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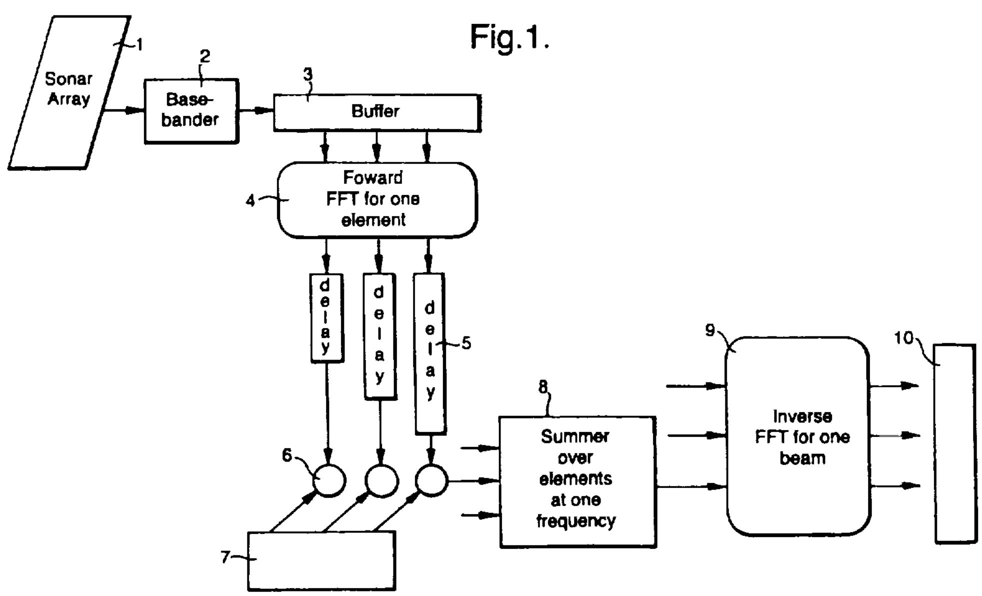
On-line:WPI, CLAIMS, INSPEC

(54) Abstract Title
Processing of signals incident on an array

(57) A method of signal processing comprises sampling a time sequence of signals incident on a plurality of elements of an array and transforming these signals between time and frequency domain, then applying a time delay to each of the transformed signals to align the signals for correlation. A separate phase shift is applied to each of these signals for beamsteering and the time aligned transformed signals are summed, then inverse transformed between time and frequency domain to produce a beamformed and correlated output signal. The method is carried out using a signal processing device which comprises an array (1), a transform (4) for transforming signals incident on the array between time and frequency domain; a delay (5) for applying a time delay to each of the transformed signals to align the signals for correlation; a phase shifter (6) for applying a phase shift to the aligned signals; a summer (7) for summing the time aligned transformed signals, and an inverse transform (8) for transforming the summed output between time and frequency domain.

Fig.1.



