Multiuser Code Acquisition in Multipath Channels

Milica Stojanovic
Massachusetts Institute of Technology
Cambridge, MA 02139
millitsa@mit.edu

Lee Freitag
Woods Hole Oceanographic Institution
Woods Hole, MA 02543
lfreitag@whoi.edu

***Abstract*— An adaptive algorithm for code acquisition in a direct sequence CDMA system is proposed. The algorithm is based on simultaneous channel estimation for multiple users. Its operation is demonstrated with up to four users transmitting over a 2 km shallow water channel within a 24 kHz acoustic bandwidth. Experimental results demonstrate the possibility of successful acquisition in the presence of strong multipath, with acquisition times on the order of a few hundred chip intervals.**

I. Introduction

Direct sequence (DS) code-division multiple access (CDMA) is considered for an underwater acoustic mobile network because of its superior properties in view of multipath resolution, soft capacity limit, possibility to provide low probability of intercept (LPI) mode of communication, and relaxed synchronization requirements as compared to time-division multiple access (TDMA). The last two properties, however, are contingent upon the system's ability to acquire the code timing of multiple users without the aid of pilot signals. Instead of using a dedicated signal, such as an out-of-band tone, before transmitting a data packet, each user simply joins the system without an announcement. In such a situation, the receiver must acquire the codes of all active users directly from the signal it receives before actual data detection can begin. Code acquisition involves determining the starting epoch of the spreading codes within the received signal.

Multiuser code acquisition in a multipath channel has received comparatively little attention in the relevant literature which mostly addresses the single-user problem, and often does so in idealized conditions (e.g., see [1] and references therein for an overview of recent work in radio communications). For underwater acoustic channels, a single-user code acquisition algorithm was proposed in [2], and its performance was tested through simulation.

While it is possible to apply the single-user algorithm to a multiuser scenario, interference from other users will limit its performance, because this algorithm uses only the knowledge of a single, desired user's spreading code. In contrast, a multiuser algorithm exploits the knowledge of all the users' spreading codes. In this paper, a multiuser acquisition algorithm is proposed, and its performance is demonstrated using experimental data. Similarly as in the single-user case, the code acquisition problem is regarded as the channel estimation problem; however, while the algorithm [2] considers multipath interference only, the algorithm presented here explicitly takes into account multiple-access interference as well.

The paper is organized as follows. In Sec.II the channel model and the acquisition algorithm are introduced. Sec.III describes the experiment and presents the results of real data processing. Finally, Sec.IV summarizes the conclusions and identifies the areas of future work.

II. Receiver Algorithm

A. System model

We are looking at a system with I users, where each user transmits a signal of the form

$$u_i(t) = \sum_{k=0}^{N_{acq}L-1} q_i(k) g_T(t - kT_c) \tag{1}$$

where T_c is the chip duration, $q_i(k)$ is the chip sequence, and $g_T(t)$ is the transmitter filter shaping pulse, usually chosen as a square-root raised cosine pulse, so that the baseband signal is bandlimited to $\pm 1/T_c$. During the time allocated to acquisition, no data modulation is present, and the sequence $q_i(k)$ represents the spreading code, periodically repeated N_{acq} times. For a code of period L, we thus have that $q_i(k+L) = q_i(k)$, for all integers k. The chips take values from a binary or a quaternary alphabet with unit amplitude. In an underwater acoustic system with a limited bandwidth, the chip rate is usually kept constant, while the data rate changes in accordance with the chosen spreading factor L.

The signal from each user passes through a multipath channel with an impulse response $c_i(t)$, and in addition experiences a phase distortion $\theta_i(t)$ caused by the system mobility. After passing through a receiver filter $g_R(t)$, the resulting signal is given by

$$v_i(t) = \sum_k q_i(k) h_i(t - kT_c) e^{j\theta_i(t)} \tag{2}$$

where $h_i(t) = g_T(t) * c_i(t) * g_R(t)$ is the overall channel corresponding to the ith user. This signal represents the ith user's contribution to the received signal, which is given by

$$v(t) = \sum_{i=1}^{I} v_i(t) + z(t) \tag{3}$$

where $z(t)$ is zero-mean additive noise.

