Designing and evaluating the performance of a wireless sensor network for environmental noise monitoring applications.

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Abstract
The concept of Smart Cities is an area of research that has drawn scientific attention and has been discussed thoroughly over the last few years. This is due to the fact that nowadays there is a good understanding of the requirement for monitoring of parameters that affect people's behavior and wellbeing. From an environmental perspective noise is undoubtedly an important factor with immediate effects on peoples' lives. However, continuous monitoring in big cities can prove to be a very complex and expensive task if traditional methods are followed. That is because of the complexity and diversity of modern urban environments. On the other hand modern technologies have made possible the creation of low cost Wireless Sensor Networks which can provide a viable alternative. The information harvested from such networks can be used to inform the public but also assist urban planners in their decision making. Hence it is a fundamental requirement that data quality is uncompromised. In this work the design and calibration procedures of a versatile and reliable low cost Wireless Sensor Network created at NPL are presented. All necessary tests prior to deployment as well as tools that ensure network stability are outlined. Moreover, data from a first deployment at a high impact railway construction site in the center of London are presented and discussed.

Keywords: Noise Monitoring, Wireless Sensor Networks, Smart Cities, Performance Assessment

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1 Introduction

As population numbers residing in urban environments increase with time the requirement for monitoring of stressors affecting people's well-being becomes more pronounced. Noise is a form of pollution that has an immediate impact on peoples' health and behavior [1], [2] [3]. Whether it is due to increased traffic, airport or construction site operation, noise is the predominant component of modern urban soundscapes. Hence it is important to monitor and understand how it evolves in order to potentially predict and prevent undesired effects caused by it.
So far noise monitoring has relied on traditional methods and equipment such as baseline surveys involving trained professionals and expensive sound level meters. However, the increased costs of such solutions prohibit extensive noise surveys and so the time and spatial resolution achieved is usually low.
The utilization of modern technologies can provide solutions to the problem described. Nowadays capable computers like Raspberry Pi, Arduino, Banana Pi etc. have become surprisingly cheap while low cost sensor implementations like MEMS microphones are improving rapidly. Thus it is not surprising that lots of scientific interest has been generated towards the transformation of low cost components into measurement grade systems that can be used to form inexpensive sensor networks [4], [5], [6], [7], [8] [9]. Such networks can comprise of great numbers of individual measurement stations which are capable of providing continuous data streams. The information collected can then

be used in various ways, from assisting city planners in decision making, to providing real time noise readings to the public. For this reason it is important that their operation is not compromised and so a great amount of effort has been channeled towards the integrity evaluation of the data gathered [10], [11], [12].
In this paper the design of a Raspberry Pi based noise monitoring network is presented. An overview of the complete system is given while particular focus is placed on the assessment of the processing units' performance. Finally a brief presentation is given of early data gathered by the first deployment of the network at a construction site in central London.

2 Network overview

The aim of the overall design was the use of commercially available components which can be appropriately combined in order to produce a low cost yet powerful and physically resilient solution. The network should comprise of numerous compact, easily installed and calibrated measurement stations capable of sampling the soundscape with high frequency, process the collected samples to extract various acoustic parameters and transmit the resulting data to a remote server. On the server, the data should be further processed in order to produce information that could finally be presented to the end user of the network.