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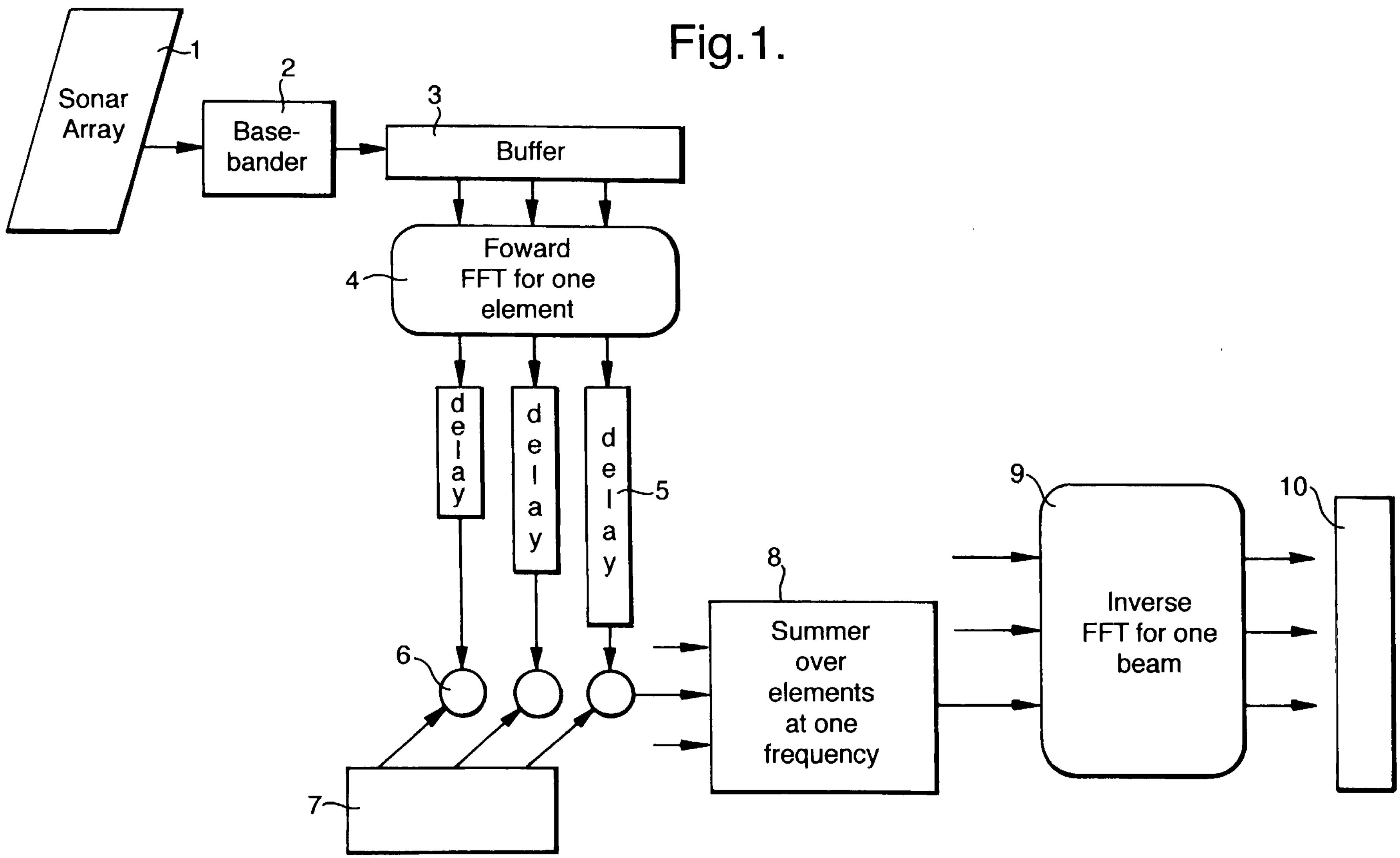
