SONAR BEAM-FORMING WITH AN ARRAY PROCESSOR IN REAL TIME

by

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ABSTRACT A real-time sonar beam-forming system is described which consists of a minicomputer as host computer and an array processor. Beam-forming is accomplished by using the FFT algorithm in the processor.

The system is also connected to a multi-channel PCM magnetic tape recorder. Processing can be done either on-line or off-line. During sea trials the entire equipment is installed on board a research vessel.

An acoustic towed array supplies up to 40 input signals to the beam-former. The AP 120 B from Floating Point Systems is used as the array processor.

INTRODUCTION

Up until now, real-time beam-formers have usually consisted purely of hardware. The inadequate throughput of standard minicomputers prevented their us. Software solutions are made possible only by employing an array processor, which is specially suitable for processing large arrays of data. In contrast to the general purpose computer, an array processor has several arithmetic units and a "pipelined" structure. This allows operations on data arrays to be carried out much faster than the corresponding individual operations. In particular, address calculations and memory access are separated from the arithmetical computation and run in parallel with it. An array processor can thus work several hundred times faster than a typical general purpose computer.

In the following a report will be given on the realisation of a beam-former with the aid of an array processor; the following boundary conditions must be fulfilled:

- The sound signals are received with a line array having N equidistant hydrophones or hydrophone groups.
- The incident signal wave-front is plane.
- The unwanted noise signals at the individual hydrophones are mutually uncorrelated.

Under these conditions, the time-delay beam-former is the ideal detector. However, it is not an easy task to realise this in a digital computer. For a given sampling rate there is only a discrete number of exact beam directions. These are also called "natural" or "synchronous" beams [1]. There must therefore be a significantly higher sampling rate as required due to the sampling theorem, or use must be made of an interpolation process.

For realisation in the array processor, therefore, a process is used in which the delays are replaced by phase rotations [2, 3, 4].

In the following, a beam-former will carry out both beamforming and beam-scanning.

1. Overview of the Hardware

The overall beam-forming system consists of

- o the acoustic towed array
- o the data recording equipment, and
- o the computer system for on-line beam-forming.

Fig. 1 shows the scheme of the data recording equipment consisting of

- o the PCM modulator
- o the magnetic tape recorder
- o The PCM demodulator.

The data can be taken from the output of the demodulator in analogue and digital form. The overall PCM equipment can process up to 40 individual channels. The digitalised data are fed directly to the array processor.

The array processor computes the directional pattern and passes the result on to the host computer. The host computer is employed merely to output data to appropriate devices.

2. Beam-Forming with FFT

2.1 Narrow-Band Signals

Fig. 2 is intended to clarify the notations which we will use.

Assuming narrow-band signals, it is possible to replace the delays τ_n required for beam scanning, by phase rotations $2\pi f_0 \tau_n$ (f_0 = centre frequency of the received signal). This can be done very easily with the aid of FFT (Fast Fourier Transform) [2, 3]. The representation of the received signal in complex form is required, which can be achieved for example by quadrature sampling.

This leads to the expression

(1) $$u(\vartheta) = \sum_{n=0}^{N-1} u_{on}(t)\, e^{j[\varphi_n(t) - 2\pi n \Delta x \frac{f_0}{c} \sin\vartheta]}$$

By simple rearrangement, we obtain

(2) $$u(\vartheta_m) = \sum_{n=0}^{N-1} u_n\, e^{-j2\pi\frac{mn}{N}}$$

where

$$u_n = u_{on}(t)\, e^{j\varphi_n(t)}$$

and

$$\vartheta_m = \arcsin\frac{m \cdot c}{N \cdot \Delta x \cdot f_0} \quad .$$

$u(\vartheta)$ is also referred to as "angle spectrum" and $\frac{f_0}{c}\sin\vartheta$ as "spatial frequency". The increment between two angles ϑ_m and ϑ_{m+1} is in fact the angular resolving power of the antenna. For finer sampling of the angle range one simply has to fill up the series in (2) with zero elements. We thus obtain

(3) $$u(\vartheta_m) = \sum_{n=0}^{\tilde{N}-1} \tilde{u}_n\, e^{-j2\pi\frac{mn}{\tilde{N}}}$$

where

$$\tilde{u}_n = \begin{cases} u_n & \text{for } n \leq N-1 \\ 0 & \text{for } n \geq N \end{cases}$$

The calculation of (3) is done with the aid of FFT. For this purpose $\tilde{N}$ must be chosen as a power of 2. It can be seen that we can thus process any number N of received signals where

$$N \leq \tilde{N} .$$

2.2 Broad-Band Signals

The principle of phase compensation can be applied only to narrow-band processes. At first sight it would therefore seem that it is no longer possible to process broad-band signals. However, if all N received signals are split up into a large number of narrow-band processes, then it is once again possible to use FFT, although a large number of band-pass filters is needed for this.

