ARCHIVES OF ACOUSTICS
13, 1-2, 67-88 (1988)

REAL TIME SIGNAL PROCESSING TECHNIQUES IN A DUAL BEAM SONAR SYSTEM FOR FISH STOCK ASSESSMENT

R. S. MITCHELL, A. STEPNOWSKI*, J. DROBNIK, D. R. GREEN

Seastar Instruments Ltd. Argo Bldg. Bedford Institute of Oceanography, Dartmouth, Nova Scotia, Canada B2Y 3Y9

*Technical University of Gdańsk, Institute of Telecommunications, 80-952 Gdańsk, Poland

Advances in VLSI technology including A/D converters, large high speed memories and digital signal processing devices, combined with a dramatic improvement in high performance single board computer SBCs, have radically altered design possibilities for sonar systems [1, 2]. It is now possible to centralize the system functions in one controller and transfer much of the signal processing from hardware to software. Furthermore, digital signal processing techniques may be employed to improve the system's performance and reliability. In this paper, we describe a new stock assessment dual beam sonar designed to take advantage of modern computer architectures and digital signal processing techniques. We also discuss some original processing enhancements to the basic dual beam concept both in hardware and software which have been built into the sonar system. The system is flexible so that the same hardware is employed for various modes. The transfer of functions from hardware to software combined with a hihgly efficient transmitter has yielded a low cost compact system.

Rozwój technologii VLSI bardzo dużej skali integracji, a w szczególności przetworników A/D, bardzo dużych i szybkich pamięci oraz układów do przetwarzania sygnałów w połączeniu z dramatycznym rozwojem mikrokomputerów, w radykalny sposób zmienił możliwości projektowania systemów sonarów [1, 2]. Obecnie stało się możliwym zgrupowanie większości funkcji w systemie jednego kontrolera oraz szerszego programowego przetwarzania sygnałów w miejsce przetwarzania sprzętowego. Należy dodać, że zastosowanie cyfrowego przetwarzania sygnałów poprawia jakość i niezawodność systemu. W artykule podana została ocena dwuwiązkowego sonaru zaprojektowanego pod kątem korzyści jakie oferuje nowoczesna architektura komputerowa i technika cyfrowego przetwarzania sygnałów. Przedyskutowane zostaną pewne nowe oryginalne modyfikacje, tak programowe jak i sprzętowe, przetwarzania sygnałów w porównaniu z klasycznym podejściem do dwuwiązkowego systemu sonaru. Opracowany system jest elastyczny tak, że opracowany hardware może mieć wiele zastosowań. Przeniesienie funkcji operacyjnych z rozwiązań sprzętowych do programowych (software) w połączeniu z bardzo skutecznym nadajnikiem pozwoliło opracować system tani i zwarty (compact).

1. Introduction

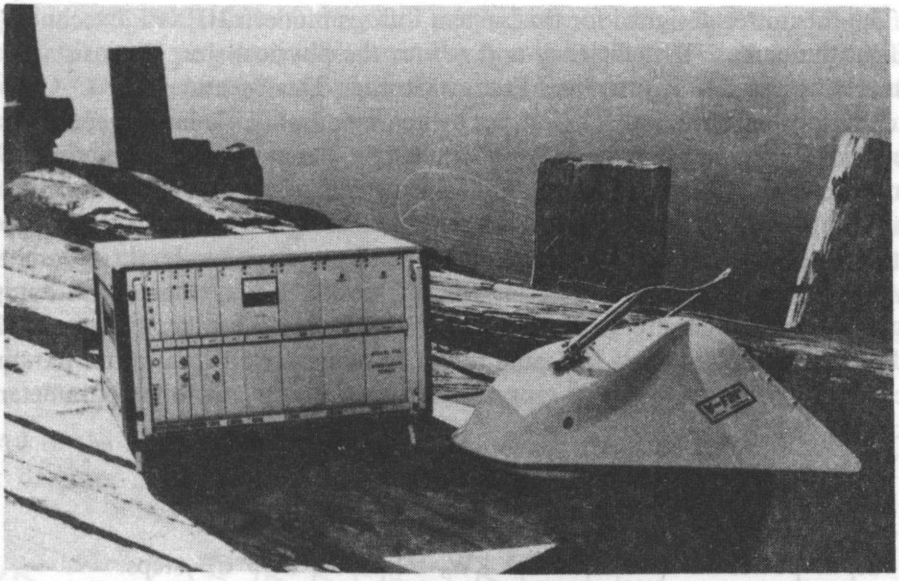
The utilization of dual beam configuration in hydroacoustic systems has proven to be an accurate and reliable means for target strength estimation from received echos. Conventional sonar systems for quantitative fish stock assessment either have limited real time capabilities or require dedicated extensive separate hardware to perform echo counting, target strength estimation and/or echo integration [3, 4].

Over the last few years, there have been major improvements in VLSI technology resulting in the availability of low cost digital signal processing devices and high performance microcomputers. The benefits to the system design engineer have been profound. Large reductions in hardware real estate and enhancements in system performance and flexibility have resulted. The availability of high performance 16 bit microprocessors has changed the design philosophy of microprocessor systems from that of a few dedicated task oriented machines arranged in a sequential or pipeline fashion to that of a high speed parallel bus with one or more high performance machines. Complementing the improvements in machine architecture have been advances in software, namely real time executives enabling a single processor to perform many different tasks simultaneously. Furthermore, international standards for the multiprocessor bus systems have resulted in the availability of numerous inexpensive single board computers.

This paper describes the application of these advances in hardware and software to the design of dual-beam sonar system for fish stock assessment. A high speed A/D converter in conjunction with digital signal processing devices results in relaxes the specifications of the analog hardware, allows the use of a digital Finite Impulse Response (FIR) filter, and transfers the Time Variable Gain (TVG) function provision requirements to a digital processor which results in improvement of the receiver performance and in the overall system flexibility. Emphasis on software design has yielded a flexible system that is compact, low cost and user-friendly, and which combines all basic functions required for acoustic surveying of fish populations, namely: echo-sounding (fish-echo detection), echo-integration, echo-counting and target strength estimation.