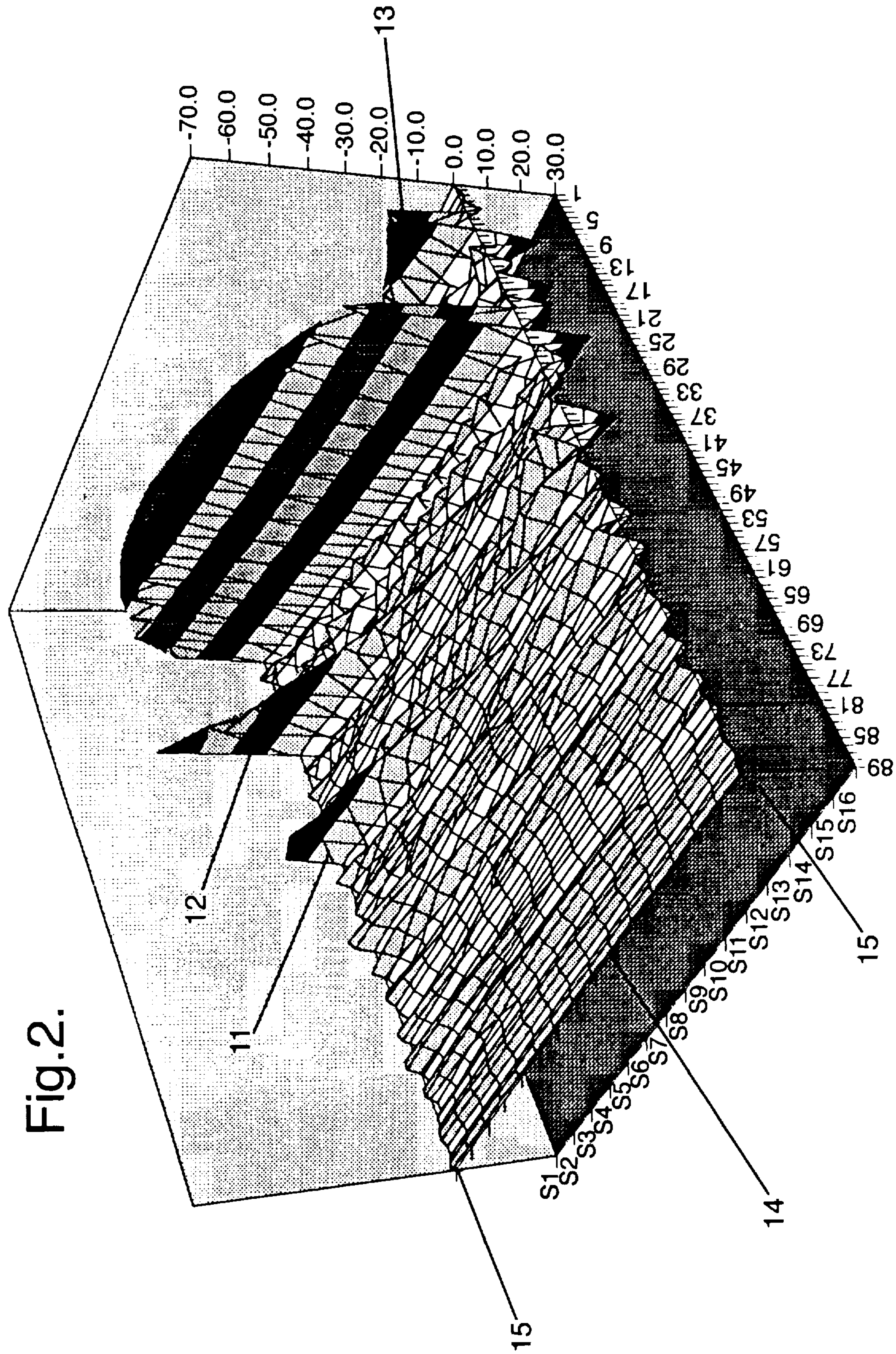
At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

