The design and calibration of low cost urban acoustic sensing devices

Charlie Mydlarz
Center for Urban Science and Progress, New York University, New York, USA, cmydlarz@nyu.edu

Charles Shamoon
New York City Department of Environmental Protection, New York, USA, charlessh@dep.nyc.gov

Melody Baglione, Michael Pimpinella
Vibration and Acoustics Lab, The Cooper Union, New York, USA, melody@cooper.edu, pimpin@cooper.edu

Summary
The urban sound environment of New York City (NYC) can be, amongst other things: loud, intrusive, exciting and dynamic. As indicated by the large majority of noise complaints registered with the NYC 311 information/complaints line, the urban sound environment has a profound effect on the quality of life of the city's inhabitants. To monitor and ultimately understand these sonic environments, a process of long-term acoustic measurement and analysis is required. The traditional method of environmental acoustic monitoring utilizes short term measurement periods using expensive equipment, setup and operated by experienced and costly personnel. The proposed project takes a different approach to this application by implementing smart, low-cost, static, acoustic sensing devices based around consumer hardware. These devices can be deployed in numerous and varied urban locations for long periods of time, allowing for the collection of longitudinal urban acoustic data. The varied environmental conditions of New York City make for a challenge in gathering calibrated sound pressure level data for prospective stakeholders. The wide variations in temperature and humidity affect microphone sensitivity and response, which can increase the likelihood of the generation of erroneous sound pressure level readings. This paper details the sensors' design, development and potential future applications, with a focus on the calibration of the devices' Microelectromechanical systems (MEMS) microphone in order to generate reliable decibel levels at the type/class 2 level.

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1. Introduction

Noise pollution is an increasing threat to the well-being and public health of city inhabitants [1]. The complexity of sound propagation in urban settings and the lack of an accurate representation of the distribution of the sources of this noise have led to an insufficient understanding of the urban sound environment. The presented project aims to continuously monitor and ultimately understand these urban sound environments. Examples of the long term goals of the project, include how sound impacts on the health of a city's population, correlates with urban problems ranging from crime to compromised educational conditions, and how it affects real estate values. While a number of past studies have focused on specific contexts and effects of urban noise [2], no comprehensive city-wide study has been undertaken that can provide a validated model for studying urban sound in order to develop long-lasting interventions at the operational or policy level.

With its population, its agency infrastructure, and its ever-changing urban soundscape, NYC provides an ideal venue for a comprehensive study and understanding of urban sound. To achieve this goal an initial network of low cost acoustic sensing devices [3] were designed and implemented to capture long-term audio and objective acoustic measurements from strategic locations throughout the city using wireless communication strategies. These prototype sensing devices currently incorporate a quad-core Android based mini PC with WiFi capabilities, and a Microelectromechanical systems (MEMS) microphone. The initial goal is to develop a comprehensive cyberphysical system that provides the capability of capturing, analyzing and wirelessly streaming environmental audio data, along with its associated acoustic features and metadata. This will provide a low-cost and scalable solution to large scale calibrated acoustic monitoring, and a richer representation of acoustic environments that can empower a deeper, more nuanced understanding of urban sound based on the identification of sources and their characteristics across space and time. As part of this goal, work is ongoing to equip the sensors with state-of-the-art machine listening capabilities such as automatic sound source identification through the development of novel algorithms [4].

2. The revision of the noise code of NYC

The NYC Department of Environmental Protection (DEP) revised the Noise Code for the first time in 30 years. The Noise Code bill was passed by the NYC Council unanimously on Dec 21, 2005 and signed by the Mayor on Dec 29, 2005. There was almost universal praise for the collaboration between the City, the real estate industry, construction groups, hauling, utilities, and nightlife industries, neighborhood groups and the City Council. The legislation protected NYC's legacy as the "City that never sleeps" while making sure that New Yorkers can get some peace and quiet.

