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A.T. ONE DAY SYMPOSIUM ON DIGITAL VS ANALOGUE FILTERS AND THE IMPLICATIONS ON L.O. DESIGN

24 May 1984

Summary

1. Analogue or Digital Filters? Implications on System Design *Alec Little/Terry Percival*

As a result of the Correlator Workshop, the proposed IF signal configuration is:

1. 160 MHz, 1 bit sampled, or

2. 80 MHz, 2 bit sampled, or

3. a time multiplexed signal consisting of:
(a) 40 MHz (continuum), 2 bit sampled, and
(b) 10 MHz (tied array and narrow band line), 4 bit sampled, and
(c) one of the following:
20 MHz (wideband, line), 2 bit sampled
10 MHz (narrowband, line), 4 bit sampled
5 MHz (narrowband, line), 4 bit sampled
2.5 MHz (narrowband, line), 4 bit sampled
1.25 MHz (narrowband, line), 4 bit sampled
0.625 MHz (narrowband, line), 4 bit sampled

3(a) is required for calibration of line observations and 3(b) is needed for the tied array. We need to decide where we will place the IF filters. The options are:

(i) all at the antennas, or
(ii) all at the central site, or
(iii) some at the antennas and some at the central site

If all the IF filtering is done at the antenna, a number of signals will need to be multiplexed onto each IF line. Additionally, the filters will need to be small as space at the antenna is limited, and 36 IF filters are required per antenna.

Another consideration is that fine tuning of the observing frequency should be in steps of about 10% of the bandwidth. Thus, as the narrowest bandwidth at the antennas is 650 kHz, one of the local oscillators at the antenna must be tunable in 50 kHz steps. This requirement is eased if the narrowest bandwidth at the antenna is increased and the narrower bandwidths are generated at the central site with further mixing.

The first part of the IF system mixes the RF band down to a 160 MHz band at 160-320 MHz. The IF system we are considering is the part which takes the 160 MHz band and produces the IF signals described above. The initial design for the IF system, based on the correlator workshop report [1], is shown in Fig. 1. This has digital filtering for bandwidths less than 10 MHz.

The alternative, all analog system with all the filters at each antenna, is shown in Fig. 2. The disadvantage of having all the IF filters at each antenna is that they are in different environments and may be subject to different temperature variations.

An alternative to the single digital filter of Fig. 1 is shown in Fig. 3(a). This has four digital filters with fixed coefficients, of which only one filter needs to run at 20 MHz, and the next at lower clock rates and hence we could use a cheaper technology.

Fig. 3(b) is a block diagram of an "ideal" system. A feature of this sytem is that the 40 MHz IF is 4-bit digitized and transmitted to the central site where (fast) tunable, bandpass, digital filters are used to select the narrower bandwidths. This system would be very flexible, give good stable bandshapes, but would be prohibitively expensive. Note that with this sytem, the LO at the antenna only needs to be tunable in 5 MHz steps.

There is an analog equivalent of this, shown in Fig. 3(c), where the 40 MHz IF is reconstituted from the 4-bit signal and then mixed and filtered at the central site. One advantage of this approach is that only one synthesizer is required for each of the four IF's rather than requiring that the synthesizers at each antenna be phase coherent. There may be a problem with glitches and/or loss of sensitivity in the D/A conversion.

One final comment on the subject of filters:

<u>F</u>oolish
<u>I</u>mmortals
<u>L</u>ook
<u>T</u>o
<u>E</u>asy
<u>R</u>emedy

Paul Galico

Discussion

Rick Forster: A 4-bit 40 MHz IF signal may be required for the tied array.

2. <u>Analog Filters</u> *George Graves*

A lot of the material presented here has been written up in more detail [2,3].

An antenna-based, all-analog IF system is shown in Fig. 5. The image filters in front of the mixers are not difficult to realize, but the bandpass filters before the samplers present more of a problem. To illustrate the problem, we will consider the 160 MHz band which is situated at 160-320 MHz. The 160 MHz band could equally be situated at 0-160 MHz or 320-480 MHz. If we design a bandpass filter with corner frequencies of 160 MHz and 320 MHz, and sample this band at 320 MHz, the frequency bands below 160 MHz and above 320 MHz will be folded (aliased) into the IF band as shown in Fig. 4(c). The portion of the band where the image band is less than -40 dB (an A.T. requirement) is considerably less than 160 MHz. However, if we set the corner frequencies inside the 160 MHz band, as in Fig. 4(d), we can use a larger portion of the band.

