The State of the Art in Underwater Acoustic Telemetry

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***Abstract*—Progress in underwater acoustic telemetry since 1982 is reviewed within a framework of six current research areas: 1) underwater channel physics, channel simulations, and measurements; 2) receiver structures; 3) diversity exploitation; 4) error control coding; 5) networked systems; and 6) alternative modulation strategies. Advances in each of these areas as well as perspectives on the future challenges facing them are presented. A primary thesis of this paper is that increased integration of high-fidelity channel models into ongoing underwater telemetry research is needed if the performance envelope (defined in terms of range, rate, and channel complexity) of underwater modems is to expand.**

***Index Terms*—Acoustic signal processing, diversity methods, reviews, underwater acoustic communication, underwater acoustic propagation, underwater acoustic telemetry.**

I. Introduction

THE UNDERWATER acoustic telemetry channel is bandlimited and reverberant which poses many obstacles to reliable, high-speed digital communications. Prior to the late 1970's, there were a few published attempts of acoustic modems. Analog systems were developed, which were essentially sophisticated loudspeakers, but they had no capability for mitigating the distortion introduced by the highly reverberant underwater channel. Paralleling the developments applied to the severely fading radio frequency atmospheric channels, the next generation of systems employed frequency-shift-keyed (FSK) modulation of digitally encoded data [1], [2]. As an energy-detection (incoherent) rather than phase-detection (coherent) algorithm, FSK systems were seen as intrinsically robust to the time and frequency spreading of the channel. The use of digital techniques was important in two respects. First, it allowed the use of explicit error-correction techniques to increase reliability of transmissions. Second, it permitted some level of compensation for the channel reverberation both in time (multipath) and frequency (Doppler spreading). The remainder of the decade saw steady improvements in these systems. As processor technology improved, variants of the FSK algorithm that exploits the increased demodulation speeds were implemented. While signaling alphabets are much larger today, the incoherent FSK modems in use have no fundamental differences from those early ones. However, there have been tremendous strides in hardware design since their introduction. Technical issues such as signal generation, demodulation speeds, and the frequency agility required by high-bandwidth systems (e.g., filters) initially posed serious obstacles but have been largely overcome by the relentless progress in processors. Power efficiency, however, remains a concern for remote transmitters.

Incoherent systems, however, retain a fundamental trait that pressed the scientific community to consider other modulation methods despite the reliability of FSK modulation. The inefficient use of bandwidth of incoherent systems coupled with the limited availability of bandwidth underwater makes them ill-suited for high-data-rate applications such as image transmission or multiuser networks except at short ranges. Larger data rate-range products required the use of coherent modulation.

Communication channels may be coarsely divided into two categories according to the performance-limiting quantity, power, or bandwidth. The division is important in that differing modulation strategies are appropriate for each. While some underwater communication channels are, in fact, power-limited (the long-range low-rate SOFAR channel being one example), most telemetry applications are bandwidth-constrained. As such, bandwidth efficient coherent signals play a central role in current research. The emergent use of coherent phase-based systems in the last decade is quite surprising considering the prevailing view in the early 1980's that the time variability and the dispersive multipath of the ocean simply would not allow such modulation schemes. The potential improvements in bandwidth efficiency (data rate/signal bandwidth) stimulated researchers to challenge this view, especially with the rapidly developing capabilities for high-speed digital signal processing.

The early 1990's have yielded a plethora of published coherent systems that moved acoustic telemetry into the horizontal ocean channel. The seminal work [3], [4] succeeded due to the use of a powerful receiver algorithm that coupled a decision feedback adaptive equalizer with a second-order phase-locked loop. Using quadrature phase-shift-keyed (QPSK) modulation, a data link of 1000 bit/s at 90 km was demonstrated. The frontier of underwater telemetry now finds researchers attempting to communicate in ever more challenging channels such as littoral areas and surf zones. If attempts in such dynamic environments are to succeed, the community must increase its understanding of the temporal and spatial coherence of signals with the bandwidths and frequencies typical of telemetry waveforms.

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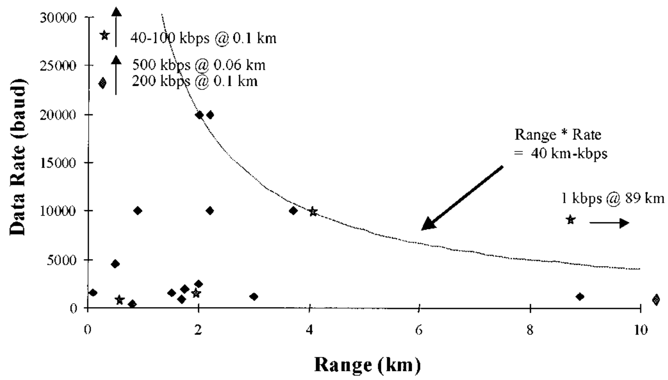


Fig. 1. Published experimental performance of underwater acoustic telemetry systems is summarized in this plot of range (kilometers) versus rate (kbit/s). The channels vary from deep and vertical to shallow and horizontal. In general, the high rate or high range results are for deep channels while the cluster of low range, low rate results are for shallow channels. Modems developed by the research community are represented with diamonds while stars denote commercially available systems. The range*rate bound represents an estimate of the existing performance envelope. While there are exceptions, most reviewed systems are bounded by this performance limit.

One metric for comparing telemetry systems is the maximum attainable range–rate product. The published results of many modems with both academic and commercial origins are summarized in Fig. 1. The reader is cautioned that this figure is based on a large aggregate of channels, some challenging and some not. Focusing on the historical growth of data rate and range, however, does not bring to light other important advances in the underwater acoustic telemetry field.

While there have been several summary articles published in recent years [5]–[8], we seek to capture the published literature in the context of specific underwater telemetry research avenues. As such, we will now examine the progress and challenges in the areas of channel physics, equalization and carrier recovery, diversity exploitation, error control coding, underwater networks, and alternative modulation strategies.

II. TELEMETRY AND CHANNEL PHYSICS

Acoustic telemetry over most channels encountered in the ocean involves propagation through a random time-varying medium. Seldom is the additive Gaussian noise channel model appropriate for representing signal propagation at telemetry carrier frequencies. Perhaps the only exception to this are short-range direct path channels in the deep ocean. The literature on propagation in random time-varying media is extensive, and much of it is too detailed to be useful as a robust representation for telemetry systems. In this section, we discuss aspects of the channel physics which are especially germane to acoustic telemetry. There are, however, many additional aspects of the channel physics which must be included in the design of a telemetry system. Important ones are the ambient noise environment and transmission loss; these are well documented [9]. The dominant features of the acoustic telemetry channel are: 1) it is bandlimited due to absorption with most systems operating below 30 kHz and 2) it is reverberant with spreading in both travel time and Doppler for all but a few systems operating over short ranges along a single direct path. Just as in telemetry over electromagnetic channels, there is no single design of an acoustic telemetry system appropriate for all environments. Maximum ranges and data rates will both be functions of the channel physics. Much of this was not well recognized in the design of early telemetry systems where electromagnetic systems were cloned for underwater applications.

There have been numerous simulations of acoustic telemetry channels which attempt to incorporate the channel physics to varying degrees of fidelity, e.g., [10]–[18]. Many other papers analyzing algorithm performance also contain simulations. Unfortunately, high-fidelity modeling of the channel physics is complicated at the ranges, carrier frequencies and bandwidths, and variability scales which are relevant. Direct simulation based upon ray paths needs to include many micropaths and account for boundary and internal scattering which can rapidly lead to ray chaos; modal representations at telemetry frequencies and bandwidths are prohibitively large and do not easily incorporate range dependencies. Many of these codes, e.g., SAFARI/OASIS implicitly assume coherence among the multipaths which is most likely not satisfied at telemetry frequencies; alternatively, parabolic equation simulations across the bandwidth of a telemetry system are numerically intensive for the rates at which the multipath structure of the medium changes. Most of the propagation codes developed for low frequencies must be used with some care. First, these codes typically compute a channel transfer function and/or impulse response. Spatially, they implicitly assume full coherence among all the rays or modes excited which is usually not appropriate at telemetry frequencies. Temporally, the coherence across frequency, especially assumptions about the phase, is seldom addressed. Effective signal design for an acoustic telemetry system requires an understanding about the spatial and temporal coherence of the channel. In addition, these codes do not typically introduce time variability. One can understand why this is often so because of the extensive computations required for a full bandwidth simulation at telemetry frequencies; nevertheless, the adaptation capability of a system is essentially embedded in how it responds to time variability. There have, however, been some attempts made for realistic channel simulations using Gaussian beams and similar approximate representations [19], [20]. Temporal fluctuations are induced by source/receiver motion and medium variability, and modeling these is important to understand adaptation issues. Spectral representation for scattering such as those developed for internal waves [21] and surface waves [22] have been used, e.g., [6]; these have been relevant primarily to deep water where good acoustic models of the scattering are available. Bottom roughness and the scattering it induces is also important. First, the bottom introduces travel time spreading when it is rough; secondly, it couples to Doppler spreading when source/receiver motion is present since it modulates the impulse response of the channel randomly.

Communication over random time-varying media where the signal is time spread, or extended in time, and Doppler spread where the signal is frequency spread, or extended in bandwidth, is a well-studied problem in radar, electromagnetic communications, sonar, and radio astronomy. There are several useful characterizations which have as their basis the channel being represented as

$$r(t) = s_r(t) + n(t) \tag{1}$$

where $r(t)$ is the observed signal, $n(t)$ is an additive noise term,[1] and $s_r(t)$ is the output of a system with a random time-varying impulse response, or

$$s_r(t) = \int b(t, x) s_t(t - x)\, dx \tag{2}$$

where $s_t(t)$ is the transmitted signal, e.g., a set of pulsed tones for an multiple phase shift keying (MFSK) system or a differential phase shift keying (DPSK) modulated waveform. The channel is modeled with a random impulse response $b(t, x)$, where the first argument t incorporates the time variability, often referred to as the Doppler index and x as the range, or travel time, index.

This model has a long history for representing random channels. It was first used for modeling the reverberation found in active sonar [23]; the report by Price and Green on a communication technique for multipath channels is a classic development of the model wherein the "Rake" receiver was first described [24]. Additional applications may be found in radar astronomy [25]. Several concise summaries may be found in a review article by Bello [26] and the texts by Kennedy [27] and Van Trees [28]. In spite of having a rich history of its application in modeling communication over electromagnetic channels, its absence is noteworthy in the recent work on communication over the underwater channel. The literature, in fact, contains numerous approaches to represent channel effects. Many of these are specific to the modulation signals employed, so it is difficult to extrapolate results to other signals. Impulse responses have been measured, but little has been done to convert them to stochastic measures which are needed for robust system design. Conversely, acousticians have made numerous models of transmission loss, coherence, and fading, as well as impulse responses, but they have not been in a form useful for designing a telemetry system.

For digital systems, it is useful to use a discrete representation, or

$$s_r[n] = \sum_m b[n, m] s_t[n - m] \tag{3}$$

where $s_t[n]$ is the modulated signal. The sampling rate is often chosen to be at the symbol rate; however, the rate may be an integer multiple of the symbol rate when fractional equalizers are implemented or even dynamically adjusted to compensate for Doppler shifts.

The second moments of $b(t, x)$ are quantities which describe how the channel spreads, or redistributes, power in Doppler and travel time, and they can often be related to the propagation physics. They provide a stochastic model for the spreading processes. The temporal correlation of the output signal is given by

$$\begin{aligned} K_{s_r}(t_1, t_2) = \iint & K_b(t_1, t_2, x_1, x_2) \\ & \cdot s_t(t_1 - x_1) s_t^*(t_2 - x_2)\, dx_1\, dx_2 \end{aligned} \tag{4}$$

where

$$K_b(t_1, t_2, x_1, x_2) = E[b(t_1, x_1) b^*(t_2, x_2)] \tag{5}$$

is the covariance of the time-varying impulse response.[2] Two assumptions, termed the wide sense stationary, uncorrelated scattering (WSSUS), are made which are usually appropriate in acoustic telemetry. The first assumes that scattering at two travel time delays x_1 and x_2 is uncorrelated. At the frequencies used for acoustic telemetry systems, this is usually appropriate. Signals arriving at different travel times have propagated over distinct paths, the multipaths, and therefore have been subjected to different random inhomogeneities. This assumption can breakdown at low frequencies where wave phenomena become important or for very short separations in travel time where the paths coalesce. The second assumes that the time variability represented by the Doppler index t is a wide sense stationary random process. This is also usually appropriate for acoustic telemetry. The time variability is introduced as the result of either: 1) wave motion, such as surface or internal waves, or 2) source/receiver motion which often can be separated into a mean Doppler shift introduced by the mean range rate between the source and receiver plus a random fluctuation about this mean. In addition, the motion modulates the impulse response between the source and receiver, and this is typically a complicated function at the carrier frequencies employed. These lead to

$$\begin{aligned} K_b(t_1, t_2, x_1, x_2) &\to K_b(t_1 - t_2, x_1) \delta(x_1 - x_2) \\ &= K_b(\tau, x_1) \delta(x_1 - x_2) \end{aligned} \tag{6}$$

where $\tau = t_1 - t_2$. One may consider the quantity $K_b(\tau, x)$ to represent an ampltitude modulation of the complex transmission loss with a travel time delay x. Physically, this suggests a random superposition of paths with essentially the same travel time at the resolution scales consistent with the bandwidth of the transmitted signal.

There are several Fourier transforms of K_b which have useful intuitive properties for modeling the acoustic telemetry channel. The scattering function, also termed the delay Doppler power spectrum, is given by

$$S_b(f, x) = \int K_b(\tau, x) e^{-j2\pi f\tau}\, d\tau. \tag{7}$$

[1] Typically, acoustic telemetry systems are "reverberation, or clutter, limited" and not "noise limited." Hence, additive noise is not the dominant concern for most acoustic telemetry systems except for covert ones where low probability of intercept is an issue. It is, nevertheless, important in determining the performance of a telemetry system.

[2] In characterizing the channel, s_t is the basic signaling waveform from which the entire message sequence is constructed by modulating it in some form. The signal s_t may be a pulsed tone, a sequence of pulses assembled according to a code, or a combination of tones, again with some form of coding. Modulation methods which have been used for acoustic telemetry include phase shift keying (PSK), frequency shift keying (FSK), and amplitude modulation (AM).

This function describes how the signal power is redistributed in range x and frequency f, i.e., an impulsive input is distributed, or spread, along the travel time axis according to

$$S_b(x) = \int S_b(f, x)\, df \tag{8}$$

and a tone is distributed, or spread, according to

$$S_b(f) = \int S_b(f, x)\, dx. \tag{9}$$

The scattering function is an important characterization for coherent modulation systems. Its travel time extent determines the number of taps required for equalizers to span the multipath spread; moreover, the distribution within this span indicates the multipath arrival structure where one can allocate the tap spacing and delete taps where there is no power. This is the concept underlying sparse equalizers. In addition, the Doppler spread at each range delay indicates the bandwidth required, hence the time constants for any adaptive equalizer. The time-varying impulse response model and the WSSUS assumptions need to be applied with some care in wide-band acoustic telemetry systems. First, the effects of source/receiver motion may not be well represented by a simple Doppler shift. A common criterion for this is

$$(WT)^{-1} \gg \frac{v}{c} = M \tag{10}$$

where WT is the time–bandwidth product of the signal or, more specifically, of the matched filter in the front end of the receiver, and v/c is the ratio of the source/receiver motion to the speed of sound or the acoustic Mach number. Since $c = 3000$ kn, v/c is usually approximately 10^{-3}, so relatively large time–bandwidth products can be represented. This extends to assumptions implicit in the WSSUS model since it is based upon a Doppler shift representation.

Similarly, one obtains the two-frequency correlation function by transforming with respect to the travel time index, or

$$H_b(\tau, \nu) = \int K_b(\tau, x) e^{j2\pi\nu x}\, dx. \tag{11}$$

This function describes the correlation between the complex envelopes for two sinusoids separated by ν at a lag τ, i.e., two sinusoids sampled at the same time have a correlation $H_b(0, \nu)$ and a sinusoid has a complex envelope with correlation $H_b(\tau, 0)$. The two-frequency correlation function is particularly important for MFSK systems since it indicates the minimal spacing between the tonal components needed to maintain statistically uncorrelated envelopes for diversity.

The scattering function, the two-frequency correlation function, and other channel characterizations related by Fourier transform [26] are independent of the transmitted signals used once the carrier frequency is established, although probing or measuring them does depend upon the signal used; hence, they are robust and can be employed to evaluate competitive signaling strategies. If one further assumes Gaussian statistics, which is often appropriate, then probability densities, fading and phase fluctuation rates, and other quantities of interest can be determined. The most significant advantage of the scattering function model is that it provides a robust signal-independent model for channels operating at a given carrier frequency and bandwidth. One might argue that such a model is not available *a priori*; while this is often the case, the scattering function provides a framework for environmentally adaptive systems where it may be measured dynamically.

