SAMPLE RATE CONVERSION FILTERS - ADA 24/96

Sample Rate Conversion Filters

Introduction
The Delta-Sigma Converters of the ADA 24/96 operate at 6.144MHz (48/96kHz sample rate) or 5.6448MHz (44.1/88.2kHz) at the ends facing the analog world. But the processing, transmission and storage of high-quality audio material are often done at 48 or 44.1kHz. Conversion to/from this low sample rate involves a series of digital filters removing the signal components above one half the sampling frequency (the Nyquist frequency) at all points in the chain. The properties of the lowest filter stage converting to/from 44.1 or 48kHz has profound influence on sound quality, and getting down to 44.1kHz digital and back to analog without severe loss of "air", transparency and spatial definition is usually considered impossible. Furthermore, experience suggests that the optimum design of this critical filter is program material dependent. And of course there is always the matter of taste.

Therefore, when we designed the ADA 24/96, we did not just do a careful circuit design around a set of state-of-the-art 24-bit converter chips. We threw in the powerful Motorola 56303 DSP chip as well, enabling us to offer you a choice of different, carefully optimized filters for the super-critical conversion stage from 96 to 48kHz.

The 5 different filter types described below have been optimized individually for each of the two target sampling rates (44.1 and 48kHz), enabling us to take full advantage of the (comparatively) more relaxed design conditions at 48kHz. The optimization involved A/B comparison to a direct analog transmission of live musical instruments.

Linear phase - non linear phase
When aiming for perfect sound reproduction, the best one can hope for is a "straight wire": Nothing added, nothing removed and nothing changed. The output waveform is an exact replica of the input waveform. In technical terms this means

- Infinite signal-to-noise ratio and no distortion
- Linear phase response "from DC to light".
- Dead flat magnitude response "from DC to light"

How does digital audio transmission and processing score compared to this ideal?

- With state-of-the-art 24-bit converters and no-less-than 24-bit processing, the signal-to-noise ratio today is sufficiently close to "infinite" and with proper dithering, distortion is no longer considered a problem, even at low levels.

- Linear Phase is easily obtained: Perfect digital transmission is essentially just copying or storing and retrieving numbers, and if filtering is required, linear-phase FIR filters are the simplest kind.

- The Sampling Theorem puts a dramatic restraint on bandwidth: Any remaining signal components above the Nyquist frequency (only 22.05 kHz at 44.1 kHz sampling rate) turn into aliasing. Therefore the magnitude response has to drop sharply just above 20 kHz.

So far most designers of digital audio equipment have settled for 2 out of 3: Fought noise as best they could and kept the phase response linear. However, when the conditions change so dramatically from "Ideal" to "Having to give up on one goal out of three", there's no law saying that the other two goals don't move! And - according to our experience - the linear-phase goal does indeed move! Given the necessity of a sharp hi-cut filter, a perfectly linear phase response is not optimal!
With a bit of mathematical and psycho-acoustical reasoning, this is not surprising:
The sharp hi-cut filter is bound to add a lot of ringing to the system's impulse response. The phase response affects the distribution of the ringing in time: Linear phase implies a time-symmetric impulse response with equal amounts of ringing before and after the actual impulse. Human hearing is not time-symmetric, as anyone who has ever played a tape backwards will know. On the other hand, we are not insensitive to phase distortion either, so a compromise has to be found.

The Filters

Through weeks of repeated listening tests and phase, as well as magnitude response adjustments, we came up with the Natural and Vintage filters that use slightly non-linear phase responses.
The filters include some that are aliasing-free in the sense that they achieve full attenuation at the Nyquist frequency where aliasing starts to occur, as opposed to the cheaper half-band filters often used, that are only 6 dB down at the Nyquist please see fig. 1 and 2.

