THINK INDIA JOURNAL

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

ANALYSIS AND STUDY OF DOPPLER SPREAD CHANNELS FOR UNDERWATER ACOUSTICS COMMUNICATION

Manwinder Singh *, Kamal Kumar Sharma *, Daljeet Singh *

*School of Electronics & Electrical Engineering

Lovely Professional University, Phagwara,Punjab, India

manwinder.25231@lpu.co.in

ABSTRACT: Underwater acoustic channels are characterized by time-varying and multi-path features. Underwater experiments conducted to test communication systems are expensive. Simulation techniques that model underwater acoustic signal in a realistic sense, which is still an open issue, are useful for verification of new communication techniques before experiments and for comparison of different systems in the same scenario. In this paper, we propose a new approach to model the underwater acoustic channel for communication. This approach is unified for any kind of movements in any scenario in the sense that the time varying channel is introduced according to the movements. The viability of this approach is evaluated by modeling signal of real experiment in both shallow water and deep water scenarios. It is shown that the numerical results in simulation are well matched to those in the experiments from the aspects of modeling channel Doppler spread, the signal for matched field localization, and the effect of motion fluctuations.

Index Terms— underwater acoustics, channel modeling, Doppler spread, channel simulator

1. INTRODUCTION

Communication through underwater acoustic channels is difficult due to the time varying multi-path structures of the channels, considerable Doppler spread and shift caused by the low speed of sound (1500 m/s) and the unavoidable relative movement between transmitters and receivers [1]. In the literature [2, 3, 4], many techniques have been proposed to overcome the difficulties in underwater acoustic communication. However, due to the expensive cost of experimentation and the time varying feature of the channel, examining and comparing the performance of these techniques in the same underwater environment is difficult to conduct. Therefore, impulse responses. However, for a fast moving source or a source moving for a long distance, the structure of the impulse responses, i.e., the number of eigen path and the delay of each eigen path, may change significantly. In these cases, it is difficult to find a proper statistic approximation of these variations.

Thus, Doppler effect is not considered in such channel simulators. A more advanced channel simulator [8] models the fluctuations of the amplitude and phase of each eigenpath, which are assumed to be caused by

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

source and receiver motions. However, this model fails to give a physical relation between these fluctuations and the motions, where y(t) is the received signal and h(t) is the channel impulse response. To take account for the Doppler effect in the simulated signal caused by transmitter and receiver motions, we consider the channel impulse response as time variant in the sense that different source and receiver positions have different transmission channels. therefore, it cannot be used to model signal for arbitrary source and receiver motions. However, for a fast moving source or a source moving for a long distance, the structure of the impulse responses, i.e., the number of eigen path and the delay of each eigen path, may change significantly. In these cases, it is difficult to find a proper statistic approximation of these variations.

In this work, we propose a unified approach for simulating underwater acoustic signal propagating through a Doppler spread channel caused by transmitters and receivers motions. The basic impulse responses are interpolated to obtain intermediate impulse responses to match the sampling rate of the source signal. The simulated received signals are then modeled by computing the convolution of the impulse responses and the trans- mitted signals. In this approach, the time variant of the channel is introduced according to the movement, and the impulse responses can be calculated provided that the environmental parameters are known. Therefore, the proposed simulation approach can deal with arbitrary movement in arbitrary scenario.

In the next section, a brief description of the proposed approach is given. The methods of computing and interpolating the impulse responses are discussed. In section 3 and section 4, the proposed approach is applied to model signals in a shallow water experiment SWellEx-96 Event S5 and a deep water experiment in the Pacific ocean, respectively. The simulation results are compared with the experiments. Finally, section 5 draws the conclusions.