A central issue for designing acoustic telemetry systems is to match the channel physics to the parameters of the system. Beyond the well-understood issue of attenuation *versus* frequency [29] and how it establishes a "curtain" for the maximum range, typically set at [30]

$$\alpha(f_0)R < 10 \text{ dB} \tag{12}$$

where $\alpha(f_0)$ is the attenuation at the center frequency and R is the range, there are a number of other aspects of the propagation which impact the scattering function and two-frequency correlation function.

The impact of the travel time and Doppler spreads for a signal is often classified by being underspread or overspread. This is often classified in terms of a travel time spread channel, a Doppler spread channel, or a doubly spread channel. Typically, acoustic telemetry systems are spread in travel time and sometimes in Doppler as well.

The important scaling for the travel time spreading depends upon the reciprocal of the extent of the multipath spread. For shallow-water channels, this is typically of the order of 100 ms, implying a frequency correlation length of 10 Hz. The signal components which propagate efficiently are below the critical grazing angle at the seafloor, so the late-arriving high-angle components are strongly attenuated. Moreover, absorption losses impose an exponential range curtain primarily due to boundary interactions, which is a strong function of the geology of the bottom in the 10–20-kHz band typical for acoustic telemetry. If one wants to operate lower carriers and accept the lower bandwidths, relatively long ranges (~100 km) have been obtained [4].

Deep-water channels separate into two categories according to the range/depth ratio. For small ratios, the multipath spread is quite small, especially if the surface and bottom reflected paths are baffled; consequently, the two-frequency coherence is on the scale of kilohertz. This channel is probably closest to the classical memoryless additive white Gaussian noise channel. In fact, some of the highest published data rates have been obtained over this channel. Long-range, i.e., large range/depth, ratios are time spread as a result of signficant multipath. The extent of the spreading is a strong function of the depths of the transmitter and receiver. If the transmitter is near the sound channel axis, all rays and/or modes are excited, especially the late-arriving axial ones. If the receiver is similarly positioned, it records all these paths as well. Although one might suggest not operating near the axis, these are the paths which propagate over long ranges with the least amount of transmission loss, hence, leading to higher signal-to-noise ratios (SNR's). Transmitters and/or receivers positioned near the surface have less multipath spread since the late-arriving axial rays/modes are not coupled.

Applied directly, these long multipath spreads imply two-frequency coherence on the scale of 1 Hz since, at frequencies used for telemetry, the ray paths/modes are uncorrelated; however, there is a subtle issue if one resolves the individual paths. At the temporal scales of the resolved paths, the frequency coherence scales are much larger, typically of the order of 100 Hz.

If a channel has a Doppler spread with bandwidth B, it has a fading time constant on the order of $1/B$; therefore, if a signal has symbol duration T, then there are approximately BT uncorrelated samples of its complex envelope. When BT is much less than unity, the channel is said to be underspread in Doppler, while, if greater than unity, it is overspread. When underspread, the effects of the Doppler fading can be ignored; however, one still must track mean Doppler shifts due to source/receiver motion for demodulation in coherent systems and shifting frequency bins in incoherent systems. If the channel is overspread, one needs to pursue incoherent combinations of the uncorrelated components of the envelope. For incoherent MFSK telemetry systems, one wants to use tone durations less than $1/B$ which typically can be done. For coherent systems, the individual symbol duration is underspread, and $1/B$ sets the adaptation rate for the equalizer.

There are several references where the issue of underspread *versus* overspread signaling is discussed [24], [26]–[28], [31]. The choice of signaling falls within two domains, which are characterized by the available bandwidth and SNR. While appropriate for the additive white Gaussian noise channel, Fig. 2 (taken from Proakis [31]) represents the choice well. Here the channel capacity as a function of SNR per bit (E_b/N_0) is plotted as well as the bandwidth efficiency (R/W) of several modulation methods at an E_b/N_0 corresponding to a symbol error probability of 10–5. The channel capacity C is given by

$$\frac{C}{W} = \log_2\left(1 + \frac{CE_b}{WN_0}\right) \tag{13}$$

where W is the bandwidth, E_b is the energy per bit, and N_0 the noise spectral density. A system operating with $R/W < 1$, where R is the data rate (bit/s), is appropriate for a power-limited channel as the modulation uses excessive bandwidth but may operate at lower power levels. The orthogonal signaling methods in underwater telemetry usually rely on incoherent rather than coherent detection methods with implications for the required E_b/N_0. If $R/W > 1$, the modulation methods (amplitude and phase modulation generally) are suitable for bandwidth-limited channels as they use bandwidth efficiently but require excessive power compared to orthogonal signaling.

The nominal Doppler shift induced by source/receiver motion is given by

$$\Delta f_d \approx 0.35 \text{ Hz/(knot} * \text{kHz)}. \tag{14}$$

The impact of transmitter/receiver motion is seldom simply a shift since there are usually several paths which couple them, so there is a spread of Doppler shifts which is determined by the spread of the phase delays of the several paths. There are two effects associated with Doppler spreading that need to be differentiated. In the first, there is a simple frequency translation which is relatively easy for a receiver to compensate for. In the second, there is a continuous spreading of frequencies which leads to a truly Doppler-spread, not shifted, signal. It is far more difficult for a receiver to compensate for this effect.

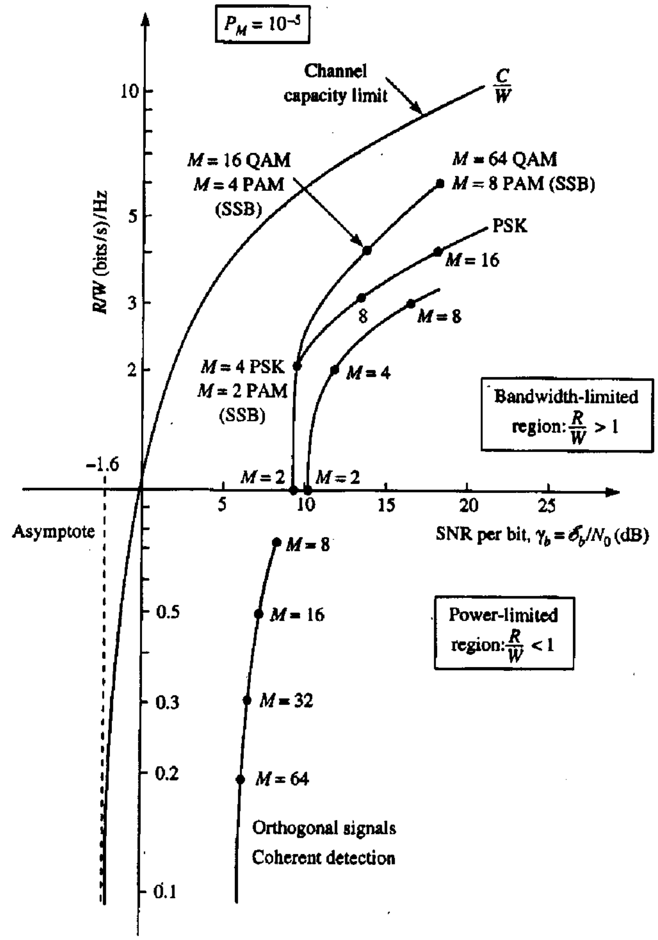


Fig. 2. Bandwidth efficiency η_b in bit/s/Hz as a function of SNR per bit 9(energy per bit/noise) as given in [31]. Data is given at a constant error probability of 10—5. Orthogonal signaling (FSK) is shown with η_b less than 1 while "efficient" methods such as PAM, QAM, and PSK are shown with η_b greater than 1. [Reprinted with publisher's permission.]

Both the sea surface and/or the sea floor may be rough which leads to travel time spreading. The usual scale of comparison for when this roughness becomes significant is the Rayleigh parameter

$$s = 2\pi \frac{h_w}{\lambda} \sin\theta_0 \tag{15}$$

where h_w is the surface rms roughness and θ_0 is the grazing angle. When $s \ll 1$, the surface is smooth and responds as an "acoustic mirror" and one has coherent multipath interference. Alternatively, for $s \gg 1$, the surface is acoustically rough and differential range spreading is introduced. In the case of the sea surface, Doppler spreading which depends on the temporal spectrum of the sea surface may also be introduced.

The sea surface and its associated processes such as bubble entrainment and source/receiver motion are the dominant mechanisms leading to Doppler spreading. At high sea states, the surface is rough. There is an extensive literature on sea-surface Doppler effects for telemetry frequencies [32]. For telemetry systems, fluctuation bandwidths were suggested in [7] based upon results of [33] caused by the motion of the sea surface. These are given by

$$B_w = 2f_w\left[1 + \frac{4\pi f_0 \cos\theta_0}{c} h_w\right] \tag{16}$$

where

w	wind speed in m/s;
$f_w = 2/w$	the wave frequency in hertz;
$h_w = 0.005w^{5/2}$	the wave height in meters;
f_0	carrier frequency in hertz;
θ_0	incident grazing angle;
c	sound speed in m/s.

The author suggests that, for successful adaptive tracking, one needs to underspread by $B_w T_s < 10^{-2}$ [7]. Typical carrier frequencies around 10–20 kHz and 1–2 kbit/s lead to this criterion being satisfied for modest sea states. (The issue of the number of degrees of freedom in an equalizer is also involved; equalizers with a large number of taps are more complex and tend to track at a slower rate.) This is representative of the scale of spreading; however, there are a number of processes such as Bragg scattering at the lower frequencies, etc., that also influence the extent of Doppler spreading.

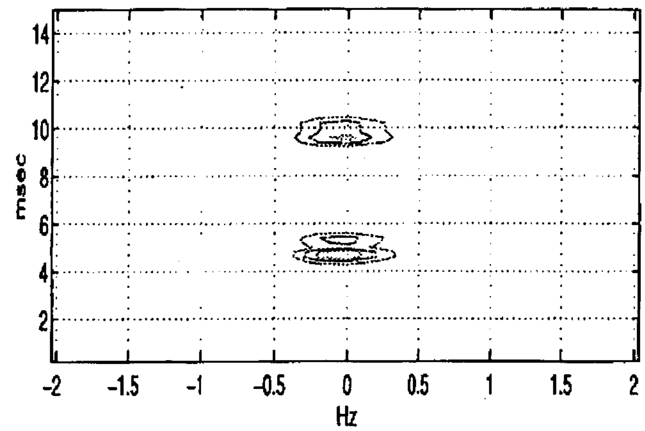
Source/receiver motion effects are an important mechanism since only a small change in separation need occur to traverse a wavelength which then modulates the multipath interference. Typically, the paths of concern are the interference between a direct and a surface-reflected path. In the case of an isovelocity sound speed field with the source and receiver at the same depth, the phase difference is given by

$$\Delta\phi = 4\pi \frac{f_0 d \sin\theta_0}{c} = 4\pi \frac{d \sin\theta_0}{\lambda_0} \tag{17}$$

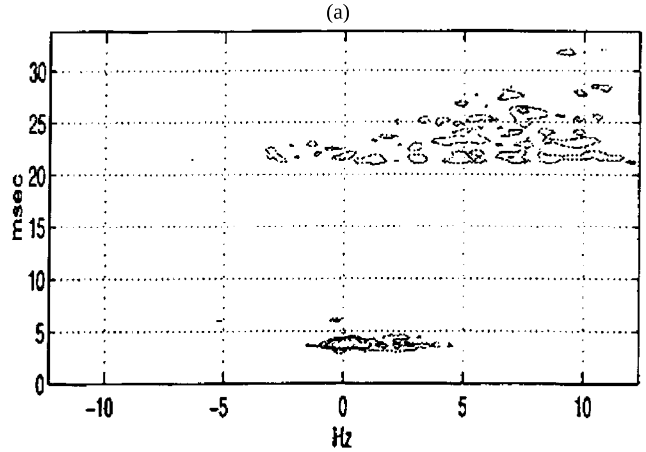
where d is the depth of the receiver and λ_0 is the wavelength, typically on the scale of 10–30 cm for telemetry frequencies. Consequently, very small changes in either depth (source or receiver moving with a surface float or boat) or apparent grazing angle (change in source/receiver range) lead to substantial interference fluctuations.

Several authors have used acoustical propagation models in attempts to provide characterizations which can be coupled to environmental parameters. Some of these are accompanied by supporting experimental data. References [6] and [11] introduce the Λ–Φ parameterization used to characterize fluctuations from internal waves on a single path in terms of saturated, partially saturated, or unsaturated. They then argue that the multipath environment in deep water is an uncorrelated superposition of individual paths. This model is appropriate for totally refracted paths in the deep ocean which encounter neither the sea surface nor seafloor.

A number of authors have published channel characterization papers. Rice [34] summarizes the signaling issues for acoustic telemetry systems. He tabulates the conditions for under versus overspread conditions, but does not include any channel physics or data as part of his discussion. Badiey measured Doppler spreads (coherence times) for two short range sites (approximately 200 m) one off New Jersey, the other in



(a)



(b)

Fig. 3. (a) This scattering function was measured in an Arctic environment (ice-covered surface with no bottom interaction) using a telemetry signal with a 15-kHz carrier and 2.5-kHz bandwidth. Maximal length sequences were used as the probing signal. (b) This scattering function was measured in deep water near the Bahama Islands on a windy day. The frequency spreading induced by the rough sea surface is evident. The contours are in 3-dB increments.

Delaware Bay from tripods, hence eliminating source/receiver motions [35]. The data are for relatively low frequencies, up to 3 kHz. At 1-kHz coherence times are very long, on the scale of 100 min, however, just a shift to 3 kHz led to coherence times on the scale of 1–2 min. Nevertheless, these are quite long for a telemetry system. He also identified the multipath arrival structure and monitored through the tidal cycle, demonstrating the modulation induced by the height of the water column. In [36], the authors set up a time-varying impulse model for the telemetry channel, essentially duplicating the original formulation of the scattering function without referencing the wide-sense stationary uncorrelated scattering assumptions.

The best collection of scattering function measurements from a number of locations were made in [18]. Several important physical phenomena are indicated including the sparse multipath, Doppler spreading due to surface waves, and source motion. Two examples of these measurements are shown in Fig. 3. These were then used to establish the parameters for both incoherent MFSK and coherent DPSK receivers.

Several papers have presented the time variability of the channel impulse response. In [37], the impulse response for a short-range (1 km) shallow-water path (Woods Hole Harbor, MA) was measured. These results were instrumental in the development of the coherent, adaptive equalizers. Similarly, in [38], channel impulse responses were estimated for propagation down slope between a fixed source and receiver. In addition, probability densities for the envelope and phase were also measured which are consistent with what one expects for Gaussian complex envelopes.

III. Underwater Channel Simulations

An increasing number of researchers have begun to use underwater acoustic channel simulations as an adjunct to their modem developments. The purpose is commonly to aid in evaluation of signal processing algorithms in an attempt to increase the success of field experiments. Less common are attempts to use these models to explicitly relate time-varying ocean processes to telemetry performance and gain true insight. The distinction will become clear when we examine how physical effects are usually incorporated into these models. While there are numerous modeling techniques for underwater acoustic wave propagation including modal decompositions, parabolic equation methods, wavenumber integration algorithms, and finite difference solutions, the telemetry community has focused almost exclusively, and appropriately, on ray theory. In the following survey of published channel simulations, the discussion will be loosely grouped on how time variability is accounted for in the models. A short commentary on future modeling challenges will follow.

The simplest simulations are based on an entirely deterministic ray trace that may or may not account for movement of the transmitter, receiver, or sea surface. In a series of papers, Zielinski *et al.* model a shallow-water constant sound-speed channel [15], [39], [16]. Boundary interactions are characterized by a reflection coefficient determined by sea surface roughness and ray angle. Time variability is not explicitly considered. Zielinski introduces the signal-to-multipath ratio (SMR) as a metric that governs coherent system performance in the absence of equalization. Based on the impulse response, he defines the SMR as the ratio of coherent power in a symbol interval around the direct arrival (the signal) to the coherent power outside of the interval (the multipath). The issue of defining suitable metrics that may parameterize telemetry system performance is a significant challenge facing the community in the coming years.

As a next step in complexity, several researchers have developed models that associate a frequency shift with each eigenray based on defined boundary and platform motions. Using a three-dimensional (3-D) ray trace algorithm, Appleby describes a model that generates successive realizations of an impulse response for a channel with a moving boundary and endpoints [10]. The model evaluation is limited to a single sequence of impulse responses evolving over some tens of seconds for a particular geometry. Essebbar chose to focus on the impact of transmitter motion for systems employing arrays of high directivity [12]. In this model, a piecewise linear sound velocity profile with no range dependence forms the channel. The sole linkage to channel dynamics is contained in the relation between array pointing variance and swell frequency and amplitude. In a later article, the model is enhanced to allow for motion-induced Doppler effects [13]. These simple models are valuable for investigating channel features such as total temporal and frequency spread. Such information is essential in designing a modulation strategy for a specific channel. More complex issues such as adaptive algorithm performance and bit error rates (BER's) rely directly on accurate channel statistics which are considerably more difficult to model.

