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Underwater Acoustic Communication

Milica Stojanovic

Electrical and Computer Engineering Department
Northeastern University
Boston, MA 02115

index terms: acoustic; communications; coherent; equalization; channel estimation; phase synchronizations; multipath; Doppler; sparse channels; diversity combining; beamforming; multiuser detection; interference suppression; time-reversal; multi-input-multi-output (MIMO) processing; multi-carrier modulation; OFDM; adaptive modulation; underwater networks.

The need for underwater wireless communications exists in applications such as remote control in off-shore oil and gas industry, pollution and climate monitoring in environmental systems, defense, collection of scientific data recorded at ocean-bottom stations and unmanned underwater vehicles, speech transmission between divers, and mapping of the ocean floor for detection of objects and discovery of new resources. Wireless underwater communications can be established by transmission of acoustic waves. The underwater acoustic communication channels, however, have limited bandwidth, and often cause signal dispersion in time and frequency [1], [2], [3]. Despite these limitations, underwater acoustic communications are a rapidly growing field of research and engineering.

Acoustic waves are not the only means for wireless communication underwater, but they are the only ones that can travel over longer distances. Radio waves that will propagate over longer distance through conductive sea water are the extra low frequency ones (30 Hz-300 Hz) which require large antennae and high transmitter powers [4], while higher-frequency signals will propagate only over very short distances (few meters at 10 kHz) [5]. Optical waves propagate best in the blue-green region, but in addition to attenuation, they are affected by scattering, and are limited to distances on the order of a hundred meters [6]. Narrow laser beams are power-efficient but require high pointing precision, while simple light-emitting diodes are not as power-efficient. Thus, acoustic waves remain the single best solution for communicating underwater, in applications where tethering is not acceptable and anything but a very short distance is to be covered.

Sound propagates as a pressure wave, and it can easily travel over kilometers, or even hundreds of kilometers, but to cover a longer distance, a lower frequency has to be used. In general, acoustic communications are confined to bandwidths that are low compared to those used for terrestrial radio communications. Acoustic modems that are in use today typically operate in bandwidths on the order of a few kHz, at a comparably low center frequency (e.g. 5 kHz centered at 10 kHz) [7]. While such frequencies will cover distances on the order of a kilometer, acoustic frequencies in the 100 kHz region can be used for shorter distances, while frequencies below a kHz are used for longer distances. Underwater acoustic communication over basin scales (several thousand kilometers) can be established in a single hop as well; however, the attendant bandwidth will be only on the order of 10 Hz [8]. Horizontal transmission is notoriously more difficult due to the multipath propagation, while vertical channels exhibit less distortion [9]. Frequency-dependent attenuation, multipath propagation, and low speed of sound (about 1500 m/s) which results in a severe Doppler effect, make the underwater acoustic channel one of the most challenging communication media.

The idea of sending and receiving information underwater is traced back all the way to the time of Leonardo Da Vinci, who is quoted for discovering the possibility to detect a distant ship by listening on a long tube submerged under the sea. In the modern sense of the word, underwater communications began to develop during the second World War, for military purposes. One of the first underwater communication systems was an underwater telephone, developed in 1945 in the United States for communicating with submarines [10]. This device used a single-sideband (SSB) suppressed carrier amplitude modulation in the 8 kHz-11 kHz frequency range, and it was capable of sending acoustic signals over distances of several kilometers. However, it was not until the development of VLSI technology that a new generation of underwater acoustic communication systems began to emerge. With the availability of compact digital signal processors (DSPs) with their moderate power requirements, it became possible for the first time to implement complex signal processing and data compression algorithms at the submerged ends of an underwater communication link.

During the past two decades, significant advancements have been made in the development of underwater acoustic communication systems in terms of their operational range and data throughput (overview articles [12]-[15] document the history of these developments). Acoustically controlled robots have been used to replace divers in performing maintenance of submerged platforms [16]; high-quality image transmission from the bottom of deepest ocean trenches (6500 km) to a surface ship was established [17], [18]; and data telemetry over horizontal distances in excess of 200 kilometers was demonstrated [19], [20].

As efficient communication systems are developing, the scope of their applications continues to grow, and so do the requirements on the system performance. Many of the developing applications, both commercial and military, are calling for real-time communication with submarines and autonomous, or unmanned underwater vehicles (AUVs, UUVs). Setting the underwater vehicles free from cables will enable them to move freely and refine their range of operation. The emerging communication scenario in which the modern underwater acoustic systems will operate is that of an underwater data network consisting of both stationary and mobile nodes [15]. This network is envisaged to provide exchange of data, such as control, telemetry and eventually video signals, between many network nodes. The network nodes, located on underwater moorings, robots and vehicles, will be equipped with various sensors, sonars and video cameras. A remote user will be able to access the network via a radio link to a central node based on a surface station.

Towards achieving these goals, current research and development efforts are focusing on the development of efficient communications and signal processing algorithms, design of efficient modulation and coding schemes, and techniques for mobile underwater communications. In addition, multiple access communication methods are being developed for underwater acoustic networks, and network protocols are being designed for long propagation delays and strict power requirements encountered in the underwater environment. Finally, data compression algorithms suitable for low-contrast underwater images, and related image and video processing methods are expected to enable their near real-time transmission through band-limited underwater acoustic channels.

System requirements

The achievable data throughput, and the reliability of an underwater acoustic communication system, as measured by the bit-error rate, vary from system to system, but are always subject to bandwidth limitations of the ocean channel. Unlike in the majority of other communication media, the use of underwater acoustic resources has not been regulated yet by standards, except for those that protect the marine life.

In the existing systems, there are usually four kinds of signals that are transmitted: control, telemetry, speech and video signals.

Control signals include navigation, status information, and various on/off commands for underwater robots, vehicles and submerged instrumentation such as pipeline valves or deep ocean moorings. The data rates up to about 1 kilobit per second (kbps) are sufficient for these operations, but very low bit-error rates may be required.