2.1 Hardware

A Raspberry Pi 2 Model B computer was selected as the basis of the data acquisition and processing units. The technical characteristics of the Raspberry Pi 2 Model B include a 900 MHz quad-core ARM Cortex-A7 CPU, 1 GB RAM, 4 USB ports, 1 Ethernet port and 40 GPIO pins which make it a very versatile board about the size of a credit card. The GPIO pins and Ethernet and USB ports as well as the abundance of available peripherals for Raspberry Pi computers provide easy connectivity using standard network cables, Wi-Fi USB antennas, USB 3G/4G dongles or even XBee/ZigBee modules. The above together with the low price of the board and the vibrant Raspberry Pi user community were particularly important factors for the selection of the embedded computer.
The microphone was based around a Cirrus Logic WM7132PE analogue silicon sensor which offers 65 dB signal to noise ratio, 93 dB dynamic range with acoustic overload at 126 dB (THD<10%) and low power consumption. The sensor was placed in a ¼" enclosure specially designed at the National Physical Laboratory, U.K. using NPL's patented acoustic filter technology to produce a standard ¼" sized microphone meeting IEC 61672 class 1 frequency response specifications [13] between 100 Hz and 18 kHz. The dimensions of the package allow for calibration using standard calibrators with a ¼" adapter fitted.
For the sound-field sampling the Cirrus Logic element 14 audio interface was chosen. The specific audio interface is built around the WM5102 audio codec which is a low-power system designed for portable audio applications with good quality analogue to digital converters. The audio card can provide a microphone bias voltage level of 2.8 V which matches the 1.5 V - 3.7 V working range requirements of the microphone. It also provides Programmable Gain Amplifiers (PGA) for the line/mic input and achieves 95 dB of dynamic range when the PGAs are set to 0 dB gain. This is more than enough for any urban environmental noise application. Moreover it supports sampling rates up to 192 kHz. It is designed to plug firmly on to the Raspberry Pi's 40 pin extended GPIO connector forming a compact double deck board and it also provides a GPIO expansion header where additional modules like temperature sensors can be mounted. It comes with a customized Cirrus Logic Linux distribution that following appropriate configurations was used as the operating system of the processing units.

As mentioned there are a number of available options for data transmission solutions supported by Raspberry Pi computers and depending on the application one can choose whichever seems more suitable. For the described network mobile communication standards such as 3G and 4G were thought to be most appropriate since the system is intended mostly for urban applications where good network coverage is usually achieved. The selected module was a Huawei 3G USB dongle. All electronic components were fitted in a 13x13cm IP65 enclosure to form the water and dust proof, resilient measurement station presented in Figure 1. IP65 compression glands were used to seal the power and microphone cable openings and bent aluminum tubes were used for mounting the microphone away from the unit in order to minimize undesired effects on its response. The overall manufacturing cost of each measurement unit was less than £200 as of April 2016.

Figure 1: NPL's Raspberry Pi based wireless sensor network's measurement station.

2.2 Software

The measurement application running on each measurement station was programmed in Python and consists of three main processes: the recorder, processor and transmitter. The recorder continuously collects 200 ms long audio buffers through the audio interface at a sampling frequency of 44.1 kHz. The processor passes the collected buffers through an A-weighting filter and subsequently a series of 1/3 octave band filters and extracts broadband and 1/3 octave A-weighted Leq levels. Then the transmitter compresses the processors outputs and transmits them to the server. The result is a continuous stream of one broadband and twenty six 1/3 octave band (40 Hz-12.5 kHz), A-weighted Leq measurements every 1/5 of a second. All three processes operate separately but communicate internally to make sure that the algorithm functions as expected. If the internet connection is dropped the computed levels are stored in text files on the SD card of the Raspberry Pi from where they can be accessed remotely via SSH once the connection is restored. This guarantees that no measurements are lost unless of course the SD card reaches its maximum capacity before the internet connection is restored. Nevertheless a 16 GB SD card requires several months of continuous data storage to reach its maximum capacity.

From the server's side the incoming data are handled by a PHP API and stored into the main table of a PostgreSQL database. A number of PHP algorithms running on the server pick up the high resolution data from the main table and compute broadband and 1/3 octave 1 second, minute, hour and day averages as well as 1 minute L_{min}, L_{max}, L_{10}, L_{50} and L_{90} statistical levels. The outputs are stored in separate tables on the same database. This way the network provides a complete measurement solution for all common acoustical parameters used in environmental noise monitoring surveys. The aggregation intervals can be easily adjusted to serve the purposes of any application.

Finally the user can view the live data stream together with a map of the measurement stations' locations through a fully responsive web interface. Among other functionalities the web interface provides accessing, plotting and downloading capabilities for all stored data as well as visualizations of useful statistical measures about the network's activity.