A much simpler solution is again offered by FFT, which can be performed very quickly in an array processor. The incoming signals are split up into time elements of length T_W and then filtered with the aid of FFT. A "spatial FFT" is then carried out for every frequency. In this way a directional pattern is obtained for every frequency band. One generally wants just one directional pattern. For this purpose the narrow-band angle spectra must be recombined. Simple summation cannot be used, since the relationship between space frequency or beam number and angle of incidence depends on the frequency. This fact is illustrated in Fig. 3 by means of an example. Summation must therefore take place along the lines ϑ = const.

Let us point out here a feature of this processing in the reception of noise signals. For linear antennas in general a definite directional correspondence is possible with monofrequent signals only if the distance between the hydrophone groups is less than half the wavelength. At high frequencies grating lobes occur. This limitation does not apply to noise signals. The state of affairs shown in Fig. 4 is thus obtained for the

FFT processing used here. Assuming that the hydrophones act as point objects, the space frequencies are repeated periodically in accordance with the spatial sampling theorem. The lines ϑ = const. are therefore also periodic. Above the frequency $f = \frac{c}{2\Delta x}$ portions of the neighbouring spatial frequency ranges are folded for the first time into the range under consideration. The summation of the narrow-band directional functions therefore has to take place on the lines

$$(4) \qquad f = \frac{c}{\sin\vartheta}\left(f_x + k f_{xN}\right) \quad , \quad k = \ldots -2, -1, 0, 1, 2 \ldots$$

where f_x = spatial frequency

f_{xN} = spatial Nyquist frequency.

The length of the time windows T_W determines the loss of directional gain compared with the ideal time delay beam-former.

Simple considerations [3] lead to the following condition:

$$(5) \qquad T_W \gg \frac{L}{c}$$

L = antenna length

or, if we limit the evaluated angle range to $\pm\vartheta_{max}$,

$$(6) \qquad T_W \gg \frac{L}{c}\sin\vartheta_{max}$$

If the actual windows length is

$$(7) \qquad T_W = \varepsilon \frac{L}{c}\sin\vartheta_{max} \quad , \quad \varepsilon \geqslant 1$$

then for the loss in directional gain D_{LOSS} in the reception of noise signals, we can calculate

$$(8) \qquad D_{LOSS} = 10\log\left(1 - \frac{1}{3\varepsilon}\right)$$

This loss is 1.8 dB for $\varepsilon = 1$ and drops rapidly as ε increases.

For deterministic signals there is a loss of up to 6 dB, but this can be completely avoided by increasing ε and in processing of overlapping time windows.

3. Realisation with an Array Processor

When realising the real-time beam-former with an array processor, attention must be paid to the nature of the signals to be evaluated. It is not possible to deal with all types of signals in the course of this lecture. In the following, therefore, we shall confine our remarks to the processing of broad-band noise signals.

3.1 Algorithm

Fig. 5 illustrates the structure of the FFT beam-former. The temporal and spatial FFT's are shown. Then the envelopes of the signals are calculated by means of rectification and low-pass filtering. The summation on the lines ϑ = const. follows.

Fig. 6 shows a sketch of the chosen algorithm. We shall mention only the important operations here (for a more detailed treatment, see [4]).

N input signals are sampled and subjected to temporal FFT. For beam-forming, a band of interest with band width

$$B_{BF} < \frac{f_s}{2} \tag{9}$$

is selected from this and is reordered in the spatial direction. If required, amplitude shading can be carried out at the same time. The extra computer time is of no great importance.

Zeros are used for filling in to give interpolated values, and the spatial energy spectrum is calculated. Low-pass filtering follows for every point.

While computation is taking place, the data for the next computation cycle are read in per DMA (direct memory access).

The reading in of data is briefly interrupted after a freely selected number of such computation cycles, in order to make a single calculation of the broad-band directional pattern. This is transferred to the host computer, which manages the output devices. There should be an output every one or two seconds.

