Towards Robust Adaptive Acoustic Communications

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Abstract - **Modulation and detection techniques used for underwater acoustic communications include phase coherent (PSK and QAM) and noncoherent (FSK) techniques. The objective of this work is the development of algorithms that can select the best technique for a given channel. The system of interest operates in shallow water over several kilometers in the 10-15 kHz band. A wideband probe is used at the beginning of each data packet to measure the multipath and the Doppler spread, as well as the SNR. The choice of modulation is based on these measurements. The spread factor of the channel determines whether phase coherent communications are possible. If so, an equalizer is employed to combat any intersymbol interference. If the channel varies too rapidly, noncoherent signaling is chosen. Guard intervals are used to eliminate the effect of multipath. Computer simulations and theoretical analysis are presented to quantify the performance of candidate modulation methods for varying channel parameters.**

I. INTRODUCTION

For the past several decades, underwater acoustic telemetry has relied on the use of FSK modulation and noncoherent demodulation for data transmission. This type of signaling generally provides robust performance in the presence of channel time dispersion (multipath) and fading, but is hampered by its low bandwidth efficiency. The development of powerful DSP processors during the past decade has made it possible to implement complex adaptive equalizers that can be used in conjunction with phase coherent modulation and demodulation (modem) techniques, such as PSK and QAM, to achieve significantly higher bandwidth efficiencies. However, phase coherent modulation techniques are more sensitive to the effects of channel multipath and fading and, as a result, their performance degrades more rapidly under severe channel distortion.

The problem that is being addressed in this paper is the development of algorithms that select the most appropriate modem techniques from a class of modem methods, based on measurements of the underwater acoustic channel characteristics. The modem techniques include both phase coherent techniques such as PSK and QAM, and noncoherent techniques that employ FSK modulation.

Communication systems that adaptively select modulation formats have been extensively studied for wireless radio channels (e.g., [1]) and have also been considered for underwater acoustic channels. An underwater acoustic multimodulation system described in [2] has a capability to use three modulation formats: PSK for vertical channels, and chirp and frequency hopping for horizontal channels. This system is not adaptive, in the sense that modulation selection is made by an operator. A remotely controlled unit is then programmed via an acoustic link. More recently, adaptive modulation methods for underwater acoustic communications have been suggested in [3]. In this reference, an experimental system is described in which a handshaking protocol is used to adaptively set the signal power on a packet-by-packet basis, but the system is based on a single FSK type modulation format. Our study is aimed towards the development of a multimodulation system in which the choice of a modulation method and its parameters is made autonomously, based on the instantaneous measurements of signal quality.

General approaches to modulation selection are discussed in Sec.II. In Sec.III. a model of a time-varying multipath channel is developed. Performance of noncoherent modulation/detection techniques is addressed in Sec.IV. and performance of coherent modulation/detection in Sec.V. Finally, concluding remarks are given in Sec.VI.

II. GENERAL APPROACH TO MODULATION SELECTION

Our approach is based on the assumption that data is transmitted in packets of some nominal size, and the communication link between the transmitter and the receiver is half duplex. Each packet of information symbols is preceded by a header, which includes a channel probe that is used for measuring the characteristics of the channel. Based on such a measurement, the receiver decides on the appropriate modem technique and corresponding data rate. This information is conveyed to the transmitter for use in the transmission of the next packet. Although changes in modulation and data rate may be made on a packet-by-packet basis, it is anticipated that in a static environment, where the transmitter and receiver platforms are fixed, changes in the type of modulation/demodulation technique will be very infrequent. However, if one or both platforms are moving, it is expected that changes in the type of signaling method used for data transmission will be more frequent.

Two types of modem techniques are considered, namely, phase coherent and nonchoherent techniques. In the case of noncoherent modulation and demodulation, we consider coded M-ary FSK (MFSK) with square-law, soft-decision decoding and detection. In particular, Hadamard-coded M-ary FSK is considered. As an enhancement to this modem, we also consider the introduction of a nonbinary outer code, which has the capability of providing an additional 5dB to 6dB coding gain in a Rayleigh fading channel. In such a modem, the parameters that can be adapted to the channel characteristics include the data rate, through control of the amount of code redundancy and the channel bandwidth, and the selection of the time guard bands.