2. System architecture

A photograph and block diagram of the dual beam sonar system is shown in Figure 1 and 2 respectively. The central single board computer (SBC), based on the 68010 microprocessor is the heart of the system. Its responsibilities include transmitter and receiver control as well as data collection and real time analysis. A host personal computer, IBM PC/XT or compatible is also connected to the 16 bit Versatile Modular European (VME) based system. The PC provides a user-friendly, menu driven interface between the operator and the central computer. The PC may



Sonar unit Towed Body with transducer

FIG. 1. Dual beam sonar system (Host Computer Workstation not shown)

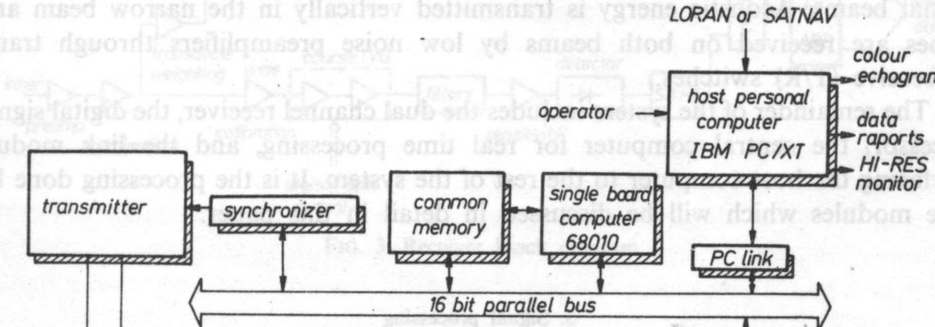


FIG. 2. Dual beam sonar system block diagram

take control of the system for testing and diagnostics. Various peripherals may be connected to the PC for storage of raw or processed data, graphics presentation, user interface and navigation equipment interfacing. A typical configuration consists of a 20 Mbyte hard disk, a high resolution colour monitor, a colour chart recorder and a Loran C serial interface.

The trasmitter designed for this system utilizes modern HEXFET technology to significantly increase the efficiency and reduce the physical size. It consists of three sub-modules: AC/DC Converter, Energy Storage Device and DC/AC Converter (switching power amplifier). The output switching amplifier is implemented in a full wave bridge configuration, and is controlled by the synchronizer module. When commanded by the central computer, the synchronizer will generate necessary control (appropriate transmit gates) to the output bridge network, so the high power burst will be routed to the transducer load. The output power level is regulated by the AC/DC converter the output of which is controlled by the central computer using an eight bit digital to analog converter (DAC). The central computer also loads the desired carrier frequency and ping rate. Consequently all transmitter parameters are computer programable. The ranges of these computer controlled parameters are as follows:

ping rate	$f_p = 0.1–10$ Hz,
pulse length	$\tau = 0.1–10$ ms,
carrier frequency	$f = 30–450$ kHz,
output power	$P = 0.075–2.4$ kW, selectable in 3 dB steps.

The transducer, which is physically mounted inside the towed body, consists of two sections arranged in concentric configuration, to form the narrow and wide coaxial beams. Acoustic energy is transmitted vertically in the narrow beam and echoes are received on both beams by low noise preamplifiers through transmit/receive (T/R) switches.

The remainder of the system includes the dual channel receiver, the digital signal processor, the central computer for real time processing, and the link module interfacing the host computer to the rest of the system. It is the processing done by these modules which will be discussed in detail in this paper.

3. Signal processing

Processing done on the received echo signals can be divided into two categories. The first category is referred to as signal processing where the same algorithm is performed independent of the signal characteristics. The second class is referred to as data processing in which processing is done conditionally on the value of the digital samples. In general, signal processing is done by dedicated hardware and data processing is done in software by either the central computer or the host computer.

The echoes received by the sonar transducer are subjected to two stages of signal processing. The first stage is the dual-channel receiver analog preprocessing (signal conditioning) which performs equalization of sensitivities in both channels, transducer aperture weighting, course TVG, filtering, demodulation, sampling and analog to digital conversion (ADC). In the second stage, the multiplexed digital receiver outputs are applied to a hardware digital signal processor (DSP) which performs the digital TVG correction, digital filtering in FIR filter, sampling rate decimation, amplitude thresholding and preliminary bottom, detection. The output of the digital signal processor is buffered in a first-in first-out (FIFO) queue, which in turn is served by the central computer where real time data processing of the echoes is performed.

The central computer can operate in a number of modes to implement echo counting and integration algorithms with or without target strength estimation. The computer also has direct control of the receiver and signal processor to set up the required configuration for the each mode of operation. The output of the central computer is sent to the host PC through a common large memory also on the system bus.

3.1. *Analog signal processing*

All of the analog signal processing is done on the dual channel receiver, the block diagram of which is shown in Figure 3.

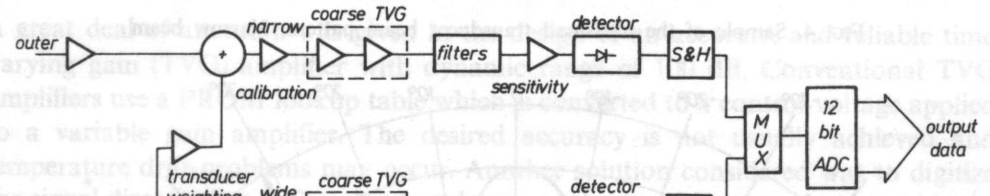


FIG. 3. Receiver block diagram

3.1.1. Transducer Weighting. The first stage of the dual channel receiver buffers the preamplifiers outputs for the inner and outer transducer sections. After channel sensitivity equalization is done, transducer aperture shading is performed by summing the weighted combination of the individual channels. For the low frequency (e.g. 50 kHz) system version, both sections are formed by an array of individual pre-stressed sandwich resonators, arranged in five concentric rings. For the high frequency (e.g. 120 kHz) version, a single inner disc and outer ring elements constitute the transducer's aperture. In the present system design only two-step shading was applied, to reduce the side lobe level in receive narrow beam, by appropriate sensitivity weighting in the variable gain buffer amplifiers of the receiver. However, optional weighting capabilities are provided for transmit beam pattern and also multi-step weightings may be applied for both transmit and receive beams.

