MULTI-FUNCTIONAL OVERSAMPLING DIGITAL FILTER LSI FOR DIGITAL AUDIO USE

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## AN AUDIO ENGINEERING SOCIETY PREPRINT

# Multi – functional Oversampling Digital Filter LSI for Digital Audio Use

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## Abstract

Oversampling D/A conversion systems are widely used in digital audio equipments to improve reproduction audio signal quality. The authors have developed an 8 – times oversampling digital filter CMOS LSI with higher performance and several other functions in a chip. 8 – times oversampling (interpolation) digital FIR (Finite Impulse Response) filtering, digital de – emphasis IIR (Infinite Impulse Response) filtering and digital attenuation are the main funcιions of this LSI.

This paper describes the system costruction and performance and the digital signal processing of the developed digital filter LSI.

## I. Introduction

In the audio and audio – visual field, various new equipments which use digital techniques have been developed and manufactured, for example, CD (Compact Disc) players, PCM recorders, DATs (Digital Audio Taperecorders), digital surround processors, pre – amplifiers with digital audio inputs, BS (Broadcast Satellite) tuners. One of the most important technologies is the digital signal processing. And its progress depends on LSI technology.

Now the quarity of A/D and D/A conversion systems are thought to be a main factor of the performance of digital equipments. And oversampling method have been generally used in audio signal reproduction systems. Recently 4 – times oversampling digital filters have been adopted in many CD players. The function of this digital filters are to operate the data to be interpolated and to insert these data between the input data. In the frequency domain, the results mean that the image noise spectra above the audio band which is inevitably contained in digital signal is pushed away much higher frequency range. Thus analog post-filters following D/A convertors can be easy to suppress image noise above audio band.

We have developed a 2-times oversampling digital filter LSI in 1984 [1] and a 4-times oversampling digital filter LSI in 1985. And in 1987 we have developed a 1/2 decimation digital filter LSI for oversampling A/D conversion system of digital recording equipments [2].

We have recently developed an 8-times oversampling digital filter LSI with much higher performance and several other functions by 2-micron molybdenum gate CMOS process. We have designed this digital filter for some purposes below.

(1) Suppressing the quantization noise produced by both the internal digital signal processing and the output roundoff as much as possible, not to spoil the 98dB dynamic range of 16-bit PCM input signals.
(2) Digital signal processing of de-emphasis and attenuation etc. which have been done by analog curcuits.
(3) Corresponding to multiple system clocks and multiple sampling rates.
(4) Preventing from the reduction of audio quality caused by clock jitter of digital data transmission.

## 2. Construction

Fig.1 illustrates the signal flow of digital signal processing system of this LSI. This system is made up of six digital signal processing parts below.

(1) 1st 2-times oversampling FIR digital filter
(2) 2nd 2-times oversampling FIR digital filter
(3) De-emphasis IIR filter
(4) 3rd 2-times oversampling FIR digital filter
(5) Digital attenuation
(6) Noise shaping

In the following, each part of digital signal processing system will be discussed in terms of their functions and characteristics.

## 3. Oversampling Digital Filter

The function of oversampling digital filter is to suppress the quantizing noise above the audio baseband. Using this filter, high-performance reconstruction filters are not necessary for D/A conversion systems. And the oversampling filter can be designed to have a linear phase response by using linear phase FIR (Finite Impulse Response) filter.

Our developed 8 - times oversampling filter is realized by the cascade of 3 linear phase FIR filters of which each function is 2 - times oversampling. The 1st stage is 153 - tap FIR filter, the 2nd stage is 29 - tap FIR filter and the 3rd stage is 17 - tap FIR filter as illustrated in Fig.1. And Fig.2 illustrates how the 8 - times oversampling is realized by the cascade of 3 FIR filters. The total frequency response is that tha passband ripple is within ±0.00005dB and the stopband attenuation is abcve 110dB. The total transfer characteristics of this 8 - times oversampling filter is illustrated in Fig.3.

By the way, there are three kinds of sampling frequencies to be recommended as standards. These are 48kHz, 44.1kHz and 32kHz. And required passband - widths are 22kHz, 20kHz and 15kHz respectively. The permitted maximum sampling frequency of this LSI is 48kHz. Both the bandwidths of passband and stopdand in this 8 - times oversampling filter are shown in Table 1 at each sampling frequency.

