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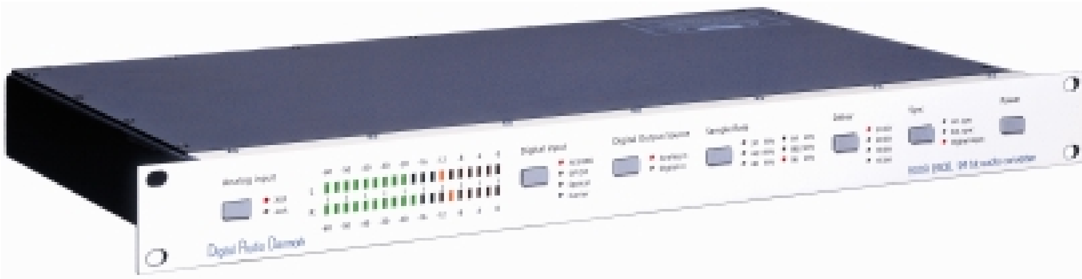


How to avoid Aliasing Intermodulation Distortion (AID)
The invisible distortion in digital audio

- **If an ADC (the components used for the analog to digital conversion) receives frequencies at more than the half sample-rate, it will mirror the frequencies around the half of the sample frequency, and therefore create new a-harmonic frequencies.**

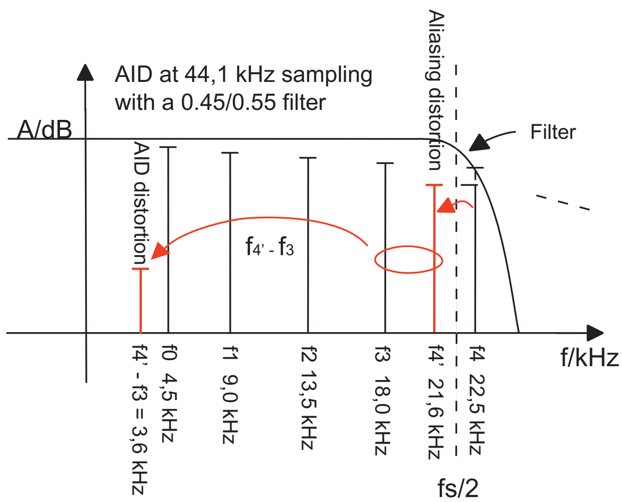
- **In order to have as wide frequency response as possibly most high-end AD converters use a 0.45/0.55 filter. This “Brick Wall” filter starts at 45 % and has full attenuation at 55 % of the sample frequency. At 44,1 kHz sampling the filter starts at 19,85 kHz and has full attenuation at 24,26 kHz.**

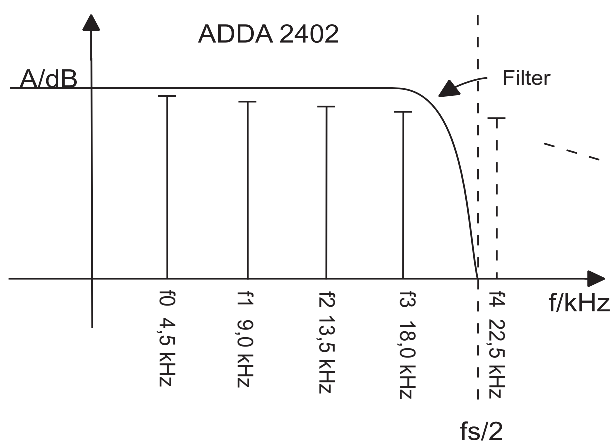
- **This means that at 44,1 kHz sampling the frequencies between 22,05 kHz (the half of the sampling frequency) and 24,26 kHz will be mirrored in the area between 19,84 and 22,05 kHz. By themselves these frequencies is inaudible.**

- **However, the a.-harmonic mirrored frequencies will, when reproduced in a loudspeaker, intermodulate with the audible signal and create new audible frequencies typical in the area between 1 and 5 kHz. This kind of distortion is called Aliasing Intermodulation Distortion (AID). The invisible distortion in digital audio.**

- **The only solution is to have full attenuation at the half of the sample frequency.**

- **The ADDA 2402 has full attenuation at the half of the sample frequency and therefore no AID.**





Distortion Effects from Aliasing in Digital Audio

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ABSTRACT

When converting audio between the analog and digital audio domains, the well-known effects from alias signals have to be taken into consideration. Generally, audio converters do not have adequate filtering to eliminate this kind of spectral distortion. During recent works it has been concluded that the sound quality of digital recordings can be drastically improved when applying efficient stop-band filtering.