In the case of phase coherent modulation, the types of signal modulation that are considered are PSK and QAM. A convolutional code of rate 1/2 may be included with phase coherent modulation. The receiver employs a decision-feedback equalizer (DFE), which is initialized at the beginning of each packet by a training sequence. The equalizer adaptation algorithm is taken from the RLS family.

The choice of the type of modulation, phase coherent or noncoherent, is based on the channel parameters extracted from the channel measurement. The channel probe signal is a wideband signal that provides a measurement of the channel scattering function from which we extract the channel multipath response and the Doppler spread of the resolvable multipath components. The channel is modeled as a tapped-delay-line filter with randomly time-varying coefficients. The value of the Doppler spread is related to the rate of time variations in the channel response. If the channel impulse response is varying rapidly, the adaptive equalizer must also have the capability of tracking the rapid time variations. Hence, the degree to which the equalizer can obtain a reliable estimate of the intersymbol interference (ISI) caused by the multipath is determined by the spread factor of the channel, defined as the product between the Doppler spread B_d and the multipath spread T_m. This factor must be of the order of 0.001 or smaller to ensure that the adaptive DFE obtains a reliable estimate of the ISI. Hence, the channel spread factor of the primary (large energy) resolvable multipath components and their corresponding total signal power will be used to decide on whether or not phase coherent communications would provide reliable transmission and the corresponding predicted error rate performance.

When channel time variations and multipath conditions preclude the use of coherent modulation, the receiver determines that coded noncoherent MFSK will be the method for transmitting and receiving the data reliably. Based on the channel multipath and Doppler spread, the receiver determines the guard bands, and the amount of redundancy, hence the data rate, that will be needed to achieve the desired level of performance.

If the Doppler spread is below the critical value, the task of the adaptive modem is to decide on the exact modulation format. This decision includes the choice of the modulation and coding method, as well as the choice of the bit rate. The resulting symbol rate plays a major role in determining the performance of an adaptive equalizer. The information gathered from the channel probe is used to determine the parameters of the adaptive equalizer. In particular, once the symbol rate is chosen, the multipath spread determines the number of equalizer taps, while the Doppler spread influences the choice of the tracking parameters.

III. CHANNEL MODELING FOR SIMULATION

The Doppler power spectrum of the channel can be modeled either experimentally or theoretically. There are several theoretical models that are widely accepted in the theory of digital communications over time-varying channels. These models are the Jakes' model and the auto-regressive (AR) models. The Jakes' model [4] is one of the most commonly used models for describing wireless radio channels. A second-order AR model has been considered as a good match for the mobile radio channel [5]. These models provide an analytic representation of the Doppler power spectrum of the channel. We consider for purposes of simulation a second-order AR model.

A test channel may be defined to reflect the geometry of a system. The system of interest to the present study is a shallow water network with bottom-mounted nodes separated by distances of several kilometers. The channel is characterized by a direct and a number of reflected paths. The relative delay between the longest propagation path and the direct path defines the multipath spread of the channel, which can be evaluated using a simplified expression

$$T_m(n) \approx \frac{2n^2h^2}{cr}, \; nh/r << 1 \tag{1}$$

where n is the number of surface reflections. Let us consider a test channel with a depth h=50 m and a range r=5 km. The speed of sound is taken to be c = 1500 m/s. Using practical spreading, and the absorption coefficient for the center frequency of 12.5 kHz, the average path amplitudes, calculated from the channel geometry are 1, 0.5, 0.08, 0.005, etc. Even though the loss due to reflection is neglected in this calculation, the path energy decays rapidly with increasing travel distance. For clarity, we define the test channel as the channel containing the direct and one surface-reflected path. The multipath spread in this case is $T_m = 2h^2/cr$=0.67 ms. The two paths are taken to be independently time-varying, and each is modeled using a second-order Gauss-Markov model, with a specified Doppler spread. Fig.1 shows the simulated characteristics of the test channel. The direct path has the Doppler spread of 0.001 Hz, while the surface-reflected path has the Doppler spread of 1 Hz. The Doppler spread is defined as the 3 dB bandwidth of the Doppler power spectrum of the complex path gain fading process.

