An Oversampling Sonar Receiver with Digital Filtering

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Abstract:

This document describes a sonar receiver that uses digital signal processing techniques to simplify analogue design and allow the receiver to be easily reconfigured for different operating frequencies. Signal processing techniques used include digital filtering, envelope generation using the Hilbert transform, and oversampling which permits the use of simple (low order) analogue filters and, as a secondary benefit, increases effective A/D conversion resolution. The receiver incorporates a low noise differential preamp, a second order passive bandpass filter, an 80 dB variable gain stage and a second order low pass antialias filter. A National Instruments data acquisition (NIDAQ) card installed in a personal computer provides the A/D and the control signal for the receiver gain. National Instruments LabWindows CVI ("C" for Virtual Instruments) is used as the software development tool to provide an intuitive virtual control panel on a PC monitor and the library of digital signal processing functions used in the project.

Résumé:

Ce document décrit un récepteur de sonar qui emploie des techniques de traitement de signal numérique pour simplifier la conception analogue et pour permettre au récepteur d'être facilement modifié pour différentes fréquences de fonctionnement. Les techniques de traitement des signaux utilisées incluent le filtrage numérique, la génération d'enveloppe employant la transformée de Hilbert, et le suréchantillonnage qui permet l'utilisation de filtres analogues simples (mode d'ordre faible) et, comme avantage secondaire, améliore la résolution de la conversion analogique-numérique efficace. Le récepteur comprend un préampli différentiel à faible bruit, un filtre passe-bande passif du second degré, un étage de gain variable de 80 dB et un filtre anti-repliage passé-bas de second ordre. Une carte d'acquisition de données de National Instruments (NIDAQ) installée dans un ordinateur personnel fournit les signaux analogique-numérique et de commande pour le gain du récepteur. National Instruments LabWindows CVI ("C" pour Virtual Instruments) est l'outil logitiel de développement utilisé pour fournir un panneau de commande virtuel intuitif sur un moniteur de PC, ainsi que la bibliothèque des fonctions de traitement de signal numérique utilisées dans le projet.

INTRODUCTION

The SSG (Sonar Systems Group) 38 kHz sonar receiver was built to obtain a high quality sonar receiver whose parameters were totally under our control to provide a reference sonar for bottom classification work. The design relies heavily on digital filtering to reduce the complexity of the analogue filtering and also uses digital techniques in the generation of the echo envelope which can be a weak point in a strictly analogue system.
The system is designed to “piggy back” on a commercial sonar to use its transmitter and transducer for its own purposes without loading or affecting the operation of the host sonar.

SYSTEM OVERVIEW

The SSG sonar receiver hardware consists of the following components:

- A small shielded enclosure for the basic analogue processing hardware to interface to a host sonar transducer and a PC.
- A commercially available high speed multipurpose PC data acquisition card.
- A software control and data acquisition program developed using National Instrument’s LabWindows / CVI program development package which is based on the ‘C’ language and targeted to the development of instrumentation and control systems. Sonar controls appear as a virtual instrument on the monitor with “knobs”, “switches, etc. to facilitate control.

DESIGN PHILOSOPHY

A hardware design was desired that would minimise component changes (e.g. hardware filters) required to accommodate the various sonar frequencies in use which typically range from about 12 kHz to 200 kHz or higher. To achieve this goal, the analogue receiver was designed to be wide-band except for a single optional band-pass filter early in the amplifier chain to act as a preselector and suppress out of channel interference (e.g. a sounder operating at another frequency) which could cause amplifier overloading and hence masking of the desired signal. All further filtering and signal processing would be done digitally except for the necessary antialiasing filter which would be set at a frequency high enough to accommodate most commonly used frequencies. The minimum sample rate therefore is determined by the antialiasing filter design and upon how much signal one expects to see in the antialiasing filter stop band. This generally necessitates very high initial sample rates compared to a sonar where analogue filters are totally responsible for bandwidth limiting. In the SSG Sonar Receiver, the maximum A/D sample rate of 1.25 MHz combined with available computing power resulted in the antialiasing passband terminating at 50 kHz. However, with the selection of higher sampling rate boards and the use of high speed digital signal processors, this technique can readily be extended to 200 kHz. The operating frequency of 38 kHz, a standard CHS frequency for mid-water depth sounding, was chosen since it is a good choice for acoustic seafloor classification as it provides some bottom penetration, although other commonly used frequencies less than 50 kHz such as 12 kHz and 24 kHz could readily be accommodated.

