UNDERWATER ACOUSTIC COMMUNICATION

The need for underwater wireless communications exists in such applications as remote control in the off-shore oil industry; pollution monitoring in environmental systems; collection and telemetry of scientific data recorded at ocean-bottom stations and by underwater vehicles; speech transmission for divers; mapping of the ocean floor for detection of objects and discovery of new resources, and military situations involving submarines and autonomous vehicles. 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 (1–6). Despite these limitations, underwater acoustic communications are a rapidly growing field of research and engineering.

Acoustic waves are not the only means for underwater wireless communication, but they are the best known so far. Radio waves that will propagate any distance through conductive sea water are the extra low-frequency ones (30 to 300 Hz) that require large antennas and high transmitter powers. Optical waves do not suffer as much from attenuation, but they are severely affected by scattering and absorption. Transmission of optical signals requires high precision in pointing the narrow laser beams, which are still being perfected for practical use. Hence, in applications where tethering is not acceptable, acoustic waves remain the single best solution for communicating underwater.

The idea of sending and receiving information underwater dates all the way back to the time of Leonardo Da Vinci, who is credited with discovering the possibility of detecting 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 World War II for military purposes. One of the first underwater communication systems was an underwater telephone, which was developed in 1945 in the United States for communicating with submarines (3). This device used a single-sideband (SSB) suppressed carrier amplitude modulation in the frequency range of 8 to 11 kHz; it was capable of sending acoustic signals over distances of several kilometers. Low rate acoustic communications based on binary frequency shift keying have been in use since the early 1970s for controlling acoustic releases. Acoustic tomography signals have also been used for many years in transmissions over horizontal distances of several thousand kilometers (26). However, not until the development of very large-scale integration (VLSI) technology did 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 few years, significant advancements have been made in the development of underwater acoustic communication systems with respect to their operational range and data throughput (6). Acoustically controlled robots have been designed to replace divers in performing maintenance of submerged platforms (7); high-quality video transmission from the bottom of deepest ocean trenches (6500 m) to a surface ship was established (8); and acoustic telemetry over horizontal distances in excess of 200 km was demonstrated (9).

As efficient communication systems are developing, the scope of their applications continues to grow, as 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 uncrewed, underwater vehicles (AUVs, UUVs). Freeing the underwater vehicles from cables will enable them to move unencumbered and refine their range of operation. While point-to-point acoustic communication systems remain the ones with most important applications, the emerging communication scenario of the future is that of an underwater data network consisting of both stationary and mobile nodes. 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.

With the aim of achieving these goals, current research is 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 considered for underwater acoustic networks, as well as the design of network protocols suited 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 processing methods (10) are expected to enable image transmission through band-limited underwater acoustic channels.

System Requirements

The achievable data throughputs and the reliability of an underwater acoustic communication system as measured by the bit-error rate vary from system to system. However, they are always subject to bandwidth limitations of the ocean channel. By contrast to the majority of other communication media, the use of underwater acoustic resources has not been regulated by standards.

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. Data rates of up to about one kilobit per second (kbps) are sufficient for these operations, but very low bit-error rates may be required (5).

Telemetry data are collected by submerged acoustic instruments such as hydrophones, seismometers, sonars, currentmeters, and chemical sensors; also included are low-rate image data. Data rates on the order of one to several tens of kilobits per second are required for these applications. The reliability requirements are not as stringent as those 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 between divers. While the existing commercially available diver communication systems mostly 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 because digital transmission is expected to provide better reliability. Transmission of digitized speech using linear predictive coding (LPC) methods requires rates on the order of several kilobits per second to achieve close-to-toll quality. The bit error rate tolerance of about 10^{-2} makes it a feasible technology for poor-quality band-limited underwater channels (11,12).

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 s (8). Further reduction of the required transmission rate seems to be possible by using dedicated compression algorithms, e.g., the discrete wavelet transform (10). There have been reports recently of compression ratios in excess of 100 : 1. On the other hand, underwater acoustic transmission of television-quality monochrome video would require compression ratios in excess of 1000 : 1. Hence, the required bit rates for video transmission are greater than ten kilobits per second, and possibly up to several hundreds of kilobits per second. Performance requirements are moderate, as images will have satisfactory quality at bit-error rates on the order of $10^{-3} - 10^{-4}$. Compression of speech and images places more stringent requirements on the bit-error rates than with uncoded signals.