The received signal is sampled at the Nyquist or higher rate, corresponding to N_s samples per chip interval. A total of LN_s samples are collected to form the received signal vector at time kT_c, $\mathbf{v}(k) = [v(kT_c + (LN_s - 1)T_s \dots v(kT_c + T_s) v(kT_c)]^T$, where $[\cdot]^T$ stands for transpose. This vector can be expressed in terms of individual user's contributions as

$$\mathbf{v}(k) = \sum_{i=1}^{I} \mathbf{v}_i(k) + \mathbf{z}(k) = \overline{\mathbf{v}}(k) + \mathbf{z}(k) \tag{4}$$

The ith user's signal vector is modeled as

$$\mathbf{v}_i(k) = \sum_{m} \mathbf{h}_i(m) q_i(k-m) e^{j\theta_i(k)} \tag{5}$$

where the channel vector is defined as $\mathbf{h}_i(m) = [h(mT_c + (LN_s - 1)T_s) \dots h_i(mT_c + T_s) h_i(mT_c)]^T$. The vector $\mathbf{h}_i(0)$ is called the reference channel vector of the ith user. Note that the reference vector is defined as the zero-shift vector in reference to the (zero-shift) code $q_i(k)$. The total time span of this vector is LT_c, and as long as this span is greater than the multipath spread of the channel, all the significant channel components are contained within the reference vector. Assuming that the code length is chosen to satisfy this requirement, we note that the vector $\mathbf{h}_i(1)$ can be obtained from the reference vector by shifting its components down by N_s and filling the top with N_s zeros. The other channel vectors can be obtained similarly. Thus, if the reference vector is known, the signal $\mathbf{v}_i(k)$ can be reconstructed. This is the key observation used to develop the acquisition algorithm.

B. Channel estimation

The acquisition algorithm in essence deals with estimating the channel responses of all users. When each reference vector $\mathbf{h}_i(0)$ has been estimated, the strongest of its components indicates the delay at which the code $q_i(k)$ is aligned with the incoming signal, or, equivalently, the starting epoch of that code.

Assuming that at some point in time, kT_c, we have the estimates of the channel vectors $\hat{\mathbf{h}}_i(m,k)$, as well as the phase estimates $\hat{\theta}_i(k)$, we can form an estimate of each user's signal as

$$\hat{\mathbf{v}}_i(k) = \sum_{m} \hat{\mathbf{h}}_i(m,k) q_i(k-m) e^{j\hat{\theta}_i(k)} \tag{6}$$

The sum of these signals will be an estimate of the overall signal mean,

$$\hat{\overline{\mathbf{v}}}(k) = \sum_{i=1}^{I} \hat{\mathbf{v}}_i(k) \tag{7}$$

For a given user u, we now form the signal

$$\hat{\mathbf{v}}_{0u}(k) = \mathbf{v}(k) - \sum_{i \neq u} \hat{\mathbf{v}}_i(k) - \sum_{m \neq 0} \hat{\mathbf{h}}_u(m,k) q_u(k-m) e^{j\hat{\theta}_u(k)} \tag{8}$$

where the first of the two terms subtracted is an estimate of multiple-access interference (MAI), and the second is an estimate of self-interference caused by multipath, i.e. intersymbol interference (ISI). If the estimates are perfect, this signal is equal to

$$\mathbf{v}_{0u}(k) = \mathbf{h}_u(0) q_i(k) e^{j\theta_u(k)} + \mathbf{z}(k) \tag{9}$$

which represents an interference-free contribution of user u, as both MAI and ISI have been canceled. Note that the estimate of interference-free signal (8) can also be represented as

$$\hat{\mathbf{v}}_{0u}(k) = \mathbf{v}(k) - [\hat{\overline{\mathbf{v}}}(k) - \hat{\mathbf{h}}_u(0,k) q_u(k) e^{j\hat{\theta}_u(k)}] \tag{10}$$

This signal provides the basis for multiuser channel estimation procedure, which capitalizes on the fact that

$$\mathbf{h}_u(0) = E\{\mathbf{v}_{0u}(k) e^{-j\theta_u(k)} q_u^*(k)\} \tag{11}$$

When estimation errors are present, the signal $\hat{\mathbf{v}}_{0u}(k)$ will contain additional noise. However, as long as the estimation noise is zero-mean, substitution of estimated quantities into the above expression is justified. An unbiased adaptive channel estimator for user u is given by

$$\hat{\mathbf{h}}_u(0,k+1) = \lambda \hat{\mathbf{h}}_u(0,k) + (1-\lambda) \hat{\mathbf{v}}_{0u}(k) e^{-j\hat{\theta}_u(k)} q_u^*(k) \tag{12}$$

where $\lambda \in [0,1)$ is the algorithm parameter, normally chosen close to 1.