Fig.1.



Sonar Array
1
Base-bander
2
Buffer
3
Foward FFT for one element
4
delay
delay
delay
5
6
7
Summer over elements at one frequency
8
Inverse FFT for one beam
9
10

Fig.2.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57
61
65
69
73
77
81
85
89
S1
S2
S3
S4
S5
S6
S7
S8
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S11
S12
S13
S14
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S16
11
12
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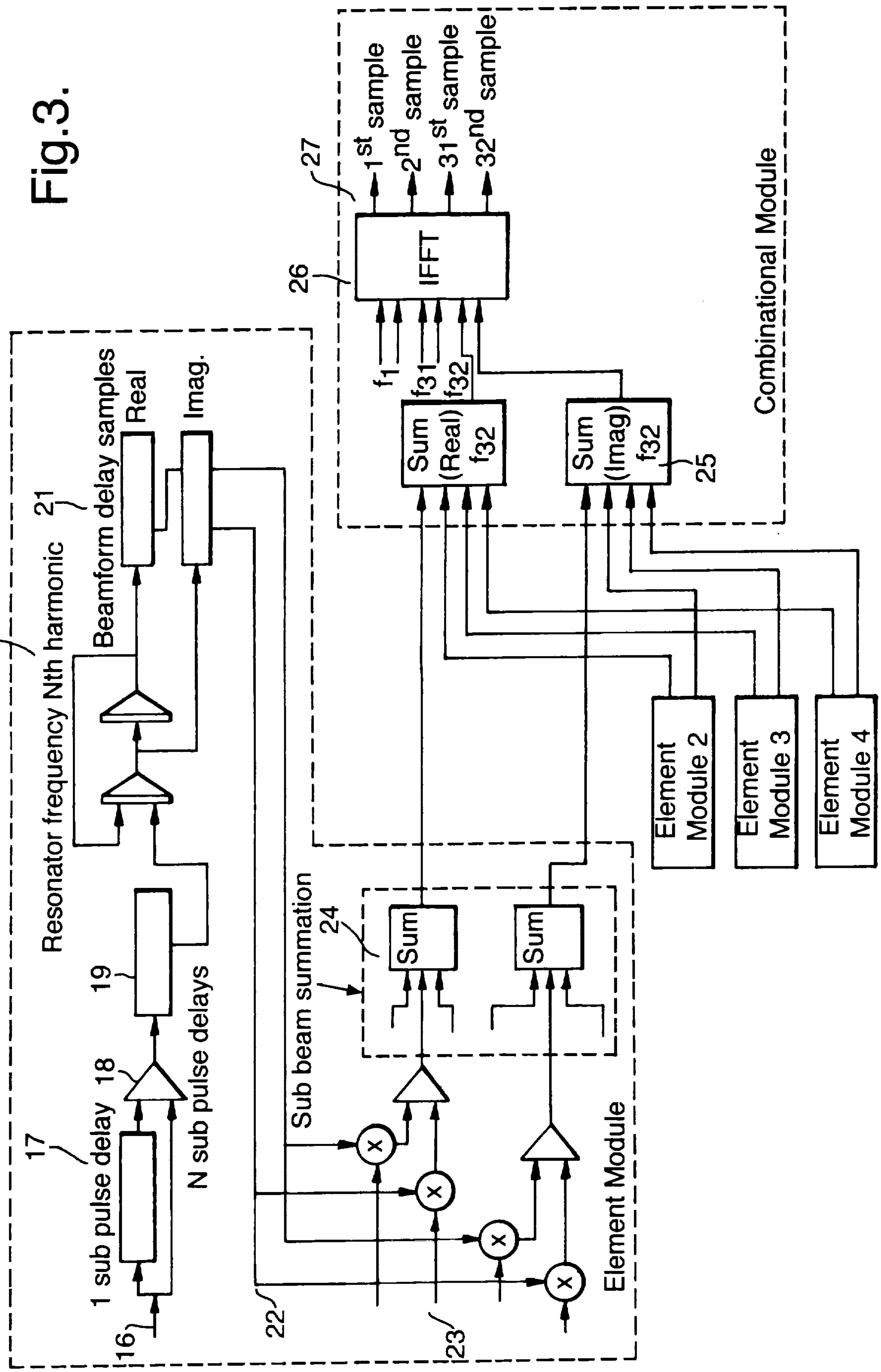
14
15
15

Fig.3.



Resonator frequency Nth harmonic
20
Beamform delay samples
21
Real
Imag.
1 sub pulse delay
17
18
N sub pulse delays
19
16
22
23
Sub beam summation
24
Sum
Sum
Element Module
Element Module 2
Element Module 3
Element Module 4
Sum (Real) f_{32}
Sum (Imag) f_{32}
25
f_1
f_{31}
f_{32}
f_{32}
IFFT
26
27
1^{st} sample
2^{nd} sample
31^{st} sample
32^{nd} sample
Combinational Module

METHOD AND APPARATUS FOR SIGNAL PROCESSING

This invention relates to a method of signal processing, in particular for receiving wide band signals.