3 Performance assessment

The final units were subject to thorough testing in order to benchmark the measurement grid's performance. Both hardware and software were assessed and the results are presented in the following paragraphs. It should be mentioned once again that for this paper focus is placed on the processing units' and algorithms' performance assessment rather than the microphone's which is the subject of a different study.

3.1 Processing time

High temporal resolution is of great importance in acoustic measurements. Fast integration periods are commonly used to capture sudden sound events which can otherwise appear to be much longer in time and lower in amplitude giving false impression about the nature of the sound field. Hence one of the main goals when designing the measurement application was the creation of an algorithm that can produce real time measurements at fractions of a second. This was achieved by using parallel processes for the various tasks which were easily handled by the four cores of the Raspberry Pi. The upper limit in the number of measurements per second would be set by the processing time required for each buffer. Of course the smaller the buffer the lower the processing time however there is a lower limit to this posed by the processing power of the unit, the architecture of the algorithm and the programming language used. For the described system a comfortable 200 ms integration time was selected resulting in 5 broadband and 1/3 octave band measurements per second.
In order to assess the performance of the processor two time measurement points were selected in the algorithm; right after the buffer capture and right after the processor's output. The difference in time would characterize its performance. The results for 200 ms long buffers over 4.5 consecutive days of operation are presented Figure 2.

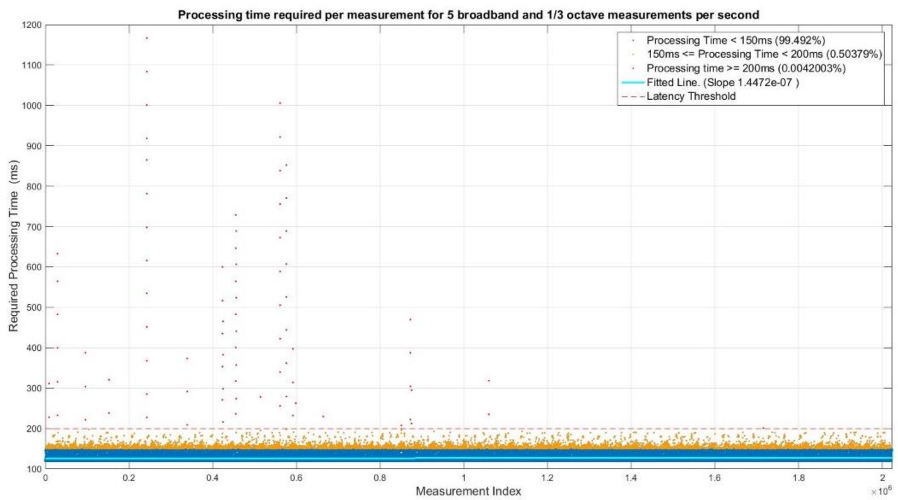
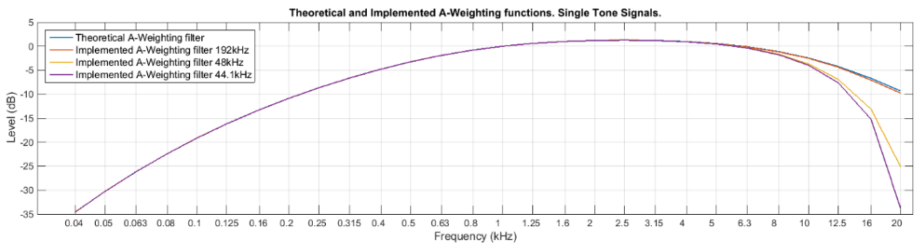


Figure 2: Processing time for 5 broadband and 1/3 octave frequency band measurements per second.

The processing time threshold was set at 200 ms which equals the time require to capture one buffer. The recorded times were divided in three categories: Lower than 150 ms, between 150 ms and 200 ms and above the 200 ms threshold. Any value above 200 ms introduces a delay in the processing of the next buffer which in turns introduces a delay in the following one and so on. If this happened continuously it would result in an asynchronous process. However as seen in the graph 99.5% of the recorded times were below 150 ms, about 0.5% was between 150 ms and 200 ms while only 0.0045% were over 200 ms. A closer inspection would reveal that those very few processing cycles exceeding the threshold would only introduce delays for the following 12 cycles at max. That means that the process would catch up with real time within two to three seconds. Also the fitted line presents a slope of $1.37e^{-07}$ with an offset of 126.6 ms indicating that there is no upwards trend in the data and thus guaranteeing that the process operates comfortably at real time.

