Synchronized Intelligent Buoy Network for Underwater Positioning

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***Abstract*—This paper addresses the development of a new generation of lightweight intelligent buoys. These buoys are used to support underwater acoustic positioning systems, but were also designed to be elements of portable coastal observatories for short term deployments. We will present the main features of a buoy prototype including the physical structure, the computational system and algorithms developed to support operations. The paper also shows how to take advantage of this new tool to implement different navigation algorithms for AUVs.**

I. Introduction

The accurate estimation of the position of autonomous systems has always been a key research area in robotics and it is particularly challenging in the case of Autonomous Underwater Vehicles (AUVs). Many methods are used to face this problem and, among them, the Long Baseline method (LBL) stands out for the ability to provide absolute coordinates with bounded accuracy. This method uses a network of acoustic beacons previously deployed on the sea-floor with known GPS coordinates. The underwater vehicle can compute its underwater position by acoustically measuring the ranges to the beacons. Such ranges are determined by measuring the elapsed time between an interrogation acoustic signal sent by the AUV and the arrival of the responses sent by the beacons. The absolute position is computed using a triangulation of these ranges, together with the GPS coordinates of the beacons.

Two of the major problems associated to underwater positioning systems are the position update rate and power consumption. Both these problems can be optimized by working with different approaches to the positioning algorithms. To better understand these issues lets take a simple example for consideration where an AUV needs to be positioned. Assume for simplicity a baseline defined by two buoys, B1 and B2. The buoys and AUV form a equilateral triangle with a 3 kilometer length per side. With a traditional method, the vehicle sends an interrogation acoustic ping, which triggers the response of the beacons. Taking for the speed of sound in water 1,500 meters per second, it takes about 4 seconds for a single position update if both beacons respond to the same interrogation signal, or even more if using a different scheme. Such a long time between position updates may degrade navigation performance, specially in case of disturbances such as currents. This could be improved by applying more elaborated schemes like sending multiple interrogation signals between position updates. However it would yield a higher complexity of the navigation algorithm and an overhead on AUV power consumption.

Fig. 1. The MARES Autonomous Underwater Vehicle

As far as logistics are concerned, the installation of permanent sea-floor beacons is a relatively complex task, and for some years now, a new approach emerged where these beacons morph into fast deployable buoys equipped with acoustic transducers and GPS receivers. Although this yields improvements in terms of logistics it also raises the necessity of calibrating the network parameters for each new operational environment. This is due to the high variability of the acoustic proprieties in the operational scenarios. These variations are strongly related with oceanographic characteristics like salinity, temperature and currents, as well as background noise originated by natural and artificial sources [1], [2], [3], [4], [5].

In this work we will present the prototype of a synchronized intelligent buoy (SIB) developed in our labs that takes the acoustic buoys one step closer to integrated nodes of a sensors network, *i.e.* key elements in portable coastal observatories. Some of the main features of this prototype will be shown in section II and in section III the mechanisms used for clock synchronization. Next, in section IV, an automatic calibration algorithm of the acoustic network that only became possible with the introduction of synchronization between the nodes of the network will be demonstrated. And finally we will also describe examples of navigation algorithms that are possible to implement in this new system, based on the AUV MARES[6] (Fig.1) and using two synchronized intelligent buoys.

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II. Synchronized Intelligent Buoy

With the growing developments in underwater monitoring it becomes increasingly important the evolution of the acoustic beacon to an active node of the sensors network that composes the basis to a portable coastal observatory. In that sense it is desirable to possess a modular architecture capable of managing a wide range of sensors and of being easily adapted to different types of missions. In order to achieve this, we designed the buoy architecture around an expandable single board computer (Fig.2). However the integration of such system begins to raise issues related with its operational autonomy. A compromise between performance and power consumption must then be attained, that can be accomplished by the simultaneous use of low power electronics and the inclusion of an integrated power production module.

