Simultaneous Round-Trip Time-of-Flight Measurements With Encoded Acoustic Signals

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***Abstract*— In relative positioning systems, with the aim of estimating object positions, distances among them are computed in a cooperative way, usually by measuring times-of-flight from the signals that they emit. These emissions are often synchronized with additional signals or suitable hardware that acts as a temporal reference. In this paper, a ranging system is presented where only acoustic emissions are used to compute the distances between objects or nodes. Thus, an organization and operation algorithm is proposed, which provides a temporal reference to the acoustic emissions carried out by every node. In this way, distances are computed by determining the temporal relation between a request of emission from a coordinator node and the corresponding answers emitted by the other nodes. In order to simultaneously detect the acoustic emissions, the signals are encoded with complementary set of sequences allowing multisensory operation and accepting low signal-to-noise conditions. With this measurement scheme, additional signals and high accuracy clocks often used for synchronization can be eliminated, thus reducing hardware complexity, power consumptions, and possible interferences with other systems (i.e. if radio frequency signals are used). The simulation and experimental results show that sub-centimeter accuracy can be obtained with the proposed ranging scheme.**

***Index Terms*— Acoustic signal processing, distance measurement, direct-sequence code-division multiple access, relative localization, sensor network.**

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I. Introduction

RELATIVE localization techniques [1] require the design of sensor architectures which allow computing the distances among the objects in a low scanning time, before the environmental conditions or the relative position among nodes change. This means that an active cooperation and simultaneous operation among nodes is necessary to collect and distribute the distance information. The signals or the physical channels used to obtain the spatial relations depend on the application, power consumptions and size constraints of systems. In the case of indoor localization systems [2], ranging mechanisms often use radio frequency (RF), or acoustic signals.

When RF signals are used as sensory technology, two different techniques can be applied to measure the distance among objects. Some determine the received carrier signal level, called RSSI (Receive Signal Strength Indicator). The distance is computed by assuming a channel attenuation model where the level of the emitted signal usually decreases with distance. This attenuation model strongly depends on the environmental conditions; also, the emitted power cannot be accurately determined, so the estimation of the real signal attenuation becomes difficult [3]. On the other hand, the distance measurements can be computed from the times-of-flight of the emissions between pairs of objects by assuming a channel propagation speed for RF waves. This principle of measurement is less dependent on environmental conditions and more accurate than the previous ones. Three different versions of this method can be applied, depending on the used synchronization scheme: direct measurement of time-of-flight (TOF), round-trip time-of-flight (RTOF) or difference in times-of-flight (DTOF). Nevertheless, the aspects related to timing and synchronization of RF systems should be carefully taken into account, since the high propagation speed of electromagnetic waves extremely increases hardware complexity [4].

Acoustic ranging are widely used in localization systems [5]–[9], taking advantage of low cost, small sizes and good features in emission and reception patterns of sensors, even for those frequencies slightly over the audible band. Among their drawbacks, acoustic signals are influenced by solid objects, which can obstruct the emissions providing errors in measurements if non line-of-sight (NLOS) conditions appear. In order to mitigate this error, high-level strategies should analyze the geometric coherence of the global information collected by the sensor systems.

The relative positioning of objects by means of acoustic emissions allows to obtain a good accuracy and precision by using small and low-cost sensory nodes [1]. The cooperative mobile robot teams are a good example of relative positioning in indoor spaces. In [10], [11] the TOF measurement of the ultrasonic signals emitted by every node is used to compute the distances between the team members. These ultrasonic signals are synchronized by means of a RF link. In the DOLPHIN system [12], a ranging mechanism based on ultrasonic TOFs is proposed, also synchronized by RF signals. Fixed beacons are used with the purpose of determining the absolute positions by measuring the distances from the objects to them. Those nodes, which are in NLOS with the reference beacons, can determine their positions by measuring the distances with the other nodes previously located and then used as new reference beacons.

The distributed mobile computing systems are a new field of application for relative positioning systems extended during last years; e.g. in [1] a localization system for general-purpose computers (GPC), such as notebooks or personal digital assistances (PDAs) has been developed. In this case a sensor node, which carries ultrasonic transducers, is connected through USB ports to the computer. The distances among objects are computed by measuring the TOF and also the angle of arrival of the emitted ultrasonic signals. These sensory modules also perform inter-node communications to distribute the data collected by all of them. In [9], [13] localization systems for mobile computers are described. In these cases, the acoustic and RF transducers included in GPCs or PDAs are used as sensory system. By means of TOF or DTOF measurements, the distances among the objects can be computed. The synchronization algorithms become important due to the computational load, software, and hardware latency associated to mobile devices.

In addition to the considerations about the ranging mechanism in location systems, it is also remarkable to detect simultaneous emissions transmitted from different independent sources, with reduced interference among them. In many cases [7], a binary encoding scheme, called Direct Sequence Code Division Multiple Access (DS-CDMA) [14], is used to discriminate the emissions carried out simultaneously by the sensors or nodes. In this scheme it is necessary to be careful with the signal processing algorithms, trying to avoid an increase in the complexity of hardware to be implemented.