2.1. Recognition & awards

The National Academies including the National Academy of Engineering reviewed the NYC Noise code and favorably discussed it in their extensive committee report [5]. On page 118 it states that the NYC Noise Code is a modern noise code and: "a good starting point for upgrading existing noise laws or creating new ones." The Noise Pollution Clearing House, a well-known noise control organization presented a prestigious award to Mayor Bloomberg for his support for the 2005 Revision of the NYC Noise Code at the 10th Anniversary of Noise Pollution Clearinghouse. The National Safe In Sound Hearing Loss Prevention Award was also awarded to the DEP in 2010, recognizing Innovation in Hearing Loss Prevention in the Construction Sector. The NYC-DEP were recognized for their combined efforts in developing, implementing and overseeing the NYC Construction Noise Mitigation Rule. The rule, which is a result of a Mayoral charge to update the New York City's Noise Code, established noise emission limits and mitigation measures for all city construction and also proactively addressed work-related exposures.

2.2. Complaint procedure

Generally the Noise Code is complaint driven. Inspectors are dispatched to the location of the complaint to determine the ambient sound level and the amount of sound above the ambient. In 2003, in a related endeavor, a non-emergency telephone number 311 was established in New York. It provides an easy-to-remember telephone number to attain access to municipal services. Dialing this number allows city residents to obtain important non-emergency services through a central, all-purpose phone number quickly and effectively for issues such as noise complaints, heating issues and parking regulations. The largest US 311 Citizen's Service Hotline operation that exists operates in NYC. Citizen noise complaints to 311 established itself as a major quality-of-life issue.

2.3. Measurement

Measurements are taken by certified and trained inspectors who actually observe violations. They use type 2 sound level meters (SLMs) for most inspections. The SLM readings are read as Lmax with the meter set to slow response. Measurement of sound levels shall comply with standards established by the American National Standards Institute specifications for sound level meters S1.4-1971. The department measures using the A & C weighting filter. The DEP also has a section of maximum allowable dB for 9 different frequencies.

3. Acoustic measurement

In order for a piece of equipment to be suitable for acoustic measurement purposes, it should comply with the sound level meter (SLM) standard IEC 61672-1 [6]. This includes, for example, tolerance limits for a device's frequency response, self-generated noise and linearity. Two "type" specifications are defined where type 1 devices, designated Precision, are intended for accurate sound measurements in the field and laboratory, type 2 devices, designated General Purpose, are intended for general field use. The overall accuracy of the device is determined by its "type" rating. In the US, the general minimum type specification for use in noise surveying is type 2. It is not the intention of this paper to prove that this sensor network can be used to generate legally enforceable acoustic data for a location, but the data that it can provide will be a real-time, continuous and accurate indication of the acoustic conditions in which each sensor inhabits. This data stream will help to inform and augment urban noise enforcement procedures, e.g. optimizing the allocation of in-depth noise assessment personnel and equipment.

4. Hardware

The projects sensor network is based around a consumer computing platform where low cost and high power are of paramount concern. The design philosophy is based on the creation of a network that provides dense spatial coverage over a large area, through the deployment of inexpensive and physically resilient sensors [3]. At the core of the projects sensing device is a Tronsmart MK908ii mini PC running the Android 4.2, Linux based operating system. These small and versatile devices shown in Figure 1 are priced at $50USD as of March 2015 and provide a 1.6GHz quad core processor, 2GB of RAM, 8GB flash storage, USB I/O, and WiFi connectivity. The computing power offered by these units allows for complex digital signal processing to be carried out on the device, alleviating the need to transmit large amounts of audio data for processing on the project's servers.

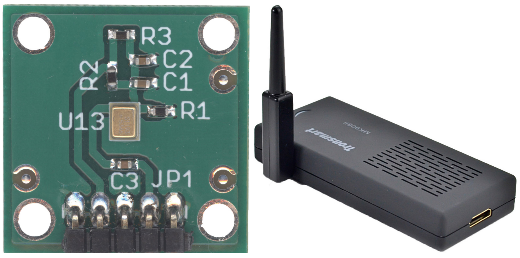

Figure 1. MEMS mic. PCB (Knowles SPU0410LR5H-QB mic. in center) & Tronsmart MK908ii mini PC

USB I/O allows for the inclusion of a USB audio device to handle all analog to digital conversion (ADC) work, thus providing the means to connect a custom microphone solution. The USB audio device chosen for this application had to be compatible with Linux based Android devices, low in price, provide input gain control and a clean signal path. The device selected was the eForCity USB audio interface which retails for $4 as of March 2015. It provides a single microphone input channel with low noise and a software adjustable input gain stage. The frequency response of the device was measured and whilst it introduces filtering with a steep roll-off below 20Hz and above 20kHz, the audible frequency range is unaffected.