CHANNEL CHARACTERISTICS

Sound propagation under water is determined primarily 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 (SNR). Time-varying multipath influences signal design and processing in a communication system, thus determining the information throughput and the 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 and depth, the absorption loss increases not only with range but also with frequency, thus setting the limit on the available bandwidth. In addition, source transducer bandwidth presents a key limitation on the system throughput.

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

Noise in the ocean consists of man-made noise and ambient noise. In the deep ocean, ambient noise dominates, while near shores and in the presence of shipping activity, man-made noise significantly increases the overall noise level. Unlike man-made noise, most of the ambient noise sources except biological sources (whales, shrimp, etc.) 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 20 dB/decade, in both shallow and deep water, over frequencies which are important to communication systems design.

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. (SNR is evaluated assuming spherical spreading, absorption according to Thorp's absorption coefficient from Ref. 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, i.e., the data throughput. 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 to 100 km, the bandwidth is limited to a few kilohertz (for a very long distance on the order of 1000 km, the available bandwidth falls to 100 Hz). A medium-range system operating over 1 to 10 km has a bandwidth on the order of several tens of kilohertz, while only at very short ranges below about 100 m, more than 100 kHz of bandwidth may be available.

Multipath

Within the 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. 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. How to use 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.

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 the horizontal underwater acoustic channels they increase to several tens of, or a hundred 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).

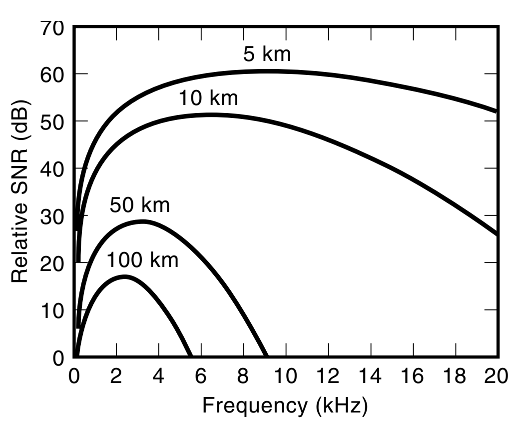

Figure 1. Frequency-dependent portion of SNR.

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. However, 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 200 m) and the region past the continental shelves (with depth up to several 1000 m). 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 toward 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 the bending of the rays toward the location where the sound speed reaches its minimum; this is 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. This channel is used for sound fixing and ranging, or SOFAR operations. Alternatively, the rays bending upwards may reach the surface, focusing in one point where they are reflected; the process is repeated periodically. The region between two focusing points is called a convergence zone, and its typical length is 60 to 100 km.

The geometry of multipath propagation and its spatial dependence are important for communication systems which use array processing to suppress multipath (13,14). 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 the basis for such propagation modeling.

Time-Variation

Associated with each of the deterministic propagation paths (macromultipaths), which can be modeled accurately, are random signal fluctuations (micromultipaths), 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 the 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 phase shifting by π. However, wind-driven waves act as the displacement of the reflection point, resulting in the signal dispersion. Equivalently, surface motion, and in particular longer waves and swell, cause amplitude variation or fading in the received signal. 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; this is significantly larger than that caused by many other phenomena. Doppler spread of a signal component of frequency f caused by a single surface-reflection occuring at an incidence angle θ is given by $0.0175(f/c)w^{3/2}\cos\theta$, where c is the speed of sound, nominally taken to be 1500 m/s, and w is the wind speed in meters per second (1). A moderate wind speed is on the order of 10 m/s. The highest Doppler spreads are most likely to be found in short- and medium-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, in general, can be expected to decrease with range.

As an example, Figs. 2 to 4 each show an ensemble of channel impulse responses, observed as functions of delay over an interval of time. These responses are estimated from experimental measurements (15). Relevant system parameters are indicated in the figures. These figures describe channel responses obtained at three fundamentally different locations with different mechanisms of multipath formation. The channel responses shown in Figs. 2 to 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. Figure 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 channel impulse response magnitude is shown as a function of delay. Each channel response reveals that two or more signals arrive at the receiver at any given time and the amplitudes and phases of distinct arrivals may vary independently in time.