Telemetry data is collected by submerged acoustic instruments such as hydrophones, seismometers, sonars, current-meters, chemical sensors, and it also may include low rate image data. Data rates on the order of one to several tens of kbps are required for these applications. The reliability requirements are not so stringent as for the command signals, and a probability of bit error of $10^{-3} - 10^{-4}$ is acceptable for many of the applications.

Speech signals are transmitted between divers and a surface station or among divers. While some of the existing, commercially available diver communication systems still use analog communications, based on single-sideband modulation of the 3 kHz audio signal, research is advancing in the area of synthetic speech transmission for divers, as digital transmission is expected to provide better reliability. Transmission of digitized speech by linear predictive coding (LPC) methods requires rates on the order of several kbps to achieve close-to-toll quality. The bit error rate tolerance of about 10^{-2} makes it a viable technology for poor quality band-limited underwater channels [21].

Video transmission over underwater acoustic channels requires extremely high compression ratios if an acceptable frame transmission rate is to be achieved. Fortunately, underwater images exhibit low contrast and detail, and preserve satisfactory quality if compressed even to 2 bits per pixel. Compression methods, such as the JPEG (Joint Photographic Experts Group) standard discrete cosine transform, have been used to transmit 256×256 pixel still images with 2 bits per pixel, at transmission rates of about one frame per 10 second [17]. Further reduction of the required transmission rate seems to be possible by using dedicated compression algorithms, e.g., the discrete wavelet transform [22]. Video transmission appears possible using the modern compression techniques of the MPEG-4 type, which can operate at bit rates below 64 kbps with moderate detection performance as images will have satisfactory quality at bit error rates on the order of $10^{-3} - 10^{-4}$ [23].

Channel characteristics

Sound propagation underwater is primarily determined by transmission loss, noise, reverberation, and temporal and spatial variability of the channel. Transmission loss and noise are the principal factors determining the available bandwidth, range and signal-to-noise ratio. Time-varying multipath influences signal design and processing, which determine the information throughput and communication system performance.

Range and bandwidth

Transmission loss is caused by energy spreading and sound absorption. While the energy spreading loss depends only on the propagation distance, the absorption loss increases not only with range but also with frequency, thus setting the limit on the available bandwidth.

In addition to the nominal transmission loss, link condition is largely influenced by the spatial variability of the underwater acoustic channel. Spatial variability is a consequence of the waveguide nature of the channel, which results in such phenomena as formation of shadow zones. Transmission loss at a particular location can be predicted by many of the propagation modeling techniques [1] with various degrees of accuracy. Spatial dependence of transmission loss imposes particularly severe problems for communication with moving sources or receivers.

Noise observed in the ocean consists of man-made noise and ambient noise. In deep ocean, ambient noise dominates, while near shores, and in the presence of shipping activity, man-made noise significantly increases the noise level. Unlike the man-made noise, most of the ambient noise sources can be described as having a continuous spectrum and Gaussian statistics [1]. As a first approximation, the ambient noise power spectral density is commonly assumed to decay at about 20 dB/decade, both in shallow and deep water, over frequencies which are of interest to communication systems design. The exception are biological sources of noise, such as snapping shrimp which lives only in certain geographical areas and produces impulsive noise within the range of frequencies used by a typical communication system [24].

Frequency-dependent transmission loss and noise determine the relationship between the available range, bandwidth and SNR at the receiver input. This dependence is illustrated in Fig.1, which shows the frequency dependent portion of SNR for several transmission ranges. (The SNR is evaluated assuming spherical spreading, absorption according to Thorp [1] and a 20 dB/dec decay of the noise power spectral density.) Evidently, this dependence influences the choice of a carrier frequency for the desired transmission range. In addition, it determines the relationship between the available range and frequency band. Underwater acoustic communication links can be classified according to range as very long, long, medium, short and very short links. For a long-range system, operating over 10-100 km, the bandwidth is limited to few kHz (for a very long distance on the order of 1000 km, the available bandwidth falls below one kHz). A medium-range system operating over 1-10 km has a bandwidth on the order of 10 kHz, while only at very short ranges below about 100 m, more than a hundred kHz of bandwidth may be available.

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

Multipath

In a digital communication system which uses a single carrier, multipath propagation causes intersymbol interference (ISI), and an important figure of merit is multipath spread in terms of symbol intervals. While typical multipath spreads in the commonly used radio channels are on the order of several symbol intervals, in horizontal underwater acoustic channels they increase to several tens, or a hundred of symbol intervals for moderate to high data rates. For example, a commonly encountered multipath spread of 10 ms in a medium-range shallow water channel, causes the ISI to extend over 100 symbols if the system is operating at a rate of 10 kilosymbols per second (ksps). Multi-carrier systems avoid this problem by transmitting in parallel on many carriers, each occupying a narrow sub-band, whose width is kept well below the coherence frequency (inverse of the multipath spread).

The mechanisms of multipath formation in the ocean are different in deep and shallow water, and also depend on the frequency and range of transmission. Understanding of these mechanisms is based on the theory and models of sound propagation. Depending on the system location, there are several typical ways of multipath propagation. It is mostly the water depth that determines the type of propagation. The definition of shallow and deep water is not a strict one, but usually implies the region of continental shelves, with depth less than about 100 m, and the region past the continental shelves, where the water gets deeper. Two fundamental mechanisms of multipath formation are reflection at boundaries (bottom, surface and any objects in the water), and ray bending (rays of sound always bend towards regions of lower propagation speed). If the water is shallow, propagation will occur in surface-bottom bounces in addition to a possible direct path. If the water is deep, as in the regions past the continental shelves, the sound channel may form by bending of the rays toward the location where the sound speed reaches its minimum, called the axis of the deep sound channel. Because there is no loss due to reflections, sound can travel in this way over several thousands of kilometers. Alternatively, the rays bending upwards may reach the surface focusing in one point where they are reflected, and the process is repeated periodically. The region between two focusing points on the surface is called a convergence zone, and its typical length is 60 km-100 km.

The geometry of multipath propagation and its spatial dependence are important for communication systems which use array processing to suppress multipath (e.g. [25]). The design of such systems is often accompanied by the use of a propagation model for predicting the multipath configuration. Ray theory and the theory of normal modes provide basis for such propagation modeling.