Because different users may have different patterns of motion, a different Doppler distortion must be expected for each user's signal. Consequently, I independent phase terms $\theta_i(k)$ have to be estimated. To do so, we associate a phase lock loop (PLL) with each user's signal. The PLL operation is guided by the error between a chip estimate and its true value (which is known during the acquisition process). The chip estimate is obtained as (prime denotes conjugate transpose):

$$\hat{q}_u(k) = \hat{\mathbf{h}}_u'(0,k) \hat{\mathbf{v}}_{0u}(k) e^{-j\hat{\theta}_u(k)} \tag{13}$$

which, up to a positive scaling constant, represents the MMSE estimate of $q_u(k)$, given the knowledge of the channel and the phase. The chip error, $e_u(k) = \hat{q}_u(k) - q_u(k)$, is used to form the gradient

$$\psi_u(k) = -\frac{1}{2} \frac{\partial |e_u(k)|^2}{\partial \hat{\theta}_u(k)} = \\ = Im\{\hat{\mathbf{h}}_u'(0,k)[\mathbf{v}(k) - \sum_{i \neq u} \hat{\mathbf{v}}_i(k)] e^{-j\hat{\theta}_u(k)} e_u^*(k)\} \tag{14}$$

The phase estimate for user u is now obtained as

$$\hat{\theta}_u(k+1) = \hat{\theta}_u(k) + K_1 \psi_u(k) + K_2 \sum_{n \leq k} \psi_u(n) \tag{15}$$

where K_1, K_2 are the PLL constants (normally, $K_2 = K_1/10$).

The quality of channel and phase estimates is controlled by the parameters λ and K_1. In addition, it can be greatly improved by introducing truncation, or sparsing of the channel estimate. Namely, only those coefficients of the channel estimate that exceed in magnitude a certain threshold are kept and updated, while the rest are set to zero. This procedure ensures that unnecessary noise is not fed back through the estimation loop, and it represents a crucial step in executing the algorithm. The truncation threshold is chosen for each user in reference to the strongest coefficient of its channel vector. Initially, full channel is estimated, and truncation begins after a certain number of iterations (e.g. twice the code length).

C. Code acquisition

The receiver algorithm so far was devoted to adaptive channel estimation and phase tracking for multiple users. Once these estimates are available, the acquisition test can be performed. The acquisition test is based on monitoring the quality of chip estimates (13). Because the system is expected to operate in conditions of high interference, or low SNR in an LPI mode, the quality of these estimates may be poor. Hence, a certain number of chip estimates, L_d, is used to form the decision variable

$$\hat{d}_u(k) = \frac{1}{L_d} \sum_{n=k-L_d+1}^{k} \hat{q}_u(k) q_u^*(k) \tag{16}$$

This soft decision variable should be close to 1, as there has been no data modulation. Alternatively, a hard decision variable can be used. For binary spreading, it is given by

$$\tilde{d}_u(k) = sgn\{Re\{\hat{d}_u(k)\}\} \tag{17}$$

The acquisition is declared when a certain number, N_{test}, of successive +1 decisions have been counted. Note that both N_{test} and L_d are design parameters, whose values do not affect the quality of channel and phase estimation, but only the acquisition statistics.

D. Algorithm summary

The algorithm implementation is defined by the following steps, carried out for $k = 1$ through some k_{end} when acquisition is declared (or until the acquisition preamble has expired):

1) Form new input signal vector $\mathbf{v}(k)$.
2) Using existing estimates, form $\hat{\mathbf{v}}_i(k)$ for all $i = 1, \dots I$. Note that it is not necessary to perform actual summations as in (6); it is only necessary to compute the top N_s elements of each vector, while the rest are obtained by down-shifting of the current elements by N_s.
3) Calculate $\hat{\bar{\mathbf{v}}}(k)$, the estimated signal mean (7).
4) For all users, $u = 1, \dots I$:
   a) form the estimate of interference-free signal (10)
   b) update the channel vectors (12) and truncate
   c) form the chip estimate (13)
   d) update the phase (14,15)
   e) calculate decision variable (16, 17)
   f) test: if all users passed, declare acquisition.