Using this filter, the pre - echo and post - echo (derived from the passband ripple) and the intermodulation effects (produced by residual stopband quantizaing noise) will become almost negligible levels. In a word, the culculation in a FIR filter is to convolve the input signals with the coefficients corresponding to the impulse response of the filter. We have decided the three FIR filters' coefficients by computer simulations. As the results, we have adopted 22 bits fixed - point word length for the filter coefficients. And the convolutions are operated by one 20X22 - bit multiplier/25 - bit accumulator at the higher rate than 13MHz.

## 4. Digital De - emphasis

In the formats of CD and DAT, digital recorded signal can be emphasized at the higher frequency range. And to reproduce these pre - emphasized signal, de - emphasis operation is needed. In these formats, pre - emphasis and de - emphasis circuits are defined by 2 time - constants. Usually these circuits are realized by analog circuits only. Fig.4 shows the de - emphasis characteristics of CD format. And Fig.5 illustrates some examples of analog de - emphasis circuit.

Generally, the transfer function of de - emphasis is represented by following equation.

$$Ha(j\omega) = \frac{1 + j\omega T2}{1 + j\omega T1} \quad --- \quad (1)$$

where $\omega = 2\pi f$ (f = analog frequency), $T1 = 50\,\mu s$, $T2 = 15\,\mu s$.

Now the phase characteristics of this transfer function is not linear phase. To replace non-linear analog filters like this to digital filters, IIR (Infinite Impulse Response) filters are more suitable than FIR filters, because IIR filters can be made by transforming analog characteristics into digital of it. We show the process of designing the IIR filter in the following.

At first, the Laplace transformation of this de-emphasis analog circuit is shown below.

$$Ha(s) = a0 + \frac{a1}{s - b} \quad --- \quad (2)$$

where $a0 = T2/T1$, $a1 = (1 - T2/T1)/T1$, $b = -1/T1$.

And the impulse response of this is represented below.

$$ha(t) = a0 \cdot \delta(t) + a1 \cdot \exp(bt) \cdot u(t) \quad --- \quad (3)$$

where $\delta(t)$ is unit delta function and $u(t)$ is unit step function.

So the unit sample response of digital filter should be as follows

$$\begin{aligned} h(n) &= ha(nT) = a0 \cdot \delta(nT) + a1 \cdot \exp(btn) \cdot u(nT) \\ &= a0 \cdot \delta(nT) + a1 \cdot (\exp(bt))^n \cdot u(nT) \quad --- \quad (4) \end{aligned}$$

where T = cycle time of sampling, n = integer.

By z-transformation of equation (4), we can get the system function of it.

$$H(z) = \frac{a0}{T} + \frac{a1}{1 - \exp(bt) \cdot z^{-1}} \quad --- \quad (5)$$

Replace $T \cdot H(z)$ to $H(z)$ and put $z = \exp(j\omega T)$,

$$H(e^{j\omega T}) = a0 + \frac{a1 \cdot T}{1 - \exp(bt) \cdot \exp(-j\omega T)} \quad --- \quad (6)$$

The equation (6) shows the transfer function of this IIR digital filter. And this filter have 3 coefficients, a0, $(a1 \cdot T)$ and $\exp(bT)$.

Fig.6 shows the system configuration of this IIR filter. In the real de-emphasis IIR filter of this LSI, $T = 1/(4 * fs)$; where fs = original sampling frequency. That means the digital de-emphasis function is operated after the 4-times oversampling digital FIR filter stage as shown in Fig.1.

By the way, in every sampling frequencies (48kHz, 44.1kHz and 32kHz) the analog de-emphasis characteristics are the same, but in digital filter to change sampling frequency means to change filter characteristics. So the IIR filter in this LSI has 3 sets of de-emphasis coefficients and changing coefficients can be controlled by the 2 external signals (FSEL1, FSEL2). Fig.7 shows the passband characteristics of the 8-times oversampling output when the de-emphasis ie ON . The deviation of this IIR filter from the ideal characteristics is following. The deviation of amplitude is within ±0.001dB, and in the phase response the deviation is within 1.5 degrees.