The measured narrow beam and wide beam patterns for 50 kHz transducer are shown in Figure 4 and figure 5, respectively. The unweighted beam patterns have

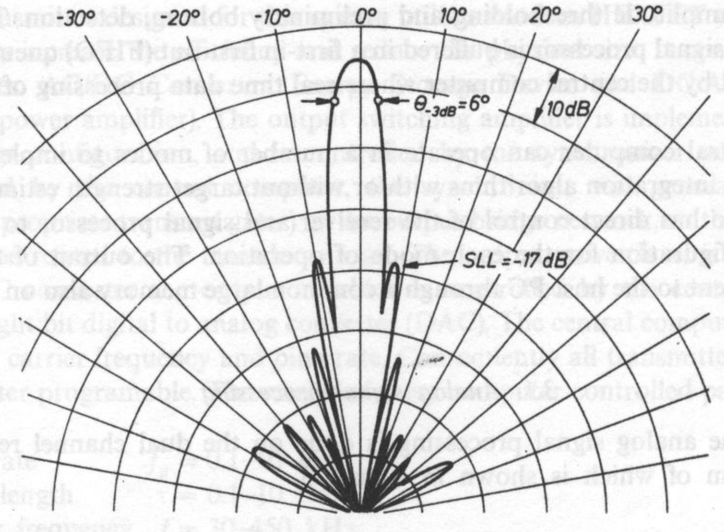


FIG. 4. Sample of the measured transducer beam pattern. Narrow beam

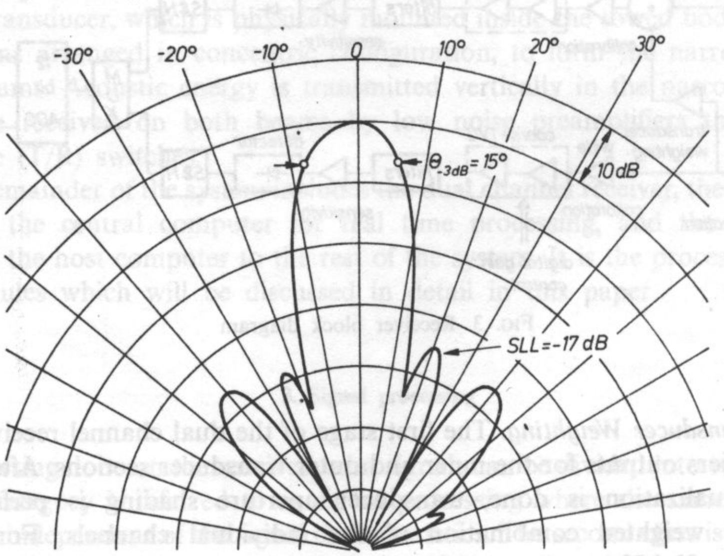


FIG. 5. Sample of the measured transducer beam pattern. Wide beam

side-lobes at approximately 17 dB. The beamwidth of the wide beam is approximately 15 degrees, while of the narrow beam is about 6 degrees both in transmit and receive modes. The simple binomial weighting applied for the receive narrow beam reduces the side-lobe level to about 20 dB.

3.1.2. Time variable gain. **The received echoes must be progressively amplified to compensate for the geometric spreading and attenuation of the acoustic wave as it**

travels to the target and back. For single fish targets, when TS estimation or fish counting is applied, the targets are considered as isotropic scatterers, and the range (R) dependent gain required to correct for the two way transmission loss is [5]:

$$G(R) = 40 \log \frac{R}{R_{\text{ref}}} + 2\alpha (R - R_{\text{ref}}) \quad [\text{dB}], \tag{1}$$

where R_{ref} reference range equals TVG start range, α attenuation coefficient.

For dense concentrations of fish (schools and scattering layers) when echo-integration is applied, the scattering from the plane is considered, which implies the one way compensation of spreading loss with range (i.e. $1/R^2$ low) and consequently the required time varied gain function has the form:

$$G(R) = 20 \log \frac{R}{R_{\text{ref}}} + 2\alpha (R - R_{\text{ref}}) \quad [\text{dB}]. \tag{2}$$

A great deal of attention was given in the design of an accurate, and reliable time varying gain (TVG) amplifier with dynamic range of 100 dB. Conventional TVG amplifiers use a PROM lookup table which is converted to a control voltage applied to a variable gain amplifier. The desired accuracy is not usually achieved and temperature drift problems may occur. Another solution considered was to digitize the signal directly, but A/D technology has not progressed far enough to provide the necessary dynamic range at a reasonable cost.

The time varying gain method in this system is a hybrid solution which combines the course TVG implemented using step gain amplifiers, and the digital TVG correction realized by high speed digital multipliers in DSP. The analog step amplifiers provide fixed gain steps of 10 dB, from 0 to 100 dB. The computer loads in a RAM a function which changes the analog gain in steps of 10 dB as a function of range. The coarseness is therefore at most 5 dB at a given range. The gain is increased by changing the resistance in the feedback network of an inverting amplifier. Two amplifier stages with dynamic range of 50 dB each are cascaded. The result is a precise, stable TVG amplifier with dynamic range of 100 dB.

3.1.3. Filtering and demodulation. The output of the TVG amplifiers is filtered using conventional multiple feedback bandpass filters and applied to a wide dynamic range envelope detector. The bandwidth of the filters is fixed at the bandwidth suited to the minimum pulse length of the transmitted signal (0.1 ms for 120 kHz, 0.3 ms for 50 kHz).

3.1.4. Digitization. The echoes from each beam are simultaneously sampled and multiplexed through a common A/D converter. The sampling rate was fixed at 80 kHz (40 kHz for each channel). The maximum bandwidth of the receiver is 10 kHz. The 12 bit converter is applied with least significant bit of 2.5 mV.

3.2 Digital signal processing

After the signals have been digitized, they are multiplexed through a common digital signal processor. The 12 bits of data for each channel corresponds to a maximum dynamic range of 74 dB. A block diagram of the digital signal processor is shown in Fig. 6. The first two sections of the processor are essentially part of the receiver. The filtered data is thresholded to remove noise and the bottom echo is detected. The data is buffered and stored in a FIFO buffer.