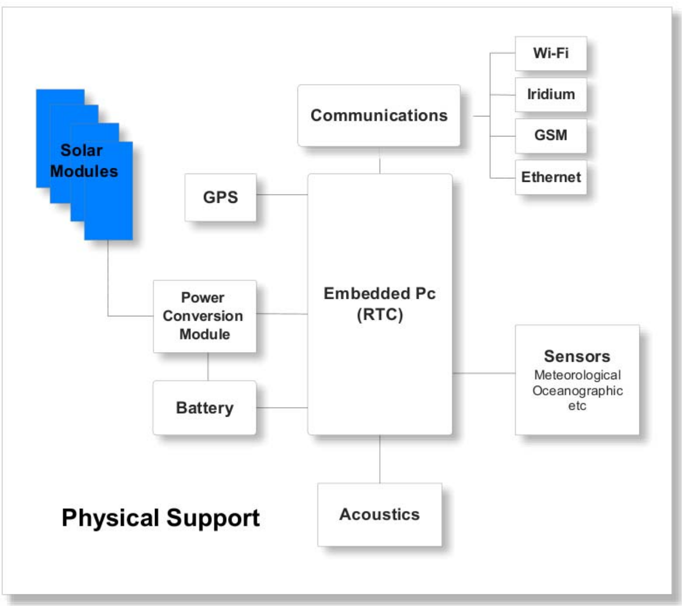

Fig. 2. SIB Internal Architecture

Bearing this in mind we then proceeded to develop a prototype of a modular acoustic buoy (Fig.3 and Fig.4) that met all the described requirements. The physical prototype was fully designed using the 3D mechanical design software, SolidWorks. It is composed of a primary flotation module, with a tower fitted with four 12 Watt solar modules, that gives shelter to a waterproof box for electronics. Connected to the tower there is an antenna module capable of supporting various types of antennas, radar deflector and even meteorologic sensors, with an embedded light signaling. Additionally, there are secondary flotation modules for extra weight compensation, and a lower section that allows the installation of the acoustic transducer and various underwater sensors.

The box for electronics is fitted with a 12 Volts battery, the low power computer, a low power UBlox RCB-LJ GPS receiver, an USB Wi-Fi dongle with an external antenna connection and a monitored power converter.

The low power computer is a Technologic Systems TS-7260 that is an ARM SBC running Debian Linux from an SD-Card, which features a 200 MHz Cirrus processor, a 10/100 Ethernet port and more importantly a real time clock. As far as interfacing is concerned, there are several other ports available to connect additional sensors or subsystems such as two USB ports expandable through a hub, two serial ports, thirty digital I/Os and two 12 bits ADC. If further connections are required expansion boards can be stacked onto the on board PC/104 bus.

As far as acoustics are concerned, each buoy integrates a receiver and a transmitter board connected to an underwater transducer in the range 20-30KHz. On the receiver board, a set of analog filters are tuned to 8 different frequencies allowing for the parallel detection and time-stamping of 8 incoming signals. On the transmitter, the waveforms are synthetically generated on a microcontroller and then amplified to provide up to 198dB re 1V/μPa of transmitted power. In the current implementation, the transmitter needs about 500 ms to fully *recharge*, and so this limits the number of *pings* to about 2 per second. Although the building blocks are similar to the ones used onboard the MARES AUV, the energy constraints are less demanding on the buoys and therefore the transmitter boards are configured to provide extra power. The overall system is reconfigurable, allowing for pre mission or on line programming of detection/reply frequency pairs, channel sensitivity, etc., in order to adapt to the environment conditions of the operation area. The ability to detect underwater acoustic signals is highly dependent on the characteristics of the water mass and also on background noise, but typically we have been using these boards to support the navigation of the MARES AUV in ranges of 1–2 kilometers.

III. Clock Synchronization

Synchronization of the internal clocks of both AUV and buoys is held prior to a mission start and it is achieved using a NTP daemon (ntpd) for Linux. The Network Time Protocol (NTP) is a protocol used in the synchronization of clocks in computers over variable latency data networks, using UDP [7].

Our system features a GPS receiver capable of providing GPS time and the pulse per second signal (PPS). However the NTP daemon can't handle these signals directly, and for this reason a second daemon called GPS daemon (gpsd) was used. This daemon gathers the feeds from the GPS receiver and provides them on two virtual network servers, one providing GPS time and the other GPS time tagged with the PPS signal, when available, allowing the NTP daemon to use standard heuristics to choose between them.