3.2 A-weighting and third octave digital filters implementation

The main function of the Processor is the A-weighting and third octave band filtering. The filtering takes place in the time domain using digital filter implementations whose response is inseparable from the sampling frequency. At the same time the sampling frequency is closely related to the processing time required for each buffer. The lower it is the faster buffers can be processed and higher temporal resolution can be achieved. However there is a limit to how low sampling frequencies should be used in order for the system to operate within the Class 1 thresholds set by IEC 61672.
Before deciding on the sampling frequency the behavior of the implemented digital filters had to be investigated. To do so three sampling frequencies were selected at 44.1 kHz, 48 kHz and 192 kHz. A signal generator was used to supply the measurement units with single tone signals and the extracted data were compared to the theoretical A-weighting values. The results are presented in Figure 3.



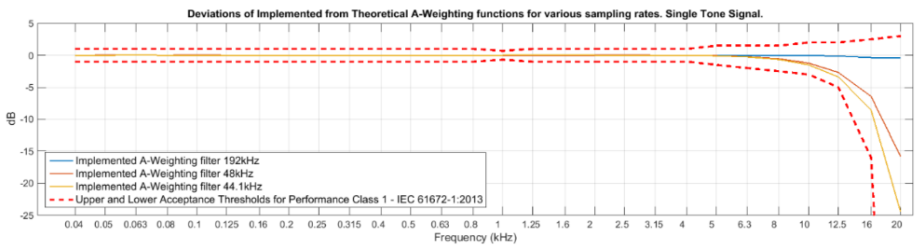


Figure 3.1: Theoretical and implemented A-Weighting filters for sampling frequencies at 44.1 kHz, 48 kHz and 192 kHz.
Figure 3.2: Deviations of implemented A-weighting filters from IEC 61672 Performance Class 1 thresholds.

As seen the higher the sampling frequency is the closer the filter's response gets to the theoretical values, however, all of them result in responses compliant with the Class 1 performance thresholds set by IEC 61672. Following the above results a 44.1 kHz sampling rate was decided for the data capture and processing and a comparison between Z and A-weighted levels was performed using broadband white noise this time in order to add the third octave band filters’ response to the test. This is depicted in Figure 4.

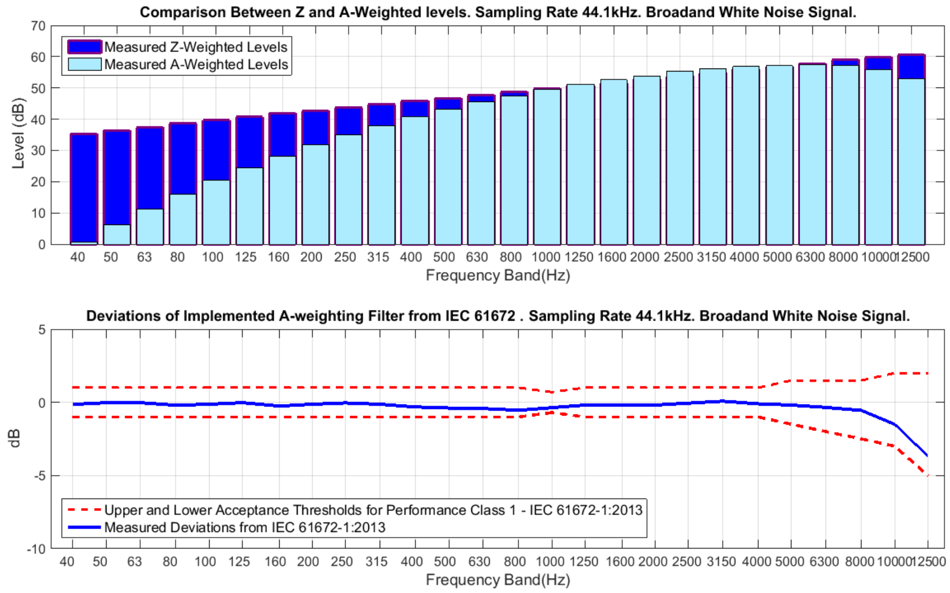
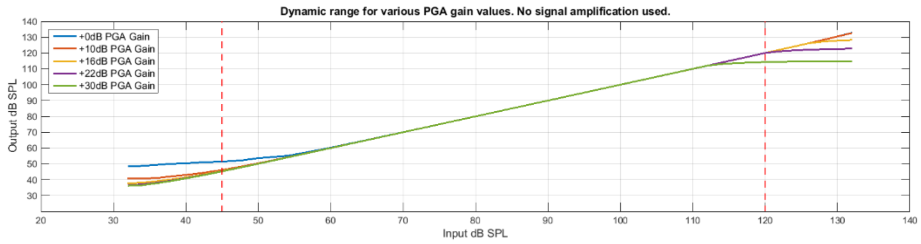


