FREQUENCY DOMAIN SONAR SIGNAL PROCESSING WITH HIGH SPEED DSP DEVICES

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1. INTRODUCTION

Typical active sonar systems perform their signal processing functions in the time domain. Considerable improvement in both performance and efficiency can be achieved by carrying out these tasks in the frequency domain. This paper demonstrates the advantage of such an approach for a narrowband active sonar. The concepts of active sonar are briefly stated in Section 2, identifying processing tasks that might be more suited to a frequency domain implementation. Section 3 presents the theory behind frequency domain processing, indicating improved solutions to the problems of active sonar. Results of a computer simulation of the system are given in Section 4, illustrating the improvements over an equivalent time domain system. Section 5 describes the implementation of a frequency domain sonar processor using proprietary DSP devices. The paper concludes with observations on the advantages and limitations of the techniques presented.

2. CONCEPTS OF ACTIVE SONAR

Active sonars detect the presence of targets by insonifying sections of the ocean with pulsed acoustic transmissions and extracting target echoes from background noise and reverberation returns at the receiver. A typical active receiver is shown schematically in Fig.1. Sonar returns are initially filtered into a narrow band centred on the transmit frequency. A beamformer then forms typically as many directional beams as there are hydrophones in the array. This yields target bearing information. The output of each beam is then heterodyned to baseband and further filtered. In active FM sonars, the signal is correlated with a prestored replica of the transmitted waveform. The time from transmission to maximum correlation indicates the propagation time, or range, to the target. The signal is then thresholded and passed to the sonar display where it is usually displayed as range vs. bearing information.

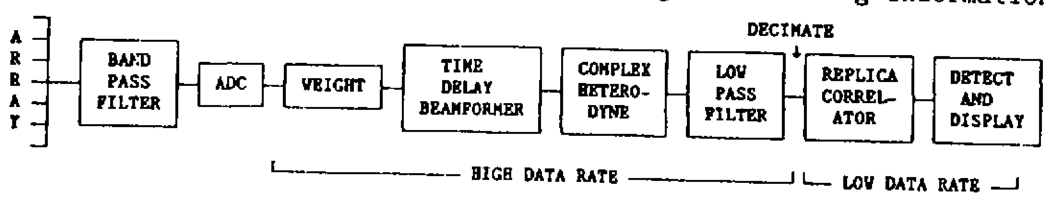

FIG.1 - SCHEMATIC OF TIME DOMAIN ACTIVE RECEIVER

Traditionally, the main processing functions of active sonar are performed in the time domain. Filtering involves the convolution of the incoming signal with the filter coefficients. Conventional beamforming is accomplished by summing the weighted and delayed outputs of successive hydrophones. The complex heterodyne operation is a multiplication of the received time series

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with a complex demodulating sinusoid. Finally, a replica correlator can be envisaged as a matched filter and is usually implemented as a time domain convolution. In addition, various ancillary functions often present in active sonars, such as demultiplexing of encoded signals, Doppler correction, etc, are commonly formulated as unwieldy time domain algorithms. It is shown in the following section that the majority of these processing functions can be performed more efficiently by adopting a frequency domain approach.

3. FREQUENCY DOMAIN SIGNAL PROCESSING

The key to the improved efficiency afforded by frequency domain processing is the Fast Fourier Transform (FFT,[1]). The FFT, and its inverse, the IFFT, enable rapid transfer between the time and frequency domains. This section shows how an FM receiver for an active towed array sonar can be constructed from a number of frequency domain processing modules connected by FFTs and IFFTs. The system is shown in schematic form in Fig.2. Complex heterodyning is combined with a low pass filter and decimation into one operation, performed prior to beamforming. This reduces the required sampling rate, and hence processing load, throughout the system. The signal is passed into the frequency domain via the FFT for the beamforming operation. A novel algorithm, the Arbitrary Frequency Fourier Transform (AFFT) is used to form beams at arbitrary steer angles. The beamformer output is returned to the time domain by the IFFT. The data is regrouped and returned to the frequency domain using a longer FFT, where it is passed through a replica correlator. Correlation in the frequency domain simply involves the multiplication of the signal and replica spectra. The final output to the display processor is a correlator-time series which is obtained by invoking a further IFFT. Each operation is now discussed in greater detail.