Fig. 1

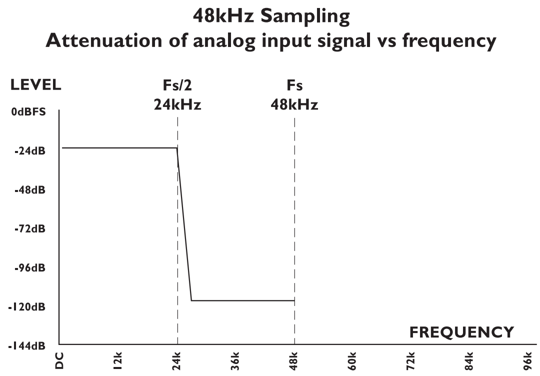


Fig. 2

48kHz Sampling
Reproduced signal with alias components

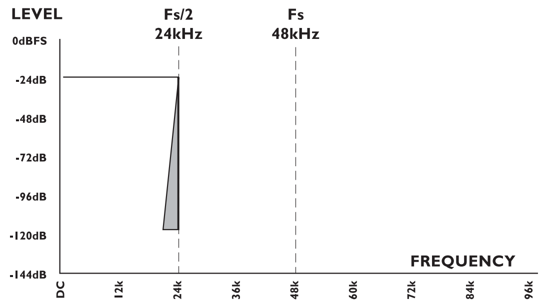


Fig. 3 illustrates how attenuation is performed just above FS/2 when operating at 48kHz sampling rate. As shown in fig. 4 alias components will be reflected around FS/2 and will disturb the high but still audible frequencies.

Fig. 3

96kHz Sampling
Attenuation of analog input signal vs frequency

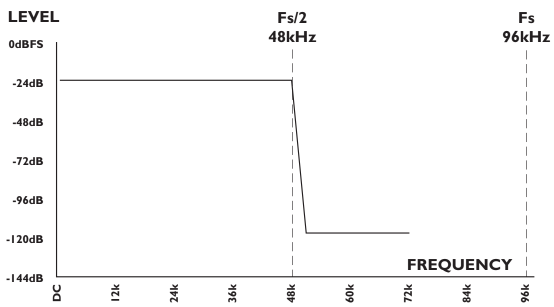
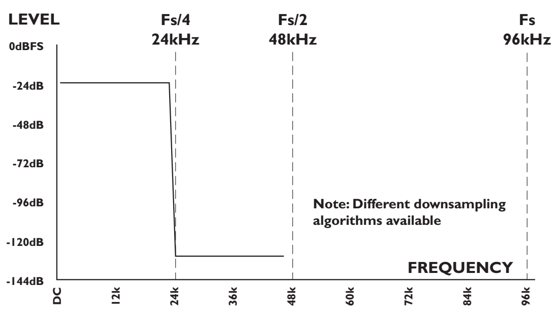


Fig. 4

96kHz Sampling
Reproduced downsampled signal



At 96kHz FS/2 is 48kHz. Since the alias components are produced just below FS/2 they are in a non-audible area when 96kHz processing is possible. This is one of the reasons why 96kHz is preferable.

"Linear" filter
These filters are linear-phase and non-aliasing (the stop-band starts below the Nyquist frequency). The pass-band response is extremely smooth and non-equiripple, extending beyond 20 kHz. With the "Linear" filters you'll have a hard time discriminating between the sound of the conversion chain and direct analog, even at 44.1 kHz!

"Natural" filter
Based on the "Linear" filter class, but with a carefully adjusted non-linear phase response, these filters obtain an almost "better-than-live" reproduction of space while retaining crystal-clear imaging and absolute tonal neutrality. The "Natural" filters too are non-aliasing.

"Vintage" filter
Based on the "Natural" filters, here we've added a bit of warmth and roundness to the treble by introducing a smoother "tube like" roll-off. This filter would be an exceptionally good choice when mastering material that seems too hard in the high-end frequencies.
These filters too are non-aliasing and non-linear phase.

"Bright" filter
These filters are something entirely different: Ultra-short impulse response, linear phase and quite a bit of deliberate aliasing produces a "digital" and slightly aggressive sound adding plenty of top-end life to e.g. Rock and Techno recordings, or giving you the feeling of air you need when you are mastering a somewhat dark sounding source material.

"Standard" filter (STD)
This filter emulates the response of typical mid-end converters: Equiripple half-band filters that are precisely 6dB down at the Nyquist frequency.