TIME DOMAIN BEAMFORMING ALGORITHM USING A NOVEL SYSTOLIC CONVOLVER ARRAY.

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1. INTRODUCTION

Conventional beamforming techniques within modern sonars require very high levels of performance from Digital Signal Processing (DSP) hardware. Complex or large sonar arrays often adopt frequency domain techniques because they offer advantages of signal processing efficiency and data management in not needing to upsample incoming sensor data. In simpler systems where upsampling or over-sampling can be tolerated, time domain processing is preferred because algorithms require only simple 'shift & add' DSP and offer direct wide band operation.

This paper describes the implementation of a systolic convolver array that has the form of a Vector Parallel Processor Unit (VPPU) optimised for the efficient implementation of time domain beamforming. This offers a combination high DSP performance together with data management that rival frequency domain methods in complex systems. Along with the descriptions of the VPPU hardware features and algorithms, this paper also presents performance results and concludes with the next steps in our development programme.

2. BACKGROUND

During the late 1980's our underwater systems department was involved in a number of demonstrator programs and products that used various time domain and frequency domain beamforming techniques. Towards the end of this period a research programme was initiated to identify a common beamforming strategy and to develop a suitable extensible DSP hardware engine.

Our early research indicated that time domain approaches had the potential to offer the best signal fidelity performance at the expense of a high processing requirement, whereas frequency domain techniques offered the best processing throughput with signal performance limitations. Experience also showed that in small systems (especially if wide band operation is required) time domain techniques offered the simplest and most straight forward and timely solutions providing the proposed hardware is sufficiently capable.

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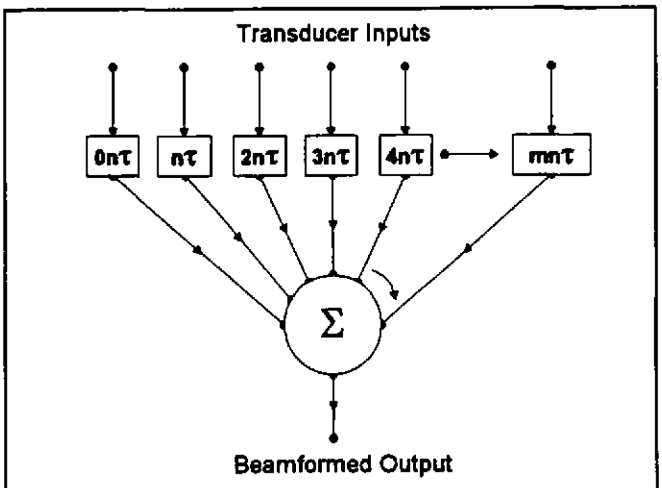

Figure 1 Time Domain Beamforming.

These conclusions, together with a product requirement for wide band operation, lead us to investigate the possibility of producing an efficient and compact time domain Beamformer. This research culminated in an adaptation of a signal reconstruction algorithm that could be used with the appropriate hardware in efficiently achieving time domain beamforming.

3. TIME DOMAIN BEAMFORMING

Consider a linear array of transducers, the outputs from these sensors can be directly summed to form a beam in a direction that is perpendicular to the line of the array. In order to steer a beam in a required direction, then the signal from each transducer must be subjected to a time delay that is a function of various physical properties of the system , its position within the array and the steer angle. This is diagrammatically represented in Figure 1. If the input from the sensors is a digitally sampled time series then a number of approaches are commonly used to perform the delay function. The first, most obvious technique is to sample the input at a sufficiently high rate that the required time delays can be accommodated by an integer number of sample periods, this is referred to as oversampling. The problem with this is that it is likely that an infinitely small sampling period may be required which may be relaxed dependant on the beam distortion, sidelobe levels and steering limitions that can be tolerated. A second problem with this technique is that, with high sampling periods compared to the Nyquist rate, high performance filters and analogue to digital converters are

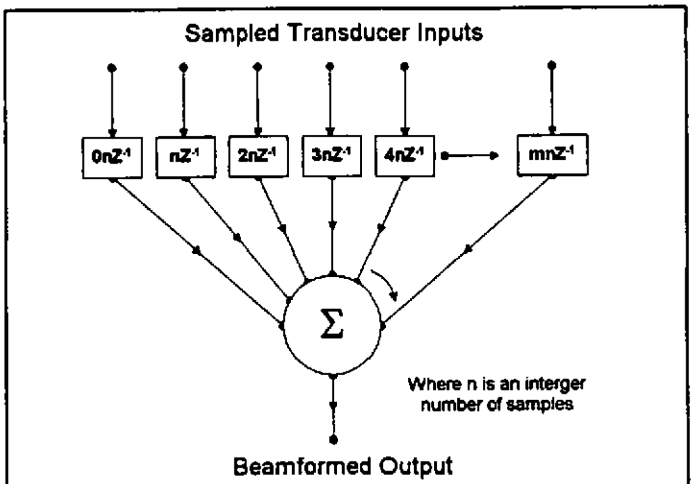

Figure 2 Beamforming using sample period delay.

