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## Underwater Acoustic Communication

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The need for underwater wireless communications exists in applications such as remote control in off-shore oil industry, pollution monitoring in environmental systems, collection of scientific data recorded at ocean-bottom stations, speech transmission between divers, and mapping of the ocean floor for detection of objects, as well as for the discovery of new resources. Wireless underwater communications can be established by transmission of acoustic waves. Underwater acoustic communications are a rapidly growing field of research and engineering as the applications, which once were exclusively military, are extending into commercial fields. The possibility to maintain signal transmission, but eliminate physical connection of tethers, enables gathering of data from submerged instruments without human intervention, and unobstructed operation of unmanned or autonomous underwater vehicles (UUVs, AUVs).

Acoustic waves are not the only means for wireless transmission of signals under water. However, radio waves that will propagate any distance through conductive sea water are the extra low frequency ones (30 Hz-300 Hz) which require large antennae and high transmitter powers. Optical waves do not suffer so much from attenuation, but they are affected by scattering. Consequently, transmission of optical signals requires high precision in pointing the narrow laser beams. While the laser technology is still being perfected for practical use, acoustic waves remain the single best solution for communicating under water in applications where tethering is unacceptable.

The idea of sending and receiving information under water, 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, 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 8-11 kHz, and it was capable of sending acoustic signals over distances of several kilometers. However, it wasn't until the development of VLSI technology that a new generation of underwater acoustic communication systems began to emerge. With the availability of compact DSPs with their moderate power requirements, it began 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 [6], in terms of their operational range and the data throughput. Acoustically controlled robots have been used to replace divers in performing maintenance of submerged platforms [14]; high-quality video transmission from the bottom of deepest ocean trenches (6500 km) to the surface ship was established [15]; and data telemetry over horizontal distances in excess of 200 kilometers was demonstrated [23].

As efficient communication systems are developing, the scope of their applications continues to grow, and so do the requirements on the system throughput and performance. Many of the developing applications, both commercial and military, are now calling for real-time communication with submarines and autonomous underwater vehicles, not only in point-to-point links, but also in network configurations. 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. This network is envisaged to provide exchange of data, such as control, telemetry and eventually video signals, between many network nodes. The network nodes, both stationary and mobile ones, located on underwater vehicles and robots, will be equipped with various 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 is focusing on the development of efficient communications and signal processing algorithms, design of efficient modulation and coding schemes, multiple access methods, and techniques for mobile underwater communications. In the related field of communication networking, the design of protocols, suited for long propagation delays and strict power requirements encountered in the underwater environment is being addressed. Finally, in the field of image processing, the development of data compression algorithms suitable for low-contrast underwater images will enable image transmission through band-limited underwater acoustic channels [7].

## 0.1 System requirements

The achievable data throughputs, and the reliability of the underwater acoustic communication system, as measured by the bit-error probability, 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.

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 high reliability is required.

Telemetry data is collected by submerged acoustic instruments such as hydrophones, seismometers, sonars, or various other instruments which collect information about currents, tides, pollution, etc, and it also may include very 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 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, 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 [17, 18].

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  $256 \times 256$  pixel still images with 2 bits per pixel, at transmission rates of about one frame per 10 second [15]. Further reduction of the required transmission rate seems to be possible by using dedicated compression algorithms, e.g., the discrete wavelet transform [16]. On the other hand, underwater acoustic transmission of television-quality monochrome video would require compression ratios in excess of 1000:1. Hence, the bit rates required for video transmission vary between 10 kbps and 500 kbps. Performance requirements are moderate, as images will have satisfactory quality at bit error rates on the order of  $10^{-3} - 10^{-4}$ .

## 1 Channel characteristics

Sound propagation under water 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, often imposing severe limitations on the system performance.

### 1.1 Range, bandwidth and SNR

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 [1].

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 channel behavior as a waveguide, which results in various phenomena, including formation of the 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 exhibits strong frequency-dependence as well as site-dependence. Generally, the inshore environments, such as marine work-sites, are much noisier than deep ocean, due to the man-made noise. 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 20 dB/decade, both in shallow and deep water, over frequencies which are of interest to communication systems design.

Ambient noise, together with frequency-dependent transmission loss, determines 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. From the figure, the relationship between the available range and frequency band also becomes apparent. 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 a 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 multipath, horizontal channels may have extremely long multipath spreads. Most notable in the long- and medium-range channels, multipath propagation causes severe degradation of the acoustic communication signals. Combating the underwater multipath to achieve a high data throughput is without exception considered to be the most challenging task of an underwater acoustic communication system.

## 1.2 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 the 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).

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, respectively. Two fundamental mechanisms of multipath formation are reflection at the boundaries (bottom, surface and any objects in the water), and ray bending (sound speed is a function of depth, and the rays of sound always bend towards regions of lower propagation speed). If the water is shallow, such as in the littoral region and the region of continental shelves, 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. Since 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 -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., [20], [21]). The design of such systems is accompanied by the use of a propagation model for predicting the multipath configuration.

Ray theory and the theory of normal modes provide basis for such propagation modeling. Recent references commonly use ray tracing for determining the coarse multipath structure for communication channel modeling [8, 9]. A different class of underwater acoustic communication systems has recently been developed [23], which do not rely on the particular multipath geometry, and is equally applicable in a variety of channels, regardless of the parameters such as range-to-depth ratio which determine the angles of incidence of multipath arrivals.

