DIGITAL TO ANALOGUE CONVERSION USING NOISE SHAPING AND OVERSAMPLING FOR DIGITAL AUDIO APPLICATIONS

Dr M.J. Hawksford and B.A.H. McCrea

University of Essex, Wivenhoe Park, Colchester, Essex

The performance of a noise shaping filter architecture in association with a digital to analogue converter is investigated using computer simulation. Results of relevance to digital audio are presented to demonstrate resolution and the decorrelation of high-level DAC errors.

1. INTRODUCTION

This paper presents results of computer simulations to show the performance advantage in using a noise shaping filter in association with oversampling. Particular attention is given to digital to analogue converter (DAC) large scale errors that are characterised by both random and systematic displacements from the optimum reconstruction levels. Noise shaping is demonstrated to induce an effective high level, spectrally weighted intelligent dither that translates DAC errors of both static and dynamic classifications to benign noise, within the bound of source data quantisation. Of particular interest is the observation that non-linear DAC errors take on a spectral image of the noise shaped dither that reduces inband distortion and is attractive because the DAC is operated open loop.

The technique of oversampling has already been successfully adopted for digital audio reproduction [1], where the most common oversampling ratios are x 2 and x 4. The principal advantages are, first, a reduction in analogue circuit complexity where reconstruction filtering is now performed substantially within the digital domain, and second, an enhancement in DAC resolution by using error correction feedback as a noise shaper. Effectively, a 14 bit DAC operating at 176.4kHz can yield the same resolution as a 16 bit DAC operating at 44.1kHz, providing exemplary linearity of the two DACs. Though the present oversampling schemes address the areas of low-level nonlinearity, they are not effective at controlling high-level errors spanning many quanta that may also have elements of dynamic distortion as a function of inter-sample transitions.

This paper presents results based upon spectral analysis of computed data sequences to show how a R^{th}-order level compaction, noise shaping algorithm can both reduce the number of bits required in the output code of the interpolation scheme and effectively decorrelate the high-level DAC errors, even when the DAC is driven open-loop as illustrated in Fig.1.

2. LEVEL COMPACTION, NOISE SHAPING ALGORITHMS

The conversion of a PCM code {Mbit/f_{s1} Hz} to {Nbit/f_{s2} Hz}, where $f_{s2} > f_{s1}$ and signal bandwidth is constant, allows N < M, by taking account of intersample averaging. The sample interpolation is assumed to be performed by a digital low-pass filter and subsequent level compaction by a cascaded noise shaping algorithm, where the basic scheme is shown in Fig.1.

The structure of the compaction algorithm [2] is shown in Fig.2 and is generalised to order R. Effectively, the algorithm is an extension of a multi-level delta sigma modulator (DSM) [3,4] but with R cascaded integrators in the forward path, together with response shaping for stability. Integration takes the form $1/(Z-1)$ and no multiplication is required which aids high speed processing. The structure was shown in reference [2] to be derived from a digital deltamodulator (and also follows the work of Tewksbury [5] and Adams [6]), which offered exact signal transparency when the input data and linear quantiser Q-law are matched. However, for high-order loop filters, such structures are sensitive to rapid changes in input signal. Consequently, the reconfiguration to a delta sigma modulator type structure [7] is preferred as a constant input-output transfer function is offered, together with desirable noise shaping characteristics (see appendix).

3. COMPUTER SIMULATION SYSTEM PARAMETERS

The following data was used to drive the computer simulation:

Sinusoidal input signal frequency = 2 kHz

Input signal amplitude = $\sqrt{2}$ volt

Noise shaper sampling frequency (f_{s2}) = 512kHz

Input quantum step = 2^{-16} volt
(i.e. 16 bit data reference 1 volt pk-pk input)

Q - quantum (Fig.2) nominal interval = 1 volt

DAC non-ideality is a superposition of both random and systematic errors, where the selected data is shown in Fig.3. The DAC is configured as a 5 bit device where the overall error bound is set in the present example to 13 bit. In practice, a parallel (non-segmented) DAC structure can be designed where accuracy should exceed 13 bit and parallelism minimise glitch generation.