Figure 4.1: Comparison between measured Z and A-weighted levels for broadband white noise input signal.
Figure 4.2: Deviations of Implemented A-weighting filter from IEC 61672 Performance Class 1 thresholds.

3.3 Linearity, gain selection and dynamic range

Linearity is of upmost importance when it comes to measurement instrumentation. In many cases though low cost components do not behave linearly for various reasons. This is particularly true for analogue to digital converters found in cheap audio interfaces. For this reason it was absolutely necessary to test the measurement unit’s behavior in order to gain trust for the output data. At the same time the linearity tests were a good opportunity to investigate the effect of the PGAs on the dynamic range and decide on the most appropriate settings.
A signal generator was used to generate a 1 kHz tone which was then passed through an attenuator before reaching the unit’s input. The generator’s output voltage and attenuator’s reduction level were matched to the sensitivity of the microphone in a way that they would allow for sufficient signal attenuation and amplification so that the whole dynamic range of the audio interface could be covered. Then a series of linearity tests were run using steps of 2 dB for 0 dB, +10 dB, +16 dB, +22 dB and +30 dB PGA gain values. Finally the effect of introducing a pre-amplification stage to the input signal chain was also examined with the PGA gain levels set at 0 dB and +10 dB. The results are shown in Figure 5.



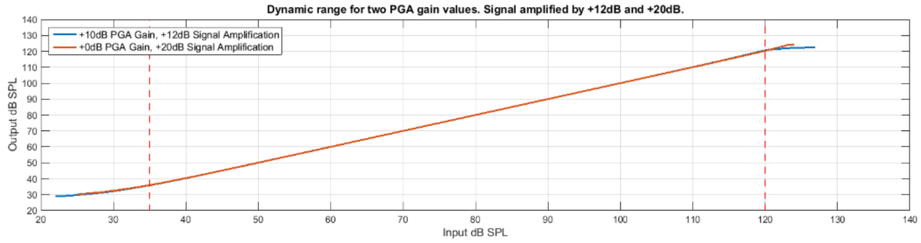


Figure 5.1: Linearity and dynamic range measurements for 0 dB, +10 dB, +16 dB, +22 dB and +30 dB PGA gain levels.

Figure 5.2: Linearity and dynamic range measurements for +10 dB PGA gain with +12 dB input signal amplification and 0 dB gain with +20 dB input signal amplification.

In the presented graphs it is clear that the system behaves linearly. It is also seen that increasing the PGA gain introduces a trade-off between the lower and upper end of the dynamic range. The lower the PGA gain the higher the maximum dB SPL levels that can be achieved. However low gain values result in increased dB SPL noise floor levels. On the other hand, high PGA gains result in lower dB SPL noise floors but also decreased maximum levels squashing the dynamic range. This makes clear that when no external signal amplification is used the PGA gain levels must be carefully selected so that the system behaves linearly over the desired dB SPL range. However introducing an external amplification stage, as presented in Figure 5.2, matches better the microphone's output and the audio interface's input voltage ranges and hence makes possible the use of low PGA gain levels achieving both low noise floor and high maximum dB SPL levels.
From the above it is concluded that when no external amplification is used the PGA gain should be set between +16 dB and +22 dB in order for the system to measure linearly from 45 dB SPL to 120 dB SPL. This 75 dB of linear dynamic range is considered sufficient for monitoring noisy environments like construction sites. When monitoring quiet environments, like residential areas, a +20 dB external amplification can achieve an 85 dB of linear dynamic range from 35 dB SPL to 120 dB SPL.

