WIRELESS SENSOR NETWORK-BASED INSTRUMENT FOR CAPTURING SOUNDSCAPE

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1 INTRODUCTION

This paper presents ongoing work on the development of acoustic instrumentation as part of a joint research project comprising Southampton, York and Newcastle Universities. The instrumentation developed in the research will be used to capture acoustic signals from indoor and outdoor measurements. The captured signals will be undergo analysis and processing to provide useful information in relation to the characteristics of the source and environment for industry. University of York is developing the source identification and separation algorithms and Southampton is undertaking the work on sound recording database and evaluation for noise legislation. Each element of the project is carried out individually at each University, as well as joint experimental work amongst the group members. All of the work will be incorporated at the final stage of the project by the evaluation of the developed system.

The use of distributed instrumentation such as sensor networks in measurement applications has been in existence for decades, motivated by military applications. For instance, research on sound source imaging, has provided a way to detect and track aircraft flying too low to be detected by radar, over areas of concern such as along international borders [1,2]. Also, the success of the hostile artillery locator (HALO) system, a project developed for the UK's Ministry of Defense [3], to locate enemy weaponry through sound source localisation is another excellent example of the use of acoustic sensor networks in military applications. This mobile system has the capability of detecting and locating acoustic sources coming from tanks or mortars as far away as 30 km. These examples show that there are useful applications, which use instrumentation systems for sound capturing and measurement of the soundscape of a particular area.

Soundscape, an emerging term used to describe the acoustic environment of a region, is an important piece of information; an example could be to identify noise problems for the particular region. To understand fully the soundscape of a region of interest it is desirable that sound measurements across the region take place simultaneously and continuously, acquiring data both during the day and at night, as there are differences in sound propagation behaviour at those times. This would ensure an effective representation of the soundscape in the region. Another challenging aspect in simultaneously measuring the sound field in a large outdoor area is the use of arrays of acoustic probes. Thus, distributed acoustic measuring systems comprising an individual processor on each of its elements controlled by a main processor can be an appropriate contender for such an application. The implementation of the distributed system also applies to the measurement of an indoor acoustic environment, which is generally less demanding than its outdoor counterpart, having environmental parameters which are more controllable. Indeed, recent developments in wireless sensor network technology allow us to meeting such requirements. In addition, the coverage of larger measurement areas with the implementation of such technology becomes possible.

2 ACOUSTIC INSTRUMENT AND WIRELESS SENSOR NETWORK STATE-OF-THE-ART

There are various acoustic measuring instruments available, such as hand-held sound pressure level (SPL) instruments. Brüel and Kjær are one of the leading commercial producers of sound and vibration measurement instrumentation. They have introduced the latest hand-held SPL instruments with the added flexibility of included application software. This sophisticated SPL meter functionality can be upgraded to suit the required applications. This hand-held SPL meter can have facility to analyse room reverberation as well as measure the source's sound intensity. Using these instruments with the use of Global Positioning System (GPS), one could measure noise at several different points simultaneously within an area, similar to the work introduced by Madrid City Council for their urban noise measurement and mapping system [4]. Another future trend in acoustic instrumentation development has been addressed by National Physical Laboratory (NPL) [5], which have produced a new generation of acoustic measuring instruments. Such instruments formed into a sensor network have to fulfil the following criteria, which will implicate on the existing standards of how sound and noise would be measured.