Associated with each of the deterministic propagation paths (macro-multipaths), which can be modeled accurately, are random signal fluctuations (micro-multipath), which account for the time-variability of the channel response. Some of the random fluctuations can be modeled statistically [1],[2]. These fluctuations include surface scattering due to waves, which is the most important contributor to the overall time-variability of the shallow water channel. In deep water, in addition to surface scattering, internal waves contribute to the time-variation of the signal propagating along each of deterministic paths.

Surface scattering is caused by the roughness of the ocean surface. If the ocean were calm, a signal incident on the surface would be reflected almost perfectly, with the only distortion being a phase shift of π. However, wind-driven waves act as the displacement of the reflection point, resulting in signal dispersion. Vertical displacement of the surface can be well modeled as a zero-mean Gaussian random variable, whose power spectrum is completely characterized by the wind speed [1]. Motion of the reflection point results in frequency spreading of the surface-reflected signal, significantly larger than that caused by many other phenomena. Doppler spread of a signal component of frequency f caused by a single surface-reflection occurring at an incidence angle θ is $0.0175(f/c)w^{3/2}cos\theta$ where c is the speed of sound, nominally taken to be 1500 m/s, and w is the wind speed in m/s [1]. A moderate wind speed is on the order of 10 m/s. Highest Doppler spreads are most likely to be found in short range links, which use relatively high frequencies. For longer ranges, at which lower frequencies are used, the Doppler spread will be lower; however, multipath spread will increase as there will be more significant propagation paths. The exact values of multipath and Doppler spreads depend on the geometry of multipath on a particular link. Nevertheless, it can be said that the channel spread factor, i.e. the product of the Doppler spread and the multipath spread, can in general be expected to decrease with range.

As an example, Figs.2-4 each show an ensemble of channel impulse responses, observed as functions of delay over an interval of time. These figures describe channel responses obtained at three fundamentally different locations with different mechanisms of multipath formation. Fig.2 shows the impulse responses recorded in deep water of the Pacific ocean, off the coast of California. In this channel, propagation occurs over three convergence zones, which span 110 nautical miles. At each fixed time instant, the figure shows a realization of the channel impulse response magnitude as a function of delay. Looking at one channel response reveals that two or more signals arrive at the receiver at any given time. The multipath delay spread in this channel is on the order of 20 ms. The multiple arrivals have comparable energy, thus causing strong ISI. The amplitudes and phases of distinct arrivals may vary independently in time. Along the time axis, variation of the channel response is observed for each given delay. In this example, significant variation occurs over the shown 15 second interval. This channel does not have a well-defined principal, or strongest arrival, as evidenced by the fact that the maximum amplitude does not always occur at the same delay. The channel responses shown in Figs.2-4 are obtained by adaptive channel estimation techniques. In particular, a recursive least-squares algorithm is applied to 4-PSK signals transmitted over the channels at rates indicated in the figures. Fig.3 shows the impulse responses obtained in shallow water of the Atlantic ocean continental shelf, off the coast of New England, over a long distance of 48 nautical miles. This example shows a channel with a well-defined principal arrival, followed by multipath of lower energy. The extent of multipath is up to 50 ms. It is worth noting that even though the extended multipath may appear to have negligible energy, its contribution to the overall ISI cannot be neglected. This channel shows a slower time-variation than the one observed in Fig.2. In contrast, Fig.4 provides an example of a rapidly time-varying channel. These response were recorded in the shallow water of Buzzards Bay near the coast of New England, over a distance of 2 nautical miles. Of the three examples shown, this channel demonstrates the fastest time-variation, which is typical of a medium-range shallow water environment.

The factor that determines the performance of a digital communication system on a frequency-spread channel is the Doppler spread normalized by the symbol rate. In underwater acoustic channels, the normalized Doppler spread can approach values as high as 10^{-2}. The implications that the time-varying multipath bears on the communication system design are twofold. On the one hand, signaling at a high rate causes many adjacent symbols to interfere at the receiver, and requires sophisticated processing to compensate for the ISI. On the other hand, as pulse duration becomes shorter, channel variation over a single symbol interval becomes slower. This allows an adaptive receiver to efficiently track the channel on a symbol-to-symbol basis, provided, of course, a method for dealing with the resulting time-dispersion. Hence, time-varying multipath causes a trade-off in the choice of signaling rate for a given channel. Experimental results obtained on a rapidly varying shallow water channel [26] demonstrate these observations.

While there exists a vast knowledge of both deterministic and statistical modeling of sound propagation underwater, the use of this knowledge in modeling of communication channels has only recently begun to receive more attention (e.g., [27], [28],[29]). Modeling of the slower variations of the locally-averaged received signal power (large-scale modeling) offers some evidence in support of a log-normal model for the channel gain [30], [31],[29]. Experimental studies of the small scale fading have offered evidence to Rician [29, 30, 32] as well as Rayleigh phenomena [33]. Combining the effects of large- and small-scale fading leads to a mixture of log-normal and Rician distributions, which can be approximated in closed form by the compound K-distribution [34]. As far as the time-correlation properties are concerned, it is generally understood that coherence times on the order of 100 ms can be assumed for a general-purpose design.

In addition to the inherent random phenomena, motion of the transmitter, receiver, or a reflection point along the signal path causes the path distances to vary with time. The resulting Doppler effect is evident as time compression/dilation of the signal, which causes frequency shifting and bandwidth spreading/shrinking. The magnitude of the Doppler effect is proportional to the ratio v/c of the relative transmitter-receiver velocity to the speed of sound. Because the speed of sound is very low as compared to the speed of electro-magnetic waves, motion-induced Doppler distortion of an acoustic signal can be extreme. AUVs move at speeds that are on the order of a few m/s, but even without intentional motion, underwater instruments are subject to drifting with waves, currents and tides, which may occur at comparable velocities. The resulting v/c ratio is on the order of 10^{-3}, which, compared to the land-mobile radio communication systems where this value os on the order of 10^{-7}, presents a striking difference. Even after proper initial synchronization and resampling of the signal, the residual Doppler frequency offset presents a challenge to mobile acoustic communications.