INTRODUCTION

When converting an analog signal to a digital signal, the analog signal is sampled with a frequency of twice the bandwidth that is required for the digital representation. To get an optimal sampling, the bandwidth of the analog signal must not exceed half the sampling frequency, also referred to as the Nyquist frequency (NF). If frequency contents are present above the NF, a mirror of the frequencies is generated at frequencies symmetrical to the NF and new frequencies are thus generated with no harmonic relationship to the tonal contents of the original signal. This phenomenon is referred to as aliasing distortion. If the distortion is only present at high frequencies, it is not likely that it can be heard, and therefore it will not degrade the signal quality. However, when the digital signal has to be reproduced using D/A coverters, amplifiers and indeed analog transducers such as loudspeakers will introduce intermodulation distortion (IMD) to the signal. IMD means that the frequencies of the signal mix together and generate new frequencies, which can be audiable if the distortion exceeds certain levels. The problem is now that when new a-tonal frequencies are added to the signal due to aliasing distortion, such signals will mix very badly with the rest of the program and thus be audiable.

The important fact of this problem is that distortion happens when the signal has both aliasing distortion and IMD above a certain level. Of course, it depends on the quality of the sampling and the quality of the reproduction equipment, and in particular the loudspeakers. This effect is referred to as Aliasing Intermodulation Distortion (AID).

Over the years discussions and papers on sound quality of digital recorded material have been many. However, at the 106th AES Convention in Munich a new thesis was presented by Mr. Richard Black [1] stating the problems of this combined distortion phenomenon.

This paper has the purpose of explaining the basis of this distortion type, which he had recently identified.

Today high quality audio A/D converters are typically based on the delta-sigma conversion principle with 64/128 times over sampling and internal digital filtering. According to the manufacturers of the converter chipset, this removes the need for an external anti-alias filter. However the filters normally implemented are so called 0.45/0.55 times the sample frequency (FS) filters. This means that the pass band goes to 0.45xFS, and the stop-band starts from 0.55xFS. In Figure 1 the frequency response is shown with 44.1 kHz and 48 kHz sample frequency.

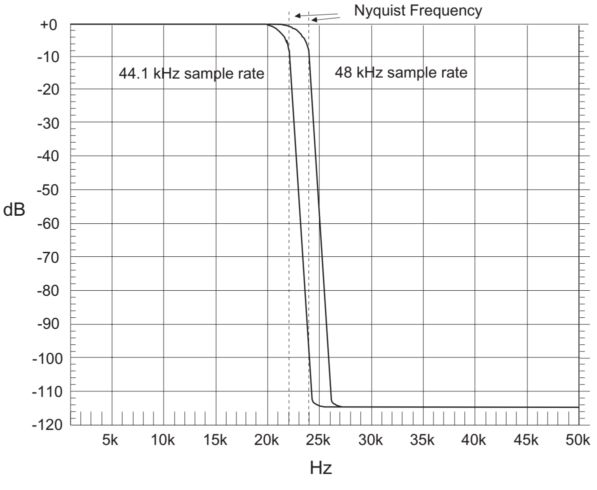

Figure 1, Frequency response of 0.45/0.55xFS filter

In the case of FS = 44,1 kHz, the stop-band will only be effective at 24,26 kHz, and the attenuation at the NF is approx. 8.5 dB. In Figure 2 a close-up of the transition band frequency response is shown.

