Multiuser Undersea Acoustic Communications in the Presence of Multipath Propagation

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Abstract— **Multipath propagation in underwater acoustic communications causes distortion to high-rate digital signals, but at the same time it provides diversity. Signals propagating from different locations to the same receiver undergo different channel effects, which provides a degree of separability. We exploit this fact the design of a multiuser communication system. Our goal is to isolate the signal of the desired user in the presence of multiuser interference without the use of typical direct-sequence code-division multiple-access methods. Instead of assigning different spreading codes to the users, the available bandwidth expansion is used for channel coding. The same code is used by every source. Powerful coding, coupled with broadband array processing is shown to provide sufficient separability for a small number of sources. The intended application is an underwater network of several sources such as AUVs. Experimental results obtained with four sources moving at speeds up to 2 m/s in shallow water show excellent receiver performance at per-source information rate of 1400 bps within a bandwidth of 5 kHz. A 1/7 rate concatenated code is employed, together with a four-element receiver array and soft-decision decoding based on multichannel decision-feedback equalization.**

I. Introduction

Multiuser communication methods for application to underwater acoustic systems are the subject of this work and an area of research that has gained much interest during the past several years. The interest in underwater communication networks has emerged with the development of reliable acoustic modems and unmanned underwater vehicles, which together open a range of possibilities for remote underwater exploration.

Underwater acoustic channel is broadcast by nature, thus requiring a multiple-access communication method to enable many sources to communicate without interference. Frequency-division multiple-access (FDMA), time-division multiple-access (TDMA) and code-division multiple-access (CDMA) are the most commonly used methods that allocate the channel resources to individual users in a deterministic manner. Another such method is space-division multiple-access (SDMA); however, this method has not been fully exploited in existing communication systems because of the lack of perfectly-pointed sources and receivers capable of tracking the mobile users.

The communication scenario of interest to this work involves a small number of underwater vehicles that communicate via a central node. FDMA, TDMA and CDMA are the primary candidates for such a situation. These techniques are known to provide the same system capacity (number of users supported within a given bandwidth) when compared on an ideal channel whose only disturbance is additive white Gaussian noise (AWGN) [1]. However, on time-varying, frequency-selective channels, CDMA is known to gain advantage over the other techniques through exploitation of multipath diversity by rake-type filtering of high-resolution signals. For these reasons, CDMA has been extensively used in mobile radio communications, and has come into the focus of several studies on underwater acoustic systems [3]-[7].

The benefits of CDMA, together with the recognition of channel distortions, have motivated our preliminary study on the use of spread-spectrum modulation for underwater acoustic communications [7]. Signaling methods based on frequency-hopping (FH) and direct-sequence (DS) spread-spectrum modulation were investigated for use in high-rate shallow water communications. Experimental results indicated that direct-sequence spread-spectrum is an advantageous technique to use in time-varying shallow-water channels, provided that there exists a method for reliable channel tracking.

The basic principle of CDMA technique is to assign orthogonal or nearly-orthogonal codes to different users. The code orthogonality is exploited at the receiver which despreads the signal, i.e. extracts the desired user's signal while suppressing the interference. Such a receiver is known as a conventional single-user receiver, and is designed for a channel whose only disturbances are additive noise and interference. The receiver performance is heavily dependent upon the orthogonality of the codes, and the system design is mostly concerned with the choice of codes with good cross-correlation properties. Examples of such codes are the Gold codes and the Kasami codes which are derived from PN sequences [2].

However, asynchronous reception and multipath propagation destroy the good cross-correlation properties of spreading codes. Both of these effects are present in underwater acoustic channels, where the conventional despreader alone does not suffice to suppress multiple-access interference. To account for these effects, as well as for the time-variability of the channels, adaptive signal processing methods must be used. Consequently, the focus of system design shifts from the spreading code selection [5] to adaptive filtering [4]. Especially beneficial to multiple-access interference suppression in the presence of multipath is array processing [3],[6]. It was shown in [3] that even with a moderate number of array elements, a decentralized adaptive receiver which has no knowledge of the spreading codes can achieve the performance of its centralized counterpart in processing multiuser signals with very low spreading gains. It is conjectured that such performance results from the spatial separation of the different users which

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induces a different channel response for each signal, allowing them to be separated despite the fact that the the spreading code orthogonality has been destroyed by multipath.

