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Design of Digital Filters for Underwater Applications

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ABSTRACT: Acoustic signals are the important means of detecting Underwater objects. Due to physical and chemical properties of propagation medium, underwater acoustic signals suffer from ambient noise and serve transmission loss. Underwater sensor signals are acquired through a data acquisition system. The data acquisition system contains a acoustic transducer, preamplifier, anti aliasing filters ADCs and Processor. The sensor signals are digitized using ADCs of different resolution. These digital signals need to be processed for filtering, smoothening and peak detection. Digital filters are preferred to analog due to less hardware, re-configurability and more accurate at low frequencies.

In the proposed work, ARM cortex R processor based hardware is used for processing of digital filters. This ARM processor contains a floating point unit. With the help of floating point unit, digital filters will be implemented. Smoothening and peak detection algorithms will be implemented. This work will be useful for the development of underwater data acquisition systems.

KEYWORDS: Acoustic, ADC, ARM Cortex R Processor, DAS.

I. INTRODUCTION

Underwater acoustics is the study of the propagation of sound in water and the interaction of the mechanical waves that constitute sound with the water and its boundaries. The water may be in the ocean, a lake or a tank. Typical frequencies associated with underwater acoustics are between 10 Hz and 1 MHz. The propagation of sound in the ocean at frequencies lower than 10 Hz is usually not possible without penetrating deep into the seabed, whereas frequencies above 1 MHz are rarely used because they are absorbed very quickly. Underwater acoustics is sometimes known as hydroacoustics.

II. LITERATURE SURVEY

FILTERS:

A Filter is a device or process that removes some unwanted components or features from a signal. Filters are of two types. Analog and Digital. An Analog filter is any filter which operates on continuous-time signals. Coming to the title, a Digital filter is a system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects

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of that signal. A digital filter system usually consists of an analog-to-digital converter to sample the input signal, followed by a microprocessor and some peripheral components such as memory to store data and filter coefficients etc.

DIGITAL VS. ANALOG:

Digital filters are not subject to the component non-linearities that greatly complicate the design of analog filters. Analog filters consist of imperfect electronic components, whose values are specified to a limit tolerance. As the order of an analog filter increases, and thus its component count, the effect of variable component errors is greatly magnified. In digital filters, the coefficient values are stored in computer memory, making them far more stable and predictable.

Because the coefficients of digital filters are definite, they can be used to achieve much more complex and selective designs – specifically with digital filters, one can achieve a lower passband ripple, faster transition, and higher stopband attenuation than is practical with analog filters. Frequency selective filters with adjustable cutoff frequencies are achievable with digital filters. Digital filters are applicable to very low frequency signals.

System complexity increases in digital processing of signals. Limited range of frequencies are available. However advantages far outweigh the disadvantages and with continuing decrease in the cost of digital processor hardware, applications of DSP are increasing rapidly.

DIGITAL FILTER:

Digital filters are classified either as finite impulse response (FIR) filters or infinite impulse response(IIR) filters. In FIR system, the impulse response sequence is of finite duration whereas in IIR system, the impulse response sequence is of infinite duration.

FIR filter: It is a filter whose present output depends on present & past input. Filter structure is as follows:

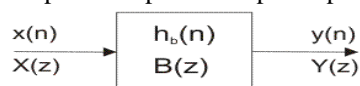


Fig: FIR filter structure

FIR filter equation:

$$y[n] = \sum_{k=0}^{N-1} h[k] \cdot x[n - k]$$

Fig: Filter equation

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{n=0}^{N-1} h[n] \cdot z^{-n}$$

Fig: Transfer function

IIR filter: It is a filter whose present output depends not only on past and present inputs but also on past output. Filter structure is as follows:

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IIR filter equation:

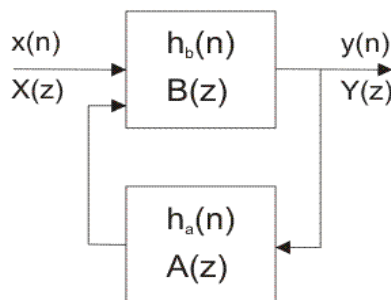


Fig: IIR filter Structure

$$Y(z) = \sum_{k=0}^N a_k z^{-k} X(z) - \sum_{k=1}^N b_k z^{-k} Y(z)$$

$$H(z) = \frac{\sum_{k=0}^N a_k z^{-k}}{1 + \sum_{k=1}^N b_k z^{-k}}$$

Fig: Transfer function

FIR filter	IIR filter
1)open loop system	1)closed loop system
2)N-1 filter order	2)N filter
3)only zeros are available	3)Poles and zeroes are available
4)FIR is always stable	4)IIR can be unstable
5) Requires more memory	5)requires less memory
6)Linear phase response	6)Non-linear phase response

Table1: Differences between FIR & IIR filters

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III. PROPOSED METHODOLOGY

UNDERWATER ACOUSTIC SYSTEM:

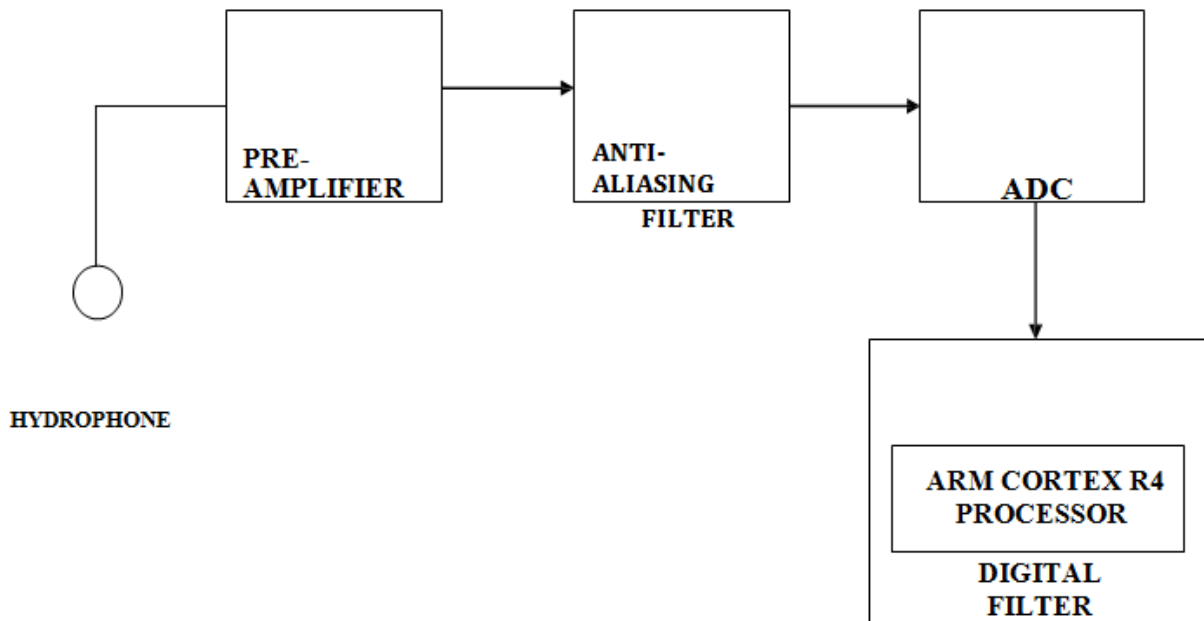


Fig: Block diagram of Underwater Acoustic System

Hydrophone:

The perception of external variations in pressure, temperature, velocity and sound wave detection is performed by a sophisticated device called a hydrophone. It is basically a sophisticated microphone designed to be used underwater. A hydrophone is a microphone designed to be used underwater for recording or listening to underwater sound. Most hydrophones are based on a piezoelectric transducer that generates electricity when subjected to a pressure change. Piezoelectric materials or transducers can convert a sound signal into an electrical signal since sound is a pressure wave.