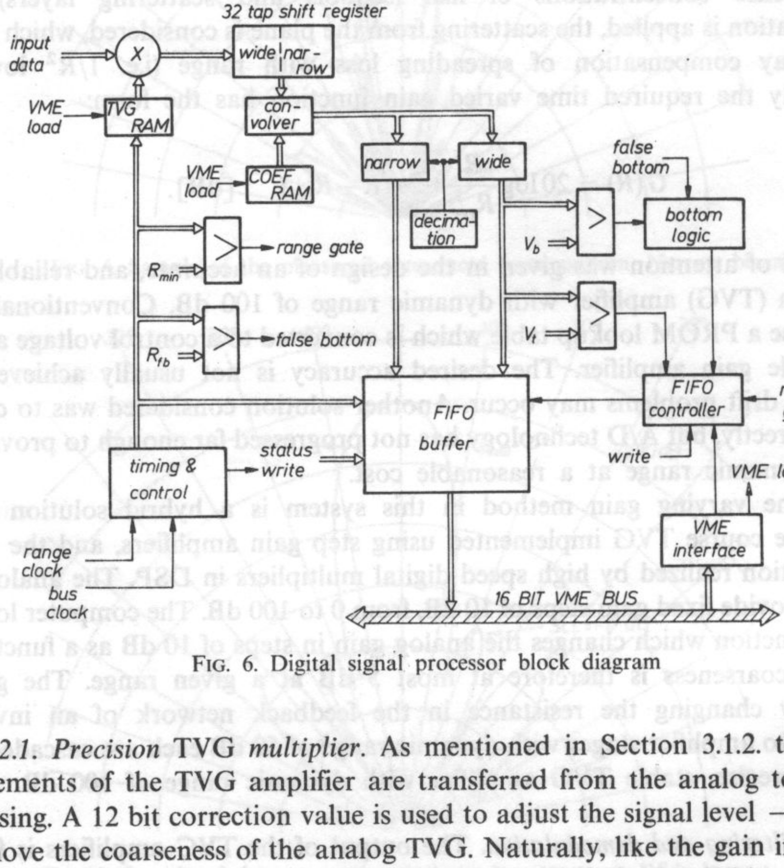


FIG. 6. Digital signal processor block diagram

3.2.1. *Precision* TVG *multiplier*. As mentioned in Section 3.1.2. the precision requirements of the TVG amplifier are transferred from the analog to the digital processing. A 12 bit correction value is used to adjust the signal level -5 to $+5$ dB to remove the coarseness of the analog TVG. Naturally, like the gain in the analog amplifiers, the correction value is time dependent and is also loaded in a RAM from the central computer.

The above scheme for TVG is very flexible: the same analog and digital hardware can be used for any spreading model, attenuation coefficient and reference range and consequently, arbitrary TVG functions which cover two decades of range are possible. The central computer simply loads the desired gain function and correction in the receiver and digital signal processor during system initialization. The attenuation constant α, and the TVG start and stop range are also programmable by the user.

3.2.2. Digital FIR *filter*. To optimize the echo detection (maximize S/N ratio) and simultaneously provide the best available resolution, the sonar system should adjust the receiver bandwidth to the different sounding pulse lengths in use. Typically, the pulse length ranges from a hundred microseconds to a tens of milliseconds, and consequently the receiver bandwidth, which should be approximately the inverse of pulse length, must vary also in the range of 1.5 to 2 decades. Conventional systems fix the number of pulse lengths and select either different filters or change the passive components of the filters. In either case, the variable pulse lengths result in redundant hardware.

Variable bandwidth is easily accomplished by a digital type filter. In the system, the bandwidth of the analog portion of the receiver is fixed at 10 kHz, which matches the minimum pulse length of 100 μs. The signals are digitized at a rate of 40 kHz so that a digital finite impulse response (FIR) filter can be used to decrease the bandwidth for longer pulse lengths. After TVG correction, the samples for each channel are stored in shift registers which are multiplexed through a common digital FIR filter. The order of the digital filter is 32 taps. For shorter pulse lengths, less than 32 samples are available, but the filter requirements are not as stringent.

The central computer sets the pulse length of the transmitted signal and loads the filter coefficients that match the pulse length into a RAM during system initialization. The FIR architecture was chosen for two reasons. Firstly, the number of samples in the received envelope is fixed and secondly the same digital multiplier used for TVG correction is used for the convolution of the filter impulse response and the delayed data.

The filter output is applied to time decimation logic which will select 1 of N samples, effectively reducing the sampling rate. The value of N is set by the central computer and can range from 1 to 16. The sampling rate can range from 2.5 to 40 kHz. Depending on the system mode of operation, the sampling rate is changed.

3.2.3. Hardware threshold detection. The filtered output for each channel is buffered and applied to a digital comparator which rejects all samples below the signal to noise (S/N) threshold, V_t. The S/N threshold is 12 bits wide and set by the central computer. The level of the threshold used for signal detection is different for each mode of operation. The central computer loads V_t in the digital signal processor and all samples for each channel are compared with this threshold. The comparator is disabled after transmission until a desired blanking range has been reached. The minimum range value is set by the central computer.

3.2.4. Bottom processing. Parallel to the S/N threshold hardware is preliminary bottom processing. The samples from the wide beam channel are thresholded with another threshold V_B to detect the echo from the bottom. The bottom threshold is loaded by the central computer. The computer can disable the bottom processor until a certain range, referred to as start of the bottom window, has been reached. This is necessary so that a large target at close range is not erroneously classified as the bottom return. If a bottom echo has not been detected within the bottom window, the digital signal processor will generate an artificial false bottom echo which is written into the FIFO buffer. Once a bottom has been detected, the digital signal processor is disabled until the next transmission.

The hardware bottom processing serves only as preliminary bottom detection. One echo is tagged as the bottom and is used by the central computer for further bottom processing as well as optional bottom lock processing for echo integration.

3.2.5. FIFO *data buffering.* All samples exceeding the S/N threshold which are received after the blanking period and before the bottom echo are stored as an alarm block in a FIFO RAM buffer. The voltage amplitudes for each channel as well as the range are written in memory. An additional status byte is written for each alarm.

Buffer status indicators such as empty, half full and full are accessible by the central computer so data collection routines may be implemented. The buffer can contain 256 echo samples greater than the S/N threshold.

If the data rate begins to increase beyond an acceptable rate, the central computer can invoke a number of safeguards so that the buffer does not fill and data is lost. These safeguards include: time decimation, increasing the threshold, filling a temporary buffer and decreasing the ping rate.

4. Real time data processing

The central computer initiates the transmission and the first processing interval after transmission is reserved for direct memory access (DMA) data transfer of the central computer data to the host computer. The software range window from the minimum range to the bottom echo is used for data collection and real time processing. After the bottom echo has been detected, the real time processing is completed for the current ping and the output data is formatted and stored in the common memory for subsequent processing by the host computer.

The system has the following modes of operation:

i) Target strength estimation with optional fish sizing/echo counting.

ii) Echo integration.

iii) Echo integration with TS estimation.

Each of the two computer's share a $512k \times 8$ common memory which is mapped on the system bus.

All of the programs are written using the C programming Language [6].

4.1 Data collection and common memory

After transmission of the sounding pulse, the central computer polls the buffer status on the digital signal processor for non-empty condition. If there is data available, the FIFO is emptied into a large data buffer in the common memory. It is this larger buffer, called the raw data queue, that is used by the central computer for real time processing.

The common memory is used for storing data structures and variables, system parameters and lookup tables, and mailbox communication between two computers.

4.2. *Target strength estimation*

Routines are executed in real time by the central computer, (68010), which calculates the target strength (TS) and back scattering cross section (σ_{bs}) statistics from the individual fish echo narrow and wide beam amplitudes contained in the raw data queue. The theoretical derivation of target strength and back scattering cross section algorithms from the dual beam system are included in Appendix I. The TS and σ_{bs} data are calculated for each accepted single fish echo using formulae (I.9) and (I.11) and various data outputs are generated and displayed.