### 1.3 Time-variation

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 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 only deformation being the phase shifting by  $\pi$ . However, waves act as the displacement of the reflection point, resulting in the dispersion of energy. Vertical displacement of the surface can be well modeled as a zero-mean Gaussian random variable, with power spectrum depending only on 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  $\theta$  is  $(0.0175/c)fw^{3/2}\cos\theta$  where  $c$  is the speed of sound, nominally taken to be 1500 m/s, and  $w$  is the wind speed in m/s [1]. This dependence is illustrated in Fig.2, in which it is assumed that  $\cos\theta \approx 1$ , as it will be the case for a great

majority of communication links in which the range is much greater than depth. Wind speed of 10 m/s corresponds to moderate winds. Doppler spreads as high as several tens of Hz 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. Nevertheless, the channel spread factor, i.e. the product of the Doppler spread and the multipath spread can be expected to decrease with range.

As an example, Figs.3-5 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. Relevant system parameters are indicated in the figures.

While there exists a vast knowledge of both deterministic and statistical modeling of sound propagation underwater, the implications this knowledge bears on the communication channel modeling has only recently received more attention (e.g., [8]-[11]). 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. While 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, there is no single model accepted to date for any of the underwater acoustic channels. Among the underwater acoustic channels, modeling of the shallow water medium-range channel has received most attention, as this channels is known to be among the most rapidly varying ones. Most authors consider that this channel is fully saturated, meaning that it exhibits Rayleigh fading [2], [4], [8]. The deep water channel has also been modeled as a Rayleigh fading channel; however, the available measurements are scarce. The question of statistical channel modeling is still a controversial one, as some authors find that both the deep and the shallow water channel fit better within a deterministic model [9].

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 the limiting value of  $10^{-2}$  for reliable coherent channel tracking. And indeed, many researchers still find that due to surface variability a reliable carrier phase reference cannot be obtained in the shallow

water channels [20], [26]. However, the implications time-varying multipath bears on the communication system design are twofold. On one hand, signaling at a high rate (short pulses) causes many adjacent symbols to interfere at the receiver, and requires sophisticated processing to compensate for the ISI. On the other hand, as pulse duration becomes shorter, channel variation over a single symbol interval becomes negligible. This allows an adaptive receiver to efficiently track the channel on a symbol-to-symbol basis, provided, of course, a method for dealing with the resulting time-dispersion. Hence, the time-varying multipath causes a trade-off in the choice of signaling rate for a given channel. Experimental results obtained on a shallow water medium-range channel [25] demonstrate these observations.

The statistical 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 proven to be useful in the design and analysis of land-mobile radio systems; however, it remains for the future to develop such models for underwater mobile acoustic channels.

## 2 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. Recently, phase-coherent modulation techniques, together with array processing for exploitation of spatial multipath diversity, have been shown to provide a feasible means for a more efficient use of the underwater acoustic channel bandwidth. These advancements are expected to result in a new generation of underwater communication systems, with at least an order of magnitude increase in data throughput.

Approaches to system design vary according to the technique used for overcoming the effects of multipath and phase variations. Specifically, these techniques may be classified according to (1) the signal design, i.e. the choice of modulation/detection method, and (2) the transmitter/receiver structure, i.e. the choice of array processing method and the

equalization method, if any. In this 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 following section.

## 2.1 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- 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, 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, separated by more than this amount 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 [12] for telemetry at a maximum of 5 kbps uses a multiple FSK modulation technique in the 20 kHz-30 kHz band. This band is divided into 16 subbands, in each of which a 4-FSK signal is transmitted. Hence, out of a total of 64 channels, 16 are used simultaneously for parallel transmission of 32 information bits (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 700 m shallow water path, where probabilities of bit error on the order of  $10^{-2} - 10^{-3}$  were achieved without coding. The system performance may be improved by using error correction coding; however, its data rate will be reduced. The multiple FSK system has been implemented [12], and is commercially available with a maximum data rate of 1200 bps. Despite the fact that these systems have bandwidth efficiency which 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 [13] which uses orthogonal frequency division multiplexing (OFDM) realized with DFT-based filter banks. This system was used on a medium-range channel; however, due to the high frequency separation among the channels (only every fourth channel is used) and relatively long guard times (10 ms guard following a 30 ms pulse), needed to compensate for the multipath fading distortion, the effective data rate is only 250 bps.

## 2.2 Systems based on 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 PSK (phase shift keying) and QAM (quadrature amplitude modulation). 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 [22]-[24]. The new generation of underwater acoustic communication systems, based on the principles of phase-coherent detection techniques, is capable of achieving raw data throughputs that are an order of magnitude higher than those of the existing noncoherent systems.

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. Its disadvantage is performance loss as compared to coherent detection. While bandwidth-efficient methods have successfully been tested on a variety of channels, real-time systems have mostly been implemented for application in vertical and the very short range channels, where little multipath is observed and the phase stability is good.

In the very short range channel, where bandwidth in excess of 100 kHz is available, a representative system [14] 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. A 16-QAM format is used, and the performance is aided by an adaptive equalizer. A linear equalizer (LE), operating under an least mean

squares (LMS) algorithm suffices to reduce the bit error rate from  $10^{-4}$  to  $10^{-7}$  on this channel.