Here in this Project, we use a hydrophone that works on same principle called Piezo-electric effect.

Pre-Amplifier (AD624):

Amplifiers increase voltage level to match the analog-to-digital converter (ADC) range, thus increasing the measurement resolution and sensitivity. Also improves the signal-to-noise ratio by magnifying the voltage level.

The AD624 is a high precision, low noise, instrumentation amplifier designed primarily for use with low level transducers, including load cells, strain gauges and pressure transducers. An outstanding combination of low noise, high gain accuracy, low gain temperature coefficient and high linearity make the AD624 ideal for use in high resolution data acquisition systems. The AD624 offers outstanding noise performance.

Filtering(Anti-aliasing filter):

Filters reject unwanted noise within a certain frequency range. Often times, low pass filters are used to block out high-frequency noise in electrical measurements, such as 50Hz power. Another common use for filtering is to prevent aliasing from high-frequency signals. This can be done by an anti-aliasing filter to attenuate signals above the Nyquist frequency. A band-pass filter to cut-off frequencies out of the frequency range of interest as well as suppressing alias effects.

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Most of under acoustic signals picked up by sensors like hydrophones are very weak, which requires a data acquisition system with high SNR performance. In order to obtain high-quality experimental data, an underwater acoustic data recording system is accomplished based on low-noise analog circuit design techniques, this system also features on high dynamic range, low distortion signal acquisition while achieving high sample rate and time storage.

Analog to Digital Converter:

Here, the analog data which is received, is converted into the digital data. Then these bands of low frequencies of digitized data are given to ARM. The digital filters are used for filtering of data of unwanted range of frequencies. Filters are used for removal of noise, and to get good linear response for different range of frequencies designed.

(AD7665):

The AD7665 is a 16-bit, 570 kSPS, charge redistribution SAR, analog-to-digital converter that operates from a single 5 V power supply. It contains a high speed 16-bit sampling ADC, a resistor input scaler that allows various input ranges, an internal conversion clock, error correction circuits, and both serial and parallel system interface ports. It features a very high sampling rate mode (**Warp**), a fast mode (**Normal**) for asynchronous conversion rate applications, and for low power applications, a reduced power mode (**Impulse**) where the power is scaled with the throughput.

TMS570LS3137 16/32-Bit RISC Flash Microcontroller: ARM Cortex R4 Processor:

The TMS570LS3137 is a high performance automotive grade microcontroller family for safety systems. The safety architecture includes Dual CPUs in lockstep, CPU and Memory Built-In Self Test (BIST) logic, ECC on both the Flash and the data SRAM, parity on peripheral memories, and loop back capability on peripheral IOs.

The TMS570LS3137 integrates the ARM® Cortex™-R4F Floating Point CPU which offers an efficient 1.66 DMIPS/MHz, and has configurations which can run up to 180 MHz, providing up to 298 DMIPS. The TMS570LS3137 has 3MB integrated Flash and 256KB data RAM with single bit error correction and double bit error detection. In addition to the built in ARM Cortex™-R4F CoreSight™ debug features an External Trace Macrocell (ETM) provides instruction and data trace of program execution. With integrated safety features and a wide choice of communication and control peripherals, the TMS570LS3137 is an ideal solution for high performance real time control applications with safety critical requirements.



Fig: TMS570LS3137 Microcontroller

IV. EXPERIMENTAL RESULTS

FIR filters:

FIR filters are the simplest filters to design. There are essentially three well-known methods for FIR filter design namely:

- 1) The window method
- 2) The frequency sampling technique
- 3) Optimal filter designs methods

Of the above three the major advantages of using window method is their relative simplicity that well defined as compared to other methods and ease of use.