In last years, the researchers efforts have been focused on the development of localization systems for those applications that are not restricted to a particular environment. In this field, several technical considerations should be solved, especially in hardware architectures. For example: the use of only one kind of signals as sensory technology without requiring another support system; the encoding scheme of multi-user signals for suitable works in different conditions; as well as the development of inter-node coordination algorithms and communications. Taking these facts into account, the main goal of this work is the design of a low-complex ranging

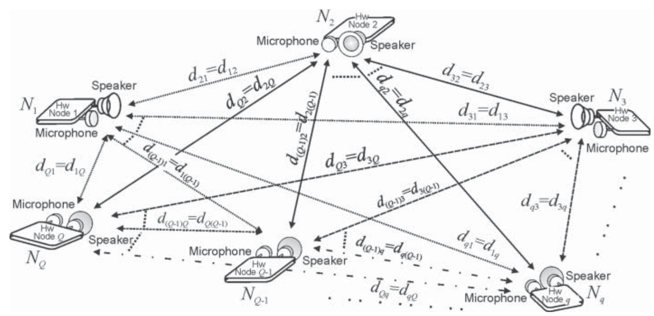


Fig. 1. Global architecture of the proposed relative positioning system.

mechanism which allows computing the distances between objects using only acoustic emissions. The main features of the proposed ranging mechanism can be summarized as follows.

So as not to include additional RF signals to synchronize the acoustic emissions, the round-trip times-of-flight among acoustic emissions have been computed to determine their distances. This technique is often used with RF signals [15] because it makes easier the synchronization of reception stages.

Since the time required by the RTOF method increases whether acoustics emissions are used, this method is simultaneously performed among the nodes in the system, and thus it has been called SRTOF (Simultaneous Round-Trip-Time-of-Flight). An organization and coordination algorithm has been proposed to provide a temporal reference system to the acoustic emissions emitted by every node.

Finally, in order to simultaneously detect the emissions from every node, the acoustic signals have been encoded according to a CDMA scheme based on Complementary Sets of Sequences [16]. In this way, it is also possible to have several users active at the same time in the system.

The rest of the paper is organized as follows: in Section II, the proposed system architecture is described, as well as the signal processing algorithms used to simultaneously detect the encoding emissions. Section III explains the method proposed to compute the distances between objects using only acoustic emissions; afterwards the ranging mechanism is analyzed. In Section IV some simulations and experimental results are described and, finally, some conclusions are discussed in Section V.

II. Proposed System

A. Global Overview

The architecture of the proposed relative positioning system is depicted in Fig. 1. It consists in an acoustic sensor network, where every node/object $N_q(q \in \{1, 2, \ldots, Q\}$ and Q is the maximum number of nodes in the system) carries a sensor node with a speaker, a microphone and the associated hardware.

The outline of the algorithms proposed to compute the positions among objects is shown in Fig. 2. The computation of the distances $d_{ql}(q, l \in \{1, 2, \ldots, Q\} q \neq l)$ between objects is performed by measuring the propagation time of acoustic signals emitted by every node without any additional link to synchronize them. The measurement principle is based on

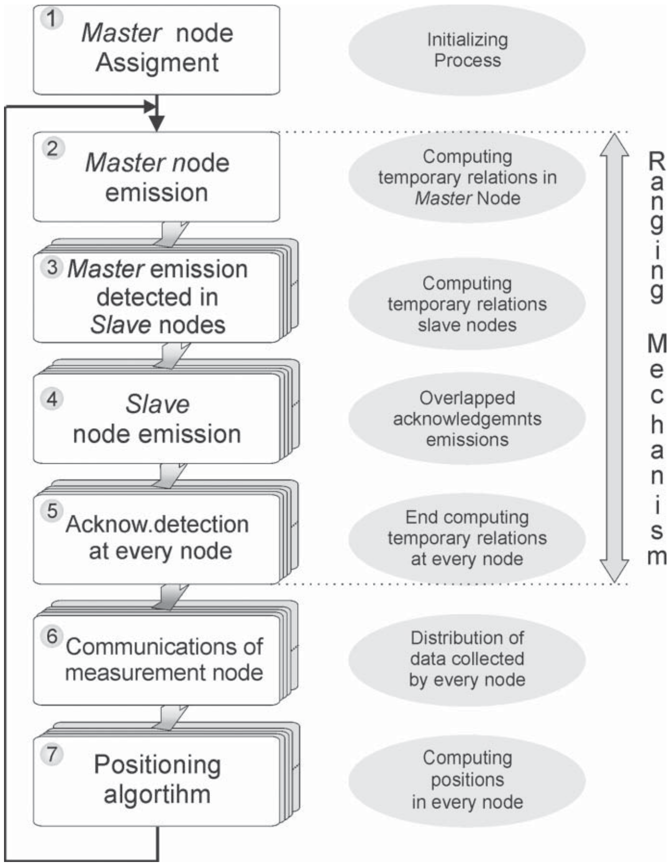


Fig. 2. Outline of the algorithm proposed to compute the relative positions among objects using only acoustic emissions in the ranging mechanism.

simultaneously computing the RTOF, between a coordinator node (*Master node*) and the other ones (*Slave nodes*).