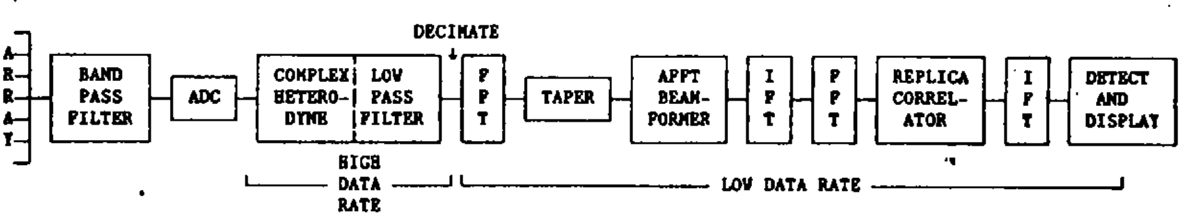


FIG.2 - SCHEMATIC OF FREQUENCY DOMAIN ACTIVE RECEIVER

3.1 Complex Demodulation

Narrowband sonar data of bandwidth B, centred on frequency f_0, enters each channel of the receiver at a data rate f_δ, high enough to adequately sample the highest frequency present in the signal. Typically $f_\delta \approx 4f_0$. A conventional sonar would demodulate the data, after beamforming at the higher data rate, by multiplying it by the complex sinusoid $\{\exp(-i2\pi n\Delta t f_0/f_\delta)\}$. The frequency domain system combines demodulation with a low pass FIR filter, having coefficients $\{c_0,\ldots,c_{N-1}\}$ and decimation by N:1 to a lower sampling rate f_Δ complex samples per second, where $f_\Delta = f_\delta/N > B$. This is done before the beamformer and is achieved using two parallel modified FIR filters per channel with coefficients $\{c_n\cos(2\pi n\Delta t f_0/f_\delta)\}$ and $\{c_n\sin(2\pi n\Delta t f_0/f_\delta)\}$. Only every Nth filter output is retained in the decimation. The resultant output is the complex analytic signal comprising frequencies from baseband to f_Δ with no negative sideband. The FIR filter is implemented in the time domain as this is considered more efficient for high decimation rates. Frequency domain filtering, achieved by multiplying the signal spectrum with the filter transfer function, is preferred where the decimation rate is small. Significant advantage is gained by demodulating prior to beamforming as all subsequent operations are carried out at the lower data rate. This advantage increases with decimation rate or centre frequency to bandwidth ratio.

3.2 Beamforming

The basebanded analytic signal is beamformed in the frequency domain. Prior to the FFT operation the data in the N_E hydrophone channels is buffered and zero-padded into blocks of N_F samples, sufficient to ensure adequate interpolation in the beamformer [2]. This redundancy is removed after beamforming. Once in the frequency domain, any multiplexing in the hydrophone data is removed (see Section 3.4). The demultiplexed hydrophone spectra are then weighted by a taper function, a(n), identical to that used by the time domain beamformer. The $N_F x N_E$ sample data block is beamformed by taking a further FFT across the hydrophone channels for each frequency cell. Thus the spatial domain beamformer output is the two-dimensional FFT of the original hydrophone-time series, x(n,l), given by

$$B(k,m) = \sum_{n=0}^{N_E-1} \sum_{l=0}^{N_F-1} a(n)\; x(n,l)\; W_{N_F}^{(k+k_s)l}\; W_{N_E}^{mn} \quad . \tag{1}$$

where

$$W_{N_E}^{mn} = \exp\,(-i2\pi mn/N_E) \quad \text{and} \quad W_{N_F}^{(k+k_s)l} = \exp\,(-i2\pi(k+k_s)l/N_F)$$

k_s is a frequency (phase) shift which ensures the beamforming is performed on the true frequency content rather than the basebanded signal, thus preserving angular resolution. The shift is given by

$$k_s = \begin{cases} [k_{min}/N_F].N_F, & f>f_0 \\ [k_{min}/N_F].N_F + N_F, & f\leq f_0 \end{cases} \tag{2}$$

