MMA Downconversion to Baseband and Sampling

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I. Introduction

Of late, there has been some discussion concerning the final downconversion/sampling scheme to be implemented for the MMA. Specifically, this memo examines the data path from a fixed IF band to a digital bit stream. This memo will consider a few options on this matter. More importantly, my intention is to introduce the variety of considerations that must be addressed in the design phase.

The term *downconversion* is appled loosely by this memo. For some of this discussion, the input band could be considered baseband for the maximum bandwidth mode of the correlator (i.e. 1 GHz bandwidith). However, this memo is more concerned with the higher frequency resolution (reduced bandwidth) correlator modes. A gemalized frequency band fragment of the available IF band is not necessarily at baseband, therefore a downconversion is necessary.

A thorough analysis to determine an optimal design must include many factors; the location of the sampling (control room versus telesope), the signal transmission and delay networks. However, this memo will put these issues aside for the moment. Instead, it will focus on some architectural options and the resulting instrumental flexibility of each option. Also, there will be some comments on the resulting effect on correlator architecture.

This discussion is organized into two sections. First is a brief and general discussion of some potentially useful features of the downconversion hardware. This introduction to the topic will focus on general options, without respect to implementation. The second will list some hardware options.

II. Desirable Downconverter Features

Naturally, an instrument with the raw power of the MMA should be coupled with the most versatile instrumentation. However, some features are easier to implement than others. This section will list some possible correlator features that could be implemented. What is needed by the design staff is a measure of the relative importance of these options. This information could affect major design decisions. If cost is left as the only criteria, many of these features would not be implemented or implemented with reduced capabilities..

1. Bandwidth/Frequency Resolution Trade-off

Every digital correlator in the world provides some degree of frequency resolution flexibility and MMA will be no exception. In general, implementation of different frequency resolution modes requires some digital multiplexing in the correlator, plus different anti-aliasing filters, to reject out-of-band spectral images. The hybrid architecture has some inherent advantages in this area by providing some flexibility using only digital multiplexers. (See Section III. 1: Filter Bank Downconverter).

The existence of bandwidth flexibility is not an issue; however to what extent is an important question. Normally, the bandwidth/frequency is traded off in factors of two. Is this an acceptable step size? What is the maximum frequency resolution which the MMA should support, and with what bandwidth? (6 **kHz** has been mentioned; is this enough?) The maximum bandwidth specification has been discussed at some length, but not the maximum resolution mode.

2. Adjustable Selection of Bandcenters and Simultaneous Measurement.

Figure 1 gives a sketch of a flexible downconverter scheme. In this example, two independent portions of the available IF are analyzed simultaneously. Also, the bandwidth/frequency