From the beginning, the purpose of de-emphasis is to improve the signal to noise (S/N) ratio in the higher frequency range of audio band. On the other hand any digital operations generally add new quantizing noise to the signal. So digital de-emphasis operation at the original sampling rate fs and 16-bit output (16-bit roundoff) brings about increasing quantizing noise at the higher frequency range of audio baseband and decreasing the value of digital de-emphasis. To prevent this contradiction, this LSI prepares the following functions.

(1) 8-times or 4-times oversampling output.
(2) capable of noise shaping.
(3) capable of 18-bit or 20-bit output.

By using these functions together with de-emphasis filtering, sufficient dynamic range of signal can be realized.

## 5. Digital Attenuation and Soft-Muting

The gain control of digital signal is to multiply one coefficient to the data of signal. This LSI has 0.188dB-step attenuation, so volume control is possible at 512 steps from 0dB to -96dB. This attenuation is controlled by internal two 9-bit attenuation registers for 2 channels. It is both capable to set attenuation level directly from the external serial data interface and to increase or decrease the level step one by one. Thus by controlling the attenuation registers' contents, this function can be used for a digital volume. The function of attenuation is operated by the 20x22-bit multipler and a look-up table ROM (from logarithm to linear) and shift registers.

Furthermore, by using this attenuating function the soft-muting is possible. There are two muting modes in this LSI. One is soft-mute and the other is direct-mute. In case of soft-mute mode, as the muting function is operated softly, the generation of noise by muting ON or OFF is suppressed. The slope example of soft-muting is shown in Fig.8. The slope draws an exponential curve.

## 6. 18 - bit Input and 18 - bit or 20 - bit Output

This digital filter LSI takes MSB first serial data format for the input /output data. The input data may be specified as 2's complement and the output data may be specified as 2's complement or COB (Complementary Offset Binary). In the formats of CD and DAT, the word - length of PCM signal is 16 - bit. And the dynamic range of this digital signal is about 98.1dB. But, processing this signal by any circuits means increasing quantization noise. So the dynamic range of the processed signal is decreased below 98. 1dB. We have taken account of this incresing of quantization noise when designing this LSI.

As the digital signal processing is done at the rate of original sampling frequency (fs) and output word - length is 16 - bit, the output roundoff noise will be rather large. Fig.9 shows the quantizing noise level produced by roundoff at various oversampling rates and output word - lengths. In Fig.9 the quantization noise level is considered only within audio baseband. So the output roundoff noise becomes less as higher as the oversampling rate and as larger as the output word - length. Furthermore in case of using digital de - emphasis and/or digital attenuation, it is necessary to prevent decreasing the dynamic range of signal as little as possible ,

In that meaning this LSI can output 18 - bit or 20 - bit output. On the other hand, the reason for capability of 18 - bit input is following. If the 16 - bit souce signals are processed at the original sampling rate fs (for example, digital tone - controlling, digital equalization and digital reverberation etc. ) and oversampled by this LSI, the roundoff noise after the pre - processing is not negligible shown in Fig.9. By incresing output word - length from 16 - bit to 18 - bit, the roundoff noise level is decreased at the rate of 1/4.

## 7. Noise - shaping

Quantization noise is produced at both the cases, when analog signal is digitized and when digital signal is rounded off at the less bits. But, in the case of sampling frequency is much higher than audio baseband, quantization noise can be reduced by noise shaping operation. This operation is to compensate for the error associated with the quantization of the preceding sample by adding the error to the succeeding sample with reversed porarity. As the results, the shape of noise density as a function of frequency is modified as the density is decreased in the audio baseband.

In this LSI, the 1st order noise shaping operation is capable to be used by external mode selection. At the noise shaping mode, the internal 25 bits data is rounded-off to N bits (where N is 16, 18 or 20), and then the rounding-off error (25 - N) bits is added to succeeding data. Fig.10 illustrates the noise shaping characteristics of this LSI.

## 8. Architecture

The block diagram of this LSI is shown in Fig.11. This diagram is made in the aspect of function. The main function blocks are the filter operation and the volume operation blocks. Other blocks are input/output interface, output controllor, mode controllor, volume controllor, timing controllor etc. The filter and volume operations are carring out by using the same 20x22-bit multiplier and 25-bit accumulator. The mode is mainly set by an external micro-processor.