The current state-of-the-art in coherently modulated systems which use the deep ocean vertical path is represented by an image transmission system [15] developed in Japan. It is a differentially coherent, 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 developed in France [26]. This system uses a binary DPSK modulation at a data 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 [17]. This system uses a code excited linear prediction (CELP) method to transmit the speech signal at 6 kbps. The modulation method used is quadrature DPSK. A decision-feedback equalizer, operating under LMS algorithm is being used in the pool tests. Field tests have not been reported yet. A similar approach is considered in [18].

For the applications in shallow water medium-range channel, a DPSK system [19] 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 in a spreading bandwidth of 10 kHz is achieved. Such high spreading ratios are justified in interference-suppression applications.

Current state-of-the art in phase-coherent communication systems is represented by the prototype system implemented at the Woods Hole Oceanographic Institution in the U.S.A. [27]. This system is based on purely phase-coherent modulation and detection principles given in [22]. The modulation format is QPSK, and 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 the under-ice shallow water environment. The ISI caused by the shallow water multipath is processed by a decision-feedback equalizer operating under an RLS (recursive least squares) algorithm.

### 3 Signal processing methods for multipath compensation

To achieve higher data rates, the more sophisticated 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 ISI. Three main approaches have recently 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. An approach pursued at the University of Birmingham, U.K. [20] focuses on the use of transmitter arrays to excite only a single path of propagation. Long arrays and careful positioning are required to ensure complete absence of multipath, since no equalization is used in this system. Equalization is not deemed feasible in this work, since it is considered that the shallow water test channel exhibits Rayleigh fading which is too rapid for an adaptive equalizer to follow. Instead, a receiving array is employed to compensate for the possible pointing errors. Binary and quaternary DPSK signals were used achieving data rates of 10 kbps and 20 kbps, respectively. The estimated bit error rate (BER) was on the order  $10^{-2} - 10^{-3}$ , depending on the actual channel length. In general, it was found that the technique proposed is more effective at shorter ranges.

An approach for multipath rejection using adaptive beamforming at the receiver end is being investigated at the University of Newcastle, U.K. [21]-. The prototype beamformer uses an LMS type of algorithm to adaptively steer nulls in the direction of a surface reflected wave. Similarly as in the case of the transmitter array, it was found that the beamformer encounters difficulties as the range increases relative to depth. To compensate for this effect, the use of an equalizer was proposed to complement the performance of the beamformer [21]. The equalizer is of a decision-feedback type, and it operates under the LMS algorithm whose low computational complexity permits real-time adaptation at the symbol rate. A separate

waveform is transmitted at twice the data rate for purposes of time-synchronization. The system was tested in shallow water at 10 kbps, and showed the estimated BER of  $10^{-2}$  without, and  $10^{-3}$  with the equalizer.

The final approach we shall describe is based on the use of purely phase-coherent detection methods, which have been developed at Northeastern University and the Woods Hole Oceanographic Institution, Massachusetts, U.S.A [22]-[24] for application in underwater acoustic channels with severe multipath. These signal processing methods are based on joint synchronization and equalization for combating the effect of phase variations and ISI. The method of fractionally spaced decision-feedback equalization is used with an RLS algorithm. It also incorporates spatial signal processing based on diversity combining, which will be discussed in more detail below. The phase-coherent methods have been tested in various underwater environments. 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.

### **3.1 Design example: multisensor 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 [22, 33]. The fact that the strongest multipath component may not be well defined makes the extraction of carrier reference a difficult task in the underwater channel. To establish coherent detection in the presence of strong multipath, a technique based on simultaneous synchronization and multipath compensation has been presented in [22]. This technique is based on joint estimation of the carrier phase and the parameters of a decision-feedback equalizer (DFE), where the optimization criterion is minimization of the mean-squared error (MSE) in the data estimation process. In [23], the equalizer/synchronizer structure has been extended to include a number of input array channels. This implementation with spatial diversity combining has shown superior performance in a variety of channels, as well as

potentials for dealing with several types of interference. In Fig.6, 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, and frame-synchronized using a known channel probe (usually a short 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 ( $T_s = T/2$ ), since the signals are shaped at the transmitter using raised-cosine spectrum shaping which limits their bandwidth to less than  $1/T$ . 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 channels are phase-shifted by the amount estimated in the process of joint equalization and synchronization. After coherent combining, the ISI resulting from the previously transmitted symbols (postcursors) is cancelled in the feedback section of the equalizer. This receiver structure is applicable to any linear modulation format, such as M-PSK, or M-QAM, the only difference being in the way in which symbol decision is performed. In addition to combining and equalization, signal processing at the receiver includes the operation of decoding if the signal at the transmitter was encoded. Trellis coded modulation, compatible with PSK and QAM signals, provides an effective means of improving performance on a band-limited channel [36]. In addition to coded modulation, error correction coding may be employed.

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 the parameters. This error is the difference between the estimated data symbol which is input to the decision device, and its true value. During the initial training mode, the true data symbols are known. After the training period, when the receiver parameters have converged, the on-line symbol decisions are fed back to the equalizer and used to compute the error. The adaptive

algorithm which has proven to be effective in different underwater acoustic environments is a combination of the second-order digital phase-locked loop (PLL) for the carrier phase estimates, and the RLS algorithm for the multichannel equalizer tap weights. The 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 [24].

The receiver is adaptively adjusted to coherently combine the multiple signal arrivals, and thus exploit both the spatial and the 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 in this manner usually has to be processed by a separately optimized equalizer, needed to compensate for the residual ISI which arises because the beamformer cannot completely eliminate the interference represented by the broad-band replicas of a high-rate signal. 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 equally applicable to a variety of underwater acoustic channels, regardless of the channel's range-to-depth ratio. In applications where large arrays are available, the use of a pre-combiner reduces receiver complexity, while preserving the multichannel diversity gain.

