A DSP Based Underwater Communication Solution

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Abstract - The last several years have seen rapid advances in DSP (Digital Signal Processor) architecture technologies. These advances have made it possible for DSP's to be used for applications previously considered unsuitable for them. This paper describes an underwater communication radio that utilizes a DSP to perform all of the audio signal processing as well as radio functions. The primary advantage of utilizing a DSP lies in its flexibility. Because it is programmable, new code can be downloaded to change radio functions or to add a new algorithm. The current generation of DSP's have sufficient processing power to perform several concurrent signal processing operations including filtering, noise removal, channel equalization, AGC, etc., in addition to signal modulation and demodulation. This paper describes several of those functions and is primarily intended to showcase a DSP based solution as well as to illustrate what is achievable.

INTRODUCTION

Traditional electromagnetic wireless communication is nearly impossible in the underwater medium because of the high attenuation of electromagnetic waves. However, high frequency sound provides a viable alternative and has been used successfully in a variety of underwater communication systems. Most of the current underwater radio solutions utilize analog techniques. This paper outlines an alternative and improved digital implementation. It is organized as follows: Section I gives a brief introduction to Digital Signal Processing (DSP) techniques and elucidates the advantages of the digital approach. Section II describes a communication system built entirely from digital blocks and points out a number of enhancements and features that can only be implemented using Digital Signal Processing. This paper aims to whet the appetite of the reader and is not intended to be a complete design reference.

I: Introduction to Electrical Digital Signal Processing

Real world analog signals are continuous in both time and amplitude. Examples of analog signals include sound, pressure, temperature, position, etc. Transducers are used to convert analog signals such as pressure, temperature, acceleration, etc. into electrical signals (voltages or currents) in order to be processed electrically. This processing involves amplification, filtering, and linear and non-linear processing. Some common drawbacks with traditional analog implementations include accuracy limitations due to circuit complexity, device tolerances, and sensitivity to electrical noise. Degradation of performance and reliability changes with time due to device imperfections and unwanted dependence on environmental factors such as temperature and mechanical disturbances. Analog executions also make it difficult to implement non-linear and time-varying operations. Digital Signal Processing addresses most of these limitations, and although digital techniques have their own drawbacks, they are confined primarily to executing high data-rate, high frequency operations. For audio and frequencies in the less than 100KHz range, Digital Signal Processing provides far superior solutions over conventional Analog Processing.

In layman terms, Digital Signal Processing is a branch of engineering that allows us to replace circuits containing active and passive elements by an A-to-D converter, a numerical processor and a D-to-A converter. An Analog to Digital converter (ADC) converts a continuous signal into a discrete set of numbers (the digital signal) that represent the signal using a process called sampling. A Digital to Analog converter (DAC) converts a digital signal to an analog one by the process of reconstruction, which is essentially passing the digitized signal through a low pass filter. The digital signal is internally represented in the DSP using binary digits (bits). Obviously the greater the number of bits for each sample, the closer its resemblance to the amplitude of the original analog signal.

Shannon (1948) was the first to show that information in a signal is a finite, discrete subset of a finite continuous signal and furthermore described what needed to be done in order to extract this discrete data without losing any of the information present in the original signal. His paper on information theory is considered to be the starting point of modern day signal processing, although Joseph Fourier had earlier laid down the foundations in 1807 when he formulated what we now call the Fourier transform. In the 1930s, Nyquist, in another important paper, formulated the Sampling Theorem, which states that sampling an analog signal at twice the highest frequency encountered in the signal is theoretically enough to completely reconstruct the signal without any loss of information. Together, Shannon and Nyquist produced

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the two most fundamental ideas, which launched practical Digital Signal Processing. The discovery of the Fast Fourier Transform (FFT) in the 1950's by Cooley and Tukey along with the advent of the microprocessor in the 1970's and the DSP chip in the 1980's enabled further rapid advances.

The advantages of "going digital" are numerous. First, it is possible to get as much precision as is needed for most real-world problems by changing the word length of the DSP. The operations are perfectly repeatable and sensitivity to electrical noise is low (but not zero). Dynamic range can be improved vastly (almost infinitely) by using floating-point number representations. While, speed is a limiting factor in both DSP's and analog devices, advances in silicon technology and DSP architecture (parallel processing, faster memories, wider buses, etc.) are continually making DSP's faster. The biggest advantage of DSP's lies in their programmability. If there are new requirements to be added, new code can be downloaded into the same design without any hassle, and this can be done without limitations.