As an intermediate step to direct modeling of time-varying ocean processes, some researchers have chosen to make statistical assumptions for the individual arrivals in a given impulse response. For instance, a common premise is a Rayleigh distribution for amplitude and a Gaussian distribution for phase or arrival time, or equivalently a complex Gaussian envelope. Using precisely this premise, Galvin [40] simulates a 14-m-deep isovelocity channel over a 900-m range with realistic bathymetry. For that example, however, the predicted BER was an order of magnitude worse than the measured rate. Geng assumes Ricean statistics for the amplitude and a decaying two-sided exponential distribution of arrival time for each eigenray [14]. While the motivation for the simulator structure is discussed in depth, no model results are given. Gray *et al.*randomly and independently attenuate each arrival according to a Gaussian probability density function [41]. By doing this for successive realizations, frequency spreading is simulated. By introducing time variability explicitly into their simulations, these researchers are able to approximate physical effects that, in many cases, pose the limiting obstacle to successful telemetry. There are, however, two notable drawbacks to an assumed statistical model. First, the distribution moments are free parameters and appropriate values are simply not known for many channels. Second, error events under benign conditions are typically dominated by extreme events associated with the tails of the distribution. While the statistics near the mean may be well approximated by Rayleigh, Ricean, or Gaussian statistics, the diversity of noise and distortion sources in the ocean channel may lead to subtle but important statistical differences away from the mean. Therefore, one must ensure that the simulations are consistent with measured data for extremal events, i.e., the tails of the distribution.

Models with an explicit incorporation of true ocean dynamic processes are beginning to be presented in the open literature. The interaction of rays with a moving, wind-forced sea surface featured in a ray-theory-based simulation developed by Eggen [18]. While the derived (rather than assumed) statistics still depend on specified parameters, utilization of a physical ocean model allows parameter selection based on a relatively wide body of experimental work. Plaisant examined the perturbation effects of internal waves [42]. The depth and travel time variance of individual eigenrays is analytically derived from the prescribed internal wave spectrum. A sequence of impulse response realizations is then generated based on the derived eigenray statistics. A similar approach for including turbulence effects is introduced by Bjerrum-Niese [11]. Amplitude and phase variances for each eigenray are derived from an assumed turbulence process in accordance with the methodology discussed by Flatte [21]. While the explicit linkage between models and ocean processes is a significant advance, we find two concerns with the methodology. First, by deriving statistics for each individual ray, an implicit assumption is made that the rays are uncorrelated over any time interval (uncorrelated scatter). Considering the prevalence of adaptive, coherent receiver algorithms, the coherence of signals and the validity of the WSSUS model are critical concerns not adequately addressed by these models. A second related issue is the need to model the true bandwidth of the dynamic process. As an example, a slow frequency wander and a nearly instantaneous frequency spread have vastly different implications even if the process variances are the same.

In summary, underwater acoustic channel simulations are an important emerging research area. Several modeling aspects are clear candidates for increased emphasis. Metrics, such as SMR, must be defined to parameterize telemetry system performance. Explicit linkage between physical ocean processes and wave propagation are needed if the models are to augment the value of field experiments. Finally, adequate treatment of signal coherence (time variability of the channel) may require substantial modification of existing modeling approaches.

IV. Receiver Structures

While the substantial attenuation of underwater communication signals as well as pervasive noise sources (anthropogenic, biological, and wave phenomena) often conspire to reduce available SNR, the phenomenon of reverberation, in both time and frequency, has tended to dominate the evolution of receiver strategies for underwater acoustic telemetry. As becomes evident from the following review, incoherent receivers have generally sought to avoid reverberation issues using classical methods while coherent receivers have struggled to accommodate reverberation with new powerful adaptive algorithms. This dichotomy of recent research effort was apparent in the literature survey performed in support of this review in that only a single publication was found to focus on an incoherent receiver algorithm while 28 publications emphasized a proposed coherent receiver algorithm. Overcoming reverberation effects in the pursuit of higher data rates, in fact, has been and continues to be a consistent theme. In this section, we separately describe the evolution of incoherent and coherent digital underwater acoustic communication receiver architectures in the last 15 years while citing relevant publications. System performance measures such as data rate, maximum range, and error rates along with an environmental description were often given in the publications. As such, the historical growth of performance is noted along with the evolution of receivers.

A. Incoherent Digital Receivers

Owing to the predominantly linear nature of propagation in the underwater channel, the frequency content of telemetry signals remains largely contained within its original band whereas the amplitude and phase of the signal can vary widely in both space and time due to reverberation effects as well as fluctuations in the water properties. That observation naturally led to the use of FSK as the incoherent modulation method of choice.[3] FSK systems use distinct tonal pulses to denote digitized information. Decisions regarding which tones are sent are based on energy detected at the output of narrow-band filters. These filters may be implemented in analog or digital form. The band edges are derived from the known modulation and, in some systems, modified to account for any detected Doppler shift. Early system designs and implementations used low bit rates (<200 bit/s) [1], [43]. The low rates were driven both by the relatively low data rate requirement of command and control missions as well as the computational capability of existing digital processing hardware. Morgera's simulated design already employed the classical techniques of guard times (delays between reuse of a tone to allow temporal reverberations to expire) and multiple frequency diversity (the simultaneous use of several tones to combat fading, or coherent destructive interference, on any single tone). His simulated channel was characterized by a transmission loss, a temporal spreading between 0.5 and 2.4 s, and a frequency spreading between 0.02 and 0.2 Hz. Garrood describes 40 bit/s experimental performance in 50 m of water out to 2 nautical miles under quiet conditions.

The Digital Acoustic Telemetry System (DATS) [2], [37] offered data rates up to 1200 bit/s in conjunction with channel coding for error protection. The tonal codewords had the desirable property of being of equal energy [44]. These results were obtained in the harbor at Woods Hole, MA. While the underlying receiver strategy was identical to earlier implementations (guard times and frequency diversity), the additional performance was largely attributable to a higher center frequency as well as a more powerful microprocessor allowing the use of multiple frequency shift keying (MFSK) as well as a phased array beam. Frequency hopping in this system mitigated the data throughput penalties of guard times.

An extremely low-data-rate (~2 bit/s), presumably highly reliable, private industry system was developed for monitoring and control of oil wellheads [45]. Redundant transmissions and a restricted codebook composed of four tone "words" led to derived statistics, namely a 0.95 probability of receiving a correct command and a 10^{-7} probability of executing a false command. Anecdotal results are given suggesting adequate performance in 100-m depths out to 2 nautical mile ranges.

A 75-bit/s FSK system using 1.5 kHz of bandwidth centered on 12 kHz is described [46] for transmission of sensor data from a platform buried in sediment on the ocean floor. The error performance of the so-called "Super Doppler" system varied widely during sea trials. No channel probes were available to allow explanation of the error rate. The use of continuous frequency sweeps rather than multiple discrete tones for underwater telemetry was first introduced by Hill [47]. The benefits of a sweep are essentially identical to those derived from MFSK systems (e.g., robustness to fading) although the hardware implementation may be simpler. Two current commercial systems, produced by Edgetech, Inc., and Orca Instrumention, respectively, also employ frequency sweeps in their modulation scheme. Despite the nearly continuous work on incoherent digital receivers during the 1980's, no substantive improvements in incoherent receiver design, except increased complexity (largely introduced through increased alphabet sizes) owing to increased processing power, were presented in the 1980's.

[3]While we will focus on telemetry missions, FSK architectures were, and still are, widely used to trigger acoustic release systems. Emphasizing reliability rather than data rate, development of acoustic releases has not been a driver in acoustic modem development.

TABLE I
SUMMARY OF SEVERAL SALIENT METRICS FOR INCOHERENT TELEMETRY SYSTEMS IS TABULATED HERE FOR EACH OF THE REFERENCED SYSTEMS

Principal Investigator	Data Rate (bps)	Coding Scheme (Redundancy)	Bandwidth (Hz)	Bandwidth Efficiency	Range (km)	Prob. of Error
Morgera (1980)	0.5	Golay Code	50	0.01	N/A_{SIM}	N/A
Garrod (1981)	40	Tones/Bit >1	N/A	N/A	4.0_S	$<10^{-2}$
Catipovic (1984)	1200	Hamming	5000	0.24	3.0_S	$\sim10^{-2}$
Jarvis (1984)	<2.3	Time Repeat	6000	$<4.0e^{-4}$	2.0_D	N/A
Coates (1988)	75	None	1500	0.05	5.0_D	$\sim10^{-3}$
Hill (1988)	360	Tonal Chirp	5500	0.07	6.0_D	N/A
Freitag (1990)	2500	Convolutional	20000	0.13	3.7_D	$\sim10^{-4}$
Freitag (1991)	600	Tones/Bit > 1	5000	0.12	2.9_D	10^{-3}
Mackelburg (1991)	1250	Tones/Bit > 1	10000	0.13	2.0_D	N/A
Scussel (1997)	2400	Hadamard	5120	0.47	10.0_{SIM}	N/A

N/A indicates the data was not available in the published reference. SIM indicates simulated results rather than experimental. Ranges with an "S" subscript indicate a shallow-water result while a "D" subscript indicates a deep-water result, typically a vertical channel. Error probabilities are simply typical of what the authors report. Although bandwidth efficiency is redundant given the data rate and bandwidth, it is included to emphasize the inefficient nature of these systems.

In the last ten years, incoherent receiver technology development has been marked by ever more efficient and powerful hardware. Reliable long-term autonomous operation was the objective of one Woods Hole Oceanographic Institution (WHOI) experiment [48] with a 6-month mooring deployment of a 600-bit/s system described. The Utility Acoustic Modem (UAM) under development at WHOI, under the sponsorship of the Office of Naval Research, represents the state of the art in compact hardware design featuring a 42- in^3 form factor, 100 mW of standby power, and an onboard 60-MHz TMS30c44 processor. Designed for autonomous vehicle and mooring use, the UAM has demonstrated 200 bit/s over 5.4 nautical miles in shallow water off of Cape Cod, MA, using MFSK modulation while it is capable of supporting any linear modulation scheme. The availability of powerful microprocessors led to the demonstration of a 5000-bit/s MFSK system [49]. The overall bandwidth of 20 kHz required to achieve that data rate, however, limited its range to about 2 nautical miles. The Adjustable Diversity Acoustic Telemetry System (ADATS) achieves the more modest goal of 1250 bit/s out to 2 nautical miles using 10 kHz of bandwidth [50]. The wide range of channel conditions present in the ocean is acknowledged through a user-controllable tradeoff between frequency diversity and data rate.

The Telesonar system being developed in concert by Datasonics, Inc., and the U.S. Naval Command, Control and Ocean Surveillance Center features 128 simultaneously available tones [51]. A key element of the Telesonar system is the use of a Hadamard code to select tones for each word and a convolutional code to select a hopping pattern for the tonal alphabet [52]. The particular algorithm spawns from the work of Proakis on incoherent coded modulation [53]. The increase in processing power that moved the community from the eight tones available to DATS to the 128 tones available to Telesonar has allowed the same underlying receiver algorithm to employ increasingly complex alphabets and coding approaches to improve performance.

Table I details the salient characteristics of each of the incoherent systems referenced above. If available, the metrics given for each system are bit rate, coding scheme, range, and bandwidth. In the case of coding schemes, redundant transmissions in time or frequency are the only fundamental technique. If that redundancy is controlled by a classical channel coding method, it is so indicated. Clearly, these systems were not intended to optimize any one of these performance measures and, as such, readers are cautioned against making conclusions regarding the contribution of each work based solely on Table I.

With no fundamentally improved incoherent receiver strategies on the horizon, the main challenge facing the incoherent community is to adaptively optimize the classical modulation parameters in response to the *in situ* environment in an effort to maximize range, rate, and reliability. None of the systems reviewed currently offer an *in situ* adaptation capability for determining channel reuse time and setting these parameters. As a result, systems are designed to operate in the harshest expected environment and, thus, suffer from unnecessary, often substantial, bandwidth and power inefficiencies.

B. Coherent Digital Receivers

Phase-coherent underwater acoustic communication systems, in contrast to incoherent systems, have evolved considerably in the last two decades. Intersymbol interference (ISI) mitigation and the underlying, often dynamic, mechanism of multipath propagation have posed the primary obstacles, as discussed in the earlier section on coherence. In fact, the initial significant milestones in coherent receiver development, namely the use of decision feedback equalization (DFE) and phase-locked loops (PLL's), were driven by the complexity and time variability of ocean channel impulse responses. Whereas incoherent receivers studiously avoid the ISI effects of reverberation, coherent receivers must actively mitigate it to preserve a reliable phase reference. In this section, we trace the development of coherent receivers up to the now ubiquitous jointly optimized PLL and DFE structures. We then summarize current efforts to reduce the complexity of such algorithms and improve their performance. Alternative receiver strategies to the DFE-PLL structure will then be reviewed. Finally, we conclude with prospects for future development in this area.

As a prelude to the task of estimating and tracking the absolute phase of the transmitted signal, differential phase shift keying (DPSK) serves as an intermediate solution, in terms of

TABLE II
A SUMMARY OF SEVERAL SALIENT METRICS FOR DPSK TELEMETRY SYSTEMS IS TABULATED HERE FOR EACH OF THE REFERENCED SYSTEMS

Principal Investigator	Data Rate (bps)	Bandwidth / Carrier (kHz)	Bandwidth Efficiency	Range (km)	Prob. of Error
Mackelburg(1981)	4800	8 / 14	0.6	4.8_D	10^{-6}
Olsen (1985)	2000	2 / 10	1.0	6.0_D	$<10^{-3}$
Mackelburg (1991)	4800	6 /11	0.80	10.0_{SIM}	N/A
Howe (1992)	1600	10 / 50	0.16	0.1_S	$<10^{-3}$
Fischer (1992)	625	10 / N/A	0.06	N/A	N/A
Suzuki (1992)	16000	8 / 20	2.0	6.5_D	10^{-4}
Jones (1997)	20000	10 / 50	2.0	1.0_D	10^{-2}

N/A indicates the data was not available in the published reference. SIM indicates simulated or design results rather than experimental. Ranges with an "S" subscript indicate a shallow-water result while a "D" subscript indicates a deep-water result, typically a vertical channel. Error probabilities are simply typical of what the authors report.

bandwidth efficiency, between incoherent and fully coherent systems. DPSK encodes information in the signal phase relative to the previous symbol rather than to an arbitrary fixed reference and may be referred to as a partially coherent modulation. As with PSK, one can use an alphabet of N distinct levels (N-DPSK). While this strategy substantially alleviates the carrier phase-tracking requirements, the penalty is an increase in error probability over PSK at an equivalent data rate. The decision metric is based on the difference in two measurements and, thus, increases the incoherent noise component over a single measurement. For binary DPSK and binary PSK, the difference amounts to an effective decrease in SNR of 3 dB when signaling over an additive white Gaussian noise (AWGN) channel. Nevertheless, coherent communication in the 1980's relied on DPSK as a compromise between the benefits of coherent modulation and the consequences of a time-variant ocean communication channel. The receiver structures were typically simple. The real passband signal from the hydrophones was coherently demodulated to yield both in-phase (I) and quadrature (Q) signal estimates. A slicer, or decision device, compares the phase of each symbol to that of the previous symbol. In some instances, equalizers were included but generally served to reduce already modest error rates rather than make the difference between a functional and nonfunctional modem.

One of the earliest descriptions [54] briefly relates a test of 4-DPSK telemetry from the surface to a bottom mooring in 4500 m of water. Error rates less than 10^{-6} were obtained for transmissions up to 45° from vertical at a data rate of 4800 bit/s. As was typical of these early systems, no provision was made for equalization although baffling was often essential to mitigate nearby reflections in geometrys where the nodes were near the surface or bottom. A later system [55] was similarly designed for operation over a deep, vertical 6-km channel. 2-DPSK modulation with a data rate of 2000 bit/s was tested with error rates of 10^{-5} obtained. Error rates were substantially higher as the source was displaced 3 to 6 km horizontally but that was, perhaps, a consequence of the high directivity of the source which remained in a vertical orientation. The raw digital data was encoded with the classical techniques of BCH and Reed–Solomon codes. Yet another system [50], dubbed AUSS, developed by the Naval Ocean Systems Center employed 2-DPSK and 4-DPSK to obtain data rates of 1200 and 4800 bit/s over 6 kHz of bandwidth. Operation was restricted to deep, vertical channels. While the phase-tracking burden is lightened by DPSK, these systems are as prone as any coherent system is to ISI, which presumably explains their confinement to deep, vertical channels. An application of 2-DPSK to horizontal or vertical channels with the special property of having a clear time interval between the direct path arrival and subsequent path arrivals [56] was tested over a short (100-m range), shallow (13-m depth) channel. An average data rate of 1.6 kbit/s was obtained using 3-ms 10-kbit/s pulses designed such that the burst ends before the first significant multipath arrival. The system failed at 200-m range because the delay between the first and second arrival became too short. Variants of this approach are currently being examined by researchers at Lockheed Martin/Sanders and WHOI.

In an early application of spread spectrum techniques to underwater communication, Microlor, Inc., developed a 625-bit/s DPSK encoded system that overlays a 16-chip/bit spreading code prior to transmission [57]. Known as direct sequence spread spectrum modulation, each element (chip) of a bit sequence (code) is multiplied by the symbol to be transmitted. The result is transmitted such that the symbol rate remains the same, thus a K-chip/bit code results in a spreading of the bandwidth by a factor of K. In principle, this spreading code resolves reverberation on a scale of 0.1 ms. Unfortunately, no experimental results are given.

