DRAFT TC–NCA–18

HIRDLS

HIGH RESOLUTION DYNAMICS LIMB SOUNDER

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Reviewed By:

The principles guiding the signal processing filter definition are:
1) Do minimal attenuation to signals containing desired information, that is from spatial frequencies from zero to the value corresponding to about 1 cycle per km;
2) Attenuate to the maximum possible extent any noise which can add to the signal;
3) Do as much as possible of the filtering digitally to minimize sensitivity to drift, imperfections in components, and lack of knowledge of the filter function;
4) Preserve all the precision present in the observations;
5) Send more than the minimum number of samples to the ground so that additional filtering can be done or not as desired for a particular application;
6) Keep the data rate reasonable (by definition, reasonable means less than 40 kbps);
7) Keep the required onboard computer power reasonable;
8) Make filter characteristics programmable to fit particular observations.

From these principles, we derive the following characteristics of the signal processing method:

A high precision infrared system requires the presence of a chopper; the chopping of the radiation limits the attainable frequency response to less than half the chopper frequency. Therefore the highest frequency of determination of an intensity is the value over one chopper cycle; to make this determination digitally requires digital sampling twice per chopper cycle and digital demodulation (in the simplest form, taking the difference between successive samples).

Since a digital system is perforce sampled and noise at frequencies higher than the Nyquist frequency will alias into the signal band, it is necessary to have analog filtering before the digital sampling. However, principle 3 implies that this filter should be as broad as possible. Since we want only to retrieve the fundamental of the chopper waveform (see TC-NCA-05), we can filter out all signals below $0.5*f_{ch}$ and above $1.5*f_{ch}$where $f_{ch}$is the chopper frequency. This will eliminate any aliasing into the signal band of $0\text{-}0.5*f_{ch}$ after demodulation. The bandpass analog filter should have maximum flatness over the passband and should not have too many poles, in order to minimize sensitivity of the response to small changes in component values. A good practical compromise is a four pole Butterworth bandpass filter with 3 dB points at $0.5*f_{ch}$ and $1.5*f_{ch}$.

The synchronous demodulation will be done by taking the difference between two successive samples, or preferably by taking the difference between a sample and the average of the preceding and following samples.

Anticipating the result below for the sample rate, we can derive the required precision of the A/D converter. It is anticipated that the ratio of the largest signals to the NEN in global mode will be about 30,000. It will be seen that the digital filtering process results in an improvement of the SNR in the output point of about sqrt(12) compared to the single sample value. Thus the SNR in a single sample is less than 9,000. The digitizing error (rms) is the LSB/sqrt(12), and adds in quadrature to the detector noise. Hence, the

digitizing noise degrades the SNR by only 1% provided LSB/sqrt(12) <0.1*MSB/SNR. This implies that MSB/LSB>25,000 and so a 16 bit A/D converter is adequate.

The input samples to the digital filter will be 16 bit values of the demodulated signal at the chopper frequency. The minimum output rate is determined by the scan rate and the requirement that there be a minimum of 5 samples per field of view. The rate should be set to this minimum in accordance with principle 6. The input samples contain information at frequencies up to some maximum f_{max} which is about 1 cycle/field of view and noise up to the cutoff frequency of the analog filter, $f_{ch}/2$. Since the output frequency f_{out} is less than the chopper frequency, noise at frequencies above $f_{out}/2$ can alias back into the signal band. We have the following relationships:

$$0 \leq f_{sig} < f_{max} < f_{out}/2 < f_{ch}/2.$$

There will be noise present at frequencies up to $f_{ch}/2$ which can alias to frequencies in the signal band. The input sample stream must be digitally filtered to remove noise at the frequencies that can alias when the stream is sampled at the output rate. The digital filter must be designed for minimal attenuation up to fmax and good attenuation at all frequencies above f_{out}-f_{max}, the lowest frequency that can alias into the signal band. A sharp cutoff at $f_{out}/2$ is ideal, and can be approximated arbitrarily closely by a FIR (finite impulse response) filter, but the steeper the cutoff at $f_{out}/2$, the more terms in the filter function must be employed and the greater the computer burden. However, if the ratio of f_{max}/f_{out} is small, there can be a wide range of frequency over which the attenuation can take place, with correspondingly reduced computational load.

The input samples will be 16 bit results from the A/D converter, and the arithmetic must be done in higher precision to avoid the buildup of roundoff errors. The output samples can be 16 bit samples.

In the actual case, if the filter transmission is high over the signal band and it attenuates the lowest troublesome noise frequency (f_{out}-f_{max}) by a factor of 10 (and higher aliases by more), this will increase the noise by only 1% above the unaliased case. This should be satisfactory minimum performance.

For the nominal (CDCR) design, the chopper frequency is 503 Hz and the global scan rate is 17 km/sec, so that the output sample rate is 5 samples per 1/17 sec, or 84 Hz, one-sixth of the chopper frequency. The highest signal frequency is 1 cycle/km or 17 Hz. Thus, the analog filter is a bandpass filter with 3 dB points at 250 and 750 Hz. The digital filter passes d/c. to 17 Hz unattenuated, has a transition from 17 to 67 Hz, and has less than 10% transmission above 67 Hz. The Nyquist frequency is 42 Hz, and 67 Hz aliases to 17 Hz.

These considerations lead to the following requirements to be included in the IRD:

The analog output of the detector preamplifiers shall be filtered with a bandpass filter of the Butterworth type with four poles, with 3 dB points at 0.5 and 1.5 times the chopper frequency.

The signals will be sampled twice per chopper cycle, synchronous with the chopper waveform, and demodulated to detect the fundamental frequency of the chopper in the waveform.

The signals will be converted to digital form by a 16 bit analog-to-digital converter, having been scaled so that the largest anticipated signals will use only 80% of the dynamic range of the converters.

The frequency of the input digital sample stream will be the chopper frequency; the frequency of the output stream will be some submultiple of the chopper frequency high enough to have a minimum of 5 output samples per field of view at the actual scan rate.

Before output, the digital signals will be filtered by a low pass FIR digital filter with sufficient number of taps that the attenuation shall be flat (0±0.1dB) from d.c. to f_{max}, and shall be at least 10dB at all frequencies greater f_{out}-f_{max}, where f_{out} is the frequency of samples (from a single channel) in the output stream and f_{max} is the frequency corresponding to 1 cycle per km in object space. The filter calculations shall be done in 32 bit arithmetic and the input and output samples shall be 16 bit integers. The filter shall be symmetric in time and have not more than 33 taps.

The maximum frequency for output samples (from all of the radiometric channels) will be 1750 samples per second. In those operational modes where this is not sufficient bandwidth to output 5 samples per field of view, provision should be made to output data from a subset of the channels.

The coefficients of the digital filter and the selection of channels to output shall be programmable to match any operation modes consistent with the overall constraints defined above for sample rate and number of taps.

The ITS will add more detail such as the linearity of the A/D, aperture and aperture jitter of the sample-and-hold, phase stability with respect to the chopper, etc.