CHIRP SOUNDING THE SHALLOW WATER ACOUSTIC CHANNEL

Gareth Cook and Anthony Zaknich

Centre for Intelligent Information Processing Systems
Department of Electrical and Electronic Engineering
University of Western Australia
Nedlands, WA 6907
email: gjc@ee.uwa.edu.au

ABSTRACT

Characterisation of the shallow water acoustic communications channel involves the analysis of sounding data. Chirp signals have many properties which make them an attractive choice for channel sounding. They are easily generated and channel responses can be processed in the time or frequency domain for channel estimation. In the rapidly varying shallow water environment time domain techniques are most appropriate. In this case weighting windows can be used to reduce clutter in the estimate. A channel sounding experiment is described which employs very simple hardware to generate and record chirp responses for offline processing.

1. INTRODUCTION

From the point of view of the communications engineer, the goal of channel sounding is to sample the time varying impulse response of a communications channel. In general, this sample is analysed (under the assumption of stationarity) to determine channel statistics which may be used to fit the channel to a known model. The shallow water acoustic communication channel is generally considered to fit within the fading multipath model (see [7], for example). In particular, the channel amplitude response is often assumed to exhibit Rayleigh or Rician statistics [1]. This assumption has been verified experimentally [9, 5] but is questioned by other researchers [2] (see [8] for further discussion). There has also been poor agreement between the performance of experimental systems and theoretical predictions based on fitting sounding data to the Rayleigh fading model [5]. It is clear that a great deal more data is required for analysis. The process of channel sounding is frequently reported to involve expensive, specialised and high powered hardware [5, 4]. In this paper we will review channel sounding concepts and highlight some of the attractive features of the chirp sounder. We present the results of a channel sounding experiment which made use of commonly available and (largely) non-specialised hardware.

2. CHANNEL SOUNDING

For linear channels, the impulse response $h(t)$ can be estimated by measuring the response of the channel to a short pulse. In general, the channel impulse response will be time varying, $h(t,\tau)$, so that h is the response of the channel at delay t to an impulse at time τ. In this case, the time varying impulse response of the (underspread) channel can be estimated by measuring the response of the channel to a train of pulses. Periodic pulse sounding, as it is known, requires narrow pulses (for good delay resolution) of high amplitude (for good signal-to-noise ratio). These requirements can often place prohibitive demands on sounding hardware. It is for this reason that pulse compression techniques are used.

2.1. Pulse Compression Techniques

In pulse compression techniques, the channel response is correlated with a delayed replica of the sounder to produce the impulse response estimate. The technique can be described identically in terms of linear filtering, where the channel response is passed through a matched filter. Firstly we excite the channel $h(t)$ with the sounder $x(t)$ to produce a response $y(t)$.

$$y(t) = h \otimes x(t) \tag{1}$$

Here $\otimes$ denotes the convolution operator. The channel response is then passed into a matched filter which will have an impulse response (by definition) $x^*(-t)$. The asterisk denotes complex conjugation and is included for generality.

$$\begin{aligned} \hat{h}(t) &= y \otimes x^*(-t) \\ &= h \otimes [x \otimes x^*(-t)] \end{aligned} \tag{2}$$

The matched filter output, $\hat{h}(t)$, is the channel impulse response estimate. From the second line of 2 it is clear that the estimate differs from the true impulse response through a convolution with the bracketed term. Examination of this bracketed term will reveal it to be the autocorrelation function of the sounder $x(t)$. The accuracy of the estimate will therefore be dependent on the autocorrelation of $x(t)$. In an ideal case the autocorrelation function would be the Dirac delta function (ie. an impulse) so that $\hat{h}(t) = h(t)$.

2.2. Sounding Waveforms

The process of channel sounding is identical to that in the conventional sonar (or radar) detection problem. It is only in the application to which the output is directed that they differ. As such, the set of active sonar waveforms is where we look to choose a sounding signal.