The percentage of the band to the Tchebyscheff corner frequencies is given in the following table.

FILTER ORDER	RIPPLE (dB)	BANDWIDTH %
6	0.1	71
8	0.1	82
6	0.2	74
8	0.2	84

For filters with large ripple (>~0.5 dB) the phase slope increases towards the filter edge. Additionally a large ripple leads to greater losses and this distorts the passband shape. One advantage of digital filtering is that the phase and attenuation are independent.

The IF system was designed with bandpass filters because the best commercial image reject mixers can achieve only about 20 dB image rejection: at the VLA, 30 dB image rejection is achieved using an 8th order, broadband, 90° phase shifting network. The local oscillator frequencies in the IF system were chosen to be submultiples of 640 MHz to simplify LO generation and restrict the self-generated interference to harmonics of submultiples of 640 MHz. For IF bandwidths <5 MHz, a 20 MHz sampling clock will be used to oversample the IF signal and the unwanted samples can be discarded. This avoids the use of lower frequency sampling clocks which would produce a more closely spaced comb of interfering signals.

For optimum phase stability, the IF system should be designed with low order, relatively broadband filters and broadband amplifiers. Mixer phase can vary with LO power. We will need to look at the phase stability of RF switches. The stability of some K & L tubular filters has been investigated.

An accurate, detailed analysis of the cost of a module has not been made. Following is an approximate costing of the components shown in Fig. 5.

ITEM		NO.	COST	TOTAL EACH ($)
Amplifiers		20	25	500
		6	50	300
Filters		9	400	3600
		4	200	800
Attenuators:	fixed	10	15	150
	preset	6	100	600
	ALC	6	200	1200
Mixers		2	40	80
		1	20	20
Switches:	3PST	3	300	900
	4PST	2	400	800
	2PST	2	40	80
Power splitters		3	40	80
Detectors, Synch. Demod.		6	40	240
Connectors: (backplane)	RF input	1	30	30
	LO input	2	30	60
Samplers		5	30	150
		Total (per module)		$9630

24 IF modules will be required. If digital filtering were used to obtain the 0.625 MHz to 5 MHz bandwidths, this cost would be reduced by $3200 per module, or $77000 in total.

Discussion

Terry Percival: The cost saving would be even greater if a 4-bit 40 MHz signal were sent back to the central site as some ALC's and some other components would not be required.

Alec Little: It would be much nicer to have 40 MHz, 4-bits and do everything back at the control building. This would avoid the need for time multiplexing on the optical fibres, but we may lose sensitivity.

John O'Sullivan: Have elliptic filters been considered?

George Graves: Not as yet, but they should be.

Ray Norris: I don't see why we need 8-pole filters. At the lower bandwidths we don't have to go out to the edge of the band as we have 8000 channels of baseline available. Very rarely will we want those 8000, so if we really want to get out to say, 2.4 MHz, all we need to do is go to 5 MHz and throw out half the channels. But at the higher bands we certainly want to get every bit of sensitivity we can.

Mike Kesteven: You have built a very expensive filter - the correlator - why not use it all the time?

George Graves: What percentage of the VLA's correlator channels are used?

Rick Forster: Typically, 80% of correlator channels are used.

George Graves: If digital filtering is used, we may have a problem finding an output which is proportional to input power to use as an input to the AGC system.

3. <u>Digital Filters</u>

Two documents on this subject [4,5] have already been circulated. A block diagram of the basic system, shown in Fig. 6, is taken from [4]. Andy Henderson has looked at implementing this scheme [5]. Each filter, capable of filtering one 10 MHz IF band to give a 5 MHz, 2.5 MHz, 1.25 MHz or 625 kHz output, would require about 430 TTL devices on four boards and would cost about $3000. We would require 24 of these units for the Compact Array and a further 8 units for the Tied Array - a total of 32 units which would cost about $100000.