As shown in the block diagram (Figure 1), the signal from the transducer is first routed to a discrete three opamp instrumentation amplifier which has a high input impedance and low input noise. This eliminates the requirement for an input transformer to provide a differential input and common mode noise rejection. The gain of this stage is low (unity) but it isolates the low input impedance of the following gain stages from the transducer and will accept an input of up to 5Vpp which can occur in shallow water. Following this is the optional 38 kHz passive bandpass filter which provides rejection of out-of-band signals which could overload the following high gain stages. This is the only hardware component that must be changed for the selection of any other operating frequency below 50 kHz and can in fact be omitted at the expense of increased susceptibility to out of band interference. The next section of the circuitry consists of a voltage controlled amplifier which has a gain range of 0 to 80 dB and which is used to implement both fixed and time varying gain (TVG) functions. The final stage in the receiver enclosure is a second order low pass antialias filter which provides attenuation of ~40 dB at the Nyquist (450 kHz) frequency before analogue to digital conversion at a rate of 900k samples/second. The combined attenuation of this filter, the sonar transducer, the passband filter, and the natural attenuation of acoustic energy with increasing frequency were deemed adequate to prevent aliasing of frequencies above Nyquist for most situations.

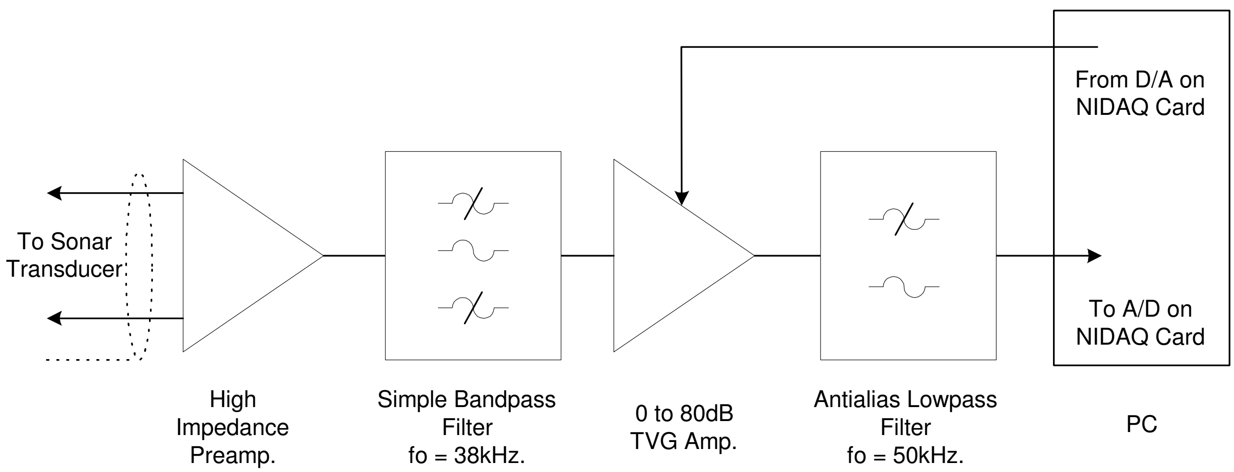


Figure 1 Block Diagram

DATA ACQUISITION BOARD (NIDAQ)

The final hardware section of the receiver is a National Instruments multi-function data acquisition board (NIDAQ card) which plugs into a PCI slot of a desktop PC. The board is able to digitize at 12 bits at up to 1.25M samples/second and write this data to computer memory using DMA (direct memory access) functions. In addition there is a 2048 byte on-board RAM cache which is loaded with the values necessary to control the TVG amplifier via a 12 bit D/A to provide 20 LOG TVG for the sonar. Simultaneous input signal acquisition and output of the TVG control voltage are accomplished through a logic trigger signal supplied from the analogue processor board to one of the board's digital I/O control lines.