The method of multichannel equalization was successfully used to process the signals transmitted over a variety of underwater channels. Experimental results include data rates of 2 kbps over three convergence zones (110 nautical miles) in deep water; 2 kbps over 50 nautical miles in shallow water, and 40 kbps over 1 nautical mile in a rapidly varying shallow water channel.

## **4 Active research topics**

At this stage in the development of underwater acoustic communication techniques, with the feasibility of high rate communications established, a number of research topics are foreseen which will influence the development of future systems. These 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.

## 4.1 Reducing the receiver complexity

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

The problem of reducing the receiver complexity may be addressed on two levels: the algorithmic and the structural level. For application in time-varying channels, the receiver, whether it is based on multichannel combining, equalization, or both methods, must use an adaptive algorithm for adjusting its parameters. Two commonly used types of algorithms are based on the LMS and the RLS estimation principles.

In a majority of recent studies, the LMS-based algorithms are considered an only alternative due to their low computational complexity (linear in the number of coefficients  $N$ ) [18],[21], [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  $N$  is usually large in shallow water medium- and long-range applications (more than 100 taps is often needed for spatial and temporal processing). In addition, the standard LMS is very sensitive to the choice of the step-size. To overcome this problem, 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 complexity. The quadratic complexity of the standard RLS algorithm is too high when large adaptive filters need to be implemented. In general, it is desirable that the algorithm be of linear complexity (number of computations per iteration proportional to the number of parameters), a property shared by the fast RLS algorithms. A numerically stable fast RLS

algorithm based on [28] has been used for off-line signal processing [22]-[24], while the real-time implementation [27] uses a square-root RLS algorithm of [29]. Despite its quadratic complexity, the advantage of this algorithm is that it allows the receiver parameters to be updated only periodically, rather than every symbol interval, thus reducing the computational load per each detected symbol. In addition, the updating intervals can be determined adaptively, based on monitoring the mean squared error. Such adaptation methods are especially suited for use with high transmission rates. At high symbol rates, the long ISI requires large adaptive filters, therefore increasing the computational complexity. However, the time interval between two transmitted symbols decreases, eliminating the need to update the receiver parameters every symbol interval. Also, the square-root RLS algorithm has excellent numerical stability, which makes it a preferable choice in many applications. A different class of adaptive filters, which also have the desired convergence and numerical stability properties, are the lattice filters which use RLS algorithms. These algorithms have been proposed in [31], but have not yet been applied to underwater acoustic channel equalization. Choosing an appropriate receiver adaptation method will receive more attention in the future acoustic modem design.

Regardless of the adaptive algorithm used, its computational complexity is proportional to the number of receiver parameters (tap-weights). Rather than focusing on low-complexity algorithms only, one may search for a way to reduce the receiver size. Although the use of spatial combining reduces residual ISI and allows shorter length equalizers to be used, a broadband combiner may still require a large number of taps to be updated, limiting the practical number of receiving channels to only a few. In [24] a method is presented which allows multichannel equalization of a large number of input channels. This is accomplished by using the multichannel equalizer of Fig.6 in which the spatial pre-combiner reduces a large number of input channels to a smaller number for subsequent multichannel equalization. The parameters of the spatial pre-combiner and those of the multichannel equalizer are optimized jointly to minimize the MSE in the data symbol estimate. More than one channel at the output of the combiner is usually required to achieve a substantial processing gain; however, the reduced number of channels at which the full processing gain may be achieved is often low (e.g., three), leading to an interesting conclusion that reduction in complexity may be

achieved at no cost in performance. This is explained by the fact that the multipath structure is not independent among the array sensors. In addition to the reduced computational complexity, smaller adaptive filters result in less noise enhancement, contributing to improved performance.

A different approach in the design of reduced-complexity receivers has been investigated in [32], where the focus is made on reducing the number of equalizer taps. A conventional equalizer is designed to span all of the channel response. However, if the channel is characterized by several distinct multipath arrivals separated in time by intervals of no reverberation, an equalizer may be designed to have fewer taps, corresponding only to the significant portions of the channel response. Such method, termed sparse equalization, was applied to detection of QPSK signals recorded in the Arctic waters. Results reported in [32] show an order of magnitude reduction in computational load. 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 which are naturally sparse, discarding the low-magnitude equalizer taps in fact results in improved performance since no unnecessary noise is processed.

## **4.2 Interference cancellation and multiuser communications**

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

Recent advances in noise cancellation techniques for underwater acoustic communication systems have been presented in [33]. In this study the performance of several methods for cancellation of band-limited white noise and multiple sinusoidal interference were investigated. It was found that the same receiver structure as that of Fig.6 was most effective in cancelling the interference and noise while simultaneously detecting the desired signal. Noise cancellation is performed simply by feeding a reference of the noise signal to one of the mul-

tichannel combiner inputs, while cancellation 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 notch interfering signal out, and extract the desired signal.