This LSI is designed to be free from the jitter of input data and clocks. This digital filter is operated by system clock. The system clock can be selected among 192fs, 384fs, 256fs and 512fs. Output data of 2 channels are given at independent output terminals. And as the output data and clocks are synchronized by the system clock, their timing resolution is almost the same of system clock. So if the system clock is generated by the X'tal near this LSI, the jitter of the output data and clocks to D/A convertors and sample/hold circuits will be very small. And the output bit clock rate is either 192fs or 256fs at 8-times oversampling. The timing diagram of the output is shown in Fig.12.

At last, this LSI is fabricated by 2-micron molybdenum gate CMOS process. The chip is 7.8mm x 7.1 mm size and is integtated about 60,000 transistors. Fig.13 shows a photograph of this chip.

## 9. Conclusion

A multi-functional 8-times oversampling digital filter for digital audio use has been described about the functions and characteristics. This digital filter is developed to be employed high-performance oversampling D/A coversion systems. Particuraly we have taken into consideration the quantizing noise by output roundoff and internal digital sinal processing.

And we have tried to process de－emphasis filtering and attenuation digitally in a chip. Thus this digital filter can be applicable in various system configurations and interfaces. The authors hope this digital filter will advance the quality of the digital audio equipments.

## 10. Acknowledgments

This LSI and this article would not have been possible without the enthusiastic cooperation from many individuals and companies. The authors are grateful for their help.

## References

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[2] S.Kakiuchi, H.Iizuka, M.Chijiiwa and T.Ohtsuka, " Application of Oversampling A／D and D／A Conversion Technique to R－DAT," Proc. 83rd AES Convension, New York, Oct.1987.

