IMPLEMENTATION OF UNDERWATER COMMUNICATION WITH TMS320VC5509A DSK

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Abstract

Underwater acoustic communication is a rapidly growing field of research and engineering as the applications. Especially it is necessary for submarine-ship, submarine-submarine and ship-submarine communications. 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. In this paper, we implement an prototype for underwater acoustic communication via the transmitter and receiver sides on two TMS320VC5509A DSK boards with using upper sideband - suppressed carrier (SSB) and selectable carrier frequencies 8,0875 kHz - 39 kHz including the frequencies defined in the STANAG 1074.

1. Introduction

Acoustic based underwater communication systems have been developing to provide underwater sound and telegraph communication by using sound carriers.

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 communication solutions utilize analog techniques. "[2]"

2. Underwater Sound

Sound is the disturbance of the medium – here water – travelling in a 3 dimensional manner as the disturbance propagates with the speed of sound. The sound is defined as a plane wave when the sound propagates in a single direction i.e. the lines for uniform phase are straight. "[5]"

Acoustic impedance is perhaps the most basic concept of underwater sound because its definition is a constitutive equation (one from which others are derived) for underwater sound propagation. The relation is:

$$p = Z_a.u -> Z_a = \rho.c \tag{1}$$

This definition is analogous to Ohm's law for electrical circuits i.e. V=R.I and you can often think of particle velocity, acoustic impedance and sound pressure in the same way. It shows that particle velocity and pressure are in phase in a plane sound wave. Acoustic intensity – power (Pa) per unit area (Aa) or energy flux - is used to describe levels of underwater sound e.g. an echo, a whale's call or a signal from a remote transducer. The intensity of a plane harmonic wave is:

$$I_a = \frac{P_{rms}^2}{\rho.c} = \frac{P_a}{A_a} \tag{2}$$

The daily term "a high sound" refers to a sound with a high intensity. A reference intensity I_{ref} has been defined in order to enable direct comparison of the loudness of sound, and the reference intensity used in underwater acoustics is that of a plane harmonic wave with an rms-pressure of 1µPa, which, for ordinary seawater, with c≈1500m/s and ρ≈1000kg/m3 gives

$$I_{ref} = \frac{(1\mu Pa)^2}{1000.1500} = \frac{10^{-12}}{15.10^6} = 0,667.10^{-18} \tag{3}$$

The intensity level (IL=how high a sound is) is the intensity of the sound wave given in decibels relative to the reference intensity of 1µPa plane wave rms-pressure.

$$IL = 10\log\frac{1}{0,667.10^{-18}}\, dBre1\mu PA \tag{4}$$

Sounds originating from acoustic sources are measured in intensity level, which decreases as the distance to the source is increased due to transmission loss (TL) i.e. spreading and absorption:

$$IL = SL - TL = SL - 20\log(r) - \alpha(r - 1m) \tag{5}$$

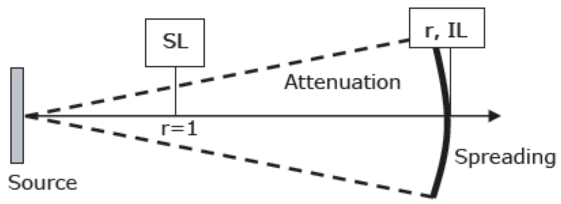

Fig. 1. Schematics of underwater sound transmission. "[5]"

Fig. 1 shows the underwater sound transmission. Spherical spreading is most common and valid in the far field given that the source is placed far enough from any large structure. Cylindrical spreading occurs for example in shallow waters when the bottom and the surface reflects the sound and forces it to spread like a cylinder. When the sound is completely bounded (e.g. inside a pipe) it cannot spread and only absorption remains in the formula for transmission loss. The last term of the transmission loss is the attenuation, which increases very significantly with the frequency and furthermore varies with pressure, temperature, salinity and acidity. "[5]"

The speed of sound is a very important parameter in any echo-sounding system where a range is determined based on the elapsed time and the speed of sound. The speed of sound can be approximated with a simple formula:

Table 1. Simple formula of the speed of sound

Description	Equation	Remarks	Limits
Speed of sound in seawater	$C = 1449.2 + 4.6T - 5.5 \cdot 10^{-2}\ T^2 + 2.9 \cdot 10^{-4}\ T^3 + (1.34 - 10^3\ T)\ (S - 35) + 1.6 \cdot 10^{-2} D$	C = speed of sound [m/s] T = temperature [°C] S=salinity [ppt] D=depth [m]	0≤T≤ 35°C 0≤S ≤ 45ppt 0≤d≤ 1000m

3. Single Side Band Modulation

Single-sideband modulation (SSB) is a refinement of amplitude modulation that more efficiently uses electrical power and bandwidth. One method of producing an SSB signal is to remove one of the sidebands via filtering, leaving only either the upper sideband (USB), the sideband with the higher frequency, or less commonly the lower sideband (LSB), the sideband with the lower frequency. We obviously have to translate the audio signal upward in frequency and preserve its spectral content within the band we want the transmitted signal to occupy. If we wish to produce an upper-sideband (USB) signal, we want the carrier and lower sideband to be suppressed as much as possible. Were we able to translate the spectrum of our cosine wave with its symmetrical positive- and negative-frequency components upward in frequency far enough, we would have two positive frequencies separated by twice the original signal's frequency. For a real signal, this is exactly what happens when it is applied to an analog mixer: Both sum and difference frequencies are generated. "[1]"

Upper Side Band

$$USB(t) = m(t)\cos(\omega_c t) - m_h(t)\sin(\omega_c t) \tag{6}$$

Lower Side Band

$$LSB(t) = m(t)\cos(\omega_c t) + m_h(t)\sin(\omega_c t) \tag{7}$$

If we want to create a signal having a one-sided spectrum from a real input signal, such as from the microphone, we need to shift all the frequency components in the sampled signal by 90º. Fortunately, in DSP, we have a way to do that: the Hilbert transformer. Recall that an FIR filter with a symmetrical impulse response exhibits a constant, frequency-independent delay. It turns out a filter with an anti symmetrical impulse response that is, with h(0) =–h(L–1), h(1) = –h(L–2), and so forth produces a linear phase response, too, but with a phase response exactly 90° different from the symmetrical impulse-response filter. This is exactly the type of filter we need to generate the components of an analytic signal.

Since the hilbert transformer includes not only a 90° phase shift, but also a fixed delay of L/2 sample periods, we need an L/2 delay in the leg that does not contain a phase shift. The delay through the two paths is then equal and the only difference between the two signals produced is the 90° phase shift. The non-phase-shifted signal is called I, the phase-shifted signal is called Q. Together these signals form our analytic signal I + jQ. "[1]"

3.1. Analytic Filter-Pair

We have seen how complex mixing translates signals in frequency with a one-sided spectrum. We will use this fact to our advantage in creating an analytic filter pair. Each filter will have the same frequency response as the other; they will differ only in their phase responses. "[1]"

We begin by designing a low-pass filter having the desired transition-band characteristic, H(w); we obtain its impulse response, h(t). Multiplying the impulse response by a complex sinusoid of angular frequency w_0 results in two sets of coefficients one for the real part, and one for the imaginary part:

$$h_I(t) = h(t)\cdot\cos[\omega_0 t] \tag{8}$$

$$h_Q(t) = h(t)\cdot\sin[\omega_0 t] \tag{9}$$

Which is a BPF centered at w_0. The I filter has a phase response differing 90° at every frequency from the Q filter. The frequency translation theorem works just as well on the responses of filters as it does on real signals. To perform this transformation of the L coefficients of the prototype LPF, we calculate new coefficients according to:

$$0 \le k \le L-1 \tag{10}$$

$$hI(k) = h(k)\cdot\cos\left[\omega_o\left(k - \frac{L}{2} + \frac{1}{2}\right)t_s\right] \tag{11}$$

$$hQ(k) = h(k)\cdot\sin\left[\omega_0\left(k - \frac{L}{2} + \frac{1}{2}\right)t_s\right] \tag{12}$$

Where t_s is the sampling period. When the low-frequency transition band is placed near zero frequency, as we would like for SSB, the BW of each BPF is approximately twice that of the prototype LPF.