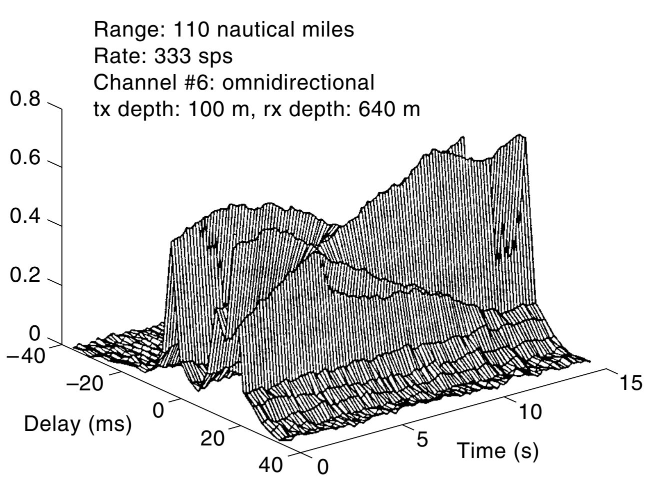

Figure 2. Ensemble of long-range channel responses in deep water (approximately 2000 m) off the coast of California, during the month of January. Carrier frequency is 1 kHz. Rate at which quaternary data symbols used for channel estimation were transmitted is given in symbols per second (sps).

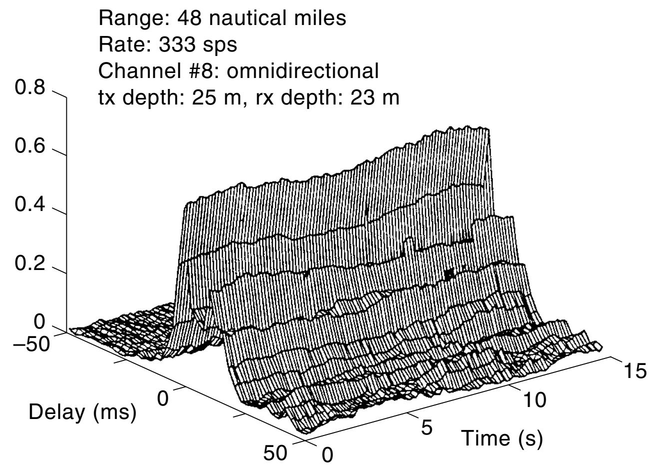

Figure 3. Ensemble of long-range channel responses in shallow water (approximately 50 m) off the coast of New England, during the month of May. Carrier frequency is 1 kHz.

The multipath delay-spread in this channel is on the order of 20 ms and the multiple arrivals have comparable energy, thus causing strong ISI. Along the time axis, variation of the channel response is observed for each given delay. In Fig. 2, 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. Figure 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 responses were recorded in the shallow water of Buzzards Bay near the coast of New England, over

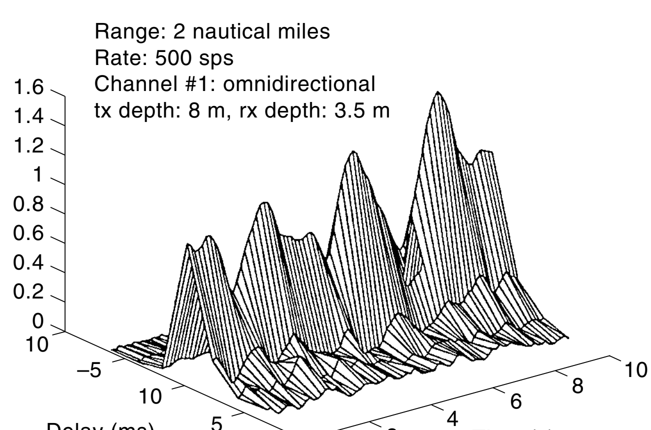

Figure 4. Ensemble of medium-range channel responses in shallow water (approximately 20 m) near the coast of New England, during the month of February. Carrier frequency is 15 kHz.

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 have values as high as 10^{-2}. While the Doppler spread describes time-variation of the channel response, multipath spread describes its time-dispersion. The implications of the resulting time-varying multipath dispersion on the communication system design are twofold. On 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, there is 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 (16) demonstrate these observations.