4.2.1. Echo detection and classification. The hardware amplitude S/N threshold (sec 3.2.3.) and software time windowing are used as a decision criteria for detection and parameter estimation of the echo pulses. Discrimination of the echoes as those from single or multiple fish targets is done. For each detected echo, an output echo data block is generated and stored in common memory and is available for further processing and/or transfer to the host computer.

A. Extraction of echo pulses – peak amplitude determination. The raw data queue is comprised of alarm blocks which are linked together. Sequential alarm blocks are searched for either a gap in range or a hardware tagged bottom alarm block. A gap is detected when successive alarms have ranges that differ by more than 1 sampling interval. As the raw data queue is being searched for end of the current echo pulse, the peak amplitudes and the pulse length for each beam is determined. Simultaneously, the pulse lengths are compared with a maximum pulse length (determined from the sounding pulse) for multiple fish target detection. If the pulse length is less than the maximum length, the amplitudes are stored in the echo pulse record and the echo pulse is initially classified as a single fish target. Once a gap is detected, the echo pulse record is complete and contains:

1) array of narrow-beam amplitudes (limited to maximum length),
2) array of wide-beam amplitudes (limited to maximum length),
3) narrow-beam peak amplitude,
4) wide-beam peak amplitude,
5) pulse length for each beam,
6) range for maximum peak,
7) single/multiple flag.

The echo pulse record is ready for further analysis for valid single fish detection. The above algorithm is illustrated by the flowchart in Figure 7.

B. Time windowing – pulse width determination. Once the echo pulse has been extracted and formatted in an echo pulse record with single target status, it is subjected to final pulse width check for classification as single, multiple or noise. The half-peak amplitude (−6 dB) pulse width criterion is illustrated by the flowchart in

Fig. 8. The peak amplitudes in the echo pulse record are halved and the amplitude arrays are searched to determine the number of contiguous samples for which the half-peak amplitude levels are exceeded. The resultant pulse lengths are then compared to a preset time window, adjustable automatically for varying pulse lengths. The time window resolution is the period of the carrier frequency, (i.e. 20 μs for 50 kHz, 8 μs for 120 kHz) and there is no ambiguity due to the synchronism of the transmit gate with the carrier frequency. The adjustable number of periods and sampling rate results in increased system flexibility and reliability of the pulse width classification criterion as the time window adapts optimally to the different operating conditions (e.g. S/N ratio).

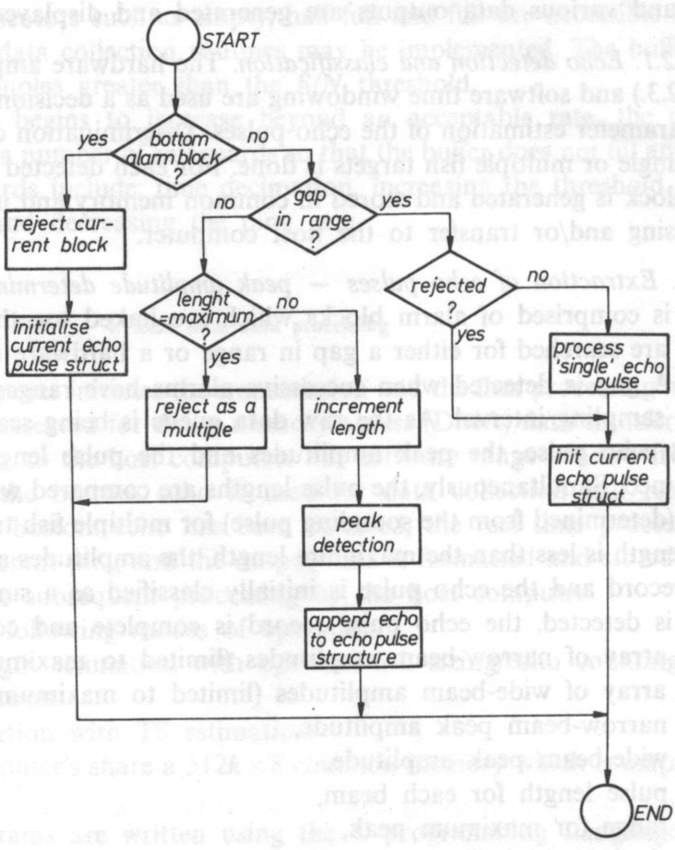


FIG. 7. Process alarm block in Raw Data Queue

Finally, if the half-peak amplitude pulse lengths are within the allowed time window, the echo pulse record is classified as a single fish target. If either the pulse length is greater than the maximum pulse length or less than the minimum pulse length, the echo is rejected as multiple or noise respectively.

C. Amplitude rato determination — beam pattern threshold. Those echo pulse records which are classified as single fish targets are subjected to beam pattern threshold criteria. The ratio of the squared peak-amplitude voltages (I.3) is compared with a computer-set beam pattern threshold T_B (I.4). Those echoes for which the squared voltage ratio does not exceed -3 dB are rejected. In addition, any

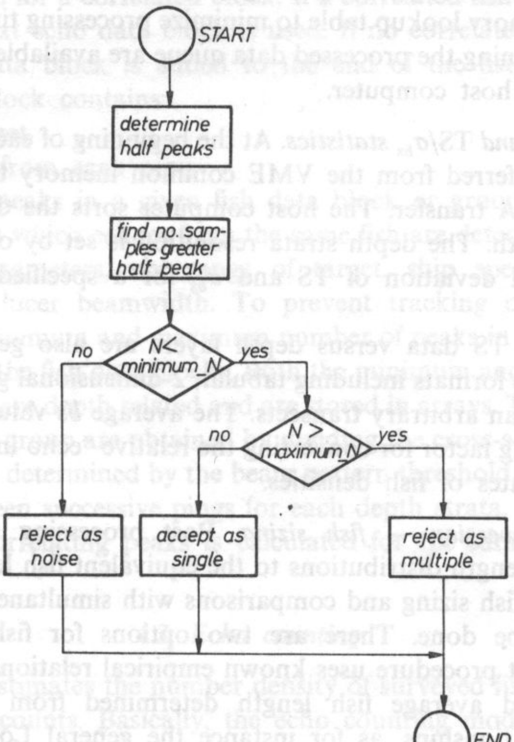


FIG. 8. Time windowing flowchart

"suspicious" echoes ($V_n > V_w$) are also rejected. As a result, the sampled volume becomes better defined and confined to the central portion of the beams. This assures the validity of the dual beam theory and permits the removal of the bias in final target strength estimates [7].

Those echo pulse records which pass the beam pattern threshold are finally considered as single fish targets, and due to their redundancy are reduced to an echo data block which contains:

1) ping number,
2) range,
3) narrow beam peak amplitude,
4) wide beam peak amplitude.