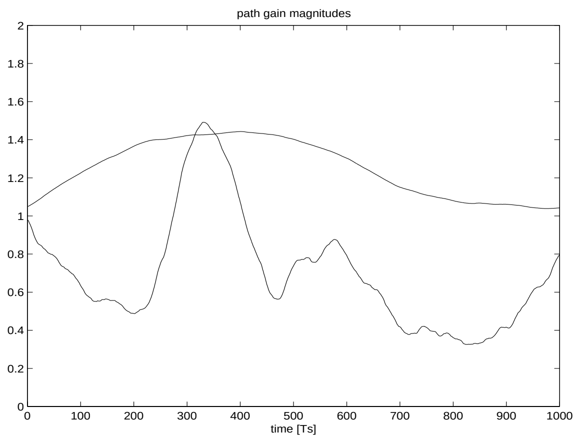

Figure 1. Simulated characteristics of the test channel. Doppler spread is 0.001 Hz on the direct, and 1 Hz on the surface-reflected path. Ratio of average path amplitudes is 1/2. Relative path delay is 0.67 ms, corresponding to one surface reflection at depth 50 m and range 5 km. Channel sampling rate is 600 samples/s. Second-order Gauss-Markov model with critical damping is used for each path.

IV. PERFORMANCE OF CODED MFSK MODULATION WITH NONCOHERENT DETECTION

In this section we evaluate the performance of Hadamard-coded MFSK in a Rayleigh fading channel model which consists of two independently fading paths with a differential path delay of T_m seconds. The first path has a power level of σ_1^2 and the second path has a power level of σ_2^2. The power levels of the two paths are normalized so that

$$\sigma_1^2 + \sigma_2^2 = 1 \tag{2}$$

The MFSK tones have duration T which is selected such that $T >> T_m$. To completely eliminate ISI, we employ a time guard band equal to the multipath spread T_m. As a result, a certain amount of signal energy is lost due to the time guard band. The parameter that provides a measure of the percentage of lost energy is

$$\rho = 1 - \frac{T_m}{T} \tag{3}$$

The error probability for the coded MFSK signals is upper bounded as [6]

$$P_M \leq 2^k P_2(L) \tag{4}$$

where

$$P_2(L) = \left(\frac{1-\mu}{2}\right)^L \sum_{k=0}^{L-1} \binom{L-1+k}{k} \left(\frac{1+\mu}{2}\right) \tag{5}$$

and

$$\mu = \frac{2\gamma_b R_c}{2(1+\gamma_b R_c)} \tag{6}$$

Here, γ_b is the average SNR per bit, $R_c = k/n$ is the code rate and $L = d_{min}/2$ is the diversity in the code with d_{min} as the minimum distance.

Below, we illustrate the results for the Hadamard H(20,5) code, which has a minimum distance d_{min}=10 and, hence, provides an order of diversity equal to $d_{min}/2$=5. The codeword error probability is plotted in Fig.2 for σ_2^2/σ_1^2= 0.7, with ρ as a parameter.

We observe from the graph that the energy loss due to the use of time guard band is very small. By repeating this computation for different values of σ_2^2/σ_1^2, we found that the effect of scaling the energy ratio σ_2^2/σ_1^2 of the two paths has a negligible effect on performance. The error probability decreases as the reciprocal of the SNR per bit raised to the fifth ($d_{min}/2$) power. A codeword error probability of 10^{-6} is achieved with an SNR per bit of approximately 21.5 dB.