III. Experimental Results

A. Signal design and the experiment

DS spread spectrum signals were designed for four users. Kasami codes were used, with period 15, 63 and 255. Transmission was organized in packets, and for purposes of acquisition, 5120 unmodulated chips were allocated at the beginning of each data packet. Pulse shaping was performed using square-root raised cosine pulses with roll-off factor 0.25. The chip rate was 19,200 chips per second. The signals were modulated onto a carrier of 33 kHz, thus occupying the bandwidth between 21 kHz and 45 kHz. The sampling frequency used for generating the transmitted signals was 96 kHz. In addition to the multiuser signals, channel probes were transmitted at regular intervals between the data packets. Channel probe consisted of a frequency sweep spanning the entire 24 kHz bandwidth in 100 ms.

The signals were transmitted during an experiment that took place in the fall of 2003, off the coast of Elba in Italy. The signals of four users at a given code length were transmitted sequentially, with approximately 24 seconds between the packets' starting times. The next group of signals, corresponding to a different code length, was transmitted a minute or so later. The range between the transmitter and receiver was 2.3 km, and the water depth was approximately 100 m. The signals were recorded at a depth of 30 m.

Multiuser test signals were generated after the fact, by adding a desired number of signals. All the signal processing at receiver was performed in complex baseband, using 2 samples per chip.

B. Channel characteristics

Fig.1 shows snapshots of the channel response magnitudes obtained by matched filtering to the frequency sweep probe at the beginning of each group of multiuser signals. Clearly, the channel exhibits strong multipath and a moderate delay spread. In most of the cases recorded, there were two pronounced multipath components, separated by approximately 12.5 chips. These components are followed by distant echoes of decaying energy. From comparison of the three responses, it is evident that the channel is time-varying. The first arrival appears as strongest in the first recording (preceeding the L=15 group of four users), yielding to the late arrival in the second recording (L=63) and regaining strength the third recording (L=255).

To illustrate the channel variation within a data packet, cross-correlation was performed using one period of the spreading code over the entire received acquisition preamble. The result for L=63 code is shown in Fig.2. We note that the position of significant channel coefficients does not change with time, while their relative strength varies with time.

C. Performance results

A number of test cases were constructed to demonstrate the performance of the acquisition algorithm. We begin by looking at the single-user scenario, and illustrate the algorithm performance using the L=63 data set corresponding to user 1. This is the same data set whose channel response is characterized in Fig.2. The algorithm performance is depicted in Fig.3.

Shown on the left hand side is the performance of the estimator. The top graph shows the channel estimate magnitude at the end of the acquisition preamble. There are two pronounced multipath arrivals, corresponding to the two arrivals observed earlier from probe filtering. A single coefficient is kept to represent the early arrival, while six coefficients represent the late (principal) arrival. The truncation threshold was set to 0.5, i.e. all coefficients whose amplitude was less than half of the maximal one were discarded. The time evolution of the

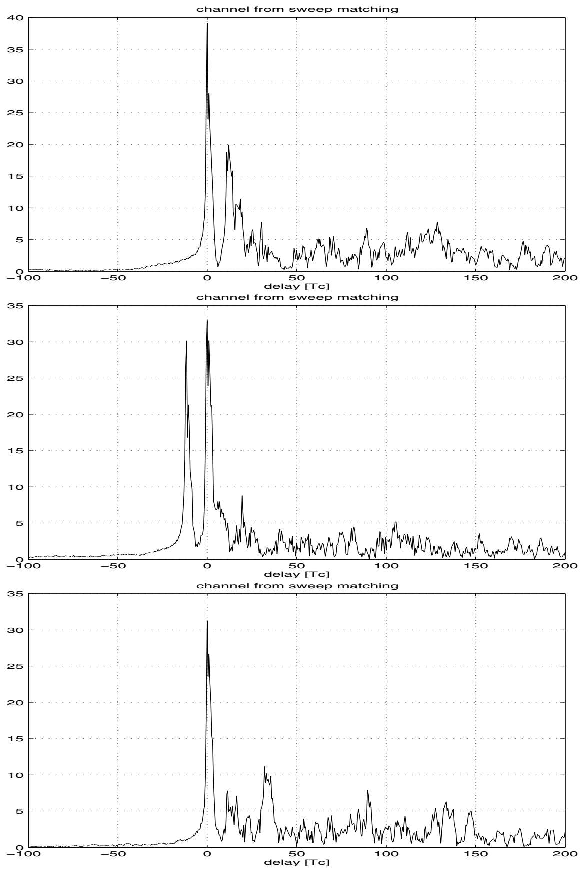

Fig. 1. Channel responses, estimated from the frequency sweep preamble prior to receiving the L=15, 63, and 255 group of multiuser signals (shown top to bottom, respectively).

