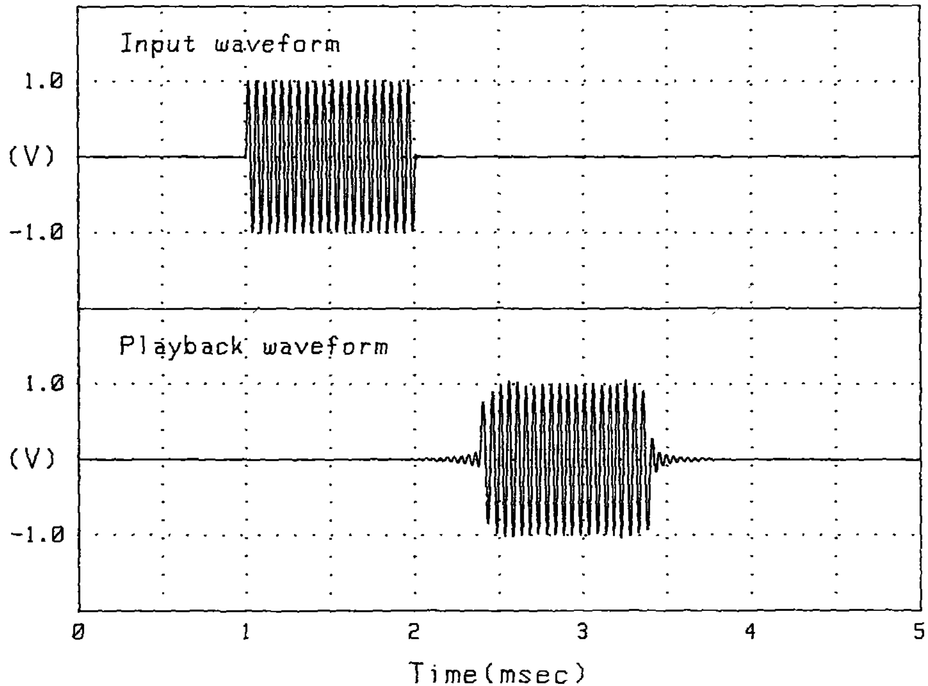

Fig. 12-(a) Playback waveform with A/D and D/A oversampling
(Input signal : 20kHz tone-burst signal)

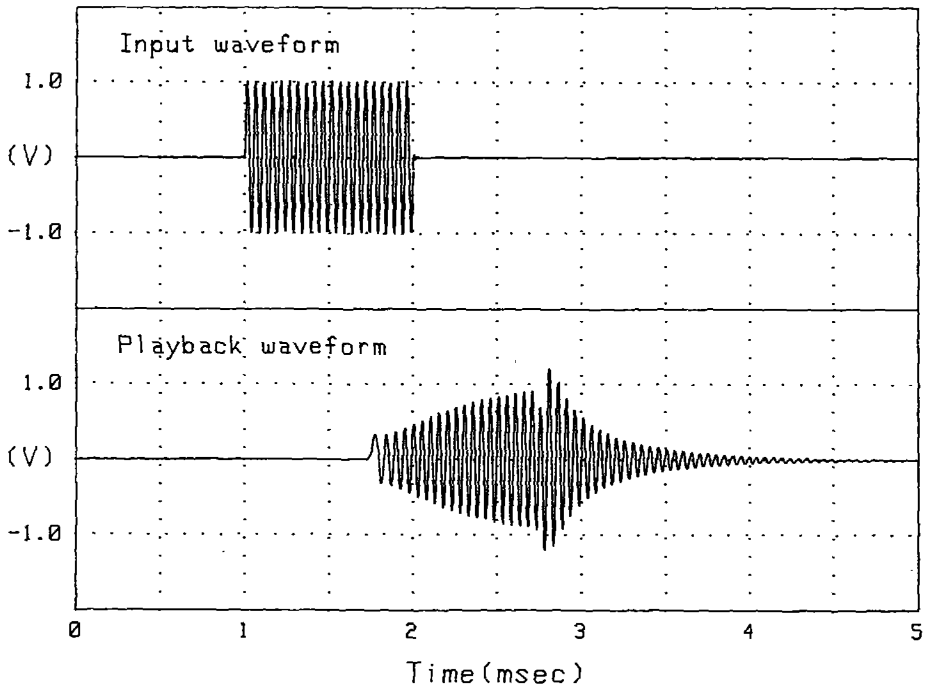

Fig. 12-(b) Playback waveform with D/A oversampling only
(Input signal : 20kHz tone-burst signal)