3.4 Temperature and humidity impact measurements

Any permanently installed environmental monitoring system is certain to be exposed to varying weather conditions that can have negative impacts on its normal functioning. As a result it is important to assess how such systems behave under extreme environmental conditions prior to deployment. In order to evaluate the performance of the described noise monitoring units a lab based test was performed using NPL's temperature and humidity chamber. A signal generator was used to supply a constant level signal to a measurement station placed inside the chamber while the temperature and humidity settings were varied. Each set of measurements was taken after the environmental conditions inside the chamber had stabilized for more than three hours. Table 1 summarizes the results which show that even extreme conditions such as 40.4 C^0 and 90% humidity had no significant impact on the operation of the unit.

Table 1: Varying temperature and humidity level dependencies.

Temperature (C^0)	Humidity (%)	Measured Level (dB)
6.8	80	72.40
15.7	30	72.38
21.4	70	72.37
40.4	90	72.34

4 Early data from first network deployment

The first deployment of the described noise monitoring network took place at a high impact railway construction site in London. Being in the center of the city at an area that is under heavy re-development the site is surrounded by a number of other construction sites and is nearby several sensitive locations. This deployment was intended to monitor the noise generated by the construction works and develop algorithms for noise source separation aiming to identify the contribution of the specific site to the noise levels existing in the area. For this purpose 17 monitoring stations were installed in key locations.
At the time of writing the network is still in the process of initialization. Due to the size and amount of processes taking place on site not all units were possible to initialize simultaneously. After two days from installation 5 stations have started transmitting data allowing for a very limited data analysis but a good demonstration of the capabilities of the web interface. Figure 6 presents a screenshot of the interface's Charts tab.

Figure 6: Screenshot of the web interface's Chart tab presenting data collected from the railway construction site over the first two days.

In the above figure the upper plot presents broadband Leq (A) levels averaged over 1 minute. The high temporal correlations are obvious while the noise level difference between night-time and day-time is distinguished quite clearly and appears to be of the order of 10 decibels. This is more easily seen in the lower right plot which is used as close inspection tool for any of the selected measurement units. The lower left pie chart shows the activity of the selected units as a percentage over the expected activity for the examined time period.

5 Conclusions

Modern technologies have made possible the creation of low cost wireless sensor networks using off the self components. Careful selection and combination of separate elements together with some good understanding of their capabilities and limitations can lead to the creation of robust and powerful systems like the one described.

The Raspberry Pi 2 Model B has proved to be a versatile computer that supports many connectivity options and can be used as the foundation for a Linux based measurement system. Moreover the specifications of the Cirrus Logic interface make it an appropriate choice for most urban noise monitoring applications. Based on these two components and careful algorithm development the designed units manage to achieve 5, A-weighted, broadband and 1/3 octave band Leq measurements per second which provides high temporal resolution. The collected data can then be used to extract any environmental noise parameter like aggregated and statistical levels once they have reached the database. Finally the created web interface enhances the network by giving easy access to the stored data from any internet connected device.

Operating at 44.1 kHz the processing units achieve real time processing and digital filters complying with the Class 1 acceptance limits specified by IEC 61672. They also perform linearly providing an adjustable, wide dynamic range which is considered more than enough for most environmental noise monitoring applications. Additionally extreme environmental conditions do not seem to have any significant impact on the units.

Finally the data gathered by the first real life deployment seem promising although it is still quite early to perform any meaningful analysis on them. One of the first lessons learnt, however, is that deployments in such environments can be challenging as they require the coordination of a great number of people and thus sufficient time should be allocated during the planning phase of such projects.

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