Knowledge of statistical channel models has proven to be useful in the design and analysis of land-mobile radio systems, and first attempts at modeling the distribution and the correlation functions of the underwater acoustic mobile systems have been made [29]. More is certainly to come on this topics, and on related issues in channel coherence in time, frequency, and space.

System design

To overcome the difficulties of time-varying multipath dispersion, the initial design of underwater acoustic communication systems relied on the use of non-coherent modulation techniques, i.e. frequency-shift keying (FSK) with energy detection. In the early 80's, a system known as DATS (Digital Acoustic Telemetry System [35]), provided the basis for the first generation of commercial digital acoustic modems. Today, coded FSK is used in several acoustic modems, including the Woods Hole Oceanographis Institution's "micro-modem" [7] and the Teledyne-Benthos' "telesonar type B" modem [36]. While FSK relies on simple energy detection (non-coherent detection), and thus offers robustness to channel impairments, its bandwidth utilization is not efficient. Motivated by this fact, research in the 90s focused on investigating phase shift keying (PSK) and quadrature amplitude modulation (QAM) for underwater acoustic channels. These modulation methods offer more bits/sec per Hz of occupied bandwidth, but require a receiver that can track the channel and compensate for the time-varying multipath and phase distortion (coherent detection). That work resulted in a channel equalization/synchronization method [19], which forms the basis of a second generation of "high-speed" acoustic modems. Through the last decade, these modems have been used routinely in operations involving both stationary platforms and autonomous underwater vehicles (AUVs), over vertical and horizontal links at bit rates of about 5 kbps. Bit rates in excess of those available with the operational modems have been demonstrated as well, but these results are in the domain of experimental research. Research remains active on improved, and ever more sophisticated channel estimation and equalization methods for single-carrier as well as multi-carrier broadband systems.

Approaches to system design vary according to the technique used for overcoming the effects of intersymbol interference and signal phase variations. Specifically, these techniques may be classified according to (1) the signal design, i.e. the choice of modulation/detection method, and (2) the transmitter/receiver structure, i.e. the choice of array processing method and the equalization method, if any. In the following section, the design of several systems which have been implemented is described. While most of the existing systems operate on the vertical, or the very short-range channels, the systems under development often focus on the severely spread horizontal shallow water channels. Signal processing methods used in these systems are addressed in the subsequent sections.

Systems based on noncoherent modulation

Noncoherent detection of FSK (frequency shift keying) signals has been used for channels exhibiting rapid phase variation such as the shallow water long-range and medium-range channels. To overcome the ISI, the existing noncoherent systems employ signal design with guard times, which are inserted between successive pulses to ensure that all the reverberation will vanish before each subsequent pulse is to be received. The insertion of idle periods of time obviously results in a reduction of the available data throughput. In addition, because fading is correlated among frequencies separated by less than the coherence bandwidth (the inverse of the multipath spread), it is desired that only those frequency channels which are separated by more than the coherence bandwidth be used at the same time. This requirement further reduces the system efficiency unless some form of coding is employed so that the adjacent, simultaneously transmitted frequencies belong to different codewords. As an example, the system [37] for telemetry at a maximum of 5 kbps used a multiple FSK modulation technique in the 20-30 kHz band. This band was divided into 16 subbands, in each of which a 4-FSK signal is transmitted. Hence, out of a total of 64 channels, 16 are used simultaneously for parallel transmission of 32 information bits (2 information bits per one 4-channel subband). This system has successfully been used for telemetry over a 4 km shallow water horizontal path, and a 3 km deep ocean vertical path. It was also used on a less than

1 km long shallow water path, where probabilities of bit error on the order of $10^{-2} - 10^{-3}$ were achieved without coding. Despite the fact that bandwidth efficiency of this system does not exceed 0.5 bps/Hz, noncoherent FSK is a good solution for applications where moderate data rates and robust performance are required. An improved FSK system [38] used 128 subbands and employed coding. The essence of the coding method is a Hadamard H(20,5) code, in which each 5 input bits are encoded into 20 output bits (the minimum distance of this code is 10). The encoded bits dictate the choice of active subbands for transmission of the given codeword. The 20 subbands that are simultaneously used are chosen (among the 128 available) to be maximally separated, which ensures the least correlated fading, and thus provides diversity on time-varying underwater channels. Because of their robustness and simplicity of implementation, noncoherent signaling methods remain an essential part of acoustic modems used in field operations.

Systems based on coherent and differentially coherent modulation

With the goal of increasing the bandwidth efficiency of an underwater acoustic communication system, research focus has shifted towards phase-coherent modulation techniques, such as PSK and QAM. Phase-coherent communication methods, previously not considered feasible, were demonstrated in the early 90's to be a viable way of achieving high-speed data transmission over many of the underwater channels, including the severely time-spread horizontal shallow water channels [19],[20].

Depending on the method for carrier synchronization, phase-coherent systems fall into two categories: differentially coherent and purely phase-coherent. The advantage of using differentially encoded PSK (DPSK) with differentially coherent detection is the simple carrier recovery it allows. Most of the systems that employ DPSK methods have been used in vertical and very short range channels, where little multipath is observed and the phase stability is good.

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

Deep ocean vertical path channel was used by an image transmission system [17]. This is 4-DPSK system with carrier frequency of 20 kHz, capable of achieving 16 kbps bottom to surface transmission over 6500 m. Field tests of this system indicated the achievable bit error rates on the order of 10^{-4} with linear equalizer operating under an LMS algorithm.

Current state-of-the art in phase-coherent underwater communications is represented by the system [7]. This system is based on purely phase-coherent modulation and detection principles [19] of 4-PSK signals. The signals are transmitted at varying rates up to 7 kbps, depending upon the coding method used. The system's real-time operation in either stand-alone configuration, or as a node of a network, was demonstrated in varying environments, including shallow and deep water, under-ice and ocean-trench operations [18]. To overcome the ISI caused by multipath propagation, the system uses a decision-feedback equalizer operating under an RLS (recursive least squares) algorithm.

