Performance of high-rate adaptive equalization on a shallow water acoustic channel[a)]

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Multipath propagation in underwater acoustic channels causes intersymbol interference in the transmission of digital communication signals. An increase of the transmission rate on a multipath channel results in longer intersymbol interference, which ultimately limits the performance of a phase-coherent digital communication system. Recent experimental results, however, show a seemingly surprising result: an increase in transmission rate resulted in improved system performance. An explanation for this phenomenon is found in the time variation of the ocean multipath. In strongly fluctuating shallow water channels, higher transmission rates allow for more frequent sampling of the rapidly varying channel, thus resulting in a better tracking capability of the receiver. Experimental results obtained in shallow water show a substantial improvement in performance of QPSK coherent detection over a 1-mile range, as the data rate is increased from 5 to 20 kilobits per second. A theoretical analysis based on stochastic channel modeling supports the experimental observations. © *1996 Acoustical Society of America.*

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INTRODUCTION

Over the past several years there has been an increased interest in underwater acoustic (UWA) communications. This is partly due to the increasing need for wireless underwater communications in various nonmilitary applications such as off-shore oil industry, environmental monitoring, or recovery of scientific data recorded at benthic stations, as well as military applications which call for communication with submarines and autonomous underwater vehicles. These, and the many new applications to be developed, require wireless underwater transmission of digital information at high data rates.

The feasibility of high data rate, phase-coherent transmission through severely band-limited UWA channels has been demonstrated only recently.[1,2] The principles of phase-coherent signal detection given in Refs. 1 and 2 serve as a basis for current state-of-the-art communication link design. The adaptive receivers which achieve bandwidth-efficient UWA communications are based on a method of simultaneous diversity combining, phase recovery and equalization, necessary for combating the effects of spatial and temporal variations of underwater multipath. These receivers were shown to have excellent performance on a class of horizontal, long-range deep and shallow water channels. Current achievements cite data rates up to 1 kilobit per second (kbps) over three convergence zones in deep water (110 nautical miles), and up to 2 kbps over 50 nautical miles in shallow water.[1,2]

As a continuation of work on coherent UWA communications, an experiment was set up in the fall of 1993 to test the performance of the mentioned receiving techniques on medium-range (approximately 1 nautical mile) shallow water channels, at data rates as high as 30 kbps. The medium-range shallow water channel is known to be the most rapidly time-varying of all the acoustic channels.[3,4] Nevertheless, excellent performance results were obtained using adaptive phase-coherent communication techniques. During the course of the experiment it was observed that an *increase* in transmission rate resulted in *better* receiver performance. Such behavior may seem surprising at first since it is known that the most prominent, and indeed the most difficult, aspect of the shallow water channel is its multipath, whose effect on digital transmission is enhanced by increasing the symbol rate. It is for this reason that the majority of the existing communication systems are confined to very low transmission rates.[5] To be explicit, it is known that if the signaling rate is increased on a given multipath channel (or equivalently, the symbol duration is shortened), a larger number of received symbols will overlap, causing longer intersymbol interference (ISI), and thus degrading the receiver performance.

A relatively simple explanation for the phenomenon observed can be found in the time variability of the medium-range ocean channel. Besides the ISI, the channel time variations are another factor affecting the system performance. While long-range channels may be characterized as moderately time-varying, medium-range shallow water channels suffer from the surface variability and may exhibit time variations rapid enough to cause significant degradation in the channel tracking mechanism of the adaptive receiver algorithm which adjusts the receiver parameters every symbol interval. On such channels, higher transmission rates allow for faster sampling of the time-varying channel response, which may ultimately result in better overall receiver performance through more accurate channel tracking.

The purpose of this article is to present the experimental

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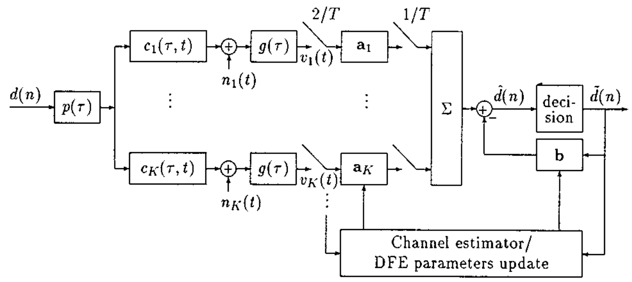

FIG. 1. Baseband system and channel model.

results which demonstrate that signaling at a higher rate can result in improved performance of an acoustic communication link, and to theoretically support this interesting phenomenon. The material is organized as follows. In Sec. I we overview the theoretical background which relates the performance of an adaptive equalizer to the time-variability of the channel, showing, for a particular case of a Rayleigh fading channel, the system tradeoffs that can lead to improved performance when the signaling rate increases. In Sec. II we present the experimental results that assert this fact, and in Sec. III we summarize the conclusions.