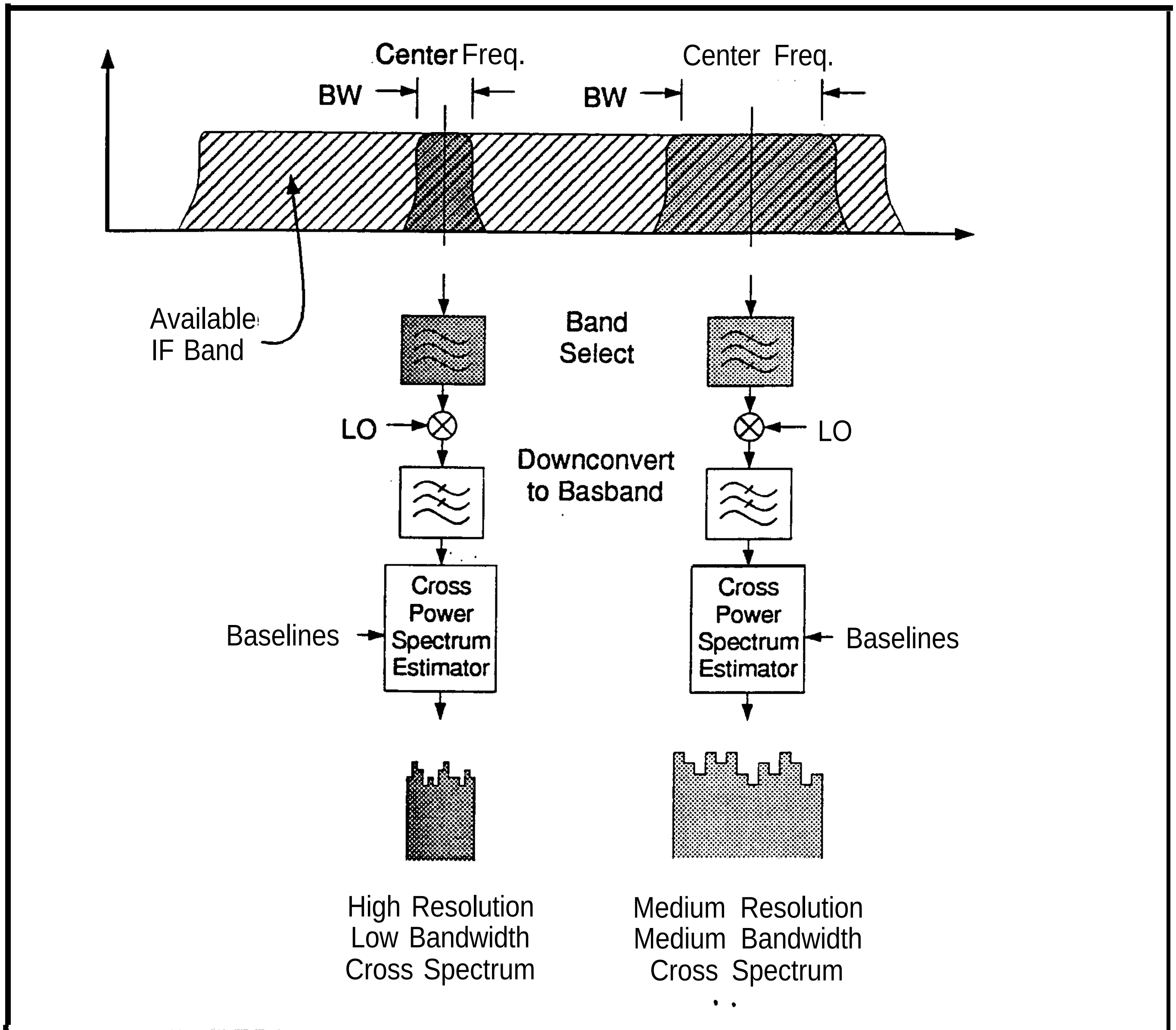

Figure #1 - Flexible downconversion for MMA correlator

resolution of each separate measurement is controlled independently. Ideally, the band-center of each subsection should be fully adjustable. Similarly, the bandwidth/frequency resolution of each should be fully adjustable. Most architectures will allow some of this flexibility, but have limited adjustment ranges. For example, most correlators only allow bandwidth and frequency resolution in factors of 2. Similarly, some architectures would allow simultaneous observations of a different band, but limit the band centers to fixed positions. This would allow **only** one measurement band to be adjusted for optimum placement. The additional (simultaneously measured) bands would be constrained to fixed center frequencies. For example, a fixed filter bank type hybrid correlator would tend to dissect the IF into frigid sub-bands. Resolution within a sub-band can be increased, but the width and position in the IF is fixed. Another question is, how many independent measurement band are needed? The specifications indicates only two per polarization, is this enough? In certain architectures, it may be easier to implement 4 or 8 independent sub-bands.

3. Separate Continuum Correlator

This issue will not be covered in detail at this point, but is related to this discussion. Some people have indicated a preference for a separate, very wide band continuum correlator. If this. causes the IF bandwidth to increase, it will increase the usefulness of the flexibility described in the last two sections. A larger IF band will place more spectral features within the available band, thus improving the case of simultaneous observations of multiple frequency bands within the IF. Also, some of the downconverter architectures may be more conducive to implementing a continuum correlator.

II. Downconversion Methods

Next, the discussion will focus on some available techniques and how they affect the flexibility discussed in the last few sections.

1. Filter Bark Downconverter (Hybrid Architecture)

This approach separates the available *IF* band by using a *bank* of analog filters to cover the input bandwidth. This architecture is normally utilized by hybrid correlators. Each filter module is responsible for a given portion of the input bandpass. In most implementations, the width of each filter is identical, thus subdividing the input bandwidth into an even number of sub-bands (or spectral chunks). In this architecture, one sampler is required for each filter module. Much of the filter hardware is identical between neighboring channels: the samplers will be identical and the bandwidth would be the same. (This homogeneity is normally implemented to save costs, but is not necessary.) Each downconversion module is responsible for a different section of the IF input band, therefore small differences are necessary.

In general the filter widths will be fixed, thereby limiting some flexibility. By reallocating correlator hardware, the resolution of a given sub-band can be increased (See Figure 2). Thus, one section of the available IF is not measured, while another has increased resolution. By allowing some flexibility in managing the digital correlator hardware, many possible bandwidth/resolution modes are possible. For example, if 4 filter modules are available, then it would be possible to allocate 3 correlator modules to one band (which is 1/4 of the total possible processed bandpass), while leaving one in place. Thus, half the original bandpass is ignored, 1/4 is processed at the normal resolution, while 1/4 is processed at 3 times resolution (figure 2).

In the simplest (and cheapest) implementation, the hybrid filter chunks are fixed within the