Particular applications of signal processing include active remote sensing, synthetic aperture processing and spread spectrum communication in the fields of sonar, radar, and space communications. There are various known methods of carrying out the processing steps. Signals incident perpendicular to a straight line array may be summed coherently. Signals which are incident at an angle will have a different phase at each element of the array, but the array can be steered electronically to compensate for this, so in time domain beamforming, an electronic time delay is applied to the signal from each array to enable them to be summed coherently. In frequency domain beamforming, the signals from each array are transformed to the spatial frequency domain in which the skew in the incident sine wave corresponds to the spatial frequency. This transformation is usually achieved by applying a Fast Fourier Transform (FFT) to the set of array signals.

Frequency domain beamforming is preferable because it requires a lower sampling rate, the Nyquist frequency, two samples per cycle, compared with the multiple samples needed for time domain beamforming. However, a problem of using spatial FFT's is that they only work for narrowband signals. For wide band signals, the angle of electronic steer depends on the frequency. There is a requirement for the systems described to be able to process wide bandwidth pulses to achieve short range resolution and low reverberation levels. In sonar systems the signal reflected from the surface, seabed and water column is termed reverberation; it masks low target strength echoes; this problem is related to the area insonified and thus to the range resolution. A short pulse (with good range resolution), however, only has a small energy and can be affected by noise sources, such as from pre-amplifiers in the equipment or from rain. Commonly, a long, wideband pulse is transmitted and the received pulse is compressed by correlating it with the transmitted pulse. This is very computationally intensive and the processing required increases with the bandwidth.

In accordance with a first aspect of the present invention, a method of signal processing comprises sampling a time sequence of signals incident on a plurality of elements of an array; transforming these signals between time and frequency domain; applying a time delay to each of the transformed signals to align the signals for correlation; applying separate phase shifts to each of these signals for beamsteering; summing the time aligned transformed signals; and inverse transforming the summed output between time and frequency domain; whereby a beamformed and correlated output signal is produced.

The method of the present invention is able to produce a beamformed and correlated output signal by summing time aligned transformed signals. The sampled signals are derived from a transmitted signal which is constrained to comprise discrete sub-sections, each sub-section comprising an independent band of frequency components of bandwidth less than the bandwidth of the signal and wherein each sub-section has a duration greater than or equal to the inverse of its bandwidth

Typically, the bandwidth b of each sub-section is less than or equal to the total bandwidth B divided by the square root of the product of the signal bandwidth and the transmitted pulse length, T, i.e. $b <= B / (BT)^{-2}$

Typically, the duration t of each sub-section is greater than or equal to the total duration T divided by the square root of the product of the signal bandwidth and the transmitted pulse length, T, i.e. $t >= T / (BT)^{-2}$

Preferably, the signals are sampled at a sampling rate of twice the signal bandwidth.

Preferably, the time delay applied to each signal is an integer multiple of the duration of each transformed signal.

Preferably, the signals comprise quantised chirp signals.

Preferably, the component frequencies of the quantised chirp signals are harmonically related and have equal durations.

Preferably, alternate sinusoidal components of the aligned signals are converted into cosinusoidal components by applying a suitable phase shift, whereby a single correlation peak is generated when the final output is generated by Fourier synthesis.

Preferably, the phase shifts for beamsteering are precalculated and stored in a look up table.

In accordance with a second aspect of the present invention, a signal processing device comprises an array; transform means for transforming signals incident on the array between time and frequency domain; delay means for applying a time delay to each of the transformed signals to align the signals for correlation; phase shift means for applying a phase shift to the aligned signals; and summing means for summing the time aligned transformed signals, and inverse transform means for transforming the summed output between time and frequency domain; whereby a beamformed and correlated output signal is produced.

