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A METHOD OF ACOUSTIC PHASE CALIBRATION USING SMALL, SLOW SAMPLED, DATA SETS.

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INTRODUCTION

In this paper we shall be discussing a practical method of acoustic phase calibration for line arrays using Discrete Fourier Transforms. The method could, with some modification, be applied to arrays of other profiles. Earlier calibration techniques have often resorted to a qualitative assesment of the characteristics of typical broadside beams, this practical procedure will produce quantitative results which lead to a better understanding of the array behavior.

PHASE CALIBRATION

Multi-transducer sonar arrays have been used for many years to identify the direction of coherent noise sources in a marine environment. Time or frequency domain beamformation is used to improve sensitivity in the direction of the source. Beam characteristics are influenced by the phase matching of elements within the sonar array. Reference 1 describes the effects upon beam characteristics of array element phase mismatch. The ability to predict beam characteristics will enable the formulation of array performance quantifiers, such as the Directivity Index (see reference 2).

If sonar arrays are to function in a predictable manner it is imperative that the phase and sensitivity of the individual elements are closely matched. In some systems it is possible to build up the arrays from closely matched elements but in others the arrays have to be completed prior to the acoustic calibration. In the latter the variance of the individual elements should be taken into account in the electronics behind the acoustic elements. In either case the behaviour of elements may change during construction and post construction array calibration will be required to establish this.

Urick, reference 2, describes calibration as 'the determination of the response as a function of frequency and direction'. We define array phase calibration to be the determination of array element phase mismatch. In situ acoustic phase calibration is a difficult procedure with the following problems. Firstly, phase calibration is achieved by the comparison of the outputs from elements simultaneously excited by a sinusoidal accoustic signal. This procedure relies upon a sinusoidal waveform arriving at each element of the array with matched phase.

A METHOD OF ACOUSTIC PHASE CALIBRATION.

The transmitter is therefore required to be far enough away to produce a wavefront with negligible curvature at the array face. Secondly, it is also important that the signal recorded should be as long and noise free as possible with minimal multi-paths, which would corrupt the signal. Signal multi-paths can most easily be avoided by ensuring that the nearest reflective surfaces are a considerable distance from the array and signal transmitter, although this is often impractical.

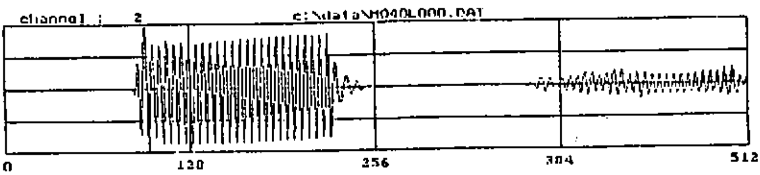


Fig 1 shows the desired signal form with a strong sinusoidal signal followed, after a discrete interval, by the first multi-path. This signal is an example of the data collected by Ferranti in array phase calibration trials at Lake Kelk near Bridlington.

Prior to the advent of digital techniques arrays were calibrated with analogue processors, such as correlators, which were used to determine the time variance between elements, and hence the phase mismatch. The use of analogue correlators requires the duplication of hardware for each channel to be simultaneously considered. The correlation equipment has to be calibrated to ensure the phase matching of the correlators, as a result it is normal to match only pairs of array elements. However this method results in a very arduous procedure to establish the phase matching of elements in a large array

Having simultaneously recorded a signal pulse from each channel of a sonar array, phase mismatch between elements can be measured by comparison of the waveforms with a reference. Comparison of two sinusoidal waveforms to measure phase shift can be achieved by identifying the time at which each signal crosses a given point, typically zero. Phase extraction by time of zero-crossing can be very accurate, but relies upon very high sampling rates and low noise levels. At sampling rates less than twice Nyquist the accuracy becomes very coarse.

Our problem was therefore to extract signal phase from short sinusoidal signals, sampled relatively slowly, in a noisy environment. Our solution is based upon Fourier Transforms and permits the reduction of sampling frequency to two times Nyquist, while maintaining an acceptable phase resolution.