Discussion

Warwick Wilson: The cascaded digital filters suggested by Terry Percival (see Fig. 3) would be more expensive as it would require four digital filters per IF.

John O'Sullivan: Even though a lower decimation factor can be realized with a shorter digital filter, the multipliers and adders need to go faster to produce the wider bandwidth output. A decimation 2 filter is capable of doing all the decimations down to 16. There will be problems with the implicit resampling at the output of each filter and the additional corrections required.

Warwick Wilson: The costing was done on the basis of low-pass filters. For bandpass, the cost may go up by a factor of 1.5 or 2.

Regarding Terry Percival's suggestion that the filters could be done in VLSI, it should be noted that the cost of developing a VLSI deisgn is about $100000. It really only becomes viable if a lot of circuits are used.

George Graves: If a low-pass digital filter is used, the IF band will need to be mixed down to baseband. However, if 40 dB rejection of the image bands is required, an image reject mixer can not be used. So we will have an image filter which will set the IF bandshape at low frequencies and the digital filter will determine the shape of the high-frequency cutoff. In the case of the 625 kHz IF band, most of the bandshape will be determined by the low frequency roll off of the 10 MHz IF band. Consequently, a bandpass digital filter must be used.

John O'Sullivan: I agree.

Alec Little: Would this system work if we went to a bandpass?

Jon Ables: When the Parkes correlator, which has about 25000 IC's, was first turned on, it had a mean-time-between-failure of about 15 minutes.

4. <u>Discussion</u>

John O'Sullivan: How would the system design change if we could ease the 40 dB image rejection requirement to, say, 20 or 25 dB?

Terry Percival: We would mix straight down to baseband and use low-pass filters.

Ray Norris: Is there any hope of a 312.5 kHz bandwidth?

Alan Young: The advantage of putting the IF filters at the central site is the ease of subsequent system modification.

Alec Little: It is possible to bring the IF signals back to the central site as an analog signal on the optical fibres, but the fibres work best digitally. Also, the phasing problem is more easily solved if the IF signals are transmitted digitally.

We were told by Peter Napier that filters only cost about $50 but we haven't yet found any for that price.

Richard LaCasse: 4th order low-pass custom design filters for the Mark III VLBI cost $50-$100 from Allen Avionics.

Jon Ables: The cost of digital systems is often many times the cost of the components. Has allowance been made for cost of engineering?

John Brooks: Yes.

Alan Young: It didn't look big enough.

George Graves: The phase change with temperature of some 30 MHz filters with air-cored inductors has been measured. The results indicate that we could expect 0.2°/°C for an 8-pole 0.625 MHz bandwidth filter at 5 MHz. The phase over the whole band seemed to change by the same amount when the filter was heated.

Mike Kesteven: You want the IF bandshapes to be identical from antenna to antenna. This is easy with digital filtering; how good are analog filters?

Alec Little: If the IF bands have slightly difference centre frequencies, you will reduce the effective correlator bandwidth a little.

Ray Norris: The problem is the phase.

Terry Percival: Filter manufacturers tune for amplitude response, not phase.

John O'Sullivan: Do they really do it that way?

Alec Little: George, were you able to measure the phase slope on more than one filter?

George Graves: Yes, I had two of each and they both did the same thing, but we didn't compare one with the other.

Alec Little: It is obvious we don't know enough about analog filters.

Alan Young: Peter Napier said we should try to get filters very the same from antenna to antenna for good self-calibration.

Alec Little: What accuracy do we need for phase closure?

John O'Sullivan: Of the order of 1° difference - half the band 0°and half the band 1° phase - then you have something of the order of cos 1° amplitude closure.

Ray Norris: In practice, you calibrate for closure errors empirically after the map has been made. To do it properly, you have to measure the closure error as a function of frequency for every filter on every baseline. On MERLIN the narrowest IF bandwidth is 78 kHz, and the IF filtering is done at the central site.

Alan Young: Peter Napier said that the IF filters at the VLA were not good enough - they need to be matched to better than 0.25 dB and 2°.

John O'Sullivan: The Westerbork filters were better than 1°, which was primarily phase slope and could be calibrated out. Near the band edge there was a 5° variation, but only over a small range of frequencies.