A trait shared by all the coherent receiver structures described thus far is that they strive to avoid or suppress ISI. The following systems employ equalizers that seek to undo the effects of ISI. An application with high bandwidth requirements, namely transferring an image from a deep submersible vehicle to the surface, was studied using 4-DPSK modulation over a deep vertical channel of 6.5 km [58]. Images containing 61 440 pixels (requiring 32 000 symbols) were transmitted every 10 s (3200 bit/s) with an average BER of 10^{-4}. The system consumed 8 kHz of bandwidth centered on 20 kHz. A linear, adaptive fractionally spaced equalizer followed by a second-order PLL was used. Few details are given and, as such, the effectiveness or even the need for the equalizer is impossible to assess. Another DPSK system is under development by the University of Birmingham [59]. The system emphasizes autonomous operation over depths under 1 km and allows for three modulation levels (2-, 4-, and 8-DPSK). The system uses 10 kHz of bandwidth centered at 50 kHz. As is typical for many reports, experimental performance is described only generally with claims that 2- and 4-DPSK "worked" and 8-DPSK "did not work." Table II summarizes the reviewed DPSK systems.

None of the investigators reported using any coding in conjunction with the DPSK systems except for Mackelburg, who used temporal redundancy, and Suzuki, who incorporated a discrete cosine transform to compress the images. While these early phase-coherent systems achieve higher bandwidth efficiencies [data rate (R)/required bandwidth (W)] than their incoherent counterparts, the expected increase in range–rate products has not been achieved, presumably due to the restrictions on channel geometry. For that to happen, transmission over substantial ranges in a horizontal channel needs to be demonstrated and, in turn, true ISI compensation, or equalization, is required.

PSK encodes information in the signal phase relative to a fixed phase reference. Over comparable AWGN channels with a fixed transmission power and equivalent data rates, the BER of a PSK system is lower than that of a DPSK system. This performance advantage can be used to achieve equivalent performance over longer ranges or with lower power. The cost for many channels, however, is the need to continuously track the phase and amplitude variability of the signal due to fluctuations in the channel impulse response. A multitude of equalizer and receiver implementations has been presented in the last decade. A comprehensive review of these potential approaches was prepared by Proakis [60] and offers a concise tutorial to the interested reader. In all cases, demodulated, digitally sampled data is linearly combined with filtered versions of past data and, possibly, past decisions to yield symbol estimates. Variations of this paradigm can be parameterized in terms of three features.

1) The filtering may be purely linear with only a tapped delay line of the data streams or nonlinear with the addition of a feedback loop of filtered previous decisions.
2) The method for adaptively estimating and updating the filter weights may range from the low complexity and slow convergence of the least mean square (LMS) algorithm to the high complexity and rapid convergence of the recursive least squares (RLS) algorithm.
3) The burden of carrier phase tracking is either carried by the equalization filter or an external PLL.

Other features, such as blind versus decision-directed equalization and transversal versus lattice structures, differentiate the published algorithms much less frequently with transversal, decision-directed approaches predominantly implemented.

As an aid to subsequent discussions, a common and effective algorithm that combines decision feedback equalization with a second-order PLL will be described. The essential elements of that receiver are a set of conventional feedforward taps that sample the received pressure signal, a set of feedback taps that provide previous symbol decisions as well as introduce a nonlinearity into the filtering, and a PLL that attempts to relieve the tap weight adaptive algorithm of the phase-tracking task. The feedforward sampling is usually done at an integer multiple of the symbol rate (e.g., twice) allowing the adaptive algorithm to perform fine scale synchronization. Under many circumstances, the most prominent time-varying feature of the signal is a mean variable Doppler shift. The PLL is capable of estimating and compensating for this phase offset in a rapid, stable manner leaving the equalizer to track the complex,

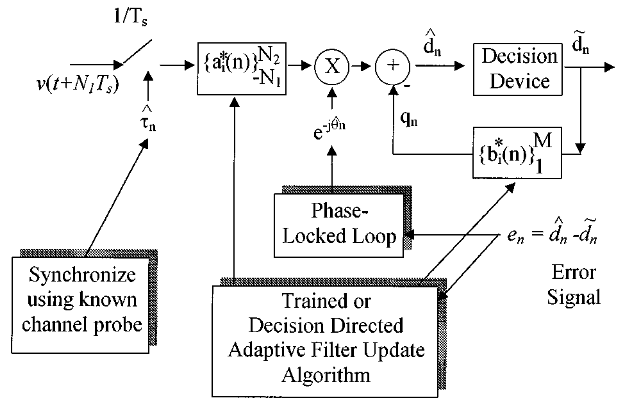


Fig. 4. This schematic illustration of a canonical coherent receiver algorithm is taken from [4]. The feedforward tap weights $\boldsymbol{a}$ filter the incoming data ν while the feedback tap weights $\boldsymbol{b}$ compensate for residual intersymbol interference. An estimate of the carrier phase θ is used to provide phase compensation while a coarse tuning estimate τ synchronizes the process. The total number of taps updated is $N_1 + N_2 + M$.

relatively slowly varying channel response. The stability of the coupling between these two adaptive processes is still an open research question [18]. The algorithm structure [4] is shown in Fig. 4. If a hydrophone array is available, the structure may be modified slightly to provide an individual set of feedforward taps for each input data stream. Alternatively, a preprocessing step may be included that combines some or all of the hydrophone elements. Algorithm features (e.g., spatial beamforming and diversity combining) that deal explicitly with multiple spatial channels will be considered in the later section on diversity. These and other modifications to the canonical structure presented will be discussed in conjunction with a review of the appropriate references below.

The common modulation approaches encode information in the phase and amplitude of the signal. QPSK denotes signaling with four phase symbols equally spaced from 0 to 2π. 8PSK employs eight equal-amplitude equally spaced phase symbols from 0 to 2π. 8QAM and 16QAM have 8 and 16 possible symbols that vary in both amplitude and phase from each other and are typically uniformly distributed over an amplitude range defined by a signal power limitation.

The discussion begins with receivers that are similar to that shown in Fig. 4 and then turns to a description of alternate approaches. One of the earliest examples of truly phase-coherent underwater acoustic communication was developed to telemeter images and commands between a surface ship and a subsea robot [61]. Quadrature amplitude modulation (QAM) with a 16-symbol constellation was used to transmit 500 kbit/s over a range of 60 m. The link used 125 kHz of bandwidth centered at 1 MHz and typically contained two principal arrivals, a direct path and a surface bounce separated by about 11 symbols. A symbol spaced single-channel decision feedback equalizer using a LMS update algorithm was able to reduce the output mean square error from –10 to –16 dB on average, thereby reducing BER's from 10^{-4} to 10^{-7}. While the receiver contained many of the features of current systems, it was employed on a relatively benign and highly structured channel. A similar system was used for telemetry over a 3.5-km-deep vertical channel [62].

The structure differed from that used by Kaya in that no feedback loop was used and sampling occurred at twice the symbol rate, yielding a fractionally spaced equalizer. The modulation was 4- and 8-level PSK with a 10-kHz bandwidth centered at 25 kHz. Performance results are purely anecdotal with no detailed descriptions. In both of these examples, the equalizer was not required for communication at modest error rates ($<10^{-3}$) largely because of the simplistic nature of the channels and solely served to increase reliability.

The first demonstration of phase-coherent underwater acoustic communication over representative horizontal channels was published in a series of two papers [3], [4]. Both reports describe results based on the canonical coherent receiver structure described in Fig. 4. The first paper is noteworthy in that it develops the proposed receiver structure from classical signal processing theory beginning with the multichannel combining and detection approach that is optimal under a maximum likelihood criterion. Given the substantial length of intersymbol interference[4] present in many practical underwater channels, the author points out the unacceptable complexity of maximum likelihood sequence estimation and proposes equalization in its place. An approach is presented whereby carrier tracking and equalization are separated, allowing separate tracking strategies, and jointly optimized against a least square error criterion. Experimental results are given for transmission of QPSK, 8PSK, and 8QAM with 0.3 to 1.0 kHz of bandwidth centered at 1.0 kHz. Transmissions were made in deep-water convergence zone conditions as well as shallow-water (<50 m) conditions over ranges of 89 nm in shallow water and 203 nm in deep water. Error rates were less than 10^{-4}. A suggestive observation was made whereby a packet was not decodable with a single hydrophone channel but was decodable with multiple channels, pointing to the value of spatial diversity.

A linear equalizer with jointly optimized carrier recovery was proposed for a European image telemetry application (~150 kbit/s) [63]. No experimental or simulated results are given. The algorithm does include an adaptive update of the PLL gain but, without any results, its value is difficult to ascertain. A prototype digital acoustic underwater phone was recently described [64]. Sampled voice data is compressed and modulated onto a QPSK alphabet. The data is actually differentially encoded but the equalizer is, in fact, identical to that proposed by Stojanovic. The carrier frequency is 60 kHz, the bandwidth is 3 kHz, and decoding is accomplished with a decision feedback fractionally spaced equalizer. The only results given are anecdotal performance over a 40-m range in a 10-m-deep laboratory tank where output MSE was –12.3 dB in one case. In a related paper [65], the same system is used to compare the LMS and RLS update strategies. The authors noted the known convergence advantages of the RLS algorithm but found no steady-state advantage, presumably due to the lack of channel dynamics. Experimental results were given, however, for a 4-km vertical channel where an output MSE of –19.5 dB was achieved for one 500-point data set.

[4]Channel delay spreads can range up to hundreds of milleseconds. With typical symbol rates of 1–5 kHz, this leads to ISI exceeding 100 symbols in some difficult shallow-water channels.

Yet another version of the canonical receiver was tested in Loch Duich, Scotland, over a 0.9-km range and 100-m depth [66]. Hardware constraints required the use of binary DPSK although the equalizer operated as with BPSK signaling. No explicit carrier phase tracking was used as all platforms were moored and a front-end beamformer was incorporated although not described in this paper. The use of a self-optimized LMS update algorithm, with adaptively modified step gains, was described.

An alternative approach to filter weight update in conjunction with the canonical DFE-PLL structure was undertaken by the U.S. Naval Undersea Warfare Center [67]. The authors' implementation of a stabilized fast transversal filter is claimed to have the stability and convergence properties of the RLS update but with $O(N)$ complexity instead of $O(N^2)$ where N is the number of filter taps. A synopsis of numerous experimental results is given noting acceptable error rates ($<10^{-3}$) over ranges from 500 to 8000 m, data rates from 1.1 to 2.2 kbit/s, and both shallow- and deep-water conditions.

Another investigation of update algorithms compares an $O(N)$ RLS algorithm, classical LMS, and a version of LMS with adaptive step gain [68]. The equalizer is linear with no feedback section or explicit carrier phase recovery and provides for multiple input channels with separate LMS step gain for each. Results are given for large depths in the Mediterranean Sea over a 50-km range and BPSK modulation with a 212.5-Hz bandwidth centered at 1.7 kHz. Received signals had a 100-ms delay spread and an average SNR of 18.5 dB. Based on 30 000 symbols, the authors conclude that the fast RLS approach is numerically unstable while the LMS and optimized step gain LMS give comparable results. In [69], a the authors describe the performance of the same receiver following a brief derivation of optimal MSE for both linear and DF equalizers. With a seven-element hydrophone array spaced between depths of 100 and 300 m, a 200-bit/s BPSK waveform was successfully decoded over a 50-km range. Under similar conditions, a nine-element array spaced between depths of 148 and 151 m was unable to provide reliable communications, again pointing toward the value of spatial diversity. The authors cite some evidence suggesting that the equalizer is capable of compensating for the time dilation and carrier phase drift induced by Doppler effects in this particular experiment.

A final implementation of the canonical coherent receiver was designed by Jarvis to transmit 900 to 1800 bit/s over a 4-km range in shallow water and 8-km range in deep water [70]. The carrier frequency and modulation method are not given. Five field experiments are described with mixed results. In one case, excessive Doppler variation over the packet precluded decoding largely because no PLL was used and the adaptive equalizer was unable to track the time variability. In another case, a late arrival in the impulse response generated excessive filter lengths, again precluding successful decoding. The design goals were met, however, in the final test.

These authors emphasize a design criteria that the current community of researchers also deem valuable, namely an algorithm that can autonomously initialize itself without human intervention. Many aspects of the equalization process are quite sensitive to such parameters as step factors, forgetting

TABLE III
A SUMMARY OF SEVERAL SALIENT METRICS FOR COMPLETELYCOHERENT TELEMETRY SYSTEMS IS TABULATED HERE FOR EACH OF THE REFERENCED SYSTEMS IN THIS SECTION

Principal Investigator	Modulation Method	Data Rate (kbps)	Bandwidth / Carrier (kHz)	Range (km)	Prob. of Error
Kaya (1989)	16QAM	500	125 / 1000	0.06_D	$<10^{-7}$
Suzuki (1985)	4, 8PSK	20-30	10 / 25	3.5_D	$<10^{-4}$
Stojanovic (1993,1994)	4, 8PSK 8QAM	0.6 – 3.0	0.3 – 1.0 /10	$89 – 203_{S,D}$	$<10^{-2}$
Goalic (1994)	QPSK	6	3 / 60	0.04_S	N/A
Labat (1994)	QPSK	6	3 / 60	4.0_D	N/A
Tarbit (1994)	BPSK	20	20 / 50	0.9_S	$\sim 10^{-3}$
Jarvis (1995)	B, QPSK	1.1 – 2.2	0.6 – 2.2 / N/A	$0.5 – 8.0_{S,D}$	$<10^{-3}$
Capellano (1996)	BPSK	0.2	0.2 / 7	50_D	$<10^{-4}$
Capellano (1997)	BPSK	0.2	0.2 / 7	50_D	$<10^{-4}$
Jarvis (1997)	N/A	0.9 – 1.8	N/A / N/A	4.0_S , 8.0_D	$<10^{-4}$
Freitag (1998)	QPSK	1.67 , 6.7	2 ,10 / 3,25	4.0_S , 2.0_s	N/A

N/A indicates the data was not available in the published reference. SIM indicates simulated or design results rather than experimental. Ranges with an "S" subscript indicate a shallow-water result while a "D" subscript indicates a deep-water or line-of-sight result. Recall that the bandwidth efficiency (η) of BPSK is 1.0, QPSK, 8PSK/8QAM is 3.0, and 16QAM is 4.0. Error probabilities are simply typical of what the authors report.

factors, and filter support, with proper selection crucial to reliable performance. The approach of these authors was to repeatedly reprocess an initial block of data with a range of equalizer settings, thereby determining an optimal selection for the remainder of the data. The optimality criterion was to achieve the smallest output MSE over the specified range of equalizer parameters. Successful autoinitialization was experimentally demonstrated in that the parameter set selected at the outset was held constant throughout the test with all packets successfully decoded.

A practical implementation of a phase-coherent acoustic communication system onboard an autonomous underwater vehicle (AUV) has been described demonstrating the maturity level of this technology [71]. QPSK modulation at center frequencies of 3 and 25 kHz with coded data rates of 2500 and 10 000 bit/s, respectively, was employed. Reliable two-way communication was demonstrated to 4 km (3 kHz) and 2 km (25 kHz) range in water depths of 10–30 m. Table III offers a synoptic view of the preceding references with the usual metrics of modulation, bandwidth, carrier frequency, and range. As these references demonstrate, the common element for successful coherent underwater acoustic communication systems is an adaptive equalizer structure while the inclusion of a feedback section and a jointly optimized PLL often dramatically improve performance.

A number of straightforward, yet quite important, modifications to the canonical structure have been proposed and tested in recent years. Each addition is geared toward reducing the overall computational complexity or increasing equalization and tracking performance of the receiver algorithm. One should note that these two goals are not mutually exclusive as excessive adaptive filter lengths are associated with increased noise levels and poor convergence rates. The potential combination of lengthy reverberation and time-variant channel characteristics in the underwater acoustic channel create a clear design tradeoff for the filter adaptation algorithm. In many cases, this part of the algorithm requires the bulk of available computational resources, thereby heightening its significance.

The selection of LMS versus RLS update algorithms based on channel complexity has been investigated [72], [73]. Unfortunately, the authors only give comparative anecdotal examples without giving any quantitative algorithm selection measures. As acoustic communication between moving surface and underwater vehicles has been and remains an active application area, another area of research has dealt with Doppler compensation. The stability of the interaction between a linear transversal equalizer and a PLL has been analytically treated [18]. Under certain circumstances, that interaction is shown to be intrinsically unstable. One approach to overcoming mean Doppler effects involves a preprocessing step for each block of data [74]. The proposed approach involves three steps. By correlating against a known training sequence, the mean Doppler shift is estimated for each channel. Based on this, the data is phase compensated for the gross Doppler shift and resampled. Although the equalizer is capable of accomplishing these tasks in principle, the preprocessing steps serve to reduce the tracking burden on the adaptive filters. Finally, the canonical DFE-PLL algorithm decodes the data, mitigating any residual or differential Doppler effects.