Signal processing methods for multipath compensation

Coherent systems fall into two types: single-carrier and multi-carrier systems. In single-carrier systems, a broadband information-bearing signal is directly modulated onto the carrier and transmitted over the channel. A typical high-rate acoustic signal occupies several kHz of bandwidth over which it experiences uneven channel distortion. This distortion must be compensated at the receiver through the process of equalization. Multi-carrier modulation bypasses this problem by converting the high-rate information stream into many parallel low-rate streams, which are then modulated onto separate carriers. The carriers are spaced closely enough such that the channel appears as frequency-flat in each narrow sub-band. After demodulation, each carrier's signal only has to be weighted and phase-synchronized, i.e. a single-coefficient equalizer suffices per carrier. Each of these methods has its advantages and disadvantages when it comes to practical implementation: single-carrier systems are capable of faster channel tracking but they need high-maintenance equalizers; multi-carrier systems are efficiently implemented using the fast Fourier transform (FFT), but they have high sensitivity to residual frequency offsets.

To achieve higher data rates, single-carrier systems based on phase-coherent signaling methods must allow for considerable ISI in the received signal. These systems employ either some form of array processing, or equalization methods, or a combination thereof, to compensate for the distortions. Three main approaches have been taken towards this end. The first two approaches use differentially coherent detection and rely on array processing to eliminate, or reduce multipath. The third approach is based on purely phase-coherent detection and the use of equalization together with array processing for exploitation of the multipath and spatial diversity.

Array processing for multipath suppression has been used both at the transmitter and at the receiver end. Transmitter arrays can be used to excite only a single path of propagation, but very large arrays are required. To overcome the need for a large array, the use of parametric sources has also been studied [39]. These highly directive sources rely on the nonlinearity of the medium in the vicinity of a transducer where two or more very high frequencies from the primary projector are mixed. The resulting difference frequency is transmitted by a virtual array formed in the water column in front of the projector. A major limitation of such a source is in its high power requirements. High directivity implies the problem of pointing errors, and careful positioning is required to ensure complete absence of multipath. These systems have been employed in shallow water channels where equalization is not deemed feasible due to rapid time-variation of the signal. Instead, a receiving array is employed to compensate for the possible pointing errors. Binary and quaternary DPSK signals were used achieving data rates of 10 kbps and 20 kbps, respectively, with a carrier frequency of 50 kHz. The estimated bit error rate was on the order $10^{-2} - 10^{-3}$, depending on the actual channel length. In general, it was found that the technique is more effective at shorter ranges.

Multipath rejection using adaptive beamforming at the receiver end only in another possibility. The beamformer [25] uses an LMS algorithm to adaptively steer nulls in the direction of a surface reflected wave. Similarly as in the case of the transmitter array, it was found that the beamformer encounters difficulties as the range increases relative to depth. To compensate for this effect, the use of an equalizer was considered to complement the performance of the beamformer. The equalizer operates under an LMS algorithm whose low computational complexity permits real-time adaptation at the symbol rate. A separate waveform is transmitted at twice the data rate for purposes of time-synchronization. The system was tested in shallow water at 10 kbps, using a carrier frequency of 50 kHz, and showed the estimated bit error rate of 10^{-2} without, and 10^{-3} with the equalizer.

A different method, based on purely phase-coherent detection, uses joint synchronization and equalization for combating the effect of phase variations and ISI [19, 20]. The equalization method is that of fractionally spaced decision-feedback equalization, used with an RLS algorithm. The system incorporates spatial signal processing in the form of multichannel equalization based on diversity combining. The phase-coherent methods have been tested in a variety of underwater channels with severe multipath, showing satisfactory performance regardless of the link geometry. The achieved data rates of up to 2 kbps over long range channels, and up to 40 kbps over shallow water medium-range channels, are among the highest reported to date. Below, these methods are discussed in more detail.

Multichannel signal processing for coherent detection

In many of the underwater acoustic channels multipath structure may exhibit one or more components which carry the energy similar to that of the principal arrival. As the time progresses, it is not unusual for these components to exceed in energy the principal arrival (e.g., see Fig.2). The fact that the strongest multipath component may not be well defined makes the extraction of carrier reference a difficult task in such a channel. To establish coherent detection in the presence of strong multipath, a technique based on simultaneous synchronization and multipath compensation may be used [19]. This technique is based on joint estimation of the carrier phase and the parameters of a decision-feedback equalizer, where the optimization criterion is minimization of the mean-squared error (MSE) in the data estimation process. In addition, the equalizer/synchronizer structure can be extended to include a number of input array channels [20, 40]. Spatial diversity combining has shown superior performance in a number of channels, as well as potential for dealing with several types of interference. In Fig.5, the multichannel equalizer is shown, preceded by an additional pre-combiner, which may or may not be used depending on the application and the number of available received channels.

The input signals to the baseband processor are the A/D converted array signals, brought to baseband using nominal carrier and lowpass filtering. The signals are frame-synchronized using a known channel probe (usually a short Barker sequence transmitted in phase and quadrature at the data rate). Baseband processing begins with downsampling, which may be carried out to as few as 2 samples per symbol interval ($N_s = 2$), since the signals are shaped at the transmitter to have a raised-cosine spectrum which limits their maximal frequency to less than $1/T$. Since there is no feedback to the analog part of the receiver, the method is suitable for an all-digital implementation.

For applications where transmitter and receiver are not moving at a high speed, but only drifting with water, no explicit adjustment of the sampling clock is needed. It will implicitly be accomplished during the process of adaptive fractionally spaced equalization. The front section of the equalizer will also perform adaptive matched filtering and linear equalization. To correct for the carrier offset, the signals in all channels are phase-shifted by the amount estimated in the process of joint equalization and synchronization. After coherent combining, the ISI resulting from the previously transmitted symbols (postcursors) is canceled in the feedback section of the equalizer. This receiver structure is applicable to any linear modulation format, such as M-PSK, or M-QAM, the only difference being in the way in which symbol decision is performed.