Bob Batchelor: The mixture of analog and digital systems sounds like a problem.

George Graves: One K & L tubular RF filter has been measured - its phase change by about 0.08°/°C.

Alec Little: What will happen to the temperature in the antennas if the air conditioning is turned off when the antennas are moved?

Dennis Cooper: We expect the air temperature in the cabin will rise 5-10°C.

Jon Ables: The K & L tubular filters are made on a computer-driven lathe and therefore they should be very closely identical.

Rick Forster: It is possible that phase drift in the mixers and amplifiers is a greater problem than in the filters. This is the case at Westerbork.

Jon Ables: There is a description of the series of temperature stability tests done on a series of components at the VLA [6].

Terry Percival: The initial VLA design did not consider the phase stability of amplifiers, mixers and switches. The A.T. will be designed with this sort of thing in mind.

John O'Sullivan: Westerbork took a lot of care with their phase stability and their phase stability was finally limited by their paramps.

Rick Forster: The analog filters seem to have enough stability to give 30 dB of dynamic range except near the edge of the band, and if you can afford to drop those channels you may be chasing the wrong problem.

Alan Young: My experience is that filters drift much more than mixers.

5. L.O. Systems

(a) The unified clock *Jon Ables*

Consider the two receivers, A and B, shown in Fig. 7. Receiver A is stationary, receiver B is moving up with velocity v, and there is a plane wave, of frequency f, propagating towards the receivers from the top of the diagram. Each receiver is equipped with a tape recorder which records the received waveform directly. The frequency of the plane wave seen by recorder B is doppler shifted to $f(1+v/c)$, so the speed of the capstan on the recorder in receiver B is $1+v/c$ times that of recorder A. Thus, if one period of the plane wave is recorded on a length ℓ of tape at receiver A (tape speed is ℓf), one period of the plane wave at receiver B, that is $1/f(1+v/c)$ seconds, is recorded on a length

$$(1+v/c)\ell f \cdot 1/f(1+v/c) = \ell$$

of tape, that is, the same length of tape as receiver A.

If we now read the tapes at the same speed we will be able to correlate the signals directly because we will have compensated for the doppler shift in the signal seen by receiver B. Notice also that if receivers A & B initially receive the plane wave simultaneously, the loop of tape between receiver B and the correlator will grow and automatically provide the required delay. In the A.T. the function of the recorders running at different speeds is performed by samplers with different sample clock frequencies, and a FIFO is substituted for the loop of tape between receiver B and the correlator.

When we do two element interferometry, the main beams of the two telescopes projected on the sky are broken up into interference bands, or fringes, due to the interference. If the two receivers are separated by a time τ in the direction of propagation, the fringe pattern is stationary. But, when one of the receivers moves in the direction of the source relative to the other, the fringe pattern races across the primary beam. In many interferometers the frequency of one of the LO's is shifted to slow down or stop the fringes. It is stated without proof that, to stop the fringes, the LO frequency is doppler shifted by exactly the same factor as the speed of the capstan of the recorder in receiver B is changed for automatic delay compensation.

If the capstan clock and the first LO are derived from separate synthesizers, we can lock the synthesizers to the same reference and doppler shift the reference because the output frequency of a synthesizer is simply a multiple of the reference frequency. This also applies to any other LO's there might be in the receiver system.

One effect which is ignored in traditional systems is that, because the received frequencies are doppler shifted, the bandwidth of the bandpass filters should also be doppler shifted. If a digital filter is placed after the sampler the digital filter's bandwidth will doppler shift as required. This is, at present, only a small effect and has been successfully ignored until now, but if we want to do synthesis from satellites in orbit about the Sun, where the relative velocities will be much higher, and people want to do OH work (somebody always wants to do OH work at some ridiculously fine bandwidth), it may become important.

The unified clock principle has nothing to do with the fact that the fringes are stopped; even if you bring them to a steady rate, this effect still works.

Traditionally delay tracking and fringe stopping are taken as being independent problems - in fact the delay is simply the integral of the doppler effect which you have taken out to stop the fringes.