I. THEORETICAL BACKGROUND

The equivalent baseband channel and system model we are considering is shown in Fig. 1. The receiver employs a K-sensor array which provides spatial diversity. Each of the K propagation channels connecting the transmitter with the receiver array is modeled as a multipath channel. To simplify the mathematical treatment, in some of the available analyses a UWA communication channel is modeled as quasi-time-invariant (e.g., Ref. 6), and it is consequently assumed that the channel impulse response is perfectly known at the receiver. However, this is not the case in practice when the receiver adaptively estimates the channel response. As it will be shown, the errors in channel estimation, caused by the fact that the channel is time-varying, bear significant impact on the receiver performance. For a rapidly varying channel, the assumption of perfect knowledge of the channel response may lead to an overly optimistic performance estimate. Therefore, we model the UWA channel as a time-varying multipath channel. The impulse responses of the K channels at time t are given as $\{c_k(\tau,t)\}_{k=1}^{K}$, where τ is the delay variable in which the multipath spread is measured.

The receiver structure is that of a multichannel decision-feedback equalizer (DFE). The DFE is used in a variety of time-dispersive fading channels due to its low computational complexity and near-optimal performance.[7] In a multichannel configuration, it provides an efficient method for combating the severe ISI encountered in virtually all of the horizontal UWA channels.[2] Among these channels, which include both deep and shallow water channels, the medium-range shallow water channel is by far the most rapidly varying one. It is known that the critical issue for the equalizer performance in a rapidly varying channel is the tracking capability of the underlying adaptive algorithm. A standard algorithm that has been used for the adaptation of the equalizer coefficients is the RLS algorithm whose tracking properties are determined by the value of the forgetting factor which is used for exponential weighting of the past data. The sensitivity of the receiver performance to the time-variation of the channel is best illustrated in the case of a medium-range shallow water channel, where experimental results have shown that slightest changes in the value of the forgetting factor significantly affect the receiver performance.[1]

Although a vast body of literature has been devoted to the analysis of the multichannel equalizer performance, an assumption commonly made is that the receiver has perfect knowledge of the channel responses at all times. Among the first references on this subject is Ref. 7 where the performance of a multichannel DFE is analyzed for the case of a slowly varying troposcatter channel. Recently, a similar analysis was presented in Ref. 6 for a time-invariant UWA channel model. However, this assumption may not be appropriate for rapidly varying channels, such as medium-range channels, where the performance degradation due to channel estimation errors has to be taken into account. Thus, rather than assuming knowledge of the particular channel realization, we shall only assume knowledge of its statistical properties. Under such a constraint, we wish to characterize the performance of a multichannel DFE, showing its dependence on the channel dynamics.

An in-depth mathematical analysis of the multichannel DFE performance based on stochastic channel modeling when the channel is assumed to be Rayleigh fading is presented in Ref. 8. Below, we summarize the results of this analysis which support our experimental observations. The Rayleigh fading assumption in underwater acoustics is justified for the fully saturated medium-range channels.[3,4] As defined in Ref. 4, a saturated or strongly fluctuating channel is one in which the signal on each of the propagation macro-paths, or rays, splits into a number of resolvable micro-paths. The condition of full saturation occurs when the randomly fluctuating micro-paths are uncorrelated, leading to statistical characterization of the signal traveling on this path as having a Rayleigh distributed envelope and a uniformly distributed phase. The amount of saturation of one channel can be related to its range and frequency, showing that a medium-range high-frequency channel fits within the model of a fully saturated channel where the signal on each macro-path is subject to Rayleigh fading.[3]