While there exists a vast knowledge of both deterministic and statistical modeling of sound propagation underwater, the implications of this knowledge on the communication channel modeling has only recently received more attention (17–20). A time-varying multipath communication channel is commonly modeled as a tapped delay line, with tap spacing equal to the reciprocal of twice the channel bandwidth, and the tap gains modeled as stochastic processes with certain distributions and power spectral densities. Although it is known that many radio channels fit well within the model of Rayleigh fading, where the tap gains are derived from complex Gaussian processes, to date there is no widely accepted single model for any of the underwater acoustic communication channels. Modeling of the shallow-water medium-range channel has received the most attention, as this channel is known to be among the most rapidly varying ones. Most researchers consider this channel to be fully saturated, meaning that it exhibits Rayleigh fading (2,4,17). The deep-water channel has also been modeled as a Rayleigh fading channel; however, the available measurements are scarce, often making channel modeling a controversial issue (18).

The channel measurements available today focus mostly on stationary communication scenarios. In a mobile underwater acoustic channel, vehicle speed will be the primary factor determining the time-coherence properties of the channel, and consequently the system design. Knowledge of a statistical channel model has proved to be useful in the design and analysis of land-mobile radio systems, and such models for underwater mobile acoustic channels await future development.

SYSTEM DESIGN

To overcome the difficulties of time-varying multipath dispersion, the design of commercially available underwater acoustic communication systems has relied so far mostly on the use of noncoherent modulation techniques and signaling methods which provide relatively low data throughput. Phase-coherent modulation techniques, which use equalization and array processing to compensate for the channel impairments, have only recently been shown to provide a feasible means for a more efficient use of the underwater acoustic channel bandwidth (6). These advancements are expected to result in a new generation of underwater communication systems, with at least an order of magnitude increase in raw data throughput.

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 the signal design, i.e., the choice of modulation/detection method, and the transmitter/receiver structure, i.e., the choice of array processing method and equalization method, if any.

This section describes the design of several systems which have been implemented. 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 following section.

Systems Based on Noncoherent Modulation

Noncoherent detection of frequency shift keying (FSK) signals has been used for channels exhibiting rapid phase variation, i.e., the shallow-water long- 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 vanishes before each subsequent pulse is to be received. The insertion of idle periods of time obviously results in a reduction of the available data throughput. In addition, due to the fact that fading is correlated among frequencies separated by less than 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.

A representative system (21) for telemetry at a maximum of 5 kbps uses a multiple FSK modulation technique in the 20 to 30 kHz band. This band is divided into 16 subbands, in each of which a 4-FSK signal is transmitted. Hence, from 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 been used successfully 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} to 10^{-3} were achieved without coding. The system performance may be improved by using error correction coding; however, its data throughput will be reduced. An acoustic modem based on multiple FSK is commercially available with a maximum data rate of 1200 bps. 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. As such, these methods are being further developed, and a system has recently been implemented (22) which uses orthogonal frequency division multiplexing

(OFDM) realized with DFT-based filter banks (discrete Fourier transform). This system was used on a medium-range channel; however, due to the high frequency separation among the channels (only every fourth channel is used) and relatively long guard times (10 ms guard following a 30 ms pulse), needed to compensate for the multipath fading distortion, the effective data rate is only 250 bps.

Systems Based on Differentially Coherent and Coherent Modulation

With the goal of increasing the bandwidth efficiency of an underwater acoustic communication system, research focus over the past years has shifted towards phase-coherent modulation techniques, such as phase-shift keying (PSK) and quadrature amplitude modulation (QAM). Phase-coherent communication methods, previously not considered feasible, were demonstrated 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 (9,16,23,24).

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; however, it has a performance loss compared to coherent detection. Most of the existing systems employ DPSK methods to overcome the problem of carrier phase extraction and tracking. Real-time systems have been implemented mostly for application 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 (7) operates 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.

Transmission over a very long range has been used for ocean acoustic tomography (25). With a carrier frequency of 200 Hz, and a 20 Hz bandwidth, information was transmitted in this system at less than 1 bps over 4000 km in deep water.

A deep ocean vertical path channel is used by an image transmission system (8). This is 4-DPSK system with carrier frequency of 20 kHz, capable of achieving 16 kbps bottom to surface transmission over 6500 m. The field tests of this system indicate the achievable bit-error rates on the order of 10^{-4} with linear equalizer operating under an LMS algorithm.

Another example of a successfully implemented system for vertical path transmission is that of an underwater image and data transmission system (26). This system uses a binary DPSK modulation at a rate of 19.2 kbps. The carrier frequency of 53 kHz was used for transmission over 2000 m.