(b) Impact of the unified clock on the L.O. *Terry Percival*

Fig. 8(a) shows the mixing chain if the observing band is 4.6 - 4.76 GHz; Fig. 8(b) shows that, for the band 9.24 - 9.4 GHz, exactly the same local oscillator frequencies are used. The difference is that for C-band the first mixer does a lower-sideband conversion; and for X-band it does an upper-sideband conversion. Thus 9.24 - 9.4 GHz is the image band for the 4.6 - 4.76 GHz observing band.

A triple conversion scheme is used, rather than a single conversion straight down to, say, 600 MHz, so that the very wide tuning bandwidths of the A.T. can be accommodated. It also gives a good separation between the observing band and the image band so that a simple 3 or 4 pole image filter can be used to give 40 dB image suppression.

At 4.6 GHz, on a 6 km baseline, the fringe frequency varies from -6 Hz to +6 Hz. For convenience we take the fringe rate to be 1 Hz/GHz, that is 4.6 Hz at 4.6 GHz. Fig. 9(a) shows the mixing chains for two antennas which are eventually correlated. The signal to the left-hand receiver is doppler shifted, so the band 4.6 - 4.76 GHz in the right-hand antenna corresponds to the band 4.600 000 004 6 - 4.760 000 004 76 GHz in the right-hand receiver. The reference frequency for the right-hand mixer chain is doppler shifted by -1 Hz/GHz, so the local oscillator frequencies are 6.999 999 993 GHz, 1.599 999 997 4 GHz and 0.479 999 999 52 GHz, and the sampling clock is at 0.319 999 999 68 GHz. After mixing as shown in Fig. 9(a), the output of the left-hand mixer chain is 4.6 Hz - 0.160 000 004 76 GHz, and the output of the right-hand mixer chain is 4.6 Hz - 0.160 000 004 6 GHz. The 0.16 Hz difference between the upper frequencies of the two bands is taken out by the FIFO as described above by Jon Ables.

Fig. 9(b) shows a system where only the frequency of the last LO and the sample clock are altered: in this case the last LO is at 0.480 000 0049 2 GHz and the sampling clock is 0.319 999 999 68 GHz, as above. In this case we get identically the same output frequency bands as in Fig. 9(a). The 4.92 Hz shift in the last LO depends entirely on the observing frequency, and needs to be calculated for each of the 4 IF's. It is not clear if we need to alter both the last LO and the sample clock frequency.

It was shown in Fig. 8(b) that, if we are observing the 4.6 - 4.76 GHz band, the 9.4 GHz image frequency is also mixed down to dc. It can be shown that with the LO frequencies shown in Fig. 9(a) both a doppler shifted 9.4 GHz (9.400 000 009 4 GHz) image in the left-hand receiver and a 9.4 GHz image in the right-hand receiver will be mixed down to 9.4 Hz. This will produce a correlation from the image band. However, if the LO frequencies of Fig. 9(b) are used, the output of the left-hand receiver will still be 9.4 Hz but the output of the right-hand receiver will be 4.6 Hz. Any correlation at the image frequency will produce a correlation product at 4.8 Hz which will integrate to zero over a few seconds. In short, if we doppler shift all the LO's and the sampling clock, both the observing band and the image band will be fringe stopped; but if we shift the frequency of just one LO and the sampling clock, only the observing band will be fringe stopped.

Fig. 10 shows the required local oscillator signals. The 320 MHz and 480 MHz signals are fixed frequencies. The 5.76 - 8.32 GHz in 640 MHz steps can be generated by putting the 640 MHz signal through a step recovery diode multiplier and lock to the required harmonic of 640 MHz. The 1.5 - 2.5 GHz synthesiser is not quite so easy as the steps are ony 50 kHz. Block diagrams of the two synthesizers are shown in Fig. 11. The synthesizer in Fig. 11(a) is locked onto the 640 MHz reference with a 10 MHz offset so that the phase detection can be done at 10 MHz rather than at dc. The 1.5 - 2.5 GHz synthesizer, shown in Fig 11(b), has a 10 MHz reference which is multiplied up in a comb generator, mixed with the output of the YTO, and then a phase comparison is made with 50 kHz. So synthesizers are not trivial devices. Fig. 12 is a block diagram of the reference generator shown in Fig. 10.