The synchronization in performed before a mission and does not extend to the mission itself for two main reasons. The first one concerns the timestamps of the sensors measurements. We can't afford to have a time shift in the middle of a data collect, for example when measuring gradients, since it would corrupt all the information. The second, is linked to the type of network we're working with. When the AUV submerges all communications, except acoustics, are lost. This problem

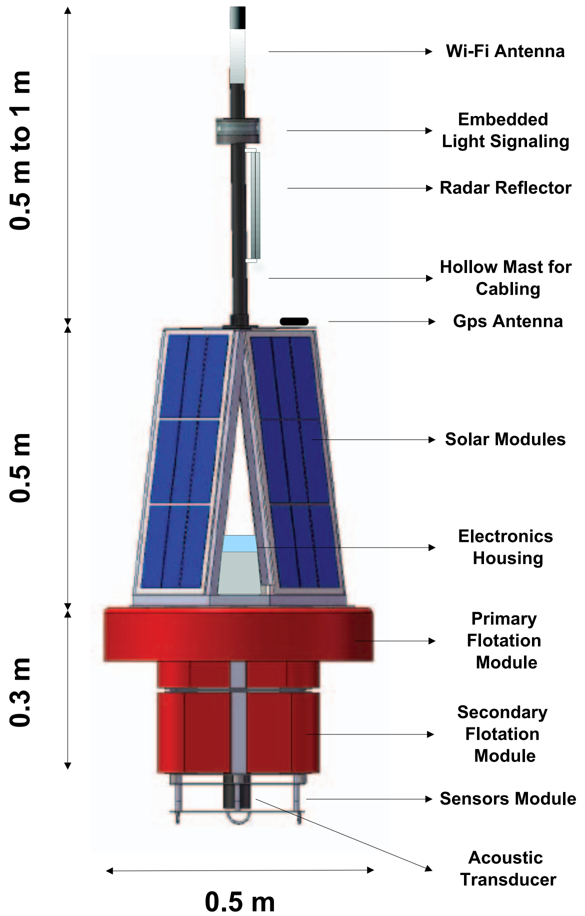

Fig. 3. SIB Model

Fig. 4. SIB Under Construction and Finished Prototype

could be overcome by introducing a custom algorithm that used acoustics to perform some kind of synchronization of the AUV clock, adapting a custom algorithm such as described in [8]. In any case, this goes far beyond our current needs since the real time clock is capable of maintaining the accuracy for the duration of our standard mission (a few hours).

The existence of synchronization between the nodes of the network (including AUVs) will allow us to do a one-way navigation, that will be address further on, and perform missions with time constrains, like for example a *rendez-vous* between two vehicles at a given time.

IV. Auto-Calibration of the Acoustic Network

The mechanisms of underwater acoustic signal transmission are extremely intricate and are dependent on many factors such as water properties and variable background noise emitted from natural and artificial sources. Systems designed for application in different scenarios should therefore have the ability to adapt to the conditions of the acoustic environment. The calibration of the acoustic network is an integrated procedure of the mission planning and supervision software, also developed in our labs, that is described in [9]. Our buoys are programmable both in terms of transmitting and receiving frequencies and it is possible to set the detection level of each frequency independently. The automatic calibration procedure must be able to determine the most suitable set of frequencies according to signal to noise ratios.

The calibration procedure is divided into two main steps, the determination of the environment noise level and the detection of the peak levels of the signal arriving from the other nodes. This is done from all the different codes emitted. For the first step, each node simply listens to the environment, without any emissions of acoustic signals. The second step is much more complicated since it requires the exchange of acoustic signals between all devices, in order to determine the most suitable configuration for operations. In previous implementations of acoustic networks this procedure was performed manually, which not only represented a time consuming task but was also prone to errors due to the unsystematic nature of the procedure.

In order to achieve a fast and systematic procedure we have implemented an algorithm that takes advantage of clock synchronization to predict the arrival of acoustic signals emitted by the remaining nodes. All the nodes can by active during the calibration process, emitting multiple acoustic signals, minimizing its execution time (Fig.5). Nonetheless it is crucial to ensure the proper functioning of the network, by respecting

