Recent Advances in High-Speed Underwater Acoustic Communications

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Abstract—**In recent years, underwater acoustic (UWA) communications have received much attention as their applications have begun to shift from military toward commercial. Digital communications through UWA channels differ substantially from those in other media, such as radio channels, due to severe signal degradations caused by multipath propagation and high temporal and spatial variability of the channel conditions. The design of underwater acoustic communication systems has until recently relied on the use of noncoherent modulation techniques. However, to achieve high data rates on the severely band-limited UWA channels, bandwidth-efficient modulation techniques must be considered, together with array processing for exploitation of spatial multipath diversity. The new generation of underwater communication systems, employing phase-coherent modulation techniques, has a potential of achieving at least an order of magnitude increase in data throughput. The emerging communication scenario in which the modern underwater acoustic systems will operate is that of an underwater network consisting of stationary and mobile nodes. Current research focuses on the development of efficient signal processing algorithms, multiuser communications in the presence of interference, and design of efficient modulation and coding schemes. This paper presents a review of recent results and research problems in high-speed underwater acoustic communications, focusing on the bandwidth-efficient phase-coherent methods. Experimental results are included to illustrate the state-of-the-art coherent detection of digital signals transmitted at 30 and 40 kb/s through a rapidly varying one-mile shallow water channel.**

I. INTRODUCTION

IN THE past five to 10 years, there has been a tremendous increase in research and development of underwater acoustic (UWA) communication systems. The growing interest in UWA communications came as a response to the rapidly growing needs for wireless (tetherless is sometimes a preferred term) underwater communications, brought in part by the broadening of applications, which were almost exclusively military, to commercial ones. Commercial applications which have received much attention lately are pollution monitoring in environmental systems, remote control in off-shore oil industry, and collection of scientific data recorded at benthic stations without the need for retrieving the instruments. As efficient communication sytems are developing, the scope of their applications continues to grow, and so do the requirements on the system throughput and performance. Many of the developing applications, both commercial and military, are now calling for real-time communication with submarines and autonomous underwater vehicles, not only in point-to-point links, but also in network configurations.

Manuscript received March 15, 1995.

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Publisher Item Identifier S 0364-9059(96)03420-6.

Due to the difficulties encountered in underwater acoustic communication channels, the most prominent of which is considered to be the multipath propagation [1]-[3], the design of modern high-speed data transmission systems has been accompanied by extensive research in both the fields of oceanographic and communication engineering. This research has aimed toward bringing together many of the well-established principles of wireless radio communications and re-considering them for application in UWA channels. However, the adverse effects underwater propagation has on the digital acoustic signals often require development of specialized communication techniques. Many solutions have been offered, and the extensive research of the past years has established UWA communications as a new field of applied engineering.

Since the last publication of the Special Issue on Ocean Acoustic Data Telemetry in the IEEE JOURNAL OF OCEANIC ENGINEERING in 1991, fundamental advances have been made in this field. Bandwidth-efficient phase-coherent communications, previously not considered feasible, were demonstrated to be a viable way of achieving high-speed data transmission through many of the underwater channels, including the severely time-spread horizontal shallow water channels [4]–[6]. The new generation of UWA communication systems, based on the principles of phase-coherent detection techniques, is capable of achieving raw data throughputs that are an order of magnitude higher than those of the existing systems [1] which are based on noncoherent detection methods.

These recent results open many new possibilities for application of UWA communications. Notable among the emerging applications is the concept of an autonomous oceanographic sampling network (AOSN) [7]. This network will provide exchange of data, such as control, telemetry and video signals, between many network nodes. The network nodes, both stationary and mobile ones, located on underwater vehicles and robots, will be equipped with various oceanographic instruments, such as hydrophones, current meters, seismometers, sonars and video cameras. A remote user, interested in gathering the various oceanographic data, is envisioned to have a direct computer access via a radio link to a central network node based on a surface buoy. Besides the design of communication links at the physical layer, i.e., on the level of a communication channel, the complete system requires a new approach to network protocol design. Major difficulties are encountered due to the long propagation times in the underwater channels. Nevertheless, advances have already been made in this direction, and the first protocols for acoustic local area networks (ALAN) have been proposed [8], [9].

This paper presents a review of the most recent developments in UWA communications, as well as the latest achievement in bandwidth-efficient communication methods developed for high-speed UWA data transmission. For a more general review of UWA communication systems, the reader is referred to [1]–[3]. As an introduction to the problem of UWA communications, Section II gives a brief overview of the characteristics of UWA communication channels and the recently proposed approaches to channel modeling, as seen from the viewpoint of communication system design. Section III is devoted to communication system design, which is largely determined by the method employed for overcoming the problem of multipath. The approaches to system design are discussed on the basis of signal design (the modulation method) and processing method (array processing and equalization methods). Examples are given of the methods which have been implemented in real-time systems, as well as of methods currently under development. In Section IV, we present the current state-of-the-art in phase-coherent system design for UWA channels with extreme multipath, such as that observed in most of the horizontal UWA links. The principles of phase-coherent signal design and detection using spatial diversity and equalization are summarized, and the system performance is illustrated through the results obtained recently on a shallow water channel. Finally, Section V describes some of the advanced research topics currently pursued in UWA communications, as well as those topics that need to be addressed in the future.

II. UWA Communication Channels

The available bandwidth in an UWA communication channel is severely limited by transmission loss which increases with both frequency and range [10], [11]. For example, a long-range system (see [1] for precise definitions) operating over several tens of kilometers is limited to few kHz of bandwidth; a medium-range system operating over several kilometers has a bandwidth on the order of ten kHz, while only a short range system, operating over several tens of meters may have available more than a hundred kHz.

Within this limited bandwidth, the signal is subject to multipath propagation through a channel whose characteristics vary with time and are highly dependent on the location of the transmitter and receiver. The multipath structure depends on the link configuration, which is primarily designated as vertical or horizontal. While vertical channels exhibit little multipath, horizontal channels may have extremely long multipath spreads. Most notable in the long- and medium-range channels, multipath propagation causes severe degradation of the acoustic communication signals. Combating the underwater multipath to achieve a high data throughput is without exception considered to be the most challenging task of an UWA communication system.

In a digital communication system, multipath propagation causes intersymbol interference (ISI), and an important figure of merit is multipath spread in terms of symbol intervals. While typical multipath spreads in the commonly used radio channels are on the order of several symbol intervals, in the horizontal UWA channels they increase to several tens, or a hundred of symbol intervals for moderate to high data rates. For example, a commonly encountered multipath spread of 10 ms in a medium-range shallow water channel, causes the ISI to extend over 100 symbols if the system is operating at a rate of 10 kilosymbols per second (ks/s).