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required along with a high bandwidth data highway. To overcome this second problem a Nyquist sampling rate may be used along with an interpolation algorithm that regenerates intermediate samples equivalent to that of the over-sampled system. This is commonly referred to as upsampling. Although this second technique has removed the problems associated with the front end high sample rate, a problem still exits with the handling and quantity of data within the signal processor. It is at this point, when this last problem cannot be solved, that frequency domain techniques are adopted. However if this explosion of data within the signal processor can be avoided then one may continue to use the time domain solution. To avoid the data explosion it is necessary to be able to derive a time delayed data series directly.

Initially let us look at interpolation by zero padding in an up sampling system. This derives equi-spaced intersample values by padding the original time series with zeros and then filtering (convolution) to remove the generated out of band unwanted components. The filtering function is, in general, a modified (sin(f))/f, sinc function, with its unity and zero crossing points having a period equal to the original sampling period and the intermediate coefficients derived at equi-spaced points corresponding to the zero pad positions in the input time series.

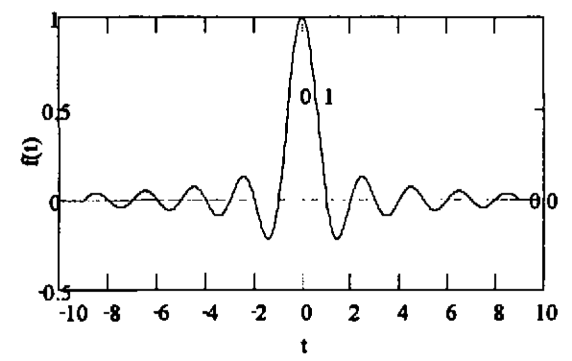

Figure 3 Sin(f(t))/f(t) function

If this operation is examined in more detail then it is realised that a lot of multiply by zeros occur and these may be used to improve performance by ignoring these operations. Also a more detailed look reveals that the only valid multiplications that take place are those that are obviously relevant to a particular output value with that particular **delay** from the preceding sampled value and that the coefficients derived are always the same for that particular delay from any preceding sampled value.

This can be shown to be identical to the basic reconstruction formula[1] that is the basis of interpolation and is defined by the infinite sum of weighted sample values

$$f(t) = \sum_{k=-\infty}^{\infty} f(kT_s) \frac{\sin[\pi(t - kT_s)]/T_s}{[\pi(t - kT_s)]/T_s} \quad (1)$$

[1] Signal processing: Discrete Spectral Analysis, Detection and Estimation - Mischa Schwartz, Leonard Shaw International student edition page 37

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This formula demonstrates that any value of f at a time t can be reconstructed from an infinite series of $f(kT_s)$ sampled at discrete points kT_s in time. In theory, therefore, we have an equation that can be used to generate a set of coefficients which are subsequently convolved with a sampled time series to generate an arbitrarily delayed reconstruction of the original series. In practice we can have very long time series data, however an infinite (or very long) set of coefficients is impossible to use. The solution to this has been to truncate equation (1) to a fixed coefficient set based on the assumption that contributions of distant sample values are small. This, however is not entirely satisfactory as this can be considered an exercise in filtering where a filter function has been windowed using a rectangular function subsequently causing significant reproduction problems due to Gibb's phenomenon. The properties of window based design of linear phase FIR filters are well known[2] and a window for our delay function can be similarly selected depending on system requirements. For simplicity our initial systems adopted a gaussian form window designed such that the maximum excursion of the most distant coefficient (from k = 0) approaches the least significant bit within the signal processing core.

$$g(t) = \exp-(\pi(t - kT_s)/10.T_s)^2 \qquad (2)$$

where k = -7 to +8 and the coefficients are represented by 12 bits.