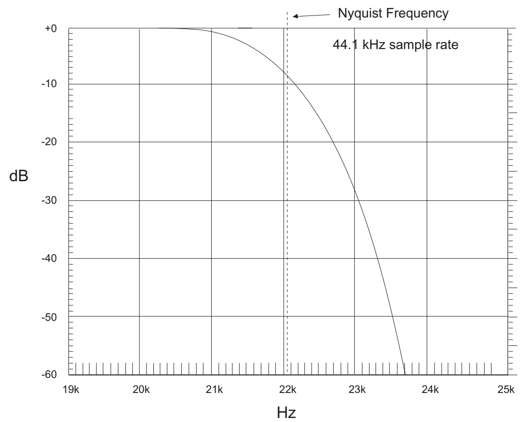

Figure 2, Transition band of 0.45/0.55xFS filter

According to the classic theory on sampling of analog signals, the spectrum of the signal will be mirrored round the NF and repeated for each multiplum of the NF, as shown in Figure 3.

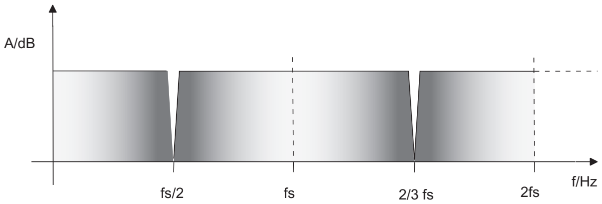

Figure 3, Spectrum of a sampled signal

If signal contents are available in the signal above the NF a mirror of the signal will fold down below the NF as an alias signal, and thus generate aliasing distortion. If a broadband signal is sampled with a 0.45/ 0.55 anti-alias filter, the spectrum can act as shown in the example in Figure 4.

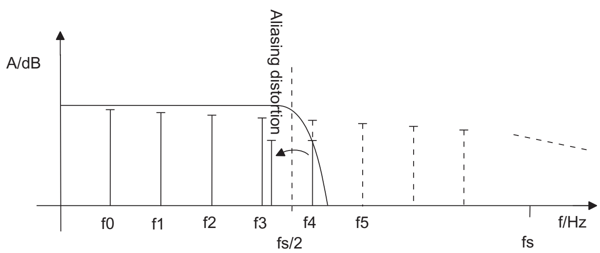

Figure 4, Aliasing distortion example

The example shows a signal with a fundamental frequency of f_0 and a number of harmonics. The 4th harmonic is just above the NF, but still in the transition band of the anti-alias filter of the A/D converter.

When the signal is sampled with an A/D converter having an anti-alias filter, with stop-band attenuation at the NF, no aliasing distortion appears.

INTERMODULATION DISTORTION

Intermodulation distortion is a type of distortion where non-linearities in the signal path generate modulation between all the frequency components of the signal. The primary source for IM distortion in the audio signal path is the loudspeaker. This is due to the non-linear behaviour of the speaker cone, which of course depends on the loudspeaker type and quality.

Modulation means that new frequencies f_{im} are generated from all positive combinations of two existing frequencies.

$$f_{im} = | \pm f_1 \pm f_2 |$$

If two frequencies are denominated f_1 and f_2, where $f_2 > f_1$, and assuming $f_2 < 2f_1$, the modulation frequencies generated are as shown in Table 1.

	f_2	$2f_2$	$3f_2$
f_1	$f_2 - f_1$	$2f_2 - 2f_1$	$3f_2 - f_1$
$2f_1$	$2f_1 - f_2$	$2f_2 - 2f_1$	$3f_2 - 2f_1$
$3f_1$	$3f_1 - f_2$	$3f_2 - 2f_1$	$3f_2 - 3f_1$

Table 1, IM frequencies

IM distortion will appear up to a very high order, but third order IM distortion is the highest order that has any significance in this application. The magnitude of the modulation products is reduced when the modulation order rises. The non-shaded areas in the table are the modulation products with the highest relevance.

IM distortion will generate a-harmonic signals when aliasing distortion is present, since the alias signal will modulate with the harmonics of the original signal. Depending on the frequency contents of the signal spectrum, some modulation frequencies will be more audible than others. If an IM frequency is near the fundamental frequency for the tonal spectrum, the effects will be quite annoying, unless the IM frequency is within the masking area, in which case it will not be heard. An example of IM distortion is shown in Figure 5.