For that, it is necessary a first stage where the *Master* condition is assigned to a given node (see step 1 in Fig. 2). Afterwards, taking advantage of the simultaneous signal detection achieved with the CDMA techniques, the *Master* node can compute temporal relations with every *Slave* node (see steps 2 to 5 in Fig. 2). Additionally, the *Slave* nodes can compute temporal relations among their emissions from the detection of *Master* request (steps 3 to 5 in Fig. 2). This process has been called S-RTOF *(Simultaneous Round-Trip-Time-of-Flight)* and is explained with more detail in Section 3. Since the information is obtained between pairs of objects, it is necessary to distribute these data, i.e. by using low-cost communication modules [17] (step 6 in Fig. 2). Finally, the last step is the computation of the positions among objects using the collected and distributed data [18], what can be carried out at every node or in a centralized system.

B. Encoding of Acoustic Emissions

DS-CDMA techniques are used to discriminate the node emissions, since every user has a unique binary code or sequence which is transmitted by a simple phase modulation. These encoded signals are detected in a receptor by performing the correlation with every available sequence in the proposed system. An encoding scheme based on Complementary Set of M Sequences ($M-$CSS) has been

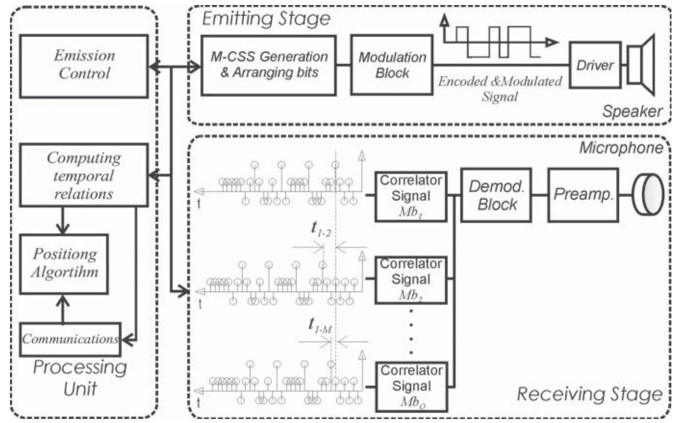


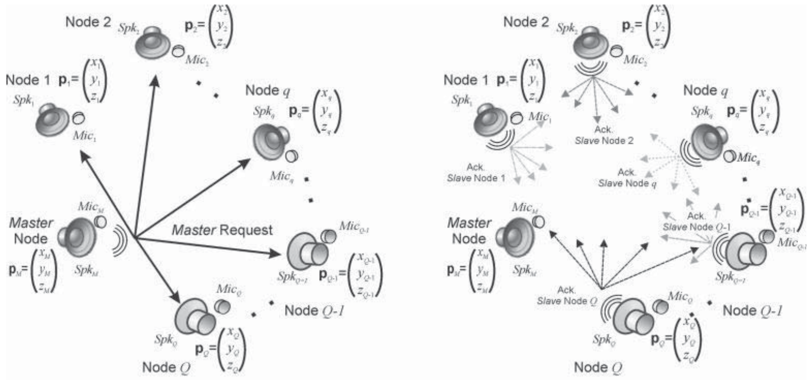
Fig. 3. Detailed block diagram of node hardware architecture.

used, where the number of sequences M is a power of two [16]. In that case, M sets can be obtained with null cross correlation, by adding the cross-correlation functions (*SCCF*) between the corresponding sequences of two sets. Furthermore, the addition of the auto-correlation functions (*SACF*) for every sequence from a set provides null sidelobes. The described properties make attractive their use in sensory systems where several simultaneous emissions are carried out from different independent sources with low signal-to-noise ratios (SNR). As opposed to the common encoding schemes used in localization, the encoding of emissions by M-CSS assigns more than one sequence to every user. In this way, the required hardware resources are increased, reason why efficient algorithms have been developed in order to minimize these requirements [19]. Also, it is necessary to analyze the most efficient and effective technique to transmit in a short time the sequence of bits used to encode every user. One simple method consists of establishing an emission order of the $M-$CSS bits and transmitting the bitstream by a BPSK modulation. According to these signal processing techniques, the emitting stage implemented in the node hardware architecture is shown in Fig. 3. This block is divided into the code generation stage for the M-CSS, the bit arranging stage and BPSK modulator. Furthermore, an emission control block is implemented to fire the acoustic signal emission, managed by the processing unit. In the reception stage, the demodulation of the signal captured by the microphone is computed after adjusting and digitalizing it. Then, the signal is processed to determine the arrival time for every acoustic emission. For that, Q correlators have been implemented according to [19], as well as the corresponding peak detectors.

Finally, a processing unit has been implemented at each node, which controls the emitting stage according to the results obtained at the correlator outputs. It also computes the temporal relations among the emissions from every node. Moreover, this unit distributes the data collected by each node with a communication block, and it computes the position among the objects.

III. S-RTOF MEASUREMENTS

Here, the ranging mechanism proposed in Fig. 2 is deeply discussed. The temporal schedule for each emission and



(a) (b)

Fig. 4. Principle of measurement based on simultaneous RTOF to compute the distances between nodes. (a) *Master* request emission. (b) Acknowledgment at *Slave* nodes.

reception is depicted to obtain the equations that describe the temporal relations among the transmissions from every node.

