8.3A COHERENT INTEGRATION

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Coherent integration is essentially a digital filtering process and was first applied to MST radar observations by WOODMAN and GUILLEN (1974). It is simple to implement with either hardware or software and is appropriate for the very narrow band (in a sense; see below) signals usually received by MST radars. By filtering the signal before performing spectral processing, the computations required for FFT or similar analysis are greatly reduced. Coherent integration does not increase the signal-to-noise ratio per unit bandwidth in the signal band; it simply filters out much of the wideband noise, something that could also be done (slightly better, in fact, but at far greater cost) by full FFT processing of the raw signal.

A pulsed MST radar samples many altitudes at the interpulse period (IPP), giving a matrix (say) of sample voltages $V(h_i,t_j)$. The samples from differing h_i's are uncorrelated, but those from the same h_i but different t_j's are correlated; the correlation time of the fluctuations in the atmosphere responsible for the scattering may be as long as a substantial fraction of a second. In contrast, the bandwidth of the raw signal entering the receiver is large (e.g. about 1 MHz for the 1 μs pulse required to achieve an altitude resolution of 150 m). It is this mismatch between the 1-MHz-wide raw signal and the few Hz-wide scattering process (i.e., the signal bandwidth that would be measured by a bistatic CW radar) that the coherent integration (digital filtering) process partially corrects.

In the MST radar literature, at least, coherent integration has come to mean replacing N consecutive voltage samples from a given altitude with their sum, i.e.,

$$W(h_i,t_k) = \sum_{j=k}^{k+N} V(h_i,t_j)$$

Subsequent analysis is done on the Ws. The number of samples which need to be processed and/or stored is thereby reduced by the factor N, a number which is typically between ten and a few hundred. In other words, most of the operations required in full FFT processing are replaced by simple additions, which can be done very rapidly by special purpose hardware if desired.

To analyze what coherent integration actually does to the signal, it is simplest to consider it to be made up of two separate operations: (1) filtering via a running average (a filter with a unit impulse response of duration T), followed by (2) sampling at intervals T, where T is N times the IPP. This interval represents a drastic undersampling of the original wideband signal.

The first operation multiples the voltage spectrum of the original signal by sinX/X, where $X = \pi fT$ and f is the frequency in Hz. The second operation leads to frequency aliasing with a window $-1/2T \leq f \leq 1/2T$; i.e., signals at the frequencies f and $f \pm n/T$, where n is any integer, are summed together. (This aliasing is in addition to the aliasing introduced by the initial sampling at the IPP.) The combination of these two processes has a few features which may be worth noting here.

(1) Since the filtering function has nulls at $f = \pm n/T$, the zero frequency (dc) point in the final aliased spectrum is the true dc component, with no aliased contributions.

(2) The square of the filter function falls to 0.5 for $f = \pm 0.44/T$.

(3) The first two 'sidelobe' maxima in the filter function at $f = \pm 1.5/T$, $\pm 2.5/T$ are at -13.5 dB and -17.9 dB, respectively.

(4) The combined filtering and aliasing process is such that a spectrum which is flat before coherent integration will still be flat afterwards; i.e., white noise emerges as still white.

In practice, the desired signal in the time series usually changes quite slowly, with a maximum frequency component, say, of f_{max}, and N is chosen so that $f_{max} \ll 1/T$. Then coherent integration increases the apparent signal-to-noise ratio by a factor N, essentially because all but one Nth of the original wideband noise is filtered out. And, to reiterate, exactly the same result would be achieved (but at greater cost) by Fourier transforming the original unintegrated signal, retaining only the interesting part.

One final point: Since coherent integration preserves phase information, it can be interchanged with pulse decoding. So when phase coding is used for pulse compression, the decoding can be done after integration, reducing the number of operations required by a factor of N.

REFERENCE

Woodman, R. F. and A. Guillen (1974), Radar observations of winds and turbulence in the stratosphere and mesosphere, J. Atmos. Sci., 31, 493-505.