Commencement of text for subsequent pages

4. TIME DOMAIN PERFORMANCE OF COMPACTION ALGORITHM

To illustrate the time domain performance, Fig.4 shows four output sequences of the noise shaping filter for orders R = {1,2,3,4}. As the filter order is advanced, output activity progressively increases where for R = 1, five levels are excited and for R = 4 approximately 15 levels are excited. It should be observed that the constellation of output samples take on a random-like structure that include the majority of DAC levels in the conversion process. However, the noise shaper forms an intelligent high-level dither that, when averaged, yields an accurate estimate of the input data sequence. To show the distribution of levels excited over the computational period, Fig.5 illustrates the histograms corresponding to the orders R = {1,2,3,4}in Fig.4. It is important to observe that, by limiting the input signal to just over one Q-quanta, the pseudo-random activity offers histogram distributions almost independent of signal level, a factor which enables decorrelation of DAC errors that is particularly effective for higher order systems (e.g. R $\gg$ 4).

5. SPECTRAL DISTRIBUTION OF ERROR DERIVED FROM A NON-IDEAL DAC

The output "DAC error" illustrated in Fig.1 enables the DAC only distortion to be interogated, where the results are shown in Fig.6(a), (b). As indicated in Section 3, the DAC error is the superposition of both random and systematic distortions. The error spectra were estimated for input data derived from a R = 4 noise shaper. Fig.6(a) shows the error spectrum for the random component, while 6(b) shows the spectrum for the non-linear contribution. It can readily be observed that the random element broadly results in a flat spectrum while the non-linear contribution takes on a spectrum that is weighted by the noise shaped activity derived from the noise shaping filter. Consequently, systematic errors translate into a noise-like spectrum and usefully exhibit a spectrum with a lower density within the audio band.

The spectral plots shown in Fig.7 illustrate the overall DAC errors using the error example of Fig.3 for orders R = {1,2,3,4}. As anticipated the total error rises slightly as the order is advanced due to the greater DAC activity.

6. SPECTRAL DISTRIBUTION OF ERROR DERIVED FROM OVERALL SYSTEM

The final results shown in Fig.8 demonstrate the overall system spectral distortion that includes both the noise shaper error and the DAC error, for R = {1,2,3,4}. Spectral estimates were made both before and after the DAC and superimposed for comparison.

7. CONCLUSION

This paper has demonstrated the performance of an R^{th}-order compaction algorithm that is a derivative of a digital, multi-level DSM coder. The compaction algorithm was presented both as a method of reducing the number of bits in the oversampled format without incurring a loss of input data and to decorrelate large-scale errors in DACs.

Unlike DSM, the loop order of the compaction algorithms can exceed two without incurring a stability penalty, providing the quantiser Q does not saturate. The results show that as R is advanced Q activity increases with a corresponding improvement in noise shaping characteristics. Although the DAC is outside the feedback loop, the high-level, near constant power input translates DAC errors to benign noise, which is a most desirable attribute for high performance digital audio, especially when DAC errors are a function of intersample dependence yielding dynamic distortion. Our investigation has concluded that for R $\geq$ 4 decorrelation is complete.

There are three principle error sources within the system. Overall, there is the imposed bound dictated by quantisation of the input data. Second, there is the error determined by the noise shaper/compaction algorithm and, thirdly, the DAC errors. For the present oversampling ratio of x 50, noise shaper distortions are reduced as the order R is advanced. However, the increased DAC activity then yields a rising error contribution, though this is partially offset by the systematic errors being noise shaped, exhibiting a lower spectral density within the audio band. However, for a practical order R = 4, the DAC activity is less than 5 bit wide; consequently, parallel structures can be formulated with low glitch and area distortion that can readily operate at the necessarily high sampling rates. The results suggest that an accuracy of 16 bit is readily achievable which can be upwards expanded within practical constraints to 18 bit or better targets. In particular, the decorrelation of systematic errors is considered a significant advantage, together with the minimisation of idle channel patterns which is realised using the higher order noise shaper.