2. ACOUSTIC CHANNEL SIMULATOR FOR COMMUNICATION

In a time invariant channel, signal at a receiver generated by a stationary source is characterized by the convolution of the source signal and the channel impulse response:

$$y(t) = \int_{-\infty}^{\infty} s(\tau)h(t-\tau)d\tau, \qquad (1)$$

Therefore, the received signal for a moving communication system can be characterized by:

$$\begin{aligned} y(t) &= \int_{-\infty}^{\infty} s(\tau)h(x(\tau), t-\tau)d\tau, \\ &= \int_{-\infty}^{\infty} s(\tau)h(\tau, t-\tau)d\tau, \end{aligned} \qquad (2)$$

where x(t) is the source/receiver position, which is a function of time.

In digital signal processing, the continuous signal is converted to discrete form by sampling at some

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

specific frequencies f_S. Rewriting (2), we get:

$$y(nT) = \sum_{i=-\infty}^{\infty} s(iT)h(iT, (n-i)T), \quad (3)$$

where $T = (1/f_S)$.

2.1. Generating Channel Impulse responses

Eq. 3 yields impulse responses at different sampling time. The wave equation for point source s(t) in a range independent scenario, is given by [10]:

$$\nabla \cdot \left(\frac{1}{\rho} \nabla p\right) - \frac{1}{\rho c^2(z)} p_{tt} = -\frac{s(t)}{r} \delta(z - z_{s,r}), \quad (4)$$

From the point of computing the impulse responses, there are mainly two ways, one of which is to compute the acoustic field for a source which is generating a continuous harmonic signal [11]. A straightforward method for doing these is the normal mode method [6]. The impulse responses are then obtained from Fourier synthesis of the resulted complex acoustic field for different frequencies. The other way of computing the channel impulse responses is using ray tracing method [7], which is able to compute the eigenpath, starting from the source and hitting the receiver in the end. The propagation delay and the complex amplitude of each eigenpath can be used to compute the impulse response for a given bandwidth.

There are many well developed programs for solving the wave equation and generating the results that we are interested in. By providing accurate environment descriptions, these acoustic field computation programs can provide results very close to those in realistic cases [12]. In this work, we use a normal mode method KRAKEN [6] for field computation at low frequencies, which pro- vides accurate acoustic field with relative fast speed.

2.2. Interpolation for Channel Impulse responses

According to (3), the impulse responses and the source signal should be sampled at the same rate as the received signal, which results in the same sampling rate on the movement trajectory. For each sample of the trajectory, a channel impulse response should be generated.

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

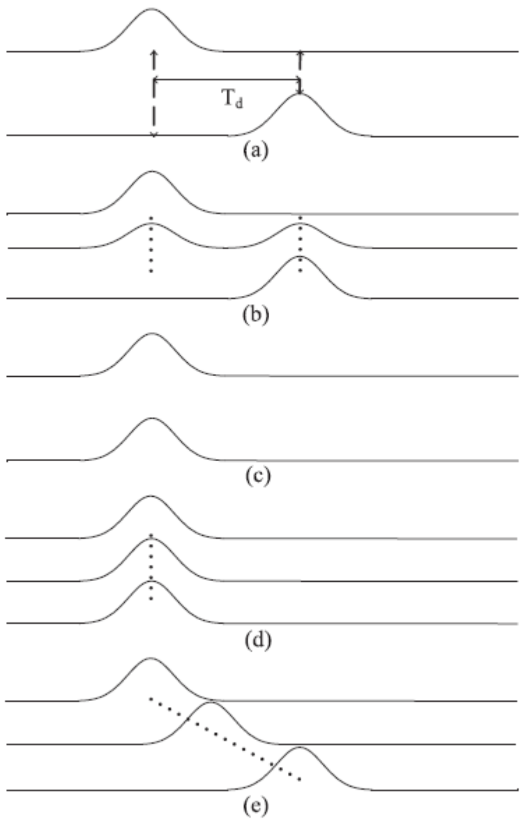


Fig. 1. Interpolation for impulse responses: (a) Basic impulse re- sponses before interpolation; (b) Direct interpolated impulse re- sponses; (c) Impulse responses with removed relative delay; (d) Interpolated impulse responses after removing relative delay; (e) Final

However, computing channel impulse responses by solving wave equation at all sampling time requires large amount of field computation (To model a 10-sec signal at a sampling rate of 12k Hz, over 10^5 impulse responses are needed). Therefore, to reduce the computational complexity, we sample the trajectory at a lower rate, and only compute a smaller number of basic impulse responses interpolated impulse response by field computation, based on which, the intermediate impulse responses, required by (3), can be calculated by interpolation. In such case, for the previous example, assumed that the source is moving at a constant speed of 5 m/s, we can sample the range trajectory at 5 Hz. Only 50, instead of 10^5, impulse responses are calculated by field computation.