The 640 MHz comb generator shown in Fig 11(a) produces not only harmonics required, which are in the range 5.76 - 8.32 GHz, but harmonics up to 18 GHz at greater than -40 dBm, as shown in Fig 13. If the 640 MHz reference signal is doppler shifted, all the harmonics from the comb generator will also be doppler shifted. The 10 MHz comb generator in the 1.5 - 2.5 GHz synthesizer generates harmonics of 10 MHz up to about 5 GHz.

We will now consider what happens if one of the harmonics of 10 MHz, say 2.3 GHz, gets into the 1.5 - 2.5 GHz local oscillator signal. This interfering signal may be 40 or 50 dB below the carrier but it will not be -120 dBc. We will consider the case where the L.O.s are set to observe the 4.6 -4.76 GHz source. It is shown in Fig. 14(a) that, if all the LO's are doppler shifted, the interfering signal will produce a spurious correlation. It is not clear if this correlation will vary slowly with time or be constant, depending upon the type of delay system, but it will definitely be at less than .1 Hz. Fig 14(b) shows the case corresponding to Fig. 9(b) where only the frequency of the last LO and sampling clock are offset. In this case the spurious correlation will be at 4.5 or 4.6 Hz and will integrate out.

Referring again to Fig 11, it is possible to put a frequency offset on the 10 MHz reference signal rather than a doppler shift on the input signal to the comb generator. This will mean that the harmonics from the comb generator will not be doppler shifted.

There is one last comment I would like to make on the subject of radiation and that is that at best only 90 dB isolation can be achieved between modules in different racks. That means that an LO at +10 dBm will produce interference at -80 dBm.

In summary I see the following disadvantages in the Unified Clock scheme:

1. Artefacts due to image frequencies are not smeared out.

2. Frequency combs and "birdies" produce coherent interference.

3. Start phases for tied array are not possible.

4. More difficult to implement a master reset for the clocks.

5. Makes phase switching of LO's more difficult.

The following is a description of interference.

> In the early days of radio design, the mixing process was not universally understood. Many receiver designs ignored the multitude of spurious signals generated by the mixer. As these spurious signals approached and crossed the desired signal, a beat note was generated. As the receiver was tuned across the band, a series of mysterious chirps and tweets could be heard. These "cross-over" spurious products were labelled "Birdies" by the early designers.

When you are searching for "birdies" you feel a bit like this bird hunter: he has found a birdie but it isn't the one he should be worried about.

BIRDIE!

At the VLA they specified that non-phase switched spurious signals should be 63 dB below the noise level in a 50 MHz band. This would stop the sensitivity from being reduced by more than 10%. On the AT, this would mean that spurious signals at the input to the first mixers would need to be less than -120 dBm; and spurious signals radiated in through the front end would need to be even lower.

Discussion

Terry Percival: If the spurious signals are not doppler shifted then they can be 30 dB greater. If we were to doppler shift all the LO's we would be throwing away the equivalent of 30 dB isolation.

Terry Percival: The LO is usually phase switched to reduce the interference problems; and the faster we phase switch the less interference there is.

George Graves: It is best to do the phase switching in the receiver as early as possible.

Terry Percival: Phase switching the RF signal is better than phase switching the LO, but is more difficult. We also have problems with amplitude imbalance when we phase switch the RF.

Jon Ables: The unified clock only refers to doppler shifting the sampling clock. After seeing that doppler shifting the clock made the tracking automatic, I realized that the same principle applies to any clock in the system and, if you had multiple LO's, you could doppler shift all the LO's. I never thought the idea that all the clocks could be unified in the same way as the sampling clock was very practical because I never thought it could be done that way. I think the idea of unifying all the clocks should be called the Grand Unified Clock scheme.

(c) An Alternative System *John O'Sullivan*

We have some specifications on images and the like which are almost impossible and this has forced us into some expensive decisions with regard to front-ends and the like.