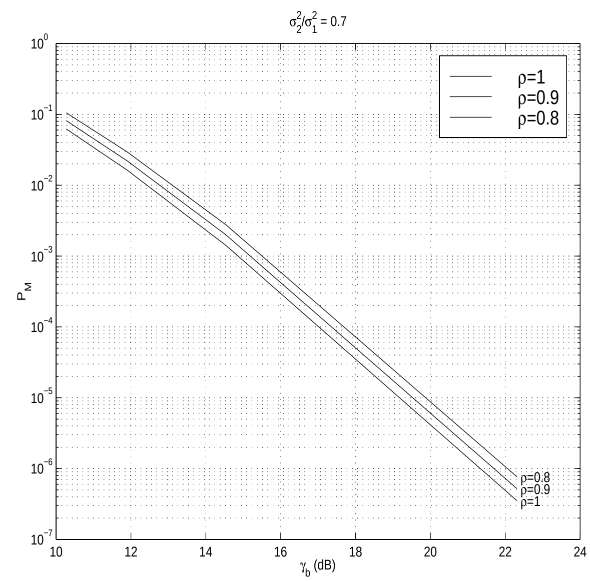

Figure 2. Bit error rate for Hadamard-coded MFSK with a two-tap channel model. Path power ratio is 0.7.

The performance of the Hadamard-coded MFSK modem can be significantly enhanced by the use of an outer code, which has the effect of increasing the minimum distance of the overall concatenated code. In particular, let us consider the use of a dual-5, nonbinary convolutional code as described in [6]. This is a rate 1/2 code which results in a factor of four increase in the minimum distance of the overall code. Since the Hadamard H(20,5) code has a minimum distance of $d_{min} = 10$, the concatenated code has a minimum distance of 40. Hence, it provides an order of diversity equal to 20, when soft-decision decoding is employed at the receiver.

Fig.3 illustrates the performance of the concatenated code compared with the performance of the H(20,5) code. Also shown in the figure is the performance obtained by repeating the H(20,5) codewords twice, denoted as 2H(20,5), which is an alternative method to the use of a concatenated code. However, the repetition code, although it has the same code rate as the concatenation of the dual-5 code with H(20,5) code, yields a minimum distance of 20. Hence, the order of diversity achieved by the repetition code is only one-half of the diversity achieved by the concatenated code. Note that at an error rate of 10^{-6}, the concatenated code yields an improvement of about 8 dB over the H(20,5) code and about 4 dB over the repetition of the H(20,5) code.

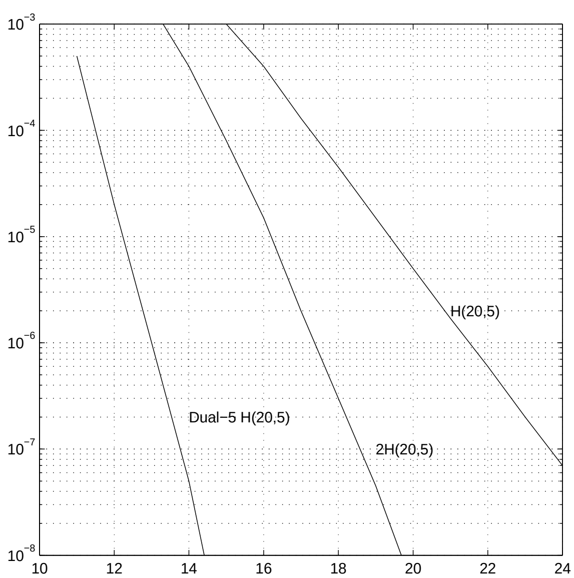

Figure 3. Performance of H(20,5) code, its repetition, and its concatenation with a dual-5 nonbinary convolutional code.

V. PERFORMANCE OF COHERENT MODULATION/DETECTION

To assess the performance of coherent modulation/detection methods for a variety of channel conditions and system parameters, a simulation has been conducted. The system performance is evaluated on the test channel defined in Sec.III.

Linear modulation with variable symbol rate is used. The signals are filtered at the transmitter using a raised cosine filter with a roll-off factor of 0.25. After passing through the channel, AWGN is added to the simulated signals, with a specified SNR. The channel is time-varying, but its statistical parameters (multipath and Doppler spread) are fixed. The receiver uses an adaptive, fractionally-spaced DFE. The DFE coefficients are updated using an RLS algorithm, jointly with a second-order digital phase-locked loop (PLL).