3.2 Program Sequence

Fig. 7 shows a diagram of the program time-squence. It can be seen that the main work - data intake, computation and filtering - is done in the array processor, and that the host computer just attends to the output of data. All activities take place simultaneously; data intake is interrupted briefly just for the calculation of signal power.

Parameters can be input to match the program to the external conditions. They determine internally

- o the antenna length
- o the frequency band evaluated
- o the time between outputs
- o the choice of output devices
- o the nature of the output.

Standard output devices are the refreshing display and the grey scale recorder.

3.3 Memory Capacity Requirement

A prerequisite for real-time beam-forming is simultaneous computation and data intake. Memory organization is therefore as follows:

- o Input buffer
- o Working memory
- o Result accumulator

The size of the input buffer is fixed by

(10) $N_{IN} = f_s \cdot T_W \cdot N$

The necessary sorting operations and the faster "not in place" FFT require a working memory of double the size.

(11) $N_{WO} = 2 \cdot N_{IN}$

The size of the result accumulator is determined by the bandwidth evaluated and the degree of interpolation in the spatial FFT.

(12) $N_{AC} = B_{BF} \cdot T_W \cdot N$

A small storage region is also required for constants.

The memory requirement is determined mainly by the product of the sampling frequency and the window length. For

$$f_s \cdot T_W = 256,$$

beam-forming with up to 16 channels can be performed with a 16 K data memory.

For 40 channels, more than 32 K is required.

3.4 Speed of Operation

The computation time T_c also depends mainly on the product

$$f_s \cdot T_W$$

according to a function

(13) $T_c = \alpha \cdot N \cdot f_s \cdot T_W + \beta \cdot B_{BF} \cdot T_W$

where α, β are hardware constants.

If the computation times quoted for the AP 120 B are summed, we obtain

29.3 ms

for 8 channels and $f_s \cdot T_W = 256$. For every additional 8 channels there are another 9 ms. A little more time is needed for some minor management tasks. The computation time measured for 8 channels was 30 ms.

For gapless reading in with 8 channels there is thus a maximum sampling frequency of

8.5 kHz per channel

and with 40 channels it is still

3.9 kHz per channel.

The single calculation of the broad-band directional pattern requires another 17 ms.

4. Modifications

A large variety of modifications can be effected by means of simple program changes. Thus, for example, output can take place with linear or logarithmic scales and with any normalisation. The extra time required for conversions to dB, for example, is only 1.5 ms per output and is thus of no great significance.

The use of time windows or frequency weighting can also be implemented with little trouble. The Hanning window can be quoted here as an example. With a very slight programming effort the additional execution time for 256 points is about 200 μs.

However, a reformatting and reordering operation is required in any case. With a little more programming effort, therefore, the Hanning window can be implemented with no extra time requirement.

Conclusions

Beamforming is possible by means of the combined temporal and spatial FFT. This type of signal processing is ideally suited for an array processor. The system described in this paper was already successfully tested on board a research vessel. The current system is able to process up to 16 individual input channels. The maximum band-width is fixed to 1 KHz. It is planned to increase the memory capacity so that the processing of up to 40 channels is possible in real time.

REFERENCES

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[3] R. Diehl:
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[4] Walter G. Wagner:
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BL 4609, 1979, confidential, Krupp Atlas-Elektronik.

DISCUSSION

C. Richardson You describe computation of the broadband directional noise spectrum within the AP. However, the linear array may be subject to low velocity coherent energy, i.e., $k > \omega/c$ [k = wave number or spatial frequency], which is, therefore "outside" the directional arc. Have you investigated this? (The question relates to array design rather than operational performance.)

W.G. Wagner What we are dealing with in this paper is the implementation of a beamforming algorithm with an array processor. In the introduction I gave the boundary conditions belonging to the physical situation.

J.M. Griffin You stated that the memory requirements for the work area is twice the channel-input sample size product. Since only one channel is transformed at a time why was so much memory allocated?

W.G. Wagner The input data is in channel order first and then in time order, so a re-ordering algorithm is required. For simplicity, and in order to reduce programming effort, more memory space than necessary was used. As it is intended to expand the number of input channels, the program structure had to be modular so no extra programming was required.