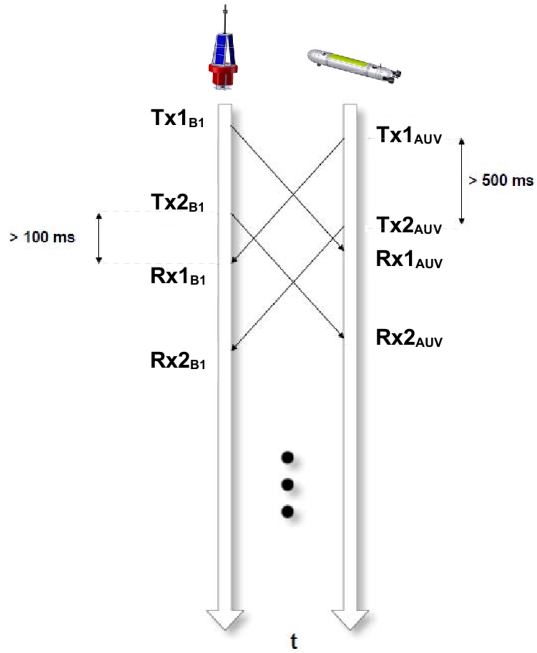

Fig. 5. Acoustic Signals Exchange Scheme. Note that multiple signals can be traveling through the medium at the same time

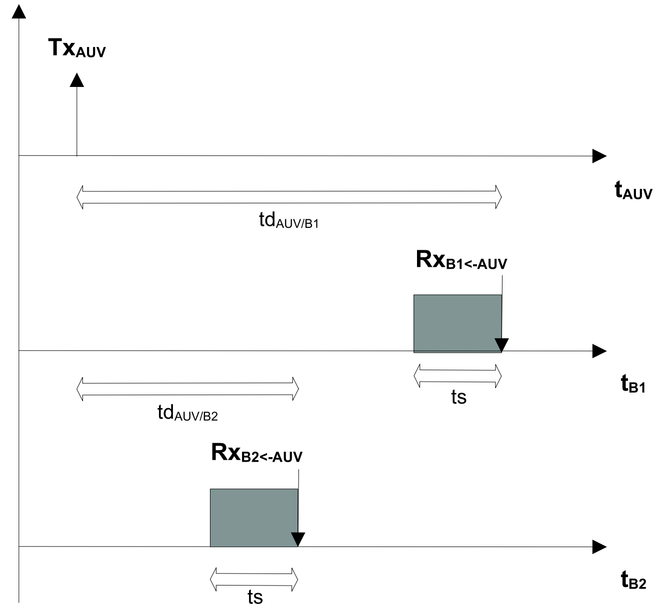

Fig. 6. Acoustic Signal Emission Constraints. During t_s the receiving buoys should not emit any signal.

the restrictions of the acoustic hardware. Also, regarding the emission of acoustic signals at the same frequency, to ensure a collision free execution. For this purpose a set of mathematical equations were deduced, that restrict the emission of acoustic signals by the nodes of the acoustic network.

The hardware constraints are related to the systems maximum rate of acoustic signals emission and the period of deafness following the emission of an acoustic signal. For now on they will be referred as t_c and t_s, respectively. If we take once more for consideration the network composed by the AUV MARES and a baseline defined by two buoys, B1 and B2, we can derive the relation between emission and reception of an acoustic signal by the nodes.

The emission of an acoustic signal by the AUV at time $T_{x_{AUV}}$ will lead to a reception in buoy 1 at a time given by

$$R_{x_{B1\leftarrow AUV}} = T_{x_{AUV}} + t_{d_{B1/AUV}}$$

where $t_{d_{B1/AUV}}$ is the signal flight time between B1 and the AUV. The same signal will arrive at buoy 2 after a different time $t_{d_{B2/AUV}}$ such as represented in figure 6.

To avoid the reception of the acoustic signal from the AUV during the deafness period of buoy 1, the following must be respected,

$$T_{x_{B1}} < R_{x_{B1\leftarrow AUV}} - t_s \quad \vee \quad T_{x_{B1}} > R_{x_{B1\leftarrow AUV}}$$

Generalizing this equation to any node of a network with n elements, we will have,

$$T_{x_k} < T_{x_m} + t_{d_{k/m}} - t_s \quad \vee \quad T_{x_k} > T_{x_m} + t_{d_{k/m}},$$

$$\forall k \vee m \leq n \quad with\ k \neq m$$

This equation means that a node from a network can only emit an acoustic signal prior to an instant equivalent to the reception of another acoustic signal sent by any other node of the network minus its deafness period, or immediately after the reception of that signal.

The above relationships describe time slots when it is possible to emit acoustic signals. It is now necessary to define the moment when each node starts emission and, if possible, a pattern for periodic transmissions. For that a new set of equations had to be deduced.

