

# The tradeoff of 192 kHz sampling

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## Abstract

Choosing a sample rate for audio conversion between analog and digital domains, the effects from over-sampling factor, filter characteristic, frequency response, aliasing intermodulation distortion, and pulse response has to be taken into consideration.

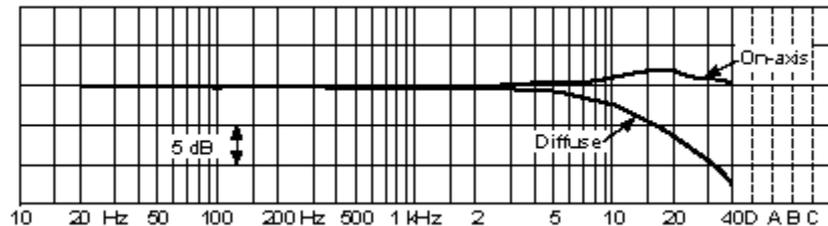
Generally, most A/D converters (ADC) capable of 192 kHz sampling uses a 0.45/0.55 filter, insufficient of giving the full benefit of the capacity in the format.

## Introduction

It is commonly recognized that the human ear fail to hear frequencies exceeding 20 kHz. Even distinguishing between frequencies above 12 kHz is complemented with difficulty. It is also recognized that an ADC will mirror frequencies higher than  $sf/2$  around  $sf/2$  and create a new frequency =  $sf - f$  (aliasing). To minimize this problem the ADC has a built-in filter to cut-off frequencies exceeding  $sf/2$ . Most converter manufacturers employs a 0.45/0.55 filter which commence suppression at 45% of  $sf$  and achieves full attenuation at 55%  $sf$ . By applying such a Brick Wall filter a linear frequency response of 21,6 kHz at 48 kS/s, 43,2 kHz at 96 kS/s and 86,4 kHz at 192 kS/s can be achieved.

To estimate/value the loss obtained by an A/D and D/A conversion it is useful to monitor the frequency response from a high-end microphone.

Frequency response DPA 4007:



As it appear the DPA 4007 records frequencies up to 40 kHz. Hereafter it suppresses at 6 dB pr. octave.

However, only the fewest loudspeakers/monitors are capable to reproduce such a broad frequency specter. E.g. a Dynaudio DM6A commence suppression at 30 kHz.

On top the amplitude of most sound sources in a recording are suppressed by a minimum of 45 dB at 40 kHz.

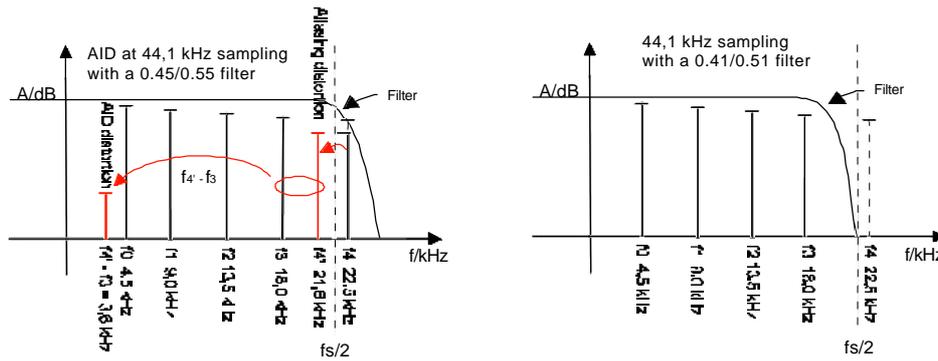
Still the trained ear distinguishes recordings at 24 bit/48 kS/s, 24 bit/96 kS/s and 24 bit 192 kS/s. Thus, the frequency response alone does not determines the quality of a digital recording.

## Aliasing Intermodulation Distortion (AID):

One important factor first pointed out by Mr. R. Blake is a pheromone called Aliasing Intermodulation Distortion (AID):

At 44,1 kS/s a 0.45/0.55 filter starts at 19,85 kHz and has full attenuation at 24,26 kHz. Thus, at 44,1 kS/s 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. Separately, these frequencies are inaudible.