where $k_{min} = [f_{min}.N_F.\Delta T]$, and $f_{min} = f_0 - B/2$ is the minimum frequency of interest. $[\cdot]$ denotes taking the integer part. The beamformer output is returned to the time domain via the IFFT and the beamformer-time series is reconstructed using the overlap-add method [3]. The procedure thus described limits the beamformer to forming N_E beams in fixed directions, k, equally spaced in $\sin(\Theta)$, Θ being the angle from broadside [4]. The implementation adopted in this paper uses the AFFT to enable N_B ($<=N_E$) beams to be formed in arbitrary steer directions. Here the spatial frequency k is not constrained to take an integer value. Beamforming is performed in two parallel streams - one representing the integer part of the spatial frequency, the other the noninteger part. The channels are recombined using prestored coefficients peculiar to the particular set of steer angles. The signal is again returned to the time domain using the IFFT. Frequency domain beamforming requires fewer operations than time domain, especially for large numbers of hydrophones and/or beams. Time domain beamforming suffers from interpolation errors unless the waveform is vastly oversampled or a nonlinear interpolation scheme is employed. Either solution is costly in hardware terms. Such an interpolation scheme is not required in the frequency domain [4]. In addition, the frequency domain beamformer can be made less susceptible to hydrophone failures. The spectrum of the missing hydrophone can be readily estimated from a linear combination of the spectra of adjacent hydrophones. This could not be performed efficiently in the time domain.

3.3 Replica Correlation

The beamformed data comprises N_B complex time series sampled at the reduced data rate f_Δ. The data has been returned to the time domain to regroup it for the higher resolution operations in the correlator. The data in each beam is buffered and zero padded into blocks of N_C complex samples prior to an FFT into the frequency domain. N_C must be sufficiently large to adequately cover the bandwidth of the transmit waveform. The redundancy introduced by zero padding is removed after the correlator. Own ship Doppler can be removed from the beamformed data prior to correlation (see Section 3.5).

Target detection is achieved when the correlation between the beamformer output, $b(n\Delta t,\Theta)$, and the replica, $r(n\Delta t)$, exceeds a preset threshold. The discrete correlation function is implemented in the time domain as

$$\rho(\tau) = b(n\Delta t,\Theta) \star r^*(n\Delta t) = \sum_{n=0}^{N_C-1} b(n\Delta t+\tau,\Theta)\ .\ r^*(n\Delta t) \tag{3}$$

The value of τ when $\rho(\tau)$ exceeds the detection threshold gives the target range $R = c\tau/2$. Eq(3) can be reformulated in the frequency domain as a block multiplication,

$$DFT\{\rho(\tau)\} = = B(k,m)\ .\ R^*(k),\ k=0,\ldots,N_C-1. \tag{4}$$

Once this has been performed the data is returned to the time domain by an IFFT. The correlator-time series in each beam is reconstructed using the overlap-save technique [2] to remove the earlier redundancy. The correlator output is then passed on to detection and display processing.
The advantages of frequency domain correlation are a reduction in computational load and arbitrarily good frequency resolution, determined by the size of the FFT rather than the length of the transmitted waveform.

3.4 Demultiplexing
The data from individual hydrophones may contain timing errors introduced by the process of sampling and multiplexing in the array. This effect can easily be removed by applying a frequency dependent phase shift to each of the hydrophone spectra. An equivalent time domain system would demultiplex the signals within the beamformer, incorporating the time delay correction into the already inaccurate interpolation scheme. Errors in the demultiplexing scheme manifest themselves as spurious sidelobes in the beam pattern.

3.5 Own Doppler Nullification
The system described above is designed to process signals from a towed array, the performance of which is affected by the forward motion of the towing vessel. This will introduce a Doppler shift in the transmitted pulse which manifests itself as a mismatch between the received signal and replica spectra, and hence a loss in the gain of the system. This problem is trivially resolved in the frequency domain implementation. When a received signal, containing no Doppler shift, is basebanded and the analytic signal is formed, frequencies from f_0-B/2 to f_0 are mapped onto the range B/2 to B, while frequencies from f_0 to f_0+B/2 occupy the band from dc to B/2. Doppler shift further rotates the spectrum by an amount