A. Master Assignment

The first step in the proposed scheme (see step 1 in Fig. 2) consists in the selection of the node in charge of starting or coordinating the measurement process, which is called *Master* node. Since all the nodes are equal in their hardware architecture and functionality, anyone of them can start the measurement process. Therefore, in order to determine which node takes the control, a Carrier Sense Multiple Access technique (CSMA) has been used. The complexity of this stage can be reduced by a pre-assignation of *Master* condition according to the particular features of each node. After establishing the *Master* condition, the ranging mechanism starts and does not stop until the assignment of a new *Master* node becomes necessary.

B. Principle of Measurements

For a set of Q nodes distributed in a certain environment (see Fig. 4), let assume that the *Master* node (consider without loss of generality the node $q = 1$) has a position with coordinates (x_M, y_M, z_M), described by the vector $\mathbf{p}_M$. On the other hand, the *Slave* node positions are given by vectors $\mathbf{p}_q$ (where $q \in 2,\ldots,Q$ and Q is the maximum number of nodes in the system). At a given instant of time T_S, the *Master* node emits its coded acoustic signal, called *Master Request*, through its speaker denoted as Spk_M (Fig. 4.a). At T_S the time computation starts until the detection of the *Slave* node emissions.

The *Master Request* is detected at every *Slave* node, by means of their microphones Mic_q, at different time instants according to their distribution in the environment. In response to this request, every *Slave* node transmits its characteristic code, called *Ack. Slave Node*, which travels towards the *Master* and also towards the rest of *Slave* nodes (Fig. 4.b). Based on this strategy and taking advantage of the used coding scheme properties, it is possible to compute temporal relations in the *Slave* nodes from the detection of the *Master Request* until the detection of every *Ack. Slave Node* from the other *Slave* nodes. These temporal relations are called pseudo-time-of-flight (pTOF) [20] since they are not a direct TOF measurement of the signals emitted by the nodes.

C. Measurement Analysis

The measurements carried out by the S-RTOF algorithm depend, not only on the distances between objects, but also on some intrinsic parameters, that perturb the measurements. It is necessary to analyze the pTOFs measured at every node, where possible delay sources can be considered in order to mitigate their effect on the distance computation and to improve the accuracy of the node position estimation. In this case, for the measurement of the pTOFs, it is necessary to consider the delays introduced by the algorithms proposed for transmitting and detecting the acoustic signals [13]. Particularly, there is a delay during the transmission from the start time of pTOF computation, until the signal is emitted by the node transducer. This delay is provided by the hardware that generates the signal to be transmitted, and also by the speaker response time. In the same way, in the reception stage there is a delay associated to the microphone response and to the detection of the coding assigned to each user (sampling frequency, latency of signal processing algorithms, etc.). In next subsections, the two types of measurements computed by the proposed system are analyzed by the timing diagrams of the receptor outputs (see Fig. 5). The pTOFs are computed considering the *Master* node and two generic *Slave* nodes q and l, where $q, l \in \{2,\ldots,Q\}$.

1) Measurements in Master Node: The *Master* node measures the pTOFs between two generic *Slave* nodes and itself in this way (see Fig. 5 for details at each instant of time):

a) The process starts with the *Master Request* emission (instant A). It is necessary to consider the delay for the emission stage, denoted as $tspk_M$.
b) The *Master Request* is detected at the *Slave* nodes at instants B (node q) and C (node l). The delays in the microphones' response, $tmic_q$ and $tmic_l$, are also taken into account.The TOF between A and B required by the *Master Request* signal to reach the *Slave* node q is given by $TOF_{M-q} = ||\mathbf{p}_M\text{-}\mathbf{p}_q||/c$, where c is the speed