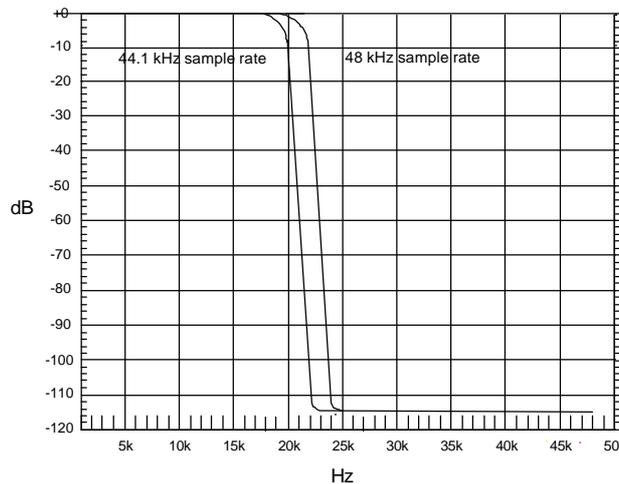
However, the a-harmonic mirrored frequencies will, when reproduced in a loudspeaker, intermodulate with the audible signal and create new audible frequencies typical in the area between 1 and 5 kHz. This kind of distortion is called Aliasing Intermodulation Distortion (AID).



The AID is minimized by raising the sample-rate from 48 kS/s to 96 kS/s or 192 kS/s because the amplitude of the audio at 43,2 kHz ( 96 kS/s), and 86,4 kHz ( 192 kS/s) is substantially attenuated compared to the amplitude at 21,6 kHz (48 kS/s).

Another method to avoid AID is to use a 0.41/0.51 filter. This filter has full attenuation at half the sample-rate and therefore no AID.

The characteristic of an 0.41/0.51 filter is showing below:



In our minds the 0.41/0.51 filter is far more transparent than the 0.45/.45 filter because the absence of AID is considerably more influential than the weak suppression of the highest treble both at 48 kS/s and at 44,1 kS/s.

### Filter characteristic and Pulse Response

Giving the frequency response of the vast majority of loudspeakers/monitors, the missing ability of the human ear to identify frequencies above 20 kHz and the fact that the amplitudes of frequencies higher than 40 kHz are small (typically below  $-45$  dBfs) there is no argument for a sample rate at 192 kHz exclusively founded on broader band width.

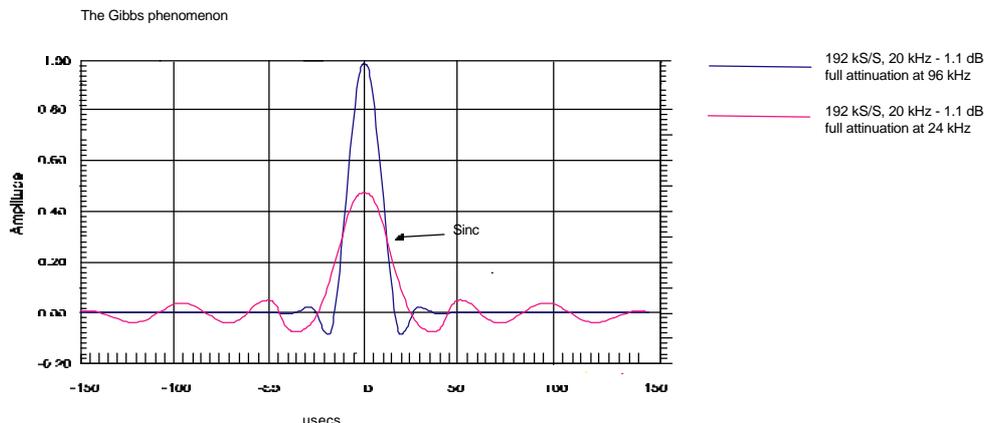
Another issue of potential importance is the ability of the AD/DA converter to reproduce a pulse (a click) .

The signal analysis reveal that the pulse response of an AD/DA converter is a function of the steepness of the applied filter used to avoid aliasing while converting. An aliasing filter creates a pre/post ringing. As a result of this ringing the A/D and D/A converter has a delay ( a little less than 1 ms at 48 kS/s with a 0.45/0.55 filter). The ringing is a result of the mathematical equations whereby the filter is constructed and is as it is unavoidable, if the filter is used to minimize aliasing.

Generally a steep filter has more ringing than a less steep filter.

It is possible to create a low pass filter without ringing. A Gaussian low pass filter lacks ringing but compromises by achieving full suppression at more than 200 kHz, if the filter is to suppress 0,1 dB at 20 kHz. If a filter rolls off faster it starts to ring.

The figure below shows the pulse response for two different filters running at 192 kS/s. One filter has full suppression at 24 kHz while the other has suppression at 96 kHz. Both filters suppresses 0,1 dB at 20 kHz.



As is appear the amplitude is half the size using a filter with full suppression at 24 kHz. In other words, a considerable part of pulse energy is lost in the pre/post ringing of the steep filter.

Giving a small amplitude and a frequency outside the audible area, it has been discussed whether or not this ringing leaves audible effect. It has been emphasized that an improved pulse response improves the stereo perspective. As well there's posted theories that the human ear not only is sensitive to frequency, but also to energy. More research – however – is necessary to determine these statements in a satisfactory manner.

