Anti-Aliasing Filters for Oversampling Analog to Digital Converters Preprint 3566 (D4-5)

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Anti-aliasing Filters for Oversampling Analog to Digital Converters

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Most oversampling D-A converter manufacturers recommend the use of a simple 1st order filter in order to implement anti-aliasing protection. A demonstration, using both theory and experimental work, is given to prove that for demanding professional applications, this is not enough. A corresponding filter selection criteria is derived.

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0 Introduction

With the traditional approach of using Nyquist sampled multi-level quantizer A-D converters, highly complex analog filtering was needed on the converter input in order to avoid aliasing problems. Typically, a 11th order Chebyshev filter was employed, leading to severe phase problems in the passband. More recently with the advent of oversampled converters, filter requirements were considerably relaxed, for example to a 7th order Butterworth to provide protection to a 2X oversampling converter [1]. Latest developments on one-bit quantizer converters employ still larger oversampling ratios, 64X being typical. Manufacturers of such integrated converters are unanimous in saying that a single R-C 1st order filter placed at the converter input is enough in order to offer sufficient anti-aliasing protection [1][2][3][4].

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1 Nyquist rate converters

Basically two groups of A-D converters exist. The first of the groups employ Nyquist rate conversion. This means that input signals are sampled at a frequency f_s, fact which restricts the maximum available bandwidth for input signals to about $f_N=f_s/2$, f_N being called the Nyquist frequency. As examples of this converter group we have the conventional successive approximation, flash and double integration converters. All employ quantizers with a precision equivalent to the required performance. Process of the conversion is as outlined in fig. 1. The nature of sampling is equivalent to multiplying the input signal by an hypothetical carrier signal having frequency components nfs, where n=0..infinity. This in turn implies that input signals in excess of the Nyquist frequency will be seen at the output as sum and difference components relative to the frequency of the hypothetical modulated carrier. The difference components may have frequencies that fit in the baseband and create components unrelated to the useful input signals, that appear as gross distortion, intolerable to the ear, and known as aliasing distortion. To prevent this, baseband signals must be strongly low-pass filtered. This implies that a guard bandwidth must be created between the maximum allowable frequency in the baseband and the Nyquist frequency f_N, and that the filter has to reach an attenuation equivalent to the required precision at f_N. Normally this leads to the use of a very steep filter of the elliptic type with poor phase and transient response characteristics. Let's take a typical example for the sake of clarity:
If the sampling frequency of a 16 bit converter is 48kHz (widely used for professional applications) and the useable bandwidth is to be 20kHz, f_N will be 24kHz. The filter must pass intact (or say, not to attenuate more than 0.1dB) a 20kHz signal, but must attenuate a 24kHz signal by about 96dB, coming from the following formula:

1) $Att=20\log_{10}(2^{-n})$

where n is the number of bits. The process is pictured in fig. 2 and will need for an elliptic filter of 11th order, bearing a net 700 degrees phase shift at 20kHz. This is of course unacceptable by present quality standards.

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2 Oversampling converters

The second group of converters work by sampling the input signal at a multiple of the sampling frequency used by the Nyquist criteria. In this case, no guard band is necessary, because the distance between the baseband and the aliased frequency bands will be greater and directly proportional to the oversampling ratio. Anti-aliasing filter requirements can thus be considerably relaxed, conducting for example to a 7th order Butterworth for a 2X oversampled converter. Modern oversampled converters employ 64X oversampling [3][4][5], and since the distance between the usable 20kHz bandwidth and the first aliased frequency band is so great that manufacturers recommend the use of a single low-pass filter at the input to do the anti-aliasing job. All the converters in reference use noise-shaping techniques and a 1-bit quantizer inserted into an integrating feedback loop, which conducts to great amounts of out-of-band noise that must be filtered out. Anti-aliasing is partly effected through the decimator filters, usually of the FIR type, which are optimized both to reduce the sample rate and to provide the necessary noise and anti-aliasing rejection. Amplitude and phase response can be made arbitrarily flat, depending on filter complexity.