Directly interpolation on the basic impulse responses is problematic, especially when the sampling rate of the trajectory is much smaller than that of the signal and the structure of impulse response changes significantly due to the movement. As shown, in Fig.1 (a), due to the movement in range by 1 m, the relative delay between the first arrivals of the two impulse responses is about $T_d = 0.67$ ms, which results in approximately 8 intermediate samples for sampling rate at 12k Hz. The intermediate impulse

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

responses interpolated, probably with two multipath components that corresponds to the arrivals of the basic impulse responses, will not contain any multipath component at the intermediate samples where the arrival should exist, as shown in Fig.1 (b). In order to achieve valid interpolation results, we removed the relative delay Td between these two impulse responses before interpolation, shown in Fig.1 (c). We also need to make sure that the sampling rate of the trajectory is high enough so that the basic impulse responses of adjacent samples have similar multipath structures after removing the relative delays. In this case, we can achieve accurate interpolation results based on impulse responses of several neighboring samples, as shown in Fig.1 (d). In this work, we use the local B-spline for interpolation. After interpolation, the relative delay of each intermediate impulse response, which is estimated according to Td, is added (Fig.1 (e)).

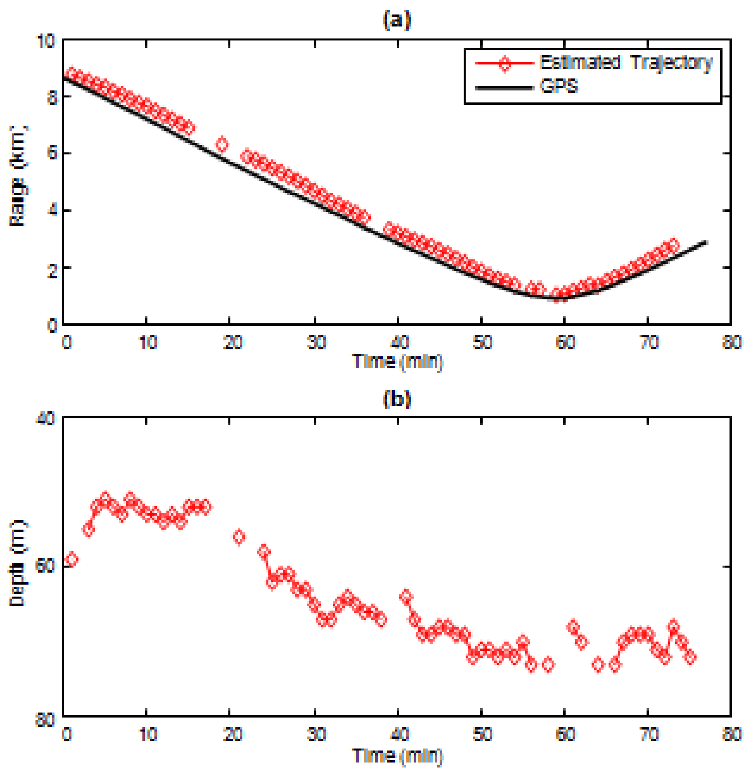
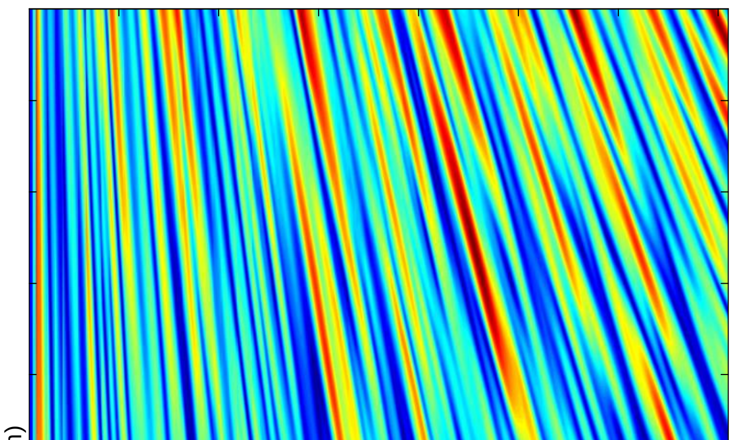