For the model shown in Fig. 1, the DFE parameters are optimized using a minimum mean-squared error (MSE) criterion, subject to the constraint that only the sampled channel estimates and their covariance matrices are known. The parameters that need to be determined based on this knowledge are the tap weights of the feedforward equalizer filters and those of the feedback filter. There are K feedforward filters with the tap-weight vectors $\mathbf{a}_k$ operating on the sampled input signals (sampling is performed at twice the maximal signal frequency), and one feedback filter with the tap-weight vector $\mathbf{b}$ operating on the sequence of previously detected symbols. At the equalizer output, an estimate $\hat{d}(n)$ of the transmitted data symbol $d(n)$ is formed, based on which the final symbol decision $\tilde{d}(n)$ is made. The effect of erroneous previous decisions is neglected in the analysis, as its contribution is expected to be much smaller than that of the combined noise, residual ISI and channel estimation errors. It is, however, included in the experimental results presented in the following section.

The signal at the input to the kth equalizer branch is modeled as

$$v_k(t)=\sum_n d(n)h_k(t-nT,t)+\nu_k(t) \tag{1}$$

where $\{d(n)\}$ is the sequence of M-ary data symbols transmitted every T seconds, $h_k(\tau,t)$ is the overall impulse response of the kth channel including any transmitter filtering $p(\tau)$ and receiver filtering $g(\tau)$, and $\nu_k(t)$ is the additive Gaussian noise observed after receiver filtering. The input noise processes $\{n_k(t)\}$ are assumed to be independent white Gaussian processes. For such a model of the input signal, the optimal (in the MSE sense) values of the equalizer tap vectors are a function of the sampled channel response vectors $\mathbf{h}_k(n)$. When the channel is not known, the filter tap weights are determined from the channel estimates $\hat{\mathbf{h}}_k(n)$, which are obtained adaptively on-line, based on the observation of the input signal and the recovered data symbols.

For many practical estimation methods that may be employed, including the MSE estimator, the channel estimates are unbiased and orthogonal to the channel estimation errors $\boldsymbol{\varepsilon}_k(n)=\mathbf{h}_k(n)-\hat{\mathbf{h}}_k(n)$. Under such conditions, the optimal equalizer parameters are shown to depend on the channel estimate in a similar way in which the conventional equalizer parameters depend on the true value of the channel response. However, the MSE between the true and the estimated data symbol,

$$\mathrm{MSE}=E\{|d(n)-\hat{d}(n)|^2\}, \tag{2}$$

is now increased. In the absence of estimation errors, the factors degrading performance are the input noise and the residual ISI.[7] When the process of channel estimation is not perfect, the equivalent signal power, represented by the sampled channel covariance matrix, is reduced by the covariance of the channel estimation error. At the same time, the equivalent noise covariance, represented by the covariance of thermal noise, is increased by the covariance of the channel estimation errors which propagate through the feedback filter.[8]

When the channel is Rayleigh fading, the probability of bit error which characterizes receiver performance can be obtained in closed form. It depends on the value of the channel estimation error covariance. To specify this value, we must focus on a particular channel estimation technique. We accordingly consider a Rayleigh fading process described by a Gauss–Markov model. The model parameters determine the channel autocorrelation properties, and therefore its covariance and the rate of time variation. The model parameters can be measured at the receiver, and we focus on the best case when these parameters are perfectly known. Although not true, this assumption is far more realistic than the assumption of a completely known channel. When the model parameters are known, they can be used to realize the channel estimator as the Kalman filter, which is the optimal estimator for the case of white input noise. Consequently, the performance of the receiver which uses Kalman filtering for channel estimation may be regarded as a bound on the performance of other, less complex receiver algorithms which are more frequently used in practice. In particular, one such algorithm is the RLS algorithm for which the experimental results will be presented.