As a first step toward a model based approach, a generalized receiver that explicitly estimates and tracks a discrete version of the delay-Doppler-spread function has been proposed for use in time-variant severely Doppler spread channels [18]. For the canonical DFE, several structural modifications have been suggested for reducing algorithm complexity [75]. First, a sparse updating technique only adapts the filter weights when the MSE exceeds a certain threshold, thereby reducing computations for more stable channels. Second, when convergence and stability requirements permit, more efficient update methods such as LMS or fast RLS may be used instead of RLS. Third, sparse feedback tap placement may be employed when the impulse response has a lengthy yet sparse structure which, in fact, can frequently occur in shallow-water as well as in deep-water convergence zones. Using experimental data, the authors demonstrate that each of these approaches can separately achieve a 75%–90% reduction in computational load. Their specific test employed QPSK modulation and 5-kHz bandwidth with a 15-kHz carrier in a 3-km-deep ocean channel over a 3-km range. Other authors have also considered sparse equalization. Kocic [76] considers the optimal sparse parameterization for a given channel. More recent work considers approaches whereby the tap placement and sparsing is adaptively modified during the decoding process [77]. In another example that highlights a modification to the canonical structure, a block DFE architecture is described that can accommodate fractional spacing and carrier recovery [78]. Simulated results comparing the block-DFE with the canonical DFE showed a 60% reduction in computational complexity with improved error rates equivalent to an SNR increase of 1–2 dB. Interestingly, one feature of coherent systems pervades the community but its use is rarely explained. All reported systems employ a packetized structure organized into a channel probe for detection, a fixed sequence for equalizer training, and a data segment with each packet independently treated by the receiver. Any issues associated with a continuous operation mode, therefore, remain largely unexplored with the exception of known numerical instabilities associated with the RLS algorithm. As an example, the variability of underwater channels is such that the equalization filter support may have to be periodically updated.

While the canonical DFE-PLL with the modifications described above has emerged as an effective receiver algorithm for a wide-range of ocean channels, results have been reported for alternative approaches, specifically blind algorithms that can operate without training sequences and maximum likelihood sequence estimation approaches for channels with short delay spreads. Gomes proposed a blind equalization technique combined with a linear transversal filter with a separately updated predictive decision feedback filter operating on the error signal between the slicer and transversal filter outputs [79]. The feedback filter serves to whiten the input to the slicer. Successful performance was shown via simulation as well as with experimental data using BPSK data of 120-Hz bandwidth on a 53-kHz carrier over 1-km range in shallow water. Some difficulties remain in compensating for Doppler effects over the block of data. Earlier work by the same authors sought to apply neural network techniques to the blind equalization problem with successful results from the same data set [80].

Another group of researchers has recently proposed a blind structure that places the recursive whitening filter before the transversal filter [81] which is, in turn, followed by a PLL. At a predetermined threshold value of some criterion, the recursive filter is switched to the position of the feedback loop in the canonical DFE, and a decision-directed mode is adopted. The authors claim that this transition is stable for sufficient SNR resulting in only small perturbations to the filter values. If channel conditions rapidly deteriorate, the algorithm reverts to a blind mode, thereby recovering where a canonical DFE would require retraining. Results are given for a 20 000-symbol packet transmitted over a vertical link using 4-QAM modulation with 3 kHz of bandwidth and a 12-kHz carrier frequency, demonstrating recovery from a rapid channel deterioration.

In another publication, blind equalization of multiple channel data was considered [82]. Bessios proposes a bank of linear transversal filters for each channel with a three-component cost function. Those components are the conventional constant modulus term, a term that compares the equalizer output covariance to the known signal covariance, and an error term that differences the equalizer outputs. A brief numerical result is reported but convergence properties are difficult to assess. Although preliminary results are promising, blind equalization approaches have yet to demonstrate applicability to the range of conditions the canonical DFE has successfully encountered with experimental data.

Maximum likelihood sequence estimation (MLSE) has received modest attention in the last few years. One approach proposed joint channel estimation and data recovery for the underwater channel [83]. Acknowledging the overwhelming complexity of the joint optimization problem, an iterative scheme is described whereby the impulse response is assumed and then the data is decoded via the Viterbi algorithm. Using the decoded data, channel identification is performed and the process repeats. Simulated results are given for a channel described by four delay taps. Application of the approach would

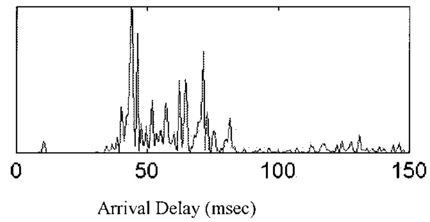


Fig. 5. Sample impulse responses from the New England continental shelf ACOMMS ATD experiment. The modulation was QPSK at 1250 symbols/s with a 2.5-kHz carrier frequency. It decoded using 957 parameters derived from eight hydrophone channels and feedback filter.

seem to be restricted to channels with reverberation times small compared to a symbol and also stable over many symbols. A more recent work also seeks to apply a Viterbi decoder with a potentially truncated number of ISI states but relies on an initial training sequence to estimate the channel for use in the metric calculation [84]. Experimental results are given for BPSK modulation with a 500-Hz bandwidth and a 1-kHz carrier. The site depth was 4 km and the range was 63 km. MLSE techniques would seem to be, in general, inappropriate for channel duration beyond some tens of symbols (less if larger symbol constellations are used) due to the exponential growth in complexity.

In considering the impressive body of research into coherent underwater acoustic communication that has taken place in the last several years, one is struck by the emphasis on ever more complex algorithms as a solution to ever more challenging channels. As high-rate communication in littoral and surf zone environments from relatively high-speed platforms is considered, this approach may fail as multichannel tap delay line representations lead to overparameterization of the ocean channel. An example may serve to clarify this point. As part of the Acoustic Communications Advanced Technology Demonstration funded by the U.S. Navy Advanced System Technology Office, a large database of acoustic transmissions over the New England continental shelf from moving platforms has been collected.[5] The impulse response shown in Fig. 5 was encountered in one test. Successful decoding of the 1250-symbol/s QPSK data stream was accomplished with eight hydrophone channels and a 957 tap filter updated with a self-optimized LMS routine. By examining the eigenvalue distribution of the time-averaged filter covariance matrix, however, one concludes that only 68 true degrees of freedom are present in the channel. Bridging this parameterization gap is perhaps the largest challenge facing the coherent communications community. Such approaches as parametric modeling of the channel, precombining spatial filters, or models for tap correlation properties may yield a reduction in receiver degrees of freedom and a concurrent increase in equalization performance, but the community must await future research to even know what is possible.

[5]The ACOMM ATD, with Sanders, A Lockheed Martin Company, as prime contractor, seeks to transition coherent underwater modem technology to the U.S. Navy. Communication between surface vessels, submarines, and AUV's has been demonstrated in many channels using current USN sources.

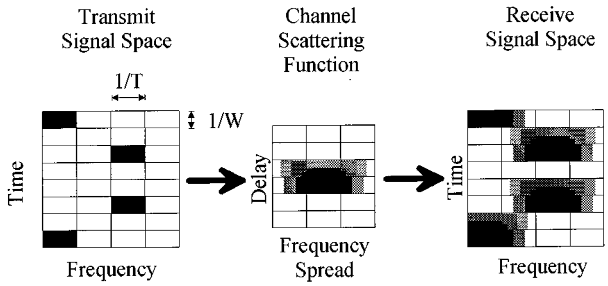


Fig. 6. A signal space representation clarifies the impact of the channel scattering function on the transmitted waveform. The required temporal and spectral separation for diversity signaling may be related to the scattering function extent. As such, the issue of signal design and diversity may be linked through a signal space formalism, such as is shown here.

V. Diversity Usage

Classical diversity in a communication system refers to the availability of multiple, uncorrelated measurements of the transmitted signal. These measurements may be taken over different frequency bands, temporal spans, or spatial apertures. Such diversity is a powerful tool in combatting the effects of fading channels characterized by a complex amplitude scaling that is a random variable leading to periods of low SNR [27]. From a purely statistical perspective, diversity is achieved when the measurement separations in frequency, time, space, or bearing are sufficient to ensure the random variables characterizing the measurements are uncorrelated, thereby reducing the probability of inadequate SNR on all measurements simultaneously. Before we discuss the literature regarding diversity usage in underwater acoustic telemetry systems, we remind the reader of a useful physical framework for considering diversity.

A communication engineer considers the underlying physical channel with the intent of determining an optimal (by some measure) manner of occupying the total frequency and temporal spans available for the signal, the signal space. Consider a rectangular grid depicting the transmit signal space. Each block represents a resolved cell. The time space of each cell is $T_{\rm cell}$, the frequency of each cell is $W_{\rm cell}$, the total signal duration is T, and the signal bandwidth is W. Fundamentally, $T-{\rm cell}$ may be no less than $1/W$ and no more than T while $W_{\rm cell}$ may be no less than $1/T$ and no more than W. The signal space may contain, at most, WT resolved cells.

A two-dimensional (2-D) convolution of the channel scattering funtion with this transmit signal space yields the received signal space with the number of resolved cells again dictated by the time–bandwidth product (WT) of the transmitted signal. The scattering function will "blur" information between cells in accordance with the temporal spreading extent L and the frequency spreading extent B. The process is described pictorially in Fig. 6. In general, one would add two more dimensions to the signal space representing the resolvable bearings, set by the overall aperture, and range of bearings, set by the hydrophone spaceing.

We now tie the notion of diversity into this physical linear system model. In underwater acoustics, fading is generally the result of destructive interferences of multiple propagation paths. For narrow-band signals such as FSK tones, the discrete, stable paths dictated by ray theory, and unresolved by tonal signals, may lead to this interference. Even for wide-band signals, which may be able to resolve these paths, channel fluctuations may be so severe that individual ray paths may consist of uncorrelated components and exhibit fading even when resolved. This latter condition is referred to as saturated propagation [21]. In contrast, interference between the signal cells is described as intersymbol interference for temporal overlap and cochannel interference for frequency overlap. If the temporal and frequency extent of the scattering function is L and B, respectively, then transmit signal cells spaced in time or frequency by more than $1/B$ and $1/L$, respectively, will typically yield a substantially different linear combination of the unresolved propagation paths, making simultaneous fades for several transmit cells unlikely. As such, one may achieve, at most, frequency diversity of order WL or time diversity of order BT. The previously defined condition of overspread versus underspread may be graphically viewed as whether L and B exceed $T_{\rm cell}$ or $W_{\rm cell}$, respectively. These measures of the number of diversity channels available (BT and WT) in telemetry systems are useful and prevalent but, in the absence of information regarding coherence between the paths, does not ensure diversity in the classical sense. In fact, invocation of a wide-sense stationary uncorrelated scatter (WSSUS) assumption for the scattering function is implicit in these diversity measures.

A design tradeoff clearly exists where the requirements of data throughput, decoding complexity, and error probability impose competing constraints on the signal design. Independent use of the resolvable tones in the available bandwidth maximizes data throughput but affords no protection against fading of any one tone. Redundant use of the tones introduces diversity, assuming tone spacing $>1/L$, and increases reliability but at the expense of data throughput. By decreasing symbol duration below L or, a greater challenge, increasing frequency resolution below B, one can begin resolving subsets of the propagation paths and generate diversity channels. A subtle, yet important, point is that when $W \gg B$ the prospect of adaptively estimating the fading coefficients becomes possible. This permits coherent combining of the diversity channels. We have included this somewhat tutorial discussion to emphasize the need to integrate channel descriptions (via the scattering function) with waveform design (via signal space models) in a unified framework. This is particularly useful for understanding the role of diversity.

A taxonomy that may be applied to diversity distinguishes the separating dimension of the diversity signals (frequency, time, spatial location, or bearing) and whether the diversity arises from resolving subsets of the channel scattering function (implicit diversity) or redundantly sending information in an uncorrelated part of the signal space (explicit diversity). A common example of implicit diversity is the ubiquitous "Rake" receiver [24]. In the original implementation, each bit to be transmitted was represented by one of two waveforms with bandwidths greater than $1/L$. This allowed a series of delayed matched filters to resolve subsets of the multipath which, in turn, could be coherently combined, thereby reducing the fading phenomena. Implicit diversity that resolves frequency spread is less common. Although implicit diversity does not *require* sacrificing bandwidth efficiency, many practical implementations (such as the Rake receiver) do, in fact, significantly spread the underlying data signal in frequency to facilitate simple receiver implementations such as correlators and avoid intersymbol interference. Rather than obtaining diversity channels by resolving channel-induced spreading, explicit diversity redundantly uses the available signal space to obtain multiple linear combinations of all the propagation paths. Explicit diversity *requires* a reduction in bandwidth efficiency, a serious issue in the bandwidth-limited underwater acoustic channel. MFSK systems redundantly use transmit signal space cells separated in frequency. Classical coding of the time series redundantly uses cells separated in time. In many cases, however, the coding used does not spread the information over time spans greater than the coherence time of the channel and, therefore, do not provide diversity, merely coding gain against noise. It is, of course, possible to combine these approaches [51]. Furthermore, the redundancy need not be simple repetitive coding [31]. Diversity combining using hydrophone arrays is an explicit spatial diversity with the attractive feature of not consuming additional portions of the time–frequency signal space. As noted earlier, the combined use of channel scattering functions and a signal space provides a nearly complete framework for understanding diversity in underwater telemetry systems.

Explicit frequency diversity has been a component of digital underwater acoustic telemetry since the inception of the field. One of the earliest descriptions in the open literature was the DATS [2]. In that case, a rate 1/2 code spread 4 bit of information over 8 of 16 frequency tones. For an experiment in 9 m of water over an 800-m range, frequency separations of 2 kHz near 50 kHz yielded independent paths. Most MFSK systems in the literature, however, do not measure or estimate the coherence bandwidth. Instead, the MFSK architecture simply accommodates the introduction of redundancy over frequency providing coding gain that may, or may not, stem from diversity. The distinction becomes significant when modeling error probability. Many of the MFSK systems described earlier in the review claim to invoke frequency diversity. In the absence of detailed coherence measurements, however, one may only speculate on the direct value of frequency diversity to any given telemetry system. As such, our discussion of explicit frequency diversity here is quite brief in spite of the pervasive presence of the technique in underwater acoustic telemetry. One simply cannot infer the role of diversity without careful channel probes. The current trend toward *in situ* channel probing in conjunction with incoherent modulation systems will serve to clarify the role of frequency diversity in future telemetry systems by allowing explicit channel coherence measurements.

Explicit temporal diversity has been employed to a limited extent. The FSK system of Jarvis [45] previously described includes a repetition of tones 1.6 s apart. A later phase-coherent system also described by Jarvis [70] transmits a redundant copy of the signal packet. Experimental results described temporal separations of 1 s. Although this degree of separation likely afforded the system diversity gain, the absence of coherence time, or equivalently frequency spreading, measurements makes the assessment difficult. Except possibly for the case of telemetry from platforms with modest to high velocity, coherence times in many underwater channels are large enough (or, equivalently, Doppler spreads are small enough) to require unacceptable latency for any system employing explicit temporal diversity, thereby limiting its use.

While explicit frequency and temporal diversity have become relatively mature additions to telemetry modems over the last two decades, the use of explicit spatial diversity has only recently become notable. A single omnidirectional transducer inherently excites the entire available spatial spectrum. This simultaneously achieves both explicit and implicit diversity. Exciting multiple rays yields explicit bearing, or angle-of-arrival, diversity while the spreading of the signal from a single transducer to multiple spatial locations gives implicit diversity (depending, of course, on the spatial coherence between the sensors). A receiving hydrophone array allows sampling of this spatial spectrum. With adaptive beamforming, a receiver may exploit the explicit bearing diversity (if $W \gg B$) while many multichannel combining techniques (both coherent and incoherent) leverage the implicit spatial location diversity. Catipovic proposed a use of implicit spatial diversity in an MFSK system where incoherent square law combining is applied to the spatial inputs [85]. The weights of each channel are derived from an estimate of its error probability in a similar manner to maximal ratio combining. Some experimental data suggested a vertical spatial coherence length of 35 cm for a signal between 15 and 35 kHz over an 800-m harbor propagation path. An anecdotal result for 20-m separation is given where the single receiver probability of error (Pe) was 0.03 while the Pe for the diversity combiner was 0.0006.

Explicit bearing diversity is more common among coherent systems, typically taking the form of an adaptive beamformer (narrow-band or broad-band). All of the work reviewed here presents algorithms and data suitable for spatial diversity processing but, in fact, fails to address coherence issues and, therefore, preclude any conclusions about the contribution of diversity itself. Henderson describes a pair of six-element transmit and receive arrays deployed to a 10-m depth over a muddy bottom and separated by 100 m [86]. Two adaptive algorithms are used to eliminate all but the direct path arrival. In this case, no diversity usage is described as the nondirect arrivals are nulled out and not made available to the decision process but diversity processing would have been possible. A similar system was tested in a 100-m-deep Scottish loch over a 1-km range [87]. A four-element array with a four-wavelength aperture was, once again, used to isolate a direct arrival and null other propagation paths.