Recent advances in digital underwater speech transmission are represented by a prototype system described in Ref. 11. This system uses a code-excited linear prediction (CELP) method to transmit the speech signal at 6 kbps. The modulation method used is 4-DPSK. A decision-feedback equalizer, operating under an LMS algorithm, is being used in the pool tests. Field tests have not been reported yet. A similar approach is considered in Ref. 12.

For the applications in a shallow-water medium-range channel, a binary DPSK system (27) uses a direct-sequence spread spectrum method to resolve a strong surface reflection observed in the 1 km long, 10 m deep channel. The interfering reflection is only rejected and not used for multipath recombining. Data throughput of 600 bps within a bandwidth of 10 kHz is achieved. Such high spreading ratios are justified in interference-suppression applications.

Current state-of-the art in phase-coherent underwater communications is represented by the system implementation (28). This 4-PSK system is based on purely phase-coherent modulation and detection principles (23). The signals are transmitted at 5 kbps using a carrier frequency of 15 kHz. The system's real-time operation in configuration as a six-node network was demonstrated in an under-ice shallow-water environment. To overcome the ISI caused by shallow-water multipath propagation, the system uses a decision-feedback equalizer operating under a recursive least squares (RLS) algorithm.

SIGNAL PROCESSING METHODS FOR MULTIPATH COMPENSATION

To achieve higher data rates, bandwidth-efficient 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 channel 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 (3). To overcome the need for a large array, the use of parametric sources has been studied extensively (13). 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 the rapid time-variation of the signal. Instead, a receiving array is employed to compensate for the possible errors. Binary and quaternary DPSK signals were used achieving data rates of 10 and 20 kbps, respectively, with a carrier frequency of 50 kHz. The estimated bit error rate was on the order 10^{-2} to 10^{-3}, depending on the actual channel length. In general, it was found that this technique is more effective at shorter ranges.

Multipath rejection using adaptive beamforming at the receiver end only is another possibility. The beamformer (14) uses an LMS algorithm to adaptively steer nulls in the direction of a surface-reflected wave. Similarly as in the case of the transmitter beamformer, 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 is of a decision-feedback type, and it 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. An estimated bit-error rate of 10^{-2} was observed without the equalizer and with the equalizer the bit-error rate was 10^{-3}.

A different method, based on purely phase-coherent detection, uses joint synchronization and equalization for combating the effect of phase variations and ISI (23,24). The equalization method is 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 highly time-spread underwater channels, showing superior 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. These methods are discussed in more detail in the following section.

Design Example: 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 (23) may be used. 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 channels (9,24). Spatial diversity combining has shown excellent performance in a number of underwater environments, as well as potentials for dealing with several types of interference. In Fig. 5, the multichannel equalizer is shown, preceded by an additional precombiner, which may or may not be used depending on the application and the number of available input 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 sequence, such as Barker code, transmitted in-phase and quadrature at the data rate). Baseband processing begins with downsampling, which may be carried out on as few as two samples per symbol interval ($N_s = 2$), because the signals are shaped at the transmitter to have a raised-cosine spectrum which limits their maximal frequency to less than $1/T$. Because 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 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 the channels are phase-shifted by the amount estimated in the process of joint equalization and synchronization. After coherent combining, the ISI resulting from the previously transmitted symbols (postcursors) is cancelled in the feedback section of the equalizer. This receiver structure is applicable to any linear modulation format, such as M-PSK, or M-QAM, the only difference being in the way in which symbol decision is performed. In addition to combining and equalization, signal pro-