In addition to combining and equalization, signal processing at the receiver includes the operation of decoding if the signal at the transmitter was encoded. For example, in a DSP implementation of the receiver two coding methods are used: concatenated coding of an outer Reed Solomon code and an inner cyclic block code (Hamming, BCH), and punctured convolutional coding with interleaving. Alternatively, trellis coded modulation, compatible with PSK and QAM signals, provides an effective means of improving performance on a band-limited channel.

The receiver parameters that are adaptively adjusted are the weights of the pre-combiner, the tap-weights of the feedforward filters, the carrier phase estimates, and the tap-weights of the feedback filter. A single estimation error is used for the adaptation of all parameters. This error is the difference between the estimated data symbol at the input to the decision device, and its true value. During the initial training mode, the true data symbols are known. After the training period, when the receiver parameters have converged, the on-line symbol decisions are fed back to the equalizer and used to compute the error. The adaptive algorithm used to update the receiver parameters is a combination of the second-order digital phase-locked loop (PLL) for the carrier phase estimates, and the RLS algorithm for the multichannel equalizer tap weights. The complexity of the multichannel equalizer grows with the number of receiver array sensors. For this reason, the spatial pre-combiner may be used to limit the number of equalizer channels, but still make use of the diversity gain. The pre-combiner weights can be estimated jointly with the rest of adjustable parameters. The details of the joint adaptation are given in [40].

The receiver is adaptively adjusted to coherently combine the multiple signal arrivals, and thus exploit both spatial and temporal, or multipath diversity gain. In this manner, it differs from a receiver based on adaptive beamforming which is adjusted to null out the signal replicas arriving from angles different than that of the desired path. The signal isolated by a beamformer usually has to be processed by a separately optimized equalizer to compensate for the residual ISI which arises because the beamformer cannot completely eliminate the multipath interference. Since it is not constrained by angular resolution, the method of multichannel equalization may be used with as few as two input channels, and is applicable to a variety of underwater acoustic channels, regardless of the range-to-depth ratio. In applications where large arrays are available, the pre-combiner reduces receiver complexity, while preserving the multichannel diversity gain.

The method of adaptive multichannel combining and equalization was demonstrated to be effective in underwater channels with fundamentally different mechanisms of multipath formation. Experimental results include data rates of 2 kbps over three convergence zones

(200 km or 110 nautical miles) in deep water; 2 kbps over 90 km (50 nautical miles) in shallow water, and up to 40 kbps over 1-2 km in rapidly varying shallow water channels. This method is enhanced by the use of a pre-combiner [40] which reduces a large number of input channels to a smaller number for subsequent multichannel equalization. By careful design, full diversity gain can be preserved by this technique. More than one channel at the output of the combiner is usually required, but this number is often small (e.g., three). The fact that diversity gain may be preserved is explained by multipath correlation across the receiver array. In addition to the reduced computational complexity, smaller adaptive filters result in less noise enhancement, contributing to improved performance.

Interference cancellation and multi-user detection

Sources of interference in underwater acoustic channels include external interference and internal interference, generated within the system. External sources of interference include noise coming from on-board machinery or other nearby acoustic sources, as well as the propulsion and flow noise associated with the underwater vehicle launch process. Internal noise, which has signal-like characteristics, arises in the form of echo in full-duplex systems, and in the form of multiple-access interference generated by other users operating within the same network.

Methods for cancellation of interference in the form band-limited white noise and multiple sinusoidal interference were investigated in [41]. It was found that the multichannel receiver structure of Fig.5 was effective in canceling the interference while simultaneously detecting the desired signal. Noise cancellation is performed simply by providing a reference of the noise signal to one of the multichannel combiner inputs, while cancelation of the sinusoidal interferer may be performed even without the reference signal. By virtue of having the training sequence, the multichannel combiner has the capability to adaptively filter the interfering signal out, and extract the desired signal.

A multiple-access communication system represents a special case of structured interference environment, such as that arising in a code-division multiple access system based on direct sequence spread spectrum modulation. Similarly as before, the adaptive multichannel receiver of Fig.5 was experimentally shown to have excellent capabilities in the role of a multiuser detector [42]. For systems with high spreading gain, such as those used to provide additional low probability of detection (LPD) needed for operation in hostile environments, signal processing is performed at the chip rate (as opposed to symbol-rate) to accommodate the time-variation of the channel that can be significant during one symbol interval, thus taking advantage of the available processing gain [43].

Time-reversal

A different approach to learning the channel has been pursued through a technique called time-reversal or phase conjugation [44]. This technique uses a time-reversed replica of a received signal waveform to implement a filter matched to that waveform, and can operate either passively or actively. Passive time-reversal resides at the receiver side only, where its role is to acquire a probe signal and use it to perform low-complexity front-end matched filtering prior to multichannel equalization or interference cancellation [45, 46]. In contrast, active time-reversal operates at the transmitter side, where its role is to time-reverse the feedback signal and use it as the basic pulse (basic transmit waveform) that will best match the channel. By doing so, the transmitter, typically equipped with a large array, focuses its energy not only in time, but also in space (see [44] and references therein). In repeated actions of this type, both ends of the link can focus their energy. Passive time-reversal has also been used for multi-user detection [46].

Coding, turbo equalization and advanced channel estimation

With the feasibility of high rate communications established, research has been extremely active on a number of interesting topics [47]. Single-carrier modulation/detection is being improved using powerful coding and turbo equalization methods, [48], [49], while multi-carrier modulation/detection methods, which we discuss in the next section, have emerged as a viable alternative to single-carrier broadband modulation. Both types of systems have been extended to multi-input multi-output configurations that provide spatial multiplexing (the ability to send parallel data streams from multiple transmitters), and bit rates of several tens of kbps have been demonstrated experimentally.