Fig. 2. (a) Range trajectory; (b) Depth trajectory.



ISSN: 0971-1260

Vol-22-Issue-17-September-2019

Fig. 3. Acoustic pressure spectrogram over the $10-710$ Hz for a range of $1.8-2.4$ km (the source depth and the receiver depth are fixed at 54 m and 94.125 m, respectively.)

Fig.3 shows the frequency spectrum we obtained using KRAKEN program from a range of 1.8 km to 2.4 km for fixed source and receiver depths. From this figure, the acoustic waveguide invariant β is about 1.1, which is evaluated according to [17]:

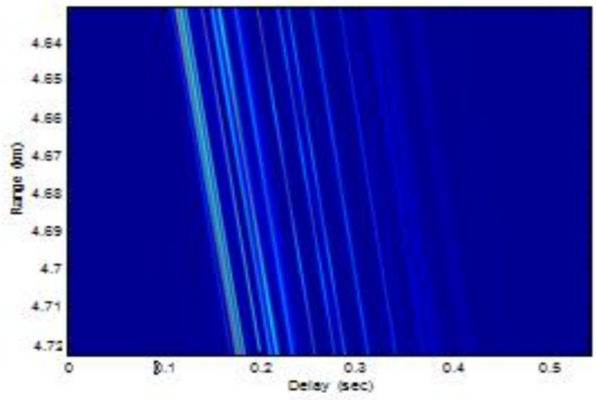
$$\Delta f / f = \beta(\Delta r / r) \tag{5}$$

where ($\Delta f / f$) and ($\Delta r / r$) are, of striations of constant sound intensity, the rate of change of the phase along the waveguide and the envelope group delay, respectively. This reasonable value of [β] further verifies the reliability of the field computation.

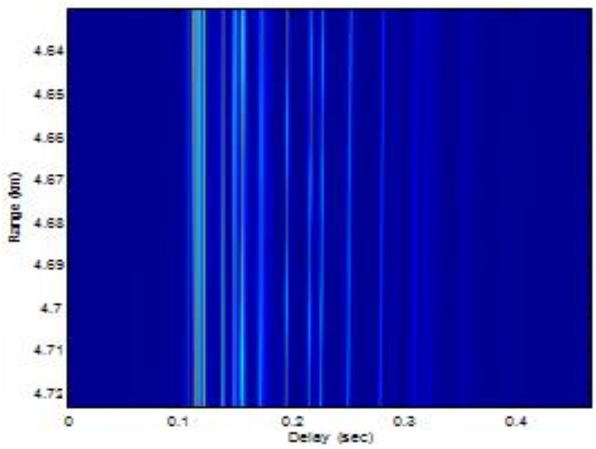
To ensure the impulse responses we generated have enough length to cover all the multi-path components and the propagation time, we computed the frequency responses in a band- width from 0 Hz to 750 Hz with an increment of 0.125 Hz, based on which the inverse Fourier synthesis were computed to produce 8-sec impulse responses.

ISSN: 0971-1260

Vol-22-Issue-17-September-2019



(a)



(b)

Fig. 4. Computed basic channel impulse responses at different source ranges before interpolation: (a) with relative propagation de- lay; (b) without relative propagation delay (Horizontal axis is the delay in second and the vertical axis is range in km.

3. NUMERICAL RESULTS

In this work, we apply the proposed simulation approach to model the data received by the VLA in the SWellEx-96 Event S5 experiment. The impulse responses for different source positions, as required in (3), were generated according to the estimated positions in Fig.2.