If we have a mixer and the LO is doppler shifted, then both the required signal band and the image band come out with the right fringe rate. If the source observed is not quite in the centre of the field, the observing band comes through with the expected fringe rate and the unwanted image band comes through with the inverted sign of the fringe. When we assign the latter fringes to a position in the sky, we get a ghost image at a point which corresponds to the source reflected in the field centre. This suggests one solution, which is to give the LO an additional fringe rotation. This is equivalent to observing with an offset field centre so that the ghost image, when reflected in this field centre, is offset from the desired image. The difficulty with this system, which is equivalent to putting a continuous phase rotation on the LO, is that it requires that a lot of computation be done after the correlator.

The reason that we require that the image be suppressed by 40 dB is that with sufficient signal-to-noise it is possible to get better than 40 dB dynamic range, or internal consistency. If you have spurious images before you start processing, you can be sure it will be 40 dB dynamic range in name only.

One way out of the problem is to put a 0/90° phase switch in the LO and a 0/-90° phase switch after the correlator. The 0/-90° phase switch after the correlator can be implemented by swapping sine and cosine and changing the sign of one of them and so must be performed after the lag-frequency transformation. The desired observing band will always be processed with 0° phase change and the unwanted image band will be 0/180° phase switched.

Another problem arises when the A/D converters before the correlator have small dc offsets, or when there is some cross coupling between the input signals to the A/Ds. The detection sensitivity of the A.T. (ratio of system temperature to apparent source temperature) is about 140-160 dB after one day's observing, and this severely restrains the permissible dc offsets or cross coupling. The standard solution to this problem is to use a 0/180° phase switch before the possible cross coupling sources and after the A/D converters.

6. Discussion

Alan Young: We have to control the phases to reset them, so phase switching could be implemented with little difficulty.

John O'Sullivan: We want to be able to work in a phase-coherent mode for the tied array, so we must be able to add in 0/90° phase switching relatively easily.

Alec Little: One question which needs to be asked is how much rejection we get by doing this.

George Graves: One problem with phase switching occurs when the phase switch rate and the fringe rate are the same or a factor of 2 different. When this happens the interferring signal gets phase switched back in phase every time it rotates out of phase and causes a large spurious correlation.

John O'Sullivan: You will have to use 2 or 3 levels of phase switching.

When there are more than two telesopes we have to introduce a Walsh function sequence.

Jon Ables: John O'Sullivan has forgotten to mention that we will have a full-blown, bi-directional, time-resolved, digital correlator. This greatly lessens the requirement on the dc offsets for the A/D converters or the cross coupling before the A/D converters which is obviously the same thing because the latter produces a correlated dc which is identical to the correlation you would get if you had a dc offset.

(Secretary's note: It is not clear to me that the cross coupling before the A/D converters would produce the same effect as dc offsets in the A/D converters.)

In the digital correlator we slide the signals past one another and compute the correlation function as a function of time. This is eventually Fourier Transformed. If you have a dc offset in the A/D converters, it produces constant offset in the correlation function, and a delta function at the origin in the Transform.

John O'Sullivan: You are forgetting the sidelobes of the delta function.

Jon Ables: No, you don't have any sidelobes at dc.

Ray Norris: You have low-frequency sidelobes. On MERLIN we have to throw out the bottom end of the bandpass.

John O'Sullivan: You don't get sidelobes if you don't do windowing because those particular sidelobes are always on sample points.

Jon Ables: Yes, you are right. This thing doesn't have zero width. However, you are going to discard the lower end of the band. All of the pure dc goes into the first part. There are other things which cause modulation sidelobes. But having a number of correlators, as we will have, is better than having just one complex correlator.

Alec Little: For MOST, it is disastrous because it lifts the whole map up.

John O'Sullivan: Phase switching is not the only way of solving the A/D offset problem - you can make counts of the outputs of the A/D converters.

Jon Ables: I am just pointing out that the simple mathematics of what we are doing causes a powerful rejection of this effect.

John O'Sullivan (?): The other thing is the cross-coupling which may be before or after the A/D converters and this is what causes problems in the VLA correlator.

Jon Ables: The VLA correlator doesn't look like this. They could use their set up to give this sort of thing but when they work in the continuum mode they don't develop the correlation function as a function of time.

Mike Kesteven: But this doesn't alleviate the cross coupling.

Jon Ables: Well, some part of it you do. If you couple some part of one channel into the other channel you produce a uniform value correlation.

John O'Sullivan: No, no.