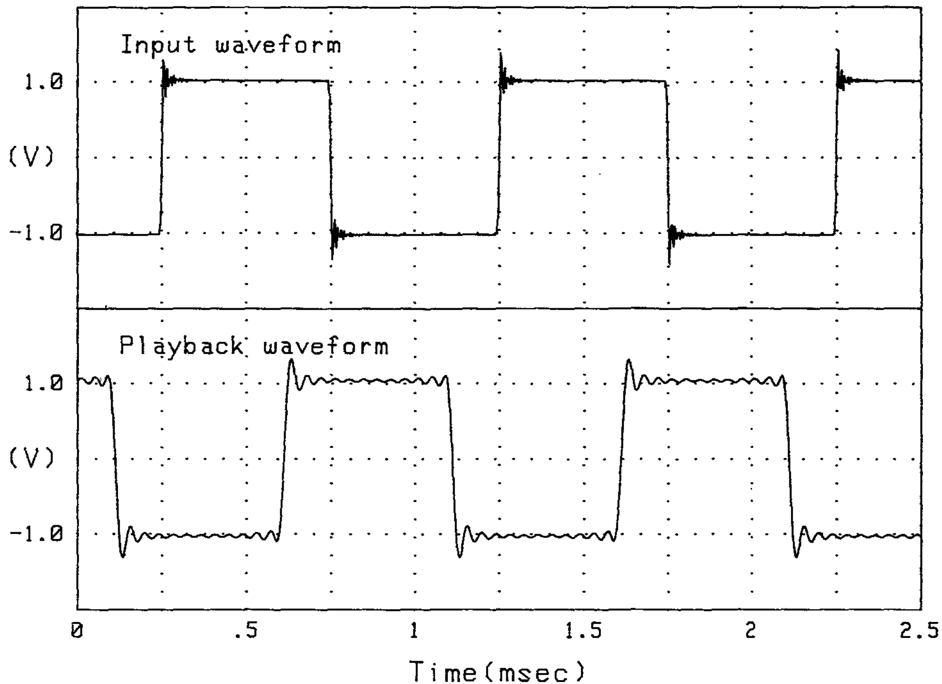

Fig. 13-(a) Playback waveform with A/D and D/A oversampling
(Input signal : 1kHz square wave signal)

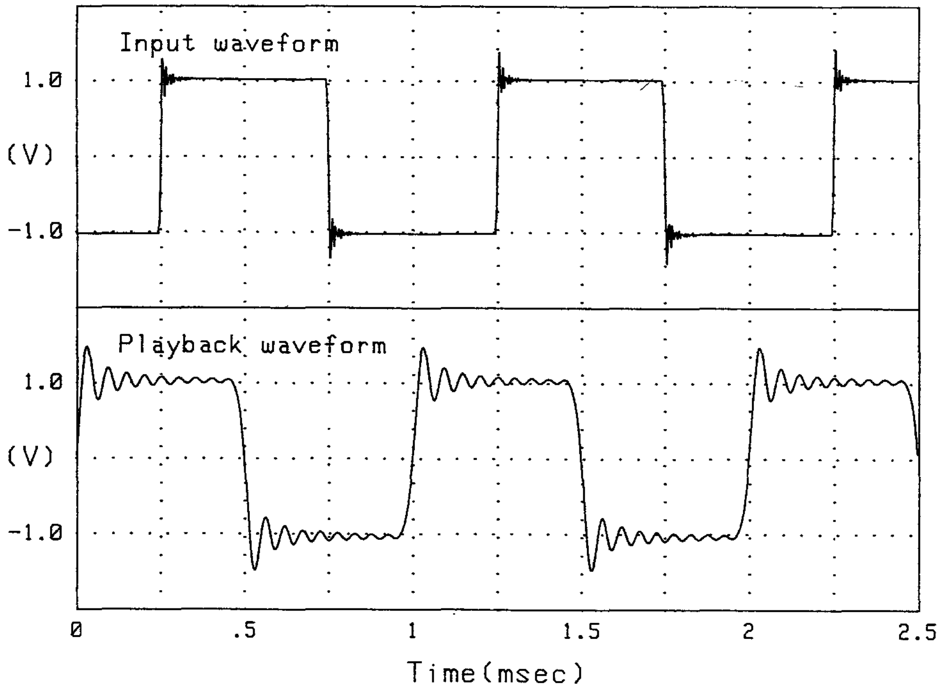

Fig. 13-(b) Playback waveform with D/A oversampling only
(Input signal : 1kHz square wave signal)