Performing time-varying operations like adaptive filtering is trivial using DSP's but extremely difficult using analog devices. In contrast with analog filters, there is no drift in performance with time as DSP arithmetic is absolutely stable. For many complex practical applications, DSP based solutions will consume less power than their analog equivalent circuits while occupying much smaller real estate. Electromagnetic Interference, which needs careful analysis and attention in the analog domain, is almost non-existent in the digital domain. While conventional analog processing seems to be limited by technology, Digital Signal Processing is limited only by the ingenuity of the engineer.

Any mathematical operation or algorithm that can be described can be performed on a digital signal by a numeric processor (i.e., a Digital Signal Processor). A significant amount of research has gone into defining mathematical operations and algorithms for a variety of applications. For communications, the most useful ones include Mixing, Synthesis, Filtering and Spectral Analysis. Filtering is an especially useful component of signal processing. Broadly speaking, a filter is a selective network, passing certain frequencies while attenuating others. This is achieved by selectively combining weighted past and present signal samples to generate the current output sample. Another useful tool is spectral analysis, which allows us to determine which frequencies are present in a signal and also the shapes of the signal magnitude profiles in the frequency domain.

Spectral analysis has enabled the development of several neat tricks that can be done only in the digital domain. A relevant example of this is speech enhancement. Speech can be made more intelligible using frequency domain analysis and selectively enhancing certain frequencies. While this is conceptually simple, it is difficult to provide a low-cost implementation of this. Switched capacitor filters provide an efficient solution in some cases but are limited in their ability to do complex filtering. Another such digital-only algorithm is adaptive noise reduction. By using matched filters whose responses model environmental noise, a diver's speech can be "compensated" and made to sound as clear as if he were speaking in a room. These are but a few examples of practical applications that DSP has enabled that were/are not available in the analog world.

However, all that glitters is not gold. DSP's are not without their disadvantages, the most notable being the distortion introduced by sampling. This is termed aliasing. While aliasing can be greatly minimized by careful sampling, it cannot be completely eliminated in a practical environment. Also, the ADC's and DAC's used for sampling and reconstruction may be expensive if high speeds and precision are of utmost importance. While the costs of DSP hardware are decreasing every year, the same cannot be said of software development. And, even with current technology, programmable DSP's remain impractical for many high frequency RF applications. In spite of these drawbacks, DSP's have much to offer and the advantages outweigh the drawbacks by a considerable measure.

II: A DSP-Based Underwater Communication System

A commercial underwater communication system that uses ultrasonic frequency for voice communication has been developed. The objective has been to replace underwater radios that use outdated analog technology. This section describes the basic building blocks of a DSP-based underwater communication system.

The block diagram of an underwater communication system shown in Fig. 1a for the transmit side and in Fig. 1b for the receive side. This particular configuration uses an amplitude-modulated single-sideband approach. One of the advantages of a DSP-based system is that if needed it may be changed into an independent sideband system without any change in hardware, by just downloading new code.

As shown in Fig. 1a, the input ADC converts the analog voice signal from the diver's microphone into a digital signal. A useful rule of thumb for calculating the number of required bits is that each bit adds about 6db of dynamic range. Therefore, 16 bits give a theoretical dynamic range of about 96db, which is sufficient for most audio applications. A bandpass filter limits the frequency range of the audio signal and attenuates out of band noise. Next, a voice activity detector (VOX) is used to gate the voice signal into the modulator. A notch filter may be used to remove hum from the unmodulated signal, and a more complex (optional) noise filter can be used to compensate for the