The most commonly used sounding waveforms to be found in the literature are short continuous wave (CW) pulses [4, 9, 5] and pseudorandom noise (PN) sequences [6, 3]. The autocorrelation functions of these sounders both have a triangular envelope. For pulse duration T and unity amplitude, the autocorrelation function of the short CW pulse has a base of width $2T$ and height $T/2$. For a $M > 1$ pulse PN sequence, the autocorrelation function will have a base of width $2T$ and height $(M+1)T/2$. The processing gain achieved through the pulse compression technique is self evident.

When sounding a time varying channel the multipath arrivals can have an associated Doppler shift as well as a distinct delay. We are therefore not strictly concerned with the autocorrelation of the sounder but with the ambiguity function of the sounder. The ambiguity function of the chirp waveform is undoubtedly less “ideal” than that of a PN sequence. However, the sensitivity of the chirp sounder to Doppler shifted multipath is not expected to limit its performance in the experiment described later.

It has been asserted that in the horizontal shallow water channel frequency spreading is low (less than 0.5 Hz) when the transmitter and receiver remain fixed [6]. In this type of environment chirp signals are well known to posses the best delay (range) sensitivity. Furthermore, a train of chirp responses can be processed to obtain both delay and Doppler estimates. The benefit in using chirp signals which will be focussed on in this paper is the option to employ windowing (either unilateral or bilateral) to reduce the sounder self-noise.

2.3. Linear FM (Chirp) Sounding

Chirp sounding is categorised as a pulse compression technique. However, there are two principal ways in which the sounder response can be processed to provide the channel estimate. The first is the matched filtering technique described earlier. The second involves multiplying the received channel response by a delayed replica of the sounder.

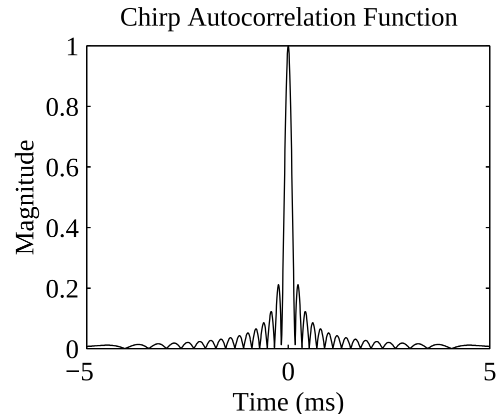

Figure 1: The envelope of the sounder autocorrelation function

Through this multiplication, the various multipath components will produce distinct beat frequencies which can be mapped to the delay domain by Fourier analysis.

The fundamental difference between these techniques is that the first is a purely time-domain method whereas the second involves both time and frequency domain analysis. Implicitly, the second method requires that the sounder response is a *steady state* measurement and this places restrictions on the duration of the chirp.

In the matched filtering technique, the only restriction on the duration of the chirp is that it does not exceed the coherence time of the channel. That is, the channel characteristics cannot vary significantly over the period of one chirp. In the second technique however, the requirement for a steady state measurement implies that the duration of the chirp is significantly longer than the multipath spread of the channel. This technique is only valid then for channels which vary relatively slowly. In the experiment described in this paper, the response data was processed using the matched filtering technique. This technique has the added benefit of being asynchronous, which simplified the offline data processing.

The autocorrelation function of the sounder used is shown in Figure 1. As can be seen in the plot, there are significant secondary lobes which will introduce error into the impulse response estimate. As was mentioned earlier, weighting windows may be employed (at some expense to resolution) to reduce these sidelobes. Figure 2 shows the result of weighting the matched filter coefficients with a Hamming window.

Figure 3 shows the sidelobe reduction on a log scale for clarity. As is shown in the plots, unilateral Hamming weighting results in a maximum sidelobe level of around -40 dB while bilateral weighting reduces the figure to around -50 dB.