An example of a method of signal processing in accordance with the present invention will now be described with reference to the accompanying drawings in which:-

Fig. 1 is a block diagram for apparatus for frequency domain signal processing according to the present invention;

Fig. 2 is a 3-dimensional plot of a signal processed in accordance with the present invention; and,

Fig. 3 is a block diagram for apparatus for time domain signal processing according to the invention.

The apparatus described in this example is for signal processing from a sonar array, although it is equally applicable to communications and radar. The system of Fig. 1 operates in the frequency domain and shows a sonar array 1 which receives signals derived from a transmitted waveform reflected from objects onto each element of the array. The preferred transmitted waveform for use in the systems described is a quantised chirp waveform. It is a long pulse made up of a number of sections of equal duration, precisely equal to one cycle of a fundamental frequency, f. Each section contains a sinusoidal waveform of different frequency, the frequencies forming a harmonic series which need not be in ascending order. For a frequency which is an integer multiple h of the fundamental frequency f then h*f cycles of the h th harmonic fit exactly into one section. This allows smooth transitions from the end of one section to the start of the next.

Although, the signal in each section is nominally of the harmonic frequency h*f, the signal starts and stops after a particular time, s. The bandwidth of the signal is 1/s and the total bandwidth B of n adjacent sections is n/s. The total pulse length, T of n sections is n*s, so the BT product which is a measure of processing gain is n/s*n*s, i.e. n^2. With a quantised chirp waveform used for correlation (a matching waveform), a Fourier transform of size n separates the frequency from each section of the matching waveform into a different independent frequency channel feeding the delays 5 of Fig. 1.

The signals from the array are basebanded 2 to reduce the sampling rate needed, and sampled into a buffer 3. This operation shifts the frequency components of the received signals from the elements of the array, so that the highest frequency in the carrier frequency chirp maps to the fundamental of the basebanded chirp, and lower harmonic frequencies in the carrier chirp map to subsequent higher harmonics in the basebanded chirp. Then the basebanded time samples of the signals from each element are transformed as a block by a FFT 4 to generate complex number frequency coefficients for each frequency component. Separate time delays 5 are applied to the frequency coefficients from the FFT 4 to align the sinusoidal waveforms for subsequent Fourier synthesis of a the compressed waveform. A phase adjustment coefficient 6 is applied from a look up table 7 to the frequency coefficients to rotate their phase. The phase adjustment coefficient is dependent upon which element of the array the signal is from, the signal frequency and which beam, of plurality of beams which form an image representing an area of interest, is to be formed. The phase shifts are pre-calculated to correspond to the delay needed for a simple time domain beamformer. The complex frequency coefficients are summed 8 over all the elements and input to an appropriate frequency input of an inverse FFT 9 for the particular beam. The output 10 of the FFT is a beamformed time series of the data signals.

Implementation of the method may be achieved in either hardware or software steps. In either case a large number of small FFT's are performed and the data stored in shift registers or ring buffers to implement the alignment delays for the correlation step. Accessing a large number of data stores can be time consuming because of having to load a separate pointer to access each, but this can be overcome by implementing all delays in a single ring buffer array with a set of interlinked pointers to control the access. The processed signal 10 is output to a display through a graphical interface.

Fig. 2 shows a 3-D plot of successive blocks (numbered 1 to 89) of coherently processed output data signals. The time domain output is plotted for successive blocks of coherently processed output on a 16-sample axis (S1 to S16), and successive processing blocks at successive instants of the array data (highest frequency) sampling rate are plotted in the orthogonal axis (numbered 1 to 65). Range sidelobes (11, 12, 13) appear towards the beginning and end of each processing block, particularly for the cosinusoidal source waveform. These will manifest themselves on target echoes as additional sidelobes spaced at a sub-pulse apart, and as lower amplitude echoes whose amplitude fluctuates with the random offset of the position of the start of the processing block relative to the main target.