- Deployable in arrays
- Minimally perturbing
- Robust and inherently stable
- Can be arranged on a variety of surfaces (planar/curved)
- Placed on space-frames
- Used for multi-position, long-term monitoring
- Deployable – can be left measuring whilst unattended
- Less vulnerable to theft
- Multi-parameter sensing (pressure, intensity on 3-axes, etc.)
- Known relative positions
- Simple and low cost calibration
- Potential for self calibration or internet calibration

For decades, the signal processing community has devoted enormous research attention to seamless integration of signals from multiple resources and sources with heterogeneous sensing modalities. The signal processing literature sometimes refers to this as array processing with heterogeneous sensors; often called data fusion. There are many applications of this technique, such as signal enhancement (noise reduction), source localisation, sound characterisation, process control, and source coding. Target tracking, energy-efficient radio scheduling, beamforming, acoustic localisation, and coordinated actuation all need to be time synchronised in various forms. It would seem natural to implement such an algorithm in distributed sensor networks, and there has been great interest in doing so. However, much of the extensive prior work in the field assumes centralised sensor fusion. That is, even if the sensors gathering data are distributed, they are often assumed to be wired into a single processor. Centralised processing makes use of implicit time synchronisation, i.e. sensor channels sampled by the same processor which also share a common time base.

For example, consider the "blind beamforming" array for localising the source of sound, described by Yao et al. [6]. The array computes phase differences of acoustic signals received by sensors at different locations. From these phase differences, differences in the time of flight of the sound from the source to each sensor can be computed. This allows the sound's source to be localised with respect to the spatial reference frame defined by the array. However, the technique makes the implicit assumption that the sensors themselves are time synchronised with each other. In other words, the beamforming computation assumes that the observed phase differences in the signal are solely due to differences in the time of flight from the sound source to the sensor. The technique breaks down if there are variable delays incurred in transmitting signals from the sensor to the processor. In a centralised system where there is tight coupling from sensors to a single central processor, it is usually safe to assume that the different sensor channels are synchronised. For such an array to be implemented on a fully distributed set of autonomous wireless sensors, explicit time synchronisation of the sensors is needed.

The following four approaches for time synchronisation, namely Network Time Protocol (NTP), Reference Broadcast Synchronisation (RBS), Timing-Sync Protocol for Sensor Networks (TPSN) and Flooding Time-Synchronisation Protocol (FTSP) will be described and compared.

NTP [7] has been used extensively for networked systems requiring time synchronisation with an accuracy of milliseconds. This method can be found in internet technology, which relies on its external clock source, such as GPS. Wireless sensor networks with low precision time synchronisation can use this approach. RBS uses one of their nodes as the beacon to synchronise their clock [8]. The clients exchange their respective reception times and use them for their offset and rate of clocks differences estimated using linear regression. TPSN employs a hierarchical structure of sensor networks to create a global timescale throughout the network by pairing synchronisation along the edges of the structure [9]. This will eventually synchronise the nodes by exchanging two synchronisation messages with its reference node one level higher in the hierarchy. Because of this the message load will be higher than the previous methods. TPSN achieves double the performance of RBS by time-stamping the radio messages. FTSP combines the features offered by the previous two methods [10]. The objective is to achieve a network wide synchronisation of the local clock of the participating nodes, with an accuracy of micro-seconds. Time synchronisation is achieved between the sender and possibly multiple receivers utilising a single radio message time-stamped at both the sender and the receiver sides. Compensation for the clock drift to achieve high precision and low communication overhead is estimated using linear regression. This method supports multi-hop synchronisation, network topology changes as well as nodes failure. Table 1 gives a comparison of the methods employed for Wireless Sensor Network (WSN) time synchronisation.

Table 1 Comparison of the existing methods employed for time synchronisation[10]

	NTP	RBS	TPSN	FTSP
Message source for synchronisation	External	Internal	External	Internal
Duration	Continue	On-demand	Continue	Continue
Scope	All nodes	Subsets	All nodes	All nodes
Timescale transform or clock sync	clock sync	Timescale transform	clock sync	clock sync
Accuracy	In ms	in tens of µs	twice as RBS	in µs

3 INSTRUMENT SYSTEM DESIGN AND EXPERIMENTAL TEST

The acoustic source capturing system first studied, using soundfield microphones is shown in Figure 1. This system captures acoustic signals in such a way that a four-channel output representing a 3 dimensional sound field is reproduced. The B-Format signals produced by this type of microphone represent a three directional figure-of-eight particle velocity signal of each axis, X, Y and Z and one omni-directional pressure signal W. Therefore, this system can be used as a passive approach to predicting the direction of an acoustic source in space. Experimental tests were carried out with the objective of verifying the microphone's features, by the decomposition of the captured signals using wavelet packets transform followed by histogram selection method[11].