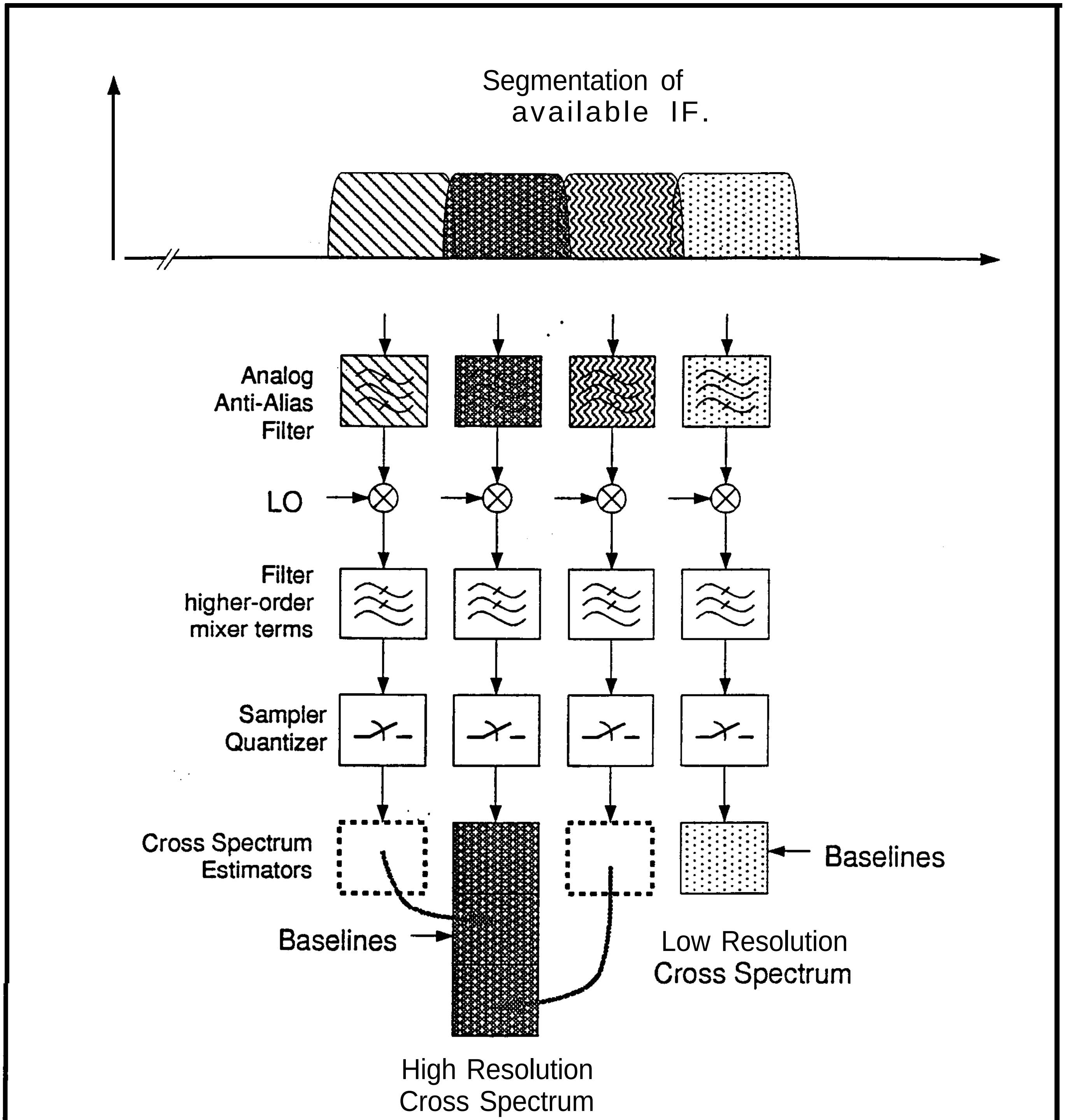

Figure 2 - Filter bank (hybrid) downconverter

IF band, thereby forcing band-center agility to the available (fixed) sub-band intervals. In this scheme the band selection is done with a fixed bandpass filter.and a fixed downconverting mixer. This creates a simple and cheap downconverter.

An alternate approach involves a mixer followed by a baseband filter. The LO will be different for each module, but the baseband filter can be identical. This has the added advantage that, by changing the LO, it is possible to select different IF chunks. Thus with LO versatility, it is possible for one homogeneous filter module to perform the functions of all the necessary downconverters. Also, the sub-band may be centered at any point in the available IF. The disadvantage to this approach is a limited image rejection specification. To remove the image requires a phasing technique that, in practice, can rarely achieve more than 25 dB image rejection. There is also some added expense in the variable LO. However, it does have the advantage of completely homogeneous downconverter modules and complete band-center flexibility.

Wide bandwidths are achieved by combining separate spectral chunks. This creates potential problems with aligning the spectral chunks. This problem will tend to drive the design to wider bandwidth sub-bands that will decrease the number of potential seams in the final spectrum. Unfortunately, increasing filter width also limits flexibility for higher resolution modes. Thus, if only four sub-bands are available (as shown in Figure 2), reallocation of sub-bands will only produce a factor 4 increase in frequency resolution, at best. Other techniques would be needed to improve the maximum possible frequency resolution.

2. Full Bandwidth Downconverter

This approach uses a switchable array of passband filters to select smaller portions of the

