Re: USB Sound Cards? Good to adapt for stuff?

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- *From*: Joerg <notthisjoergsch@xxxxxxxxxxxxxxxxxxxxx>

- *Date*: Tue, 20 Jun 2006 18:28:31 GMT

Lostgallifreyan wrote:

My ISP's news server is dropping posts, some of which later pop up (like yours right now). Would have answered sooner but couldn't see it.

> > There may be a quick and dirty option: Find out the highest frequency it can process reliably and with enough dynamic range. Then chop the slow input signal at that rate. A simple CMOS oscillator and a gate or FET can do that. You may lose some bits but hey, if it's good enough it might work.
>
> Nice. :) I think that might just do it.

Basically this is modulating your input signal onto a carrier, the chopping frequency being the carrier.

> One other thing I'm thinking is that the HPF on an IC might be disabled, if the IC designer was generous about this.

They usually aren't. They are after that one big market of audio capture and anything else is peanuts to them from a business point of view.

> The Echo Layla 20 bit audio rack unit is a VERY tempting device for laser scanner control and other things, monitoring laser power, general lab monitor and control. Echo Audio kindly sent me the I/O datasheet in PDF. I've found that the ADC is a Cirrus Logic CS5335−KSEP and with a bit of fine surgery pin 1 can be made to control DC offset, and a DC blocking cap can be bypassed. Currently, if I try to bypass the DC block cap, the result is white noise in that channel, I have NO idea why, it really is odd, that one..

Look at the DC level on the other side of the cap with a scope. If it isn't zero at all times (doubt that it is) then a bypass would force it down and one or more bias levels could go out of whack. Once they are it can take a longer time than expected to return to normal.

> If I can solve this, the CS5335 pin 1 will be a nice control, when low, it tracks an onboard op−amp's DC output and cancels it, when high, it makes the offset compensation freeze, so I can short an input, hold low for a second, then raise high to eliminate any offset between the short and the DAC input inside the CS5335, regardless of source, an extremely useful feature. I hope the Micronas or similar IC in those cheap USB thingers can be modified for this, but I suspect they might not provide a means to bypass the HPF, even though they do apparently offer some control.
>
> All this sounds tedious, but it beats having to build from scratch every time, especially when 8 ins, 10 outs, at 20 bits or better (plus 2−channel S/PDIF I/O at 24 bit), can frequently be found on eBay for £70 or so. Beside the prospect of adapting that, all ideas of self−build look as appetising as a pair of used boots on a plate.

True. If you want to cheat and can spring about $400 then this one might be a nice option:

http://www.labjack.com/labjack_ue9.html

> Your fast−chopping idea is nice though, if it works, it won't matter what the input processes are, at 20 bits I might still get to keep an accurate 16 bit log. These digital multi−channel things put out sync signals too, so I guess if I can derive something from that I can reduce bit−loss by making the chopper sample−accurate.

You can also synchronize with a nifty software PLL that "learns" when to expect the sample. But that can run astray if the signal is too low for a long time. As with building your own hardware that'll be a hassle you just might not need right now.

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Regards, Joerg

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