In an early treatment of spatial diversity processing in an underwater acoustic channel, Wen derives and analyzes an algorithm he calls the spatial diversity equalizer [17]. While the algorithm is no more than a multichannel linear equalizer, the paper makes two novel contributions. First, expressions for probability of error in the presence of additive noise and residual ISI are derived that match well with simulated results. Second, a time-invariant ocean channel is modeled using a parabolic equation method. Using this model, error probability is evaluated as a function of receiver placement, number of

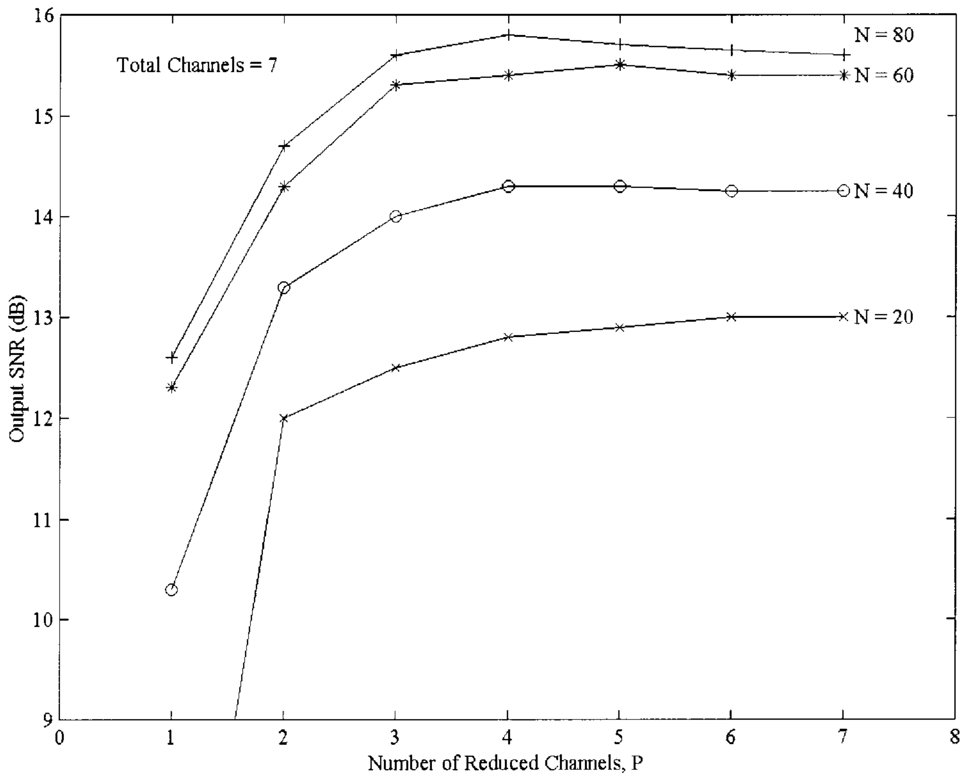


Fig. 7. Results of a study by Stojanovic [88] showing the effect on output SNR of increasing the number of reduced channels P in the proposed reduced complexity multichannel DFE. The data come from a 2-kbit/s transmission of QPSK modulation on the New England continental shelf over a 48-km range. The saturation behavior of the curves clearly points to the possibility of reducing complexity significantly without impacting performance while the rapid rise from $P = 1$ to $P = 3$ illustrates the value of multiple channels. The parameter N is the support length of the equalizing filters.

hydrophone elements, and tap length. One conclusion is that the use of hydrophone arrays substantially lessens system sensitivity to the placement of any one hydrophone. Wen notes, however, that the complexity of the processing may be high depending on the weight determination algorithm. One should note, however, that Wen examines a single realization of the environment and, as many authors do, fails to clarify the amount of performance improvement due strictly to diversity, including the implications for error probability.

Stojanovic addresses the question of complexity for multichannel equalization by comparing the performance of two algorithms [88]. The first is the full complexity jointly optimized multichannel linear equalizer. The second invokes a K to P channel narrow-band adaptive beamformer ($P < K$ and no temporal taps) followed by a single-channel equalizer for each of the remaining P channels. Experimental results are given for QPSK and 8PSK transmissions over 15–65 nautical miles on the New England continental shelf received on a 20-element vertical line array. A conventional multichannel DFE was used on three channels of the array. In that case, performance was sensitive to the precise choice of channels. By preceding the multichannel equalizer with a 10 to 3 channel beamformer, a 2-dB increase in output SNR was obtained. More importantly, the sensitivity to channel selection is eliminated with only a modest increase in complexity. A suggestive study is summarized in Fig. 7 where the output SNR for a 2 kbit/s transmission of QPSK over 48 km is shown as a function of the number of reduced channels P, beginning with $K = 7$. The curves are parameterized by the support length of the equalizing filter. As Stojanovic points out, the saturating behavior of the curves clearly points to a tradeoff between complexity and performance that is largely independent of the equalizing filter. The dramatic performance increase as more than one channel is used is also evident.

In a novel application of spatial diversity, Song extended the work of Wen to show how spatial diversity and the same receiver structure can be used to achieve channel reuse factors of 3 and 4 in a multiuser scenario [89]. In another work, experimental results for a multichannel decision feedback equalizer are given for a seven-element receive array with one wavelength spacing [90]. Complexity is addressed by using a LMS update algorithm. QPSK modulation at 20 kbit/s over 2-km ranges in 100-m water was sent to demonstrate the value of the multiple spatial channels. In one cited case, a single-channel DFE had an average output SNR of 13.2 dB, a multichannel linear equalizer achieved 14.9 dB, and the multichannel DFE reached 17.2 dB over the same data set. Once again, in the absence of spatial coherence information, it is difficult to assess the role of true diversity.

Thompson describes an experiment in the Mediterranean that performed a direct comparison between a nine-element dense array with 1/3 wavelength spacing and a seven-element sparse array with a 200-m aperture [91]. Low-rate BPSK transmissions of 212.5 bit/s with a 1.7-kHz carrier over 50 km were made. The dense array performed poorly since the SNR was low over the entire aperture. The sparse array, located separately in the water column, afforded acceptable error rates with only a single element. Use of all seven elements further boosted the output

SNR by 5 dB. The authors point to the diversity available to the sparse array but, without knowing if shadowing or fading is responsible for the signal dropout on the short array, the claim remains unsupported. Highlighting the current trend toward larger receive arrays for coherent telemetry systems, Albonico discusses results of 2- and 4-PSK signaling with a 62-kHz carrier over 1.5 km in 25-m- to 170-m-deep channels [92]. The 48-element vertical line array had a regular spacing of 2/3 wavelength. Spatial coherence functions are actually given for separations ranging from zero to nine wavelengths, revealing a coherence length of approximately three wavelengths. In addition to a study of optimal array aperture, comparisons were made between a multichannel DFE and a spatial beamformer followed by a single-channel equalizer. Both appear to perform equivalently. The case has clearly been made by the underwater telemetry community that receive arrays significantly improve coherent system performance.

Diversity continues to play a powerful and sometimes poorly understood role in successful underwater telemetry. While the use of frequency diversity in incoherent systems is quite mature, few systems have the *in situ* channel diagnostic information necessary to optimally match the data to the signal space or even quantify the available diversity. A related outstanding issue is a detailed understanding of coherence at telemetry frequencies. For instance, total channel delay spread may be a poor indicator of coherence bandwidth, yielding significant underestimates if ray arrivals are correlated. Statistical characterizations such as the channel scattering function should be used instead. Spatial diversity will continue to play a central enabling role for coherent communication over complex ocean channels. While future telemetry systems will undoubtedly utilize larger and larger arrays, researchers are already faced with a substantial complexity management task. Efficient and effective algorithms need to be developed to optimally exploit multichannel data short of introducing full jointly optimized multichannel DFE receivers. As with frequency diversity, careful and appropriate measures of spatial coherence are sorely needed. Currently, questions regarding stability and function of multichannel combiners are largely answered with pure speculation given the absence of statistical characterizations of the spatial acoustic field.

VI. Coding Applications

Coding of communication signals classically falls into one of two categories: source coding in which redundancy is removed from the information to be transmitted and channel coding in which structured redundancy is added to the signal to provide protection against errors. Both have found widespread application in underwater acoustic telemetry. As with other elements of the telemetry system, however, most efforts have been limited to a straightforward use of solutions derived from other applications. Many of the systems described already implement channel coding in the form of block or convolutional coding of the source bit stream. Having no aspects peculiar to the ocean, those efforts will not be explicitly discussed in this section. Instead, we will review recent efforts in the area of image compression that seek to balance the competing needs for high image quality and low to modest data rates, as well as four channel coding algorithms specifically proposed for use in the underwater channel. We conclude with a brief speculation on future challenges in this area.

A burgeoning application area for underwater telemetry is in control and data retrieval for AUV's. Missions such as mine countermeasures are aided by real-time transmission of images from the vehicle to the surface tender. With a single 8-bit frame of 512×512 pixels requiring 262 kbit, compression is essential for a practical implementation. Researchers from Florida Atlantic University, WHOI, and elsewhere have been comparing various compression algorithms in terms of their obtainable compression ratio and the resulting usefulness of the received image for typical ocean bottom features. Two common image compression approaches are the discrete cosine transform and the wavelet transform. A detailed discussion of applying wavelet transform ideas to underwater video compression was given in conjunction with an example whereby a video sequence of the Titanic was compressed by a factor of 100 while retaining good subjective quality [93]. A comparative analysis of discrete cosine transform and wavelet based approaches was done using the telemetry subsystem of an underwater vehicle being developed by the Defense Advanced Research Projects Agency (DARPA) [94]. Compression ratios of 50 : 1 were more typical in this application. Researchers at Florida Atlantic University are also examining compression algorithms in light of underwater imagery [95]. The emphasis in their work is on efficient vector quantization methods. Anecdotally, they achieve compression ratios of nearly 250 : 1 but the image degradation level is not well quantified. The common theme in all these reports is an effort to leverage unique aspects of underwater imagery to facilitate compression. One such aspect is that a typical image contains a few features in an otherwise uniform background. In addition, significant correlations may exist between successive images when a static seafloor is observed from a moving platform and may be exploited by frame-to-frame compression algorithms.

A common assumption in the evaluation of underwater telemetry systems is the reduction of the ocean channel to one with standard fading characteristics, either Rayleigh or Rician. This simplification often permits analytic performance estimates useful for forecasting channel coding performance. Sequential decoding of a long-constraint-length convolutional code in a simulated Rayleigh faded ocean channel was considered by Catipovic [96]. Under the assumed conditions, substantial coding gains were obtained provided the SNR per bit exceeded about 13 dB. The poor performance at a low SNR, typical of sequential decoders below a certain threshold, poses an obstacle to using a coding solution such as this for the extension of telemetry into ever more challenging environments where the SNR is severely limited.

In a recent work, a proposed mapping of available frequency tones in an MFSK system to Hadamard code words integrated the use of frequency diversity and channel coding [53]. There were three noteworthy contributions from this work. First, the use of equal weight code words (constant power) results in some simplification of the receiver (also implemented in DATS). Second, Proakis brings the techniques of concatenated coding to the underwater telemetry community, highlighting their high coding gains for a given coding complexity. Finally, and perhaps most importantly, the exploitation of frequency diversity with coding rather than simple redundant transmissions was shown to offer substantial increases in error protection against Rayleigh fading channels.

An incoherent modem under development by the Naval Command, Control and Ocean Surveillance Center, RDT&E Division, combines these two approaches by concatenating a long-constraint-length outer code with a Hadamard inner code [52]. As with the previous works, results are only given for a simulated Rayleigh faded channel. A novel scheme is proposed whereby the tonal alphabet hops from time slot to time slot in predetermined manner, enabling multiuser scenarios by giving each user a unique hopping pattern. The patterns are chosen to minimize the occurrence of "collisions" in the signal space. The coding redundancy coupled with a clipping procedure invoked when a strong user collides with a weak user can accommodate several collisions in a codeword. A comprehensive evaluation of the system must await experimental results.

A trait shared by most coding applications is the reduction in true data rate due to the introduction of redundancy. An alternate approach to obtaining the redundancy without a commensurate data rate penalty is to expand the signaling alphabet, a technique known as trellis coded modulation (TCM). The use of TCM as an outer code subsequent to coherent equalization and demodulation has been shown to significantly improve modest error probabilities in simulations [97]. The previously mentioned ACOMM ATD program has experimentally demonstrated this performance. A more compelling challenge will be to integrate TCM into the decision feedback architecture in order to lower the SNR threshold for reliable DFE operation. The relative paucity of papers discussing channel coding issues in the underwater channel should be seen as a call for more substantive work in this area.

The ocean environment poses requirements that may well advance coding techniques into heretofore unexplored regimes. Specifically, the limited bandwidth severely penalizes excessive redundancy while the substantial link latency discourages classical solutions such as automatic repeat requests. As an example, the use of long-constraint-length codes in an environment where the statistics vary on time scales comparable to the constraint length may pose challenges for low rate applications to adverse channels. As another example, attempts to expand the performance envelope of coherent telemetry systems will likely require a close integration of the DFE and decoder in decision-directed algorithms. The absence of high-fidelity channel models that capture both the reverberant and dynamic character of the ocean, which precludes obtaining estimates of the complex envelope statistics, may well hinder progress in both of these areas.

VII. Underwater Acoustic Networks

The last five years have witnessed a surge of interest in underwater acoustic networks. Although sporadic interest in multiple point communication is found in earlier literature, the relatively recent emphasis on synoptic, spatially sampled oceanographic surveillance has provided an impetus to the transfer of networked communication technology to the underwater environment. One vision, called the Autonomous Oceanographic Surveillance Network (AOSN), is promulgated by the Office of Naval Research (ONR) [98]. It calls for a system of moorings, surface buoys, and AUV's to coordinate their sampling via an acoustic telemetry network. A functioning underwater acoustic telemetry network is clearly a key component of such an architecture. While the relevant literature in this area is quite sparse, we review published accounts of the classical multiple access strategies of code division multiple access (CDMA) and time division multiple access (TDMA) as well as spatial division multiple access (SDMA), which is perhaps unique to the underwater channel.

Each of the classical multiple access strategies face obstacles when confronting the underwater channel. The comparatively slow speed of sound underwater leads to substantial latency in transmissions. Substantial one-way travel times set a significant minimum transmission time irrespective of information quantity. This latency compromises the usefulness of TDMA. The well-known bandwidth limitations pose clear difficulties to frequency division multiple access (FDMA). In addition to the spectral limitations, CDMA suffers from the severe channel reverberation that may lead to degradation of the code correlation properties, i.e., smaller codeword distances. Insufficient channel coherence time may also limit the processing gain for large spreading factors. Another unique obstacle noted by researchers is the limitation on available energy [99] which penalizes redundant transmissions. Optimization of power usage was considered by Northeastern University in demonstrating the tradeoff between many closely separated nodes and fewer widely separated nodes for a fixed range [100].

Several protocols have been suggested for use in underwater acoustic networks. Adapting a strategy used in packet radio networks, researchers from Northeastern University presented a decentralized system where the nodes adaptively learn the network parameters such as node numbers, link quality, and connectivity [101]. Modifications aimed at decreasing delay and retransmission attempts are introduced. The price of an adaptive network, however, is a high control packet overhead (2000 bit per control packet in one example). Performance in the face of frequent link outages would seem to be a concern. No experimental results are given. In another report, an asynchronous access approach is presented whereby nodes transmit packets on demand [102]. In the absence of an acknowledgment, the node waits a preset time, depending on its priority level, before transmitting again on the common channel. The network performance, either experimental or simulated, is not reported. In fact, none of the reviewed papers report typical network metrics such as latency, packet retries, or channel usage. As a first step toward a CDMA-based network, Boulanger reports on a design methodology for variable-length spreading codes [103]. Experimental results are given for a 520-Hz-bandwidth 1666-Hz-carrier spread spectrum signal. Spreading factors of 63 and 72 were considered, leading to over 18 dB of pulse compression gain at the receiver. The work did not, however, provide for multiple sources. As such, channel-imposed degradation of code correlation properties was impossible to evaluate.

A protocol particularly suited to noncoherent signaling was presented that relies on unique frequency hopping patterns for each node to allow multiple access [104]. The network is initialized with a coordinated transmission of channel probes to ascertain channel parameters followed by transition to the higher rate frequency-hopped spread spectrum protocol. Protection against residual multiple access interference (MAI) is assured through error-correction coding. As with the other reports, no network performance in terms of conventional measures is reported. Finally, a summary of several of the published attempts at underwater acoustic networks is given by Johnson *et al.* [105].

The Acoustic Local Area Network (ALAN) implemented in Monterey Canyon off the California coast is briefly described. A master node located on the surface polls a network of bottom-mounted nodes over an essentially vertical channel. A form of TDMA is implemented whereby latency is adaptively estimated and accounted for in the flow control. A similar network was evaluated in an ice-covered New Hampshire lake as well as in the Arctic. The experimental work was mostly limited to quantifying link availability rather than full-scale network performance. Nevertheless, these efforts stand out as virtually the only open ocean testing of an underwater acoustic telemetry network. It would appear that the difficulties in establishing a point-to-point link in the ocean have delayed substantive work in network architectures.