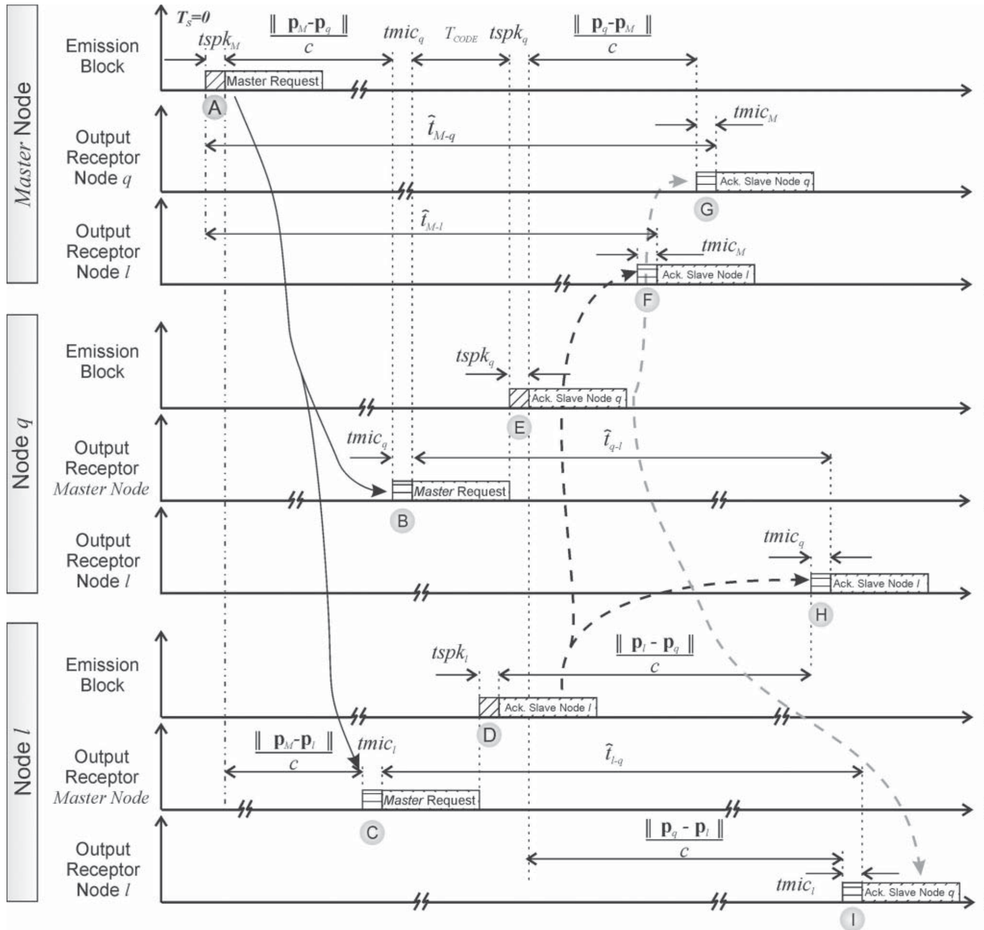


Fig. 5. Timing schedule of emissions and receptions among nodes to simultaneously compute temporal relations by means of S-RTOF.

of sound. And, similarly, for node l, $TOF_{M-l} = ||\mathbf{p}_M$-$\mathbf{p}_l||/c$.

c) The *Master Request* is validated at the *Slave* nodes, and an acknowledgement is sent by them: *Ack. Slave node* at instants D (node q) and E (node l). Now, their emission delays ($tspk_q$ and $tspk_l$) are also considered.

d) These acknowledgement emissions travel towards the *Master* where they are detected at instants F (from node q) and G (from node l). The reception delay $tmic_M$ must be also considered.

According to the four step previously considered, the pTOF computed from the *Master* node emission until the detection of the *Ack. Slave Node q* can be described as (1)

$$\hat{t}_{M-q} = tspk_M + \frac{\|\mathbf{p}_M - \mathbf{p}_q\|}{c} + tmic_q + tspk_q + T_{CODE} + \frac{\|\mathbf{p}_q - \mathbf{p}_M\|}{c} + tmic_M \quad (1)$$

where $\hat{t}_{M-q}$ is the pTOF estimated at the *Master* node in relation with the *Slave* node q, $q \in \{2,\ldots,Q\}$; and T_{CODE} is the duration of the emitted code.

Assuming that the speed of the nodes is lower than the propagation speed of the acoustic waves, it is possible to compute the distances between the *Master* and every *Slave* node rearranging (1), as follows:

$$d_{M-q} = \|\mathbf{p}_M - \mathbf{p}_q\| = \frac{[\hat{t}_{M-q} - (tp_M + tp_q + T_{CODE})]}{2} \cdot c \quad (2)$$

where tp_M and tp_q are the signal processing times required at the emission and reception stages for the *Master* node and the *Slave* node q, respectively; and $tp_M = tspk_M + tmic_M$ and $tp_q = tspk_q + tmic_q$.

2) Measurements in Slave Nodes: Considering that the *Slave* nodes are capable to detect the emissions carried out by them towards the *Master* node, it is possible to measure temporal relations among the *Slave* node emissions.

For that, this process is followed (also refer Fig. 5 for timing instants and intervals):

a) In each *Slave* node it is carried out the computation of the time from its own *Master Request* detection (instant B for node q or instant C for node l) until the detection of the other *Ack. Slave Node*, i.e. instants H (detection of *Ack. Slave Node l* in the node q) and I (detection of *Ack. Slave Node q* in the node l).

b) It is possible to compute the pTOFs $\hat{t}_{q-l}$ between two *Slave* nodes, taken into account that the transitions

A-B-H and A-C-D-H must be equal

$$tspk_M + \frac{\left\|\mathbf{p}_M - \mathbf{p}_q\right\|}{c} + tmic_q + \hat{t}_{q-l} = tspk_M + \frac{\left\|\mathbf{p}_M - \mathbf{p}_l\right\|}{c} + tmic_l + T_{CODE} + tspk_l + \frac{\left\|\mathbf{p}_l - \mathbf{p}_q\right\|}{c} + tmic_q. \quad (3)$$

Therefore, the following expression is obtained:

$$\hat{t}_{q-l} = \frac{\left\|\mathbf{p}_M - \mathbf{p}_l\right\|}{c} + \frac{\left\|\mathbf{p}_l - \mathbf{p}_q\right\|}{c} - \frac{\left\|\mathbf{p}_M - \mathbf{p}_q\right\|}{c} + mic_l + tspk_l + T_{CODE} \quad (4)$$

where $\hat{t}_{q-l}$ is the pTOF measured in the node q from the instant in which it detects the *Master Request* until it detects the Ack. *Slave* Node l; in general $q, l \in \{2,\ldots,Q\}$ and $q \neq l$.