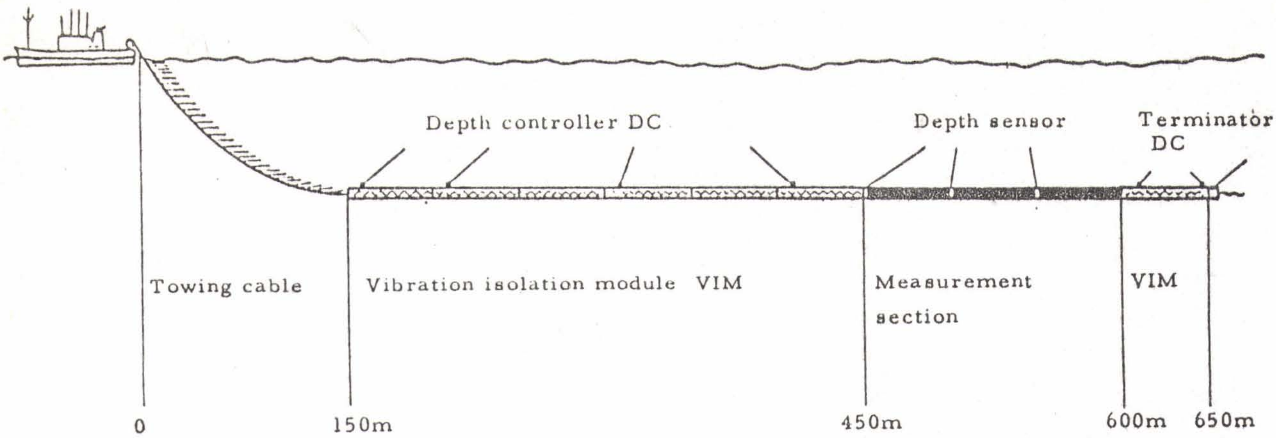

FIG. 1 TOWED ARRAY SYSTEM OVERVIEW

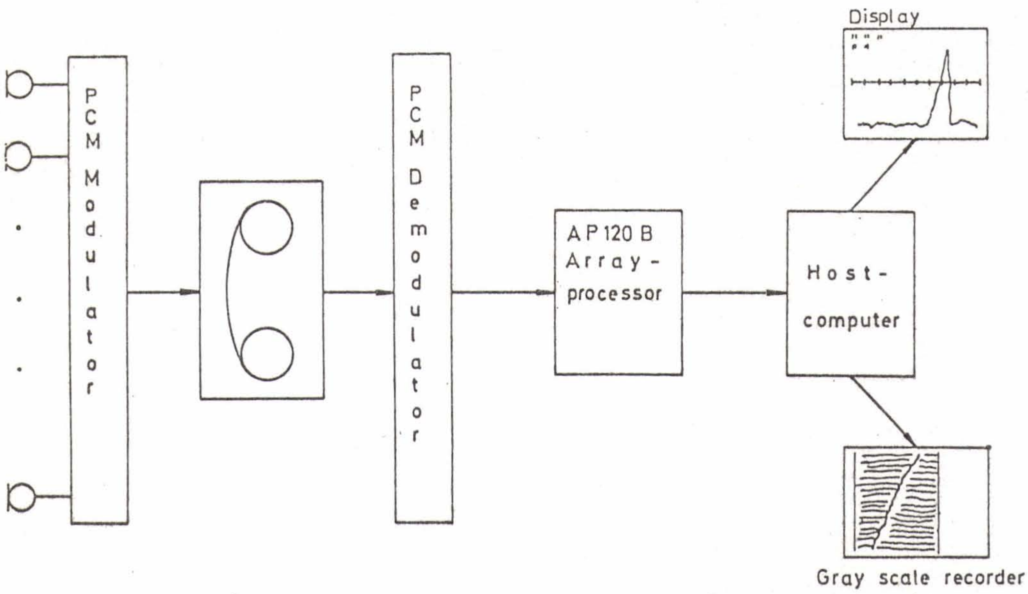

FIG. 2 HARDWARE FOR DATA RECORDING AND ON-LINE/OFF-LINE EVALUATION