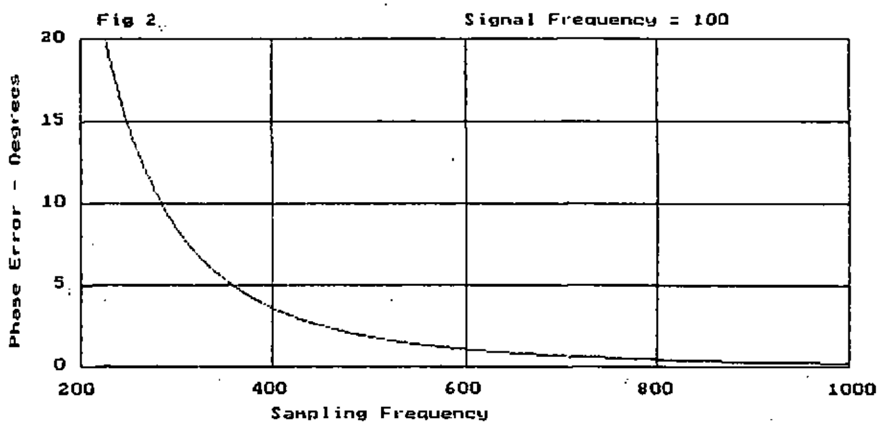


Fig 2 shows the phase errors in phase extraction by time of zero-crossing against signal to sampling frequency ratio.

The Discrete Fourier Transform (DFT) is a derivative of the Fourier Transform, used to process sampled, discrete, signals. The DFT converts data from the time domain to the frequency domain; discrete time intervals being mapped to discrete frequency intervals, or cells.

A sampled sinusoidal signal, of known frequency F and sampling frequency Fs, can be converted to the frequency domain by a DFT acting upon a finite sample set of N values. The resulting frequency domain data, consisting of N complex DFT cells, relates to the frequency content of the time domain data set. DFT cells are complex, and their values can be considered to represent a complex vector whose length is related to the power of the signal frequency and whose direction is related to the phase of the time domain signal. Extraction of signal phase can therefore be achieved by calculating the arguement of the complex vector in the signal's frequency cell.

The DFT method of phase extraction has the advantage of inherently filtering the time domain signal to remove those frequency components which do not lie within the frequency cell of interest. The presence of other, unwanted, frequencies during the calibration procedure is important as these may indicate the presence of undesirable resonances within the array. Additional frequencies can be identified by consideration of the frequency spectrum generated by the DFT procedure. DFT frequency cells are equally distributed in frequency, being centred upon a spot frequency. The frequency coverage of each cell is the sampling frequency, Fs, divided by the time domain data set length, N.

The method of DFT phase extraction has an inherent error caused by the sine x upon x response of frequency domain cells. The complex vector in a frequency domain cell, produced by a single frequency component, F, of the time domain data set, is the sum of two vectors. The primary vector is dominant and represents the true phase of the time domain frequency component. The change of phase in the primary vector, resulting from the addition of the secondary vector, is the DFT method's inherent error.

The two vectors rotate in opposite directions with changing signal frequency, thus varying the influence of the secondary vector upon the phase of the primary vector when the two are added together. Maximum phase error is achieved when the two vectors are perpendicular. Minimum phase error is achieved when the two vectors are parallel.

Fig 3

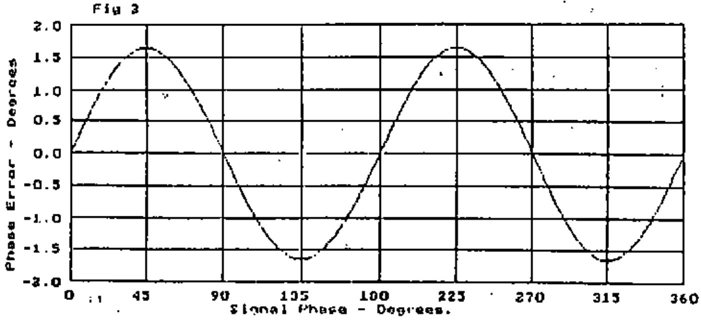
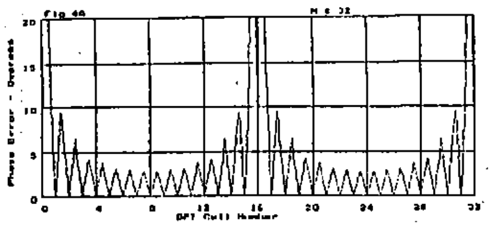
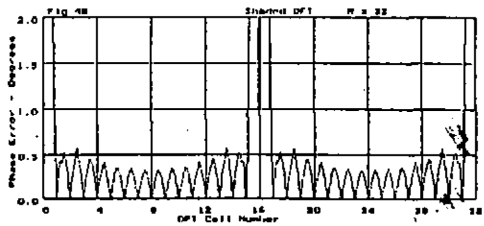