Lets admit a generic acoustic network composed of three nodes with any $t_{d_{k/m}}$ between them. These nodes will from now on be referred as x, y and z, such that $t_{d_{x/y}} \geq t_{d_{x/z}} \geq t_{d_{y/z}}$. Lets also admit that they are all synchronized with the same time referential, in which an initial moment t_0 was defined. In these terms we are now set to determine t_{x_0}, t_{y_0} and t_{z_0}, where t_{k_0} corresponds to the moment when the node k starts to emit acoustic signals.

We chose an order of transmission based on the relative distances between nodes. Assuming $t_{z_0} \geq t_{y_0} \geq t_{x_0}$ and setting $t_{x_0} = t_0$ we will have, that t_{y_0} should be assigned to one of the following intervals.

$$\begin{cases} t_0 \leq t_{y_0} < min(t_0 + t_{d_{x/y}}, t_{z_0} + t_{d_{y/z}}) - t_s \\ t_{y_0} > (t_0 + t_{d_{x/y}}) \\ t_{y_0} > (t_{z_0} + t_{d_{y/z}}) \end{cases} \quad (1)$$

And t_{z_0} should be assigned to one of the following intervals.

$$\begin{cases} t_0 \leq t_{z_0} < min(t_0 + t_{d_{x/z}}, t_{y_0} + t_{d_{y/z}}) - t_s \\ t_{z_0} > (t_0 + t_{d_{x/z}}) \\ t_{z_0} > (t_{y_0} + t_{d_{y/z}}) \end{cases} \quad (2)$$

We have restricted the moment when each node in the network starts to emit acoustic signals. Nonetheless the interference of an emitted signal in the following emissions of the remaining nodes is not considered by these equations. The question now is when to emit a new signal? And the answer to that question can be found in the characteristics of the acoustic hardware. If a node could emit acoustic signals at the maximum rate t_c beginning at an initial moment t_{k_0}, that would be the best scenario possible.
So, if we define $t_{x_1} = t_{x_0} + t_c$, t_{y_0} and t_{z_0} will be restricted to,

$$t_{x_1} - t_{d_{x/y}} < t_{y_0} < t_{x_1} - t_s - t_{d_{x/y}} \quad (3)$$

$$t_{x_1} - t_{d_{x/z}} < t_{z_0} < t_{x_1} - t_s - t_{d_{x/z}} \quad (4)$$

In the same way if we define $t_{y_1} = t_{y_0} + t_c$, we get a new constraints to t_{z_0}

$$t_{z_0} + t_{d_{y/z}} < t_{y_1} - t_s \quad \vee \quad t_{z_0} + t_{d_{y/z}} > t_{y_1} \quad (5)$$

However, equations 3 to 5 are not valid to $t_{d_{k/m}} > t_c$. Meaning, if the flight time of an acoustic signal is larger than the maximum rate of acoustic signals emission. In this case the equations will not be compatible with equations 1 to 2, because $t_{m0} < t_0$. They should then be rewritten as,

$$(int(\frac{t_{d_{x/y}}}{t_c}) + 1) \times t_c - t_{d_{x/y}} < t_{y_0} < (int(\frac{t_{d_{x/y}}}{t_c}) + 1) \times t_c - t_s - t_{d_{x/y}}$$

$$(int(\frac{t_{d_{x/z}}}{t_c}) + 1) \times t_c - t_{d_{x/z}} < t_{z_0} < (int(\frac{t_{d_{x/z}}}{t_c}) + 1) \times t_c - t_s - t_{d_{x/z}}$$

$$(int(\frac{t_{d_{y/z}}}{t_c}) + 1) \times t_c - t_{d_{y/z}} < t_{z_0} < (int(\frac{t_{d_{y/z}}}{t_c}) + 1) \times t_c - t_s - t_{d_{y/z}}$$

In which $int(\frac{t_{d_{k/m}}}{t_c}) + 1$ represents the first acoustic signal emission from k that will be affected by the emission of an acoustic signal emitted by m.