The mechanism of multipath formation depends on the channel geometry and also on the frequency of transmitted signals. Understanding of these mechanisms is based on the theory and models of sound propagation. Two principal mechanisms of multipath formation are reflection at the boundaries (bottom, surface and any objects in the water), and ray bending. If the water is shallow, propagation will most usually occur in surface-bottom bounces in addition to a direct path and reflections from other boundaries, if any. If the water is deep, as in the regions past the continental shelves, the sound channel may form by bending of the rays toward the axis of the deep sound channel where the sound speed reaches its minimum (typically at a depth of several hundred meters). Alternatively, the rays bending upwards may reach the surface focusing in one point where they are reflected, and the process repeats periodically (typical length of the so-formed convergence zone is 60–100 km). In vertical transmission channels, multipath can form by surface backscattering.

The geometry of multipath propagation is important for communication systems which use array processing to suppress multipath (e.g., [15], [34]). The design of such systems is accompanied by the use of a propagation model for predicting the multipath configuration. Ray theory and the theory of normal modes provide basis for such propagation modeling [11]. Recent references commonly use ray tracing for determining the corase multipath structure for communication channel modeling [16], [18]. A different class of UWA communication systems has recently been developed [5], which do not rely on the particular multipath geometry, and is equally applicable in a variety of channels, regardless of the parameters such as range-to-depth ratio which determine the angles of incidence of multipath arrivals.

Associated with each of the deterministic propagation paths (macro-multipaths), which can be modeled accurately, are random signal fluctuations (micro-multipath), which account for the time-variability of the channel response. Some of the random fluctuations can be modeled statistically [10]. These fluctuations include surface scattering due to waves, which is the most important contributor to the overall time variability of the shallow water channel. In deep water, in addition to surface scattering, internal waves contribute to the single-path random fluctuations [1].

While there exists a vast knowledge of both deterministic and statistical modeling of sound propagation underwater, the implications this knowledge bears on the *communication* channel modeling has only recently received more attention (e.g., [15], [16]). A time-varying multipath communication channel is commonly modeled as a tapped delay line, with tap spacing equal to the reciprocal of twice the channel bandwidth, and the tap gains modeled as stochastic processes with certain distributions and power spectral densities [13]. While it is known that mobile radio channels fit well within the model of Rayleigh fading, where the tap gains are derived from complex Gaussian processes, there is no single model accepted to date for any of the UWA channels. Among the UWA channels, modeling of the shallow water medium-range channel has received most attention, as this channels is known to be the most rapidly varying one. Most authors consider that this channel is fully saturated and thus exhibits Rayleigh fading (e.g., [1]). Recently, such statement has been confirmed in [15] and an experimental study [17]. However, it has been challenged in [18] where both the deep and the shallow water channel are found to fit better within a deterministic model. The deep water channel has also been modeled as a Rayleigh fading channel [19]; however, no measurements have been reported to support such modeling. The available measurements are still scarse, and mostly focus on a stationary communication scenario. In the mobile UWA channel, vehicle speed will be the primary factor determining the time-coherence properties of the channel, and consequently the system design. Knowledge of a statistical channel model has proven to be useful in the design of communication systems (e.g., the Jakes' model for the mobile radio channels [14]); however, it remains for the future to develop widely recognized statistical UWA channel models and the systems optimized to match those models.

The implications time-varying multipath bears on the communication system design are twofold, since the ISI in the received signal depends on both the physical channel and the duration of transmitted pulses. On one hand, signaling at a high rate (short pulses) causes many adjacent symbols to interfere at the receiver, and requires sophisticated processing to compensate for the ISI. On the other hand, as pulse duration becomes shorter, channel variation over a single symbol interval becomes negligible. This allows an adaptive receiver to efficiently track the channel on a symbol-to-symbol basis. Hence, the time-varying multipath causes a trade-off in the choice of signaling rate for a given channel: the multipath spread of the channel, as measured in symbol intervals, will be longer at a higher signaling rate, but its time-coherence, measured on the same scale, will improve. As the methods for dealing with the long ISI of UWA channels are developing, it becomes possible to use higher transmission rates and at the same time take advantage of effective reduction in the channel time-variability. These observations have recently been supported by experimental results obtained on a shallow water medium-range (approximately 1 n.mi.) channel [20]. In this experiment, QPSK signals were transmitted at rates varying from 2.5 ks/s to 10 ks/s, and processed using adaptive equalization methods described in [5]. The results showed a steady increase in the quality of performance with the increase in transmission rate. A theoretical performance analysis [21], based on a stochastic model of a Rayleigh fading channel, supports the experimental observations.

The trade-off in the choice of a signaling rate for a bandwidth-efficient communication system may be illustrated through the following example of a shallow water channel. In shallow water, where multipath is dominated by reflections from channel boundaries, the problem largely depends on the particular geometry [1]. The major source of temporal variability in shallow water transmissions is the surface scattering caused by waves. Surface scattering is in general modeled using the method of small perturbations [10], resulting in a Gaussian distributed surface height displacement. However, a simplified analysis based on considering the channel geometry and a sinusoidal surface displacement model, illustrates quantitatively the effects of surface motion on the Doppler spreading of the signal. A communication system is concerned mostly with forward scattering, and we consider the receiver and transmitter separated by a range r at some nominal depths h_1 and h_2. Wind driven waves cause changes in the height of the reflection point, $h_{1,2}(t) = h_{1,2} + \Delta h(t)$, which results in the surface-reflected path length time-variation

$$l(t) \approx l_0 + 2\Delta h(t) \cos\theta_0 = l_0 + \Delta l(t) \tag{1}$$

where l_0 is the nominal path length, θ_0 is the nominal angle of incidence, and it is assumed that the maximal surface height deviation is $\Delta h_m \ll h_{1,2}$. Due to the path length variability, the transmitted signal component $A\cos\omega t$, appears at the receiver to be phase-modulated by a waveform

$$\Delta\varphi(t) = \frac{\omega}{c}\Delta l(t). \tag{2}$$

Given the maximal deviation, or the modulation index $\Delta\varphi_m$ of the phase $\Delta\varphi(t)$, and the frequency of the waves f_w, the resulting bandwidth of the received phase-modulated signal will be (Carson's rule [14])

$$B_w = 2f_w(1 + \Delta\varphi_m). \tag{3}$$

In terms of the r.m.s. wave height h_w, the estimated Doppler spread caused by the surface waves may be expressed as

$$B_w = 2f_w\left(1 + \frac{2\omega\cos\theta_0}{c}h_w\right). \tag{4}$$

This result is a generalization of the ones obtained in [11] and [22].