4. Implementation Design

In this section, we describe the implementation of the underwater communication using DSP development board. The programming languages used for embedded system software are Assembler and C. This embedded system software was written in the (Texas, TMS320VC5509A) DSP environment which is called Spectrum Digital, TMS320VC5509A DSK.

While creating embedded system pieces, acoustic underwater communication software was created by choosing an effective and capable DSP. Telephone and telegraph operating frequencies are in the region of sound frequency, so DSP was chosen according to these frequencies.

Using CODEC integrated circuits which are typically used for sound applications, converting sound signals to digital signals and converting modulating/demodulating signals to sound signals are operated with high quality and simply.

It is very easy operation to just change or update software for new communication needs, while using DSP for modulating/demodulating operations. This is why DSP's are very effective and fast equipments for signal processing.

4.1. Functional Overview of the TMS320VC5509A DSK

The DSP interfaces to external SDRAM, Flash memory and an expansion memory interface connector through its 16-bit External Memory Interface (EMIF).

The SDRAM accesses are in 16-bit mode in chip enable 0 memory space. The EMIF provides the necessary refresh signals. The Flash accesses are in 16-bit asynchronous mode in the bottom half of CE 1 space.

An on-board AIC23B codec allows the DSP to transmit and receive analog signals. I2C is used for the codec control interface and McBSP0 is used for data. Analog I/O is done through four 3.5 mm audio jacks that correspond to microphone input, line input, line output and headphone output. The codec can select the microphone or the line input as the active input. The analog output is driven to both the line out (fixed gain) and headphone (adjustable gain) connectors.

Code Composer communicates with the DSK through an embedded JTAG emulator with a USB host interface. "[6]"

4.2. TMS320C5xx Digital Signal Processing Library

Digital signal processing library (DSPLIB) provides a set of C-callable, assembly-optimized functions commonly used in signal processing applications, e.g., filtering and transform. The DSPLIB includes several functions for each processing category, based on the input parameter conditions, to provide parameter-specific optimal performance. "[4]"

For FFT and FIR filtering, we use Texas DSPLIB library version 2.31. FIR filters are used, because they have linear phase. Filters are obtained in MATLAB and coefficients were transferred to the software.

5. DSP Implementation And Results

Fig 2 shows the DSP implementation prototype for underwater acoustic communication. DSP implementation has two 5509A DSK boards, two LCD Displays, two Keypads and two headsets. Initial implementation is performed in Matlab then run on the DSP board.

Fig 3 shows the system block diagram of the DSP implementation. As you can see there are two sides as transmitter and receiver. Each processing step on the receiver and transmitter meets each other. In our implementation, we select the microphone as the audio signal source and set the sampling rate to 96 kHz. To sample audio signals, one of the multichannel buffered serial ports (McBSPs) is configured to connect to the AIC23 codec. The audio data is transferred between the codec and the internal L2 memory through the direct memory access (DMA) channel to continuously save the raw audio data from the AIC23 codec. When one of the two buffers is filled, a DMA interrupt is initiated and the data is passed to the interrupt service routine (ISR) and then is processed.

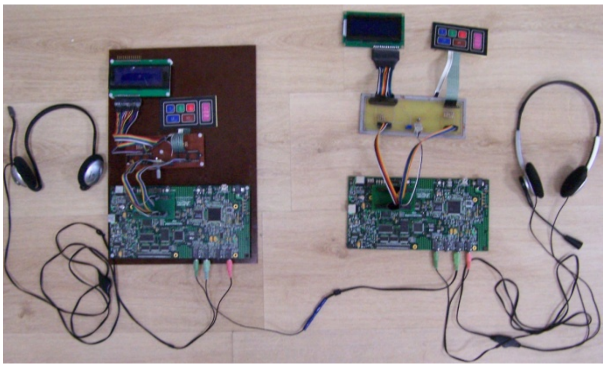

Fig. 2. The DSP based prototype for underwater acoustic communication.