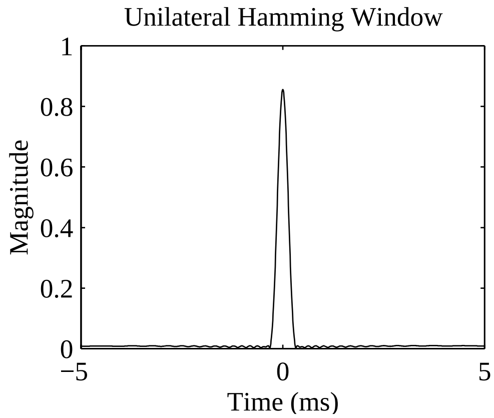

Figure 2: The envelope of the sounder autocorrelation function with unilateral Hamming weighting

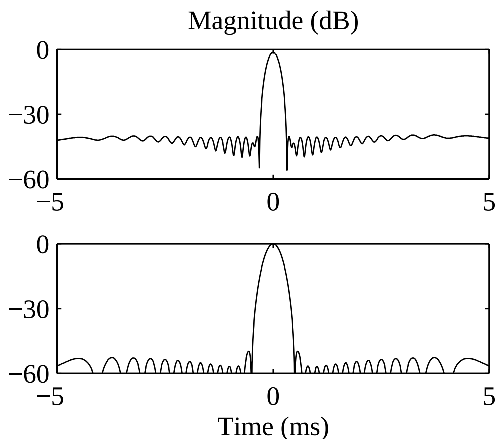

Figure 3: The envelope of the sounder autocorrelation function with (top) unilateral and (bottom) bilateral Hamming weighting

It is obviously possible to use other weighting windows to tailor the autocorrelation function. For instance a unilateral Blackman weighting has a maximum sidelobe level of -40 dB but the sidelobes near the main lobe are below -50 dB. A bilateral Blackman weighting has a maximum sidelobe level of less than -80 dB (although it has a much wider main lobe).

3. CHIRP SOUNDING EXPERIMENT

3.1. Environment

On 14 May 1997 an experimental chirp sounder was deployed between two jettys in the Fremantle Fishing Boat Harbour. Fremantle is a port city located in Western Australia. The channel was approximately 150 m long and 4 m deep with the transmit and receive transducers both located at a depth of 1.5 m.

3.2. Procedure

The chirp signal used in this experiment had a duration of 10.306 ms and had a bandwidth of 6 kHz centred at 17550 Hz. The limitations of the hardware used to generate the signal required that the signal was unweighted. The chirp repetition period was determined by the RMS power limit of the power amplifier used and was set at around 100 ms.

A chirp waveform with constant amplitude is fully defined in terms of its zero crossings. These zero crossings were generated by switching two lines on the parallel port of a laptop computer. The parallel port was connected to the input of a power amplifier operating in switching mode which in turn was used to drive the transmitter, a tuned cylindrical radially polarised piezo-electric transducer. A similar device was used as the receiver. Its output was recorded without preamplification using a Sony Digital Audio Tape (DAT) recorder. Approximately 1 minute of data was recorded.

The sounding data was transferred to computer file by recording the DAT analog output using a standard PC soundcard sampling at 44.1 kHz with 16 bit resolution. The resulting file (".wav" format) was imported into MATLAB for analysis. The entire data file was filtered using a Hamming weighted (mis)matched filter. The filter output was then divided into frames representing individual chirp responses.

3.3. Results

Examination of the data revealed that the full dynamic range of the DAT recorder was not being used. However, analysis shows that this would not have compromised the results. Figure 4 shows a segment of a typical response where an output swing from 1 to -1 represents the maximum signal level. The signal has a dynamic range of about 11 bits or 66 dB which implies that the full dynamic range of the sounder is available. Furthermore, it was observed that the channel variations were indeed slow with little change in the channel response between consecutive chirps.

Figure 5 shows the envelope of the channel response estimate over time.