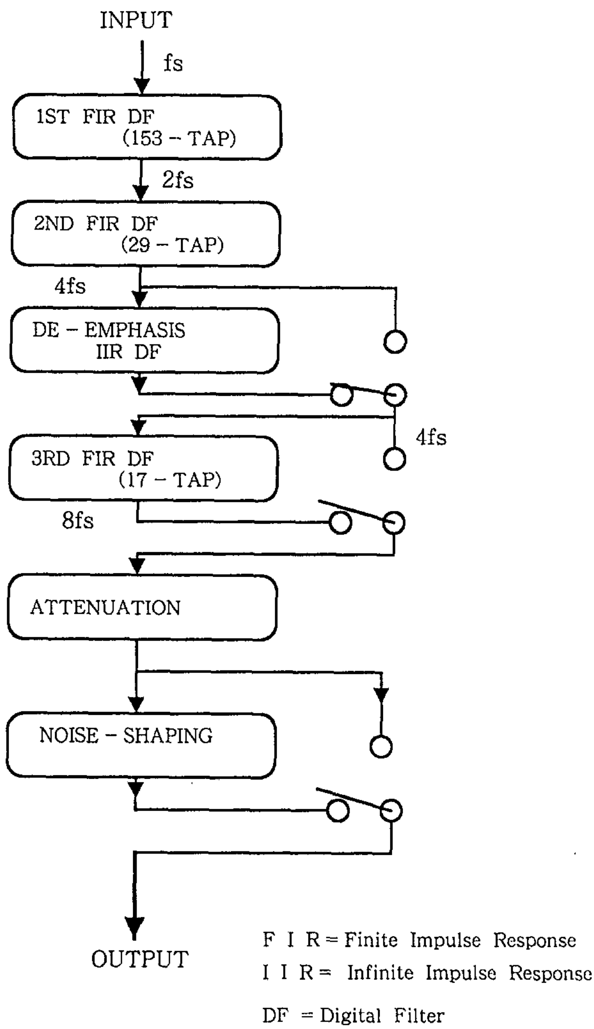



Fig.1 Signal flow of the digital signal processing system

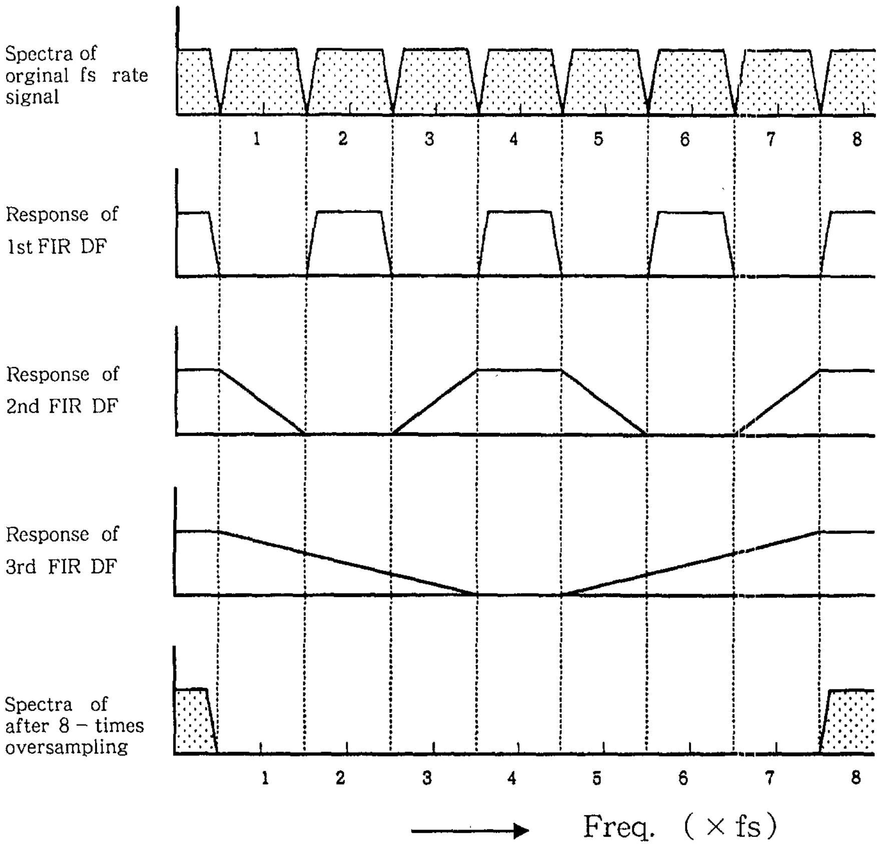



Fig. 2 8-times oversampling by the cascade of 3 FIR filters

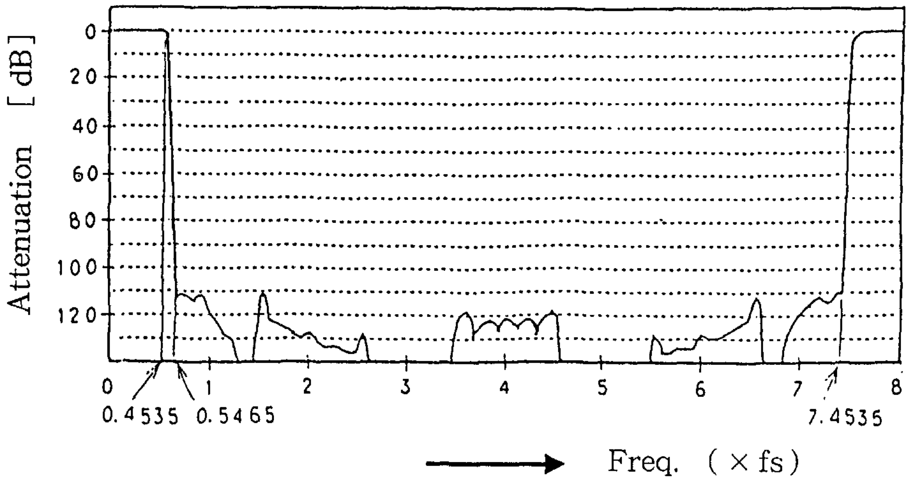



Fig.3 Transfer characteristics of the 8-times oversampling filter

Table 1 Characteristics of the filter at 3 standard sampling frequencies

| Item | | Passband | Stopband |
|---|---|---|---|
| | | 0～0.4535fs | 0.5465～7.4535fs |
| fs | 44.1kHz | 0～20.00kHz | 24.10～328.7kHz |
| | 48.0kHz | 0～21.76kHz | 26.24～357.7kHz |
| | 32.0kHz | 0～14.51kHz | 17.49～238.5kHz |
| Specifications | | Ripple | Attenuation |
| | | < ± 0.00005dB | > 110dB |