With the data collected by all the *Slave* nodes, it is possible to compute their relative distances from (4). Again, assuming that the moving speed of nodes is lower than the sound speed, these distances are obtained as follows:

$$d_{q-l} = \left\|\mathbf{p}_l - \mathbf{p}_q\right\| = \frac{c}{2} \cdot \left[\hat{t}_{q-l} + \hat{t}_{l-q} - tp_1 - tp_q - 2T_{CODE}\right] \quad (5)$$

Another way to obtain these distances derives of the use of (4) combined with (2), so

$$d_{q-l} = c \cdot \left[\hat{t}_{q-l} - (tmic_1 + tspk_1 + T_{CODE})\right] - d_{M-l} - d_{M-q} \quad (6)$$

IV. Performance Evaluation

In order to test the proposed architecture, some simulations have been carried out considering the S-RTOF ranging mechanism as well as the encoding scheme. Furthermore, an exhaustive analysis by means of Monte-Carlo simulations has been done to determine the accuracy obtained with the proposed architecture. Finally, some experimental tests have been performed to verify the obtained results.

A. SRTOF Simulations Results

The performance of the algorithms proposed to determine the pTOFs has been verified with six nodes distributed in an environment (see Fig. 6). Every node has a complementary set of four sequences ($M = 4$) with length $L = 16$ to encode the emissions; these are transmitted by using a BPSK modulation with a carrier frequency $fe = 33$ kHz. Two kinds of speakers have been considered for simulation: one of them with ideal frequency response and another with a more real acoustic frequency response centered at 33 kHz and a bandwidth of 20 kHz. The microphone has been considered with a flat frequency response.

The 4-CSS are emitted three times to reduce multipaths effects by checking the distances between successive correlations peaks in the receiving stage. In this case, the *Master* condition is assigned to the node N_1 and a Gaussian noise with SNR = 0 dB is considered. At the reception stage of each node, the signals captured by the corresponding microphones are digitalized, demodulated and correlated. Finally, a peak detector determines the instant of arrival of the encoded signals and the successive transmissions are validated by checking the intervals between them.

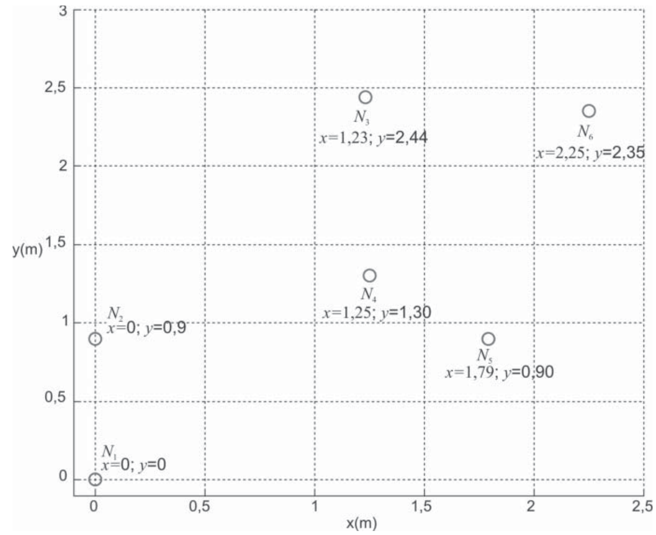


Fig. 6. Distribution of nodes in an environment to simulate the S-RTOF scheme.

Figure 7.a shows the results obtained at the correlation outputs in the *Master* node and Fig. 7.b shows the results obtained in the *Slave* node N_5 when real speakers are considered. As can be observed, it is possible to compute the temporal relations among the emissions carried out by every node under low SNR conditions. Using the times measured in the nodes N_1 and N_5 (see Fig. 7) the distances are computed assuming that all the nodes have communicated the temporal relations measured by each one to the others. Also, the signal processing times are roughly known and time invariant.

B. Monte Carlo Simulations

Taking into account the features of the proposed architecture in Section IV.A, a Monte-Carlo analysis of the performance in the pTOF computation has been carried out. According to the topology of nodes described in Fig. 6, the signals emitted by every speaker have been corrupted with zero average additive white Gaussian noise in the range of SNR = 0 dB and SNR = −3 dB, respectively. The computation of pTOF has been tested over 1000 times. Also, it is assumed that the nodes have communicated the data collected by each one of them.