4. CONCLUSION

Chirp sounding has been reviewed in relation to the shallow water acoustic channel. The use of weighting windows to modify the autocorrelation function of the sounder

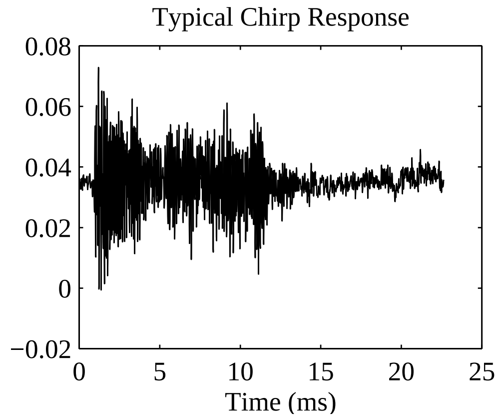

Figure 4: A typical chirp response

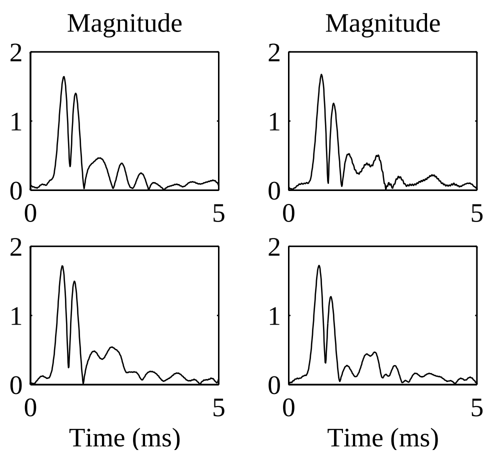

Figure 5: The channel response estimate over time. From left to right and top to bottom the estimates are at time (in seconds) $t = (0, 5, 25, 50)$

has been discussed. The process of channel sounding often involves expensive, specialised and high powered hardware. We have presented the procedure and results of a chirp sounding experiment which made use of commonly available and (largely) non-specialised hardware.

The current work has led to the identification of some areas where improvements could be made. The method used to generate the sounder waveform causes phase distortion due to the finite time resolution with which the parallel port can be switched. By using a standard PC soundcard and a linear power amplifier a much more accurate waveform can be generated. This modification would also allow the use of bilateral sounder weighting to shape the autocorrelation function.

Any setup which makes use of standard soundcards will limit the chirp's upper frequency to around 20 kHz. Since high data rate underwater communication generally requires carrier frequencies around 50 kHz, analogue to digital converters operating at suitably high sampling rates must be used. One possible solution is to make use of two PCs equipped with data logging hardware in PCMCIA format. Devices of this sort are available with sampling frequencies of 100 kHz.

5. REFERENCES

[1] Josko A. Catipovic. Performance limitations in underwater acoustic telemetry. *IEEE Journal of Oceanic Engineering*, 15(3):205–216, July 1990.

[2] A. Essebbar, G. Loubet, and F. Vial. Underwater acoustic channel simulations for communication. In *Proceedings of OCEANS '94*, pages III–495–III–500, September 1994.

[3] Lee E. Estes, Gilbert Fain, and Doris Carvalho. Simulation of a shallow water acoustic telemetry channel. In *Oceans '92*, pages 584–589, 1992.

[4] R. Galvin and R. F. W. Coates. Analysis of the performance of an underwater acoustic communications system and comparison with a stochastic model. In *Oceans '94. Oceans Engineering for Today's Technology and Tomorrow's Preservation*, volume 3, pages 478–482. IEEE, IEEE, September 1994.

[5] R. Galvin, R. F. W. Coates, L. S. Wang, and R. Stoner. Measured channel sounding characteristics and their relationship with the performance of a parametric communication system. In *Oceans Conference Record*, volume 2, pages 826–831. MTS/IEEE, 1996.

[6] Gerard Loubet and Genevieve Jourdain. Characterization of the underwater medium as an acoustical horizontal transmission channel. In *1993 IEEE International Conference on Acoustics, Speech and Signal Processing*, volume 1, pages 329–332. IEEE Signal Processing Society, IEEE, 1993.

[7] John G. Proakis. *Digital Communications*. McGraw-Hill, Inc., 3 edition, 1995.

[8] Milica Stojanovic. Recent advances in high-speed underwater acoustic communications. *IEEE Journal of Oceanic Engineering*, 21(2):125–136, April 1996.

[9] M Zheng. Experimental study on statistical characteristics of pulse transmissions in shallow water. *Proceedings of the Institute of Acoustics*, 15(9):142–151, 1993.