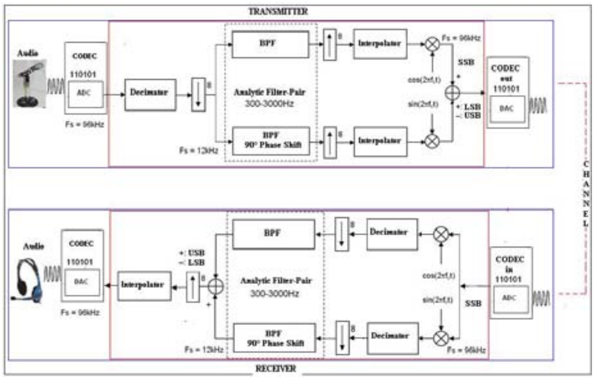

Fig. 3. Block diagram of DSP implementation

At the same time, the codec keeps sampling and saves data into the other buffer. So data sampling and processing can be done simultaneously and no incoming signals are missed even if the DSP is processing previously received data. Analog audio coming from microphone input of the AIC23 codec will be converted to a digital signal by the ADC of the AIC23 codec. Then this digital signal is sent to the DSP via McBSP serial port interface to be performed decimation/interpolation filtering, modulation/demodulation and mixer processing. Sampling frequency of the digital signal is decreased by a decimation filter to 12 kHz. This is exactly the same as running the decimation filter at the lower rate. This method is typical of those used by DSP designers to save time and effort. So, lower band pass filters and Hilbert transformation can occur in low rate. DSP receives two channels audio signal as left and right channel but only one channel audio signal is used. We used only left channel audio signal in this implementation. There are two channels as I and Q channel for signal processing. Band pass filter (300 Hz – 3000Hz) is applied to both Q&I channels. Additional Hilbert transformation is applied to the Q channel, too. Filtering operations have been performed by using the DSP's CSL library functions. After band pass filtering operation, we must increase the sampling frequency of digital data to mix with carrier signal that has high frequency. An interpolation filter is naturally needed. It is particularly convenient to choose an interpolation factor of 8 to increase the sampling rate to 96 kHz. Then a high-frequency signal is obtained by mixing with interpolated signal. Frequency of selected carrier signal is 8087.5 Hz. Our bandwidth is expanded to 48 kHz by increasing the sampling frequency of signal so that higher frequency of the carrier frequency to be sent and received. This implementation is for upper side-band modulation. According to USB modulation, we subtract the I and Q signals, we would have an USB signal. Thus, only the upper side band signal exists and the bandwidth will be used more efficiently. At this point we have generated USB signal and all operations are completed on transmitter side. Now we need to send the modulated signal to the receiver side. We can perform it by sending the modulated signal from line out of DSK1 (transmitter) AIC23 Codec to line in of DSK2 (receiver) AIC23 Codec.

Modulated signal is received on the receiver side. Firstly the received signal is multiplied by the carrier signals for I and Q channels. Then high-frequency signal will be converted to low-frequency signal. So, sampling frequency of the digital signal is decreased by the decimation filter to 12 kHz. Now we have signals with low sampling frequency. Band pass filter (300 Hz – 3000Hz) is applied to these signals on both Q&I channels. Additional Hilbert transformation is applied to the Q channel as on transmitter side. After all processing are completed for two channels according to USB demodulation, we should add the I signals and Q signals to regenerate the original signal. Sampling frequency of regenerated original signal is increased to 96 kHz. Then the digital signal is sent to AIC23 codec via the McBSP interface. Finally, the original signal is obtained but it is still digital. So it must be converted to analog signal. Digital data is converted to analog signal by DAC of AIC23 codec and the user can listen to analog audio via the codec's headset out. You can see the DSP implementation results in Fig. 4., Fig. 5., Fig. 6., Fig. 7., Fig. 8.