5. MEMS microphones

In recent years, interest in MEMS microphones has expanded due to their versatile design, greater immunity to radio frequency interface (RFI) and electromagnetic interference (EMI), low-cost and environmental resiliency [7]. Current MEMS models are generally 10x smaller than their electret counterparts. This miniaturization has allowed for additional circuitry to be included within the MEMS housing, such as a pre-amp stage and an ADC to output digitized audio in some models. The production process used to manufacture these devices also provides an extremely high level of part-to-part consistency in terms of acoustic characteristics such as sensitivity and frequency response, making them more amenable to multi-capsule and multi-sensor arrays, where consistency of individual microphones is paramount. In the proposed prototype microphone system we investigate the Knowles SPU0410LR5H-QB. The silicone diaphragm MEMS microphone has a manufacturer quoted "flat frequency response" between 100Hz and 10kHz. It requires a maximum 3.6V supply and draws only 120μA. In addition, it's quoted as having a sensitivity of -38dB re. 1V/Pa and a signal-to-noise ratio of 63dBA. In order to test the Knowles MEMS microphone a PCB shown in Figure 1 was designed and fabricated [8]. It was found in testing that the switched mode power supply noise created by the low cost AC-DC converters used to power the MEMS was unnecessarily high. To reduce this to acceptable levels an LT1086 voltage regulator was introduced to reduce the noisy USB 5V down to a clean 3.6V DC supply. The test results in this paper were gathered using the configuration described.

6. Measurements

In order to determine the proposed device's ability to generate type 2 sound pressure level (SPL) data, firstly a process of frequency response compensation was carried out. The device was then subjected to a subset of the IEC 61672-3 [9] acoustical test procedures, which describe the international standards for periodic testing of SLMs. IEC 61672-1 [6] provides the criteria for determining a complete SLM's ability to act as a type 1 or 2 device, including its directivity, which will be affected by the device and microphone housing. This extended set of tests will be performed on the final prototype sensor device in a more advanced stage of its development.

In the following set of measurements the SLM output (Larson Davis 831 - calibrated at the beginning of each measurement stage using the type 1 Larson Davis CAL200) will be used as a reference for comparison to the MEMS microphone (referred to as the device under test, DUT) readings to assess its ability to produce type 2 data. As the SLM is a type 1 certified device, it has its own set of inaccuracies associated with it. It has met the type 1 specifications within the defined tolerance bounds for that standard, thus for the DUT to meet the type 2 specifications, the type 1 tolerance bounds must be factored into the DUT assessment. For example, if the type 2 tolerance bounds for a particular measurement response are $\pm$2.0dB with the corresponding type 1 bounds at $\pm$1.0dB, the adjusted acceptable bounds for the type 2 class in this instance are $\pm$1.0dB (type 2 tolerance range of 4dB minus the type 1 range of 2dB) when using the SLM as the reference device. These will be referred to as the "adjusted tolerances". All of the following output values were generated from an average of 4 repeat measurements, where none of the test equipment was moved or altered. No discernible variations (<0.1dB) in output were observed between the individual measurements before averaging.

Measurements were conducted under low level (<20dBA), fully anechoic conditions at the Cooper Union, Vibration and Acoustics Laboratory. The atmospheric conditions in the anechoic chamber were measured at the beginning and end of the measurement process (≈2 hrs), and varied from 22-24°C in air temperature and 50-55%RH in relative humidity.