The complex spatial reverberation behavior of underwater acoustic channels gives rise to the possibility of using measurements from a spatial array of hydrophones to discriminate multiple cochannel users. The underwater acoustic channel, acting as a multimode waveguide, is quite unique in this respect. One might call this approach spatial division multiple access (SDMA). A hybrid approach has been suggested whereby a multichannel receiver is used to augment the cochannel interference suppression afforded by classical CDMA techniques [106]. Under some stringent statistical assumptions, an analytic expression for optimal mean square error of two receiver versions is derived. The first version, a centralized receiver, simultaneously makes decisions regarding all user's symbols. The second version, a decentralized receiver, has access to multiple channels but only makes decisions regarding one user. Experimental data is given for two users separated by 5 m in a shallow 18-m water column. With a 2-kHz-bandwidth BPSK signal centered at 15 kHz, two asynchronous users transmitted with spreading factors of six and a 10-dB power level differential over a range of 750 m. The system is anecdotally reported to have been successful.

Recognizing the potentially large complexity of a centralized multiuser receiver, the performance impact of using a decentralized receiver with a many-to-few spatial channel precombiner prior to a conventional multichannel DFE has been examined [107]. The authors show that substantial complexity reduction may be achieved with minimal performance loss in some circumstances. A separate data set with 1-kHz bandwidth signals and spreading factors of 3 are reported for a deep-water (2500 m depth) channel [108]. Range and bit error rates are not reported.

Multiuser detection relying *entirely* on the unique arriving spatial structure of each user's signal has been proposed using a two-stage filter [109]. The first stage is restricted to spatial taps while the second stage is restricted to temporal taps. Each is adapted at separate rates using a novel, perhaps *ad hoc*, energy metric. The algorithm was tested on a multiuser data set constructed by superimposing two QPSK single-user data sets taken in shallow water at 18 and 30 nautical mile ranges. While the signal bandwidth was 166 Hz, no carrier frequency is specified. Error rates are derived from output SNR measurements assuming Gaussian noise statistics. While presumably the effectiveness of SDMA is a strong function of the user impulse responses, this work and others only report success anecdotally for specific channels. Substantive conclusions must await more widespread testing. Nonetheless, this unique approach to underwater acoustic networks may offer promising solutions to the obstacles presented to TDMA, FDMA, and CDMA.

The combination of large latency compared to message size, limited bandwidth compared to data rate, and constrained power resources all conspire to challenge the development of a successful underwater acoustic network. In fact, as has been described earlier in this review, even point-to-point telemetry links, the fundamental unit of a network, remains an active area of research. As AUV's mature and the vision of a collection of ocean surveillance platforms acting in concert becomes closer at hand, research into underwater acoustic networks should swell. Adapting classical protocols to the unique constraints of the ocean channel and even developing completely novel approaches, as in the case of SDMA, will be the future challenge.

VIII. Alternative Modulation Strategies

FSK and QAM, in their various forms, have dominated digital underwater acoustic communication applications. Some researchers, however, have begun to explore alternative modulation schemes motivated largely by the need to mitigate temporal reverberation of the channel. We will review published applications of multicarrier modulation, parametric transduction, and several others.

Multicarrier modulation (MCM) is implemented by dividing the available bandwidth into a sequence of subchannels. Signals confined to each of the subchannels are generated and may employ any form of coherent or incoherent modulation. In fact, multiple frequency shift key (MFSK) signaling is one form of MCM. Typically, the bandwidth of the subchannels is a small fraction of the overall bandwidth leading to two principle advantages. First, the power allocated to each subchannel may be explicitly controlled. In principle, this is required to achieve channel capacity in a colored noise environment. In practice, this requires detailed knowledge of the ocean channel by the transmitter and is difficult to achieve. Second, if the inverse of the subchannel bandwidth is significantly larger than the ocean channel delay spread, the energy associated with intersymbol interference is much less than the total symbol energy and may often be neglected. One of the earliest self-proclaimed MCM implementations was, essentially, an MFSK system where binary FSK is to be transmitted over a set of subchannels [110]. An experimental demonstration used 3.6 kHz of bandwidth to transmit 250 bit/s up to 2 km in less than 30 m of water. BER's were typically less than 10^{-4}. A unique challenge facing coherent MCM is the need to estimate and track each subchannel.

The lower signal bandwidths on each subchannel, compared to the channel variability rate, severely restricts the performance of any adaptive algorithms and alternative channel estimation approaches may be needed. A recent MCM proposal employed orthogonal simultaneous pulse shapes for a data sequence and a training sequence [111]. Although their simulated results appear promising, the combined effects of the bandwidth expansion required to obtain orthogonal pulse shapes (a factor of 4) and devotion of half the power to the training sequence may well erode any advantages of the system over current approaches. Other discussions of MCM may be found in the literature but, lacking any experimental results, the value of MCM to underwater telemetry is yet to be shown.

While spread spectrum modulation has been proposed and generally discussed for applications such as multiple user access and covert signaling, a detailed report of experimentally obtained processing gains for a variety of spreading sequences has been given by Loubet [112]. A 2-kHz carrier with 500 Hz of bandwidth successfully conveyed 80.6 bit/s over 5 nm using a family of Gold codes. The 9-dB processing gain was sufficient to overcome an SNR of 0 dB except during any substantial fades. Gold codes (64 bit) were used to obtain 18 dB of processing gain, enabling operation over 45 nm with an average SNR of –5 dB. The signal parameters were slightly different with a 1.5-kHz carrier, 375-Hz bandwidth, and a data rate of 35.7 bit/s. Using a unique 72-bit code, BER's less than 0.2% were achieved under an SNR of −14 dB, implying most, if not all, of the expected 18.5 dB of processing gain was achieved. At some point, one would expect channel coherence issues to limit the amount of processing gain obtainable, but that analysis must await further research.

As a final example of an alternative modulation strategy, Sari has proposed digital pulse position modulation (PPM) [113]. Voice data with an encoded digital rate of 2.4 kbit/s is modulated onto a 70-kHz carrier. Each symbol is 1 ms in duration with the data denoted by which of eight time slots within the symbol interval contains energy. Multipath effects are explicitly ignored and test results are only given for two benign laboratory conditions. Sari also proposes joining the PPM technique with optical modulation of an underwater laser. Once again, the test results are given for exceptionally mild laboratory conditions.

Parametric transduction is not, strictly speaking, a modulation method. By exploiting a nonlinear response of the underwater medium, a small, high primary frequency source is able to transform a region of water into a large, low secondary frequency source. All of the modulation approaches discussed thus far may be projected by these sources. The primary advantage of the system is the ability to generate highly directive low-frequency signals with obvious applications to multipath reduction. The primary disadvantage is the substantial power efficiency penalty incurred. A currently funded European research initiative, PARACOM, has developed and tested parametric arrays. Coates *et al.* have built and tested a system with a 300-kHz primary frequency array that drives a 50-kHz secondary frequency [114]. DPSK signals with 10-kHz bandwidth were successfully transmitted in two benign environments, namely dockside and over 130 m in a freshwater lake. Although the results are promising thus far, definitive conclusions must await application in less benign channel conditions. In work directed toward confirming the distortion effects of the conversion process, Loubet reports results from a parametric array with a 50-kHz primary frequency and a 6-kHz secondary array with up to 8 kHz of bandwidth available [115]. He reports FSK signals conveying 2 kbit/s over a 2-km range with a BER less than 10^{-4}. Highlighting the severe power requirements, the secondary array of his system had an effective source level 40 dB less than that of the primary. A potential issue not addressed in either of these publications is the high pointing accuracy requirement that highly directive sources demand. Perhaps future results will clearly contrast parametric transduction with more common systems, allowing more useful assessments.

While these alternative modulation strategies may eventually make substantial contributions to the underwater acoustic telemetry field, careful comparisons with existing, successful, approaches in a wide variety of channel conditions are sorely needed. Many factors including power requirements, data rates, error rates, computational complexity, and range compete to define a system. Only by fairly comparing approaches within this set of common metrics may the worth of an approach be truly understood.

IX. Conclusions

The early 1980's saw telemetry systems capable of achieving a data rate * range product of approximately 0.5 km*kbit. By the mid-1990's, fielded systems were achieving nearly 40 km * kbit in shallow waters and approximately 100 km * kbit in deep waters. Much of this advance came with the introduction of coherent modulation systems and the concurrent availability of processors that could support the complex receiver algorithms. Many other important advances have been made in the areas of high-rate incoherent modulation and error control coding. Nonetheless, modem operation in more adverse channels would seem to require explicit incorporation of the underlying ocean telemetry channel physics. Acoustic propagation models tailored to telemetry applications are sorely needed. Channel characterization that captures the time variability of the channel is a necessary component of these models. With such *a priori* knowledge in hand, one may envision model-based receivers that can efficiently represent the challenging littoral and surf zone environments currently under consideration. While future research directions remain open to discussion, this review has made clear the expanded breadth of research in underwater acoustic telemetry that has grown over the last twenty years.

References

[1] D. J. Garrood, "Applications of the MFSK acoustical communication system," presented at the Oceans'81, Boston, MA, 1981.
[2] A. Baggeroer, D. E. Koelsch, K. von der Heydt, and J. Catipovic, "DATS—A digital acoustic telemetry system for underwater communications," presented at the Oceans'81, Boston, MA, 1981.
[3] M. Stojanovic, J. A. Catipovic, and J. G. Proakis, "Adaptive multi-channel combining and equalization for underwater acoustic communications," *J. Acoust. Soc. Amer.*, vol. 94, pp. 1621–1631, 1993.
[4] ——, "Phase-coherent digital communications for underwater acoustic channels," *IEEE J. Oceanic Eng.*, vol. 19, pp. 100–111, 1994.

[5] A. Baggeroer, "Acoustic telemetry—An overview," *IEEE J. Oceanic Eng.*, vol. OE-9, pp. 229–235, 1984.
[6] J. Catipovic, "Performance limitations in underwater acoustic telemetry," *IEEE J. Oceanic Eng.*, vol. 15, pp. 205–216, 1990.
[7] M. Stojanovic, "Recent advances in high-speed underwater acoustic communication," *IEEE J. Oceanic Eng.*, vol. 21, pp. 125–136, 1996.
[8] R. Coates, "Underwater acoustic communications," presented at the Oceans'93, Victoria, BC, Canada, 1993.
[9] F. B. Jensen, W. A. Kuperman, M. B. Porter, and H. Schmidt, *Computational Ocean Acoustics*. Woodbury, NY: AIP Press, 1994.
[10] S. Appleby and J. Davies, "Time, frequency, and angular dispersion modeling in the underwater communications channel," presented at the Oceans'98, Nice, France, 1998.
[11] C. Bjerrum-Niese and R. Lutzen, "Stochastic simulation of acoustic communication in turbulent, shallow water," IEEE J. Oceanic Eng., submitted for publication.
[12] A. Essebbar, G. Loubet, and F. Vial, "Underwater acoustic channel simulations for communication," presented at the Oceans'94, Brest, France, 1994.
[13] A. Essebbar and E. Vercelloni, "Simulation of communication system for underwater acoustics," presented at the Oceans'95, San Diego, CA, 1995.
[14] X. Geng and A. Zielinski, "An Eigenpath underwater acoustic communication channel model," presented at the Oceans'95, San Diego, CA, 1995.
[15] Y.-H. Yoon and A. Zielinski, "Simulation of the equalizer for shallow water acoustic communication," presented at the Oceans'95, San Diego, CA, 1995.
[16] A. Zielinski, Y.-H. Yoon, and L. Wu, "Performance analysis of digital acoustic communication in a shallow water channel," *IEEE J. Oceanic Eng.*, vol. 20, pp. 293–299, 1995.
[17] Q. Wen and J. A. Ritcey, "Spatial diversity equalization applied to underwater communications," *IEEE J. Oceanic Eng.*, vol. 19, pp. 227–240, 1994.
[18] T. H. Eggen, "Underwater acoustic communication over doppler spread channels," *Joint Program in Oceanography/Applied Ocean Science and Engineering*, p. 307, 1997.
[19] H. P. Bucker, "A simple 3-D Gaussian beam sound propagation model for shallow water," *J. Acoust. Soc. Amer.*, vol. 95, 1994.
[20] H. Weinberg and R. E. Keenan, "Gaussian ray bundles for modeling high-frequency propagation loss under shallow water conditions," *J. Acoust. Soc. Amer.*, vol. 100, 1996.
[21] S. M. Flatte, R. Dashen, W. H. Munk, K. M. Watson, and F. Zachariasen, *Sound Transmission Through a Fluctuating Ocean*, 1st ed. Cambridge, MA: Cambridge University Press, 1979.
[22] P. Dahl, "On bistatic sea surface scattering: Field measurements and modeling," *J. Acoust. Soc. Amer.*, vol. 105, pp. 2155–2169, 1999.
[23] E. C. Westerfield, R. H. Prager, and J. L. Stewart, "Processing gains against reverberation (clutter) using matched filters," *IRE Trans. Information Theory*, vol. IT-6, pp. 342–348, 1960.
[24] R. Price and P. E. Green, "A communication technique for multipath channels," *Proc. IRE*, vol. 46, pp. 555–570, 1958.
[25] J. V. Evans and T. Hagfor, *Radar Astronomy*, New York: McGraw-Hill, 1968, . .
[26] P. Bello, "Characterization of randomly time-variant linear channels," *IEEE Trans. Commun. Syst.*, vol. CS-11, pp. 360–393, 1963.
[27] R. Kennedy, *Fading Dispersive Channels*, New York: Wiley, 1969.
[28] H. L. VanTrees, *Detection, Estimation, and Modulation Theory, Pt. III*, New York: Wiley, 1971.
[29] R. J. Urick, *Principles of Underwater Sound*, 3rd ed. Los Altos, CA: Peninsula Publishing, 1983.
[30] I. Dyer, Class notes for subject 13.85, Fundamentals of Underwater Acoustics, Massachusetts Institute of Technology.
[31] J. G. Proakis, *Digital Communications*, 3rd ed. Boston, MA: McGraw-Hill, 1995.
[32] D. Dowling and D. R. Jackson, "Coherence of acoustic scattering from a dynamic rough surface," *J. Acoust. Soc. Amer.*, vol. 93, pp. 3149–3157, 1993.
[33] R. Coates, *Underwater Acoustic Systems*, New York: Wiley, 1989.
[34] J. A. Rice, "Acoustic signal dispersion and distortion by shallow undersea transmission channels," presented at the High Frequency Acoustics in Shallow Water, Lerici, Italy, 1997.
[35] M. Badiey, J. Simmen, and S. Forsythe, "Frequency dependence of broadband propagation in coastal regions," *J. Acoust. Soc. Amer.*, vol. 101, pp. 3361–3370, 1997.
[36] F. M. Caimi, R. Tongta, M. Carroll, and S. Murshid, "Acoustic impulse response mapping for acoustic communications in shallow water," presented at the Oceans'98, Nice, France, 1998.
[37] J. Catipovic, A. B. Baggeroer, K. Von Der Heydt, and D. Koelsch, "Design and performance analysis of a digital acoustic telemetry system for the short range underwater channel," *IEEE J. Oceanic Eng.*, vol. OE-9, pp. 242–252, 1984.
[38] R. Galvin, R. F. W. Coates, L. S. Wang, and R. Stoner, "Measured channel sounding characteristis and their relationship with the performance of a parametric communication system," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.
[39] A. Zielinski, R. Coates, L. Wang, and A. Saleh, "High rate shallow water acoustic communication," presented at the Oceans'93, Victoria, BC, Canada, 1993.
[40] R. Galvin and R. F. W. Coates, "Analysis of the performance of an underwater acoustic communications system and comparison with a stochastic model," presented at the Oceans'94, Brest, France, 1994.
[41] C. A. Gray, G. T. Uehara, and S. Lin, "Bandwidth efficient modulation for underwater acoustic data communications," presented at the Oceans'94, Brest, France, 1994.
[42] A. Plaisant, "Long range acoustic communications,," presented at the Oceans'98, Nice, France, 1998.
[43] S. D. Morgera, "Multiple terminal acoustic communications system design," *IEEE J. Oceanic Eng.*, vol. 5, pp. 199–204, 1980.
[44] J. F. Pieper, J. G. Proakis, R. R. Reed, and J. K. Wolf, "Design of efficient coding and modulation for a Rayleigh fading channel," *IEEE Trans. Information Theory*, vol. IT, pp. 457–468, 1978.
[45] F. C. Jarvis, "Description of a secure reliable acoustic system for use in offshore oil blowout preventer (BOP) or wellhead control," *IEEE J. Oceanic Eng.*, vol. OE-9, pp. 253–258, 1984.
[46] R. F. W. Coates, "A deep-ocean penetrator telemetry system," *IEEE J. Oceanic Eng.*, vol. 13, pp. 55–63, 1988.
[47] W. Hill, G. Chaplin, and D. Nergaard, "Deep-ocean tests of an acoustic modem insensitive to multipath distortion," presented at the Oceans'88, Baltimore, MD, 1988.
[48] L. E. Freitag, J. S. Merriam, D. E. Frye, and J. A. Catipovic, "A long term deep water acoustic telemetry experiment," presented at the Oceans'91, Honolulu, Hawaii, 1991.
[49] L. Freitag and J. S. Merriam, "Robust 5000 bit per second underwater communication system for remote applications," presented at the Proceedings of Marine Instrumentation'90, Marine Technology Society, San Diego, CA, 1990.
[50] G. R. Mackelburg, "Acoustic data links for UUVs," presented at the Oceans'91, Honolulu, Hawaii, 1991.
[51] K. F. Scussel, J. A. Rice, and S. Merriam, "A new MFSK acoustic modem for operation in adverse underwater channels," presented at the Oceans'97, Halifax, NS, Canada, 1997.
[52] M. D. Green and J. A. Rice, "Error correction coding for communication in adverse underwater channels," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.
[53] J. G. Proakis, "Coded modulation for digital communications over Rayleigh fading channels," *IEEE J. Oceanic Eng.*, vol. 16, pp. 66–73, 1991.
[54] G. R. Mackelburg, S. J. Watson, and A. Gordon, "Benthic 4800 bps acoustic telemetry," presented at the Oceans'81, Boston, MA, 1981.
[55] L. O. Olson, J. L. Backes, and J. B. Miller, "Communication, control and data acquisition systems on the ISHTE lander," *IEEE J. Oceanic Eng.*, vol. OE-10, pp. 5–16, 1985.
[56] G. S. Howe, O. R. Hinton, A. E. Adams, and A. G. J. Holt, "Acoustic burst transmission of high rate data through shallow underwater channels," *Electron. Lett.*, vol. 28, pp. 449–451, 1992.
[57] J. H. Fischer, K. R. Bennett, S. A. Reible, J. H. Cafarella, and I. Yao, "A high data rate, underwater acoustic data-communications transceiver," presented at the Oceans'92, Newport, RI, 1992.
[58] M. Suzuki, T. Sasaki, and T. Tsuchiya, "Digital acoustic image transmission system for deep-sea research submersible," presented at the Oceans'92, Newport, RI, 1992.
[59] J. C. Jones, A. DiMeglio, L. S. Wang, R. F. W. Coates, A. Tedeschi, and R. J. Stoner, "The design and testing of a DSP, half-duplex, vertical DPSK communication link," presented at the OCEANS'97, Halifax, Nova Scotia, Canada, 1997.
[60] J. G. Proakis, "Adaptive equalization techniques for acoustic telemetry channels," *IEEE J. Oceanic Eng.*, vol. 16, pp. 21–31, 1991.
[61] A. Kaya and S. Yauchi, "An acoustic communication system for subsea robot," presented at the Oceans'89, Seattle, WA, 1989.