$$f_D = 2f_0 \, (v/c) \cos \theta_m \, , \tag{5}$$

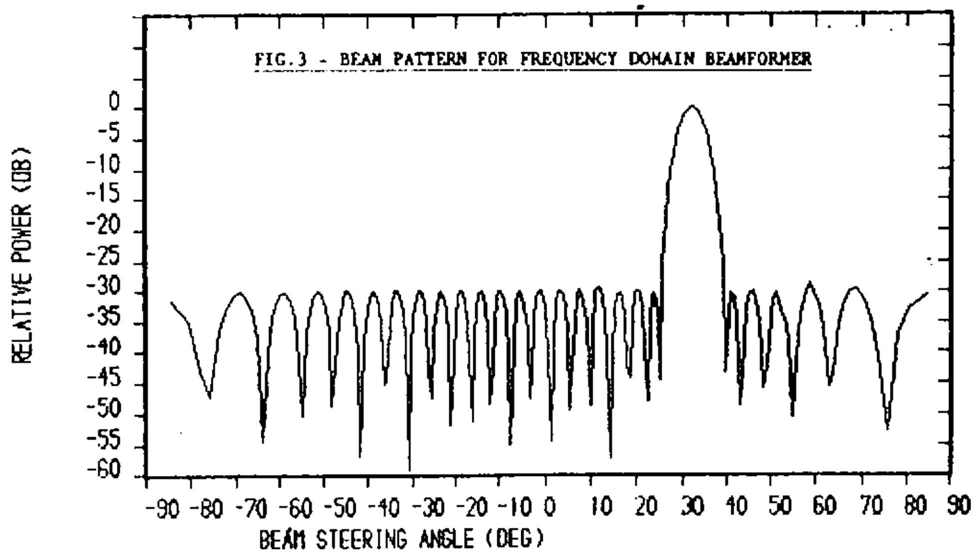
where v is the speed of the towing vessel, c is the speed of sound in water and θ_m is the beam steer angle relative to the direction of the ship's motion. In the frequency domain correlator, the data are available as beam spectra. The spectra are rotated firstly by B/2 to centre the data on f_0 and secondly by $-f_D$ to remove any Doppler shift. Correlation then proceeds as outlined in Section 3.3. A time domain system would compensate for Doppler either by extending the receiver bandwidth or the transmitted pulse length.

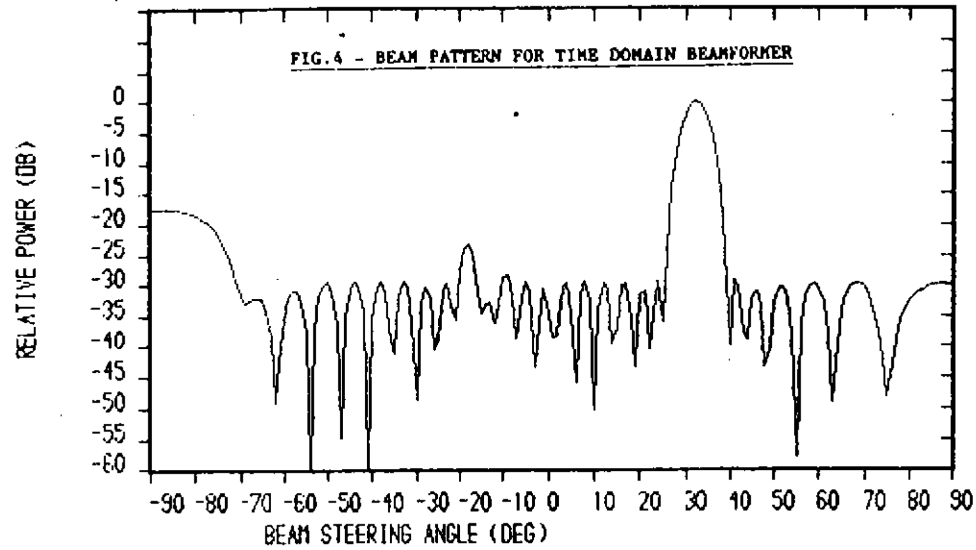
4. COMPUTER SIMULATION

The system described in the preceding sections has been modelled on a VAX computer. Performance predictions have been obtained and comparison made with existing time domain models. During the implementation phase the models have been used to refine and optimise the algorithms for real-time application.