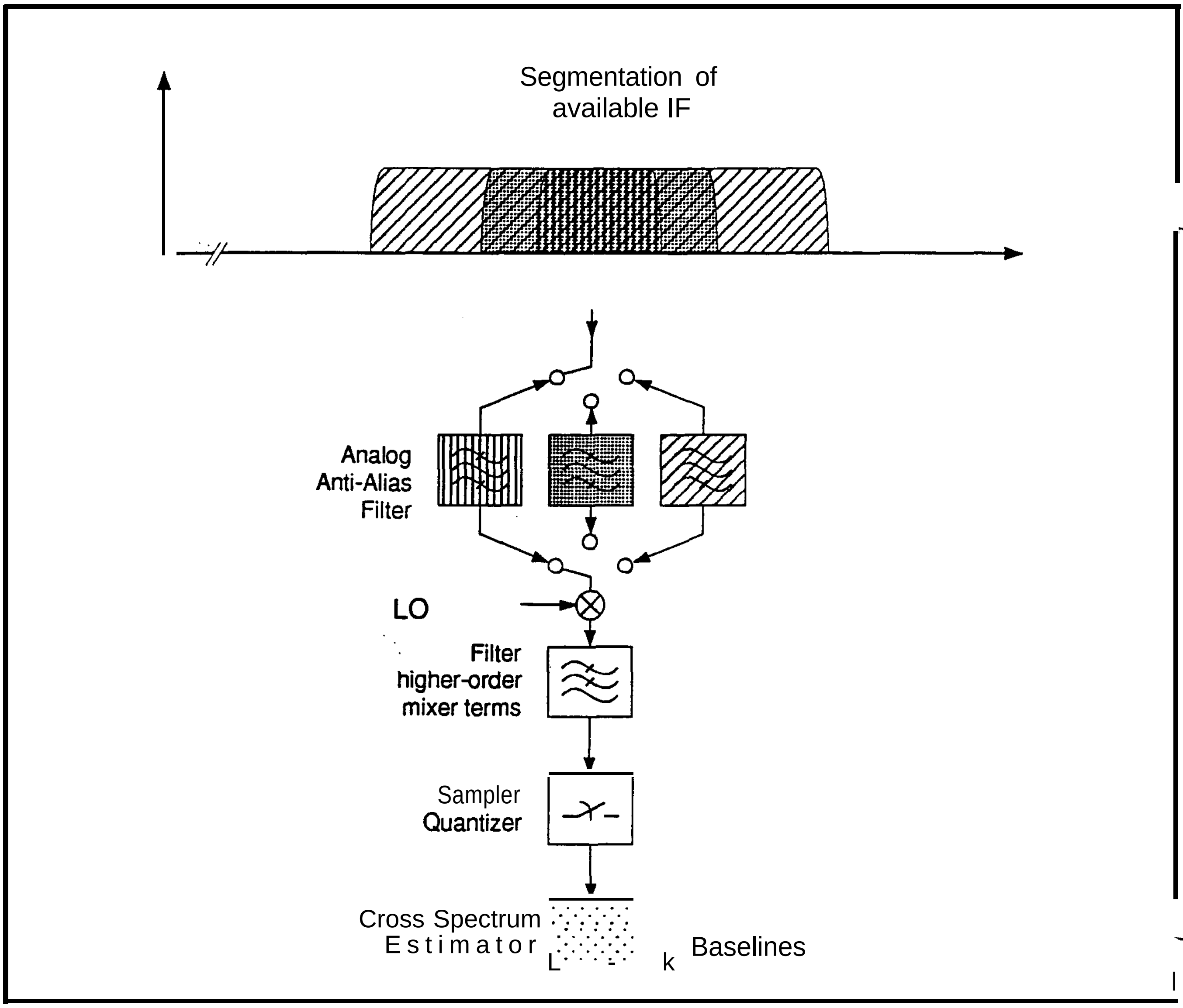

Figure 3 - Full bandwidth downconverter

available IF. These filters are switched into the circuit as needed, so that only one is active at a time. (See figure 3). In general, the filter location would be fixed within the available IF. However, it is possible to vary the center frequency by using a phasing downconverter. This would require accurate phase shifting across the entire IF band, which is difficult., To increase frequency resolution, the correlator would need to use clock slowing or recirculation. Clock slowing is fairly easy to implement, but wastes available correlator power.

To perform simultaneous observations of different sections of the IF would require two independent converters. However, this would be more expensive than the hybrid approach suggested in the previous section. The additional cost is necessary to implement the full IF bandwidth mode, with the resulting higher data rates: The hybrid mode avoids this problem by implementing the full bandwidth mode as a mosaic of low bandwidth sections.

One possible application for this scheme would be to improve the high frequency resolution in a hybrid implementation. One of the downconverter modules in the filter bank could be outfitted with some narrow anti-alias filters. Thus, low resolution modes would operate in the normal hybrid mosaicing fashion, while high resolution modes would use the specially outfitted downconverter to select narrow bands. Homogeneity of downconverters modules would be compromised, but the implementation of higher resolution modes would be simpler than a full hybrid approach. This could be especially useful if the hybrid architecture is coarse (i.e., a small number of sub-bands).

3. Double LO Downconverter

In place of the filter array described in the last section, a double mixer scheme can create a similar downconverter. In the previous discussion, the first filter was needed to select a fixed portion of the IF and reject the rest. This same effect can be achieved using two fixed filters and two LO's (at least one of which is variable). One possible setup is sketched in Figure 4. (Many variations are possible.) This scheme would increase flexibility, with slightly higher hardware cost. However, the main problem with this method is the additional LO's and analog stages, all of which can cause stability, leakage and imaging problems. However, with careful design to avoid these problems, this method can achieve some measure of downconversion flexibility.