It can be seen from Fig. 2 that the amplitude of the sidelobes is quite low near the middle 14 of the processing block, but becomes significant at the edges 15. By using just the middle section of the processing blocks (i.e. masking out the signal at the edges), and filling in the signal between adjacent processing blocks with a second set of interleaved processing blocks, the amplitude of these range sidelobes can be kept to an acceptable level. Equally sized interleaved blocks give adequate performance, although other ratios could improve the processing efficiency or the sidelobe suppression.

Fourier synthesis consists of summing sinusoid or cosinusoid waveforms to produce a given waveform. If cosines are used, two peaks are generated - one at the start of the processing block and a second at the end. Phase reversal of alternate (complex) frequency coefficients can be used to align the peaks to produce a matched filtered output of a single response in the time domain. If sinusoids are used, the summation produces a double peak straddling the mid-point of the processing block. The sinusoids can be converted into cosines by phase rotating (multiplying the complex frequency components by j), and then applying the phase reversal technique to generate a single centred peak to the matched filtered waveform.

The complex coefficients needed to produce the phase shifts for each element, harmonic and beam combination are preferably stored as a lookup table 10. The phase changes needed to align the peaks for sine or cosine matching waveforms may be incorporated into the coefficients stored in the table.

Correct alignment of frequency bands, corresponding to frequencies in each sub-section as received from the sea bed, for correlation by Fourier synthesis can be checked by displaying or printing the complex frequency components calculated at each block of the processing. The simplest test is to apply an impulsive echo which reflects the transmitted chirp back to the receiver. After n blocks have been processed (n=sqrt BT), all the frequency components of a particular beam should have their maximum amplitude simultaneously. Numeric display methods are not appropriate for checking time domain or multi-beam data, since the method of signal processing operates on wide band signals consisting of many samples, so a graphical interface is used.

In a second example operating in the time domain shown in Fig. 3, data 16 from each element of the array is transformed by delaying it by one sub-pulse in delay line 17 and subtracting in a subtractor 18, passing the output through a tapped delay 19 and separating the frequency components into separate time domain signals using state variable digital filters, otherwise known as resonators 20, to initiate a resonant response and terminate it after exactly one sub-pulse.

This operation provides time samples of the signal separated into its component frequencies, and requires sampling at least twice the Nyquist rate i.e. four times the bandwidth of the data signal. However, even this sampling rate is too low to provide a vernier time delay for multi-element beamforming. For example, to create a synchronous beamformer for a 32-element, 32-beam system with a bandwidth of 64 kHz requires of the order of 5 MHz sampling rate, whereas twice Nyquist is only 250 kHz.

It is however possible to provide vernier adjustment since the resonator provides both in-phase ('real') and quadrature ('imaginary') time domain samples 21 from the outputs of two integrators which comprise the resonator stage 20. These are used to modify the phase of the resonator output by a small angle needed to align the signals from elements to the incoming wavefront at a desired steer angle. The alignment then consists of two components - a coarse delay 22 provided by an offset in a ring buffer, and a fine or vernier component 23 produced by phase rotation using the complex components of the time domain signals.

It might be thought that the individual resonator outputs could be sampled less frequently than twice Nyquist of the wide band signal, since they each comprise only (1/sqrt(BT)) of the total bandwidth of the signal. However, this approach fails to operate accurately because of the out of band components which pass, attenuated, through the resonator components. The time signal at each resonator is a time-limited pulse of a single frequency, and in the frequency domain this maps to a sync function rather than a single frequency. Measurement of the magnitude of the complex vector shows that its amplitude is modulated within the sub pulse period when the time domain correlator resonators are fed with the matching waveform, and samples taken at Nyquist for the sub-band (i.e. 1/sqrt(BT) of the overall Nyquist rate) would be subject to similar fluctuations because the sampling instant for the lower sampling rate occurs randomly relative to the position of the peak in the echo return.