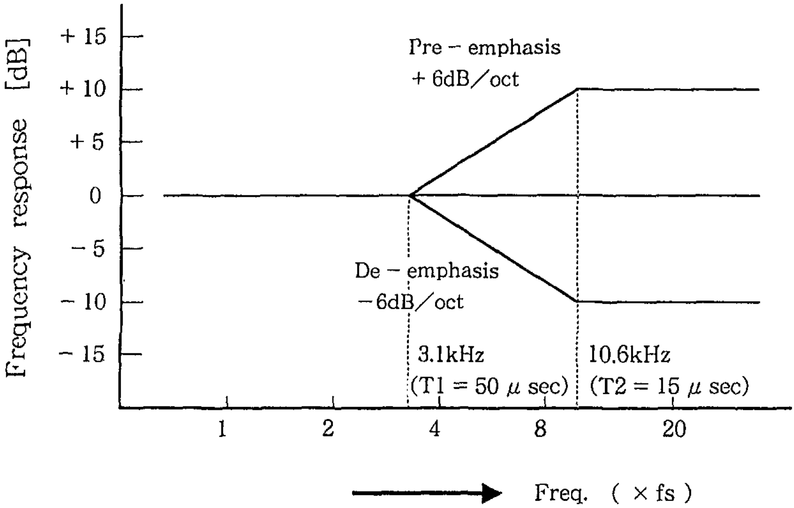
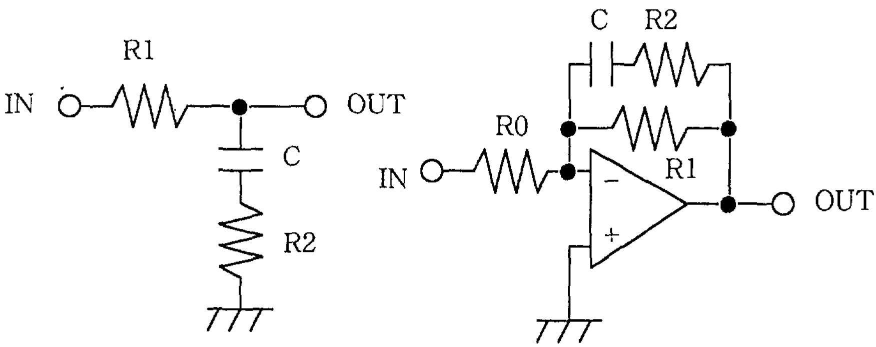