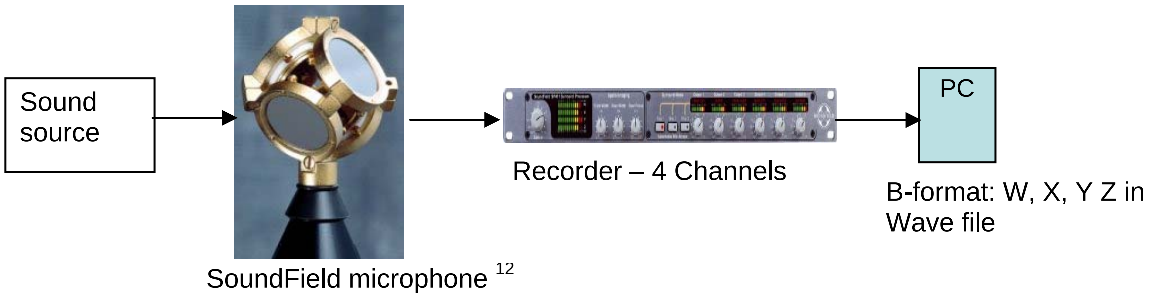

Figure 1 Instrument system arrangement implementing a nearly-coincident array of microphones known as SoundField microphone.

Acoustic source localisation experimental tests using a linear array of three microphones have been carried out and reported [13]. The objective was to assess the off-the-shelf condenser microphone's performance to accurately locating the position of the source in a reverberant room. The experiment implemented the general cross-correlation method to obtain the time difference between pairs of microphones. This procedure was then followed by a geometrical analysis to calculate the source's position. Soundfield microphones were also tested using the same approach with a result comparatively the same as that the off-the-shelf condenser microphones. Figure 2 illustrates the layout of the system arrangement. M1, M2 and M3 are the microphones positioned in-line with separation distance l1 and l2. The source position is calculated from microphone represented by distances from each of microphones R1, R2 and R3 also at the angle measured at the microphones.

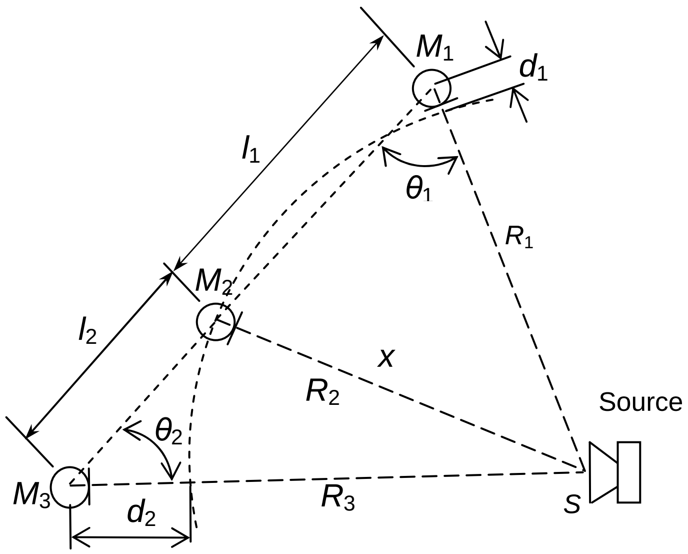

Figure 2 A linear array of three microphones arrangement to locate source position.