Fig 3 shows the variation in DFT phase error as the sinusoidal signal phase changes relative to the DFT window.

The primary vector is produced by the signal frequency component. The secondary, inherent error, vector is generated by the frequency conjugate. Signals at the centre of a frequency cell will produce conjugate values at the zero crossing points of the cells sine x upon x response, and thus have no intrinsic error. Signals whose frequencies lie halfway between the centre of two cells will have maximum intrinsic phase error as the primary vector magnitude will be a minimum, and the secondary vector magnitude will be a maximum.

The DFT assumes that the transform window contains part of a continuous signal which could be reconstructed by repeating the contents of the window. This assumption will produce a continuous signal with discontinuities every window length, unless the window length is an integer number of signal cycles. The discontinuities occur with a frequency of 1/N, where N is the window length, and appear in the frequency domain as sidelobes, to the major signal frequency, in the adjacent frequency cells. Cell leakage is, therefore, the effect whereby *some of the frequency energy leaks into adjacent cells. At the centre of a frequency cell there is no energy loss from leakage, as the window length is an integer number of wavelengths, but as the frequency moves away from the centre of the cell the energy loss through leakage increases to a maximum halfway between two cells, which gives peak discontinuity of half a wavelength.*





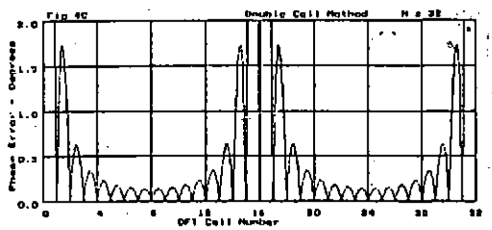


Fig 4A shows how the maximum DFT phase error varies with signal-to-sampling-frequency ratio. Note that the inter-cell peak error is at a minimum in the region of cells N/4 and 3N/4. This is because the frequency conjugate vector is furthest from the frequency component, and therefore smallest in the cells sine x upon x response.

Fig 4B shows the maximum DFT phase error, against frequency cell number, produced by the application of a Hamming window.

Fig 4C shows the maximum DFT phase error, against frequency cell number, produced by the application of the double cell method.

Fig 5

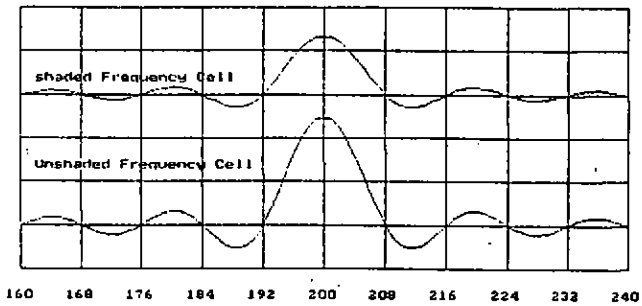


Fig 5 shows the response of a frequency cell shaded by a Hamming window in comparison with the same unshaded cell.

The intrinsic DFT phase extraction error can be reduced by shading the time domain data set to reduce the significance of data towards the edge of the DFT window, see reference 3. Shading reduces cell leakage, but also reduces the energy in the DFT window by changing the sine x upon x response. The changed cell response reduces the side-lobe amplitudes and reduces the peak amplitude of the main lobe, whilst 'squaring' its response.

Shading reduces the amplitude of both frequency vectors, but reduces the error vector more than the primary frequency vector. Shading therefore reduces the intrinsic error in DFT phase extraction.