Fig.4 Pre-emphasis and de-emphasis characteristics of CD format




$T1 = C\ (R1 + R2) = 50\ \mu\,\mathrm{sec}$

$T2 = CR2 = 15\ \mu\,\mathrm{sec}$

Fig.5 Examples of analog de-emphasis circuit

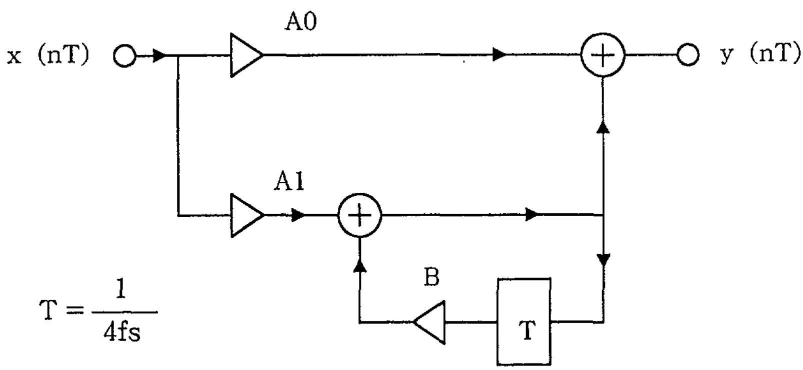



Fig.6 Signal flow graph of the IIR filter for de-emphasis

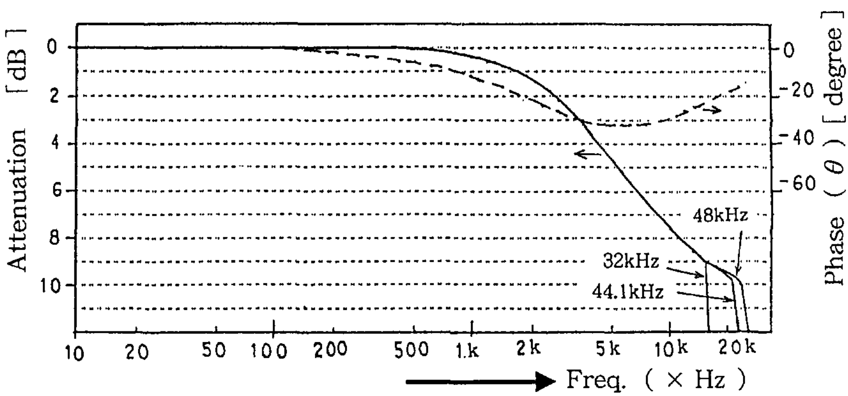



Fig.7 Passband characteristics of the 8-times oversampling output ( de-emphasis ON )

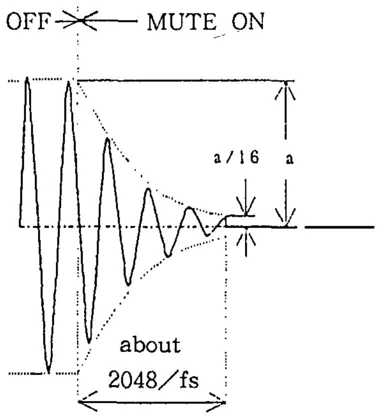



Fig.8 Attenuation characteristics of the soft-mute mode

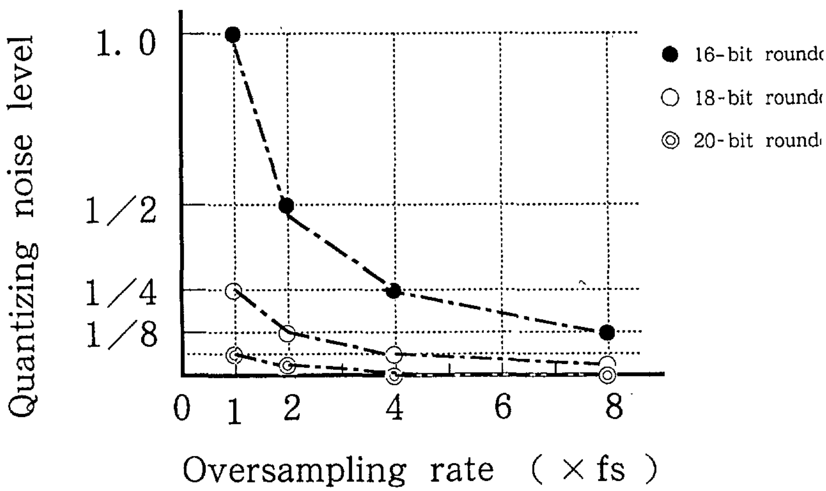
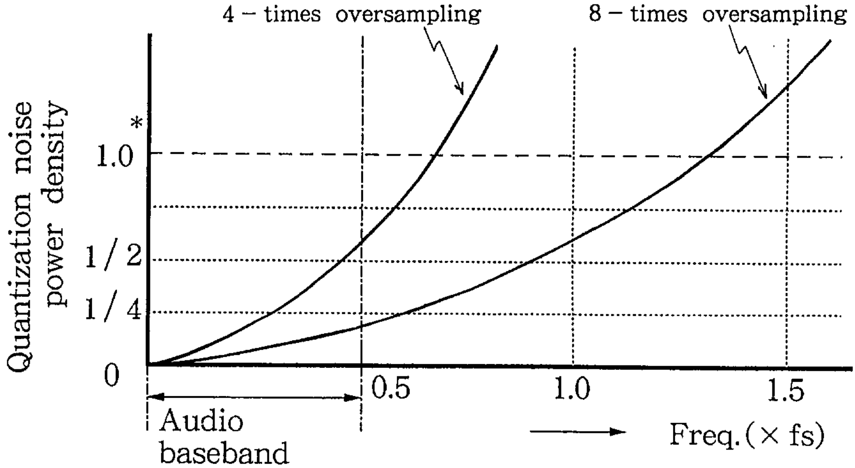
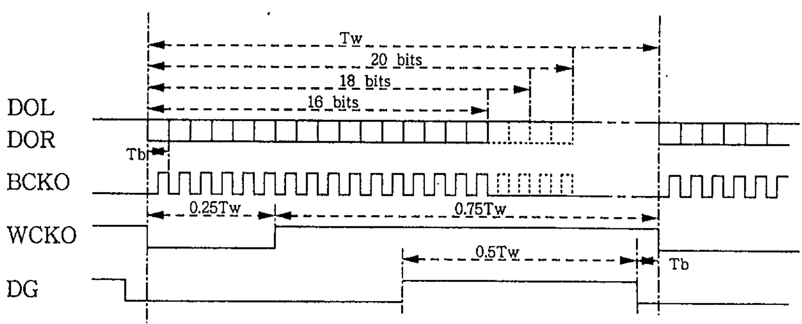