The capacity of a multiuser system is directly proportional to the available bandwidth expansion, i.e. the ratio between the total system bandwidth and the per-user information throughput. In a CDMA system design, the bandwidth expansion is allocated to spreading the spectrum of multiuser signals. However, it is known in the theory of communications that conventional direct-sequence spread-spectrum represents a special case of coding, and that the choice of PN-based spreading codes is a poor one, as better performance is available from more sophisticated coding schemes at same bandwidth expansion [1]. System design based on coding thus provides an efficient use of the limited bandwidth in a multiuser underwater acoustic system.

These results serve as a motivation for the present work, where we investigate the possibility to trade the spreading gain for a coding gain, and use a multichannel receiver to isolate the desired signal in the presence of multipath and multiuser interference. In Section II we present system design, which is based on the use of a coding method with a bandwidth expansion factor of 1/7 that is dedicated entirely to error correction coding. The receiver is a decentralized, single-user receiver which uses a multichannel decision-feedback equalizer (DFE). The decoder uses soft decisions from the DFE which, in turn, is updated using reliable decisions with a delay of one codeword. Experimental results obtained with up to four mobile users transmitting at 1400 bps within a total system bandwidth of 5 kHz in shallow water are presented in Section III Conclusions are summarized in Section IV

II. System design

A. Transmitter

Fig. 1 shows the block diagram of the transmitter.

The transmitter uses a concatenated code, composed of an outer Reed-Solomon code and an inner BCH code. Alternatively, it may use the BCH inner code only. The BCH codes are linear cyclic block codes which are frequently used in a variety of communication systems because of the large coding gain that they provide [1]. For the present application, the BCH (63,10) code was chosen. The bandwidth expansion of this code alone is $1/R_c$=63/10. Its minimum distance is d_{min}=27. Thus, the coding gain, $R_c d_{min}$, is 6.2 dB when only the inner code is used, and optimal, minimum-distance decoding is employed.

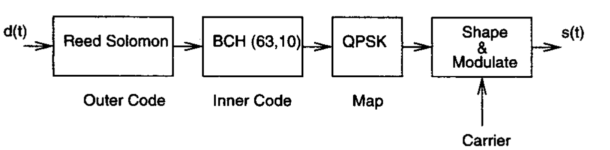

Fig. 1. Transmitter uses a concatenated coding method with a Reed-Solomon code as an outer code and a BCH code as an inner code. The rate of the Reed-Solomon code can be varied. The BCH code forms an output codeword of 63 bits for each 10 input bits. The coded bits are mapped into QPSK symbols which are used to modulate the carrier.

The BCH encoder takes input blocks of length 10 bits, which correspond to one letter of a non-binary alphabet containing a total of $2^{10} = 1024$ letters. To match this alphabet size, the length of the non-binary Reed-Solomon output codeword is chosen as 1023, while the length of the input codeword is chosen to provide a desired code rate. For the present application, a rate 9/10 Reed-Solomon code is used. Thus, the overall rate of the concatenated code is 1/7. The bandwidth expansion of this code is comparable to that of a PN-based spreading code which uses 7 chips per bit. However, as compared to a spreading code, which has no coding gain, this channel code has a coding gain of 6.2 dB or more (depending upon the outer code) but no spreading gain. Suppression of multiple-access interference is here effectively accomplished through diversity between multiple users' channels.

After encoding, the bits are mapped into QPSK symbols using Gray mapping. The transmitter pulse shape is rectangular, of duration T. The symbol transmission rate $1/T$ roughly equals the system bandwidth.

The transmission method of Fig. 1 is used by every communicating unit. There is no explicit assignment of spreading codes, which would provide orthogonality needed for single-user detection based on despreading. Instead, each user employs a different training sequence to adapt its equalizer to the transmission channel.