The processor model comprises all the modules outlined in Section 3. The data model consists of an LFM signal incident at arbitrary angle to the array and buried in Gaussian noise. The SNR is variable. The model is able to simulate multiplexed data, missing hydrophones and Doppler shift. Results indicate that the beamformer performs well even in the presence of dead sensors and at low SNR. The replica correlator is shown to give good performance in the presence of Doppler. Computer simulation suggests frequency domain processing as a robust and accurate alternative to conventional techniques.

As an illustration, Fig.3 shows the beam pattern response for a 32-element line array with -30dB Chebychev shading, steered at 32° from broadside. The incoming signal is a 2s/200Hz LFM chirp centred on 3kHz with a SNR of 60dB. The signal is sampled at 12kHz and beamformed at 240Hz. The frequency domain beamformer clearly achieves the theoretical -30dB sidelobes. Fig.4 shows the response of an equivalent time domain system beamforming at 12kHz and employing a linear interpolation scheme. Spurious lobes can be seen at a level of -23dB, some 7dB worse than the frequency domain beamformer.





5. HARDWARE IMPLEMENTATION

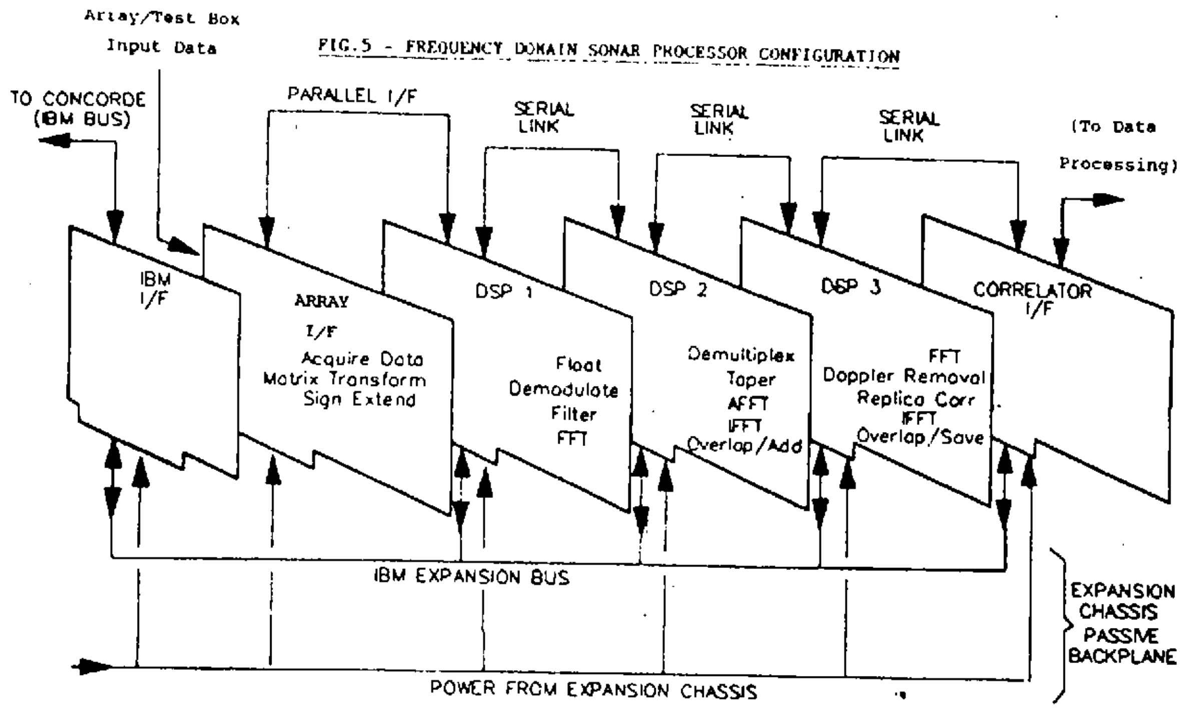
The practical realisation of a frequency domain processor has been made possible by recent advances in high speed Digital Signal Processors (DSPs). This Section describes the development of a frequency domain demonstrator by staff from Underwater Engineering at BAe. The demonstrator has been designed to interface with an existing experimental array and display processing. This is to enable an assessment of the system using real sonar data recorded on recent sea trials.