Fig. 9 Quantizing noise level produced by roundoff




* Normalized scale that quantizing noise power is 1.0 at the 4 – times oversampling without noise – shaping.

Fig. 10 Characteristics of noise-shaping




Tb = Tsys,

Tw = 24 * Tsys (at 192fs), 32 * Tsys (at 256fs)

where Tsys is the cycle time of internal system clock (192fs or 256fs).

Fig.12 A timing diagram of the outputs

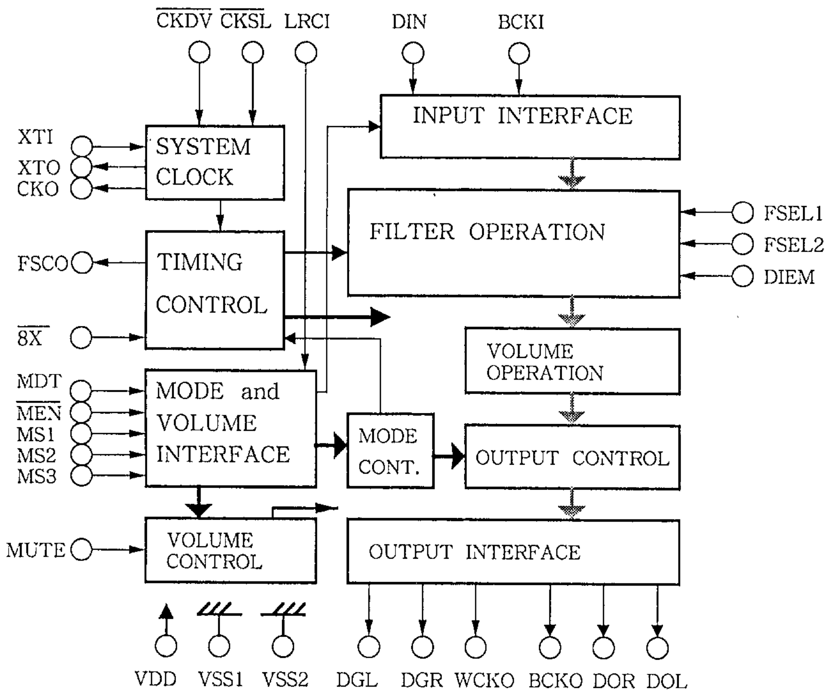



Fig. 11 Block diagram of the multi-functional digital filter LSI

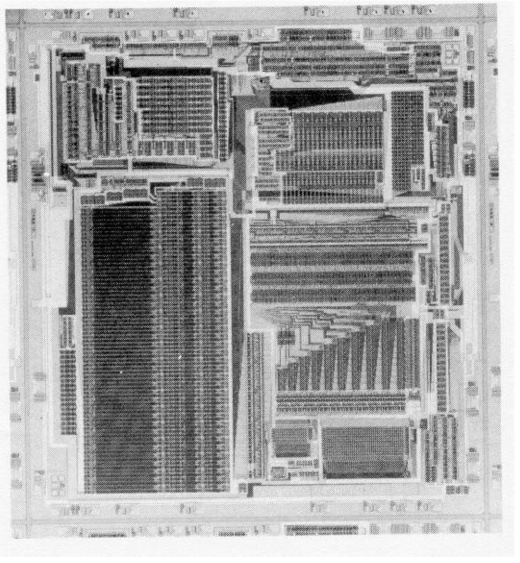


Fig.13 Photograph of the LSI chip