Using the empirical values which relate the wave height and frequency to the wind speed w in m/s [11]

$$f_w = 2/w; \qquad h_w = 0.005w^{2.5}. \tag{5}$$

Doppler broadening B_w can be obtained. As an example, consider the transmitter and receiver at the same depth $h_0 = 10$ m, $r = 500$ m, and a carrier of frequency $f = 15$ kHz. For a moderate wind speed of $w = 20$ m/s we obtain Doppler spreading at the carrier frequency due to a single surface reflection to be $B_w = 10$ Hz. Considering a signaling rate $1/T$ of 1 kHz for this channel, and possibly stronger winds, the normalized Doppler spread B_wT approaches the limiting value of 10^{-2} for reliable coherent channel tracking [21]. And indeed, many researchers still find that due to surface variability phase-coherent detection is not feasible in the shallow water channels [15], [26]. However, increasing the signaling rate on this channel effectively reduces the design parameter B_wT, thus establishing the possibility for phase-coherent detection, provided, of course, a method for dealing with the resulting ISI.

TABLE I
RECENT SYSTEM IMPLEMENTATIONS

developed by	application	channel	modulation	ISI compensation	band	data rate
Oki Elec. Ind. Comp. [27]	robot comm./contr.	very short (60 m), shallow	16-QAM	LE (LMS)	1 MHz	500 kbps
JAMSTEC [28]	image tx	vertical (6500 m)	4-DPSK	LE (LMS)	20 kHz	16 kbps
IFREMER/ ORCA [29]	image and data tx	vertical (2000 m)	2-DPSK	none	53 kHz	19.2 kbps
ENST-Br./ IFREMER [30]	digital speech tx	test pool	4-DPSK	DFE (LMS)	not rep.	6 kbps
Micrilor [31]	telemetry	medium (1 km), shallow	2-DPSK	DS-SS	30 kHz/100 kHz	600 bps
WHOI/ Datasonics [24]	telemetry	vertical and horizontal	16 × 4-FSK	none	15 kHz	1200 bps
WHOI [32]	telemetry	under-ice, shallow	QPSK	DFE (RLS)	15 kHz	5 kbps

TABLE II
RECENT ACHIEVEMENTS IN SIGNAL PROCESSING USING PHASE-COHERENT MODULATION TECHNIQUES

developed at	test channel	modulation	array	equalizer	band	data rate
U.Birmingham, U.K. [15]	medium (0.5-1.5 n.mi.), shallow	DPSK, 2-DPSK	tx and rx, fixed	none	50 kHz	10 kbps 20 kbps
U.Newcastle, U.K. [34, 35]	medium (0.5 n.mi.) shallow	4-DPSK	rx, LMS	none/ DFE (LMS)	50 kHz	10 kbps
Northeastern U./ WHOI [5, 6]	long deep (∼ 100 n.mi.) long shallow (∼ 50 n.mi.) medium (1 n.mi.), shallow	M-PSK,8-QAM M=4,8,16	rx RLS (diversity)	multichannel DFE (RLS)	1 kHz 1 kHz 15 kHz	1 kbps 2 kbps 40 kbps

III. SYSTEM DESIGN

The combined effects of multipath propagation and temporal channel variability which cause ISI and strong phase fluctuations of the signal, have led in the past to system design based exclusively on noncoherent detection and low signaling rates [1]. It has been only recently that phase coherent methods both with differential and purely coherent detection have been implemented for the first time, resulting in increased system throughputs.

Approaches to system design vary according to the technique used for overcoming the effects of multipath and phase variations. Specifically, these techniques may be classified according to 1) the signal design, i.e., the choice of modulation/detection method, and 2) the transmitter/receiver structure, i.e., the choice of array processing method and the equalization method, if any. In what follows we shall review both the design of systems that are already operational and the design which is being developed by off-line testing of various techniques. The relevant parameters used in the representative examples of system design are summarized in Tables I and II. While most of the existing systems operate on the vertical, or the very short-range channels, the systems under development often focus on the severely spread horizontal shallow water channels.

A. Systems Based on Noncoherent Modulation

Let us focus first on the signal design and the choice of modulation method. As it was mentioned, noncoherent detection of frequency shift keying (FSK) signals has traditionally been considered as the only alternative for channels exhibiting rapid phase variation such as the shallow water long- and medium-range channels. While noncoherent detection eliminates the problem of carrier phase tracking, it does not solve the problem of multipath. To overcome the ISI, the existing noncoherent systems employ signal design with guard times, which are inserted between successive pulses to ensure that all the reverberation will vanish before each subsequent pulse is to be received. The insertion of idle periods of time obviously results in a reduction of the available data throughput. In addition, due to the fact that fading is correlated among frequencies separated by less than $\Delta f_c = 1/T_m$, where T_m is the multipath spread, it is desired that only those frequency channels, separated by more than Δf_c be used at the same time. This requirement further reduces the system efficiency unless some form of coding is employed so that the adjacent, simultaneously transmitted frequencies belong to different codewords. A representative system [23] for telemetry at a maximum of 5 kb/s uses a multiple FSK modulation technique in the 20–30 kHz band. This band is divided into 16 subbands, in each of which a 4-FSK signal is transmitted. Hence, out of a total of 64 channels, 16 are used simultaneously for parallel transmission of 32 information bits (two information bits per one four-channel subband). This system has successfully been used for telemetry over a 4-km shallow water horizontal path, and a 3-km deep ocean vertical path. It was also used on a 700-m very shallow water path, where probabilities of bit error on the order of 10^{-2}–10^{-3} were achieved without coding. The system performance may be improved by using error correction coding; however, its data rate will be reduced. The multiple FSK system has been implemented [24], and is commercially available with a maximum data rate of 1200 b/s. Despite the fact that these systems have bandwidth efficiency which does not exceed 0.5 b/s/Hz, noncoherent FSK is a good solution for applications where moderate data rates and robust performance are required. As such, these methods are still receiving attention, and a method has recently been implemented [25] which uses orthogonal frequency division multiplexing (OFDM) realized with DFT-based filter banks. This system was used on a medium-range channel; however, due to the high frequency separation among the channels (only every fourth channel is used) and relatively long guard times (10 ms guard following a 30-ms pulse), needed to compensate for the multipath fading distortion, the effective data rate is only 250 b/s.

B. Systems Based on Coherent Modulation

With the goal of increasing the bandwidth efficiency of an UWA communication system, research focus over the past years has shifted toward phase-coherent modulation techniques, such as phase shift keying (PSK) and quadrature amplitude modulation (QAM). Depending on the method for carrier synchronization, phase-coherent systems fall into two categories: differentially coherent and purely phase-coherent. The advantage of using differentially coherent detection is the simple carrier recovery it allows. Its disadvantage is performance loss as compared to coherent detection [13]. While bandwidth-efficient methods have successfully been tested on a variety of channels, real-time systems have mostly been implemented for application in vertical and the very short range channels, where little multipath is observed and the phase stability is good.

