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AT/20.1/004

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File Note
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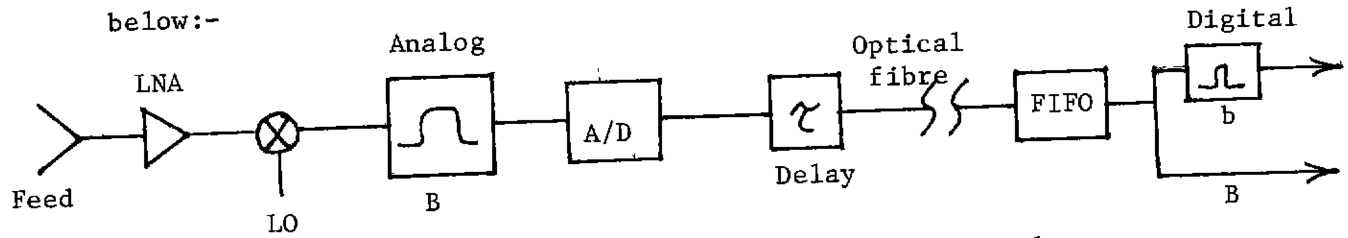
Considerations on Digital Filters for the Australia Telescope

J.R. Forster - May 23rd, 1983

A schematic diagram of an IF system using digital filtering is shown below:-



The RF signal is amplified, then mixed and analog filtered to bandwidth B. The bandlimited signal is dititized and sampled, and the delay (τ) required may be simply performed in a shift register. The properly delayed signal is then sent along optical fibre to a first-in - first-out buffer which acts as an IF line servo. This signal is finally convolved with the transform of a rectangular function of bandwidth b to form the digitally filtered signal. The advantages of digitally filtering at the backend are:

1) Both wideband (B) and narrowband (b) signals are available for correlation.

2) Identical bandshapes are obtained for digitally filtered IF's.

3) Flexibility and stability of digital system compared to banks of analog filters

The reason for digitizing the IF at the telescope rather than at the backend is basically to obtain the relative immunity to noise gained by transmitting digital signals. Furthermore, analog modulation onto optical fibres is much more difficult than digital modulation due to the highly non-linear devices used. Digitizing also offers the simple FIFO method of eliminating IF path phase errors. However, it also means that for a maximum bit rate down the IF fibre, a tradeoff between bandwidth and number of bits per sample will occur. It is the purpose of this note to investigate some of the consequences of this tradeoff.

At present the maximum bandwidth envisaged for the compact array is 160 MHz. This requires a bit rate of 320 Mbits S^{-1} for one-bit sampling. Samplers are apparently available at this speed, and mono-mode optical fibre systems appear capable of transmitting these bandwidths with low dispersion. If 320 Mbits S^{-1} is the maximum data rate, then for two-bit sampling the bandwidth B must be limited to 80 MHz. The digital correlator for the AT is expected to operate in either one-bit or two-bit mode. For two-bit mode the maximum bandwidth is therefore 80 MHz. If more levels are used in the analog-to-digital conversion, the bandwidth is further reduced. However, the correlator is not expected to operate

...2/-

at higher than two-bit mode

In terms of signal-to-noise ratio, a great improvement is gained in going from one-bit to two-bit sampling. If the Nyquist sampling interval (1/2B) is used, the S/N ratio is 0.64 in the one-bit case, and 0.88 in the two-bit case. If we account for the $\sqrt{2}$ decrease in S/N ratio for 80 MHz compared to 160 MHz bandwidth, the resulting S/N ratios for continuum correlations at 160 MHz and 80 MHz is 0.64 and 0.62. The correlator requires the same number of channels to process 160 MHz at one-bit and 80 MHz at two-bits. Therefore, apart from possible gains in the beam pattern due to bandwidth synthesis, one-bit correlation at 160 MHz does not have much advantage over two-bit correlation at 80 MHz. Furthermore, it is likely that one-bit sampled signals will prove difficult to filter digitally.