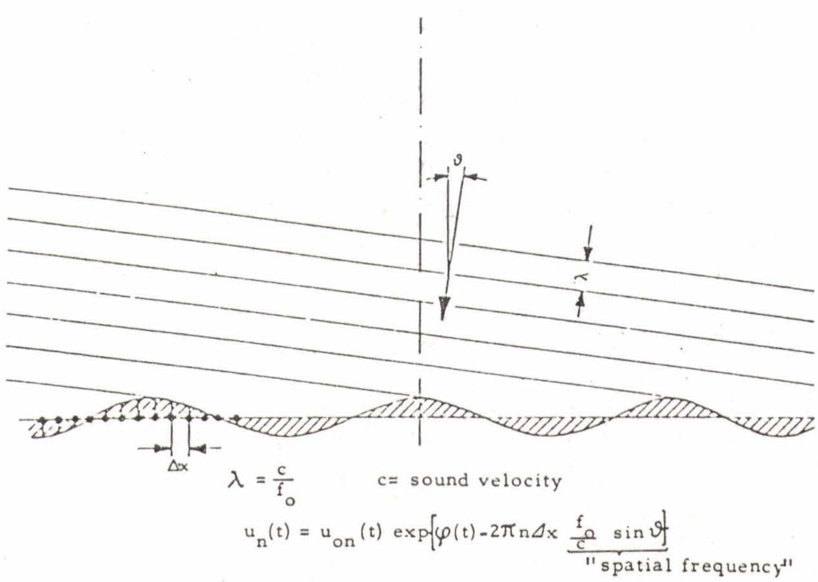

FIG. 3 NOTATIONS

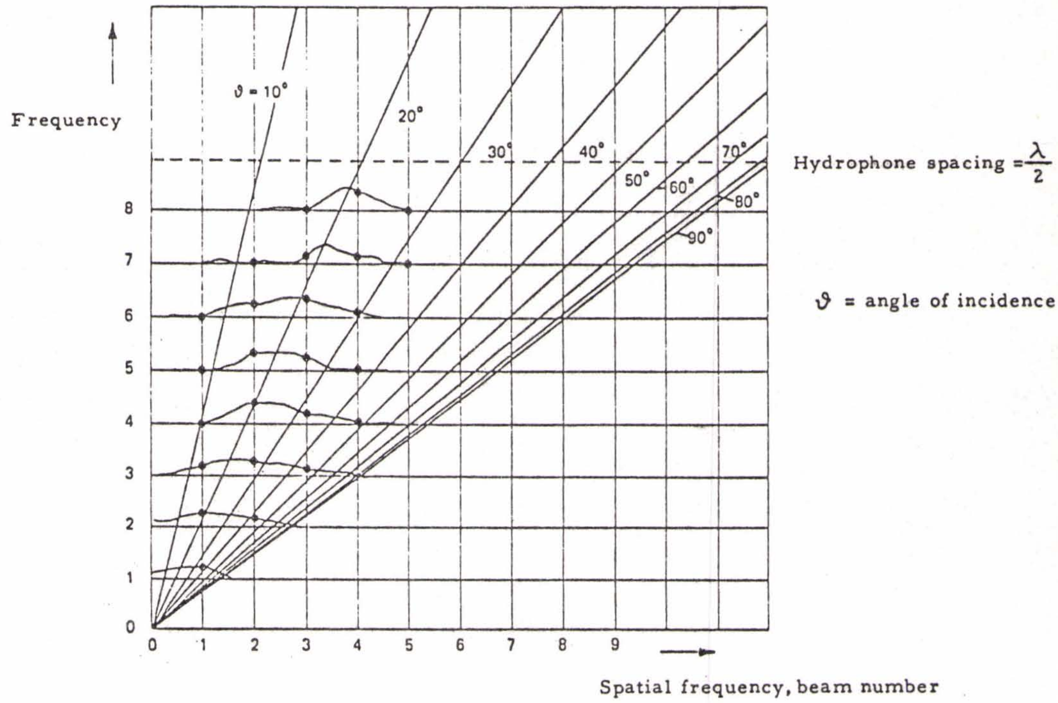

FIG. 4 RELATIONSHIP BETWEEN SPATIAL FREQUENCY AND SIGNAL FREQUENCY