4. Digital Filter Downconversion

Analog signal processing can produce a simple and cheap downconversion scheme, but with limitations on flexibility. If lots of flexibility is required, the cost and complexity of analog signal processing can become prohibitive. Digital architectures are available to replace the traditional analog processing1 These techniques can escape the limitations of the analog techniques and provide flexibility that would be impossible with analog circuits. The major advantage of this approach can be summed up with the following quote from the Hewlett-Packard Journal [2]:

> "Digital Filtering offers some powerful advantages when used with a fast Fourier transform(FFT), because it can perform frequency domain analysis in a narrow band around some arbitrary frequency."

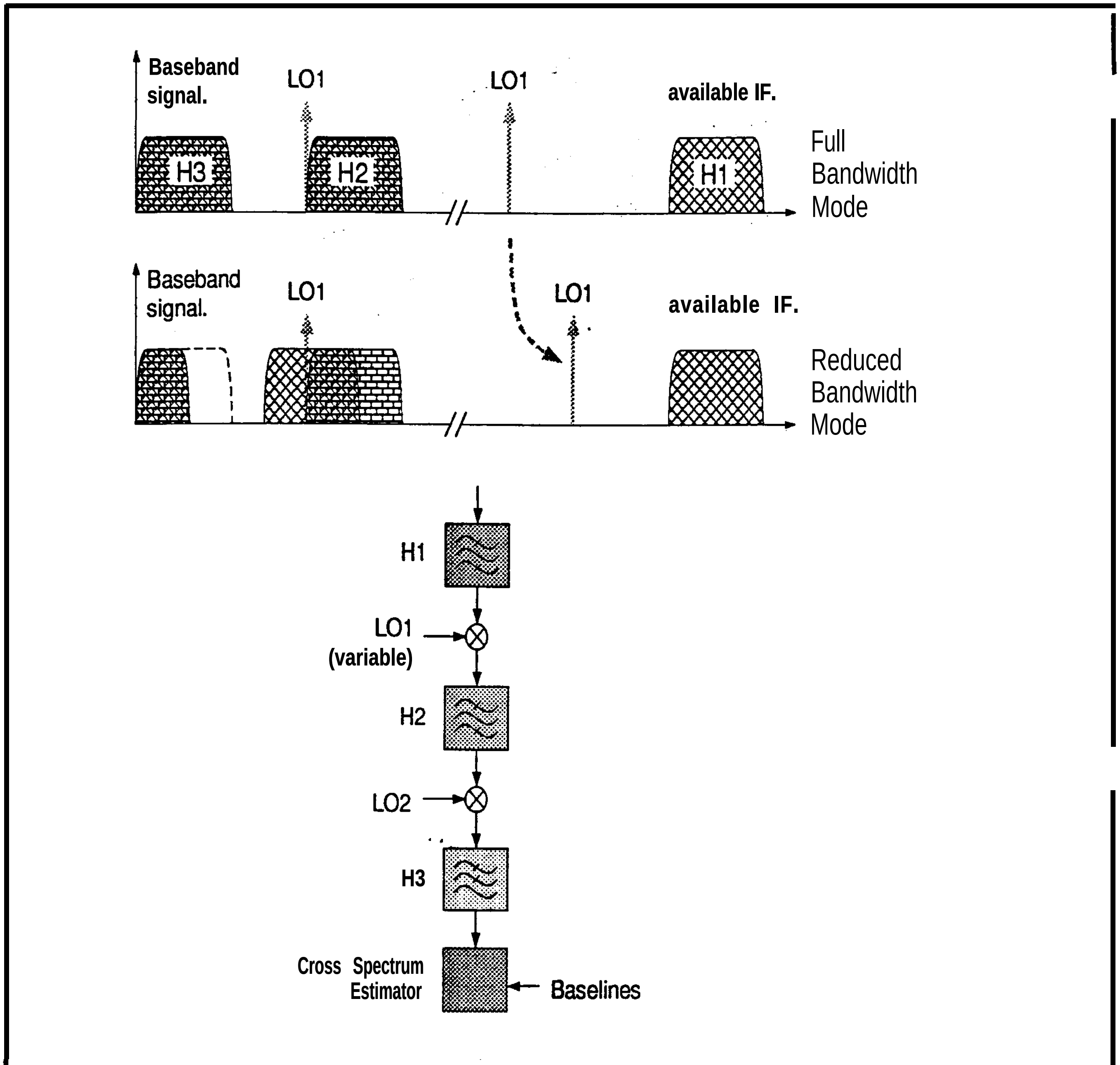

Figure 4 - Double LO downconversion

Naturally, this flexibility comes at a price, the up-front cost could be much higher than a limited flexibility analog system. However, if maximum flexibility is required, the extra cost may be justified.

To perform anti-aliasing of the input signal requires digital filtering. The other important technique of this approach is called ***sample rate decimation,*** which allows the bandwidth to be reduced after sampling. Taken together, the resulting downconverter can select virtually any portion of the available IF bandpass for processing. The rejection of the out-of-band signals is a function of the digital filter order.

Figure 5 gives a sketch of the digital filter architecture. The front-end analog filter is fixed to cover the entire IF. The sampler and quantizer operate at a fixed frequency. The downconversion is performed with a digital mixer. The mixer is implemented as a quadrature to allow separation of images. The digital filters are then used to select a portion of the available bandpass for further analysis in the FFT engine. Each digital block can-select and process a portion of the available IF. By using multiple DSP blocks, multiple sections can be processedsimultaneously. Each section is independently tunable to selection center frequency and bandwidth by changing the digital parameters.