A. *Equalizer Performance as a Function of Symbol Rate*

The task of the modem is to decide on the best modulation and data rate for a given channel. Setting the modulation format to QPSK, we explore the choice of the data rate in more detail. The test channel is characterized by the multipath spread of 0.67 ms and a Doppler spread of 0.001 Hz and 1 Hz on the two paths. These values represent the physical channel parameters that cannot be changed. Multipath spread causes ISI, while Doppler spread causes time-variation. The received signal thus experiences frequency-selective fading, or time-varying ISI. The extent of these distortions, however, can be controlled by the choice of signaling rate. To illustrate this fact, we look at three possibilities: signaling at the rate of 300 symbols per second (s/s), 3000 s/s, and at the maximum rate supported by the system bandwidth, 5000 s/s. The first choice will result in very little ISI because the multipath spread is less than the symbol duration, $T_m = 0.2T$. The other two choices result in non-negligible ISI. At 3ks/s, the ISI spans 2 symbols, while at 5 ks/s the signal experiences ISI which spans 4 symbols. At the same time, the channel time-variation observed over the duration of a single symbol decreases as the symbol rate increases.

Fig.4 shows the performance of the RLS/DFE on the test channel. The receiver parameters are listed in the figure. N denotes the number of $T/2$ fractionally-spaced feedforward taps, M denotes the number of feedback taps. L denotes the forgetting factor of the RLS algorithm; K_{f1} and K_{f2} denote the PLL proportional and integral tracking constants, respectively. Pe denotes the fraction of symbol errors in the data block. The output SNR is measured from the estimated data symbols at the equalizer output after convergence has been established. Although there is very little ISI in this channel, the receiver performance is poor. The reason for such performance is the receiver's inability to track the time-variation of the channel.

A system that employs a signal selection algorithm can measure the Doppler spread of the channel, or simply observe that the estimated phase exhibits rapid changes over the data block. Such a system can choose to transmit at a different (higher) data rate. Fig.5 shows the

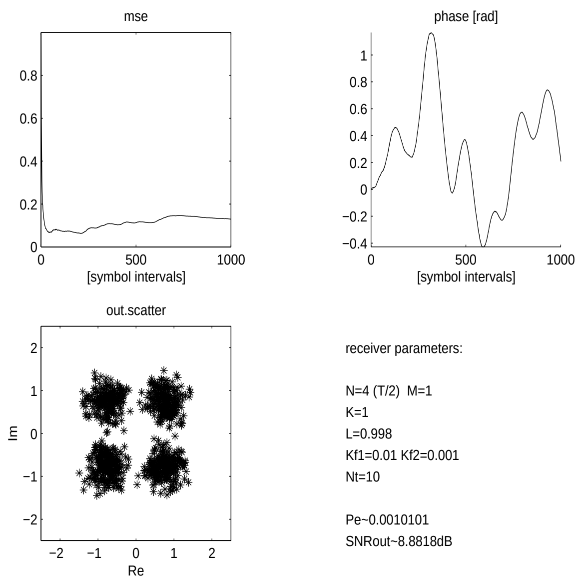

Figure 4. RLS/DFE performance on the test channel. Symbol rate R=300 s/s. Input SNR=20dB.

results of system simulation at the rate of 5000 s/s. The receiver performance shows an improvement in this case. The performance at 3000 s/s is similar. There are no errors in the data block. Note that the *physical* channel has not changed; the only parameter that has changed is the symbol rate R. Because the rate is higher, the multipath spread encompasses a greater number of symbols, and the receiver must employ an equalizer. M=4 taps suffice in the feedback section of this equalizer.