Adaptive channel estimation has received special attention as it holds the key to improved equalization. The acoustic channel is often sparse, i.e. there are only several significant paths that populate the total and possibly long delay spread. This fact has an important implication on channel estimation and the associated equalization methods. Namely, if the entire multipath spread is represented by L samples taken at intervals $1/B$, where B is the system bandwidth, fewer than L coefficients may suffice to represent the channel response. Ideally, only as many coefficients as there are propagation paths, $P < L$, are needed. Channel modeling thus becomes an important aspect of signal processing, and sparsing has been investigated for decision-feedback equalization [50, 51], turbo equalization [48, 49], and multi-carrier detection [52, 53, 54]. It is also important to note that although a greater bandwidth implies more samples needed to represent the channel, it also implies a better resolution in delay (less smearing in the observable channel response). Hence, although the attendant signal distortion is perceived as more severe, channel estimation will be more efficient if a proper sparse model is used, which may in turn lead to improved signal processing. In addition, signaling at a higher rate enables more frequent channel observations and, consequently, easier channel tracking [26].

Multi-carrier systems

Multi-carrier modulation is a technique used to combat the frequency-selectivity of the channel. This technique, in the form of orthogonal frequency division multiplexing (OFDM), has been adopted for many of the wireless radio systems, including wireless local area networks (WLAN), digital audio and video broadcast (DAB/DVB), and the next generation of cellular systems. Over the past several years, it has also come into the forefront of acoustic communications research, and several efforts at implementing an OFDM acoustic modem have emerged as well.

The appeal of OFDM lies in the computational efficiency of FFT-based processing, and in the fact that it easily scales to different bandwidths. Unlike with single-carrier systems, where the equalizer length has to be adjusted in accordance with the bandwidth B because it determines the symbol duration and hence the extent of ISI, with OFDM it simply suffices to increase/decrease the number of carriers K, i.e. the size of the FFT, while keeping the same carrier separation $\Delta f = B/K$.

In addition, by virtue of having a narrowband signal on each carrier, OFDM is easily conducive to MIMO processing [52, 56, ?], adaptive modulation [57], and differentially coherent detection [58]. However, its sensitivity to frequency offset and time-variation of the channel demands special attention. Issues related to power efficiency also need to be kept in mind, as OFDM is sensitive to non-linear distortions [59].

OFDM signal processing encompasses two stages: pre-FFT synchronization and post-FFT data detection. To account for motion-induced Doppler frequency shifting, which can amount to more than a full carrier spacing, front-end resampling is often necessary. A simple method for estimating the needed resampling rate is to measure the time between two synchronization preambles that frame several OFDM blocks and compare it to the expected frame duration [55]. Since the Doppler factor is relatively large to begin with (e.g., on the order of 10^{-3} for a relative velocity of 1.5 m/s) Doppler shifting that remains after initial resampling cannot be neglected.

Channel estimation for OFDM systems has been addressed in different forms: in one, each OFDM block is processed independently of the other blocks, thus allowing for the possibility that the channel changes completely from one block to another [53, 56], while another form exploits correlation between adjacent blocks [52, 54], which makes it advantageous on slowly varying channels. Similarly as in single-carrier systems, accurate channel estimation is the key to successful data detection in OFDM, and it benefits greatly from proper channel modeling to reduce the number of unknown parameters that need to be estimated. Methods for identification of sparse systems, such as matching pursuit and basis pursuit, which improve upon traditional least squares estimation, were found to be beneficial and well-suited to channel estimation in acoustic OFDM. These methods have been applied to both block-individual channel estimation that uses pilot carriers only, and to block-adaptive, decision-directed channel estimation [60].

Time-variability of the channel can have an adverse effect on an OFDM system. If the symbol rate (number of carriers) is increased in a given bandwidth beyond the point at which the channel remains approximatley constant during one OFDM block, inter-carrier interference (ICI) will arise. ICI equalization then becomes necessary. The problem is analogous to that of ISI equalization in single-carrier systems, except that the equalizer now operates across carriers, and typically involves fewer interfering terms. However, unlike in single-carrier systems, the problem is avoidable simply by limiting the number of carriers. Methods based on one-shot linear equalization of a full block of carriers [53, 61], as well as recursive linear or decision-feedback equalization [62], were investigated. Further improvements are available from front-end (pre-FFT) filtering, which extracts the information about the time-varying channel before it has been lost in the process of FFT demodulation [63], [58].

The majority of acoustic OFDM systems addressed to-date have focused on coherent detection and the attendant issues of channel estimation and Doppler tracking. However, a properly designed OFDM system (one in which there is no ICI) is well suited to differentially coherent detection as well. Differential encoding is preferably applied across carriers, as frequency coherence is naturally satisfied with narrow carrier spacing which simultaneously supports bandwidth efficiency. Additional forward error correction coding can also be applied, and was mostly used in the form of low-density parity check (LDPC) codes. Experimental results [58] have demonstrated the benefits of differentially coherent OFDM, whose computational complexity much lower than that of any coherent system, and whose performance can surpass that of coherent detection when channel estimation fails.

Adaptive modulation

Adapting the transmitter to the channel characteristics has been considered in different forms, including active time-reversal [44] and single-mode excitation [65]. Adaptive modulation has been considered in the context of single-carrier MIMO systems [64], and, more recently, in the context of multi-carrier systems [57], where adaptive power and/or rate control (adaptive bit loading) can be implemented easily by adjusting the amplitude and/or the modulation level of each carrier separately. The performance improvement available from these techniques is contingent on the quality of the channel state information that is fed back to the transmitter. Recent results [57], which report on an experimental demonstration of this type, suggest the possibility to isolate the more slowly varying channel parameters (i.e. the predictable propagation path gains) from the more rapidly varying ones (phases) and use them to design an adaptive modulation system.

Future work

Existing results serve as the encouragement for future developments that will include not only point-to-point links, but multi-agent underwater networks, as well as fundamental questions of system capacity. In addition to bandwidth-efficient modulation/coding and signal processing techniques, future systems will rely on dedicated data compression algorithms, accurate statistical channel models, feedback-based techniques for optimal resource allocation, and system-level integration of communications, control, sensing and navigation functions.