A multiple-access communication system represents a special case of 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. The receiver then 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 that only very low code-division processing gains are available due to bandwidth constraints, both contribute to the enhanced near-far effect in the underwater acoustic channel. The multiuser detection methods suitable for underwater acoustic channels rely on the principles of joint synchronization, channel equalization and multiple-access interference cancellation [34]. Two categories of receivers considered are centralized, in which the signals of all the users are detected simultaneously (e.g., up-link reception at a surface buoy which serves as a central network node), and decentralized, in which only the desired user's signal needs to be detected (e.g., down-link reception by an ocean-bottom node). Similarly as in the case of interference cancellation, the adaptive multichannel receiver of Fig.6 was shown to have excellent capabilities in the role of a decentralized multiuser detector, operating without any knowledge of the interfering signal. Multiuser detection techniques have been shown effective in tests with experimental data. Array processing plays a crucial role in the detection of multiuser signals, but is also associated with the problem of computational complexity.

### **4.3 System self-optimization**

A receiver algorithm must use a number of parameters which need to be adjusted according to the instantaneous channel conditions prior to turning the receiver on. These parameters

include the number and location of array sensors which provide good signal quality, the sizes of the equalizer filters, and their tracking parameters. Their optimal value depends not only on the general link configuration and location, but also on the time of operation. In addition, an increase the background noise level, caused for example by a passing ship, temporarily disable the communication. If the adaptive receiver algorithms are to be used in autonomous systems, external assistance in algorithm parameter selection, receiver initialization, training and synchronization should be minimized. It is for this reason that self-optimized receiver algorithms should be addressed in the future research.

The first steps in this direction are evident in the implementation of the self-optimized LMS algorithms for an adaptive beamformer [21],[30], which eliminates external tuning of the step-size for varying channel conditions, and the periodically updated RLS [27], 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. Since a training sequence is inserted only so often in the transmitted signal, if the receiver loses convergence during detection of a data packet, the entire packet will be lost. An alternative to periodic re-insertion of known data, which increases the overhead, methods for self-optimized or blind recovery may be considered.

A blind equalization method based on using the cyclostationary properties of oversampled received signals has recently been developed [35], which requires only the estimation of second-order signal statistics to recover the data sequence in the absence of clock synchronization. Originally developed for linear equalizers, this method has been extended to the case of decision-feedback equalizer, necessary for application in underwater acoustic channels with extreme multipath. These methods have proven 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.

## 4.4 Modulation and coding

Achieving high throughputs over band-limited underwater acoustic channels is conditioned on the use of bandwidth-efficient modulation and coding techniques [36]. The results doc-

umented in contemporary 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 the vertical channels which have minimal dispersion, their use on the horizontal channels requires further investigation. In the first place, conventional signal mapping into a high-level PSK or QAM constellation may be associated with increased sensitivity of detection on a time-varying channel. Recent results in radio communications show that certain types of high-level constellations are more robust to the channel fading and phase variations than the conventional rectangular QAM constellations [37]. The design of codes to accompany these modulations is yet to be developed. Another issue associated with the use of coded modulation on the channels with long ISI is the receiver design which takes full advantage of the available coding gain. Namely, the delay in decoding poses problems for an adaptive equalizer which relied on the feedback of instantaneous decisions. Receiver structures which deal with this problem as it applies to underwater channels are a subject of current studies.

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, where current achievements [16] report on the development of algorithms capable of achieving compression ratios in excess of one hundred. Efficient data compression, together with sophisticated modulation and coding techniques, is expected to make it possible for the first time to transmit underwater images in real time.

## 4.5 Mobile underwater communications

The problem of channel variability, already present in applications with a stationary transmitter and receiver, becomes a major limitation for the mobile underwater acoustic communication system. The ratio of the vehicle speed to the speed of sound ( $1/10^3$  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 100 km/h), making the problem of time-synchronization very difficult in the underwater acoustic channel. Apart from the carrier phase offset, the

mobile underwater acoustic systems will have to deal with the motion-induced pulse compression/dilation. Latest results report on successful missions of experimental AUVs which use commercial M-FSK acoustic modems for vehicle-to-vehicle communication [38]. For phase-coherent systems, algorithms for continuous tracking the time-varying symbol delay in the presence of underwater multipath are currently 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, which will enable the capability to remotely explore the underwater world.

## References

- [1] L.Berkhovskikh and Y.Lysanov, *Fundamentals of Ocean Acoustics*, New York: Springer, 1982.
- [2] S.Flatte, Ed., *Sound Transmission Through a Fluctuating Ocean*, Cambridge, UK: Cambridge University Press, 1979.
- [3] A.Quazi and W.Konrad, "Underwater acoustic communications," *IEEE Comm. Magazine*, pp. 24-29, Mar. 1982.
- [4] J.Catipovic, "Performance limitations in underwater acoustic telemetry," *IEEE J. Oceanic Eng.*, vol. 15, pp. 205-216, July 1990.
- [5] A.Baggeroer, "Acoustic telemetry - an overview," *IEEE J. Oceanic Eng.*, vol. 9, pp. 229-235, Oct. 1984.
- [6] M.Stojanovic "Recent advances in high rate underwater acoustic communications," *IEEE J. Oceanic Eng.*, pp.125-136, Apr. 1996.
- [7] *IEEE J. Oceanic Eng.*, Special Issue on Image Processing for Oceanic Applications, vol.22, Jan. 1997.
- [8] R.H.Owen, B.V.Smith and R.F.W.Coates, "An experimental study of rough surface scattering and its effects on communication coherence," in Proc. *OCEANS'94*, pp. III.483-III.488, Brest, France, Sept. 1994.
- [9] A.Essebbar, G.Loubet and F.Vial, "Underwater Acoustic Channel Simulations for Communication," in Proc. *OCEANS'94*, pp. III.495-III.500, Brest, France, Sept. 1994.
- [10] A.Falahati, B.Woodward and S.Bateman, "Underwater acoustic channel models for 4800 b/s QPSK signals," *IEEE J. Oceanic Eng.*, vol.16, pp.12-20, Jan.1991.
- [11] C.Bjerrum-Niese, L.Bjorno, M.A.Pinto and B.Quellec, "A simulation tool for high data-rate acoustic communication in a shallow-water, time-varying channel," *IEEE J. Oceanic Eng.*, vol. 21, pp.143-149, Apr. 1996.