In his Ph.D. thesis (1981, E.E. Dept. Uni. of Sydney) P.C. Egan investigated the properties of digitally filtered signals for the one-bit case. His study indicates that a one-bit signal may be digitally filtered without any further reduction in S/N ratio over the original factor of $2/\pi$ due to clipping. He attributes this somewhat surprising result to conservation of polarity information in the convolution process. Since only the sign bit is used in one-bit digitization, no extra noise is added by digital filtering. However, the amount of stop-band reduction does depend on the way in which the convolved signal is re-digitized. The convolved signal, which is the result of adding a lag train of one-bit signals weighted by some function (e.g. SINC for a rectangular bandpass), is multi-bit. If only the sign bit is taken, the stop-band rejection is about 8 dB. If the eight most significant bits are retained, the rejection approaches 30 dB. In the case of the AT correlator, a maximum of only two bits may be retained for correlation.

It may be expected that these results, at least regarding S/N ratio, will carry over to the two-bit input case. If that is so, the S/N ratio within the filter passband may not be degraded below 0.88, but stop-band rejection could be a problem. With two-bit input and two-bit output an improvement over the one-bit case is certainly expected. However, the filter response for this case, and for the case of four-bit input/two-bit output should be investigated. If it turns out that four-bit input signals are required for adequate digital filtering, this will limit the continuum bandwidth to 40 MHz for parallel processing with line observations.

In conclusion, it seems that before a reasonable design of the IF filter system for the AT can proceed, a number of questions must be answered. Among them are:

1) What are the spectral characteristics of a digital filter using multi-bit input and two-bit output?

2) How is the true correlation coefficient related to the measured correlation coefficient (a la van Vleck) for digitally filtered signals?

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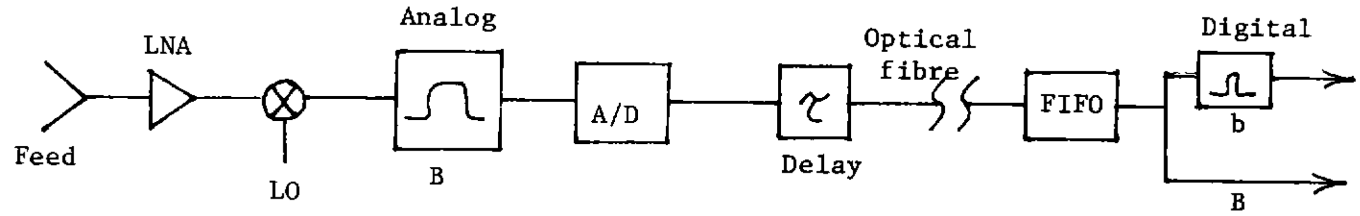

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AT/20.1/004
overall Systs. + Perform. - File Notes

AT PROJECT
DATE 25/5/83
FILE AT/05.1/004
RHF
DNC
TAC
AJP
GWB

File Note
Systems and Performance

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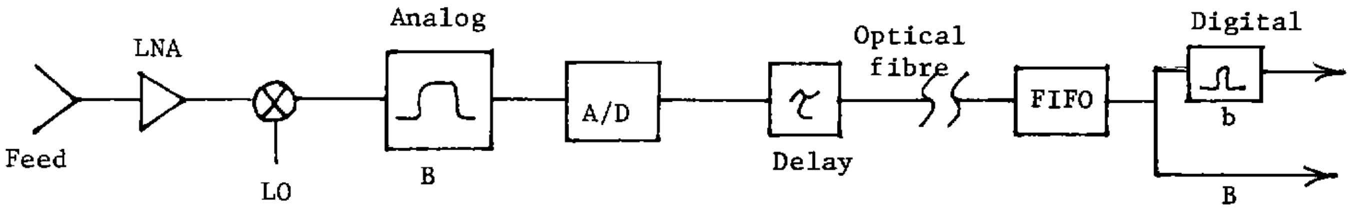

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