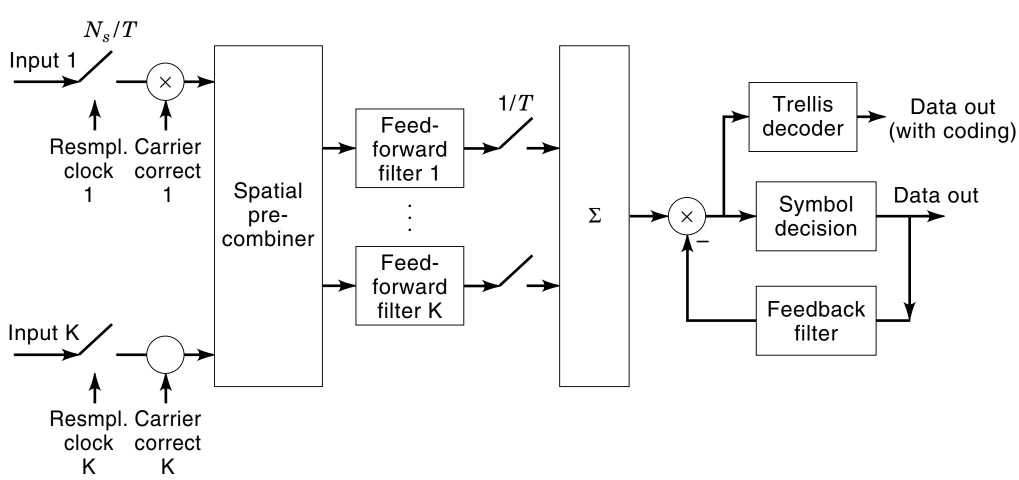

Figure 5. A multichannel equalizer for phase-coherent detection.

cessing at the receiver includes the operation of decoding if the signals at the transmitter were encoded. Trellis-coded modulation, compatible with PSK and QAM signals, is an effective means of improving performance on a band-limited channel (29). In addition to coded modulation, error correction coding may be employed.

The receiver parameters that are adaptively adjusted are the weights of the precombiner, 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 online 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 increase in the number of its input channels. For this reason, the spatial precombiner may be used to limit the number of equalizer channels, but still make use of the multichannel gain available from an array of sensors. The precombiner weights can be estimated jointly with the remaining adjustable parameters. The details of the joint adaptation are given in Ref. 24.

The multichannel receiver is adaptively adjusted to coherently combine 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 (14). The adaptive multichannel equalizer also has the capability to form spatial notches, but rather than cancelling all multipath, this receiver permits a range of solutions, judicious combination of which is determined implicitly through adaptation. Because it is not constrained by angular resolution, the method of multichannel equalization may be used with as few as two input channels. Also, it is applicable to a variety of underwater acoustic channels, regardless of the channel range-to-depth ratio. In applications where large arrays are available, the precombiner reduces the 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 to 2 km in rapidly varying shallow-water channels (6).

ACTIVE RESEARCH TOPICS

At this stage in the development of underwater acoustic communication techniques, with the feasibility of high rate communications established, numerous research topics are foreseen which will influence the development of future systems. Such topics include: reduced-complexity receiver structures and algorithms suitable for real-time implementation; techniques for interference suppression; multiuser underwater communications; system self-optimization; development of modulation/coding methods for improved bandwidth efficiency; and mobile underwater acoustic communication systems.

Reducing the Receiver Complexity

Although the underwater acoustic channels are generally confined to low data rates compared to many other communication channels, the encountered channel distortions require complex signal processing methods, resulting in high computational load which may exceed the capabilities of the available programmable DSP platforms. Consequently, reducing the receiver complexity to enable efficient real-time implementation has been a focus of active research.

The problem of reducing the receiver complexity may be addressed on two levels, i.e., the design of an efficient receiver structure, and the design of an efficient adaptive algorithm. For application in time-varying channels, the receiver, whether it is based on array processing, equalization, or both methods, must use an adaptive algorithm for adjusting its parameters. Two commonly used algorithms are based on the LMS and the RLS estimation principles.

In a majority of recent studies, the LMS-based algorithms are considered the only alternative for real-time implementations due to their low computational complexity, which is linear in the number of coefficients N (12,14,30). However, the LMS algorithm has a convergence time which may become unacceptably long when large adaptive filters are used (20 N as opposed to 2 N of the RLS algorithm). The total number of coefficients may be very large (more than 100 taps is often needed for spatial and temporal processing in medium- and long-range shallow-water channels). In addition, the LMS algorithm is very sensitive to the choice of step-size. To overcome this problem, self-optimized LMS algorithms may be used (30), but this results in increased complexity, and increased convergence time.

RLS algorithms, on the other hand, have better convergence properties but higher computational complexity. The quadratic complexity of the standard RLS algorithms is too high when large adaptive filters must be implemented. In general, it is desirable that the algorithm be of linear complexity, a property shared by the fast RLS algorithms. A numerically stable fast RLS algorithm (31) has been used for the multichannel equalizer (9). Despite its quadratic complexity, a square-root RLS algorithm (32) has been used for real-time implementation (33). The advantage of this algorithm is that it allows the receiver parameters to be updated only periodically rather than at every symbol interval, thus reducing the computational load per each detected symbol. In addition, the updating intervals can be determined adaptively based on monitoring the estimated mean squared error. Such adaptation methods are especially suitable for use with high transmission rates, where long ISI requires large adaptive filters, but at the same time eliminates the need to update the receiver parameters at every symbol interval which now has short duration. The square-root RLs algorithm has excellent numerical stability, which makes it preferable for a practical implementation. A different class of adaptive filters, which also have the desired convergence properties and numerical stability, are the lattice filters that may use either LMS or RLS algorithms. These algorithms have been proposed in Ref. 34, but have not yet been applied to underwater acoustic channel equalization. The selection an appropriate receiver adaptation method will receive more attention in the future acoustic modem design.