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3 Anti-aliasing filtering considerations

It is normally assumed when a single pole lowpass anti-aliasing filter is specified that internal digital filters will provide the necessary rejection for signals relatively near the baseband, leaving the rejection to signals far from the baseband to the single-pole filter.
When there are no signals present with significant energy at frequencies near the sampling frequency the assumption may hold. However, if we consider a professional system working at 48kHz using a 64X oversampling converter and under common broadcasting conditions or in the vicinity of other digital devices, we immediately will revise the previous assumption and will face the necessity of ensuring the required performance under worst-case conditions.
Filters used for decimation purposes are sampled filters. It appears that at least a part of the filters built in the currently available noise-shaping converters are of the comb-filter type, requiring no multiplications, followed by a more steep FIR. Their response may be optimized to afford the necessary rejection just leaving the usable band, but have the drawback of not rejecting very well the aliased bands in the vicinity of integer multiples of f_s. Thus, if the needed anti-aliasing rejection must come exclusively from the analog input filter, the requirements will be the following:

Filter passband (-0.1dB)=f_B
Filter transition band=nf_s-2f_B, where n is the oversampling ratio
Attenuation in the stopband= as given per equation 1

Let's take our 20kHz, 16 bit converter already used on the previous example, this time using 64X oversampling. Requirements as above stated conduct to a 5th order Bessel filter with 30 degrees maximum phase shift at 20kHz. The filter is not too complex and will ensure a good performance. If the designer wants to avoid the complexity and the added noise coming from the 2 operational amplifiers needed to

implement the 5th order filter using a VCVS configuration, a 3rd order Butterworth will do the job, conducting to 105dB rejection at the vicinity of the sampling frequency, and an added phase shift of 60 degrees at 20kHz. Cutoff of this filter should be placed at 44kHz, in order to introduce -0.05dB of attenuation at 20kHz.
However, sometimes the used internal chip architectures do provide some usable rejection for signals in the vicinity of f_s, coming from internal filtering. In that case, the only modification that is necessary is to deduct the rejection provided internally from the calculated attenuation value.

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4 Experimental work

We took a converter of the type mentioned in [4] and proceeded to investigate its behavior using an nfs of 128kHz, which corresponds to a net fs on the output of 2kHz. The reason to use such a low sampling rate is that it is extremely easy to generate very pure sine waves which frequency is in the vicinity of nfs, and the fact we are relatively free from the extensive shielding needed to operate a converter at 3.072MHz, needed for a net 48kHz output sampling rate. The price to pay is a strong degradation of the distortion specifications for in-band signals, probably coming from the use of a switched-capacitor integrator in the quantizer feedback loop. Degradation is thus the consequence of operating the chip clearly out of the recommended operating conditions, leading to about 9% THD at 100Hz, as pictured in fig.5.
We then proceeded to calibrate the converter in terms of amplitude and injected in its input a 127900Hz sine wave near 100% full scale, to observe its behavior in terms of aliasing. An FFT was performed on a PC directly from the digital output of the converter, using a 24 bit fixed-point DSP as both a signal acquisition system and as an accelerator. After obtaining the results directly from digital domain analysis, they were confirmed by converting the digital signal to analog, which was then passed to an external measurement system. Results are depicted in fig. 6, and show a clear 100Hz component at -55dB.
To work under the conditions stated above, we needed a filter with about 41dB of rejection at that frequency in order to meet a 16 bit performance specification. A 2nd order Bessel filter would meet those attenuation requirements, introducing negligible phase distortion, but a simple R-C filter would not do the job, as shown in fig. 8, leading to only about 20dB attenuation at the working frequency

if amplitude degradation in the baseband is kept at -0.05dB, as shown in fig. 7.
A more reliable solution is to filter at 2.2kHz using a 3rd order Butterworth, as described in the previous section. The alternative plots show the filter performance both in the baseband (fig. 7) and in the full bandwidth of interest (fig. 8). Results are as expected, being superior to the needed rejection.

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5 Conclusions

After considering the available converter types and their respective anti-aliasing requirements, establishing some guidelines for anti-aliasing filtering and confirming the assumptions through experimental work, we conclude that the conditions of actual operation of analog to digital converters have to be carefully considered in order to make a sound choice in what concerns input anti-aliasing filters. It was also shown that requirements for the new oversampling converters are not radically different from the ones for conventional converters. What really differs is the spacing between the baseband and the sampling frequency, leading to lower filter orders, but not always to the single-pole filter simplicity.

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6 References

[1] Sanchez, Clifton W., An 18-bit Dual Channel Oversampling Delta-Sigma converter, with 19-Bit Mono Application Example, Audio Engineering Society preprint, presented at the 87th Convention, New York 1989.

[2] Park, Sangil, Principles of Sigma-Delta Modulation for Analog-to-Digital converters, Motorola Inc. Application Note APR8/D, 1990.

[3] CS5326, CS5327, CS5328 A-D Converter data sheet, Crystal Semiconductor Corporation, Analog-Digital Conversion Ic's VOL1 Data Book, 1990.

[4] DSP56ADC16 16-Bit Sigma Delta Analog-to-Digital Converter Data sheet DSP56ADC16/D, Motorola Inc. 1989.