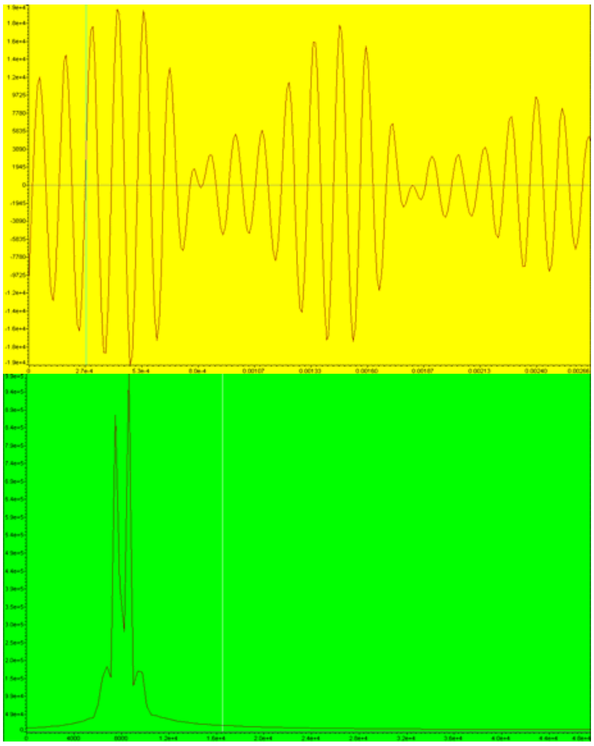

Fig. 4. Mixed message signal with carrier signal on I channel and frequency response.

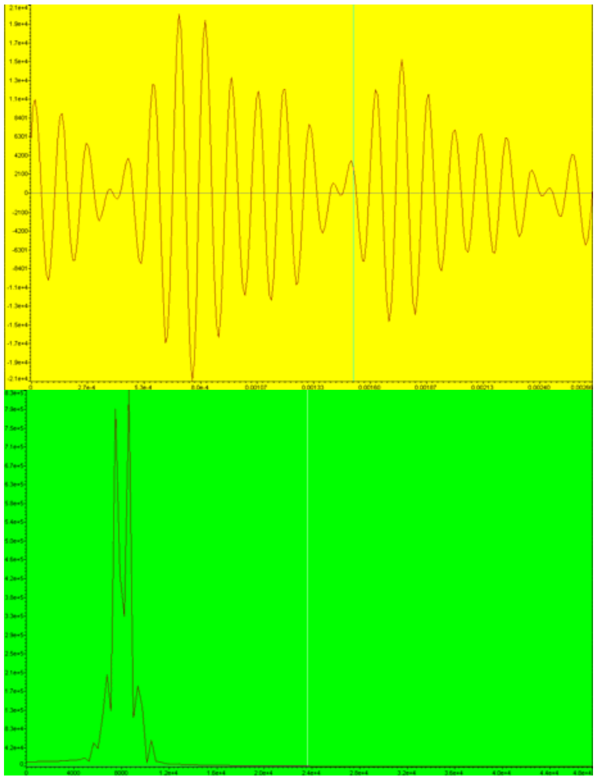

Fig. 5. Mixed message signal with carrier signal on Q channel and frequency response.

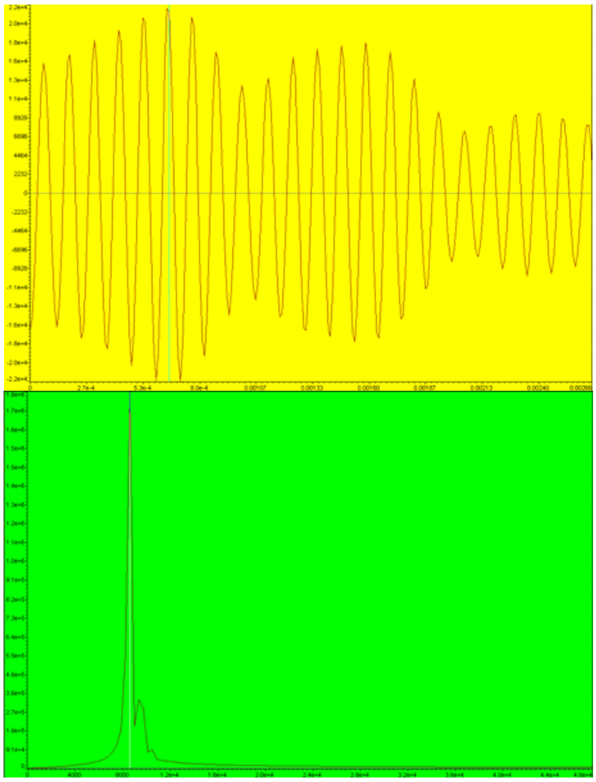

Fig. 6. Modulated USB signal and frequency response.