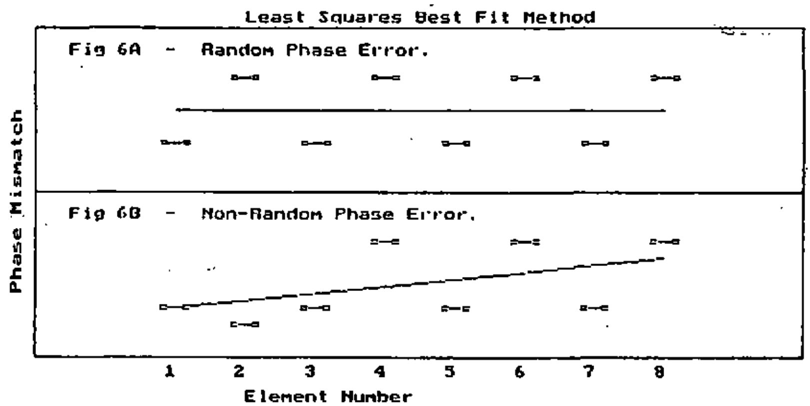
The intrinsic error in DFT phase extraction may be further reduced by utilising the energy which has leaked to the nearest frequency cell. The Double Cell method of DFT phase extraction calculates the signal phase as being the arguement of the difference between the highest amplitude frequency cell and the nearest adjacent frequency cell. The adjacent cell introduces two new vectors into the phase extraction process; the new vectors represent the proportion of the frequency energy whch has leaked into the adjacent cell from the two vectors in the main cell. The primary vector in the main cell is always a fixed distance, in phase, from the primary vector of the adjacent cell. Thus the subtraction of the adjacent cell primary vector from the main cell primary vector gives a vector whose phase is directly related to the phase of the time domain signal in the DFT window. The combined effect of the two secondary vectors is less than the effect of the secondary vector in the single cell phase extraction method. Therefore the intrinsic error in the Double Cell method is substantialy less than in the single cell method.

Having established the relative phase of elements within a sonar array, the phase data must be expressed in a form which gives a measure of the array phase mismatch. The phase mismatch, of a line array, can be expressed against a reference.

One reference which could be used is the ideal phase expected from the calibration signal's angle of interception with the array; however in practice it proves to be very difficult to measure the signal's true angle of interception. If the phase mismatch can be assumed to be randomly distributed along the array's length then a reference least-squares best-fit through the measured channel phase errors can be established. The least-squares method will produce a reference with minimum element phase deviation and is such that any faulty, or dead, channels within an array can be easily ignored.

If array phase mismatch is not randomly distributed along the array's length then the least-squares method will not produce an ideal reference. A lack of randomness in the distribution of array element phase mismatch can usualy be attributed to one or more elements having very bad phase mismatch. The randomness of phase mismatch distribution can be established statistically by considering the distribution of the phase errors about the reference.

A simple measure of the phase mismatch of an array can be established by measuring the standard deviation of each element from the reference.



Figs 6 show the effect upon the least-squares best-fit technique of non-randomly distributed phase error. Fig 6A shows the least-squares best-fit through the randomly distributed elemental phases of an eight channel array. Fig 6B shows the least-squares best-fit through the non-randomly distributed elemental phases of an eight channel array.

PRACTICAL IMPLEMENTATION

The DFT phase extraction method enables the accurate extraction of phase from sinusoidal signals at low sampling rates in relatively noisy environments.

The application of the least squares method enables the identification of channel phase mismatch relative to the array's closest approximation to broadside.

We have used the DFT phase extraction technique, together with least-squares phase matching, in our array calibration system. The system requires two PCs. The first is used to capture the data and to control the signal generation from the reference transmitter. The second, which utilises a transputer running OCCAM code, is used to process the data and provide elemental phase variance together with elemental and steered-beam responses.

The data capture and system control software is similar to many automatic test facilities currently in use, with the exception that the element outputs are simultaneously digitised using a multi-channel transient recorder.

For the 8 element arrays we have been developing the system required about 30 minutes to capture a 180 degree polar run and then about 20 minutes to calculate and plot all the elemental and steered beam responses and phase variance tables.

This approach to array calibration has provided Ferranti with a very fast and accurate method of establishing the performance of new array designs.

REFERENCES

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