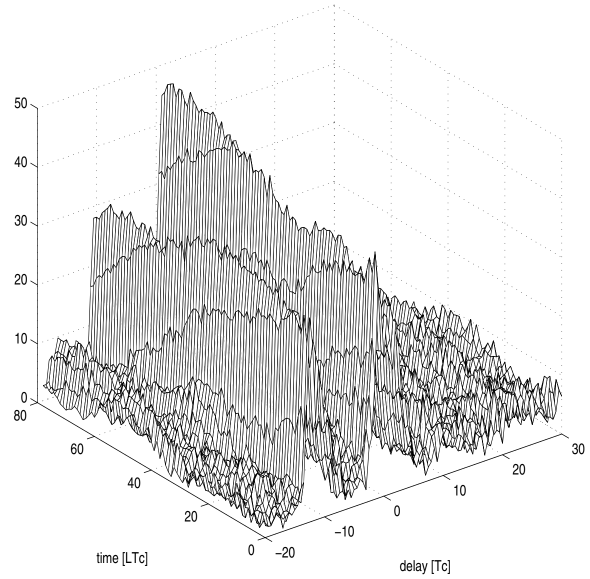

Fig. 2. Channel responses in time, estimated from the acquisition preamble. Each response represents cross-correlation between the received signal and one period of the spreading code. L=63, user #1.

significant channel coefficients is shown in the bottom graph. The middle graph shows the phase estimate and indicates the corresponding Doppler frequency offset f_d.

Shown on the right hand side are the quantities relevant for the acquisition test. The top graph represents the soft decision variable, $Re\{\hat{d}_u(k)\}$, or the "running despreader" output, while the graph below it shows the corresponding hard value. The bottom graph shows the sequence timing estimate, which is obtained from the position of the strongest channel coefficient. The estimated timing coincides with the true one. Note that for purposes of later signal detection, it is only important that the timing be acquired correctly to within one chip, or, more precisely, to within a few chips when chip-rate filtering is used for detection (as it should be the case for frequency selective channels).

The algorithm performance is shown here for an extended period of time, i.e., much longer than the time needed for code acquisition. This is done so as to illustrate the adaptive channel tracking, and to verify the algorithm stability over time. While the various estimates are shown over 5000 chip intervals, the code timing is successfully acquired already after a few code periods.

In the presence of one interferer, performance of the single-user algorithm deteriorates, and it fails completely when three users are present in the system. Multiuser detection becomes necessary in this situation.

Multiuser acquisition is illustrated using three sets of signals, corresponding to the code lengths L=15, 63 and 255. The number of users present is two, three, and four, respectively. The users are added without power adjustment, resulting in approximately equal-energy signals for all four users (as they arrive from the same location in the experiment). The signals of all users are taken with the same, arbitrarily chosen delay, i.e. their codes have the same correct starting epoch (this choice does not affect the system performance). The receiver parameters are λ=0.999 and K_1=0.0001; truncation is performed after $2L$ iterations with a threshold of 0.5, and the running despreader length is $L_d = L$.

Before proceeding to data analysis, we note that L=15 represents a marginal case where justification of the multipath assumption may be seriously compromised, as the multipath spread is certainly longer than 15 chips. However, the majority of multipath energy is contained within this short interval, which served as an incentive to test the algorithm with this data set as well.

The result of code length 15 acquisition test is illustrated in Fig.4. Interestingly, and perhaps to some surprise, acquisition is achieved easily. The channel estimator keeps a single coefficient, and there is little phase variation in this data set. Fig.4 illustrates the time evolution of the running despreader check, i.e. the hard decision variable (17) and the corresponding timing estimate for the two users. It took less than 100 chip intervals to acquire the correct timing of both users. The correct timing is maintained thereafter. In this data set, it was possible to acquire the code of two users, but not more than that.