It is interesting to observe that, in combination, sample interpolation and compaction of level realise a continuum of codes where in the limit a 1-bit serial format is generated [8,9,10] The algorithm presented is readily configured to yield a 1-bit DSM format by converting the Q parallel output data to a serial bit stream using a ROM look-up table [2]. Thus, high-order noise shaping without the usual limits on stability by quantiser saturation can be achieved. This latter application is considered optimal for digital audio, requiring the minimum of additional analogue circuitry. Also in active loudspeaker systems where digital filtering can

Commencement of text for subsequent pages

be employed for crossover networks, a 1-bit conversion is attractive as output pins on VLSI circuits are efficiently utilised that aid cost-effective design. However, the ultimate resolution of the serial format demands that the area-per-bit be kept within a close tolerance and is both symmetric between 1 and 0 pulses and independent of pulse patterns.

In conclusion, Fig.1 shows the addition of a high-level dither signal (triangular wave) between noise shaper and DAC. This was introduced to broaden the histogram and further decorrelate systematic DAC errors. However, in this application no significant advantage resulted and, in practice, it led to a slightly higher in-band noise contribution due to increased DAC activity. The inclusion of this waveform followed a recent paper by Fielder [11], though we conclude that within the present system the placement of the signal within the quantiser Q makes this technique unnecessary.

REFERENCES

1. Goedhart, D., van de Plassche, R.J., and Stikvoort, "Digital-to-analogue conversion in playing a compact disc", Philips Tech. Rev. 40, No.6, 1982, pp.174-179

2. Hawksford, M.J., "Nth-order Recursive sigma-ADC machinery at the analogue-digital gateway", 78th AES Convention, Anaheim, May 1985, Preprint 2248 A-15

3. Steele, R., Deltamodulation Systems, Pentech Press, ISBN 0727304011, pp.323-327

4. Hawksford, M.J., "Deltamodulator coder using a parallel realisation", IERE Conference Proc. 37, Digital Processing of Signals in Communications, University of Technology, Loughborough, Sept. 1977, pp.547-557

5. Tewksbury, S.K. and Hallock, R.W., " Oversampled, Linear Predictive and Noise-Shaping Coders of Order N>1", IEEE Transactions on Circuits and Systems, Vol.CAS-25/7, July 1978, pp.436-447

6. Adams, R.W., "Design and implementation of an audio 18-bit analogue-to-digital converter using oversampling techniques", J. Audio Engineering Society, Vol.34, No.3, March 1986, pp.153-166

7. Inose, H.L. and Yasuhiko, Yasuda Y., "A unity bit coding method by negative feedback", Proc. Inst. Elec. Electron. Eng., 1963, 51, pp.1524-1535

8. Candy, J.C., "A use of double integration in sigma delta modulation", IEEE Trans. Comms., Vol.COM-33, March 1985, pp.249-258

9. Stritek, P., "Prospective techniques for improved signal-to-noise ratio in digital audio conversion systems", 82nd Convention of the Audio Engineering Society, March 1987 (paper available from: Institut fuer Nachrichtentechnik und Hochfrequenztechnik, Technische Universitaet Wien)

10. Flood, J.E. and Hawksford, M.J., "Exact model for delta-modulation processes", Proc. IEE, Vol.118, No.9, Sept. 1971, pp.1155-1161

11. Fielder, L.D., "Evaluation of the audible distortion and noise produced by digital audio converters", 82nd Convention of the Audio Engineering Society, March 1987, Preprint 2424, (A-5), Fig.18

APPENDIX

Validation of compaction topology (Fig 2(b)) to meet Tewksbury's [5] optimum transfer function criterion:

The transfer function for order R of the forward path,

$H_{fp}(Z)$ of Fig. 2(b) is,

$$H_{fp}(Z) = \frac{Z^{-1}}{(1-Z^{-1})} + \frac{Z^{-1}}{(1-Z^{-1})^2} + \dots + \frac{Z^{-1}}{(1-Z^{-1})^R}$$

$$\text{i.e. } H_{fp}(Z) = Z^{-1}\sum_{r=1}^{R} \frac{1}{(1-Z^{-1})^r} = \frac{1-(1-Z^{-1})^R}{(1-Z^{-1})^R}$$

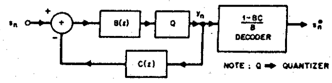

Fig. 2. General linear feedback coder/decoder.