The output echo data blocks are linked together for each ping to form the processed data queue which is analysed after the detection of the bottom echo.

4.2.2 TS *and σ_{bs} calculation.* For each current ping after the bottom echo is processed the central computer modifies field 3 and 4 in each output echo data block from peak amplitudes to σ_{bs} and TS respectively using formulae (I.9) and (I.10). The logarithm is derived using a memory lookup table to minimize processing time. The list of output echo data block forming the processed data queue are available in common memory for transfer to the host computer.

4.2.3. Data transfer and TS/σ_{bs} *statistics.* At the beginning of each ping, the processed data queue is transferred from the VME common memory to the host computer memory using DMA transfer. The host computer sorts the data according to the data value and depth. The depth strata resolution is set by operator. The average value and standard deviation of TS and σ_{bs} for a specified fish population are calculated.

Histograms of TS data versus depth layers are also generated and may be presented in various formats including tabular, 2-dimensional graphics and 3-dimensional graphics for an arbitrary transects. The average *bs* value is used to calculate the integrator scaling factor for converting the relative "echo integrator output" into the absolute estimates of fish densities.

4.2.4. TS *data conversion — fish sizing.* Post processing routines convert the estimated target strength distributions to the equivalent fish length distributions so that both acoustic fish sizing and comparisons with simultaneous trawl-caught fish length data may be done. There are two options for fish length distribution estimation. The first procedure uses known empirical relationship between average target strength and average fish length determined from a least-mean-square analysis. Some relationships, as for instance the general Love equation [7]:

$$\mathrm{TS} = 19.1 \log L - 0.9 \log f - 62 \tag{3}$$

are stored in memory lookup tables. The second method utilizes the $\mathrm{TS}-L$ relation in general form [5]:

$$\mathrm{TS} = m \log L + b. \tag{4}$$

This allows the operator to specify and enter arbitrary values of the regression coefficients. This method also allows for some post processing corrections of TS/L distributions.

4.2.5. Echo tracking — grouping of correlated peak amplitudes A post processing tracking routine is performed on successive transmissions to increase the reliability of the average target strength estimate by minimizing the effect of fish orientation (aspect). The output echo data blocks in the processed data queue from successive transmissions are sorted to form groups of echo data blocks defined as fish data blocks presumably corresponding to the same fish as seen in successive pings (the fish data block can be thought of as fish echo traces as seen on the echogram records). The grouping of the data into the ordered sequence of peaks is based on the temporal/spatial correlation of the peaks. Peaks are said to correlate if they are seen in successive pings and have range tolerance within predefined limits [8].

For each echo data block in the current processed data queue, the list of fish data block is searched for a correlated block. If a correlated fish block is found, it is appended and the next echo data block is used. If no correlated fish data block is found, a new fish data block is added to the end of the list.

The fish data block contains:

1. Range of target.
2. Array of TS from each ping.

The number of peaks in a given fish data block or group (equivalent to the number of pings from which echoes from the same fish are detected) is a function of a several system parameters, viz: range of target, ship speed, ping repetition frequency and transducer beamwidth. To prevent tracking of false echoes and merging of groups, minimum and maximum number of peaks in a group are used to window the width of the fish data blocks. Both the minimum and maximum number of peaks in the group are depth related and are stored in arrays. The average value of number of peaks in a group are obtained by dividing the cross-sectional diameter of the beam at the angle determined by the beam pattern threshold and by the distance the ship moves between successive pings for each depth strata. Similarly the range tolerance used for correlating peaks is calculated for the same depth strata and stored in an array.

4.3. *Echo counting*

Echo counting estimates the number density of surveyed fish populations from the single fish echo counts. Basically, the echo counting mode utilized the same hardware configuration as dual beam TS estimation with some software options. Conventional echo counter systems using only a single beam are biased with some uncertainties in the sampled volume determination. There is an advantage in determining the sampled volume with a dual beam system, especially when the beam pattern threshold T_b is introduced to control the beam angle. Furthermore, the grouping of peaks in successive transmissions into the "fish" data blocks in the tracking routines supplies ready data on number of single fish.

The sampled volume in a single transmission for each specified depth strata is calculated and extended to the volume sampled during the processing interval corresponding to one fish trace in the echo tracking routine. Fish number data from the echo tracker program are sorted in depth and stored in two-dimensional array indexed by range and TS/length classes.

The counting number density is formed by dividing the fish number array elements by the sampled volume and normalizing to the standard volume. The output data constitutes the fish number density averaged over the given transects as well as fish number density by depth.

4.4. *Echo integration*

Echo integration is the second fundamental mode of the system operation, allowing estimation of the average density, $\bar{d}$, of the total surveyed fish population over specified transects. This in turn, allows the estimation of the total fish quantity or biomass by multiplying the density estimate by the volume of water occupied by the surveyed population.

The same system hardware is used for echo integration and TS estimation. Due to the very high sampling rate required for the digital filter, the signal is oversampled for echo integration. The filter output rate is reduced by digital decimation. The amount of reduction corresponds to the sampling period equal to half of the pulse length in use. The system's echo-integration software performs the integration and averaging functions in a digital manner on the sampled narrow beam channel output signals. The echo integration theory is presented in Appendix II.

Before the echo-integration survey is started, the programmable parameters must be entered in the host computer by the operator. Some of the more important parameters include:

1) number of pings in sequence or sequence length in time or distance intervals,
2) S/N threshold,
3) bottom window,
4) surface blank range,
5) surface or bottom locked mode,
6) range dependent layering,
7) overall range gate,
8) scaling constants.

After the system parameters have been setup, the system will integrate all returns for each range layer for all pings in the sequence. Also, bottom tracking is automatically done. Once the specified number of pings has elapsed, the blocks of relative or absolute densities are computed using predetermined scaling factors and formulae II.5 and II.6. The sequence index is incremented, the ping counter is reset and the processing begins for the next sequence.

In echo integration surveys, bottom detection and tracking are especially important for distinguishing between fish and bottom echoes to prevent the integration of large bottom returns. For bottom tracking, the system combines the elements of hardware (section 3.2.2) and software bottom processing. Once a bottom has been initially detected in hardware (signal exceeds the bottom threshold V_b) the programmable bottom window is centred automatically or manually around this range for the next ping. The amplitude of the hardware-detected bottom is used to determine the software bottom value, which may be set at -1 to -3 dB level of the hardware bottom. The software bottom provides a more reliable estimate of the actual bottom. All samples which form the leading edge of the bottom echo pulse are deleted and not integrated. The automatic bottom tracking is normally selected as soon as a bottom echo is within the bottom window. A computer set "false bottom" is injected if a bottom echo has not been detected. In bottom locked mode system operation, the bottom layer may optionally be extended upward to some desired distance from the bottom.