In the very short range channel, where bandwidth in excess of 100 kHz is available, a representative system [27] operates over 60 m at a carrier frequency of 1 MHz and a data rate of 500 kb/s. This system is used for communication with an undersea robot which performs maintenance of a submerged platform. A 16-QAM format is used, and the performance is aided by an adaptive equalizer. A linear equalizer (LE), operating under an LMS (least mean squares) algorithm suffices to reduce the bit error rate from 10^{-4} to 10^{-7} on this channel.

The current state-of-the-art in coherently modulated systems which use the deep ocean vertical path is represented by an image transmission system [28] developed in Japan. It is a differentially coherent, 4-DPSK system with carrier frequency of 20 kHz, capable of achieving 16 kb/s bottom to surface transmission over 6500 m. The field tests of this system indicate the achievable bit error rates on the order of 10^{-4} at 15 dB of input SNR. As in the very shallow water channel, a linear equalizer operating under an LMS algorithm is used to achieve this performance.

Another example of a successfully implemented system for vertical path transmission is that of an underwater image and data transmission system developed in France [29]. This system uses a binary DPSK modulation at a data rate of 19.2 kb/s. The carrier frequency of 53 kHz was used for transmission over 2000 m.

Recent advances in digital underwater speech transmission are represented by a prototype system described in [30]. This system uses a code excited linear prediction (CELP) method to transmit the speech signal at 6 kb/s. The modulation method used is quadrature DPSK. A decision-feedback equalizer, operating under LMS algorithm is being used in the pool tests. Field tests have not been reported yet.

For the applications in shallow water medium-range channel, a coherent, DPSK system was recently developed [31], which uses a direct-sequence spread spectrum (DS-SS) method to resolve a strong surface reflection observed in the 1-km-long, 10-m-deep channel. The interfering reflection is only rejected, and not used for multipath recombining which would result in additional performance gain [13]. Data throughput of 600 bps in a spreading bandwidth of 10 kHz is achieved. Such high spreading ratios are justified in interference-suppression or anti-jamming applications.

The latest development in phase-coherent communication systems is represented by the prototype system implemented at the Woods Hole Oceanographic Institution (WHOI) [32]. This system is based on purely phase-coherent modulation and detection principles [4]. The modulation format is QPSK, and the signals are transmitted at 5 kb/s, using a carrier frequency of 15 kHz. The system is currently configured as a six-node network, and its real-time operation was demonstrated in the under-ice shallow water environment. The ISI caused by the shallow water multipath is processed by a decision-feedback equalizer operating under an RLS (recursive least squares) algorithm.

Signal processing methods for multipath compensation: To achieve higher data rates, the more sophisticated systems based on phase-coherent signaling methods must allow for the ISI in the received signal. These systems employ either some form of array processing, or equalization methods, or a combination thereof, to compensate for the ISI. Three main approaches have recently been taken toward this end. Examples of systems based on these approaches are summarized in Table II. The first two approaches use differentially coherent detection and rely on array processing to eliminate, or reduce multipath. The third approach is based on purely phase-coherent detection and the use of equalization together with array processing for exploitation of the multipath spatial diversity.

Array processing for multipath suppression has been used both at the transmitter and at the receiver end. An approach pursued at the University of Birmingham, U.K. [12], [15] focuses on the use of transmitter arrays to excite only a single path of propagation. Long arrays and careful positioning are required to ensure complete absence of multipath, since no equalization is used in this system. Equalization is not deemed feasible in this work, since it is considered that the shallow

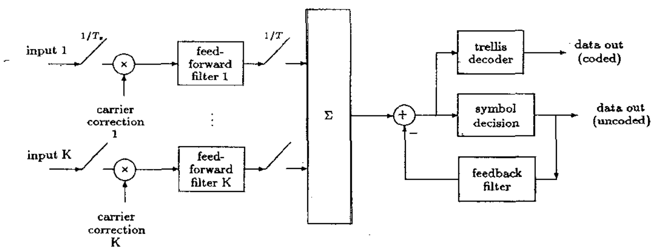

Fig. 1. Phase-coherent receiver based on a multichannel DFE.

water test channel exhibits Rayleigh fading which is too rapid for an adaptive equalizer to follow. Instead, a receiving array is employed to compensate for the possible pointing errors. Binary and quaternary DPSK signals were used achieving data rates of 10 kb/s and 20 kb/s, respectively. The estimated bit error rate (BER) was on the order 10^{-2}–10^{-3}, depending on the actual channel length. In general, it was found that the technique proposed is more effective at shorter ranges.

An approach for multipath rejection using adaptive beamforming at the receiver end is being investigated at the University of Newcastle, U.K. [34]–[36]. The prototype beamformer uses an LMS type of algorithm to adaptively steer nulls in the direction of a surface reflected wave. Similarly as in the case of the transmitter array, it was found that the beamformer encounters difficulties as the range increases relative to depth [35]. To compensate for this effect, the use of an equalizer was proposed to complement the performance of the beamformer [34]. The equalizer is of a decision-feedback type, and it operates under the LMS algorithm whose low-computational complexity permits real-time adaptation at the symbol rate. A separate waveform is transmitted at twice the data rate for purposes of time-synchronization. The system was tested in shallow water at 10 kb/s, and showed the estimated BER of 10^{-2} without, and 10^{-3} with the equalizer.

The final approach we shall describe is based on the use of purely phase-coherent detection methods, which have been developed at Northeastern University and WHOI [4]–[6] for application in UWA channels with severe multipath. These signal processing methods are based on joint synchronization and equalization for combating the effect of phase variations and ISI. The method of fractionally spaced decision-feedback equalization is used with an RLS algorithm. It also incorporates spatial signal processing based on diversity combining, which will be discussed in more detail in the following section.

As indicated in Table II, the phase-coherent methods have been tested in various underwater environments. The achieved data rates of up to 2 kb/s over long range channels, and up to 40 kb/s over shallow water medium-range channels, are among the highest reported to date.

Currently, several prototype systems are being developed based on the phase-coherent methods. The prototype system implemented at WHOI [32] is already operational. A similar prototype implementation which is currently under way is described [33]. In this reference, results of off-line processing in deep and shallow water are presented, reporting data rates of 1 kb/s to 5 kb/s, and the achieved BER of 10^{-3}–10^{-5}.

In Section IV, we focus on the method of joint synchronization and multichannel equalization, as these methods offer the possibility to establish high-speed communications over a variety of time-spread UWA channels. Results of experimental performance obtained on a shallow water channel, which were referred to in Table II, are presented.