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Windowing Techniques:

The basic idea behind window design is to choose a proper ideal frequency- selective filter and then truncate its impulse

response to obtain a linear phase and casual FIR filter. The analysis of different windows relating frequencies w.r.t magnitude and evaluation of Transition width is done.

Transition width: The **transition** band is a range of frequencies that allows a transition between a passband and a stopband of a signal processing filter.

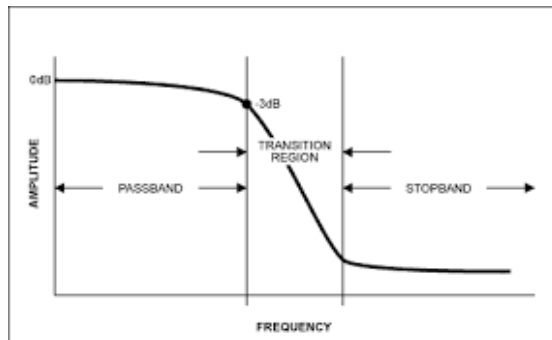


Fig: Transition width

Filters	Frequencies	Magnitude	Transition width
Hamming	25Hz 30.96Hz	-7.52dB -40dB	5.96Hz
Hanning	25Hz 31.06Hz	-6.48dB -40dB	6.06 Hz
Rectangular	25Hz 31.01Hz	-5.95dB -40dB	6.01 Hz
Blackmann	25Hz 33.08Hz	-6.26dB -40dB	8.08 Hz
Barlett	25Hz 57.26Hz	-5.90dB -40dB	32.26 Hz

Table 2: Analysis of different windowing techniques and obtaining transition width FIR filters

Filters	Frequencies	Magnitude	Transition width
Butterworth	25Hz 27.17Hz	-14.14dB -40dB	2.17 Hz
Chebyshev -I	25Hz 25.17Hz	-20.1dB -40dB	0.17 Hz
Chebyshev-II	25Hz 24.8Hz	-80dB -40dB	0.2 Hz

Table3: Analysis of IIR filters and obtaining transition width

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When FIR & IIR filters are compared FIR filters are stable. Therefore, FIR filters are being implemented in this digital filter design. By observing the above analysis we can clearly notice that from FIR filters, Hamming window has least Transition Bandwidth. So Hamming window is chosen in this project.

RESULTS:

Low pass FIR filter is designed with different range of frequencies. Here, in this paper filter range of 20Hz LPF is being presented. Software used: SIMULINK
Hamming windowing technique is implemented.

Case(i): Passband frequency: 15 Hz
(Inband) Sampling frequency: 200Hz
Cutoff frequency: 25Hz
Type of the filter: Direct form FIR

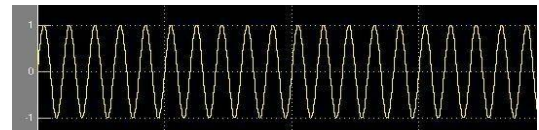


Fig: Output signal when $f=15\text{Hz}$

Order of filter: 50.

Case(ii): Passband frequency: 30Hz
(out of band) Sampling frequency: 200Hz
Cutoff frequency: 25Hz
Type of the filter: Direct form FIR
Order of filter: 50.

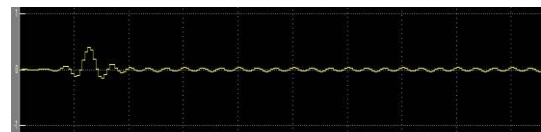


Fig: Output signal when $f=30\text{Hz}$

V. CONCLUSION

Finally, FIR digital low pass filter is implemented and results can be observed. Clearly we can observe from results that the filter has attenuated the frequency range of out of band. Similarly, the analysis of different filters is done through Simulink in order to choose the best filter to be implemented. CCS (Code Composer Studio) software is used in order to write the code and dumped through JTAG into Microprocessor. The low pass filters are designed with different range of frequencies which are very much useful for Underwater applications like target detection, Tsunami etc.

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