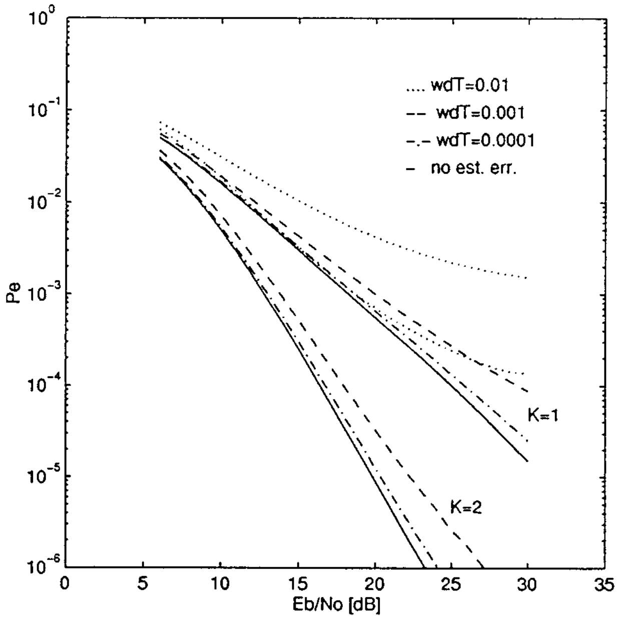

FIG. 2. Single-channel and dual diversity QPSK performance.

In the simplest case of a first-order model, with independent diversity channels, the time-variation of the channel is modeled as

$$c_k(\tau,t+T)=fc_k(\tau,t)+\xi_k(t), \tag{3}$$

where $\xi_k(t)$ is the Gaussian process noise. The fading process is specified by the parameter f which represents the value of the channel autocorrelation function at a delay of one symbol interval. The parameter f is related to the bandwidth ω_d of the Doppler power spectrum as

$$f=e^{-\omega_d T}. \tag{4}$$

Figures 2 and 3 summarize the theoretical performance results for QPSK signaling when the data symbols take values $d(n)=\pm 1 \pm j$ with equal probability. These bit error probability curves are calculated for the case when each of independent diversity channels contains two independently fading propagation paths of equal energy. The transmitter and receiver filter are taken to have rectangular impulse responses of duration T.

In Fig. 2 the probability of a bit error is shown as a function of the input SNR per bit, E_b/N_0, where E_b is the energy per bit and N_0 is the power spectral density of the input noise processes $n_k(t)$. Shown are the curves for single- and dual-channel reception, with the normalized fading rate $\omega_d T$ as the parameter. The solid curve corresponds to the reference case of perfect estimation. As the fading rate increases, the degradation from the reference curve increases. Even as the noise vanishes, a constant, nonzero value of the

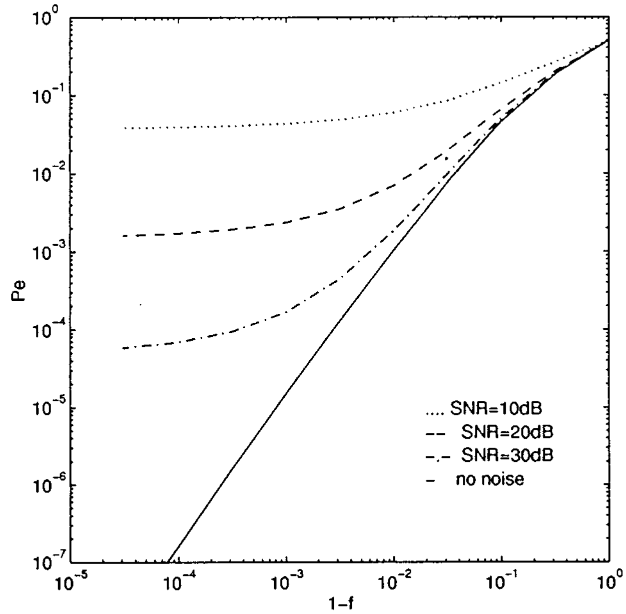

FIG. 3. Single-channel QPSK performance as a function of fading rate.

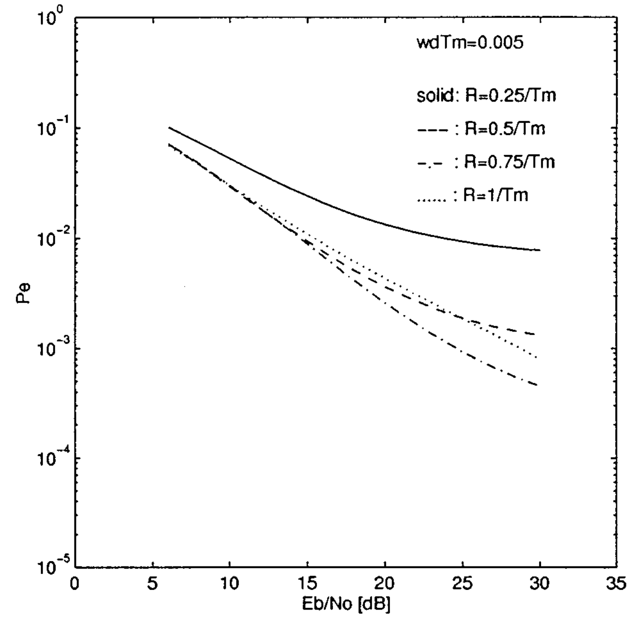

FIG. 4. Single-channel QPSK performance for varying symbol rates.

error probability remains, which is a consequence of imperfect channel tracking caused solely by the time-variations of the channel.