IV. Receiver Design for High-Speed Communications Based on Phase-Coherent Detection

In many of the UWA channels multipath structure may exhibit one or more components which carry the energy similar to that of the principal arrival. As the time progresses, it is not unusual for these components to exceed in energy the principal arrival [4], [44]. The fact that the strongest multipath component may not be well defined makes the extraction of carrier reference a difficult task in the underwater channel. To establish coherent detection in the presence of strong multipath, a technique based on simultaneous synchronization and multipath compensation has been presented in [4]. This technique is based on joint estimation of the carrier phase and the parameters of a decision-feedback equalizer (DFE), where the optimization criterion is minimization of the mean-square error (mse) in the data estimation process. In [5], the equalizer/synchronizer structure has been extended to include a number of input array channels. This implementation with spatial diversity combining has shown superior performance in a variety of channels, as well as potentials for dealing with several types of interference. The structure of the multichannel equalizer is shown in Fig. 1.

The input signals to the baseband processor are the A/D converted array signals, brought to baseband using nominal carrier and lowpass filtering, and frame-synchronized using a known channel probe (usually a Barker sequence transmitted in phase and quadrature at the data rate). Baseband processing begins with downsampling, which may be carried out to as few as two samples per symbol interval ($T_s = T/2$), since the signals are shaped at the transmitter nominally using raised-cosine spectrum shaping which limits their bandwidth to less than $1/T$. For applications where transmitter and receiver are not moving, but only drifting with water, no explicit adjustment of the sampling clock is needed. It will implicitly be accomplished during the process of adaptive fractionally spaced equalization. The front section of the equalizer will also perform adaptive matched filtering and linear equalization. To correct for the carrier offset, the signals in all channels are phase-shifted by the amount estimated in the process of joint equalization and synchronization. After coherent combining, the ISI resulting from the previously transmitted symbols (postcursors) is cancelled in the feedback section of the equalizer. This receiver structure is applicable to any linear modulation format, such as M-PSK, or M-QAM, the only difference being in the way in which symbol decision is performed. In addition to combining and equalization, signal processing at the receiver includes the operation of decoding if the signal at the transmitter was encoded. Trellis coded modulation, compatible with PSK and QAM signals, provides an effective means of improving performance on a band-limited channel [37]. In addition to coded modulation, error correction coding may be employed; however, this topic is out of our present scope.

The receiver parameters that are adaptively adjusted are the tap weights of the feedforward filters, the carrier phase estimates, and the tap weights of the feedback filter. A single estimation error is used for the adaptation of all the parameters. This error is the difference between the estimated data symbol which is input to the decision device, and its true value. During the initial training mode, the true data symbols are known. After the training period, when the receiver parameters have converged, the on-line symbol decisions are fed back to the equalizer and used to compute the error. The adaptive algorithm that has proven to be effective in different UWA environments is a combination of the second-order digital phase-locked loop (PLL) for the carrier phase estimates, and the RLS algorithm for the multichannel equalizer tap weights. The details of the joint adaptation are given in [5].

The main distinction between diversity combining and the beamforming approach described in the previous section is that in the first case, the combiner operates jointly with the equalizer to minimize the error in data symbol estimation, while in the latter case, the adaptive beamformer is adjusted separately using a reference of the transmitted signal to null out the signal replicas arriving from angles different than that of the desired path. The signal isolated in this manner is processed by a separately optimized equalizer, needed to compensate for the residual ISI. Residual ISI arises due to the fact that a classical, narrow-band beamformer cannot completely eliminate the interference represented by the signal replicas which are necessarily broad-band in high-speed UWA applications. Rather than suppressing the multiple signal reflections, the diversity combiner attempts to combine them coherently in a manner analogous to matched filtering. Its disadvantage over the beamforming approach is that its complexity increases significantly with the number of input channels. However, reduced-complexity approaches are being developed [6], which seem to incorporate best of both methods. These methods are commented upon in Section V. Since it is not constrained by angular resolution, the method of diversity combining and equalization, termed multichannel equalization, may be used with as few as two input channels, and is equally applicable to a variety of UWA channels, regardless of the channel's range-to-depth ratio. In addition, signal processing with a small number of receiving sensors may be the only alternative for many of the mobile applications, a case in which both the large transmitter and receiver beamforming arrays are not applicable.

A. Experimental Results

As the approach described above has successfully been demonstrated on the long-range channels [5], a series of experimental demonstrations was performed at WHOI during the months of January and February 1994, to investigate the algorithm performance on a highly-varying shallow water medium-range channel. Below, we present experimental results of the receiver performance on this channel at 30 and 40 kb/s.

The experiments were conducted in the Woods Hole harbor, with the transmitter and receiver separated by approximately

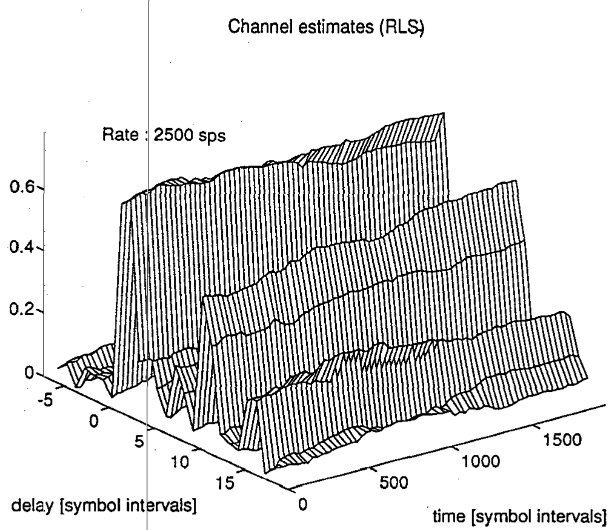

Fig. 2. Channel response in time.

one nautical mile in about sixty feet of water. The transmitter with 60° of directionality was suspended from a barge to a depth of about 10 feet, and three omnidirectional hydrophones, mounted on the dock (two near the surface, and one near the bottom) served as the receiving elements. The transducer level was 185 dB re μPa and its nominal bandwidth was 10–20 kHz.

The binary input data, representing an underwater image, were mapped into PSK and QAM signals. The signals were transmitted at a carrier of 15 kHz, using varying symbol rates. The received signals were recorded using a VME-based multichannel acquisition system, and processed off-line using Matlab.

The Woods Hole harbor channel is characterized by a multipath spread in excess of 10 ms, with the multipath structure typically comprised of two to three strong arrivals (reflections off the surface and a longer-delay reflection off the nearby pilings) and a multitude of lower-energy arrivals. A snapshot of the channel impulse response variation over a time interval of 0.4 s is shown in Fig. 2. RLS channel estimation was performed using QPSK symbols transmitted at 2.5 ksps.