The echo integration processing cycle is finished at the end of each sequence of pings. The data is sent to the host computer as a block of relative (M_j) or absolute (d_j) acoustic estimates of fish density in predefined depth layers. The host computer averages the estimates of fish density over all layers to give an overall fish density estimate.

Echo-integration may be performed in two different manners, ping (surface) locked or bottom locked.

4.5 Simultaneous echo integration and TS *estimation*

The third optional mode of system operation involves combining the two fundamental modes, which is desirable in some applications. In this case, the data are collected and TS estimation processing done to obtain an average value of σ_{bs} from those returns classified as single targets. Echo integration processing is performed on all echoes and fish densities are computed using σ_{bs} estimated from the same fish population. There are some hardware constraints that must be overcome to obtain reliable estimates. The first concern is the TVG function; TS estimation on single targets requires a $40 \log R + 2\alpha R$ while echo integration requires $20 \log R + 2\alpha R$. As mentioned earlier, the computer can generate arbitrary coarse gain function in the receiver. For this mode of operation the hardware TVG function of $30 \log \mathrm{R} + 2\alpha R$ is used and subsequent adjustment for each process is done by the central computer. The second concern is the S/N threshold set in the digital signal processor. The hardware threshold is set on relatively low level (appropriate for echo integration) and the data is thresholded again in the TS estimation routine with a higher threshold.

5. Conclusions

The application of digital signal processing techniques and the utilization of a high performance central microcomputer has resulted in a dual beam sonar system that is compact, versatile and low cost. The system hardware is used for echo sounding, echo counting, echo integration and *TS* estimation. Furthermore, the hardware can be configured so that simultaneous echo integration and TS estimation can be performed. The user interface to the system is through a user friendly, menu driven personal computer making the system both easy to use and learn.

This system represents the latest generation of equipment for fish stock survery and assessment. In addition to colour echograms, it provides virtually real time estimation of fish stocks, including 3-*D* plots of fish abundance versus position, histograms of fish size distributions and printed reports of size and quantity of fish by time or position interval. Full control of data presentation is available so that data can be manipulated and presented as desired. Both raw and processed data can be stored on the PC's mass storage device. Data can be reprocessed onshore to provide comprehensive cruise reports without any additional hardware.

6. Appendix I – target strength estimation in a dual beam sonar

The target strength, TS, estimation method utilized in the dual-beam system can be categorized as 'direct in situ technique'. In this meaning, target strength data is extracted from free swimming fish of surveyed population by removing the beam pattern factor from the each individual fish echo. The applied dual beam target strength measurement technique, which we refer to as dual beam signal processing, uses the known dual beam concept [9], however its implementation is partly original both in hardware and software.

Acoustic energy is transmitted on a narrow beam $b_T(\theta, \phi)$, and received on both narrow $b_{RN}(\theta, \phi)$ and wide beams $b_{RW}(\theta, \phi)$. The voltage waveforms of a received echo at the output of each receiver channel, assuming a time varying gain of $40 \log R + 2\alpha R$, are given by:

$$V_N(t) = k_N{}^{1/2} \sigma_{bs}(\theta, \phi) b_T{}^{1/2}(\theta, \phi) b_{RN}{}^{1/2}(\theta, \phi) \cos(\omega_0 t + N_\phi), \tag{I.1a}$$

$$V_W(t) = k_W{}^{1/2} \sigma_{bs}(\theta, \phi) b_T{}^{1/2}(\theta, \phi) b_{RW}{}^{1/2}(\theta, \phi) \cos(\omega_0 t + W_\phi), \tag{I.1b}$$

Parameters k_N and k_W are sonar system constants for the narrow and wide beam channels respectively, which include the linear form of the source level SL of the sonar transmitter and overall voltage response VR of the receiver. The target back-scattering cross section at position (θ, ϕ) is represented by $\sigma_{bs}(\theta, \phi)$. Neglecting the time dependence of the received waveforms and assuming the circular symmetry of the transducer and thus its associated beam patterns $b(\theta, \phi) \stackrel{\Delta}{=} b(\theta)$, the respective voltage squared amplitude ratio becomes:

$$\frac{V_N^2}{V_W^2} = \frac{k_N}{k_W} \frac{b_{RN}(\theta)}{b_{RW}(\theta)}. \tag{I.2}$$

If we further assume equal system constants $k_N = k_W$, equation (I.2) simplifies to the form:

$$\frac{V_N^2}{V_W^2} = \frac{b_{RN}(\theta)}{b_{RW}(\theta)}. \tag{I.3}$$

Equation (I.3) reveals the principal advantage of the dual beam configuration. The ratio of the squared narrow and wide beam amplitudes is equivalent to the ratio of the respective beam patterns for each received echo. Thus an unambiguous measure of the target position in the beam, referred to the acoustical axis of the dual beam transducer is obtained due to the unique relationship of the beam patterns ratio values and their associated variable angle. This is true under the assumption that the target observed in successive transmissions has the same reflective beam pattern (e.g. isotropic scatterers) which not necessarily must be valid for fish targets wich possess directional back-scattering beam patterns $\sigma_{bs}(\theta, \phi)$ of variable value for different dish aspect [10]. The ratio for specified angle $\theta = \theta_{TH}$, has the sense of beam pattern threshold:

$$T_B = \left.\frac{b_{RN}(\theta)}{b_{RW}(\theta)}\right|_{\theta = \theta_{TH}}, \tag{I.4}$$

which bounds the sampling volume to the cone with angle, θ_{TH} as those echoes which have the ratio of squared voltages greater than the threshold will be detected. The cone with angle θ_{TH} is such that the wide beam pattern value is close to its maximum value $b_{RW} \simeq 1$, which in turn gives the approximate ralation:

$$b_{RN}(\theta) = \frac{V_N^2}{V_W^2}. \tag{I.5}$$

Once the beam pattern factor is known, the back scattering cross section of the target can be found. Solving equation (I.1a) for σ_{bs} we obtain:

$$\sigma_{bs} = \frac{V_N^2}{k_N\, b_T(\theta)\, b_{RN}(\theta)}. \tag{I.6}$$

If we assume that the transmit beam pattern and narrow receive beam patterns are approximately equal, especially when confined to small angles (in mainlobe – see Fig. 3) formula (I.6) simplifies to:

$$\sigma_{bs} = \frac{1}{k_N} \frac{V_N^2}{b_{RN}^2(\theta)}. \tag{I.7}$$

Comparing equation (I.7) with equation (I.2) we see:

$$\sigma_{bs} = \frac{V_N^4}{V_W^2} \cdot \frac{k_N^2}{k_W^2} \frac{1}{k_N\, b_{RW}^2(\theta)}. \tag{I.8}$$