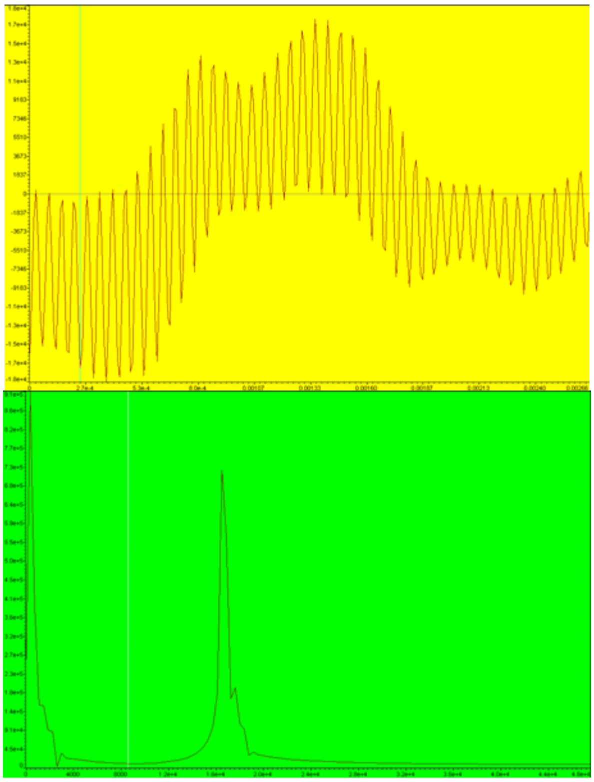

Fig. 7. Mixed received signal with carrier signal on I channel and frequency response.

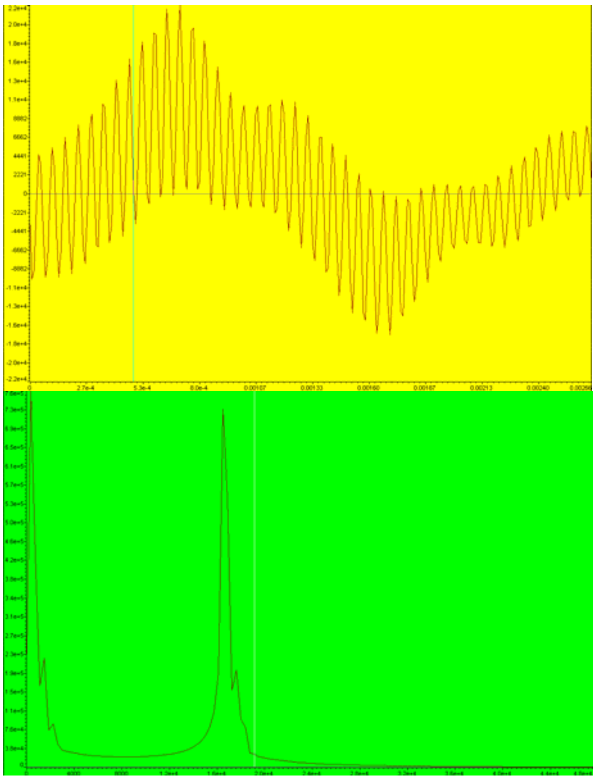

Fig. 8. Mixed received signal with carrier signal on Q channel and frequency response.

This implementation is a prototype for underwater communication. The results of lab testing are similar to Matlab implementation and meet STANAG 1074 standard.

6. Conclusions

Underwater acoustic communication is a rapidly growing field of research and engineering as the applications, which once were exclusively military, are extending into commercial fields. It is also very important because of communication requirement between submarines and ships.

In this paper, we implemented the underwater acoustic communication on two 5509A DSK boards for lab testing and have presented an alternative solution to underwater acoustic communication. 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. In conclusion, a digital system reduces the overall system size and price, while increasing performance, quality and reliability. The primary advantage of utilizing a DSP in its flexibility. Because it is programmable, new code can be downloaded to change radio functions or to add a new algorithm. All of the audio signal processing is performed in a more flexible and easy way. 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. 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.

As future work, we are motivated to pursue a hybrid DSP/FPGA-based solution to construct a real-time underwater acoustic communication.

7. References

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