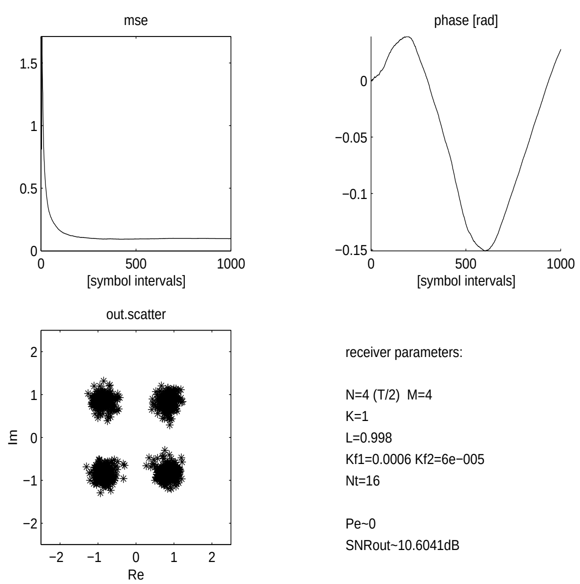

Figure 5. RLS/DFE performance on the test channel. Symbol rate R=5000 s/s. Input SNR=20dB.

In general, the equalizer should choose the number of taps $M = \lceil T_m/T \rceil$. Note also that the choice of phase tracking constants is influenced by the symbol rate. These constants need to be decreased proportionally to the increase in the symbol rate.

By further increasing the data rate, the channel changes would become negligible, and the ISI would become the major factor that limits the system performance, both through the noise enhancement and the implementation complexity with associated stability problems of long equalizers. The simulation results illustrate the tradeoff in choosing the signaling rate for a time-varying multipath channel. Limited experimental results that support these observations are documented in [7]. This reference also contains results of a theoretical analysis that quantifies the system performance in terms of the bit error rate as a function of SNR for varying Doppler spreads on a Rayleigh fading channel.

In summary, the modem should choose a signaling rate $R = 1/T$ for which the normalized Doppler spread, $B_d T$, is acceptably low *and* the ISI span, T_m/T, is minimal. If one sets the value of $B_d T$ as the system design parameter, say $B_d T = 10^{-4}$, the data rate can be determined after estimating the Doppler spread B_d. For the so-obtained data rate, the system will use the estimated multipath spread, T_m, to determine the equalizer size.

B. *Equalizer Performance as a Function of SNR*

In the above examples, we have investigated the equalizer performance as a function of ISI span and the rate of the channel time variation. The SNR was fixed at 20 dB. This value is apparently high enough to warrant a good performance, at least in the case of 5000 s/s. With descreasing input SNR, the equalizer performance will gradually degrade until the input SNR becomes too low for the equalizer to establish convergence. Below this point, coherent detection using equalization is no longer a feasible communication method.

To determine the range of input SNR for which the equalizer performance is acceptable, a simulation has been conducted using the test channel. The equalizer performance as a function of input SNR is summarized in Fig.6. Shown in this figure are the output SNR curves obtained for varying data rates. The output SNR is the value measured from the sequence of estimated data symbols at the equalizer output prior to decision making. Each output SNR value is obtained by averaging over 1000 independent channel realizations (noise and fading). The results confirm that the equalizer performance is generally better for increasing data rate. As discussed previously, the reason for such behavior is that the effective rate of channel time-variation decreases with increasing signaling rate.

To define what is meant by 'acceptable' equalizer performance, the overall system and its required bit error rate must be considered. If the system uses coding, the the role of the equalizer is to produce a sequence of es-

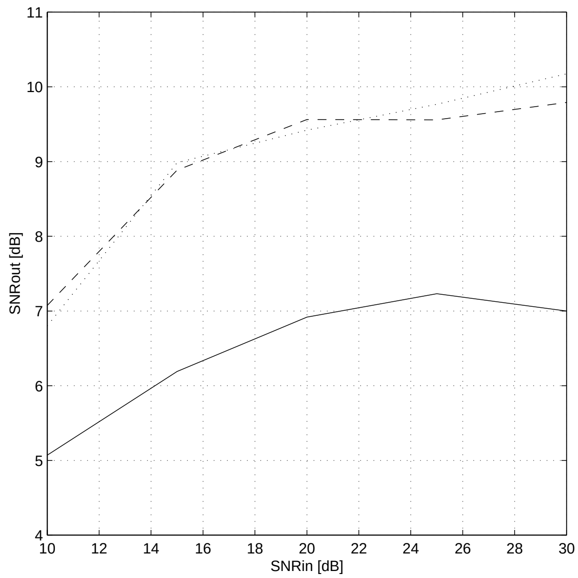

Figure 6. Equalizer performance on the test channel as a function of input SNR. Solid curve represents transmission at 300 s/s. Dashed curve represents transmission at 3000 s/s, dotted at 5000 s/s.