(diagram from Tewksbury [5])

Setting c(Z) = 1, Tewksbury's [5] model shows that for the optimum case of minimum in-band noise for a noise shaping filter of order R,

$$B(Z) = \frac{1-(1-Z^{-1})^R}{(1-Z^{-1})^R} = H_{fp}(Z)$$

i.e. the selected topologies of Fig. 2(a), (b), represent the optimum case and realise the function $H_{fp}(Z)$ without multiplication factors.

mmencement o

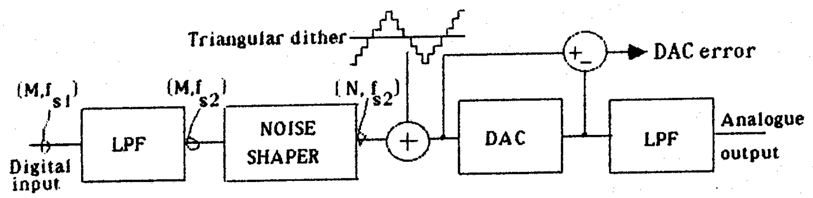

Fig.1 Basic oversampling scheme with level compaction and additive triangular wave.

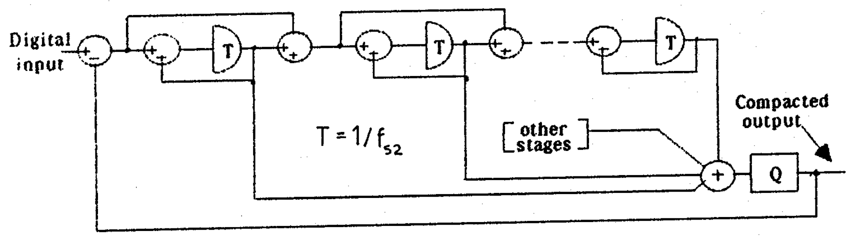

Fig 2a N-th order noise shaping filter for level compaction in DAC applications (ref 2) .

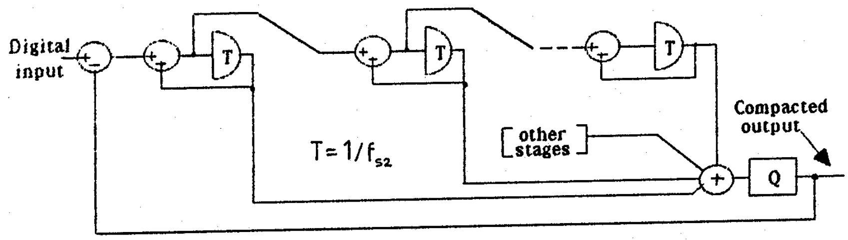

Fig 2b Simplified N-th order noise shaping filter (derived from Fig 2a) .

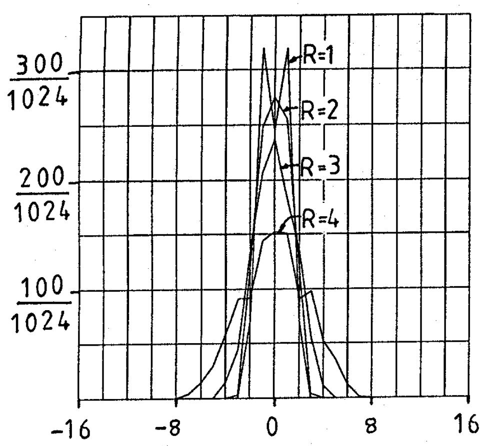

Fig. 5 Histogram distribution of quantisation levels for Fig. 4 examples for orders R = {1,2,3,4}

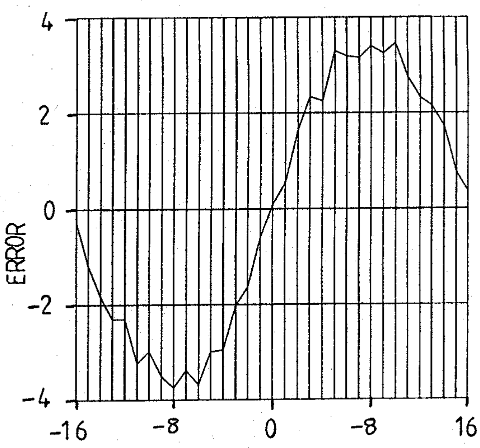

Fig. 3 DAC errors, non-linear square law + random superimposed on 5 bit output DAC (overall accuracy ≈ 13 bit)