In the case of maximum bandwidth, the DSP block could act as a simple buffer. Then, the FFT engine would measure a full bandwidth spectrum. However, a better solution is achieved by viewing ***this* architecture *as* a *digital hybrid.*** In that case, the full bandwidth can be processed by piecing together the separately measured spectral sections (one from each DSP block). This alternative scheme will reduce the cost of the digital filters by constraining the output bandwidth. This also reduces the requirements of the FFT engine. For example, if 4 DSP sections are available,

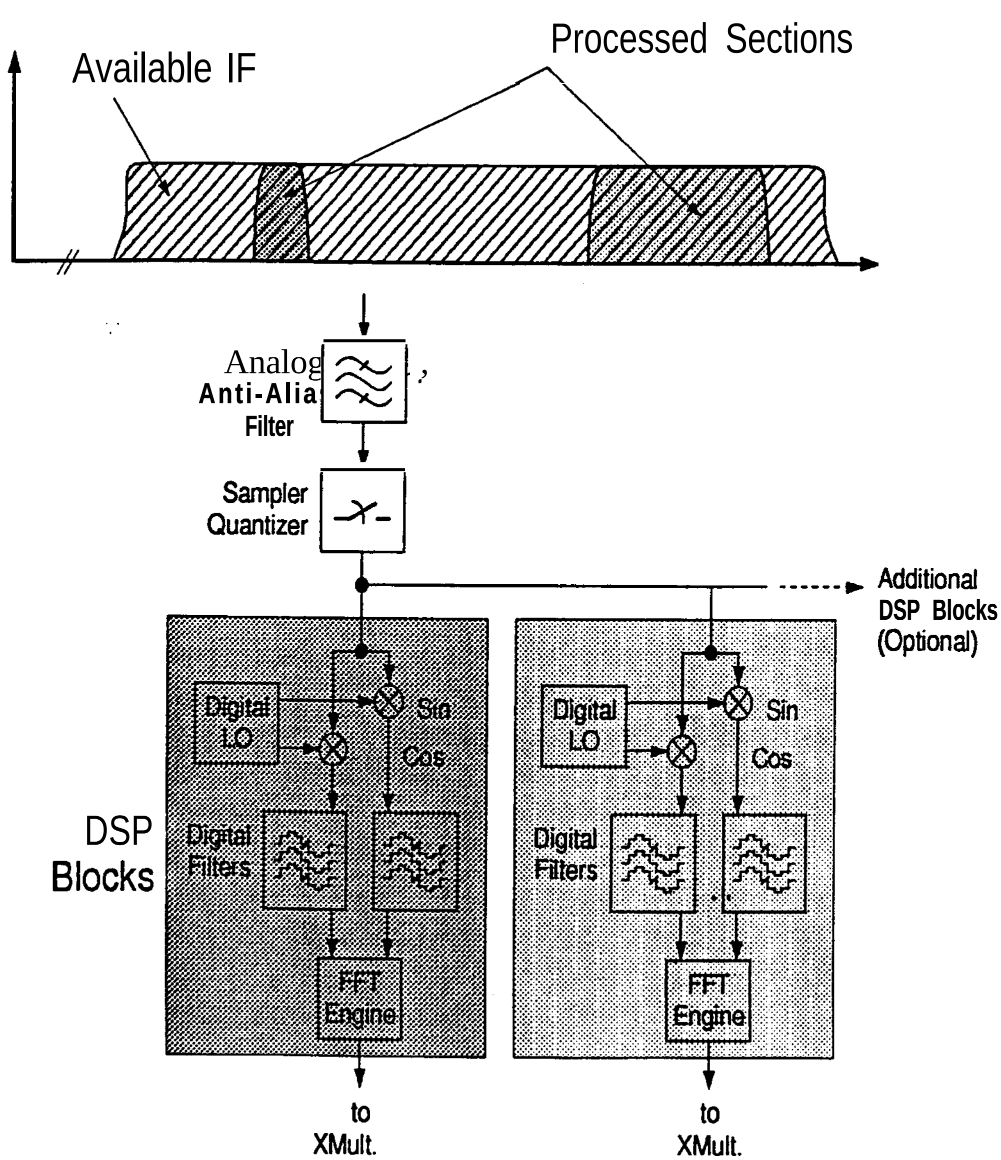

Figure 5 - Digital filter downconversion

each could process 1/4 of the available bandwidth (ignoring band-edge loss). Consequently, the 4 separate FFT engines would require only a 1/4th of the speed as compared to a single full speed FFT processor. The normal problems of hybrid correlators (piecing together of separate spectrum) will be greatly reduced because of the full digital nature of the segregation.

This approach produces an extremely agile and powerful instrument, but the cost of building the digital filters will not be trivial. These techniques are normally reserved for speech type signals with bandwidths around 20 kHz. The MMA would like to process 1 GHz, i.e., 50 times faster. A liberal application of time-division processing is needed.