DIGITAL SIGNAL PROCESSING

As mentioned, most of the signal processing has been moved from the analogue domain to the digital domain. The sample rate of 900 kHz (450 kHz Nyquist) together with the 50 kHz low pass and 38 kHz band pass offers good protection from aliasing out of band signals. However this sample rate is far greater than that required for the operating frequency pass band which is 31 kHz to 45 kHz (including filter transition bands) and is, in fact, about a factor of 30 times what is necessary i.e. 900 kHz / 28 kHz where 28 kHz is the minimum sample rate for a bandwidth of 14 kHz. The reduced sample rate is obtained by decimation and frequency translation. Since decimation reduces the sampling rate and hence the Nyquist frequency, a digital low pass filter to remove all frequencies above the new Nyquist frequency must be applied during these processes to prevent signal contamination due to aliasing.

These steps are done in the SSG Sonar in the following order:

1. An initial decimation by a factor of *five* with a 95 coefficient FIR filter with a pass band of 40 kHz and a stop band of 90 kHz. Note that this satisfies the criteria that all frequencies above the new Nyquist frequency of 90 kHz (450 kHz/5) are reduced to below the digitization noise level. This filter has a high attenuation factor since the analogue filters do not provide a high degree of rejection at 90 kHz (which is close to a standard sonar frequency of 100 kHz). See Figure 2 for the filter response of this first low-pass filter.

2. A second decimation by a factor of *two* is then applied with a pass band of 40 kHz, a transition band between 40 and 45 kHz and a stop band of 45 kHz . The sampling rate has now been reduced to 90k samples/second and the Nyquist to 45 kHz. The filter is a higher order 201 coefficient FIR filter to achieve the sharpness required to reduce all frequency components above 45 kHz to the digitization noise level (Figure 3).

3. The signal has now been band limited to 45 kHz and the sample rate reduced by a factor of ten. What is now required is to remove the frequencies below 36 kHz and to translate the frequency band of interest (31 kHz to 45 kHz which includes two 5 kHz transition bands) to a lower frequency and thus reduce the required sample rate. This is done in three steps. First the result of step 2 is multiplied by the Nyquist frequency of 45 kHz (multiplying the signal array by 1,-1,1,-1…..) which translates and inverts the spectrum as follows:

   45 kHz → 0 Hz -upper stop band from original
   40 kHz → 5 kHz -upper pass band from original
   38 kHz → 7 kHz -centre frequency from original
   36 kHz → 9 kHz -lower proposed pass band from original
   31 kHz → 14 kHz -lower proposed stop band from orignal

4. The band from 9 kHz to 45 kHz may now be removed since it is unwanted information originating from the band from 31 kHz to D.C.
   A low pass filter with a pass band of 9 kHz and stop band of 14 kHz is applied and the signal is decimated by *three* which results in a final Nyquist frequency of 15 kHz

and a sample rate of 30 kHz which is 1/30 of the original. This low pass filter is a 101 coefficient FIR (shown in Figure 4). The results of this still leave the signal spectrum inverted so a final multiplication by the new Nyquist frequency of 15 kHz re-inverts the spectrum:

0 Hz → 15 kHz -originally the stop band at 45 kHz
5 kHz → 10 kHz -originally the upper band pass at 40 kHz
7 kHz → 8 kHz -originally the band centre at 38 kHz
9 kHz → 6 kHz -originally the lower proposed passband at 36 kHz
14 kHz → 1 kHz -originally the lower proposed stopband at 31 kHz

Thus the original frequency band of 36 kHz to 40 kHz have been translated to a new frequency band of 6 to 10 kHz with two 5 kHz transition bands on either side. The information content of this final decimated signal contains the same information as the original. The decimation is done in three stages (5, 2, and 3) because digital filters become more efficient (require fewer coefficients) as the pass band approaches the Nyquist frequency. For example, if the first two filters were combined into a single filter with a decimation of 10 and the same filter properties as shown in Figure 3 , the number of mathematical calculations required would more than double.