With these constraints we are prepared to produce the algorithm that controls the emission of acoustic signals of each element of the network. At this point we can assume each node emits acoustic signals at moments known to the remaining nodes without signal collision, and we can proceed to the signal peak detection with no concerns about that subject. As said before the signal peak detection is achieved through a first order iterative method. The successive bisections method was used because of its reliability and fast progression that can be optimized with a wise choice of starting bounds. The production of logs, now possible with the automation of the procedure will allow us to optimize those bounds and even evolve to a higher order method that will ensure an even faster progression.

This method was implemented with an iterative process dependent on an evaluation of the level of trust in the signal value, based in a two to one voting scheme, therefore sending three acoustic signals for each iteration of the bisections method.

The algorithm starts by defining an interval $[N_1, N_2]$, where N_1 is a levels where there isn't a signal detection and N_2 a level where there is a signal detection. Afterwards an intermediate interval $N_3 = \frac{N_1+N_2}{2}$ is tested, unwinding from there with the mechanism of the typical bisection method, until an end condition is reached.

This procedure is executed with all the nodes of the network active, for all of its frequencies. These frequencies are allocated to each node of the network through a high level application running on one of the nodes. The application assigns the frequencies according to the following equation.

$$Time + min(t_{d_{k/m}}) > t_{ls} + max(t_{d_{k/m}})$$

This means that a frequency is assigned to a node if the current time plus the shortest distance to another node is bigger than the last time t_{ls} that frequency was sent plus the largest distance to another node. In other words, ensures only one signal at a given frequency arrives to a node at a given time.

V. Navigation Algorithms

On a typical LBL navigation scheme, the AUV interrogates an acoustic beacon and determines the range to it by waiting for a specific reply. Although a navigation filter may integrate each range independently, a complete update of the absolute position may only be computed by triangulation when the AUV measures ranges to at least 2 different beacons (providing that the AUV always remains in the same side of the *baseline*). This means that for ranges of a few kilometers, the position updates are only available every few seconds. In the meantime the navigation filter updates the position estimate based on other data (typically from IMU, doppler velocimeter or odometry), but the uncertainty in position will always grow between acoustic fixes.

For a given deployment of acoustic beacons, there are basically two ways to increase the update rate of acoustic ranges onboard an AUV and therefore reduce the growth of the position uncertainty. The first is simply to send multiple interrogations from the AUV using different acoustic codes, without waiting for every reply from the beacons. The vehicle will have to maintain a list of pending replies and estimate ranges to a beacon each time there is a match between interrogation and reply. Although the delays will always be dependent on range, the steady state update rate will be as high as the interrogation rate from the AUV. Even though the complexity of this algorithm may be manageable by the AUV, the cluttering of acoustic signals in the water may result in degraded performance due to interferences, particularly in multipath environments. Furthermore, this potential increase in update rate is achieved with an increased burden in power consumption, which is always a major concern in AUVs.

Another way of improving LBL navigation is to have the beacons sending pings synchronously and compute ranges just by one-way time of flight, such as used in [10], for example. This will provide much more measurements to the filter but it requires the AUV internal clock to be synchronized with the internal clocks of the beacons. With our synchronized intelligent buoys, all clocks can be globally synchronized before the mission, with negligible drifts during the time frame of a few hours. Note that as far as navigation is concerned, the AUV becomes a passive device, just listening to acoustic signals sent at predetermined instants, and therefore saving power as compared to any solution based on AUV interrogations. More, this synchronized scheme allows for the operation of virtually any number of AUVs in the same operational area.

VI. Conclusions

The estimation of position is a permanent research topic in underwater robotics and acoustics play a fundamental role by providing absolute measurements of distances with bounded accuracy. In order to support the development and implementation of different approaches to AUV navigation, it is critical to use a versatile platform which allows the configuration of the systems parameters and access to all available information regarding acoustic signals. It is also important to manage this information remotely and to obtain complementary data about the environment during operational missions. In this paper, we presented a prototype of a synchronized intelligent buoy that, meeting these constraints, takes the acoustic buoys one step closer to an integrated node of a portable coastal observatory. This new system has a low power SBC running Linux that provides support to a wide range of oceanographic or meteorologic sensors and also provides real time interface with all the information regarding the acoustic boards. One of the buoys main features is the ability to synchronize to a global clock ensuring the possibility of operations with time constraints. Another benefit of clock synchronization (of both AUVs and buoys) is to provide the basis for one-way travel time acoustic navigation. This will endow multiple vehicles an increased number of acoustic ranges, as compared to traditional LBL navigation.

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