The system has been implemented using a total of three AT&T, DSP32 Vector processors (one DSP32 and two of the faster, greater memory device, the DSP32C) hosted by an IBM PC compatible. As a benchmark, the DSP32 performs a 1024-point complex FFT in 8.9ms. The software is written in a combination of C and Assembler, and wherever possible, DSP functions are performed by calls to the processor's optimised DSP library. In addition, two custom-built interface cards are required. The configuration is shown in Fig.5. The transmitted FM pulse is either 1s/100Hz (nominal) or 2s/200Hz (extended) centred on 3kHz. Received data enters DSP1 from 32 hydrophones at a rate of 12kHz. DSP1 houses the heterodyne scheme, comprising two parallel, 50 tap, modified FIR filters per channel taking the data from baseband to 200Hz, and decimation by 50:1 to give a revised sampling rate of 240 complex samples per second. Also on DSP1 are data buffering to form blocks of 50 samples plus 14 zeros, and a 64-point complex FFT to take each hydrophone channel into the frequency domain. Communications with DSP2 are via a serial DMA link.

The frequency domain beamformer resides on DSP2. A DSP32C card is necessary here to meet the storage requirements of the AFFT (8192 coefficients). The spectra are weighted by a -30dB Chebychev taper. Demultiplexing using a phase shift on each channel is the next operation on DSP2. Estimation of dead element spectra is not implemented in the demonstrator. Each hydrophone spectrum is then split into two streams, one processing the integer spatial frequency, the other the noninteger part. The integer part undergoes a 32-point Hilbert Transform and a 32-point FFT, whilst the noninteger stream goes through a 32-point FFT and a 6-tap 'approximation correction' filter. The two data streams are recombined using the prestored coefficients and 28 beam spectra are selected for further processing. This represents the phase shift beamformer of Section 3.2 optimised for real-time implementation on a DSP. The final operation on DSP2 is to reconstruct the beamformer-time series using a 64-point IFFT and the overlap-add technique. The resultant beams are around 4° wide at broadside with sidelobes at -30dB. The energy in each beam is available as a histogram display or beam-time output at the PC monitor.

DSP3 is also a DSP32C, the choice dictated by the transform lengths within the correlator.

FIG.5 - FREQUENCY DOMAIN SONAR PROCESSOR CONFIGURATION



Data are prepared for the correlator by buffering into blocks of 1024 complex samples, overlapping in time by 480 samples. Doppler is removed by rotating the spectrum of a beam steered at angle θ to broadside by approximately $|9\sin\theta|$ frequency cells for each knot of towship speed. The complex replica is derived from the extended transmit pulse, suitably demodulated and decimated to a length of 480 samples. The replica is padded by 544 zeros and its spectrum obtained via a 1024-point FFT. The correlator output spectra are obtained by block multiplying the rotated beam spectra with the replica spectrum. A 1024-point IFFT returns the data to the time domain, where the correlator-time series are reconstructed using the overlap-save method.

The correlator output rate is 240Hz giving a range resolution of 3.1m. A further decimation by 2:1 (resolution 6.25m) is required to make this compatible with the existing data processing. Output is in the form of an RvB or PPI Display.

6. CONCLUSIONS

In this paper, frequency domain signal processing is presented as a viable alternative to traditional active sonar techniques. Processing modules commonly found in narrowband active receivers are described, and, for each module a frequency domain replacement is derived. In most cases, analysis and simulation show the frequency domain approach to offer a more efficient implementation, better performance or both.

The advantages offered by frequency domain processing are summarised here. A lower sampling rate is required to adequately sample the basebanded signal, and hence the processing load of the system is reduced. Beamforming in the frequency domain eliminates interpolation sidelobes and use of the AFFT allows beams to be formed in arbitrary steer directions. The algorithms are robust and the use of common processing modules allows easy expansion of the system. Frequency domain processing becomes more attractive the higher the centre frequency to bandwidth ratio or the greater the number of hydrophones in the array. It should be noted that the algorithms presented here are valid for linear array configurations operating in narrowband only.

The final section of the paper indicates how such a system has been realised using proprietary DSP devices. The system described processes sonar data in real time and has been achieved at a fraction of the hardware cost of an equivalent time domain system. Use of DSP library routines has reduced software development time considerably. The processor has been specifically developed for testing with real sonar data using an interface to BAe experimental equipment. Results of these tests will be presented at a later date.

7. ACKNOWLEDGEMENTS

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8. REFERENCES

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