timated data symbols that are reliable enough for use by the decoder. These symbols need not achieve the required bit error rate. Consequently, rather than setting a strict break-down value for the input SNR, we loosely define the useful range of input SNR to include those values for which the equalizer is at least capable of converging. Simulations reveal that this condition is met at the output SNR of about 5 dB or more. Although the receiver performance is very poor, the equalizer does maintain convergence. If coding is employed, the equalizer output may be used by the decoder to extract the coding gain and, thus, recover the transmitted information more reliably. Hence, we may say that the output SNR on the order of at least 5 dB is required for 'acceptable' performance. To relate this value to the input SNR, we must look at the curves of Fig.6. Each point shown on these curves represents an average taken among a number of different noise/fading realizations. The average is taken only among those realizations for which convergence has been established. (In cases when the equalizer is not able to converge, the output SNR cannot be measured meaningfully.) For signaling at 3000 s/s, an input SNR on the order of 10 dB is needed. For signaling at 300 s/s, the rate of failure (relative number of packets for which convergence could not be established) is high (about 0.1 throughout the investigated SNR region). Thus, coherent signaling in this case is not a good choice.

VI. CONCLUDING REMARKS

The analysis presented in this paper aims towards the development of a robust acoustic modem which will employ an algorithm for smart selection of signaling parameters to achieve a good trade-off in quality of performance and data throughput. The first decision that the modem makes is whether to use a coherent or a noncoherent modulation/detection method. This decision is based on the measurement of the channel Doppler spread B_d and multipath spread T_m. If $B_dT_m < 10^{-3}$, the modem may use a coherent signaling method. Otherwise, it resorts to a noncoherent signaling method. Coded M-ary FSK is the noncoherent method of choice. Signal parameters, such as the guard intervals are selected based on the measurement of the multipath and Doppler spread. Coherent methods use PSK or QAM modulation. An equalizer is often needed to suppress the ISI that arises in shallow water channels at higher data rates. Signaling rate $R = 1/T$ is determined to provide a low normalized Doppler spread, e.g., $B_dT = 10^{-4}$. The measured multipath spread is then used to determine the equalizer size.

The performance of coherent signaling is limited by the available SNR and the nature of ISI. To provide reliable signals for subsequent decoding, a DFE operating on a test channel was shown to require at least 10 dB of input SNR. Simulation, as well as analytical results, also show that the equalizer performance improves with increasing symbol rate. The performance of noncoherent methods was investigated analytically. These results show that for the same performance, coherent PSK requires a higher SNR per bit than MFSK. The reason for this is that the MFSK signal has higher diversity.

REFERENCES

[1] S.Sampei, "Applications of Digital Wireless Technologies to Global Wireless Communications," Prentice Hall, 1997.

[2] G.Ayela, M.Nicot and X.Lurton, "New innovative multimodulation acoustic communication system," in Proc. Oceans'94, pp.292-295.

[3] A.Rice and V.McDonald, "Adaptive modulation for undersea acoustic telemetry," in Sea Technology, May 99.

[4] W.Jakes, *Microwave Mobile Communications*, IEEE Press, 1993.

[5] P.Dent, G.Bottomley and T.Croft, "Jakes fading model revisited," *IEE Electronic Letters*, June 1993, pp.1162-1163.

[6] J.G.Proakis, *Digital Communications*, New York: Mc-Graw Hill, 1995.

[7] M.Stojanovic, J.G.Proakis and J.A. Catipovic, "Performance of a high rate adaptive equalizer on a shallow water acoustic channel," *J. Acoust. Soc. Amer.*, vol.100 (4), Pt. 1, pp. 2213-2219, Oct. 1996.

TOWARDS ROBUST ADAPTIVE ACOUSTIC COMMUNICATIONS

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