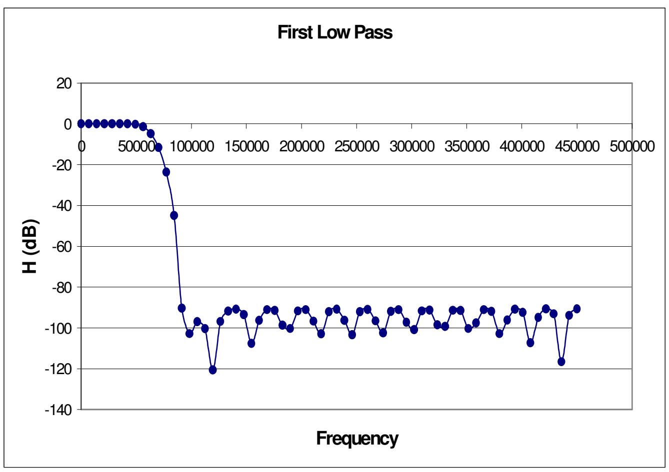


Figure 2 1st Decimation Filter

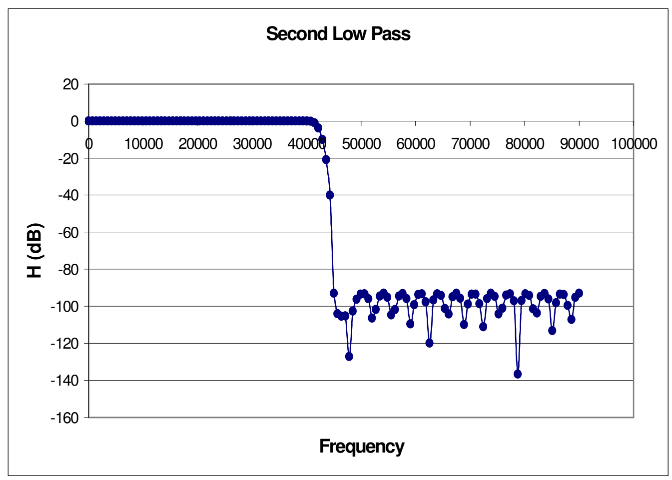


Figure 3 2nd Decimation Filter

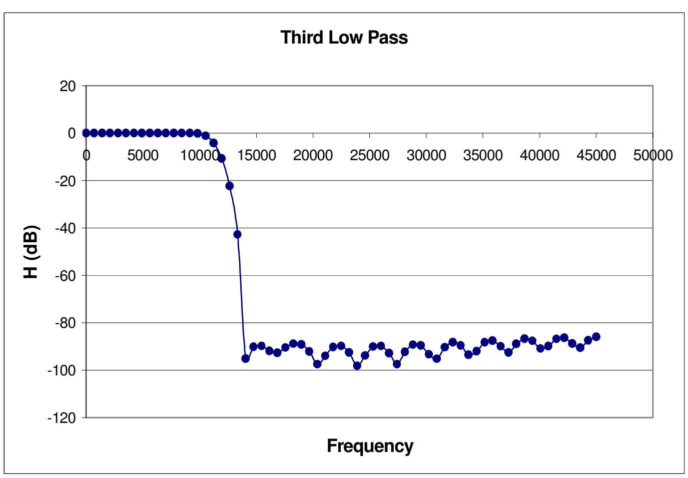


Figure 4 3rd Decimation Filter

FIR (finite impulse response) digital filters can provide extremely sharp cut-off rates, high stop band suppression, and flat passbands.

A beneficial effect of oversampling and decimation *with filtering* is the increase in the final *effective* resolution of the A/D. This results from the statistical nature of the settling of the least significant bit(s) of the A/D combined with averaging over a number of samples. The signal to noise ratio improvement is theoretically 10Log(decimation ratio). which results in a signal to noise ratio improvement in our case of 10Log(30) = 15 dB. This roughly equivalent to 2 ½ bits. Thus our original 12 bit A/D is now equivalent to a 14+ bit A/D *at the reduced sample rate*. Figure 6 shows a detected low level signal (a 38 KHz carrier 100% modulated by a 300 Hz triangular wave form) obtained as a detected output. The vertical axis is displayed at the *bit* level which have been converted to floating point early in the decimation process i.e. fractional bits are valid.