- [12] J.Catipovic, M.Deffenbaugh, L.Freitag and D.Frye, "An acoustic telemetry system for deep ocean mooring data acquisition and control," in *Proc. OCEANS'89*, pp. 887-892, Seattle, Washington, Oct. 1989.
- [13] S.Coatelan and A.Glavieux, "Design and test of a multicarrier transmission system on the shallow water acoustic channel," in *Proc. OCEANS'94*, pp. III.472-III.477, Brest, France, Sept. 1994.
- [14] A.Kaya and S.Yauchi, "An acoustic communication system for subsea robot," in *Proc. OCEANS'89*, pp. 765-770, Seattle, Washington, Oct. 1989.
- [15] M.Suzuki and T.Sasaki, "Digital acoustic image transmission system for deep sea research submersible," in *Proc. OCEANS'92*, pp. 567-570, Newport, RI, Oct. 1992.
- [16] D.Hoag and V.K.Ingle, "Underwater video compression using the wavelet transform," in *Proc. OCEANS'95*, San Diego, CA, Oct. 1995.
- [17] A.Goalic et. al, "Toward a digital acoustic underwater phone," in *Proc. OCEANS'94*, pp. III.489-III.494, Brest, France, Sept. 1994.
- [18] B.Woodward and H.Sari, "Digital underwater voice communications," *IEEE J. Oceanic Eng.*, vol. 21, pp.181-192, Apr. 1996.
- [19] J.Fischer et al., " A high rate, underwater acoustic data communications transceiver," in *Proc. OCEANS'92*, pp. 571-576, Newport, RI, Oct. 1992.
- [20] R.F.W.Coates, M.Zheng and L.Wang, "BASS 300 PARACOM: A "model" underwater parametric communication system," *IEEE J. Oceanic Eng.*, vol. 21, pp.225-232, Apr. 1996.
- [21] G.S.Howe et al., "Sub-sea remote communications utilising an adaptive receiving beamformer for multipath suppression," in *Proc. OCEANS'94*, pp. I.313-I.316, Brest, France, Sept. 1994.

- [22] M.Stojanovic, J.A.Catipovic and J.G.Proakis, "Phase coherent digital communications for underwater acoustic channels," *IEEE J. Oceanic Eng.*, vol. 19, pp. 100-111, Jan. 1994.
- [23] M.Stojanovic, J.A.Catipovic and J.G.Proakis, "Adaptive multichannel combining and equalization for underwater acoustic communications ," *Journal of the Acoustical Society of America*, vol. 94 (3), Pt. 1, pp. 1621-1631, Sept. 1993.
- [24] M.Stojanovic, J.A.Catipovic and J.G.Proakis, "Reduced-complexity multichannel processing of underwater acoustic communication signals," *Journal of the Acoustical Society of America*, vol. 98 (2), Pt. 1, pp.961-972, Aug. 1995.
- [25] M.Stojanovic, J.G.Proakis and J.A. Catipovic, "Performance of a high rate adaptive equalizer on a shallow water acoustic channel," *J. Acoust. Soc. Amer.*, vol. 100 (4), Pt. 1, pp. 2213-2219, Oct. 1996.
- [26] G.Ayela, M.Nicot and X.Lurton, "New innovative multimodulation acoustic communication system," in Proc. *OCEANS'94*, pp. I.292-I.295, Brest, France, Sept. 1994.
- [27] M.Johnson, D.Herold and J.Catipovic, "The design and performance of a compact underwater acoustic network node," in Proc. *OCEANS'94*, pp. III.467-471, Brest, France, Sept. 94.
- [28] D.Slock and T.Kailath, "Numerically stable fast transversal filters for recursive least squares adaptive filtering," *IEEE Trans. Sig. Proc.*, vol. SP-39, pp. 92-114, Jan. 1991.
- [29] F.Hsu, "Square root Kalman filtering for high-speed data received over fading dispersive HF channels," *IEEE Trans. Inform. Theory*, Vol. IT-28, pp. 753-763, Sept. 1982.
- [30] B.Geller, V.Capellano, J.-M.Brossier, A.Essebar and G.Jourdain, "Equalizer for video rate transmission in multipath underwater communications," *IEEE J.Oceanic Eng.*, vol. 21, pp.150-155, Apr. 1996.