Fig.5 summarizes the performance for code length 63. Three

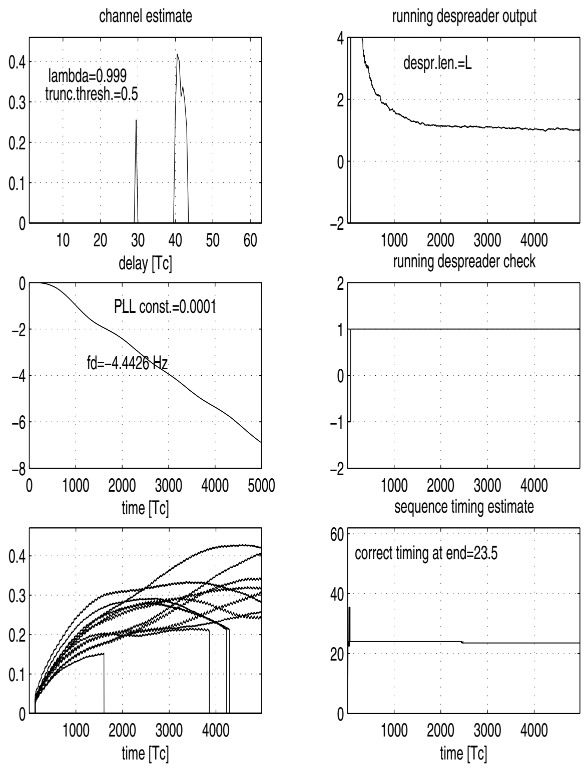

Fig. 3. Performance of the channel estimation and acquisition algorithm for a single-user, code length 63.

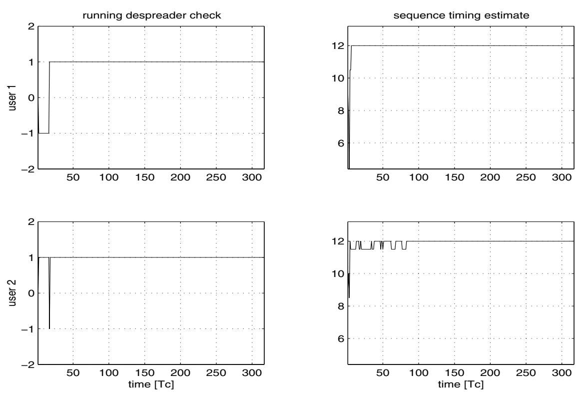

Fig. 4. Multiuser acquisition for two users, code length 15.

users are present in this case. The timing estimate of 60 chips is the correct one. Code timing of all three users is acquired after less than 200 chip intervals.

Finally, Fig.6 illustrates the performance for code length 255, with four users present in the system. Acquisition is achieved for all users after less than 400 chip intervals. At this time, all the tests have been positive for a number of iterations.

Judging by the test statistics, one may conclude that the running despreader length equal to the code period was chosen appropriately, and also that choosing N_{test} on the order of 1000 would be a good choice for the present conditions. Thus, the acquisition preamble length of 5000 was a safe design

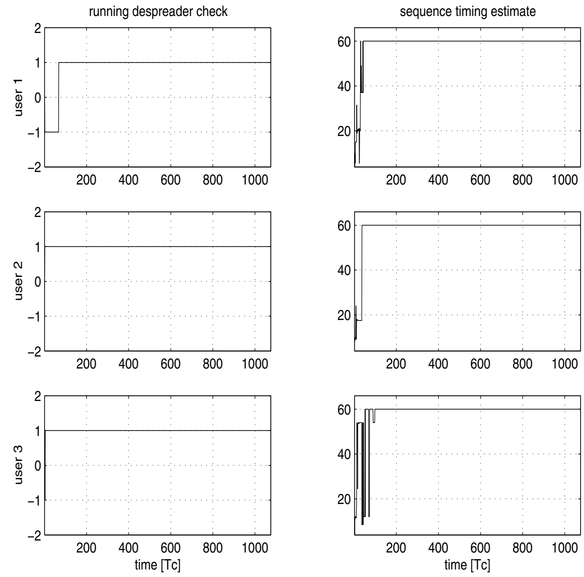

Fig. 5. Multiuser acquisition for three users, code length 63.

choice.