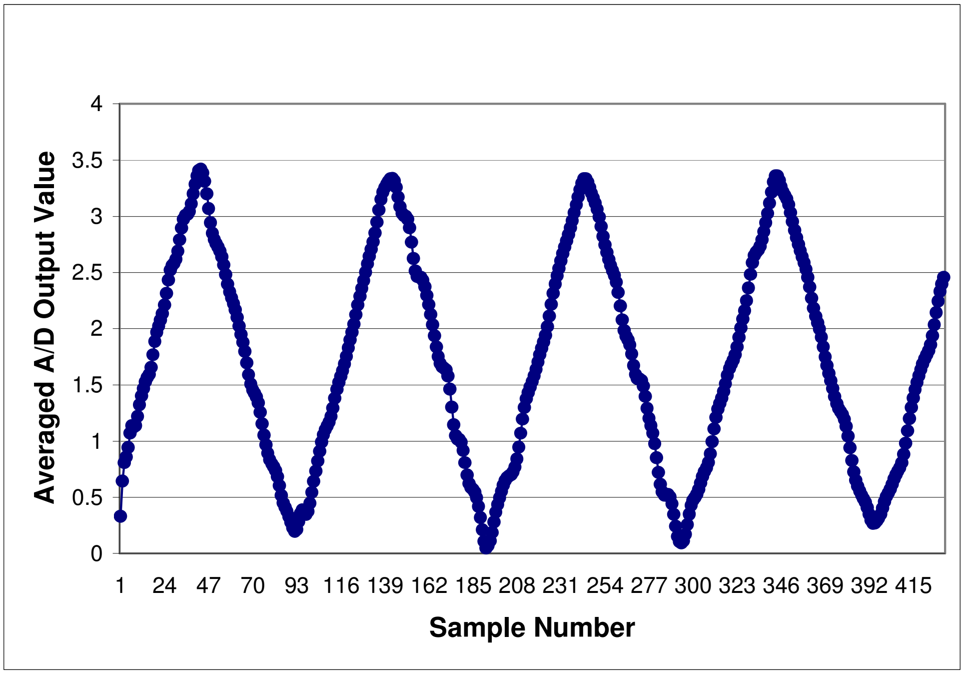


Figure 5 Low Level Digitization

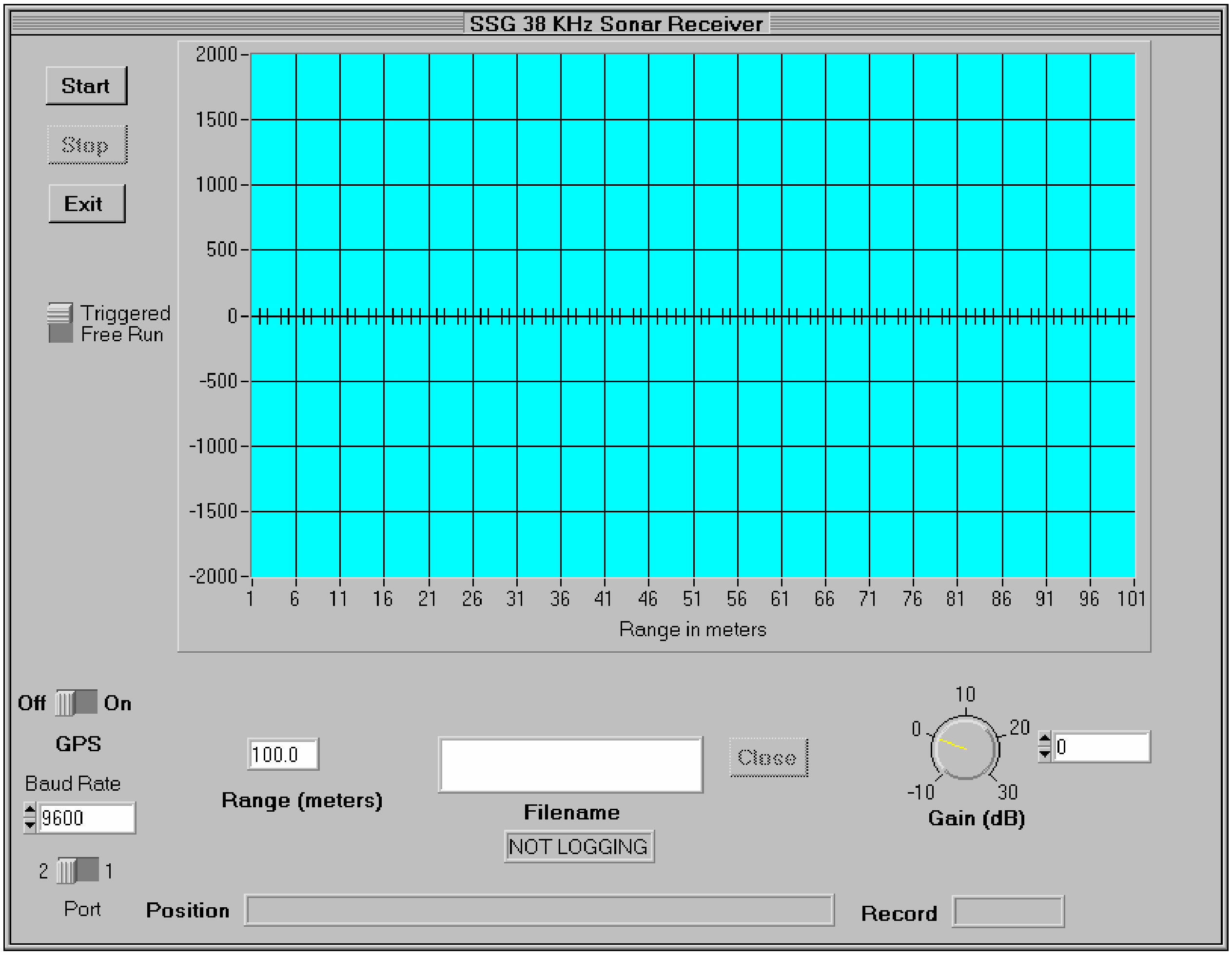
The envelope is obtained by utilizing the Hilbert Transform of the intermediate frequency (IF) signal centered at 8 kHz and is done as a post process during data processing to increase acquisition throughput. The Hilbert Transform shifts each of the signal's spectral components by -90^0 to create a -90^0 phase shifted version of the carrier with identical modulation which can be combined with the IF signal to obtain the envelope as in the following equation:

$E_i = \sqrt{s_i^{\,2} + h(s)_i^{\,2}}$ where E is the envelope, s is the original signal, and $h(s)$ is the Hilbert transform of s.

The advantage of this method over the quadrature sampling technique is that specific sampling rates are not required – the rate must only be above Nyquist. The Hilbert Transform function in the National Instruments signal processing library accomplishes this by taking a complex FFT of the original signal, multiplying positive frequency coefficients by –j, negative coefficients by j and taking the inverse FFT of the result.

THE VIRTUAL CONTROL PANEL

As previously mentioned, the operation of the sonar is accomplished via a virtual control panel that is displayed on the computer monitor as shown below in Figure 6.