4. VPPU

The previous section showed the limitations of conventional time domain techniques and developed a simple time delay function that may be used to overcome problems associated with oversampling or upsampling. Although having a time delay function it is still necessary to develop the appropriate hardware which can efficiently handle a significant amount of input data and can subsequently process the data to form beams. Recent commercial developments presented off the shelf components that, when combined, provided all the

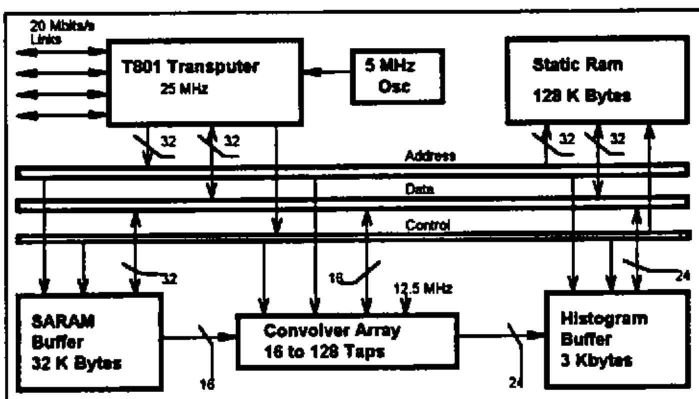

Figure 4 VPPU System Architecture

[2] Analogue and Digital Signal Processing and Coding. PM Grant et al, Chartwell-Bratt Studentlitteratur

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necessary functions required for this task and this developed into the Vector Parallel Processing Unit (VPPU).

The resulting VPPU design efficiently executes the high performance vector signal processing required for wide band time delay beamforming. The system architecture, shown in figure 4, comprises components selected for their ability to handle specific tasks required for vector processing. These consist of the Transputer host, the SARAM (Sequential Access / RAM), convolver array and a histogram buffer.

Considering the convolver array first, this device is a systolic array of 16 delays, multipliers and adders in the form of an FIR filter (or convolver) allowing the use of two banks of coefficients. The convolver array implements the function

$$y_m = \sum_{k=0}^{n-1} x_{m-k} a_k \qquad (3)$$

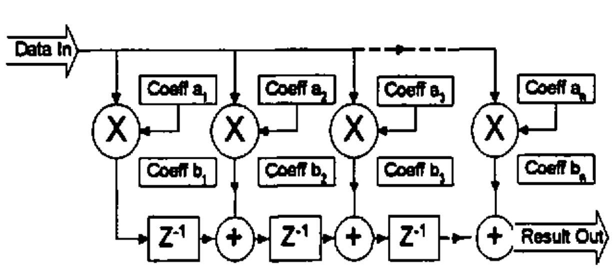

Figure 5 Convolver array

which is used to realise the time delay operation where x is the input time series, a is the product of the delay function within equation (1) and the gaussian window function in equation (2) and n is the number of stages in the convolver array. The convolver array can therefore apply a time delay to a time series data set (vector) or a set of delays to a set of data streams (vectors) if multiplexed. It should be noted that in equation (3) a history of n time series samples are required in order to receive the first valid output. This has the implication that if this device is being multiplexed then the performance of the function is reduced. This reduction in function efficiency is defined by equation (4) indicating that the longer the series the more efficient this operation will be.

$$eff = \frac{x_{length}}{x_{length} + n} . 100\% \qquad (4)$$

The VPPU is designed to be able to process sets of vectors and the supply of the vectors to the convolver array is performed by the SARAM see figure 6. This is a dual port memory device with the processor having a random access capability through one port and sequential access to the convolver array through the second port. The sequential port allows double buffer operation with a buffer chaining mode. This is achieved by allowing the user to program a set of buffer registers that define the start and end of two buffers. When set into operation a comparitor compares the end of buffer registers with the current value of the address pointer. When equal, the address contained within the next start of buffer register is loaded into the address pointer thus enabling the output of a continuous set of vectors to the convolver array.

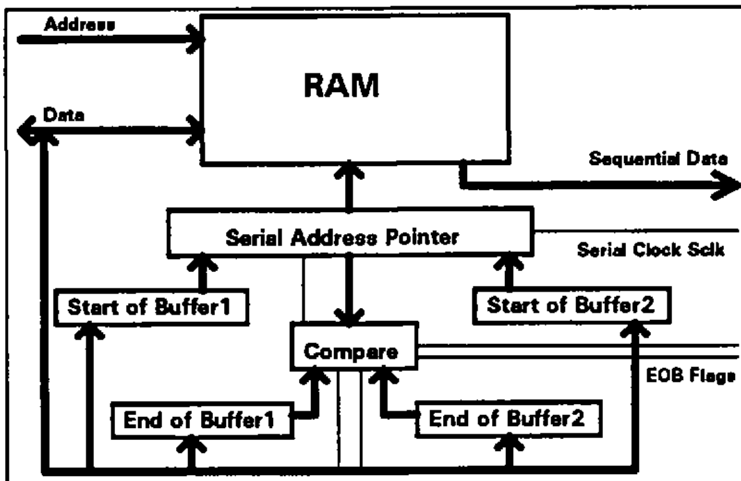

Figure 6 SARAM Function

As each vector is processed, the coefficients in the convolver array may also be updated as a double buffer operation, allowing a data vector from each sensor to have an individual time delay applied and subsequently only requiring a vector summation to complete the beamforming process. This is achieved in an image processing histogrammer device coupled with some address logic that simply converts it to a vector accumulator.