B. Receiver

The signals from various users arrive at the receiver asynchronously. Each signal is coarsely synchronized by matched filtering to its preamble, usually a frequency sweep. The signals are then processed by the receiver of Fig. 2.

The multichannel front-end processor consists of a bank of adaptive filters, i.e. feedforward equalizers that operate at twice the symbol rate, $2/T$. The filter outputs are combined, and carrier phase correction is performed at the symbol rate. The filters also incorporate adaptive resampling, which is necessary for mobile applications. Adaptive resampling is performed efficiently using polyphase filters. The resampling rate is adjusted adaptively in accordance with the carrier phase estimate. The front-end processor and the feedback filter are adaptively adjusted using an RLS or an LMS algo-

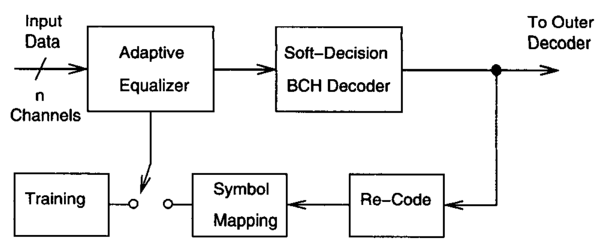

Fig. 2. The receiver employs an adaptive array and a DFE. The decoder is built into the feedback loop. The feedforward filters, the carrier phase estimate and the feedback filter are adjusted using an adaptive algorithm driven by a common error signal. The outer code employs hard-decision decoding.

rithm. Experience with real data shows a consistently better performance of the RLS algorithm with approximately 1dB-3dB gain [9].

Signaling at high bit rates (on the order of 10 kbps) in shallow water requires equalization of intersymbol interference which commonly spans tens of bit intervals. In order to implement an adaptive equalizer, reliable decisions are needed both for updating the filter coefficients and for feeding back to the DFE. In the presence of multiple-access interference, coding provides the required quality of bit decisions. However, bit decisions are available for use by the adaptive equalizer only after a decoding delay. Because the shallow water channel is rapidly time-varying, it is important that this delay be kept as small as possible. With the proposed block code and the DFE implementation, the decoding delay equals the duration of one codeword. If little change occurs in the optimal equalizer coefficients during this time, the delayed decisions can be efficiently employed. In other words, the outdated equalizer coefficients (those obtained with decoded bits of the previous codeword) suffice for processing the current data bits. Soft decisions are thus obtained on the bits of the current codeword, with equalizer coefficients that are up-to-date at the beginning of the codeword, and get progressively more outdated towards its end. When all the soft decisions of the current codeword become available, the equalizer taps can be updated using the actual error between the (delayed) received signal and the reliable decisions obtained by re-encoding the decoded bits. Processing of the next symbol now begins with equalizer taps that have been updated.

To verify the usefulness of this approach, receiver performance with experimental data is compared in two cases: when correct decisions are fed back without delay, and when true decisions are fed back, i.e. outdated equalizer coefficients are employed. Experimental results obtained with 5 kilosymbols per second (ksps) in-channel transmission rate show 0dB-0.5dB difference between the two cases, depending upon channel variability and the selected equalizer tracking rate. This loss is sufficiently small to justify the use of the proposed approach. Similar conclusions about the impact of time-variability were noted in [8], where a single-user system employing a Hamming (7,4) block code and a DFE has been experimentally analyzed for shallow water channels.

If the channel were varying more rapidly over the duration of one codeword, a different approach would have to be considered. One such approach is discussed in [4]. However, for the present system design, time-variation is slow enough to allow effective use of the coding gain.