Commercially available Mica2 motes wireless network measurement system is used in the initial work. Figure 3 shows the block diagram of the mote. The motes are interfaced with a sensor board to gain their sensing capability. However, the available sensor boards are not adequately equipped with acoustic probes that can sense the acoustic source sufficiently well to be reproduced. The sampling frequency and the resolution are too low. Figure 4 shows the captured signal reproduced by two different mote's sensor boards. From the figure it is clearly seen that the signal is accompanied by periodic spikes due to digital switching of the motes. Based on these facts an acoustic sensor module is being developed as shown in Figure 5. The module will implement a micro controller unit such Texas Instruments MSP430 Series or Digital Signal Processor (DSP). The module will be interfaced to the mote through a 51-pin connector.

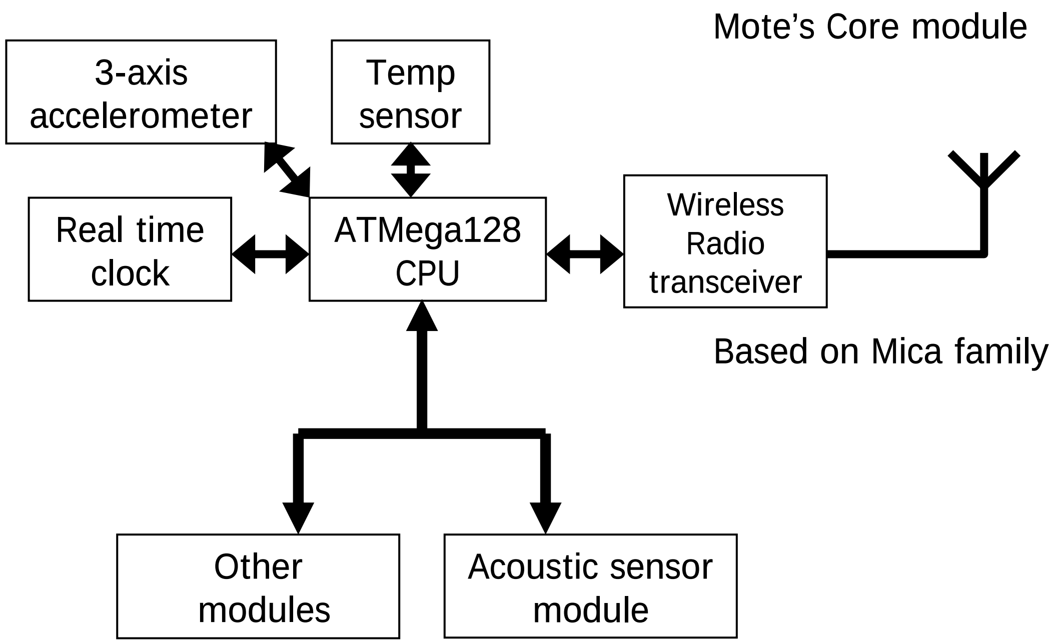

Figure 3 Mote's core block diagram showing the interface connections to other modules

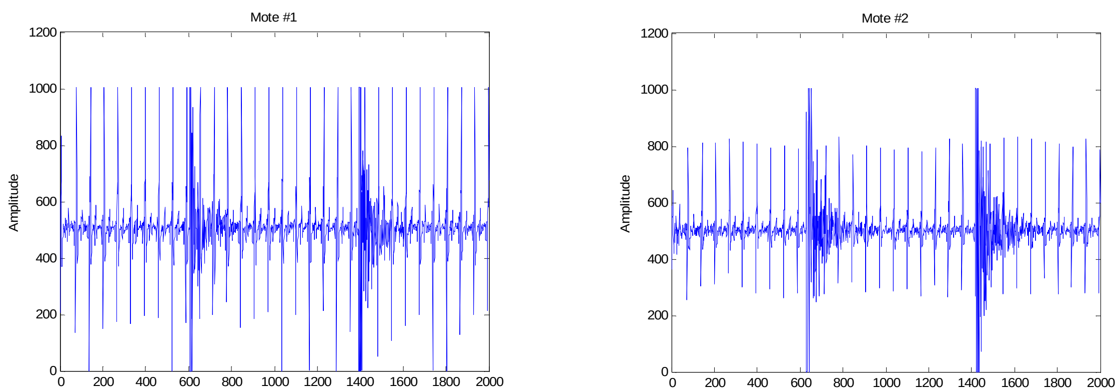

Figure 4 Captured signal by the sensor board of the Mica2 motes sampled at 3 kHz. The spikes occur due to digital switching on the motes.