[5] AD1879 High Performance Stereo 18-bit oversampled ADC data sheet v1.6, Analog Devices, May 30 1991

Fig. 1

Nyquist rate converter

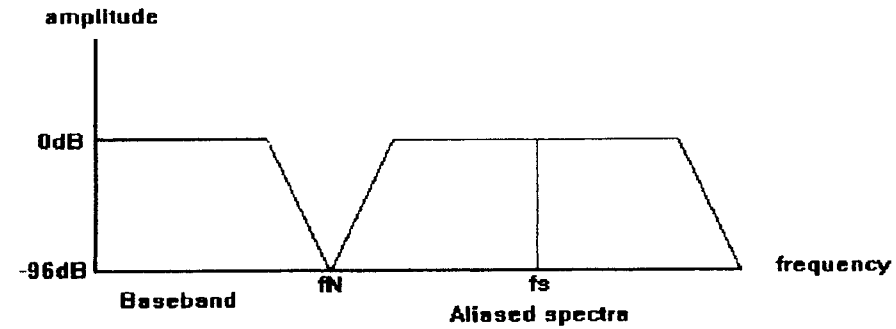

Fig. 2

Nyquist converter frequency spectra

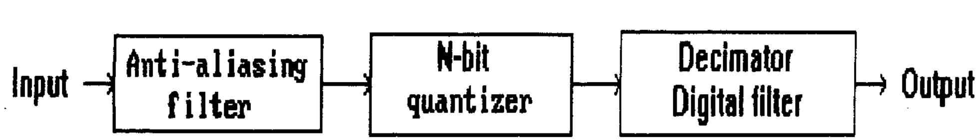

Fig. 3
Generalized oversampling converter

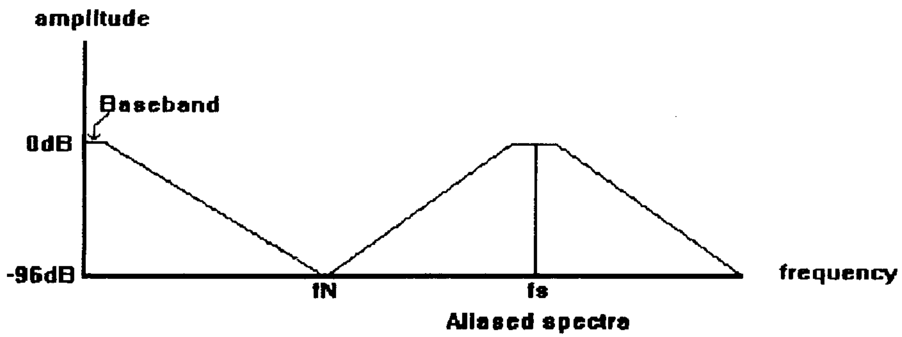

Fig. 4
Oversampling converters frequency spectra