[62] M. Suzuki, K. Nemoto, T. Tsuchiya, and T. Nakanishi, "Digital acoustic telemetry of color video information," presented at the Oceans'89, Seattle, WA, 1989.

[63] B. Geller, J. M. Brossier, and V. Capellano, "Equalizer for high data rate transmission in underwater communications," presented at the Oceans'94, Brest, France, 1994.

[64] A. Goalic, J. Labat, J. Trubuil, S. Saoudi, and D. Rioualen, "Toward a digital acoustic underwater phone," presented at the Oceans'94, Brest, France, 1994.

[65] J. Labat, "Real time underwater communications," presented at the Oceans'94, Brest, France, 1994.

[66] P. S. D. Tarbit, G. S. Howe, O. R. Hinton, A. E. Adams, and B. S. Sharif, "Development of a real-time adaptive equalizer for a high-rate underwater acoustic data communications link," presented at the Oceans'94, Brest, France, 1994.

[67] S. M. Jarvis and N. A. Pendergrass, "Implementation of a multichannel decision feedback equalizer for shallow water acoustic telemetry using a stabilized fast transversal filters algorithm," presented at the Oceans'95, San Diego, CA, 1995.

[68] V. Capellano, G. Loubet, and G. Jourdain, "Adaptive multichannel equalizer for underwater communications," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.

[69] V. Capellano, "Performance improvements of a 50 km acoustic transmission through adaptive equalization and spatial diversity," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.

[70] S. Jarvis, R. Janiesch, K. Fitzpatrick, and R. Morrissey, "Results from recent sea trials of the underwater digital acoustic telemetry system," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.

[71] L. Freitag, M. Grund, S. Singh, S. Smith, R. Christenson, L. Marquis, and J. Catipovic, "A Bidirectional coherent acoustic communication system for underwater vehicles," presented at the Oceans'98, Nice, France, 1998.

[72] L. Freitag, M. Johnson, and M. Stojanovic, "Efficient equalizer update algorithms for acoustic communication channels of varying complexity," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.

[73] V. Capellano and G. Jourdain, "Comparison of adaptive algorithms for multi-channel adaptive equalizers. Applications to underwater acoustic communication," presented at the Oceans'98, Nice, France, 1998.

[74] M. Johnson, L. Freitag, and M. Stojanovic, "Improved doppler tracking and correction for underwater acoustic communication," presented at the ICASSP'97, Munich, Germany, 1997.

[75] M. Johnson, D. Brady, and M. Grund, "Reducing the complexity requirements of adaptive equalization in underwater acoustic communications," presented at the Oceans'95, San Diego, CA, 1995.

[76] M. Kocic, D. Brady, and M. Stojanovic, "Sparse equalization for real-time digital underwater acoustic communications," presented at the Oceans'95, San Diego, CA, 1995.

[77] J. Homer, I. Mareels, R. R. Bitmead, B. Wahlberg, and F. Gustafsson, "LMS estimation via structural detection," *IEEE Trans. Signal Processing*, vol. 46, pp. 2651–2663, 1998.

[78] D. E. Lambert, N. A. Pendergrass, and S. M. Jarvis, "An adaptive block decision feedback receiver for improved performance in channels with severe intersymbol interference," presented at the Oceans'96, Fort Lauderdale, FL, 1996.

[79] J. Gomes and V. Barroso, "Blind decision-feedback equalization of underwater acoustic channels," presented at the Oceans'98, Nice, France, 1998.

[80] ——, "Blind equalization using a radial basis function neural network," presented at the Oceans'95, San Diego, CA, 1995.

[81] J. Labat, O. Macchi, and C. Laot, "Adaptive decision feedback equalization: Can you skip the training period?," *IEEE Trans. Commun.*, vol. 46, pp. 921–930, 1998.

[82] A. G. Bessios and F. M. Caimi, "Multipath compensation for underwater acoustic communications," presented at the Oceans'94, Brest, France, 1994.

[83] M. Feder and J. A. Catipovic, "Algorithms for joint channel estimation and data recovery—Application to equalization in underwater communications," *IEEE J. Oceanic Engineering*, vol. 16, pp. 42–55, 1991.

[84] G. Loubet and E. Petit, "Underwater acoustic communications with viterbi detector," presented at the Oceans'95, San Diego, CA, 1995.

[85] J. A. Catipovic and L. E. Freitag, "Spatial diversity processing for underwater acoustic telemetry," *IEEE J. Oceanic Eng.*, vol. 16, pp. 86–97, 1991.

[86] G. B. Henderson, A. Tweedy, G. S. Howe, O. Hinton, and A. E. Adams, "Investigation of adaptive beamformer performance and experimental verification of applications in high data rate digital underwater communications," presented at the Oceans'94, Brest, France, 1994.

[87] G. S. Howe, P. S. D. Tarbit, O. R. Hinton, B. S. Sharif, and A. E. Adams, "Sub-sea acoustic remote communications utilizing an adaptive receiving beamformer for multipath suppression," presented at the Oceans'94, Brest, France, 1994.

[88] M. Stojanovic, J. A. Catipovic, and J. G. Proakis, "Reduced-complexity spatial and temporal processing of underwater acoustic communication signals," *J. Acoust. Soc. Amer.*, vol. 98, pp. 961–972, 1995.

[89] B.-G. Song and J. A. Ritcey, "Spatial diversity equalization for MIMO ocean acoustic communication channels," *IEEE J. Oceanic Eng.*, vol. 21, pp. 505–512, 1996.

[90] J. A. Neasham, D. Thompson, A. D. Tweedy, M. A. Lawlor, O. R. Hinton, A. E. Adams, and B. S. Sharif, "Combined equalization and beamforming to achieve 20 kbit/sec acoustic telemetry for ROVs," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.

[91] D. Thompson, J. Neasham, B. S. Sharif, O. R. Hinton, A. E. Adams, A. D. Tweedy, and M. A. Lawlor, "Performance of coherent PSK receivers using adaptive combining, beamforming, and equalization in 50 km underwater acoustic channels," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.

[92] D. Albonico, F. Fohanno, and J. Labat, "Test of an high data rate acoustic link in shallow water," presented at the Oceans'98, Nice, France, 1998.

[93] D. F. Hoag, V. K. Ingle, and R. J. Gaudette, "Low-bit-rate coding of underwater video using wavelet-based compression algorithms," *IEEE J. Oceanic Eng.*, vol. 22, pp. 393–400, 1997.

[94] R. L. Eastwood, L. E. Freitag, and J. A. Catipovic, "Compression techniques for improving underwater acoustic transmission of images and data," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.

[95] F. M. Caimi, D. M. Kocak, G. X. Ritter, and M. S. Schmalz, "Comparison and development of compression algorithms for AUV telemetry: Recent advancements," presented at the Ocean Community Conference'98, Baltimore, MD, 1998.

[96] J. A. Catipovic and A. B. Baggeroer, "Performance of sequential decoding of convolutional codes over fully fading ocean acoustic channels," *IEEE J. Oceanic Eng.*, vol. 15, pp. 1–7, 1990.

[97] L. V. Subramaniam, B. S. Rajan, and R. Bahl, "Trellis coded modulation schemes for underwater acoustic communications," presented at the Oceans'98, Nice, France, 1998.

[98] T. B. Curtin, J. G. Bellingham, J. Catipovic, and D. Webb, "Autonomous oceanographic sampling networks," *Oceanography*, vol. 6, pp. 86–94, 1993.

[99] J. A. Catipovic, D. Brady, and S. Etchemendy, "Development of underwater acoustic modems and networks," *Oceanography*, vol. 6, pp. 112–119, 1993.

[100] J. G. Proakis, M. Stojanovic, and J. A. Rice, "Design of a communication network for shallow water acoustic modems," presented at the Ocean Community Conference'98, Baltimore, MD, 1998.

[101] J. L. Talavage, T. E. Thiel, and D. Brady, "An efficient store-and-forward protocol for a shallow-water acoustic local area network," presented at the Oceans'94, Brest, France, 1994.

[102] S. M. Smith, J. C. Park, and A. Neel, "A peer-to-peer communication protocol for underwater acoustic communication," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.

[103] C. Boulanger, G. Loubet, and J. R. Lequepeys, "Spreading sequences for underwater multiple access communications," presented at the Oceans'98, Nice, France, 1998.

[104] M. D. Green and J. A. Rice, "Handshake protocols and adaptive modulation for underwater communication networks," presented at the Oceans'98, Nice, France, 1998.

[105] M. Johnson, D. Herold, and J. Catipovic, "The design and peformance of a compact underwater acoustic network node," presented at the Oceans'94, Brest, France, 1994.

[106] Z. Zvonar, D. Brady, and J. Catipovic, "Adaptive detection for shallow-water acoustic telemetry with cochannel interference," *IEEE J. Oceanic Eng.*, vol. 21, pp. 528–536, 1996.

[107] M. Stojanovic and Z. Zvonar, "Multichannel processing of broad-band multiuser communication signals in shallow water acoustic channels," *IEEE J. Oceanic Eng.*, vol. 21, pp. 156–166, 1996.

[108] Z. Zvonar, D. Brady, and J. Catipovic, "An adaptive decentralized multi-user receiver for deep-water acoustic telemetry," *J. Acoust. Soc. Amer.*, vol. 101, pp. 2384–2387, 1997.

[109] S. D. Gray, J. C. Preisig, and D. Brady, "Multiuser detection in an horizontal underwater acoustic channel using array observations," *IEEE Trans. Signal Processing*, vol. 45, pp. 148–160, 1997.

[110] S. Coatelan and A. Glavieux, "Design and test of a multicarrier transmission system on the shallow water acoustic channel," presented at the Oceans'94, Brest, France, 1994.

[111] E. Bejjani and J.-C. Belfiore, "Multicarrier coherent communications for the underwater acoustic channel," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.

[112] G. Loubet, V. Capellano, and R. Filipiak, "Underwater spread-spectrum communications," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.
[113] H. Sari and B. Woodward, "Underwater acoustic voice communications using digital pulse position modulation," presented at the Oceans'98, Nice, France, 1996.
[114] R. F. W. Coates, M. Zheng, and L. Wange, "'BASS 300 PARACOM': A 'model' underwater parametric communication system," *IEEE J. Oceanic Eng.*, vol. 21, pp. 225–232, 1996.
[115] G. Loubet, F. Vial, A. Essebbar, L. Kopp, and D. Cano, "Parametric transmission of wide-band signals," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.
[116] A. G. Bessios and F. M. Caimi, "Fast underwater acoustic data link design via multicarrier modulation and higher order statistics equalization," presented at the Oceans'95, San Diego, CA, 1995.
[117] D. Billon and B. Quellec, "Performance of high data rate acoustic underwater communication systems using adaptive beamforming and equalization," presented at the Oceans'94, Brest, France, 1994.
[118] D. Brady and J. A. Catipovic, "Adaptive multiuser detection for underwater acoustical channels," *IEEE Journal of Oceanic Engineering*, vol. 19, pp. 158–165, 1994.
[119] D. Carvalho, F. Blackman, and R. Janiesch, "The results of several acoustic telemetry tests in both shallow and deep water," presented at the Oceans'95, San Diego, CA, 1995.
[120] J. Catipovic and A. B. Baggeroer, "Analysis of high frequency multitone transmissions propagated in the marginal ice zone," *Journal of the Acoustical Society of America*, vol. 88, pp. 185–190, 1990.
[121] J. J. Davies and S. A. Pointer, "UW communication system design for severely dispersed channels," presented at the Oceans'98, Nice, France, 1998.
[122] A. Judge, A. Taylor, A. Duguid, V. Allen, M. Hill, D. Woodroffe, J. Harrington, D. Baudais, and I. Uhrich, "Automatic data acquisition system installed in offshore canadian arctic well: Monitoring precise temperatures by acoustic telemetry," presented at the Oceans'87, Halifax, Nova Scotia, Canada, 1987.
[123] C. Kuchpil, A. L. F. Xavier, J. A. P. da Silva, and M. R. B. P. L. Jimenez, "Autonomous control system for offshore oil explotation using digital acoustic communication," presented at the Oceans'97, Halifax, Nova Scotia, Canada, 1997.
[124] W. K. Lam, R. F. Ormondroyd, and J. J. Davies, "A Frequency domain adaptive coded decision feedback equalizer for a broadband UWA COFDM system," presented at the Ocean'98, Nice, France, 1998.
[125] L. R. LeBlanc, J. M. Cuschieri, M. Singer, and P. P. Beaujean, "Coherent path measurement for characterization of the underwater acoustic channel," presented at the Oceans'96, Ft. Lauderdale, FL, 1996.
[126] S. Y. Miller and S. C. Schwartz, "Integrated spatial-temporal detectors for asynchronous gaussian multiple access channels," *IEEE Transactions on Communications*, vol. 43, pp. 396–411, 1995.
[127] R. H. Owen, B. V. Smith, and R. F. W. Coates, "An experimental study of rough surface scattering and its effects on communication coherence," presented at the Oceans'94, Brest, France, 1994.
[128] J. C. Preisig, "Reduced complexity adaptive array processing for multi-user communications," presented at the Oceans'95, San Diego, CA, 1995.
[129] B. Quellec, C. Bjerrum-Niese, L. Bjorno, B. Henderson, G. Jourdain, and A. Ishoy, "High image-data rate transmission in acoustical underwater communications," presented at the 2nd European Conference in Underwater Acoustics, Lyngby, Denmark, 1994.
[130] J. A. Ritcey and K. R. Griep, "Code shift keyed spread spectrum for ocean acoustic telemetry," presented at the Oceans'95, San Diego, CA, 1995.
[131] H. Sari and B. Woodward, "Underwater voice communications using a modulated laser beam," presented at the Oceans'98, Nice, France, 1998.
[132] P. R. Staal, "Use and evolution of a modular digital hydrophone array," presented at the Oceans'87, Halifax, Nova Scotia, Canada, 1987.
[133] G. P. Villemarette, "Programmable subsurface acoustic recording system," presented at the Oceans'88, Baltimore, MD, 1988.
[134] L. a. A. Z. Wu, "Multipath rejection using narrow beam acoustic link," presented at the Oceans'88, Baltimore, MD, 1988.
[135] K. Zhou, J. G. Proakis, and F. Ling, "Decision-feedback equalization of time-dispersive channels with coded modulation," *IEEE Trans. Commun.*, vol. 38, pp. 18–24, 1990.
[136] A. a. M. C. Zielinski, "Digital acoustic communication in multipath underwater channels," presented at the Oceans'85, 1985.

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