Fig.4(a)(b) show the impulse responses computed before interpolation (with and without relative propagation delay), which are obtained at different source ranges from 4.631 km to 4.723 km, corresponding to a 40-sec movement of the source in the experiment. In the figures, the length of each impulse response is about 0.5 sec, which contains all the significant multi-path components. We find that with a range resolution of 1 m as shown in Fig. 4(b), there is no significant change of impulse response structure, which will provide accurate interpolation results.

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

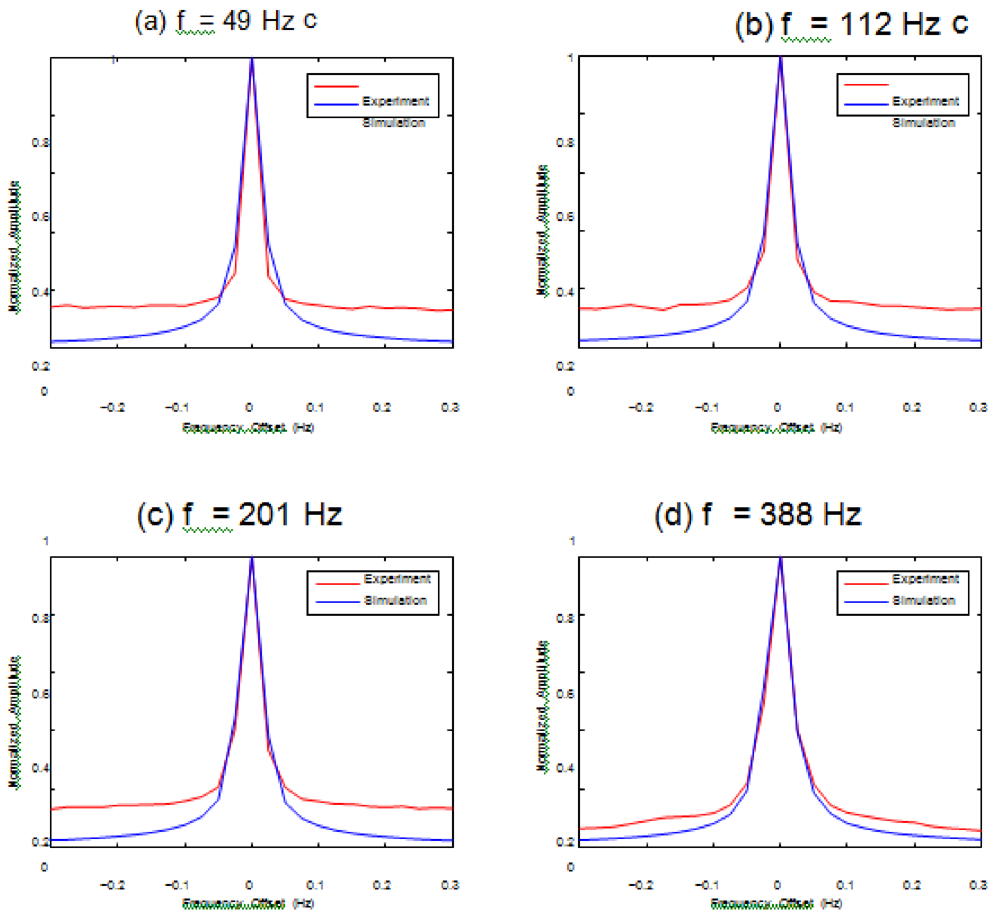


Fig. 5. Averaged Doppler spreads for different carrier frequencies f_C obtained from the proposed simulator and the experiment: (a) $f_C = 49$ Hz; (b) $f_C = 112$ Hz;(c) $f_C = 201$ Hz;(d) $f_C = 388$ Hz.

