Re: Audio Sampling Question

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- *From*: Tom Bruhns <k7itm@xxxxxxx>

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On Jul 21, 9:38 am, Eeyore <rabbitsfriendsandrelati...@xxxxxxxxxxx>
wrote:

> Guy Macon wrote:
>
> > Henry VIII wrote:
>
> > > I'm sampling high−fidelity analog audio at 44.1 kHz with a
> > > 16−bit ADC. The
> > > analog audio is noise−free for the purposes of this question.
>
> > > The ADC data stream goes to a microprocessor that
> > > compares the data in
> > > blocks of multiple samples to a previously stored set of data.
> > > When the ADC
> > > output data matches the stored data, the microprocessor
> > > generates an output
> > > pulse. Some amount of processing time "X" is needed to
> > > recognize a match
> > > and generate the pulse.
>
> > > My question, again assuming zero analog noise, is: what is
> > > the time
> > > uncertainty of the output pulse? In other words, if I split the
> > > same analog
> > > audio into two of these circuits in parallel, how much could
> > > their output
> > > pulses differ in time? Is the answer simply the clock
> > > frequency accuracy?
>
> > There are several possible sources of variation. How bad each one

> > is depends on your hardware and possibly on the nature of the signal.
>
> > 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.
>
> Graham

Sorry, Graham, that's not quite true. Because the sampling is at a high frequency in a delta−sigma converter, typically a couple MHz, the alias filtering can be very "gentle", but it IS necessary if there's any chance of high frequencies getting through otherwise. Frequencies near the sampling frequency will alias quite nicely down to audio. Things get a little more complicated if you want to use various sample rates on the ADC; there's a little fifth−order antialias filter in the HP E1433A four−channel digitizer (which can be programmed over a 5:1 range of sample rates, as I recall). The filter isn't anything special, just fixed−value parts, but even so, typically the channels match to within a very few nanoseconds.

On the other hand, with respect to the OP's question, how do you know the samples will be at the correct times, and not off by an arbitrary fraction of a sample from the reference samples? And how do you know the clock rate is identical? In other words, just HOW do you compare one digitization with another? I'm not saying it's impossible, it's just not trivial. One should NOT expect to get a set of samples that have nominally the same values as a previous set of samples of the same passage; they could be entirely different.

Cheers,
Tom

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