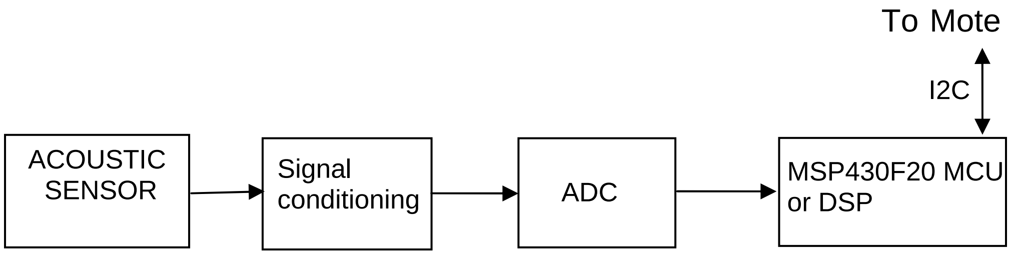

Figure 5 Acoustic sensor modules with MCU or DSP to be connected to the Mote module.

To test the feasibility of the approach, initially pair of desktop PCs was used rather than embedded DSPs. These computers contain sound cards capable of capturing acoustic signals with as good as music CD quality (44 KHz sample rate) were connected to Mica2 motes. Figure 6 illustrates block diagram of the arrangement. This system is under development for testing and evaluation.

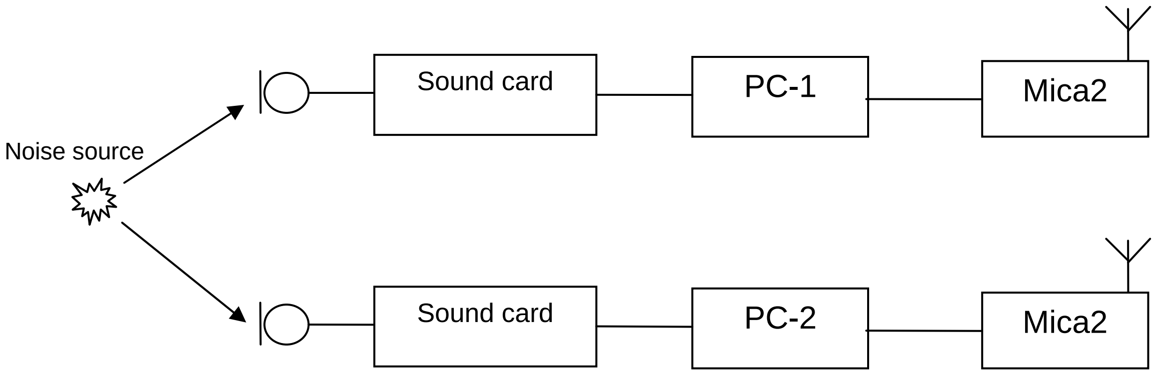

Figure 6 Wireless sensor network based instrument for sound capturing using PC as the sensor boards.

Each of the PC involved in the arrangement connects to Mica2 mote through USB port. The PC acting as the server signals the other PC to start as well as to stop capturing data simultaneously via Mica2. The capturing process records the starting time of each of the PC clock. The difference starting time of each PC compensates the data captured by this system.

4 CONCLUSION

The work on a wireless sensor network based instrument for acoustic signal capturing and the wireless sensor network state-of-the-art have been presented. An approach to distributed acoustic instrumentation design and time synchronisation work has been discussed. Some implementation and experimental study is under development. The work further designs the instrumentation system and implements it within preconditioned indoor and outdoor environments. The Universities of Southampton, York and Newcastle will undertake several experimental tests using the developed system.

5 Acknowledgement

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