In Fig. 3, the error probability is illustrated as a function of the quantity $(1-f)$, with input SNR as the parameter. For all practical values of fading rate, $1-f \approx \omega_d T$. As it can be expected, at a given value of SNR the performance degrades, i.e., the probability of error increases as the fading becomes faster. Even when the thermal noise is not present, performance is limited at finite values of $\omega_d T$. The solid line represents this limiting or saturation level of the error probability. At high fading rates ($\omega_d T > 10^{-2}$ in this example) estimation errors are seen to represent a major cause for the DFE performance degradation.

From the above considerations we see that the receiver performance on any given channel is determined by the value of the fading rate ω_d, normalized by the signaling rate $1/T$. Let us consider, in the light of such a conclusion, what happens if the signaling rate is increased on a given multipath channel specified by the fading rate ω_d. If we increase the signaling rate $1/T$, the normalized fading rate $\omega_d T$ becomes lower, which effectively reduces the estimation error penalty and results in improved performance. At the same time, however, the multipath spread, as measured in symbol intervals, becomes longer, resulting in an increase in the residual ISI penalty. This point is illustrated in Fig. 4 which shows a set of bit error probability curves as a function of SNR, with symbol rate as a parameter. In this example, the channel spreading factor is fixed at $\omega_d T_m = 0.005$, while the signaling rate is changed. Initially, the performance improves with an increase in the signaling rate from $R=0.25/T_m$ to $R=0.75/T_m$, as the effective, normalized fading rate decreases. However, by further increasing the symbol rate to $R=1/T_m$, the effect of residual ISI becomes dominant, causing the performance to deteriorate. Hence, there exists a tradeoff in choosing the signaling rate for a given time-varying multipath channel. In other words, there exists an optimal value of the signaling rate which results in the least deteriorating effect of combined estimation error and residual ISI penalty. Only if the channel were time-invariant would it be best to choose a low data rate, since it results in least ISI. If the channel is time-varying, this is no longer the case, because signaling at a higher rate allows us to observe the channel more frequently, and thus perform more accurate channel tracking. A general quantitative procedure for selecting the signaling rate to be used on a given channel would consist of two steps. In the first step, extensive measurements would be used to obtain a statistical channel model. For the so-obtained model, characterized by the Doppler spread ω_d and the multipath spread T_m, the bit error probability would then be calculated for varying values of the symbol duration T. Finally, the selection of the data rate would be made for the available SNR.

Further studies of the multichannel DFE performance on a Rayleigh fading channel also revealed the following conclusions which we only state briefly since their mathematical treatment is beyond the scope of this article. Although the use of diversity greatly improves the performance and reduces the residual ISI penalty, it increases the overall estimation error penalty, which can be easily understood from the fact that each independent diversity branch introduces new estimation errors, uncorrelated with those of other branches. Also, it can be shown that the higher level M-PSK modulation schemes are more sensitive to the channel dynamics, i.e., they suffer higher performance degradation than the binary PSK scheme on the same channel.[8]

The theoretical considerations summarized above support the experimental results obtained on a medium-range channel, which we now describe.

II. EXPERIMENTAL RESULTS

The medium-range experiment was a part of the high-rate coherent acoustic telemetry demonstration conducted at