The performance of the adaptive receiver is shown in Fig. 3 through the scatter plots of the estimated data symbols for three types of modulation methods. In each instance, the image data was divided into 2–3 packets, each containing between 3000 and 6000 data symbols. Shown in the plots are 2000 symbols from a detected packet. In the case of QPSK, the transmission rate was 15 ksps, while in the remaining two cases it was 10 ks/s. In each case, diversity of order three was used. For operation at 10 ks/s, the equalizers were designed to have 32 feedforward taps, fractionally spaced at $T/2$, and 100 feedback taps. Carrier tracking was accomplished using a single PLL for all the receiving channels, as sufficient coherence was found among them. The forgetting factor of the RLS algorithm was chosen to be 0.998 in all the cases presented. No detection errors occurred in any of the received packets, as indicated by the packet BER estimates $P_e \sim 0$. Listed by the side of each plot is also the output SNR, as measured from the estimated data symbols.

For the QPSK and 8-QAM transmissions no coding was used, accounting for the bit rate of 30 kb/s. In both cases excellent performance is observed. The 16-QAM signals were transmitted both without coding and using a rate 2/3 trellis code from [37], resulting in a maximum data throughput of 40 kb/s, and a highest bandwidth efficiency of 40 b/s/Hz. Although no detection errors were present, the receiver performance is somewhat degraded, due to the increased sensitivity of the algorithm to the time-varying multipath at higher constellation levels. However, the performance observed provides a reliable basis for application of both the coded modulation and the error correction coding schemes. As it was mentioned earlier, the data rates achieved in this experiment are among the highest reported to date on the medium-range shallow water channels.

V. Current and Future Research

With the feasibility of bandwidth-efficient phase-coherent UWA communications established, current research is advancing in a number of directions, focusing on the development of more sophisticated processing algorithms which will enable efficient and reliable data transmission in varying system configurations and channel conditions. Some of the problems that are currently addressed and will receive attention in the near future are the development of reduced-complexity receiver structures and algorithms suitable for real-time implementation, techniques for interference suppression, multiuser underwater communications, system self-optimization, development of modulation/coding methods for improved bandwidth efficiency, and mobile UWA communication systems.

A. Reducing the Receiver Complexity

Although the UWA channels are generally confined to low data rates (as compared to the radio channels), the encountered channel distortions require complex receiver structures, resulting in high computational load which may exceed the speeds of the available hardware. Consequently, reducing the receiver complexity to enable efficient real-time implementation has been a focus of many recent studies.

The problem of reducing the receiver complexity may be addressed on two levels: the algorithmic and the structural level. For application in time-varying channels, the receiver, whether it is based on a beamforming, equalization, or combined approach, must use an adaptive algorithm for adjusting its parameters. Two commonly used kinds of algorithms are based on the LMS and the RLS principles [13].

In a majority of recent studies, the LMS-based algorithms are considered an only alternative due to their low computational complexity (linear in the number of coefficients N) [30], [34]–[36]. However, the LMS algorithm has a convergence time which may become unacceptably long when large adaptive filters are used (10 N as opposed to 2 N of the RLS algorithm). The total number of coefficients N is usually large in shallow water medium- and long-range applications (more than 100 taps is often needed for spatial and temporal processing). In addition, the standard LMS is very sensitive to the choice of the step-size. To overcome this problem,

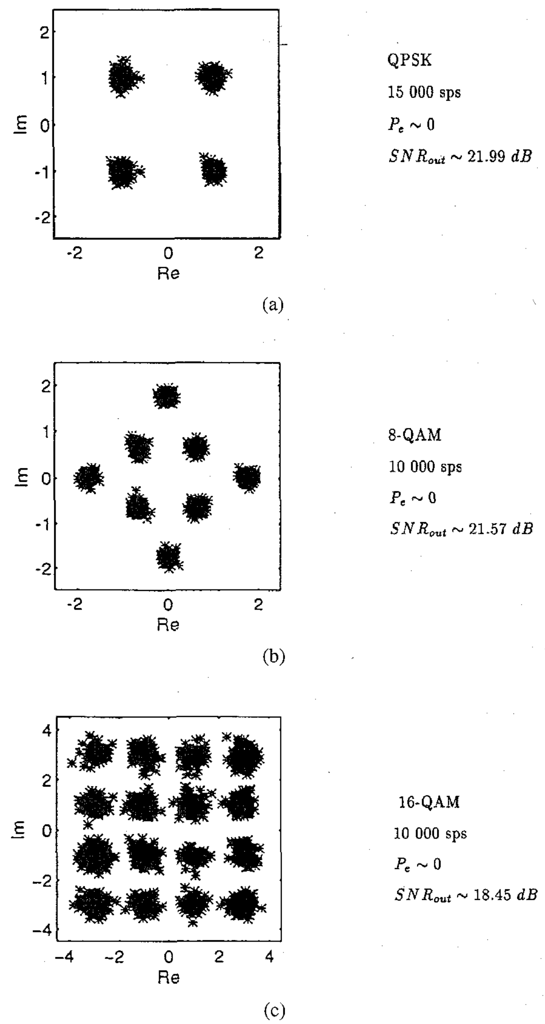

Fig. 3. Performance results: (a) QPSK at 30 kb/s, (b) 8-QAM at 30 kb/s, and (c) 16-QAM at 40 kb/s.

self-optimized LMS algorithms may be used [35], [38], but this choice results in increased complexity, and increased convergence time.

RLS algorithms, on the other hand, have better convergence properties but higher complexity. The quadratic complexity of the standard RLS algorithm is too high when large adaptive filters need to be implemented. In general, it is desirable that the algorithm be of linear complexity (number of computations per iteration proportional to the number of parameters), a property shared by the fast RLS algorithms. A numerically stable fast RLS algorithm based on [39] has been used in [4], [5] as well as for the results presented in this article. A fast modular RLS, recently developed in [40], is being used for the hardware implementation of an adaptive equalizer described in [33]. However, despite its quadratic complexity, a square-root RLS algorithm developed in [42] has been used in the real-time implementation [32], [44]. The advantage of this standard algorithm is that it allows the receiver parameters to be updated only periodically, rather than every symbol interval, thus reducing the computational load per each detected symbol. In addition, the updating intervals can be determined adaptively, based on monitoring the mse [32]. Such adaptation methods are especially suited for use with high transmission rates. At high symbol rates, the long ISI requires large adaptive filters, therefore increasing the computational complexity. However, the time interval between two transmitted symbols decreases, eliminating the need to update the receiver parameters every symbol interval. Also, the square-root RLS algorithm has excellent numerical stability, which makes it a preferable choice in many applications. A different class of adaptive filters, which also have the desired convergence and numerical stability properties, are the lattice filters which use RLS algorithms. These algorithms have been proposed in [41], but have not yet been applied to UWA channel equalization. Choosing an appropriate receiver adaptation method will receive more attention in the future acoustic modem design.

