Re: Audio Sampling Question

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- *From*: "Jon Slaughter" <Jon_Slaughter@xxxxxxxxxxx>
- *Date*: Sun, 22 Jul 2007 04:46:06 GMT

"Eeyore" <rabbitsfriendsandrelations@xxxxxxxxxxx> wrote in message news:46A2AC19.40ABF49C@xxxxxxxxxxxxxx

> Nico Coesel wrote:
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>> Eeyore wrote:
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>>>>> Guy Macon wrote:
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>>>>>> Are the analog filters identical? A slight difference in phase shift will introduce an error. Even if you use the same analog input and compare measurements at different times there could be variations due to

temperature.

Most modern audio ADCs
don't require a front end
filter.

Sorry, but that can't be true.

It is true. They have no analog front end filter.

If you are sampling, then you'll need to get rid of the frequencies which are above fs/2 otherwise you will get aliasing problems. Thats a law of physics like gravity.

Onboard digital filter.

The ADC oversamples. That's how they avoid aliasing. It digitally filters at the
oversampled rate and then downsamples to 44.1kHz.

No analogue front end filter's required so no issues with filter component tolerances.
The digital filters match perfectly of course.

Graham

You have no clue what your talking about... what a suprise!!!

The oversampling is to reduce the complexity of the analog filtering. You get aliasing no matter what unless you can be 100% sure that your signal's bandwidth is 1/2 the sampling rate. If there is any spurious noise then that will be reflected back onto the first nyquist zone and degrade the signal.

By oversampling, say, 64 times, then we have 64x as much room to roll off the highs. Then we can digitally use a FIR low pass that removes all the noise above the signal itself.

Since the gain of a simple low pass filter is rolls off about 6dB every octave, every time you double the bandwidth(sampling rate) of the sampler vs the bandwidth of the signal you get a 6dB roll off of the max frequency which means you reduce all higher frequencies that normally would be aliased by k*6dB.

If the bandwidth of the signals you are working with is B and you oversample it k times, then any aliasing will be below −k*6dB. Of course now your working with a signal that has a much larger bandwidth than what you actually need so you downsample to reduce it back down to the original bandwidth. Thats not before you digitally filter the signal to remove all the frequencies between B and k*B so they do not get aliased by into the signal. Its called decimation.

Obviously this is just another example of you talking about your ass. Maybe one day you'll get a clue and actually learn something or at the very least clean the bullshit comming from your mouth.

Jon

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