III. Experimental results

A. Experiment description

The experiments were conducted in September 2000, in the Narraganset Bay near Newport, Rhode Island. Fig. 3 shows the site map and the trajectory of the transmitter. The transmitter was mounted on a small boat that moved at speeds ranging from zero to 2 m/s within a range of approximately 0.5 km. The receiver was mounted on a stationary platform on Gould

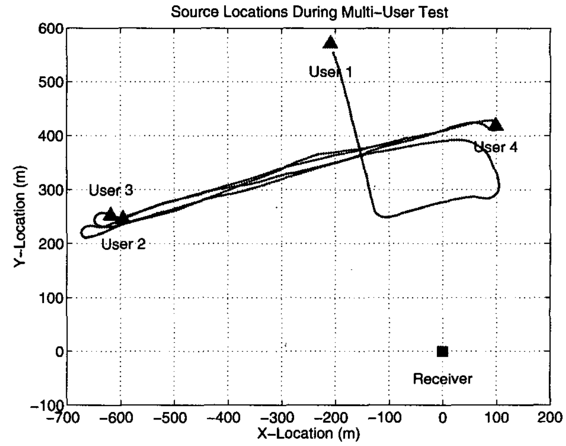

Fig. 3. Experimental site. The trajectory shows possible locations of the transmitter, which moved at speeds varying from zero to 2 m/s.

Island. The water depth was 15 m.

The signals were transmitted using the method described in Section II The carrier frequency was 23 kHz, and the transducer bandwidth was 5 kHz. The receiver was an eight-element vertical array, with 0.03 m separation between the elements. For all of the results shown, four of the eight elements were used with a resulting spacing of 0.06 m. Additional information about the acoustic array can be found in [10].

Transmission was organized in packets of length 15,000 data symbols. Signals corresponding to each user were sent to the receiver from approximately the same range, but separate locations. Individual packets were converted to baseband and stored for further processing. The multiuser signals were generated by summing the packets which are coarsely aligned in time according to the arriving ray with largest magnitude.

B. Processing results

Results of processing signals of four users transmitting from randomly chosen locations in the test area are summarized in Fig. 4. The first row of plots shown in the figure corresponds to the detection of the first user; the second row corresponds to the second user, etc. On the left hand side are the channel responses for the four different users. Each response spans about 10 ms, which is typical for shallow water. While the delay spread is similar for the four channels, the shape of each response differs.

The detector employs the method described in Section II A four-channel DFE is used with a soft-decision block decoder operating in the feedback loop. At the transmission rate of 5 ksps, the delay spread of 10 ms equals 50 data symbols. Thus, the intersymbol interference spans two codewords. The feedforward filters use 30 coefficients, while the feedback filter uses 15 coefficients. The RLS algorithm, with a forgetting factor of 0.998 is used to update the composite vector of the receiver coefficients. Two sets of results are shown in Fig. 4: single-user and four-user results. The center column shows the single-user results. The quality of performance varies depending upon the multipath composition of the channel, but in each instance decisions made directly on the DFE output have no errors.

When multiple users are present in the system, all transmit at equal power. The resulting interference appears at the receiver via the direct path and all of the multipaths. Thus, it is expected that for detection of user 1, users 2 and 3 are 'worse' interferers than user 4 because there are more paths present in their channels than in the channel of user 4. The sum of multiple users' signals was processed off-line using the receiver algorithm. The receiver uses a training signal of 1000 known symbols. During the training period, the equalizer coefficients are adjusted in the MMSE sense to extract the signal of the desired user. The receiver algorithm is first applied to extract the first user's signal, then the second user's signal, etc. The scatter plot observed in the decision-directed mode during detection of each user's signal is shown on the right hand side in Fig. 4. These plots represent the soft decisions from the DFE; thus, they do *not* reflect the gain obtained from coding. After decoding, in each of the four cases there were *no* bit errors.

Based on the above results, we may conclude that while the channels provide sufficient diversity to separate the users, the code provides sufficient gain to overcome the degradation caused by residual intersymbol interference and delayed channel estimation.

Unequal power ratio

The four-user results discussed above demonstrate good performance for signals that are approximately equal in power. However, another important case is when the users have significantly different received power levels. To demonstrate the performance of the receiver under conditions of low signal-to-interference ratio (SIR), users 2 and 4 from the above test case were combined with varying power ratios. User 2 was the desired user and user 4 was the interferer.

Table I shows the output SNR for varying signal-to-interference ratio. The results demonstrate that the receiver tolerates interference which is up to 12 dB stronger. In particular, at SIR of -9.5 dB or greater, all the errors are corrected by the BCH code; at SIR of -12 dB, there are some errors remaining after BCH decoding, but those are corrected by the outer Reed-Solomon code, and below -15 dB, reception fails due to error propagation in the DFE. The limiting value of the signal-to-noise ratio is an important design parameter for the future autonomous underwater networks which must employ some form of power control to overcome the near-far problem of single-user detection.