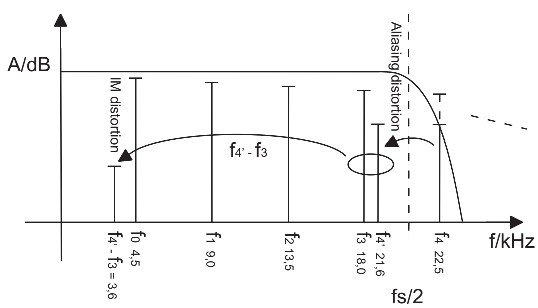

Figure 5, IM distortion example

In this case the 4th overtone of a fundamental tone of 4.5 kHz will generate aliasing distortion at 21.6 kHz. This frequency will modulate with the other tones in the signal spectrum, and first order modulation with the 3rd overtone will then generate a modulation frequency of 3.6 kHz, which is likely to be audible and make interference with the original signal. IM distortion coming from modulation between the harmonics of the signal will not generate a-harmonic distortion signals, since the IM products will have frequencies equal to the harmonic of the signal, and will therefore not be audible.

DVD audio

Recently the new sampling 96 kHz format has been defined. This format is meant to be used in the music production environments, and as the DVD linear audio format. Generally, there is an acceptance that this higher sample-rate gives a better sound quality due to the extended signal bandwidth from 48 kHz to 96 kHz. However, sampling with 96 kHz having 0.45/0.55 filters – which is currently the case for most converters – the aliasing distortion will move up in a frequency area where the IM distortion has very little effect since the harmonic contents of audio signals are insignificant above 40 kHz. An interesting test is to compare the 96 kHz sampled signals with the 44,1 kHz sampled signals having NF stop-band filtering. At 96 kHz sampling the ultra-sonic harmonics will be present in the signal giving a better quality than when sampled with 44.1/ 48 kHz even with a NF stop-band filter applied. However the primary issue when considering the different sample-rates for high quality sampling is to avoid AID, more than having a higher sample-rate.

SUMMARY

When converting audio from the analog to the digital domain, care has to be taken that stop-band filtering is applied in order to avoid aliasing distortion in the digital audio signal. Once the signal is converted to digital, the aliasing distortion products can not be removed with-out reducing the bandwidth accordingly. The aliasing distortion will, when the signal is reproduced on a set of loudspeakers having some IM distortion, generate AID. Of course good loudspeakers have lower distortion, but most hi-fi speakers will have IM distortion thus giving audible AID.

The only way to eliminate the problem with AID is to apply stop-band filtering on the A/D conversion. If this is not done AID will cause different problems depending on the application for the sampled digital audio signal.

When using 44.1/32 kHz sample-rates, the NF is close to/within the audible frequency band. This is a problem for the Compact Disc (CD). Since CDs are mastered in 44.1 kHz AID is a problem, if the A/D converters are used without proper filters. If recording is done with 48 kHz sampling, the Aliasing distortion will be at frequencies above 20 kHz. This will be a problem when working with the audio signals in the sound studio, where the monitored signals will have AID.

If digital sample-rate conversion is used to generate the 44.1 kHz master version from 48 kHz source material, AID will not occur since the alias products are filtered by the sample rate converter alias filter assuming of course that a good quality sample-rate converter is used.

Not many A/D converters are available with NF stop-band filtering. As mentioned in the abstract, the majority of converters have 0.45/0.55 anti-alias filters, which will result in AID. One converter chip is however available today with NF stop-band filtering, and that is the CS5397 from Crystal Semiconductors.

This topic of AID and the influence of sampling bandwidth on the sound quality is something that needs to be investigated further, since to our knowledge documented test results are not available. This is however a subject in which Digital Audio Denmark will conduct more research in the future.

Digital Audio Denmark A/D converters

The ADDA 2402 A/D, D/A and D/D converter from Digital Audio Denmark has implemented NF stop-band filtering for eliminating the risk of AID. This means that digital recordings can be made without aliasing distortion. However, a trade-off has to be made concerning the bandwidth of the sampled signal. When sampling with 44.1 kHz the transition band starts at 18.1 kHz, and the attenuation at 20 kHz is 12 dB. This means that the signal 3 dB bandwidth is reduced to 19.3 kHz.

References

[1] R. Blake: "Anti-alias filters: the invisible distortion mechanism in digital audio". Preprint 4966, 106th AES Convention, Munich 1999.