The phase shifted frequency coefficients aligned for correlation are summed 24 across all frequencies, and the resulting components of the summed beams are summed 25 for each element, then input to an inverse FFT 26 to generated a series of time domain outputs 27.

The method of the present invention can be applied to matched filtering and beamforming processing of wide band signals, particularly for active remote sensing, including synthetic aperture processing, and spread spectrum communication, in fields including sonar, radar, airborne sonar, and space communications.

Additionally, since the intermediate frequency-separated and spatially-separated signals are directly controllable within the process, many additional processes can be incorporated into a beamforming/correlation processor to increase the capability of the overall package. Examples are separate detection (rectification) of individual frequency bands prior to signal combination in active sonar systems. This can be used to increase the reverberation to shadow ratio or alternatively to match the range resolution of the sonar to the expected target size.. Another application is in measurement of the spectral spread of the return signal which provides a measure distinguishing point reflectors from random reflectors.

Processing efficiency using this method can be greater than conventionally available from time domain or FFT beamforming techniques, thus allowing use of wider BT products with greater processing gain against noise and reverberation. Individual beams may be separately steered, to provide high resolution windows within a lower resolution backdrop, or simply an electronically steerable window. Beam focusing, and dynamic focusing, may be achieved allowing operation in the near field of a conventional array or of a synthetic array produced by the motion of a real array. Other applications are high resolution beamforming algorithms and height finding algorithms.

CLAIMS

1. A method of signal processing; the method comprising sampling a time sequence of signals incident on a plurality of elements of an array; transforming these signals between time and frequency domain; applying a time delay to each of the transformed signals to align the signals for correlation; applying separate phase shifts to each of these signals for beamsteering; summing the time aligned transformed signals; and inverse transforming the summed output between time and frequency domain; whereby a beamformed and correlated output signal is produced.

2. A method according to claim 1, wherein the signals are sampled at a sampling rate of twice the signal bandwidth

3. A method according to claim 1 or claim 2, wherein the time delay applied to each signal is an integer multiple of the duration of each transformed signal.

4. A method according to any preceding claim, wherein the signals comprise quantised chirp signals.

5. A method according to claim 4, wherein component frequencies of the quantised chirp signals are harmonically related and have equal component lengths.

6. A method according to claim 1, wherein alternate sinusoidal components of the aligned signals are converted into cosinusoidal components by applying a suitable phase shift, whereby a single correlation peak is generated.

7. A method according to any preceding claim, wherein the phase shifts are precalculated and stored in a look up table.

8. A signal processing device; the device comprising an array; transform means for transforming signals incident on the array between time and frequency domain; delay means for applying a time delay to each of the transformed signals to align the signals for correlation; phase shift means for applying a phase shift to the aligned signals; and summing means for summing the time aligned transformed signals, and inverse transform means for transforming the summed output between time and frequency domain; whereby a beamformed and correlated output signal is produced.

The Patent Office

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Application No: GB 9718999.7
Claims searched: 1-8

Examiner: John Betts
Date of search: 13 February 1998

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.P): H1Q (QFF) G1G(GES) H4D (DRPE, DRPS, DRPV)

Int Cl (Ed.6): H01Q 3/26 G01S 15/89 G01S 13/90 13/28

Other: On-line: WPI, CLAIMS, INSPEC

Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
X	WO95/29479 A1 (Brown University) whole document	1,2,7, 8

X Document indicating lack of novelty or inventive step
Y Document indicating lack of inventive step if combined with one or more other documents of same category.
& Member of the same patent family
A Document indicating technological background and/or state of the art.
P Document published on or after the declared priority date but before the filing date of this invention.
E Patent document published on or after, but with priority date earlier than, the filing date of this application.