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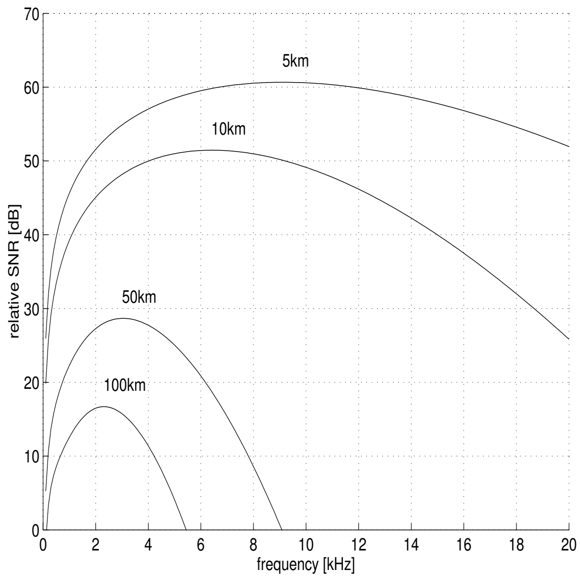

Figure 1: Frequency-dependent portion of SNR.

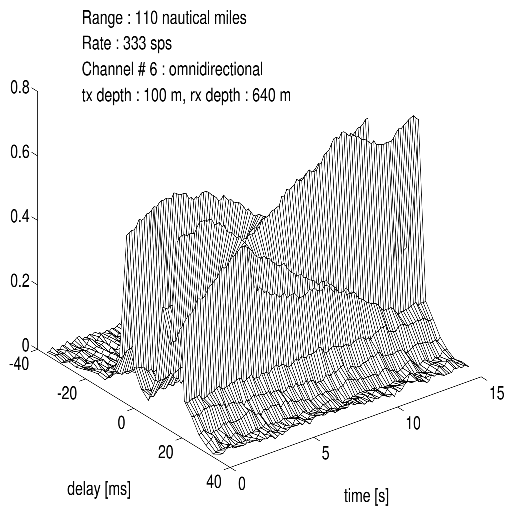

Figure 2: Ensemble of long-range channel responses in deep water (approx 2000 m) off the coast of California, during the month of January. Carrier frequency is 1 kHz. Rate at which quaternary data symbols used for channel estimation were transmitted is given in symbols per second (sps).

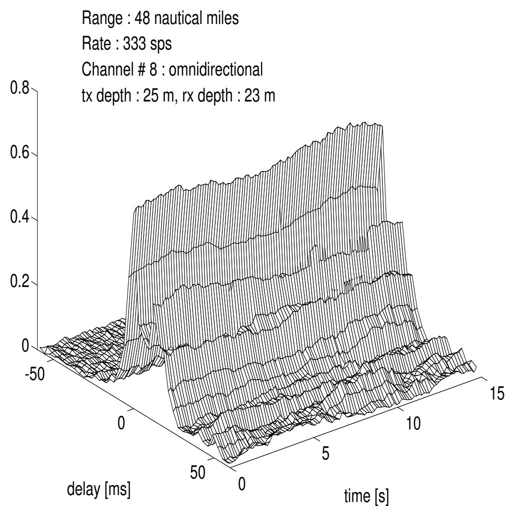

Figure 3: Ensemble of long-range channel responses in shallow water (approx 50 m) off the coast of New England, during the month of May. Carrier frequency is 1 kHz.

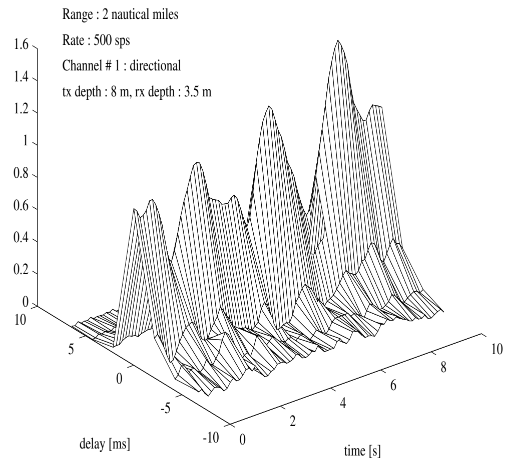

Figure 4: Ensemble of medium-range channel responses in shallow water (approx 20 m) near the coast of New England, during the month of February. Carrier frequency is 15 kHz.

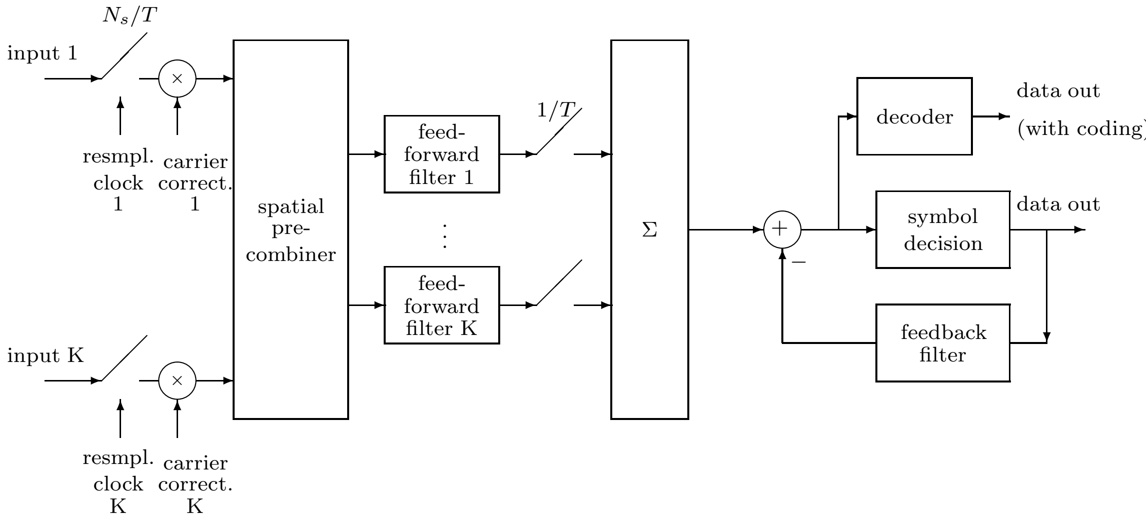

Figure 5: A multichannel receiver for phase-coherent detection.

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