Regardless of the adaptive algorithm used, its computational complexity is proportional to the number of receiver parameters (tap-weights). Rather than focusing on low-complexity algorithms only, one may search for a way to reduce the receiver size. Although the use of spatial combining reduces residual ISI and allows shorter length equalizers to be used, a broadband combiner may still require a large number of taps to be updated, limiting the practical number of receiving channels to only a few. In [6], a method is presented which allows multichannel equalization of a large number of input channels. This is accomplished by preceding the multichannel equalizer of Fig. 1 by a spatial pre-combiner which reduces the total number of input channels (say, 10) to a smaller number (say two to three) for subsequent multichannel equalization. The overall structure of the reduced-complexity multichannel equalizer is shown in Fig. 4. The parameters of the spatial pre-combiner and those of the multichannel equalizer are optimized jointly to minimize the mse in the data symbol estimate. The reduction in complexity as compared to the case where the multichannel equalizer would operate on all the input channels may be significant if the reduced number of channels is low. In the special case when the reduced number of channels is one, the receiver structure is that of a beamformer followed by an equalizer, where the beamformer and the equalizer are optimized jointly. However, more than one channel at the output of the combiner is usually required to achieve a substantial processing gain. At the same time, the reduced number of channels at which the full processing gain may be achieved is often low (two to three), leading to an interesting conclusion that reduction in complexity may be achieved at no cost in performance. This is explained by the fact that the multipath structure is not independent among the array sensors. In addition to the reduced computational complexity, smaller adaptive filters result in less noise enhancement, contributing to improved performance.

A different approach in the design of reduced-complexity receivers has been investigated in [43], where the focus is made on reducing the number of equalizer taps. A conventional equalizer is designed to span all of the channel response. However, if the channel is characterized by several distinct multipath arrivals separated in time by intervals of no reverberation, an equalizer may be designed to have fewer taps, corresponding only to the significant portions of the channel response. Such method, termed sparse equalization, is compatible with phase-coherent synchronization methods, and was applied to detection of QPSK signals recorded in the Arctic waters. Results reported in [43] show an order of magnitude reduction in computational load. By virtue of reducing the number of adaptively adjusted parameters, this approach also makes it possible to use simple updating algorithms, such as standard RLS algorithms which have good numerical stability. Finally, in channel which are naturally sparse, discarding the low-magnitude equalizer taps in fact results in improved performance since no unnecessary noise is processed.

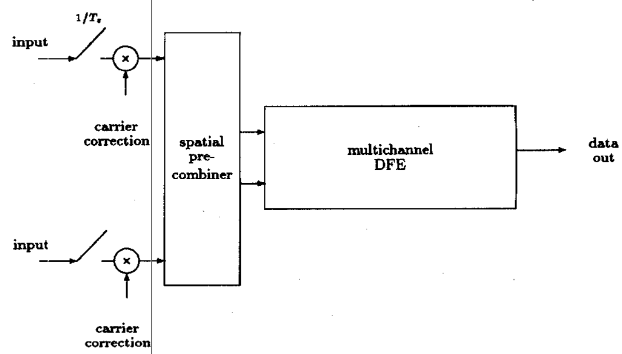

Fig. 4. Reduced-complexity multichannel DFE.

B. Interference Cancellation and Multiuser Communications

This is an active area of research since the developed UWA communication methods will find application in many interference-limited environments. The sources of interference (unintentional) include external noise and internal interference that is generated within the system. The external sources include noise coming from on-board machinery or other nearby acoustic sources, as well as the the propulsion and flow noise associated with the underwater vehicle launch process. The internal noise, which has signal-like characteristics, arises in the form of echo in full-duplex systems, or multiple-access interference (MAI) generated by other users operating within the same network.

Recent advances in noise cancellation techniques for UWA communication systems have been presented in [44]. In this study the performance of several methods for cancellation of band-limited white noise and multiple sinusoidal interference were investigated. It was found that the same receiver structure as that of Fig. 1 was most effective in cancelling the interference and noise while simultaneously detecting the desired signal. Noise cancellation is performed simply by feeding a reference of the noise signal to one of the multichannel combiner inputs, while cancellation of the sinusoidal interferer may be performed even without the reference signal. By virtue of having the training sequence, the adaptive multichannel combiner has a capability to estimate the interfering signal, reject it, and extract the desired signal.

A multiple-access communication system represents a special case of structured interference environment. Multiuser communication techniques [45] are being considered for application in the future underwater acoustic networks. Due to the limited bandwidth of the channel, in order to achieve high throughput, the network users often must share the available frequency band. Time-division multiple access may be considered, but it is associated with the problem of efficient time-slot allocation which arises because of the long propagation delays. In addition, allocation of time-slots involves a certain degree of hand-shaking which may considerably reduce the available throughput. A possible solution in such a situation is to allow the multiple users (all or within a group of prespecified size) to transmit simultaneously in both time and frequency. The receiver then has to be designed to deal with the resulting multiple-access interference, which may be very strong in an UWA network. The facts that transmission loss varies significantly with range, and that only very low code-division processing gains are available if the bandwidth is not to be waisted, both contribute to the enhanced near-far effect in the UWA channel.

Recent results in multiuser detection for UWA applications are presented in [46]. The multiuser detection methods proposed in this reference rely on the principles of joint synchronization, channel equalization and multiple-access interference cancellation. The multiuser receivers fall into two categories: centralized, in which the signals of all the users are detected simultaneously (e.g., uplink reception at a surface buoy which serves as a central network node), and decentralized, in which only the desired user's signal needs to be detected (e.g., downlink reception by an ocean-bottom node). Similarly as in the case of interference cancellation, the multichannel receiver presented in Section IV was shown to have excellent capabilities in the role of a decentralized multiuser detector, operating without any knowledge of the interfering signal. A different approach in which an array is used to reduce the MAI prior to equalization of the desired user's signal is investigated in [47]. The multiuser detection techniques, both centralized and decentralized, have so far successfully been applied to vertical channels [48] and to horizontal channels [46], [47]. Array processing plays a crucial role in the detection of multiuser signals, but is also associated with the problem of computational complexity. Research will doubtlessly follow in this area, especially on the issue of reducing the complexity of centralized multiuser detectors.

C. System Self-Optimization

Successful communication in an underwater acoustic system is conditioned on the availability of the channel. Depending on the particular environment in which a system is to operate, the communication channel may be available with varying degrees of reliability. As an example one can think of a shallow water network deployed in a region of high shipping activity. The increase in the background noise level caused by a passing ship will temporarily cut the communications off. Other, less drastic disturbances associated with the channel temporal and spatial variability must be expected to affect the quality of performance as well as to occasionally disrupt it.