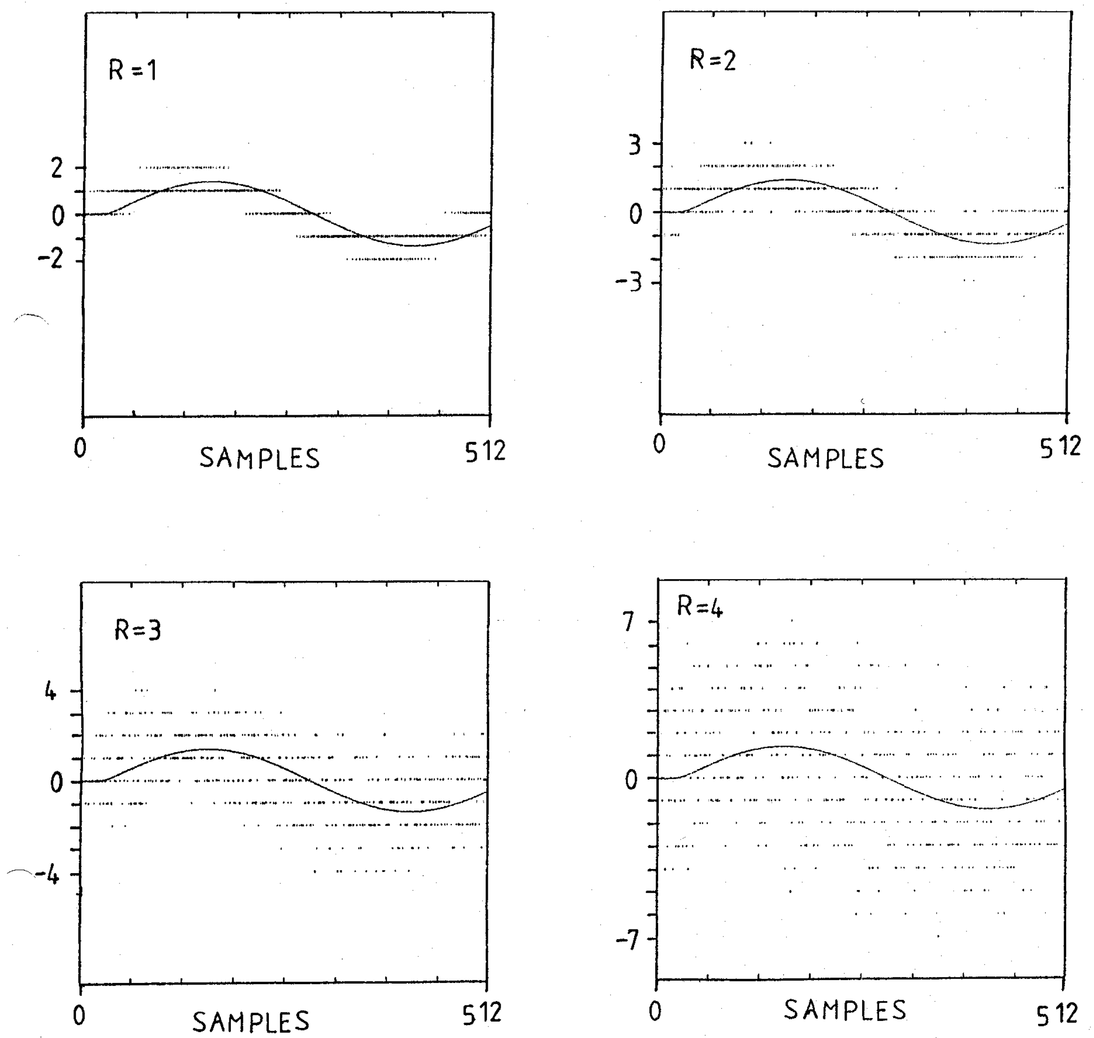

Fig.4 Time domain responses of compaction algorithm for $R = \{1,2,3,4\}$, sinusoidal excitation