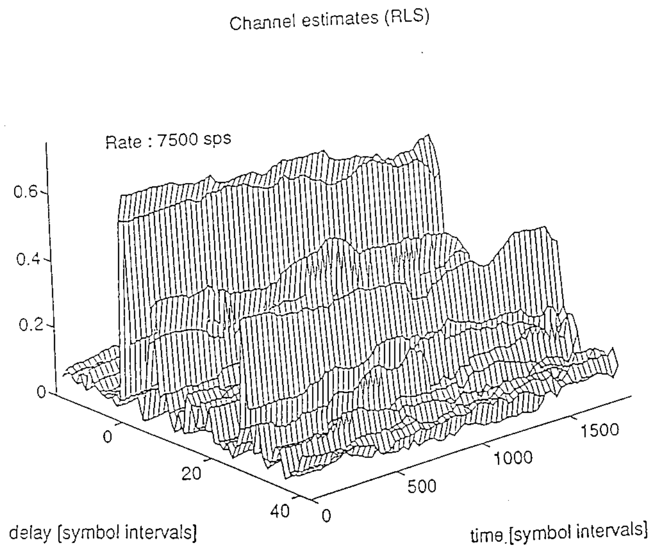

FIG. 5. Ensemble of channel responses.

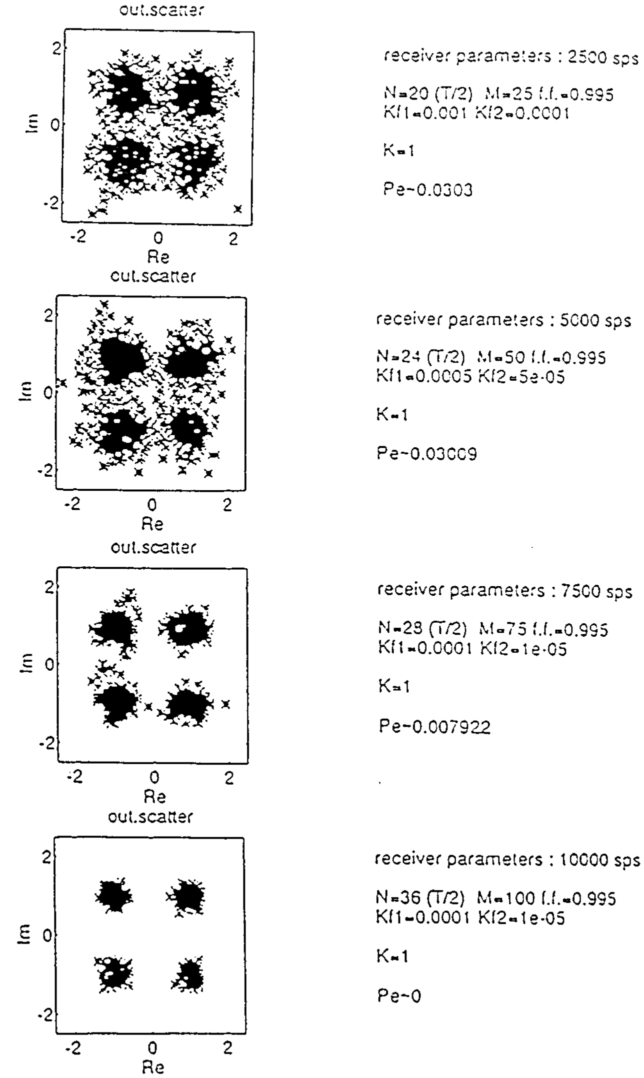

FIG. 6. Receiver performance in single-channel configuration.

the Woods Hole Oceanographic Institution during October 1993. The shallow water channel of the Woods Hole harbor exhibits a long and rapidly time-varying multipath structure. The range of transmission across this channel was approximately one nautical mile in the water depth of about six feet. The carrier frequency of 15 kHz was used with the nominal transducer bandwidth of 10–20 kHz. Three omnidirectional hydrophones (one near the bottom and two near the surface) served as receiving elements. The QPSK signals were derived from pseudo-noise binary sequences shaped by a raised-cosine transmitter filter with roll-off factor 0.5. The signals were transmitted at rates of 2.5, 5, 7.5, and 10 kilosymbols per second (ksps).

Figure 5 shows a representative set of channel responses $h(\tau,t)$, observed at one of the receiving elements over a period of 0.4 s. This shallow water channel can be assumed to be limited to a duration of 10 ms, which, depending on the signaling rate used, corresponds to 25 to 100 symbol intervals. The coherence time of the channel was estimated to be on the order of 0.1 s to 1 s, resulting in the equivalent Doppler bandwidth less than 10 Hz. At the signaling rates of 2.5–10 ksps, the equivalent normalized fading rates would be on the order of $4\cdot 10^{-3}-10^{-3}$, which is significant enough to cause a noticeable performance degradation. The impulse response of the channel usually consisted of two significant arrivals of comparable energy, accompanied by a multitude of low-energy arrivals.