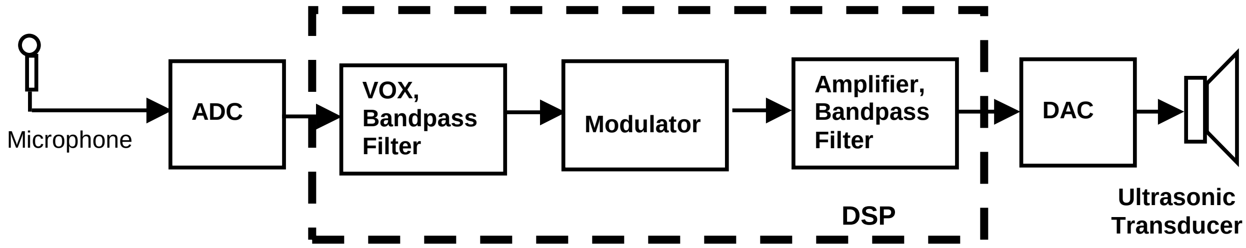

Fig. 1a. Transceiver block diagram

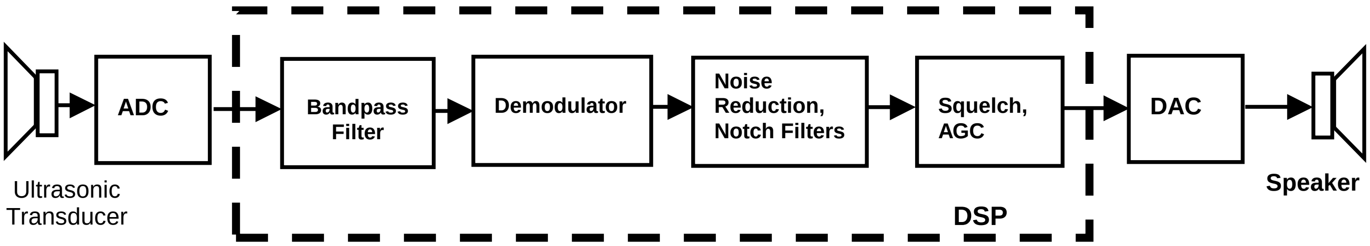

Fig. 1b. Receiver block diagram

microphone characteristics and environmental noise picked up by the microphone. The most commonly used SSB modulation techniques are the Hilbert transform method and the Weaver, or "third" method. An important factor to be considered when selecting a method is sideband suppression; both methods have their merits. It is beyond the scope of this paper to provide a detailed analysis but suffice it to say that both can be implemented in a DSP. A stable oscillator with the desired frequency range and precision can be easily implemented in the DSP to generate the carrier signal. After modulation, the signal is amplified to the desired level and bandpass filtered to improve the signal-to-noise ratio. After processing and modulation, the signal is converted back to an analog signal using a DAC. This signal is buffered and drives the ultrasonic transducer (transmitter).

The receiver side has a complementary structure. A similar ultrasonic transducer forms the input stage (receiver). An ADC digitizes the incoming signal. After this, a bandpass filter is employed to remove out-of-band channel noise. The signal can then be demodulated using the corresponding technique - either the Hilbert or Weaver method. Subsequently, a filtering algorithm is used to remove the ambient environmental noise and the noise introduced by the channel. A good noise removal algorithm should be adaptive, i.e., it should be able to adjust itself rapidly if a new environmental noise source appears. Ideally, the solution should not sacrifice voice quality for noise removal although some reduction in voice quality might be acceptable if the overall intelligibility improves. Some commonly used noise-removal algorithms include spectral subtraction, sub-band noise masking and matched filtering. A formant enhancer that selectively amplifies the voice formant frequencies is frequently used to further improve intelligibility. A squelch filter may be employed to suppress the output during silent segments, and a digital AGC can be used to set signal levels. Additionally, notch filters or comb filters can be used to remove harmonic noise components.

In summary, the advantages of using a DSP become evident when we consider the flexibility it offers. Each software module can be individually substituted when an improved module becomes available. This can be done multiple times without the need for a new hardware design. Changing from upper sideband to lower sideband transmission is as simple as changing the sign of a number inside the DSP. Shifting the carrier frequency is also a relatively simple task and all the filters can be reprogrammed to pass or attenuate the desired part of the spectrum. Another very important feature of the DSP board solution is the overall quality of speech that it can support.

The main drawbacks of the DSP based solution are its limited resolution and aliasing. And if not carefully designed, there may be unacceptable time delays. Although potentially problematical, these are not necessarily showstoppers. While it is true that analog representations have "infinite" precision and small delay, a 16 bit audio sample is in fact sufficient for radio quality speech. And while the downside of digital filters is the time delay introduced into the system, with careful coding it is always possible to compensate for system delay. Current development work is focused on creating new digital coding methods that can significantly increase the signal-to-noise ratio. Another new area of research is in modeling and canceling the effects of multi-path transmission.

CONCLUSION

This paper has presented an alternative solution to underwater radio. The proposed digital solution provides performance superior to the conventional analog one. Improvements in DSP and converter technologies have made the digital solution economically viable. This paper has provided a broad overview of the different modules needed in an underwater voice communication system and some of the digital improvements that can be added for a better sounding radio. In conclusion, a digital system reduces the overall system size and price, while increasing performance, quality and reliability.

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