### The Real World

In the previous it is stated that expanded bandwidth alone does not justify a sample rate at 192 kS/s. Exclusively considering band width a sample rate of 96 kS/s must be sufficient. It is – however – possible to improve the pulse response of an AD/DA converter by increasing the sample frequency from 96 kS/s to 192 kS/s if the extra bandwidth is used for implementing a softer aliasing filter. A softer filter obtains the advantage that a proportional smaller part of the energy from a pulse is lost in the pre/post ringing of the filter.

As a developer/manufacture of high end AD/DA converters there is few options concerning components capable of A/D conversion at 192 kS/s while obtaining a dynamic range at about 120 dB. The selection of the ADC falls on the AK5394A chip from AKM Semiconductor.

Filter characteristics AK5394A at 192 kS/s:

FILTER CHARACTERISTICS (fs=192kHz)						
(Ta = 25°C; VA = 5.0V ±5%; VD = 3.0-5.25V; fs = 192kHz, DFS0 = "L", DFS1 = "H")						
Parameter		Symbol	min	typ	max	Units
ADC Digital Filter(Decimation LPF):						
Passband	(Note 7)	PB	0		87.072	kHz
Stopband	( Note 8)	SB	104.928			kHz
Passband Ripple		PR			±0.007	dB
Stopband Attenuation	(Note 11)	SA	120			dB
Group Delay Distortion		ΔGD		0		μs
Group Delay	(Note 9)	GD		63		1/fs
ADC Digital Filter(HPF):						
Frequency response	(Note 7)	-3dB	FR	1.0		Hz
		-0.1dB		6.5		Hz

As the datasheet of the AK5394A reveals this component has a symmetric 0.45/0.55 filter at all sample rates. Using an AK5394A ADC no benefit of the improved pulse response is obtained running at 192 S/s.

Clearly it is possible for individual converter manufacturers to construct better suited filters for sampling at 192 kS/s. Due to related production costs this shows not to be the case looking at the AD/DA converters on the market today. An exception is dCS who produces their own ADC chips capable of selecting between various types of filters.

As AK5394A lacks full attenuation at  $sf/2$  using a symmetrical 0.45/0.55 filter it will generate AID at 44.1 and 48.0 kS/s unless the manufacturer has decided to sample at e.g. 88.2 /96 kS/s and hereafter down sample using a 0.40/0.50 filter.

A reason for 0.45/0.55 being chosen in most ADC's could be because of the reduced complexity and cost while implementing a filter symmetrically around  $sf/2$ .

To built a converter with a built-in aliasing filter wanting to avoid AID at low sample rates the Crystal chip CS5397 is an option. This component uses a asymmetrical 0.41/0.51 filter, however the maximum sample rate using this chip is 96 kS/s.

## Summary

Wishing an improved pulse response the benefit from increasing the sample rate to 192 kS/s is minimum using the available components of today.

Naturally a broader frequency spectrum is obtained sampling at 192 kS/s. This frequency area – however – is above the capacity of the human ear and for most part above what loudspeakers can reproduce linear. As a result the energy in the loudspeakers leaves in the shape of heat.

Of course AID is no issue at a 192 kS/s sample rate and a 0.45/0.55 filter because the amplitude of a 96 kHz is very small. But this advantage is mostly already obtained at 96 kHz.

On the other hand the study of the technical specification of the AK5394A indicates that the S/N ratio decreases at higher sample rates as a result of the expansion of the frequency band. Actually the S/N decreases by more than 3 dB that one would expect as a result of a doubling of the sample rate from 96 kHz to 192 kHz.

Real music is not about delta pulses, but pressure variations recorded by microphones. These pressure variations also have climbing time as well as microphones has a filter suppressing the highest frequencies. This is also to be taken into consideration when evaluating the total effect of the difference between 96 kS/s and 192 kS/s.

However, it is in discussable that almost any practical implementation of 192 kS/s known today does not justify the format.

New components are on the way and in the future it will be possible to create a cost effective solution which allows a soft filter at 192 kHz.

To evaluate if there is an improvement of the sound quality at 192 kHz the human ear is still the best judge and it is suggested to A/B test one on one with an analog signal as reference to determine the related transparency. In this case the expression "less is more" might prove its validity once more.

## References

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

[2] M.J.Story: "A Suggested Explanation For (Some Of The) Audible Differences Between High Sample Rate and Conventional Sample Rate Audio Material" September 1997.

[3] AKM Semiconductor, Technical specifications AK5394A  
<http://www.asahi-kasei.co.jp/akm/en/product/ak5394a/ak5394a.html>