It is now necessary to consider some practical aspects of this architecture. Implementation of the digital filters and mixers requires a larger number of multi-precision adds and multiplies. First, if we assume 4 DSP blocks are used per IF, the MMA would require 320 (40 antennas x 2 pol. x 4) DSP blocks. These blocks must operate at effectively 2 GHz (ignoring band edge loss) to handle-the full bandwidth (1 GHz). If we apply time division to work at a more reasonable speed of 60 Mhz, it would increase the number to 10,560 DSP blocks that run at 60 MHz. For a moment, I'll ignore the time division and buffering problem. Within the DSP block, are two digital filters (one for each quadrature), a quadrature mixer and FFT engine. Next, I'll attempt to estimate the number of multiprecision operations that must be made in the resulting 10,560 DSP blocks (exluding the FFT).

Total Operations @ 60 Mhz	Multiplies	Additions
FIR Filter	1.5×10^6	1.5×10^6
IIR Filter	0.44×10^6	0.44×10^6
FIR Filter (Constrained to 1/4 BW	0.38×10^6	0.38×10^6

Table 1: Operations in digital filter downconverter (does NOT include FFT or Xmult.)

The mixer consists of two multiples and a lookup table of sin/cos values. The size of the digital filter will depend on the out-of-band rejection required. The filters can be implemented as a IIR or FIR. Generally the FIR is preferred because it has linear phase, but an IIR design would reduce the number of operations. For example, a bandpass filter with 20 dB out-of-band rejection would require about 69 multiples and 68 adds[1] to implement in an FIR architecture. Conversely, a similar IIR filter would only require 20 multiplies and 16 adds[1]. Two separate filters are needed to implement the quadrature operation. The cumulative results are given in Table 1. It was mentioned earlier that constraining the output bandwidth would reduce the size of the digital filters. This result can be shown with some sample FIR equations in Table 2. The reduction in output bandwidth allows ignoring some of the possible output values. This simplification is reflected in Table 1 as a

$$X(t) = C_1\,Y(1) + C_2\,Y(2) + C_3\,Y(3) + C_4\,Y(4) + \ldots + C_N\,Y(N)$$
$$X(t+1) = \text{(unnecessary)}$$
$$X(t+2) = \text{(unnecessary)}$$
$$X(t+3) = \text{(unnecessry)}$$
$$X(t+4) = C_1\,Y(5) + C_2\,Y(6) + C_3\,Y(7) + C_4\,Y(8) + \ldots + C_N\,Y(N+4)$$

Table 2: Simplified FIR equations for 1/4th Bandwidth

reduction in operations for the FIR filter. I don't know of a similar trick for IIR filters, but it may exist.

The result is an impressive quantity of digital hardware, even before the Fourier transform or cross multiplication! Also, the 20 dB out-of-band rejection may not be enough, in which case all these numbers will increase. One advantage of the digital approach: the out-of-band rejection can be calcuated in all modes. Maybe this will allow acceptance. of a sub-optimal specification.

The digital filter approach would create an extremely powerful instrument. Spectral features within the IF could be examined with a great deal of flexibility. Many features can be examined simultaneously and with varying degrees of frequency resolution.