The outlined receiver algorithms use a number of parameters which need to be adjusted according to the instantaneous channel conditions prior to turning the receiver on. These parameters include the number and location of array sensors which provide good signal quality, the sizes of the equalizer filters, and their tracking parameters. They may be adjusted by channel sounding and fine receiver tuning, and often require operator assistance. So far, there have been no official recommendations, let alone standards, for the UWA communication system design. However, if the algorithms developed are to be used in autonomous systems, external assistance should be minimized. It is for this reason that self-optimized receiver algorithms should be addressed in the future research. The first steps in this direction are evident in the implementation of the self-optimized LMS algorithms for an adaptive beamformer [34], which eliminates external tuning of the step-size for varying channel conditions, and the periodically updated RLS [32], self-adjusted to keep a predetermined level of performance by increasing the tracking rate if the channel condition worsens. These strategies provide the receiver with the capability to adjust to fine channel changes. However, they depend on the availability of a reliable reference of the desired signal. Since a training sequence is inserted only so often in the transmitted signal, if the receiver losses convergence during detection of a data packet, the entire packet will be lost. An alternative to periodic re-insertion of known data, which increases the overhead, methods for self-optimized or blind recovery may be considered.

Most of the known methods for blind equalization [13] have been developed for synchronous systems, under the assumption that perfect clock and carrier recovery are available, while only the channel and the data sequence are unknown. These assumptions are not realistic for UWA communication systems. A method based on using the cyclostationary properties of oversampled received signals has recently been developed [49], which requires only the estimation of second-order signal statistics to recover the data sequence in the absence of clock synchronization. Originally developed for linear equalizers, this method has been extended to the case of decision-feedback equalizer, necessary for application in UWA channels with extreme multipath [50]. Fig. 5 shows a result of blind equalization of QPSK signals transmitted at 100 sps over a 48-nautical-mile shallow water channel at New England Continental Shelf. In this example, only 200 data symbols sufficed to estimate the channel response which is then used to re-establish the equalizer convergence. Further work on blind system recovery will focus on methods for array processing and carrier phase tracking.

D. Modulation and Coding

Achieving high throughputs over band-limited UWA channels is conditioned on the use of bandwidth-efficient modulation and coding techniques. The results documented in contemporary literature are confined to signaling schemes whose bandwidth efficiency is at most three to four. Higher level signal constellations, together with trellis coding are being considered for use in UWA communications. While trellis-coded modulation is well suited for the vertical channels which have minimal dispersion, their use on the horizontal channels

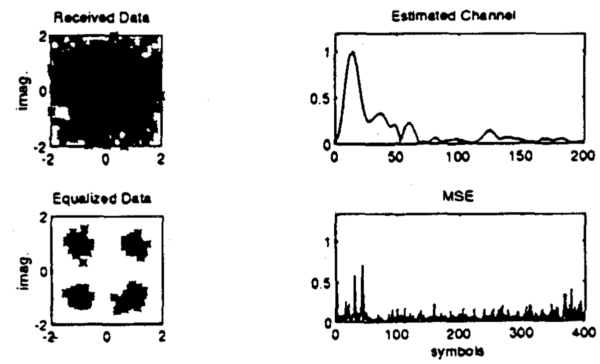

Fig. 5. Blind equalization of long-range shallow water data.

requires further investigation. In the first place, conventional signal mapping into a high-level PSK or QAM constellation may be associated with increased sensitivity of detection on a time-varying channel. Recent results in radio communications show that certain types of high-level constellations are more robust to the channel fading and phase variations than the conventional rectangular QAM constellations [51]. The design of codes to accompany these modulations, however, is yet to be developed. Another issue associated with the use of coded modulation on the channels with long ISI is the receiver design which takes full advantage of the available coding gain. Namely, the delay in decoding poses problems for an adaptive equalizer which needs the instantaneous decisions to be fed back to it. Receiver structures that deal with this problem as it applies to underwater channels have been addressed in [52].

In addition to bandwidth-efficient modulation and coding techniques, the future underwater communication systems will rely on data compression algorithms to achieve high data rates over severely band-limited UWA channels. This is another active area of research, where current achievements [53] report on the development of algorithms capable of achieving compression ratios in excess of one hundred. Efficient data compression, together with sophisticated modulation and coding techniques, is expected to make it possible for the first time to transmit underwater images in real time.

E. Mobile Underwater Communications

The problem of channel variability, already present in applications with a stationary transmitter and receiver, becomes a major limitation for the mobile UWA communication system. The ratio of the vehicle speed to the speed of sound (1/100 for a vehicle speed of 30 knots) many times exceeds its counterpart in the mobile radio channels ($1/10^8$ for a mobile moving at 100 km/h), making the problem of time-synchronization very difficult in the UWA channel. Apart from the carrier phase offset, the mobile UWA systems will have to deal with the motion-induced pulse compression/dilation. Latest results report on successful missions of experimental AUV's that use commercial M-FSK acoustic modems for vehicle-to-vehicle communication [54]. For phase-coherent systems, algorithms for tracking the time-varying symbol delay in the presence of underwater multipath that have so far been proposed include those based on jointly adaptive mmse equalization and synchronization [55] and those based on extended Kalman filtering [19]. However, no experimental evaluations of these algorithms have been reported. Efforts are currently under way to evaluate performance limitations of mobile underwater communications in experimental conditions.

VI. CONCLUSION

In this paper, an attempt was made to summarize the recent advances, as well as some of the directions for future work in UWA communications. A number of references has been provided which demonstrate a tremendous increase in research and development in this area over the past few years. During this period, fundamental advances have been made toward increasing the data throughput and reliability of UWA communication systems. The possibility to achieve phase-coherent underwater communications over a majority of UWA channels has been demonstrated for the first time, opening the door for the development of efficient UWA communication systems.

Since bandwidth-efficient modulation methods offer the possibility of achieving high data rates on the UWA channels, special attention was given in this paper to the design of phase-coherent receivers. A single receiver structure, based on multichannel adaptive signal processing, seems to offer a solution for many of the problems encountered in the UWA channels, including equalization of time-varying intersymbol interference, adaptive noise cancellation, and suppression of multiple-access interference in underwater networks. Current research is moving in the direction of developing techniques for efficient spatial and temporal signal processing to be used in real-time acoustic modems.

Aiming toward the capability to remotely explore the underwater world, many problems remain to be solved in the design of high-speed acoustic communication systems. Nevertheless, recent advances in this area serve as an encouragement for future work, and at this point it can be said that the availability of real-time underwater video from an acoustically controlled vehicle is just a matter of time.

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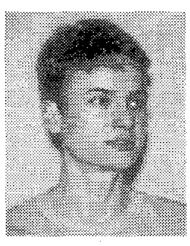

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