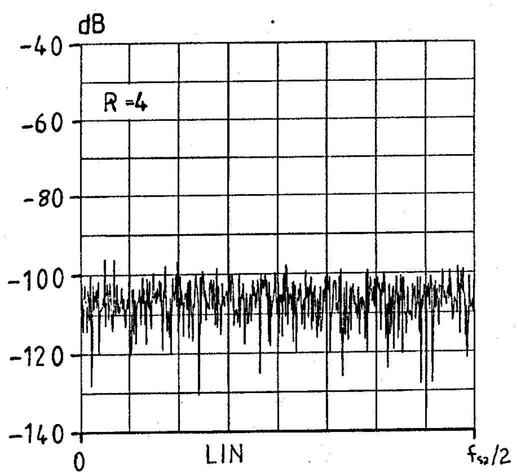

Fig. 6(a) R = 4, "DAC error" (Fig. 1) spectrum, random error only

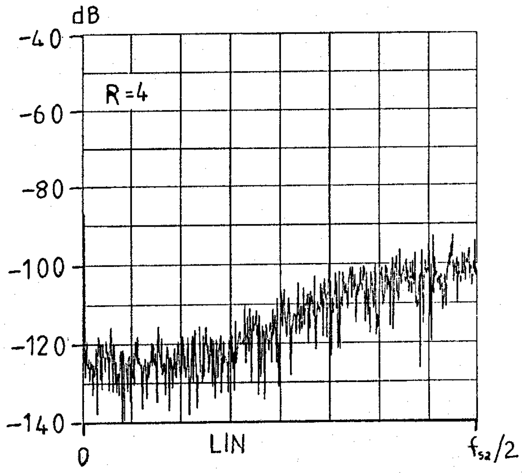

Fig. 6(b) R = 4, "DAC error" (Fig. 1) spectrum, non-linear error only

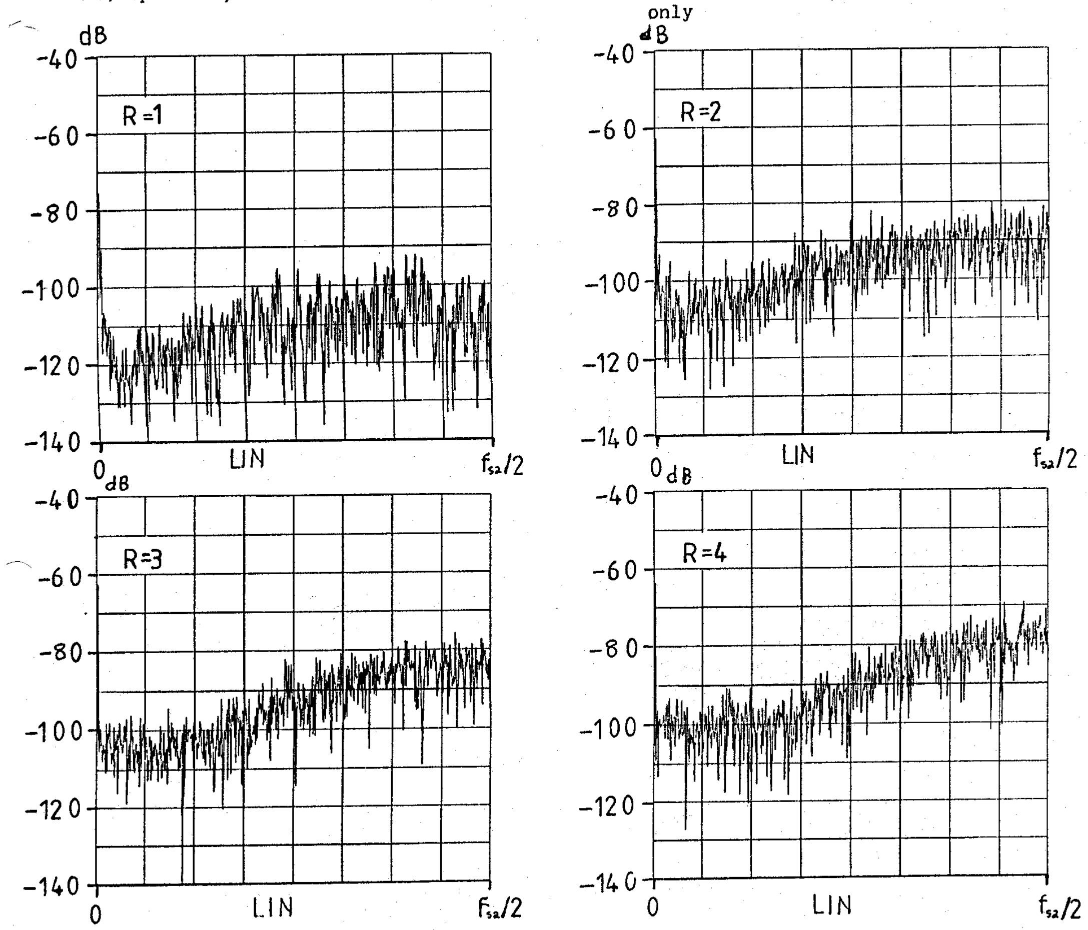