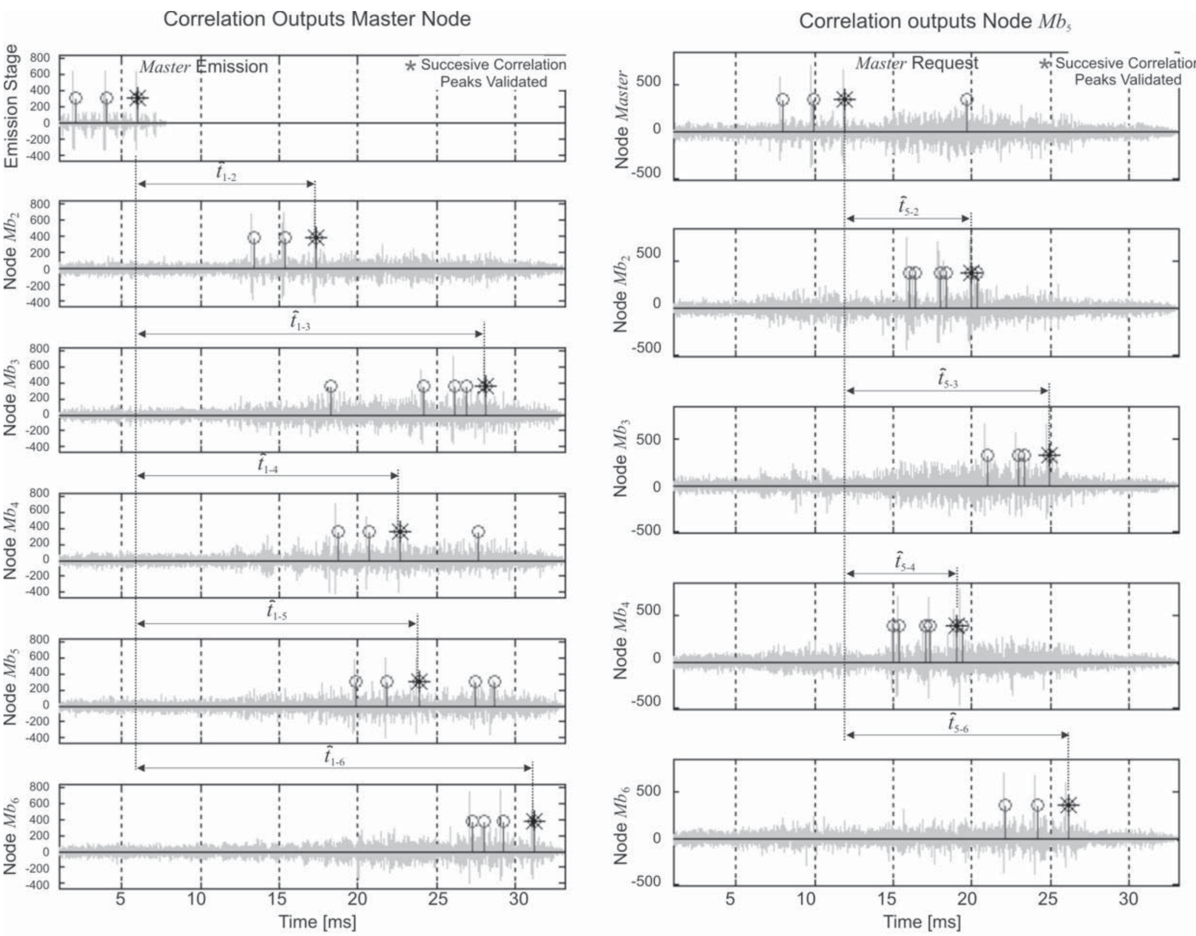
The results obtained by Monte-Carlo simulation in the computation of the distances between the *Master* and the *Slave* nodes show that these distances can be successfully computed with a percentage of 99% considering SNR = 0 dB; and of 95% in the case of SNR = −3 dB. On the other hand, considering the distances between *Slave* nodes, these distances can be correctly computed in 90% of cases with SNR = 0 dB using (5); and 80% with SNR = −3 dB. Never the less, a higher success in the distance computation can be obtained with (6), which allows the distances to be correctly determined in 99% of cases with SNR = 0 dB and 90% with SNR = −3 dB.

TABLE I

ACTUAL DISTANCE (d^r) VERSUS MEASURED DISTANCE (d^m) IN THE MASTER NODE AND IN THE *Slave* NODE N_5 WITH THE OTHER NODES OF THE SYSTEM, BY MEANS OF S-RTOF MECHANISM WITH SNR = 0 dB OBTAINED BY SIMULATION IN DIFFERENT CONDITIONS

Node q to Node l			N_1	N_2	N_3	N_4	N_5	N_6
N_1	Actual d^r_{ql} [cm]		**0**	**90**	**273,25**	**180,35**	**200,35**	**325,35**
	Measured $d^m 1l$	Average	0	90.07	273.36	180.44	200.43	325.48
	[cm] (1)	Std. dev.	0	0	0	0	0	0
	Measured $d^m 1l$	Average	0	89.8	273,09	180,08	200,23	325,16
	[cm] (2)	Std. dev.	0	4.93	8.64	8.07	6.34	10.29
N_5	Actual d^r_{ql} [cm]		**200,35**	**179**	**163,87**	**67,201**	**0**	**152,12**
	Measured $d^m 5l$	Average	200.43	179.21	164.12	67.42	0	152.29
	[cm] (1)	Std. dev.	0	0	0,042	0.01	0	0
	Measured $d^m 5l$	Average	200,23	178.49	163.95	67.19	0	151,98
	[cm] (2)	Std. dev.	6.34	11.32	5.19	3.69	0	7.95

(1) Ideal-response transducers.
(2) Real-response transducers.



(a) (b)

Fig. 7. Simulation results at the correlation outputs by means of SRTOF ranging mechanism. (a) Correlation outputs for *Master* node. (b) Correlation outputs for *Slave* node N_5.