Performance results are summarized in Figs. 6 and 7. Each plot in these figures represents the scatter diagram of the estimated QPSK data symbols $\hat{d}(n)$ on which the decisions are performed (see Fig. 1). These results correspond to the decision-directed mode of operation. The amount of dispersion around the center points $\pm 1 \pm j$ is proportional to the mean-squared error in data detection which determines the quality of performance. Listed by the side of each plot are the corresponding receiver parameters. N and M denote the number of feedforward and feedback filter tap weights, respectively. The receiver tracking parameters are the forgetting factor of the RLS algorithm (f.f.), and the phase tracking constants, $K_{f_{1,2}}$ (details of receiver design are given in Ref. 2). There was a total of 2000 symbols transmitted in each example. Note that in order to perform a fair comparison, at each signaling rate the equalizer has a different number of taps chosen so as to span the same amount of time, corresponding to the channel multipath spread. Shown in Fig. 6 are the results of single-channel ($K=1$) performance, and shown in Fig. 7 are the results of $K=3$ channel diversity reception. Going from the top to the bottom in each figure, the data rate increases from 2.5 ksps to 10 ksps. Indicated by the side of each scatter plot is the estimated probability of error P_e, obtained as the ratio of the number of erroneous symbols to the total number of symbols transmitted.

As the data rate is increased, we observe a dramatic improvement in performance in both the single-channel and the 3-channel receiver configurations, as the data estimates cluster more closely around their ideal locations. By far the best performance was obtained at the highest signaling rate used, i.e., at 10 ksps, or equivalently, 20 kbps. In each instance, a low-pass filter with a cutoff frequency $1/T$ was

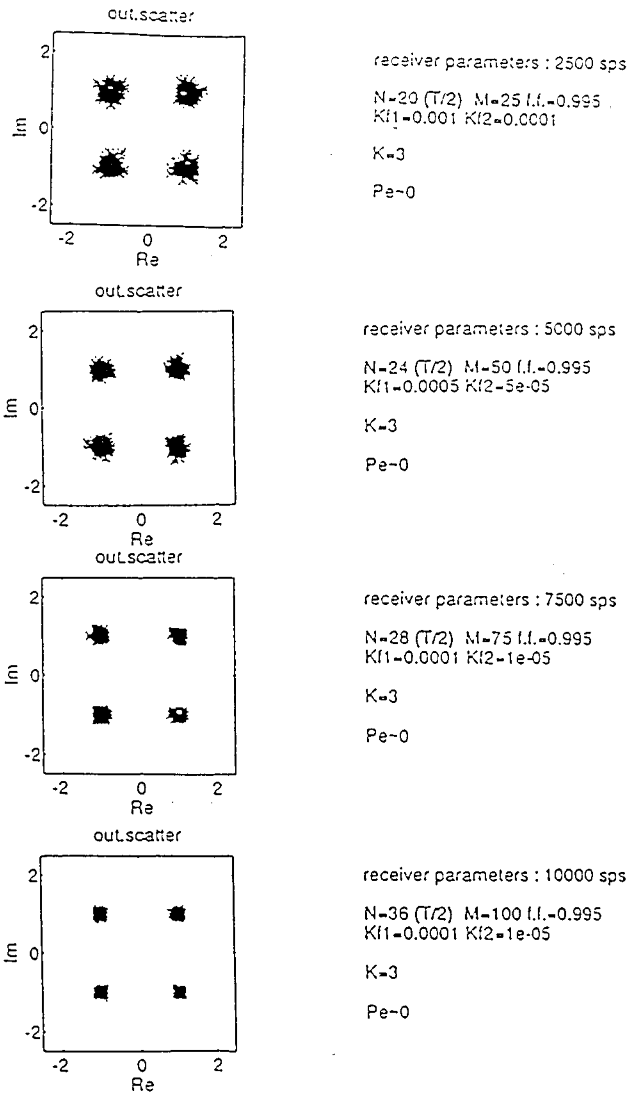

FIG. 7. Receiver performance in 3-channel configuration.

employed at the receiver front-end to limit the noise power. Note that for higher data rates *more* noise is passed to the equalizer. Hence, we explain the performance improvement by the fact that as the rate is increased, the changes in the channel, observed by an adaptive receiver on a symbol-to-symbol scale appear slower, providing for better channel tracking. In terms of the analysis presented in the previous section, changes of the symbol rate result in the changes of the *normalized* Doppler spread, $\omega_d T$, which was shown to be the figure of merit in determining the system performance.