Regardless of the adaptive algorithm used, its computational complexity is proportional to the number of receiver parameters (tap-weights). Rather than focusing only on low-complexity algorithms, one may search for a way to reduce the receiver size. Although the use of array processing reduces residual ISI and allows shorter length equalizers to be used, a broadband combiner may still require a large number of taps to be updated, thus limiting the practical number of receiving channels to only a few. The use of a precombiner (26) is a method for reducing 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. Usually, more than one channel at the output of the combiner is 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.

A different approach in the design of reduced-complexity receiver structures is to reduce the number of equalizer taps. A conventional equalizer is designed to span the entire channel response. However, if the channel is characterized by several distinct multipath arrivals separated in time by intervals of negligible reverberation, an equalizer may be designed to have fewer taps. Such a method, termed sparse equalization, was applied to detection of experimental signals, and showed an order of magnitude reduction in computational load (35,36). By reducing the number of adaptively adjusted parameters, this approach also makes it possible to use simple updating algorithms, such as standard RLS algorithms, which have good numerical stability. Finally, in channels that are naturally sparse, discarding the low-magnitude equalizer taps results in improved performance because no unnecessary noise is processed.

Interference Cancellation and Multiuser Communications

The sources of interference in underwater acoustic channels include both external and internal interference generated within the system. The external sources of interference include noise coming from on-board machinery or other nearby acoustic sources. In a specific scenario of mobile communications, these sources will also include the propulsion and flow noise associated with the underwater vehicle launch process. The internal noise, which has signal-like characteristics, arises in the form of an 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 arises in the form of band-limited white noise and multiple sinusoids were investigated in Ref. 37). It was found that the multichannel receiver of Fig. 5 was most effective in canceling the interference while simultaneously detecting the desired signal. Noise cancellation is performed by providing a reference of the noise signal to one of the multichannel combiner inputs, which may be accomplished by the use of a reference hydrophone. Cancellation of the sinusoidal interferer may be performed even without the reference signal. By virtue of having the training sequence, the multichannel combiner is capable of adaptively filtering out the interfering signal and extracting the desired signal.

A multiple-access communication system represents a special case of a structured interference environment. Due to the bandwidth limitation of the underwater acoustic channel, frequency-division multiple-access may not be an efficient technique. Time-division multiple access is associated with the problem of efficient time-slot allocation, which arises because of the long propagation delays. A possible solution in such a situation is to allow a number of users to transmit simultaneously in both time and frequency. This approach resembles code-division multiple-access; however, only very low code-division processing gains are available due to bandwidth constraints. The receiver thus has to be designed to deal with the resulting multiple-access interference, which may be very strong in an underwater acoustic network. The fact that transmission loss varies significantly with range, and the fact that low spreading ratios are available, both contribute to the enhanced near-far effect in the underwater acoustic channel. The multiuser detection methods investigated for underwater acoustic channels rely on the principles of joint synchronization, channel equalization and multiple-access interference cancellation (38). Two categories of multiuser receivers that have been considered are the centralized receiver, in which the signals of all the users are detected simultaneously (e.g., uplink reception at a surface buoy which serves as a central network node), and the decentralized receiver, in which only the desired user's signal must be detected (e.g., downlink reception by an ocean-bottom node). Similarly, as in the case of interference cancellation, the adaptive multichannel receiver of Fig. 5 was shown experimentally to have excellent capabilities in the role of a decentralized multiuser detector, operating without any knowledge of the interfering signal. Array processing plays a crucial role in the detection of multiuser signals, but is associated with the problem of computational complexity.