SIR [dB]	0	-3.5	-6	-9.5	-12	-15
SNRout [dB]	8.4	6.7	5.5	4.2	3.0	n/a

TABLE I

OUTPUT SNR FOR VARYING POWER RATIO BETWEEN USERS 2 AND 4 FROM FIG. 4. USER 2 IS THE WEAKER USER THAT IS BEING DETECTED.

IV. CONCLUSIONS

A problem faced when applying typical DS CDMA design to the underwater acoustic channel is that multipath propagation often destroys the orthogonality of the codes, thus diminishing their ability to separate the multiuser signals. Spreading gain is directly proportional to the system bandwidth which is limited in underwater applications, and, thus, increasing the spreading gain to achieve better performance is not efficient.

In this paper, we developed a multiuser system for a small number of users by allocating the available bandwidth expansion to an error-correction code, rather than the spreading code and exploiting the diversity in the communication channel to separate the multiuser signals. Signal processing based on multichannel decision-feedback equalization and soft-decision decoding of a BCH block code or a concatenated Reed-Solomon / BCH code was used. Experimental results demonstrated excellent system performance with four mobile users transmitting in very reverberant shallow water with a bandwidth expansion factor of 7.

The advantage of the proposed receiver processing method

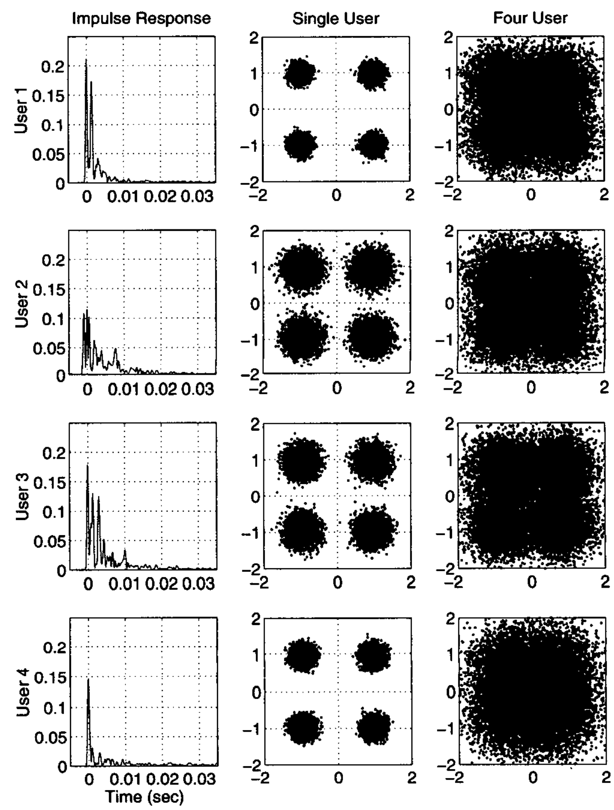

Fig. 4. Data processing results. Each user's channel response is shown on the left. Shown in the center are the scatter plots of the DFE output obtained when a single user is present in the system and that user is detected by the multichannel DFE. There are no decision errors. Shown on the right are the scatter plots of the soft decisions of a desired user's signal when four users are present in the system. The soft-decisions are provided to the BCH block decoder to provide reliable hard decisions for feedback. After BCH decoding, there are no bit errors in all four instances.

is that it does not require special spreading codes. The receiver needs only the training sequence of the desired user. The processing gain comes from coding and multipath channel diversity. Thus, the receiver algorithm is equally applicable to multirate communication systems, in which multiple users are allowed to transmit at different rates. Such systems are of interest because they offer a broader scope of applications. The choice of channel coding and optimal bandwidth allocation in the presence of multiuser interference are the subject of on-going research for shallow water networks. Future research will also include system design based on coded spread-spectrum for multiple-access communications between a large number of users.

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