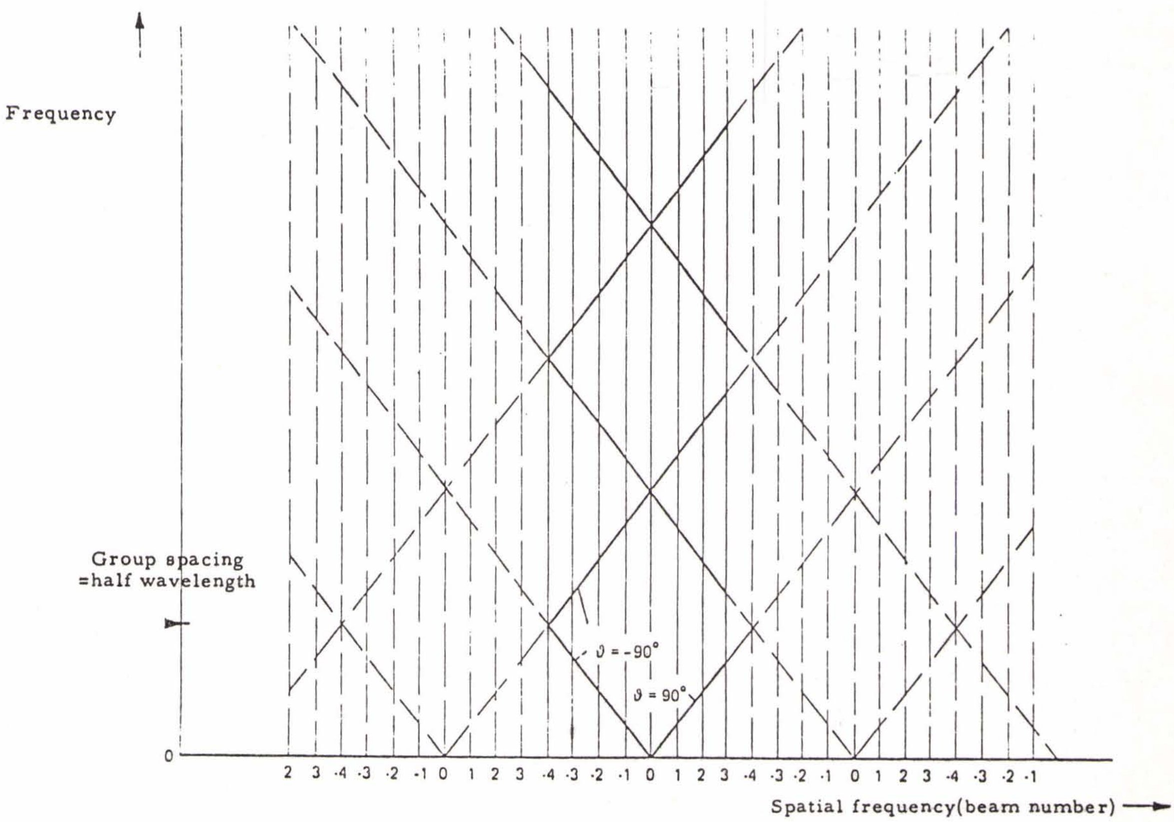

FIG. 5 SPATIAL PERIODICITY WITH FFT BEAMFORMING

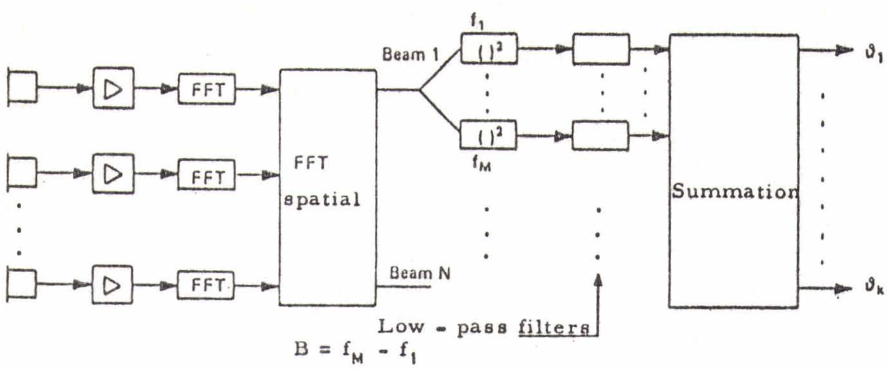

FIG. 6 FFT BEAMFORMING - STRUCTURE