The SARAM, convolver and vector accumulator comprise the DSP function element of the VPPU board. The DSP function is under the control of a programmable sequencer engine that allows the correct synchronous operation of these elements together with the asynchronous control functions from the host control processor. One advantage of this architecture is that the DSP element is self contained and may be simply coupled to any host processor type.

5. PERFORMANCE

The performance figures presented here are for the VPPU-E (Evaluation model). In this development VPPU, the DSP engine is running at 50% clock speed and is using a host processor which has insufficient performance to complete its tasks during the time for forming one beam, therefore the DSP engine is halted to allow the processor to catch up. However even with these limitations, the VPPU is able to form 15 beams, in real time, from 32 hydrophone sensors each sampled at a 12kHz rate together with the added complication that they are not simultaneously sampled. The production VPPU is estimated to perform at three times this rate giving an ability to process 45 beams at the above data rate. Figure 7 displays the resultant beam

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plot function output from the VPPU. This plot demonstrates no deformity in the side lobe levels which occur with inappropriate interpolation functions[3] or insufficiently up/oversampled time series data.

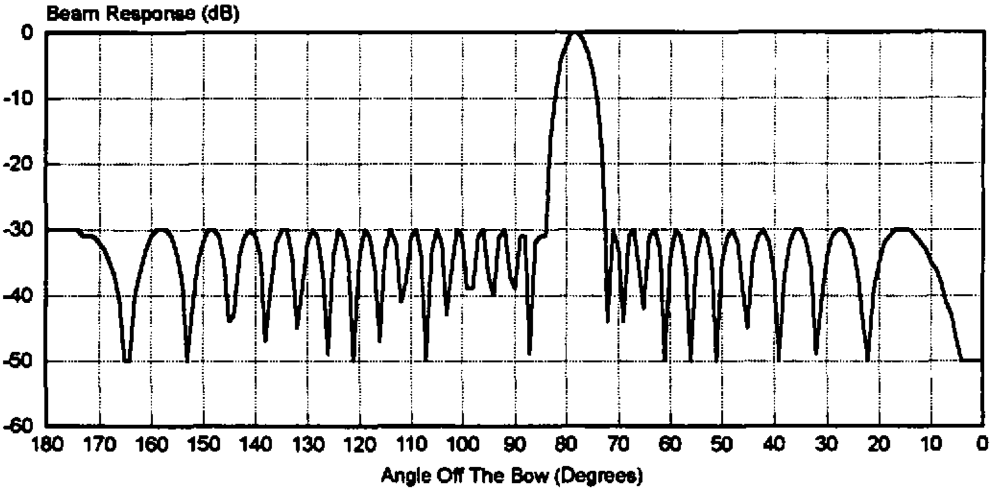

Figure 7 Beam Plot from a 32 Element Array with -30dB Dolph-Chebychev Shading

6. CONCLUSIONS

A brief history has been presented outlining our work in developing an algorithm and hardware architecture for time domain beamforming and general vector processing. The algorithm is based on the use of the fundamental sinc function that is essential in the theory of digital synthesis and the reconstruction/conversion from the digital to analogue domain. The concept of windowing has been incorporated to limit the infinite nature of the delay function in order to be able to accommodate it within practical constraints. The VPPU hardware has been described showing how a set of data vectors can be continuously passed to a systolic convolver array and then subsequently to a programmable length vector accumulator. The results demonstrate the performance, use and operation of the time delay function allowing the ability to reconstruct a delayed time series from the original sampled data stream.

[3] Beam Steering Distortion Due to Interpolation. D Butler, B Hill Plessey Company, Proceedings of the Institute of Acoustics, Advances in Underwater Acoustics: Underwater Acoustics Group Conference AUWE Portland Dec 1/2 1981

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The next stage of development is to increase the clock rate of the DSP engine and to provided an enhanced performance host processor. The DSP engine has been designed such that coupling to different host control processors is a straight forward matter allowing the use of processors appropriate to the overall design concept of the end system.

6. REFERENCES

[1] Signal processing: Discrete Spectral Analysis, Detection and Estimation - Mischa Schwartz, Leonard Shaw International student edition page 37

[2] Analogue and Digital Signal Processing and Coding. PM Grant et al, Chartwell-Bratt Studentlitteratur

[3] Beam Steering Distortion Due to Interpolation. D Butler, B Hill Plessey Company, Proceedings of the Institute of Acoustics, Advances in Underwater Acoustics: Underwater Acoustics Group Conference AUWE Portland Dec 1/2 1981