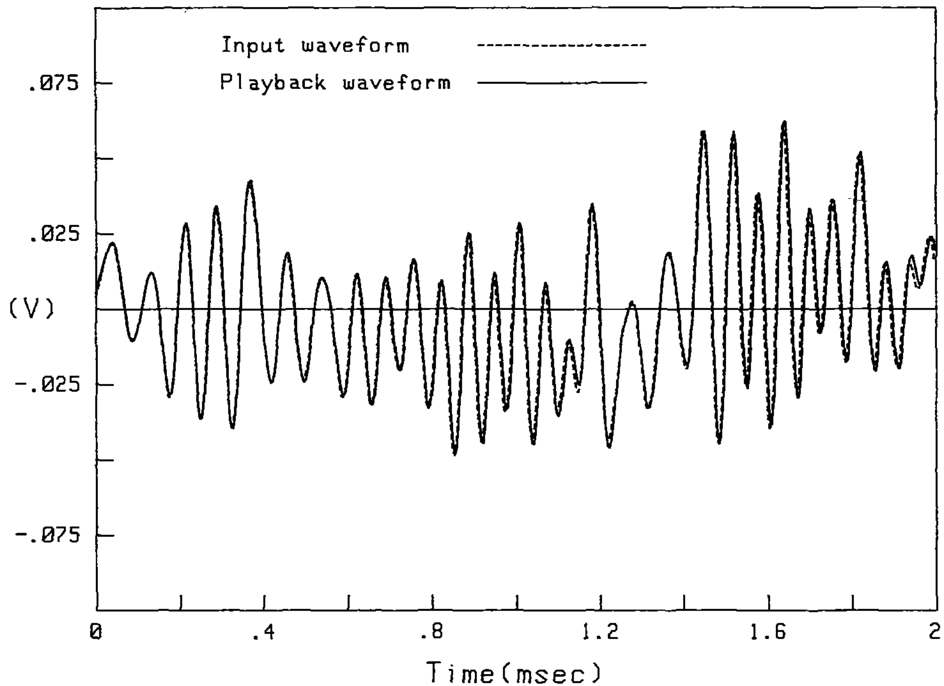

Fig. 14-(a) Playback waveform with A/D and D/A oversampling
(Input signal : music signal (Harpsichord))

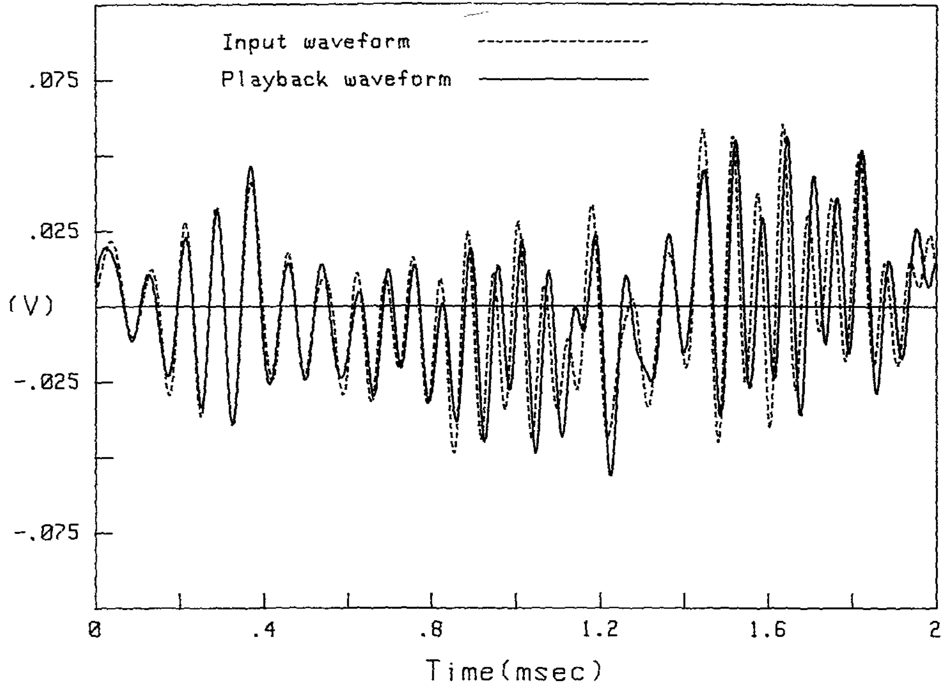

Fig. 14-(b) Playback waveform with D/A oversampling only
(Input signal : music signal (Harpsichord))

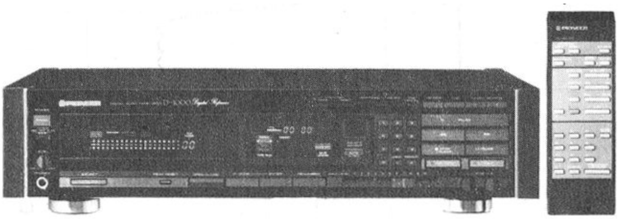

Fig. 15 R-DAT model employing A/D and D/A oversampling technique