Fig. 7 R = {1,2,3,4}, "DAC error" (Fig. 1) spectrum, random + linear

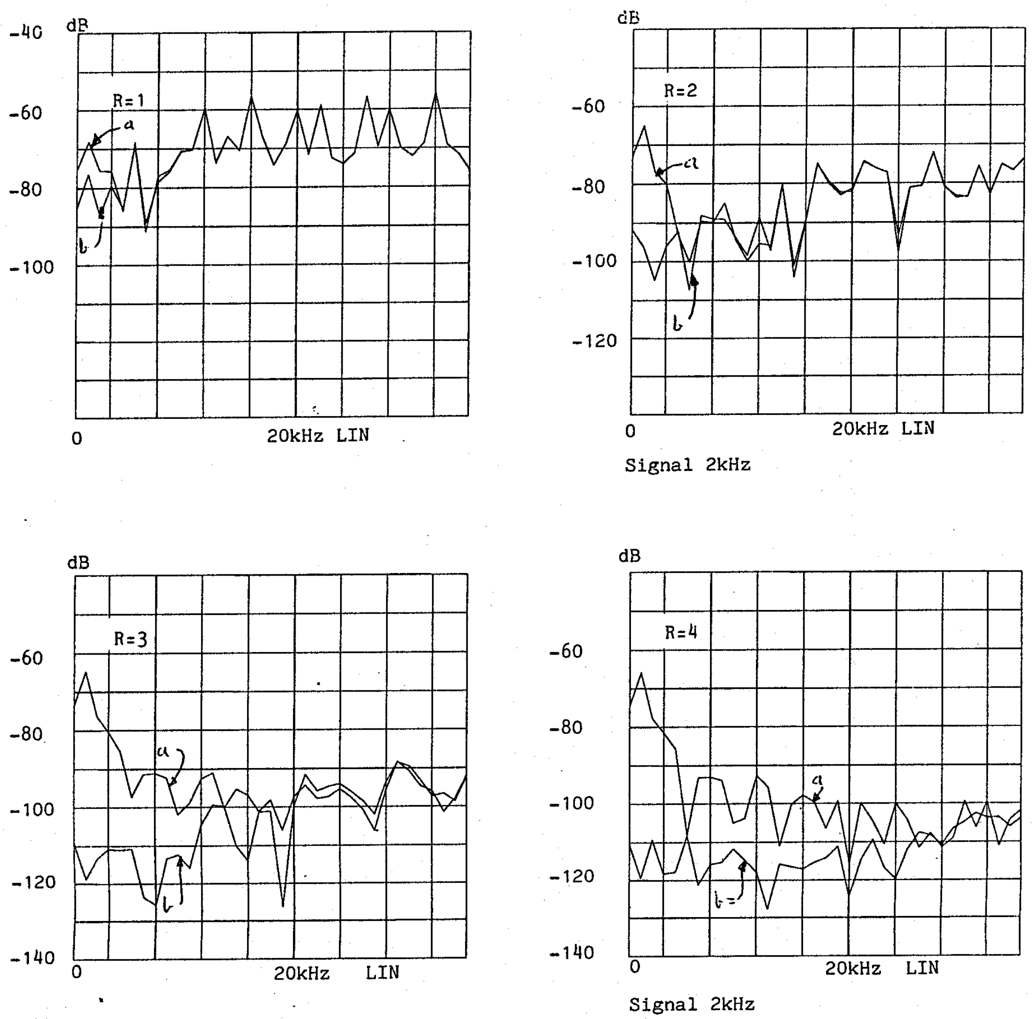

Figure 8: R = {1,2,3,4}, overall system distortion spectra before (b) and after (a) DAC (includes noise shaper distortion)