Alec Little: You could lose out on dynamic range if it was really bad and you had a steady component.

Jon Ables: The cross-coupling is really a long way down on the problem of dc offsets in the A/D converters which is quite real.

Rick Forster: At Hat Creek they use complicated Walsh functions for the phase switching system and they are able to get either sideband. I was wondering if we could use something like that.

Terry Percival: Well, we can't demodulate at 1 kHz.

Alan Young: I think we should be able to offset one LO, so that the required fringes go around one way and the image finges go the other, and then take it out later with a compensating offset on another LO.

Alec Little: We ought to make the decision to actually have this facility in the system and the possibility of phase reversal in case we need to get rid of the dc offset. The 0/180° phase switching needs to be done at the RF.

Alan Young: It is difficult to get very accurate wideband 0/180° phase change, so this will add to the expense.

Jon Ables: The precision with which you can get 180° phase switch at the front-end is usually only of the order of 1%.

John O'Sullivan: As far as the rejection is concerned, it is the demodulation which needs to be accurate.

Terry Percival: Errors in the front-end phase switch give you a loss of sensitivity.

Alec Little: Can we use phase switching to simplify the design of the mixing system.

Terry Percival: Because we have such broad tuning bands, if we want mix down to baseband we have to rely on an image reject mixer plus the phase switching to give us the 40 dB image rejection. There would be no filtering what-so-ever.

I think the 90° phase switching has application in the baseband conversion more than anywhere else.

Alec Little: Does this help the narrowband filtering problem?

John O'Sullivan: There is no need to have the multiple conversion system.

Terry Percival: May be we can go low-pass.

Alan Young: Yes, it looks like it is possible to use an image-reject mixer and low-pass fiters.

Terry Percival: The 0/90° phase switch is almost like an image-reject mixer anyway.

John O'Sullivan: We could mix the 160-320 MHz IF band down to the narrow band IF's in one conversion.

Terry Percival: Can anyone see the need for 50 dB image rejection? We can already easily get 30 dB with the mixing scheme outlined, we could get another 20 dB with phase switching.

Alec Little: How does what we have said impinge on the correlator?

Warwick Wilson: How quickly are you switching these phase switches - once per integration period?

John O'Sullivan: No, it has to be quicker than that. We need at least one full cycle of the phase switch sequence per integration period if we want to map the full field.

Alec Little: As George Graves said, it has to be rapid compared with a fringe period.

Terry Percival: If the correlator dumps data out fast enough you can demodulate the phase-switch off-line.

John O'Sullivan: If you are using Walsh function sequences, you want to independently switch 32 things, then you need 32 basic time intervals. An integral number of Walsh function sequences has to fit into the shortest integration time interval.

John O'Sullivan: The other thing with the correlator is that it means that the output of the correlators will have to go into a set of buffers - one for 0° and one for 90° or 180°. After you have done the accumulation you Fourier Transform them separately and then apply the 0/90° demodulation.

Warwick Wilson: Has it been agreed that we bring back 40 MHz, 4-bit?

Alec Little: If we bring back 40 MHz, 4-bit, then we can do all we want to do either digital or analog. That to me would be a step forward.

Warwick Wilson: What are the disadvantages?

Terry Percival: We are not sure what we can do with it once it gets back to the central site.

Alan Young: If we restore it back to analog we can use analog filters.

Terry Percival: With 4 bits we only have 16 levels, so we are probably restricted to about 5% accuracy.

Alan Young: The consensus seems to be that digital filters are not the way, so we should try to bring the 40 MHz back and do the analog filtering at the control centre. But does this lose us anything?

Alec Little: This question of analog-digital followed by digital-analog needs to be looked at.

John O'Sullivan: I think this just requires an experiment.

Jon Ables: You could use the correlator at Parkes.

Terry Percival: What you are saying is that you put on A/D and D/A in the IF line and switch it in and out.

Jon Ables: You are better using some laboratory noise sources rather than the sky.

You can use a program, written by Paul Rayner called SPECTRA.

Paul Rayner: It needs a minor modification to do cross-correlation rather than auto-correlation.

Alec Little: It looks to me as though the digital filtering is the fall-back position.

Russell Gough

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