In the single-channel configuration, signaling at the lowest rate of 2.5 ksps results in an unacceptable performance. However, by increasing the rate, the performance gradually improves, and at 10 ksps we observe a satisfactory performance. In the multichannel configuration, performance in general is greatly improved. Here, as well as in the single-channel configuration, it is clearly observed that as the signaling rate increases, the performance improves. At 2.5 ksps diversity helps to overcome the poor performance of the single-channel receiver, resulting in good quality data. At 10 ksps, the performance of the multichannel receiver is excellent. An additional advantage gained by increasing the data rate is a considerable reduction in the sensitivity of the performance to the choice of the receiver tracking parameters.

Unfortunately, at the signaling rate of 10 ksps the transducer bandwidth limit was reached, and we were not able to investigate the performance at signaling rates higher than this. We expect that by further increasing the data rate, a region in which the ISI penalty begins to dominate would be reached, and the receiver performance would start to degrade. At 10 ksps the ISI span is already very large and requires a feedback filter with 100 taps. Nevertheless, the DFE still successfully copes with this ISI. At the same time, it takes full advantage of the fine channel sampling and multipath resolution resulting from signaling at a high rate of 10 ksps.

From the above results we may conclude that although signaling at high rates causes the ISI to contaminate a large number of the received data symbols, one should not seek the solution to this problem by confining the signaling rates to low values. To deal with the multipath problem, previous generations of high-rate UWA communication systems have resorted to multichannel signaling, where the available frequency band is divided into many narrow bands, in each of which a very low-rate signal is transmitted. In this way, each of the sub-channels can be regarded as free of multipath, thus eliminating the need for equalization. An example of such a system is the 64-FSK system described in Ref. 5. However, we have seen that in coherent systems signaling at low rates implies greater penalty caused by imperfect channel estimation.

An alternative solution to the multipath problem is the broadband signaling method presented which uses equalization to combat the effects of multipath. The fact that UWA communications are generally confined to low frequencies allows for the implementation of complex receiver algorithms. Therefore, by employing a powerful equalization method, the receiver will be able to suppress the unwanted multipath effects. At the same time, it will gain an additional advantage of fast channel sampling which enables channel tracking much better than that of a narrow-band system. Hence, although the multipath represents a major problem in UWA communications, better performance of an adaptive equalizer can be achieved by using higher signaling rates in a shallow water channel. This phenomenon is explained by the fact that increasing the signaling rate effectively decreases the normalized Doppler spread of the channel, thus reducing the penalty caused by imperfect channel tracking.

III. SUMMARY AND CONCLUSIONS

It is commonly assumed that in order to overcome the effects of multipath, the transmission rate used on a shallow water channel should be kept at low values. In doing so, the implications which a low-rate signaling method bears on the problem of tracking the time-varying channel are often neglected.

The performance of an adaptive receiver on a rapidly varying channel heavily depends on the quality of the available channel estimate. Theoretical considerations described in this article support this fact by showing that higher channel dynamics result in a greater degradation of receiver performance. This analysis is based on considering receiver optimization in the case when only the channel estimate is available to the receiver. Such a case is deemed more realistic than the commonly used assumption of perfect knowledge of the random channel impulse response.

A strongly fluctuating medium-range channel may exhibit Doppler spreads high enough to cause bit error saturation at moderate signaling speeds. By increasing the signaling rate on such a channel, the channel will stay relatively constant over a larger number of symbol intervals, at the expense of introducing additional ISI. Hence, as the transmission rate is increased, the tradeoff between the estimation error penalty and the residual ISI penalty will become an important system design issue.

The results presented clearly demonstrate that despite the multipath, better performance of an adaptive equalizer can be achieved by using *higher* signaling rates. This fact leads us to conclude that low transmission rates are *not* the best solution for coherent communications over shallow water multipath channels. Not only is it possible to overcome the high-rate ISI effects by the use of sophisticated equalization methods, but signaling at high rates may result in better receiver performance by virtue of reducing the channel tracking errors.

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