6.1. Frequency response compensation

The MATLAB toolbox: Scan IR [10] was used to generate the impulse responses of the reference microphone and DUT using the swept sine technique. The signals were reproduced through a studio quality Mackie HR824 active speaker and a reference PCB 377B02 microphone and PCB 426E01 pre-amplifier (assumed to be flat in frequency response from 20Hz-20kHz) were used to subtract the room and speaker coloration from the DUT's impulse response. Reference and DUT microphones were placed at 1m from the center point of the speaker on-axis, 1.3m from the floor. The DUT impulse response was generated from an average of 10 microphone boards. Negligible differences were observed in frequency response between the 10 MEMS microphones, highlighting the part to part consistency of these devices. This averaged response was then used to design an inverse linear phase FIR filter that would allow for the time domain filtering of any test signals captured by the DUT, compensating for the MEMS microphone response. The inverse filter was regularized to prevent the filter from applying extreme attenuation or amplification at the high and low frequency ranges. The process was adapted from [11], where a tapered window between 0 and 1 is applied to the high and low extremes of the desired inverse frequency response before the FIR filter is designed. The resultant filter can be easily and efficiently implemented within the native code of each sensor's mini PC providing compensation for the MEMS microphone response in real-time, allowing for the unbiased, in-situ calculation of dBA levels.

6.2. Calibration

The DUT was mounted directly beside the calibrated reference SLM microphone, shown in Figure 2. The devices were positioned at a height of 1.3m and at a distance of 1m on-axis from the center point of the speaker.

The distance between the center of each microphone capsule is 20mm, which was found to produce negligible (<0.1dBA) variations in level response when the SLM microphone's position was shifted to match that of the DUT. The output sound pressure level in dBA from the DUT is calculated from the A weighting filtered sample values, which represent the AC voltage

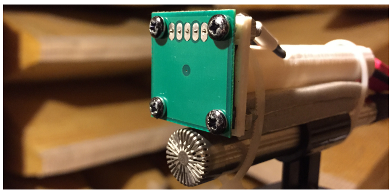

Figure 2. DUT and SLM microphones mounted

produced when presented with the calibration signal of a 1kHz sine wave at 94dBA. An offset adjustment is then applied in order to match the 94dBA SPL input level. Figure 3 shows the processes required to generate the calibrated SPL output from the DUT.

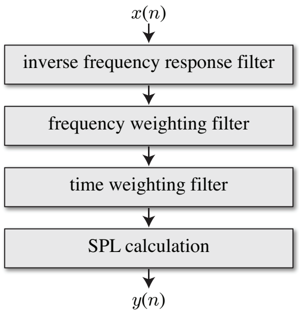

Figure 3. Block diagram of sensor's SLM functionality

6.3. Self generated noise

The DUT's self generated noise (IEC 61672-1, 5.7) was measured under low level, fully anechoic conditions, with all noise generating test equipment located outside of the chamber. Throughout the duration of the 60s measurement period, the reference SLM logged an average SPL of 22.5dBA, close to its lower limit of 19dBA. The self generated noise of the DUT was measured at **29.9dBA**. This value determines the minimum SPL the system can reliably detect. For an urban acoustic sensor in the relatively loud conditions of NYC this level is well below even a quiet suburban setting [12]. The World Health Organization (WHO) night noise guidelines for Europe [13] state that outdoor levels of 30dBA show no observed health effects on humans.

The dynamic range was then calculated using the manufacturer quoted acoustic overload point of the MEMS microphone. This results in an effective dynamic range of **88.1dBA**, with an acoustic overload point of **118dBA**. This range is more than adequate for the acoustic measurement of urban sound environments. The signal to noise ratio (94dBA @ 1kHz) of the system was measured at **64.1dBA**.