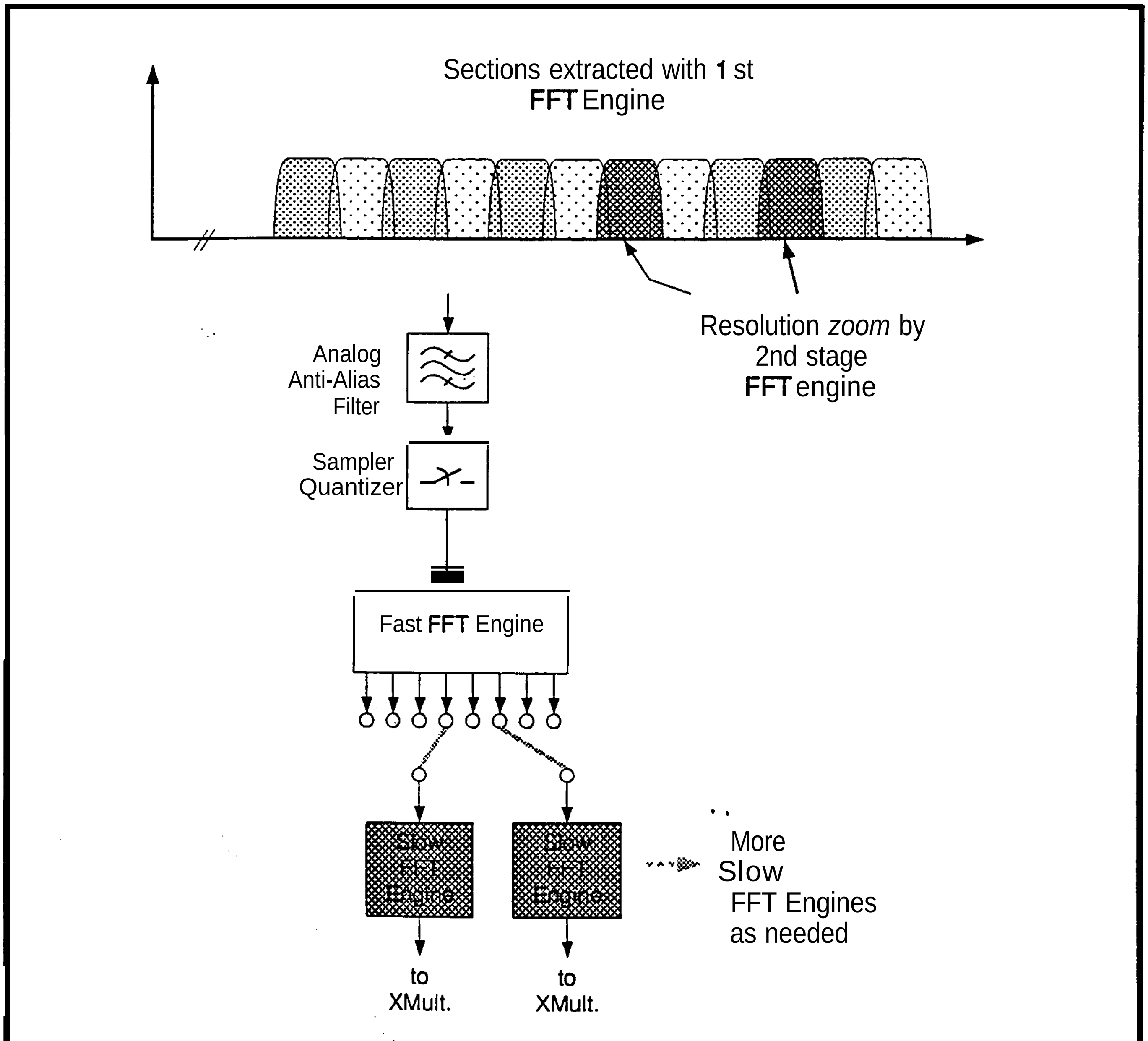

Figure 6 - Digital FFT downconversion

E. Digital FFT downconverter

Another all digital approach was suggested by Barry Clark in MMA Memo #85. This method is sketched in Figure 6. Much like the digital filter method, this replaces the conventional analog downconverters to give more frequency agility within the available IF. In this architecture, the band selection filtering is performed using a fast FFT engine. In this case, the IF band is symmetrically segregated into a fixed number of sub-bands (dependent on the length of the first FFT). Selected outputs from this first stage can be further resolved in a second layer of FFT engines. This second layer of FFT's could be followed a another layer, etc. By allowing reconfiguring of the secondary FFT engines, it create a lot of possible resolution modes. These secondary FFT's operate

Total Operations @ 67 Mhz	Multiplies	Additions
Full FFT (32 Outputs)	61,440	92,160
Full FFT (32 Outputs) with overlap	122,880	184,320

Table 3: Operations in digital FFT downconverter (does not include spectral FFT or Xmult.)

at considerably slower speeds than the initial FFT, simplifying the mode multiplexing problem.

As presented, the fast FFT produces a very coarse sample of the input spectrum. A potentially useful feature of the FFT converter scheme is the measurement of a coarse 1 Ghz spectrum, in every resolution mode. However, to fulfill this option, the front-end FFT's must be connected to a dedicated cross multiplying network (to measure cross spectra), which would require-additional, possibly unwarranted hardware expense.

An estimate of operations is needed for comparison. Like the previous discussion, this analysis will not include the 2nd stage FFT's, cross multiple network or time-division hardware. Instead, the focus will be on the 1st stage FFT. A reasonable number of frequency samples would be 32, which would divide the 1 GHz input into 31 MHz sub-bands. This requires a 64 point 1st stage FFT engine. With this arrangement, the second stage FFT's would be presented with data points at 62.5 Mhz; a reasonable rate. Back to the 1st stage, this arrangement requires 192 complex multiplications and 385 complex additions to occur during a (2 GHz/32) 62.5 MHz cycle. This gives 768 real multiples and 1152 real adds. Each Antenna/polarization would need a fast FFT, (80 total). The cumulative figures are tabulated in Table 3. The second entry includes a data window overlap to avoid loss in sensitivity.

A comparison with the digital filter method would tend to indicate a cost advantages for the FFT engine. However, it is easy to show that the FFT is simply a special case of the digital filter approach. In the architecture outlined by Figure 6, the FFT engine can be viewed as a special type of digital filter. Thus the FFT forms a fixed bank of contiguous digital filters. The FFT has computation advantages for determining many simultaneous, equally spaced filters. However, the FFT approach also places some constraint on flexibility. The FFT engine forces equal spacing and fixed bandwidth and only limited control of out-of-band rejection (using data windowing). The digital filter could give more flexibility.

V. References

[1] Schafer, Ronald, "The Math Behind the MIPS: DSP Basics", Electronics Design, Sept 8, 1988.

[2] Panek C.R. and Kator, S.F, "Custom Digital Filters for Dynamic Signal Analysis", Hewlett-Packard Journal, December, 1984.

V. References

[1] Schafer, Ronald, “The Math Behind the MIPS: DSP Basics”, Electronics Design, Sept 8, 1988.

[2] Panek C.R. and Kator, S.F, " Custom Digital Filters for Dynamic Signal Analysis”, Hewlett-Packard Journal, December, 1984.