Fig.5 compares the averaged Doppler spreads for different carrier frequencies f_C obtained by the proposed simulator with those of the experiment, both of which are obtained by averaging the frequencies spectrum over snapshots and receiver hydrophones. For both the simulation and the processing of the experimental data, we used 40 sec snapshot to produce frequency spectrum with a resolution of 0.025 Hz. Before averaging the frequency spectrums of all the snapshots, the frequency shifts of each carrier frequency are compensated so that each carrier frequency has the maximum amplitude within a frequency bandwidth $[f_C - 1, f_C + 1]$ Hz. From Fig.5, we can see that the modeled Doppler spreads are well matched to those of the experimental data. The only mismatch is due to the noise in experimental data, which was not considered in the simulation. As in Fig. 5(d), this mismatch between the modeled and the experimental curves decreases as the signal to noise ratio increases.

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

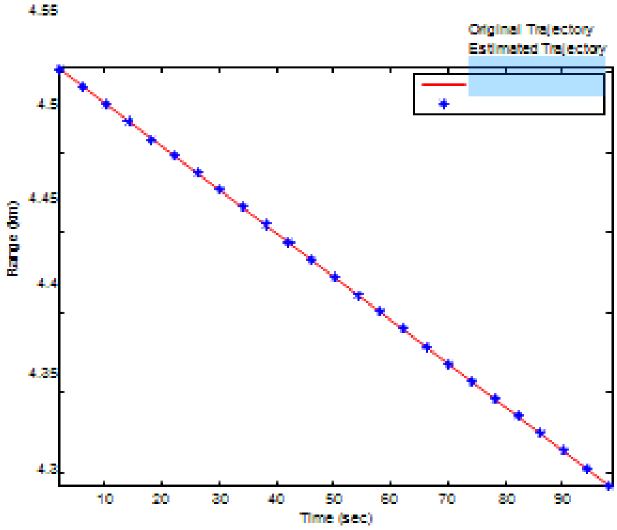


Fig. 6. Original range trajectory used for modeling the data received by the VLA and the estimated trajectory obtained by using a multi- frequency Bartlett matched-field processor.)

With the knowledge of environment parameters and the position of the receiver hydrophone array, the VLA, in the SWellEx-96 experiment, we used our proposed simulation approach to model the data received by the VLA from a moving source, which transmit- ted acoustic signal at the first 13 tones as used in the experiment. The acoustic source was at a depth of 65 m and moved according to a range trajectory as we defined, which is very similar as the experiment. A multi-frequency Bartlett matched-field processor [18] was applied to the modeled received data for source localization. Fig.6 shows that the estimated range trajectory obtained by the multi- frequency Bartlett matched-field processor is well matched to the original range trajectory as we defined for source movement.

4. NUMERICAL RESULTS FOR AN EXPERIMENT IN THE INDIAN OCEAN

Fig.8 (a) shows the estimated impulse responses obtained by employing a channel estimator [5] to the experimental data. It is observed that there are some fluctuations of delay for each multipath component, which have similar shape for all the multipath arrivals, e.g., if the delay of the first arrival at a time instant increases, the delay of later arrivals will also increase, and vice versa. This feature is due to the fluctuation of the source movement in range instead of in depth, which will result in opposite shape for adjacent multipath arrivals. Therefore, in the simulation, the source is assumed to be moving at a sinusoidal time varying speed plus a constant speed.