IV. Conclusion and Further Work

Multiuser acquisition method developed in this work is based on multiple channel estimation coupled with interference cancelation that includes both the multiple-access interference and the self-interference caused by multipath propagation. The algorithm requires that the spreading code period be longer than the multipath spread. For applications to realistic underwater network scenarios, where motion-induced Doppler distortion cannot be neglected, the channel estimation process is aided by multiuser phase tracking. The resulting algorithm, although conceptually complex, has a remarkably low implementation complexity. Computational efficiency draws on two factors: a simple adaptive channel estimation algorithm, and a vector-shifting method that eliminates the need for explicit calculations of interference components. In addition, a crucial step in the algorithm operation is truncation, or sparsing of the channel estimates, which reduces estimation error propagation, and also may significantly reduce the number of adaptively adjusted coefficients.

The algorithm operation was demonstrated using experimental data, transmitted over a 2 km shallow water channel at 19.2 kilochips per second, utilizing a 24 kHz bandwidth around the center frequency of 33 kHz. The channel exhibited very strong multipath, and up to four users were considered operating simultaneously using the Kasami codes with varying spreading factor. Excellent results were achieved in both single-user and multiple-user scenarios. In particular, it was shown that the code timing of multiple users could be correctly acquired with acquisition times on the order of a few hundreds of chip intervals.

Judging by the experimental data results, it appears that efficient channel estimation can be exploited as part of the data

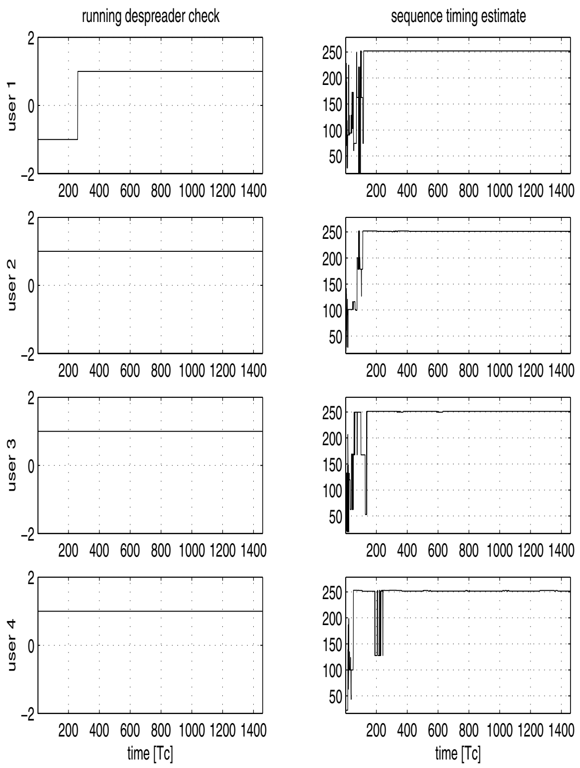

Fig. 6. Multiuser acquisition for four users, code length 255.

detection process that follows the acquisition phase. Future work should concentrate on multiuser data detection based on adaptive channel estimation. This approach differs from the usual multiuser equalization [3] in that it explicitly relies on the knowledge of channel responses of multiple users. In the single-user case, it has been demonstrated as an effective method for improving the performance as well as reducing the computational complexity of equalization when the channel is naturally sparse, as it often is the case in underwater acoustic communication systems [4]. Both of these factors – improved performance and reduced computational complexity – are crucial for the implementation of future multiple-access underwater systems based on DS CDMA.

ACKNOWLEDGMENT

This work is part of the project "Very Shallow Water Surf Zone Mine Countermeasures" (VSW/SZ MCM), supported by the Office of Naval Research (ONR).

REFERENCES

[1] E.Aktas and U.Mitra, "Single-user sparse channel acquisition in multiuser DS-CDMA systems," *IEEE Trans. Commun.*, vol.51, No.4, Apr. 2003, pp.682-693.

[2] M.Stojanovic and L.Freitag, "Acquisition of direct sequence spread spectrum acoustic communication signals," in Proc. *IEEE Oceans'03 Conference*, Sept. 2003.

[3] M.Stojanovic and Z.Zvonar, "Multichannel processing of broadband multiuser communication signals in shallow water acoustic channels," *IEEE J. Ocean. Eng.*, vol.21, No.2, Apr. 1996, pp.156-166.

[4] M.Stojanovic, L.Freitag and M.Johnson, "Channel-estimation-based adaptive equalization of underwater acoustic signals,"in Proc. *IEEE Oceans'99 Conference*, Sept. 1999.