As can be seen, the operation of the sonar is quite straight forward, with all functions visible. Knobs and slide or toggle switches are operated by clicking them and dragging the switch or control to the desired position and pushbuttons operate by clicking alone. The gain function has several methods of control- either by the rotary knob or the up-down arrow next to the adjacent readout box or by entering a number in the readout box. When a filename is entered in the Filename box, logging begins automatically. GPS GGA records may be logged with each ping if the GPS switch is set on, with the port and baud rate selected by the controls below. The desired range, entered in the Range box may be of any value up to 450m. In the present system the entire digitized water column return is logged and stored in binary format to minimize file size.

COMPUTER REQUIREMENTS

The downside of this design philosophy is the massive number of math operations required. For an example the 100 meter range has an output of 4096 samples. To arrive at this, approximately 125000 samples are taken by the A/D. The first decimation by 5 requires approximately 2.3 million multiplications and sums. The next decimation by 2 requires approximately ½ million multiplication and sums. The final decimation requires 41 thousand multiplies and sums. Fortunately the new generation of PCs have significantly faster memory bus speeds as well as math processing speeds. With an Athlon 1500XP processor and DDR memory, Windows 2000 operating system, the system is capable of processing approximately 2.5 pings per second at 100 meters range which is adequate for our purposes.

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