System Self-Optimization

A receiver algorithm must use a number of parameters that must be adjusted according to the instantaneous channel conditions before the actual signal detection can begin. These parameters include the number and location of array sensors that provide good signal quality, the sizes of equalizer filters, and their tracking parameters. The optimal values of receiver parameters depend not only on the general link configuration but also on the time of operation. In addition, an increase in the background noise level caused, for example, by a passing ship, may temporarily disable the communication. If the adaptive receiver algorithms are to be used in autonomous systems, external assistance in algorithm initialization, or re-initialization, should be minimized. For this reason, the development of self-optimized receiver algorithms is of interest to future research.

The first steps in this direction are evident in the implementation of self-optimized LMS algorithms (14,30), in which the step-size is adaptively adjusted, and in the implementation of a periodically updated RLS algorithm (28), self-adjusted to keep a predetermined level of performance by increasing the tracking rate if the channel condition worsens. These strategies provide the receiver with the capability to adjust to the fine channel changes. However, they depend on the availability of a reliable reference of the desired signal. Because a training sequence is inserted only so often in the transmitted signal, a loss of synchronization or convergence during detection of a data packet will cause the entire packet to be lost. As an alternative to periodic reinsertion of known data, which increases the overhead, methods for self-optimized or blind recovery may be considered.

A blind equalization method based on using the cyclostationary properties of oversampled received signals (39), and which requires only the estimation of second-order signal statistics, provides a practical solution for recovering the data sequence in the absence of clock synchronization. Originally developed for linear equalizers, this method has been extended to the decision-feedback equalizer, necessary for application in underwater acoustic channels with extreme multipath. It has proved successful in preliminary tests with real data (6). Further work on blind system recovery for underwater acoustic channels will focus on methods for array processing and carrier-phase tracking.

Modulation and Coding

Achieving high throughputs over band-limited underwater acoustic channels is conditioned on the use of bandwidth-efficient modulation and coding techniques (29). Related results documented in contemporary research literature are confined to signaling schemes whose bandwidth efficiency is at most 3 to 4 bps/Hz. Higher-level signal constellations, together with trellis coding are being considered for use in underwater acoustic communications. While trellis-coded modulation is well suited for vertical channels which have minimal dispersion, their use on the horizontal channels requires further investigation. First, conventional signal mapping into a high-level PSK or QAM constellation may be associated with increased sensitivity of detection on a time-varying channel. Recent results in radio communications show that certain types of high-level constellations, which permit differential encoding in both phase and amplitude, are more robust to the channel fading and phase variations than the conventional rectangular QAM constellations (40). Another issue associated with the use of coded modulation on the channels with long ISI is the receiver design which takes full advantage of the available coding gain. Specifically, the delay in decoding poses problems for an adaptive equalizer that relies on the feedback of instantaneous decisions. The use of the maximum likelihood sequence estimation for both decoding and equalization is associated with the prohibitive complexity of the Viterbi algorithm for many applications. Receiver structures which deal with these problems as they apply to underwater channels are currently being studied.

In addition to bandwidth-efficient modulation and coding techniques, the future underwater communication systems will rely on data compression algorithms to achieve high data rates over severely band-limited underwater acoustic channels. This is another active area of research, which, together with sophisticated modulation and coding techniques, is expected to provide solutions for high-rate underwater image transmission.

Mobile Underwater Communications

The problem of channel variability, already present in applications with a stationary transmitter and receiver, becomes a major limitation for a mobile underwater acoustic communication system. The ratio of the vehicle speed to the speed of sound ($1/10^2$ for a vehicle speed of 30 knots or 54 km/h) many times exceeds its counterpart in the mobile radio channels ($1/10^8$ for a mobile moving at 60 miles per hour or 100 km/h), making the problem of time-synchronization very difficult in the underwater acoustic channel. Apart from the carrier phase and frequency offset, the mobile underwater acoustic systems will have to deal with motion-induced pulse compression and dilation. Successful missions of experimental AUVs that use commercial FSK acoustic modems for vehicle-to-vehicle communication have been reported (41). For phase-coherent systems, algorithms for continuous tracking of the time-varying symbol delay in the presence of underwater multipath are under development.

While many problems remain to be solved in the design of high-speed acoustic communication systems, recent advances in this area serve as an encouragement for future work. This will make possible the remote exploration of the underwater world.

More information about underwater acoustics can be found in the following entries: OCEANOGRAPHIC EQUIPMENT; SONAR SIGNAL PROCESSING; SONAR TARGET RECOGNITION; and SONAR TRACKING.

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UNDERWATER ROBOTICS.

See UNDERWATER VEHICLES.

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