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

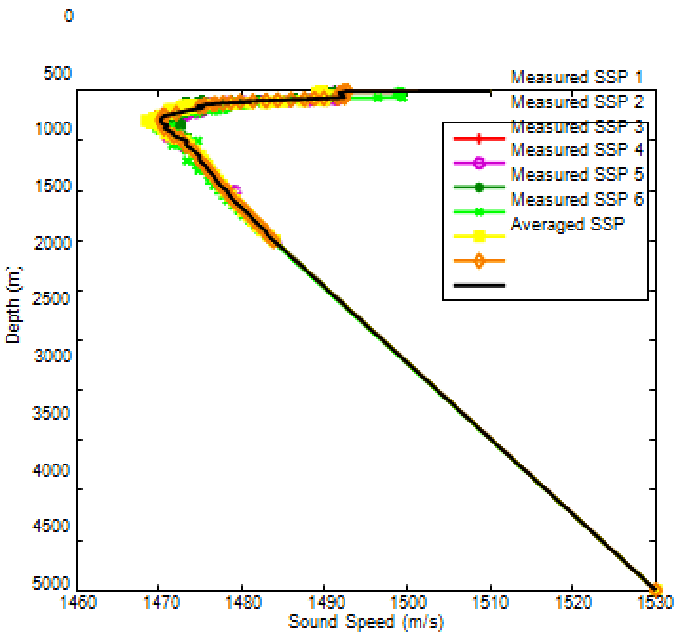


Fig. 7. Measured and averaged SSP for the Pacific Ocean experiment.

The range of the source can be characterized by:

$$r(t) = r_o - v_o t + \alpha \sin(2\pi t/T), \qquad (6)$$

where $r_0 = 42.16$ km, v_o is the constant part of the speed at 6 m/s and $\alpha = 0.6$. The variation period of the speed is set at $T = 10$ sec. The moving trajectory is sampled at every 0.5 sec, which gives basic impulse responses obtained by computing the eigen arrays using BELLHOP program for approximately every 3 m. This ensures that there is no significant change of the multipath structure for neigh- boring samples. Simulated data is generated for a period of about 6 min to compare with the experimental data.

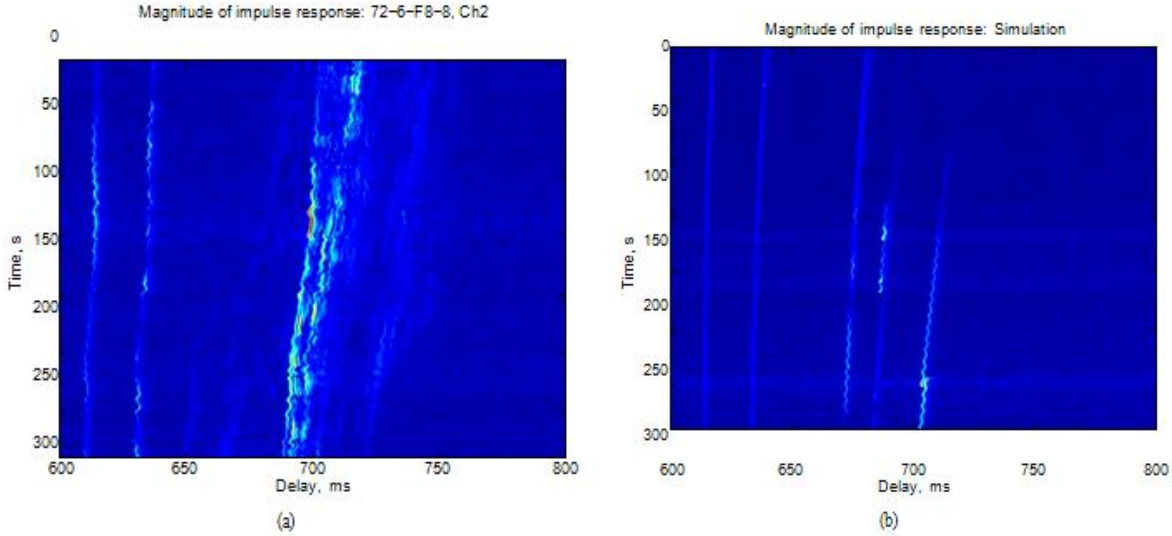


Fig. 8. Estimated Impulse responses for a period of 6 min from: (a) real experimental data; (b) simulated

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

data.

Doppler spread spectrums of the simulated data are well matched to those of the experimental data. The simulated signal can be used in matched field processing for source localization, as the real signal can do. It is also shown that the simulator can effectively take account for the motion fluctuations.