- [31] F.Ling and J.G.Proakis, "Adaptive lattice decision-feedback equalizers—their performance and application to time-variant multipath channels," *IEEE Trans. Commun.*, vol. 33, pp. 348-356, Apr. 1985.
- [32] M.Kocic, D.P.Brady and M.Stojanovic, "Sparse equalization for real-time digital underwater acoustic communications," in Proc. *OCEANS'95*, San Diego, CA, Oct. 1995.
- [33] J.Catipovic, M.Johnson and D.Adams, "Noise cancelling performance of an adaptive receiver for underwater communications," in Proc. *1994 Symposium on AUV Technology*, pp. 171-178, Cambridge, MA, July 1994.
- [34] M.Stojanovic and Z.Zvonar, "Multichannel processing of broadband multiuser communication signals in shallow water acoustic channels," *IEEE J. Oceanic Eng.*, pp. 156-166, Apr. 1996.
- [35] L.Tong, G.Xu and T.Kailath, "Blind identification and equalization based on second-order statistics," *IEEE Trans. Inform. Theory*, vol. IT-40, pp. 340-349, Mar. 1994.
- [36] J.Proakis, "Coded modulation for digital communications over Rayleigh fading channels," *IEEE J. Oceanic Eng.* vol. 16, pp. 66-74, Jan. 1991.
- [37] W.T.Webb and R.Steele, "Variable rate QAM for mobile radio," *IEEE Trans. Commun.*, Vol. COM-43, pp. 2223-2230, July 1995.
- [38] S.Chappell et al., "Acoustic communication between two autonomous underwater vehicles," in Proc. *1994 Symposium on AUV Technology* , pp. 462-469, Cambridge, MA, 1994.

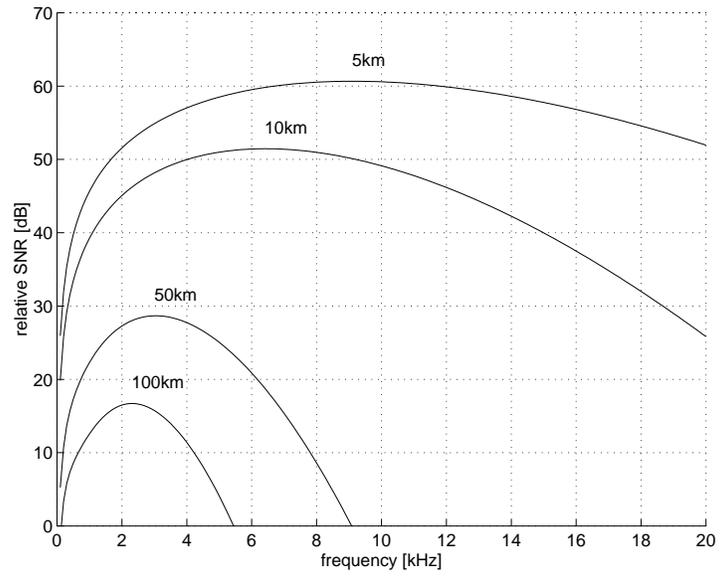


Figure 1: Frequency-dependent portion of SNR.

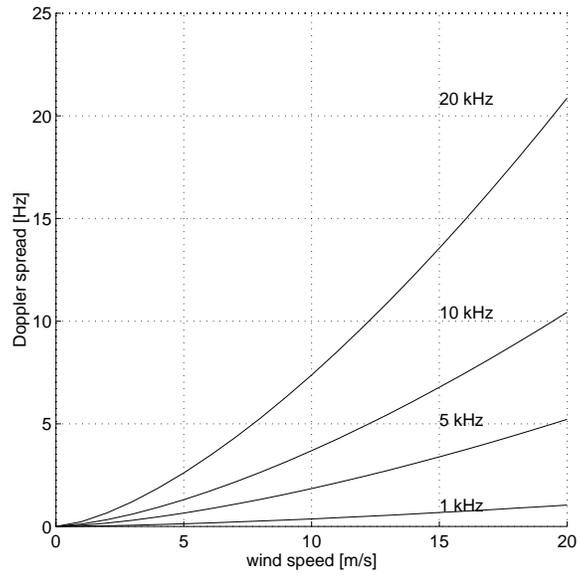


Figure 2: Doppler spread on a single surface-reflected path.

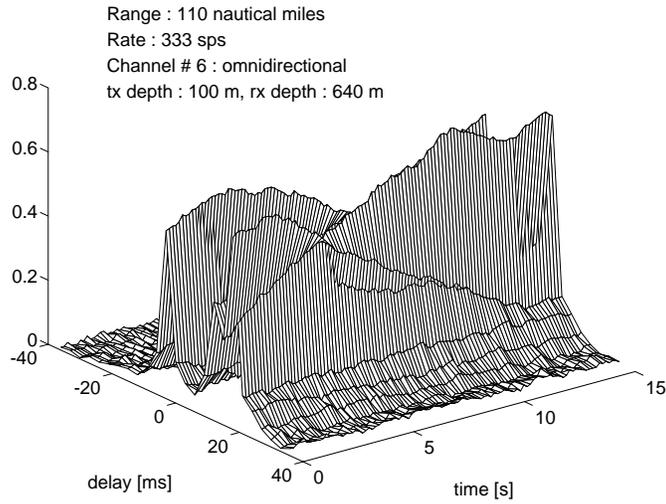


Figure 3: Ensemble of long-range channel responses in deep water (approx 2000 m) off the coast of California, during the month of January. Carrier frequency is 1 kHz.

