Subject: Digital Sampling Question
Posted by gofar99 on Sun, 20 Nov 2011 20:56:49 GMT
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Hi Everyone, I would like some information if you please. As an analog guy , digital sampling is an area that I have not yet gotten into (yet). My question regards external analog to digital interfaces. Many are USB but need not be. How can a device with 44KHZ DACs (or ADCs as the case may be) sample music or anything for that matter at say 96K or 192K. It would seem to me that one of two things would occur, either it would have to store the signal and process it off line or not be able to do it all.

Your thoughts........

Subject: Re: Digital Sampling Question
Posted by Wayne Parham on Mon, 21 Nov 2011 02:34:03 GMT
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Check out these two threads:
Non-oversampled v. Oversampled
Audio formats and converters

Subject: Re: Digital Sampling Question
Posted by gofar99 on Mon, 21 Nov 2011 04:03:52 GMT
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Wayne, Thanks, interesting links, but they seem to be focused on going from digital to analog and at the moment I'm focusing on the other end - analog to digital. Where I'm hung up is with devices that convert analog signal (audio) to digital and then feed it into a PC via USB. Many devices list a frequency of 44K HZ but say they will sample at 96,192,320K etc. What precisely are they sampling? It would seem the hardware part would limit the sampling rate.

Subject: Re: Digital Sampling Question
Posted by Wayne Parham on Mon, 21 Nov 2011 14:17:27 GMT
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Read the one called "An Introduction to Delta Sigma Converters" in one of my previous links. It has a pretty good birds-eye view of digitizing and sampling rates. It's pretty interesting, really - Things are (very) different in the newer Delta Sigma converters than they were with the old successive approximation converters. So check it out.

Subject: Re: Digital Sampling Question
Posted by gofar99 on Mon, 21 Nov 2011 19:31:30 GMT
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Hi Thanks for the links, I did read it. That was my thought the most newer devices use the D-S approach and thus can sample pretty much to any bit depth and rate if they are capable of handling the clocking rate.

Thanks

Subject: Re: Digital Sampling Question
Posted by Wayne Parham on Tue, 22 Nov 2011 07:18:36 GMT
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It's just a different method. The delta sigma method is a serialized conversion. The successive approximation method provides parallel digital output, whereas the delta sigma only provides a single bit stream.

A successive approximation A/D converter samples at twice the highest frequency needed, and each sample is given a number that sets the amplitude resolution, i.e. 8-bit, 12-bit, 16-bit, etc. If that width was 8-bits, then the entired dynamic range is represented in a range of 0-to-255, if 12-bit then it's 0-to-4095 and if 16-bit then the amplitude resolution is 65535. And again, the passband is limited to the Nyquist frequency, being about twice the sampling rate. So a successive approximation converter is pretty cut-and-dried.

The delta sigma A/D converter needs to sample much more rapidly, because it is effectively serializing the data as it converts. Instead of sampling at the Nyquist frequency and obtaining a byte or two-byte wide (parallel) representation of the amplitude, the delta sigma converter samples at a much higher rate and each sample is tested to see if it is higher or lower than the last. The analog signal is integrated and the input is compared to the integral during each sample. It's a very simple circuit that doesn't require precision - You basically just sample as fast as possible to get the most resolution out of it you can. But what you end up with is a pulse stream, a serial one-bit output.