Table 1 describes describes the results obtained in the computation of the distances between the *Master* node and every *Slave* node, and the distances computed between one of the *Slave* nodes, N_5, and the others (only this representative case has been considered and the distances among other nodes have not been included not to enlarge the table). The obtained results show that a suitable accuracy can be achieved with ideal and real transducers. A highest variability of distance

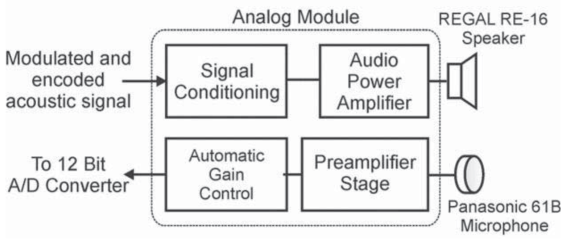
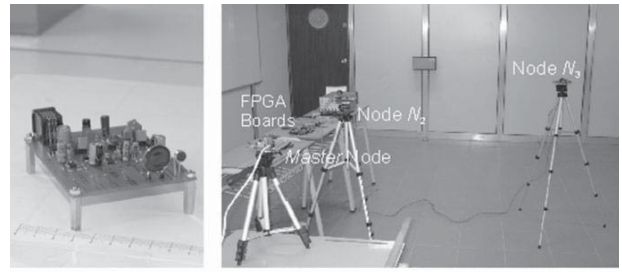
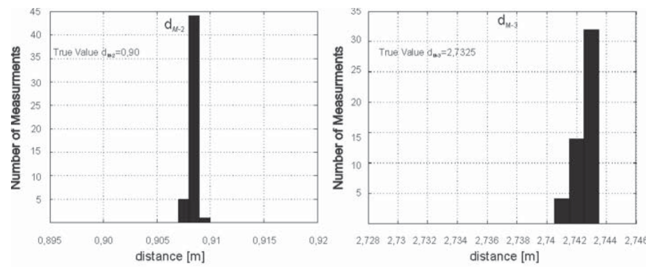


Fig. 8. Block diagram of module for pre-processing analog transducer signals.



(a) (b)

Fig. 9. Hardware implemented for SRTOF ranging test. (a) Sensor node implemented by acoustic commercial transducers. (b) Experimental test with three nodes.



(a) (b)

Fig. 10. Experimental results obtained with SRTOF on distance computation. (a) Distance computation between *Master* and node 2. (b) Distance computation between *Master* and node 3.

computation appears when a real transducer response is used, due to the effects of transducer bandwidth constraints. It may produce higher values in the side-lobes of the correlation function and in some cases a wrong validation of a peak that does not correspond to the central one.

C. Experimental Results

Experimental tests have been carried out in order to verify the accuracy obtained in the position estimation with the proposed sensory network architecture and signal processing algorithms. According to the description of the node hardware architecture in the previous section, the proposed signal processing algorithms have been implemented in a board based on Xilinx XC3S1200E FPGA [21]. The node architecture implemented in the FPGA board consists of a transmitter block, a receiver block with the corresponding A/D converter, and, finally, the block required to computethe temporal relations among emissions. Afterwards, results are sent through an USB port available in the FPGA board to a central unit to analyze the obtained data.

Hardware related to pre-processing of analog transducer signals (transducer driving and signal amplification) has been implemented in additional sensor boards (see Figs. 8 and 9a). The speaker REGAL R-16-E [22] is used as emitter. This is oriented to general-purpose applications and mobile computing devices; it has suitable sound pressure level (SPL) at frequencies above 15 kHz, where acoustic signals begin to be imperceptible for human ear.

In the reception stage, a Panasonic electret microphone 61 B has been used [23]. This presents small size, omnidirectional pattern reception and a flat response at audible frequencies and even higher than 20 kHz.

Experimental tests have been performed by using a BPSK modulation with a squared carrier symbol, whose emission period is 30 μs (carrier frequency $fe = 33.33$ kHz). In the receiver system, the acquisition frequency $f_S = 333.33$ kHz implies an oversampling $Os = 10$. In this case, a *Master* node and two *Slave* nodes have been developed and deployed in an environment as described in Fig. 9.b with coordinates $(x; y) : N_1(0; 0)$; N_2(0;0,9m) and N_3(1,23m; 2,44m).

Tests have been performed for 100 trials and the results show the lowest variability in the distance computation in the *Master* node with every *Slave* node (see Fig. 10a and 10b). In Fig. 10, a bias appears in the distance computation due to transducer bandwidth constraints and because a fixed value for the different signal processing delays has been considered, which can differ from the values in the experimental set-up.

Tests have been carried out under laboratory conditions; nevertheless, the sound speed must be temperature and pressure compensated [24] in order to convert temporal relations into distances in real conditions.

V. Conclusion

In this work, a ranging mechanism has been presented where only acoustic emissions are used to determine the spatial relations among objects, making thus unnecessary the use of additional signals to synchronize the emission of the acoustic signals. To coordinate the emissions, an algorithm has been developed, where the temporal relations between the emissions from the nodes are computed on demand from one of them that works as coordinator of the measurement process. The measured temporal relations are called pseudo-time-of-flight since they are not a direct measurement of times-of-flight. With the pTOFs, it is possible to compute the distances between objects. In the pTOF determination, it is necessary to take into account the delays related to the signal processing algorithms and hardware implementation in order to reduce the error in the distance computation. Finally, the robustness of the proposed architecture has been tested by both simulation and real experiments, verifying a satisfactory performance even under low SNR conditions (SNR $= -3$ dB).

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