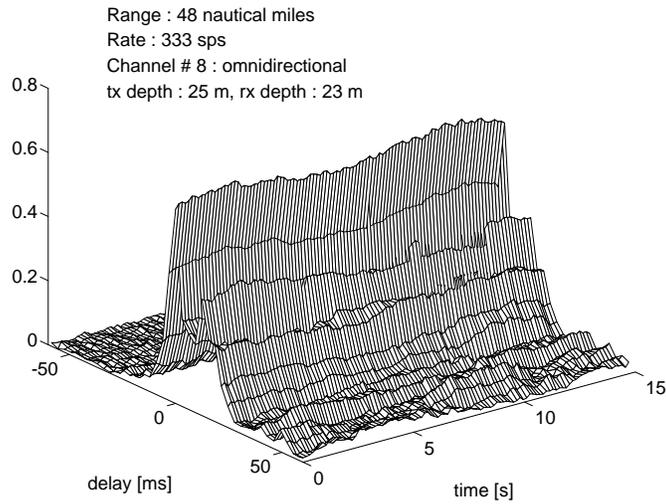


Figure 4: Ensemble of long-range channel responses in shallow water (approx 50 m) off the coast of New England, during the month of May. Carrier frequency is 1 kHz.

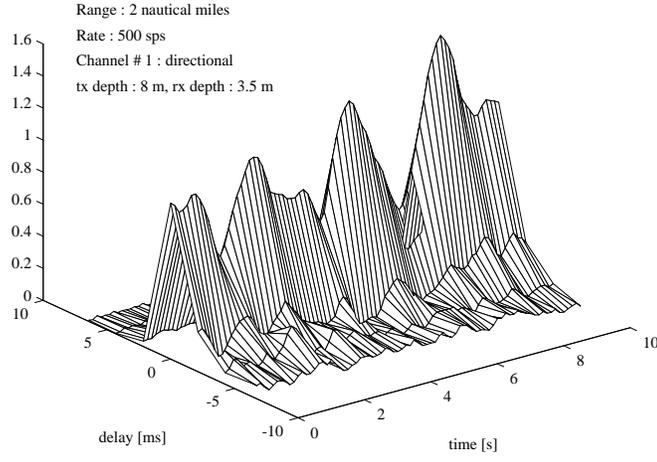


Figure 5: Ensemble of medium-range channel responses in shallow water (approx 20 m) off the coast of New England, during the month of February. Carrier frequency is 15 kHz.

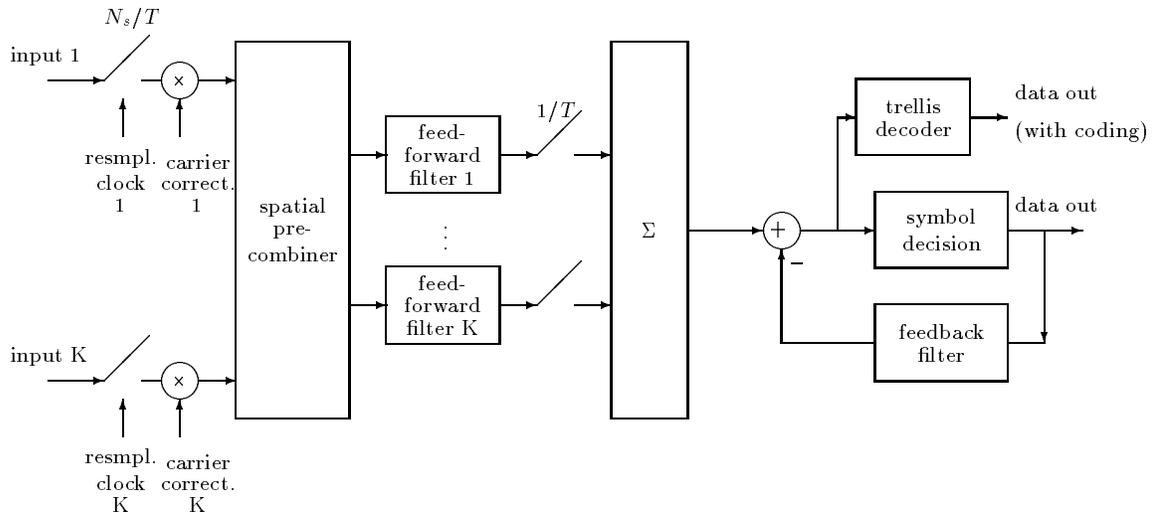


Figure 6: A multichannel receiver for phase-coherent detection.

# Contents

0.1	System requirements . . . . .	3
<b>1</b>	<b>Channel characteristics</b>	<b>4</b>
1.1	Range, bandwidth and SNR . . . . .	4
1.2	Multipath . . . . .	6
1.3	Time-variation . . . . .	7
<b>2</b>	<b>System design</b>	<b>9</b>
2.1	Systems based on noncoherent modulation . . . . .	10
2.2	Systems based on coherent modulation . . . . .	11
<b>3</b>	<b>Signal processing methods for multipath compensation</b>	<b>13</b>
3.1	Design example: multisensor signal processing for coherent detection . . . . .	14
<b>4</b>	<b>Active research topics</b>	<b>16</b>
4.1	Reducing the receiver complexity . . . . .	17
4.2	Interference cancellation and multiuser communications . . . . .	19
4.3	System self-optimization . . . . .	20
4.4	Modulation and coding . . . . .	21
4.5	Mobile underwater communications . . . . .	22

# List of Figures

1	Frequency-dependent portion of SNR. . . . .	28
2	Doppler spread on a single surface-reflected path. . . . .	28
3	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. . . . .	29

4	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. . . . .	29
5	Ensemble of medium-range channel responses in shallow water (approximately 20 m) off the coast of New England, during the month of February. Carrier frequency is 15 kHz. . . . .	30
6	A multichannel receiver for phase-coherent detection. . . . .	30