5. CONCLUSION

In this paper, we proposed a unified underwater acoustic channel simulator for communication, which can be used to model the signals in arbitrary scenario for arbitrary motion. It is useful for testing performance of new techniques before conducting field experiment and comparing different communication system in the same environment. Doppler effect is taken into account in a realistic manner by introducing the motions of source and receiver and the complexity of simulation is reduced by performing valid interpolation of impulse responses. The reliability of this simulator is successfully demonstrated by comparing the simulated data to the real experimental data in both shallow water and deep water scenarios. We have shown that

REFERENCES

[1] M. Stojanovic and J. Preisig, "Underwater acoustic communication channels: Propagation models and statistical character- ization," IEEE Commun. Magazine, vol. 47, no. 1, pp. 84–89, 2009.

[2] T.H. Eggen, A.B. Baggeroer, and J.C. Preisig, "Communication over Doppler spread channels. Part I: Channel and receiver presentation," IEEE J. Oce. Eng., vol. 25, no. 1, pp. 62–71, 2000.

[3] B. S. Sharif, J. Neasham, O. R. Hinton, and A. E. Adams, "A computationally efficient Doppler compensation system for underwater acoustic communications," IEEE J. Oce. Eng., vol. 25, no. 1, 2000.

[4] W. Li and J.C. Preisig, "Estimation of rapidly time-varying sparse channels," IEEE J. Oce. Eng., vol. 32, no. 4, pp. 927–939, 2007.

[5] D.B. Kilfoyle and A.B. Baggeroer, "The state of the art in underwater acoustic telemetry," IEEE J. Oce. Eng., vol. 25, no. 1, pp. 4–27, 2000.

[6] M.B.Porter, "The KRAKEN normal mode program(Draft)," Navel Research Laboratory, Washington DC, 1992.

[7] M.B. Porter and H.P. Bucker, "Gaussian beam tracing for computing ocean acoustic fields," J.Acoust. Soc. Am., vol. 82(4), pp. 1349–1359, Oct. 1987.

[8] A. Essebbar, G. Loubet, and F. Vial, "Underwater acoustic channel simulations for communication," Proceedings of OCEANS'94. Vol. 3. IEEE, 1994.

[9] X. Geng and A. Zielinski, "An eigenpath underwater acoustic communication channel model,"

ISSN: 0971-1260

Vol-22-Issue-17-September-2019

in OCEANS’95. MTS/IEEE. Challenges of Our Changing Global Environment. Conference Proceedings. IEEE, vol. 2, pp. 1189–1196, 2002.

[10] Michael B. Porter, “The time-marched fast-field program (FFP) for modeling acoustic pulse propagetion,” The J. Aco. Soc. America, vol. 87(5), pp. 2013–2023, May 1990.

[11] M. J. Buckingham, “Ocean-acoustic propagation models,” J. Acoustique, vol. 3, pp. 223–287, 1992.

[12] Z.H. Michalopoulou, “Matched-impulse-response processing for shallow-water localization and geoacoustic inversion,” The J. Aco. Soc. America, vol. 108, pp. 2082, 2000.

[13] M. Porter, “Bellhop gaussian beam/finite element beam code,” Available in the Acoustics Toolbox, http://oalib. hlsresearch.com/Rays.

[14] J. Murray and D. Ensberg, “The SWellEx-96 Experiment,” http://www.mpl.ucsd.edu/ swellex96/.

[15] T. Chen, C. Liu, and Y.V. Zakharov, “Matched-phase co- herent broadband matched-field processor using phase descent search,” in Tenth Eur. Conf. on Underwater Aco., ECUA, 2010.

[16] C. Liu, T. Chen, and Y.V. Zakharov, “Source localization us- ing sparsity based iterative adaptive beamforming,” in Tenth Eur. Conf. on Underwater Aco., ECUA, 2010.

[17] H.C. Song, WA Kuperman, and WS Hodgkiss, “A time- reversal mirror with variable range focusing,” The J. Aco. Soc. America, vol. 103, pp. 3234, 1998.

[18] A. B. Baggeroer, W.A. Kuperman, and H. Schmidt, “Matched field processing: Source localization in correlated noise as an optimum parameter estimation problem,” The J. Aco. Soc. America, vol. 83, no. 2, pp. 571–587, 1988.