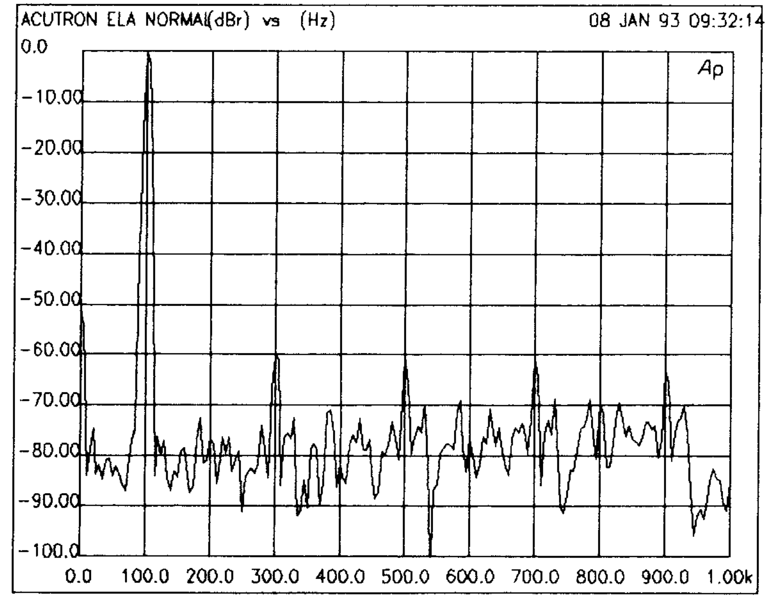

Fig. 5
Baseband signal spectral analysis

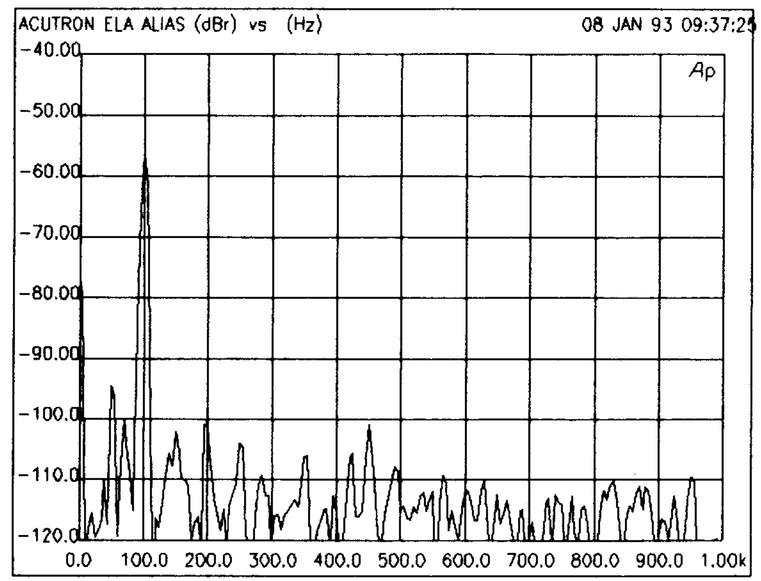

Fig. 6
Aliased signal spectral analysis

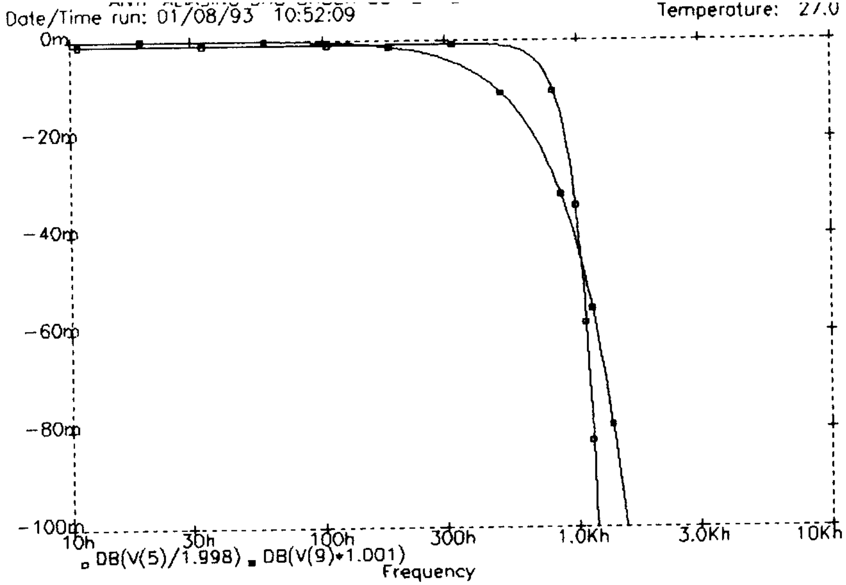

Fig. 7
Frequency response of anti-aliasing filters in the baseband

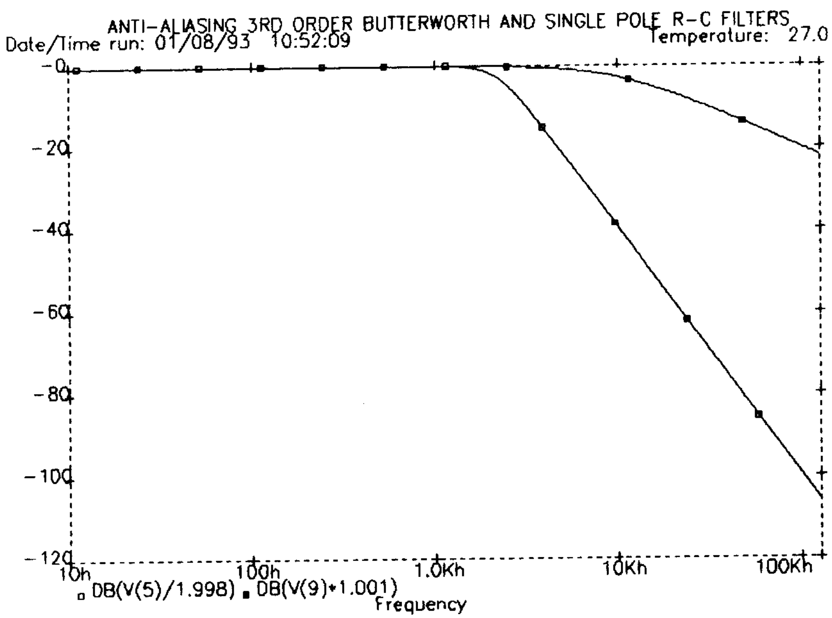

Fig. 8
Frequency response of anti-aliasing filters up to f_s