Keeping filters in focus

Digital domain filters can play an essential role when used with delta-sigma a/d converters. By **Dan Tooth**.

Delta-Sigma a/d converters (Σ) offer a number of advantages, including high resolution, high linearity and the use of a simpler analogue antialias low pass filter. However, the notion that they only need a simple one pole RC filter in front of them – as the internal digital low pass filter takes care of everything else – is only correct when the right digital filter is used in the right application.

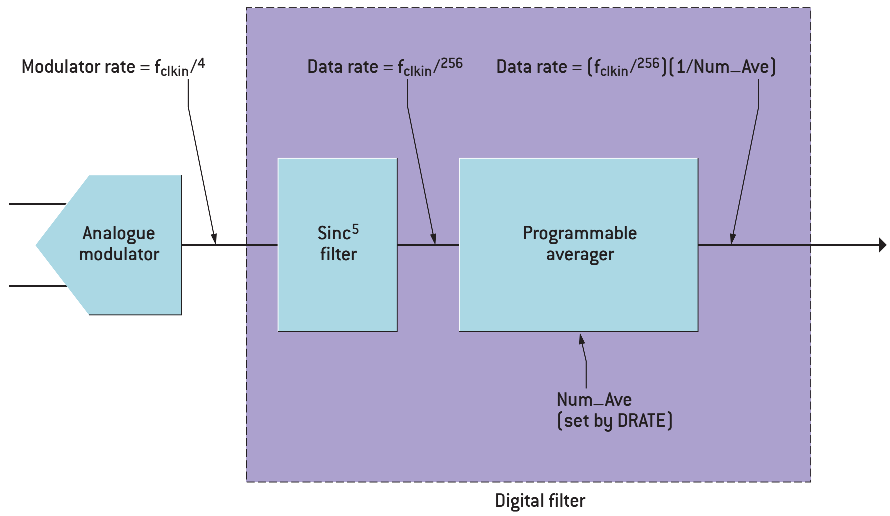
The genius of the Σ a/d is the amazing performance achieved by following a seemingly 'unpromising' 1bit modulator output with a low pass filter (and decimator) to achieve high resolution. The modulator redistributes noise to out of band frequencies, where it is attenuated by the low pass filter. The filters following the modulator must be chosen correctly.

Nyquist and Shannon showed a sampled data system is not aliased if the sample rate is greater than double the bandwidth of interest. Use of the term 'bandwidth' is correct, as this allows for undersampling, where a signal bandwidth modulating a carrier in a higher Nyquist zone is deliberately aliased back into a lower Nyquist zone.

Σ a/ds are multirate sampled data systems. You need to specify if the sample rate is at the input (modulator) or the output, as the two are usually orders of magnitude different. For this reason, they are also known as oversampling converters. Internally, different sample rates are present as the filters decimate in stages from the modulator sample rate to the output data rate.

In Figure 1, the ADS1255 input side modulator sample rate is 7.68MHz/4, or 1.92Msample/s and this yields a 1bit output. The output sample rate, often abbreviated to 'the data rate', can be selected digitally between 2.5sample/s and 30ksample/s with up to 23 noise free bits. For a 30ksample/s data rate, the output from the SINC5 filter is taken straight to the a/d output. For lower rates, the SINC5 output is passed through a selectable averaging SINC filter.

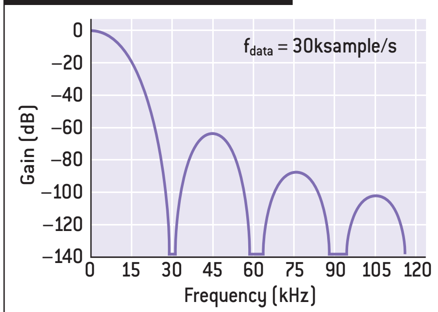
Fig 1: ADS1255 modulator and filters/decimators



The most common Σ a/d integrated FIR filter is the SINC type low pass filter. Its input data rate is the 1bit modulator output at a frequency of 1.92MHz. Its output is a 30ksample/s data rate, which implies the input has been decimated by 64.

In figure 2, there are several points to note:

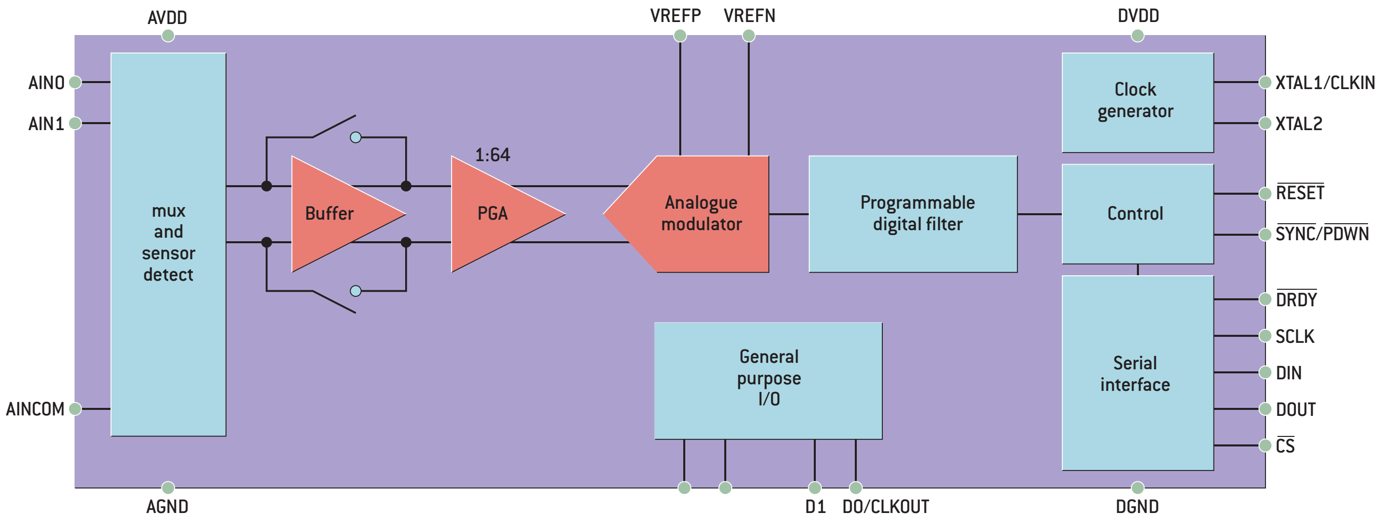
Fig 2: ADS1255 filter response



- magnitude response falls off immediately, giving a loss of about 10dB at 10kHz. This would mean a 10dB magnitude error if the desired signal is 10kHz. This is not a problem if your signal is at 0Hz
- broadband white noise is attenuated, improving SNR and resolution.
- there are notches repeating at multiples of the data rate, and
- the response has only fallen by 20dB at the Nyquist frequency of 15kHz.

Meeting the input sample rate antialias attenuation goal is easy enough. As the input sample rate is 1.92MHz, the analogue low pass filter should be set at a frequency that gives the required attenuation at the modulator's Nyquist of 1.92MHz/2, or 960kHz. For example, a three pole Butterworth MFB filter with a 30kHz corner frequency gives 90dB of attenuation at 960kHz. This could be implemented with a second order filter around a THS4521 a/d driver, followed by with a passive RC filter. The a/d driver stage

Fig 3: ADS1255 block diagram



should be given some thought so as to get good performance.

But what happens after that? The digital filters in the a/d are filtering and decimating; and decimation until the final output data rate is reached can result in aliasing. Because the signal is effectively being resampled, Nyquist's criterion must be met at each new sample rate.

For example, an unwanted input frequency tone of 16kHz on the a/d system's analogue input, will produce an aliased 14kHz output tone in the output data's FFT – more than -20dB down on its input magnitude, due to aliasing in the digital filter/decimation domain.

One solution is to place the analogue low pass filter at a much lower frequency and to use more poles. This negates one benefit of using a $\sum$ a/d, which is that analogue antialias filtering is supposed to be easier. Now, an analogue filter determines system response.

Analogue filter accuracy is hampered by manufacturing tolerances, temperature drift and the non perfect performance of the op amp which implements it. Contrast that with the predictable output of a digital filter, which can be pushed to performances that are impossible with an analogue filter.

Implementing high order analogue filters forfeits the ability to select the output rate digitally, as lower data rates will need a correspondingly lower corner frequency analogue filter.

The response of the SINC filter/decimation means it is generally poor at rejecting unwanted single frequency tones, but there is an exception. The SINC filter response has notches repeating at the output data rate. If the only unwanted tones are 50/60Hz mains frequencies, place the notches at those frequencies and get high attenuation. If the output data rate is set to 2.5, 5 or 10sample/s, notches will be seen at both frequencies. Because SINC type filters can attenuate white noise, this makes them useful where a very low frequency signal is being measured and there are no unwanted tones above the Nyquist. Systems will often pick up mains interference, but a well placed SINC filter will attenuate it.

For systems producing a wide and unpredictable range of discrete frequencies, a better solution is to use an a/d with different digital filters. As the input spectrum is wide, the user must ensure any frequency components above the Nyquist are well attenuated.

An important aspect of many-tap FIR filters is phase response. Some signal sources are not very sensitive to phase distortion; others, such as seismic, are sensitive and the ideal filter has linear phase or a constant group delay. The filter type in the ADS1281 can be linear or minimum phase. The phase relationship of a signal comprising many different frequencies is preserved when passed through a linear phase filter. Each frequency component is delayed by the same amount: 31 output data rate periods.

The linear phase steep roll off many-tap filter comes at a price, namely long phase delay and 'ringing' as the data passes through the filter kernel. And, as phase delay is fixed, it can be compensated for.

Whilst their frequency domain performance is excellent, their time domain performance is not, owing to the Gibbs Phenomenon.

Some applications may have ac frequency components, but require shape and timing to be preserved: electrocardiograms, for example. In this case, filters need to be chosen carefully to avoid aliasing and to preserve the shape of the signal in the time domain, while reducing white noise and mains frequencies.

If none of the integrated filters of available a/ds are right for your application, use an a/d with only a modulator and feed its 1bit output into a dsp software filter.

Clever techniques can be used to reduce the computation time for a multirate filter and decimator system using cascaded integrator comb filters for the first stages. These allow some aliasing, but not in the final bandwidth of interest. The final filtering and decimation stage is a many-tap FIR filter that prevents aliasing in the final bandwidth of interest and corrects the magnitude responses of the previous filters.

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