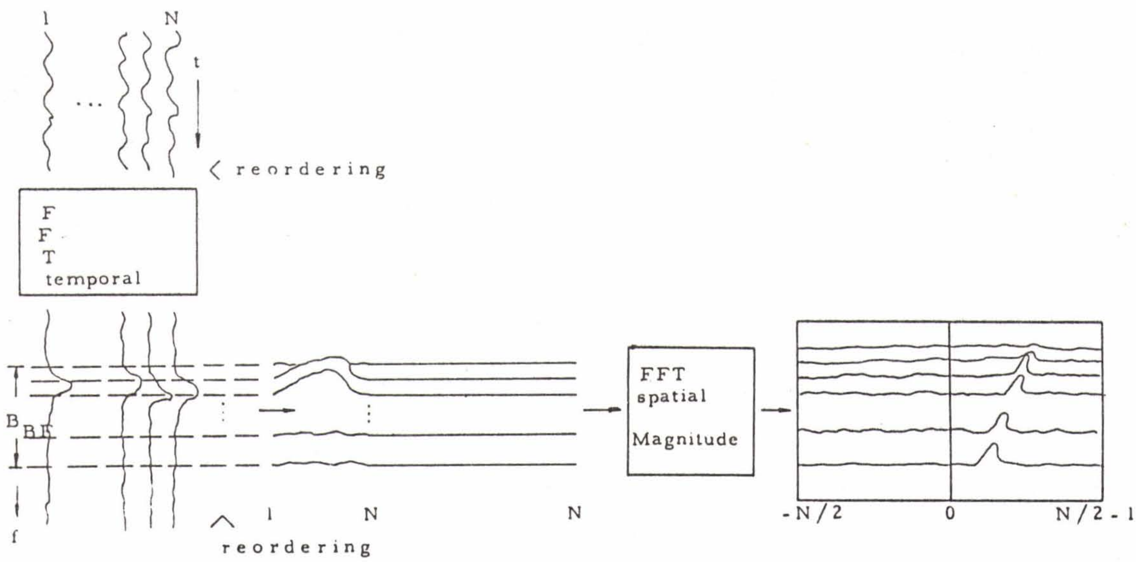

FIG. 7 SKETCH OF THE ALGORITHM

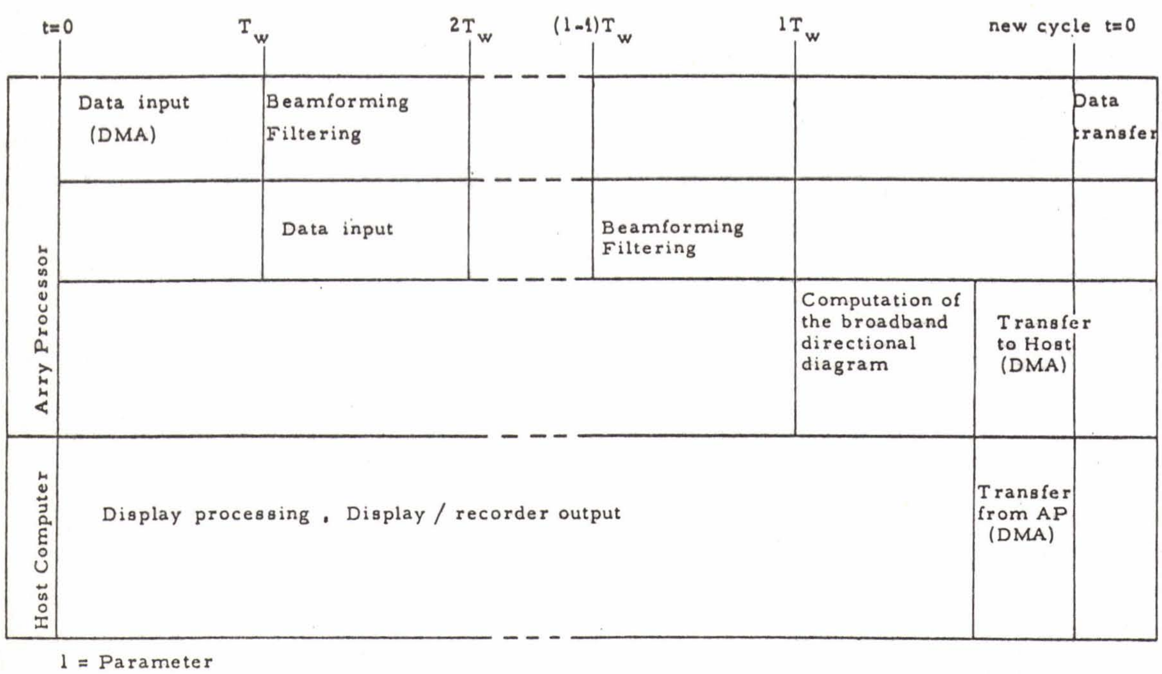

FIG. 8 TIME SEQUENCE OF BROADBAND SIGNAL PROCESSING