Recalling that system constant is equal and the wide beam pattern is unity in the angles of interest, we obtain the simple expression for back scattering cross section:

$$\sigma_{bs} = \frac{1}{k_W} \frac{V_W^4}{V_N^2}. \tag{I.9}$$

Thus, by simultaneously measuring the narrow and wide beam voltages, the unknown beam pattern factor can be removed and as a result the back scattering cross-section of observed targets can be obtained. Consequently, the target strength TS, which is related to the back scattering cross section by the simple logarithmic relation:

$$TS = 10 \log \sigma_{bs}, \tag{I.10}$$

can be extracted from the received echoes using the following equation:

$$TS = 2W - N - K, \tag{I.11}$$

where $W = 20 \log V_N$ [dB], $N = 20 \log V_W$ [dB], $K = 10 \log k_W$ [dB].

Both the back scattering cross section and target strength primary data measured for individual fish are averaged over any specified population. Depending on the system's mode of operation each of these values estimated from the corresponding distributions (histograms) is used. The average target strength data are mostly used for comparison with and/or fish length statistics for fish sizing purposes. On the other hand, the average back scattering cross section is used to convert the relative echo-integration data into the absolute estimate of fish density, i.e. to scale the integrator output (see section 4.4).

7. Appendix II – echo integration theory

Under the assumption of non-coherent first-order scattering model for the collective echo from fish concentration, and with the TVG set at $20 \log R + 2\alpha R$, the running time average of the squared echo envelope over the range layer ΔR insonified in a given ping is proportional to the volume density of the scatterers (fish) in this layer [11]:

$$\int_t^{t+\Delta t} v_e^2(t)\,dt = \text{const}\, d_{\Delta R}. \tag{II.1}$$

If, after this first integration, the averaging over the specified surveyed transect is done, the average fish density in layer ΔR is estimated by:

$$d_{\Delta R} = (\text{const})^{-1} M, \tag{II.2}$$

where M is the mean integrator output over the transect and the proportionality constant is the product of the sonar system constant C_s and the average σ_{bs} of fish surveyed in layer ΔR. Thus, if the constant C_s has been measured and the σ_{bs} is known, or can be estimated (sec. 4.2), the average integrator output yields the absolute estimate of fish density or biomass depending on the form of σ_{bs}, i.e., related to the 'pure' TS, or to the TS_{1kg} [12, 5].

For each ping, the consecutive samples within each range layer must be squared and summed. For the jth layer on the kth ping, the partial sum of N squared samples with sampling interval i is given by:

$$S_{jk} = \sum_{i=1}^{N} (v_i)_{jk}^2. \tag{II.3}$$

In successive pings, the partial sums are added and after acquiring the data for all pings in a specified sequence of p pings, the transect sums are calculated for each layer:

$$S_j = \sum_{k=1}^{P} S_{jk} = \sum_{k=1}^{P} \sum_{i=1}^{N} (v_i)_{jk}^2. \tag{II.4}$$

After averaging these values by division by the total number of pings in the transect and by the number of samples in the layer, the "integrator output" for the jth layer is obtained, representing the relative estimate of fish density in the layer:

$$M_j = \sum_{k=1}^{P} \sum_{i=1}^{N} (v_i)_{jk}^2 / n_j, \tag{II.5}$$

where $n_j = \sum_{k=1}^{P} n_{jk}$, is the total number of samples in layer j for p pings. These relative estimates are finally converted to the absolute fish density when scaled by appropriate overall system constant:

$$d_j = C^{-1} M_j, \tag{II.6}$$

where d_j — is the acoustic fish density estimate for the jth layer, $C - C_s \sigma_{bs}$ is the overall scale constant comprising the sonar system parameters and the average back scattering cross section of surveyed fish.

References

[1] KNIGHT *et al.*, *Digital signal processing for sonar*, Proceedings of the IEEE, **69**, 11 (1981).

[2] F. GOODENOUGH, *ADC Chips leap ahead both in speed and accuracy*, Electronic Design, (1986).

[3] ANON., *Model 181 dual-beam processor with Acquire and Target Strength Software*, Operator's Manual, Biosonics Inc., Seattle June (1985).

[4] R. BREDE, *Simrad QD Echo-Integrator, in*: FAO Fisheries Circular No. 778, Acoustic System for the Assessment of Fisheries, FAO, Rome 1984.

[5] A. STEPNOWSKI and J. BURCZYŃSKI, *The analysis of the calibration constant in the hydroacoustic system of fish abundance estimation*, In: Meeting on Hydroacoustical Methods for the Estimation of Marine Fish Populations, June 1979, vol. 2, Part a, [ed.] by J. B. Suomala, Cambridge, Massachusetts, C. S. Draper Laboratory Inc., 325–336, 1981.

[6] B. W. KERNIGHAN and D. M. RITCHIE, The C programming language, Prentice-Hall, New Jersey 1978.

[7] J. J. TRANYONOR, N. J. WILLIAMSON, *Target strength measurements of Walleye Pollock (Theragra Chalcogramma) and a simulation study of the dual beam method*, ICES/FAO Symposium on Fisheries Acoustics, Bergen, Norway June 1982, FAO Fisheries Report No. 300, FAO, Rome 1983.

[8] L. M. DICKIE, R. G. DOWD and P. R. BOUDREAU, *An echo counting and logging system ECOLOG for demersal fish size distributions and densities*, Can. J. Fish Aquat. Sci., **40**, 487–498 (1981).

[9] J. E. EHRENBERG, *A review of "in situ" target strength estimation techniques*, Proceedings of the ICES/FAO Symposium on Fisheries Acoustics, Bergen, Norway 1982.

[10] K. G. FOOTE, *Rather high frequency sound scattering by swimbladdered fish*, J. Acoustic Society Am., 78, 688–700 (1985).

[11] J. BURCZYŃSKI, A. STEPNOWSKI, R. SALAMON, W. MARTIN, *The hydroacoustic system on R/V*

"Professor Siedlecki" and its use in fish stock assessment by integration of thin depth layers, Rapp. P.-v. Reun. Cons. int. Explor. Mer, 170, 142–151 (1977).

[12] J. D. R. BAYONA, *Differences in the scaling of echo integrator survey results by Fisheries Research Institutes*, FAO Fisheries Circular No. 778, 107–127 FAO Rome 1984.

Received on June 1, 1987.