Table I. Acoustical signal tests in dBA, varying frequency (* indicates IEC61672-1 criteria met)

Freq. (Hz)	DUT	Ref.	Δ	Adj. tol.
31.5	44.8	45.2	0.4*	± 1.5
63	63.6	63.7	0.1*	± 1.0
125	76.6	76.2	0.4*	± 0.5
250	85.3	84.9	0.4*	± 0.5
500	90.2	89.9	0.3*	± 0.5
1k	93.9	94.0	0.1*	± 0.3
2k	93.6	94.2	0.6*	± 1.0
4k	94.1	93.3	0.8*	± 2.0
8k	93.2	90.6	2.6*	± 3.0
pink	79.9	80.0	0.1	N/A
white	87.5	88.0	0.5	N/A

6.4. Acoustical signal tests of a frequency weighting

To test the DUT's ability to produce accurate dBA output for different frequencies (IEC 61672-1, 5.5), it was mounted as in Section 6.2 and subjected to a test signal comprised of 9 steady state 20s sine waves, separated with 5 seconds of silence at octave frequencies from 31.5Hz to 8kHz. Table I shows the response from the reference SLM, the DUT, the difference between these two and the adjusted tolerance limits for type 2 devices as discussed at the beginning of Section 6. Standard deviations of the DUT measurements were <0.1dBA at all frequencies.

The DUT met all of the adjusted type 2 criteria for dBA frequency weightings when compared to the type 1 SLM. In addition, the response of the DUT and SLM were compared for a 20s, continuous level pink and white noise signal, showing a maximum difference in response of **0.5dBA**.

6.5. Long-term stability

In order to test the long term stability of the DUT, it was subjected to a 30min 1kHz sine wave at 94dBA. The measured difference between the dBA reading at the beginning and end of this period must be within the type 2 tolerance of ±0.2dBA stated in IEC 61672-1, 5.14. The DUT met this criteria, with an observed difference of **0.07dBA**.

6.6. Level linearity

The DUT was subjected to sine waves, linearly increasing up to 94dBA in level to test for the devices linear response to varying SPL's at different frequencies (31.5Hz - 8kHz in octave increments). This was carried out using an acoustical signal under anechoic conditions to test the entire systems response, as opposed to introducing an electrical signal directly into the pre-amp as per IEC 61672-1, 5.6.

For illustration, the vertical dashed line in Figure 4 shows the point at which the DUT meets the adjusted type 2 tolerance level (±0.6dB) for a 1kHz sinusoidal signal. The DUT can effectively operate within type 2 level linearity tolerances above **41.2dBA** for frequencies ranging from 31.5Hz - 8kHz. This lower limit can be reduced through the use of a lower noise microphone and pre-amp combination, as discussed in Section 7, however this lower limit would rarely be observed in the urban sound environment. The DUT was also subjected to a linearly increasing pink and white noise signal, where the type 2 lower limit was observed at **37.2dBA** and **36.6dBA** respectively, highlighting the device's broadband linear response to varying urban SPLs.

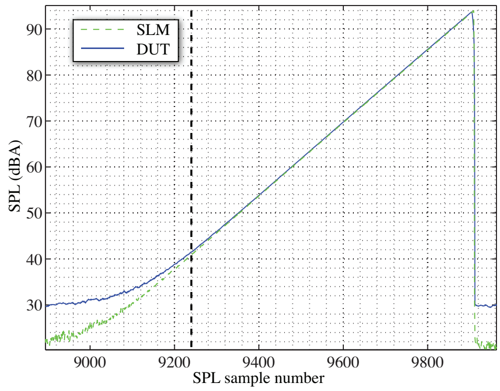

Figure 4. Linear level response of DUT vs. SLM to 1kHz sine wave upto 94dBA showing adjusted type 2 tolerance point

Table II. Acoustical toneburst tests at 4kHz, varying duration (* indicates IEC61672-1, 5.9 type 2 criteria met)

Duration (ms)	IEC61672 Δ	DUT Δ	Tol.
1000	0.0	0.0*	± 1.0
500	-0.1	0.0*	± 1.0
200	-1.0	0.0*	± 1.0
100	-2.6	-2.0*	± 1.0
50	-4.8	-4.0*	+1.0;-1.5
20	-8.3	-7.9*	+1.0;-2.0
10	-11.1	-10.9*	+1.0;-2.0
5	-14.1	-14.0*	+1.0;-2.5
2	-18.0	-18.4*	+1.0;-2.5
1	-21.0	-21.9*	+1.0;-3.0
0.5	-24.0	-25.7*	+1.0;-4.0
0.25	-27.0	-30.8*	+1.5;-5.0

6.7. Toneburst response

To test the DUT's response to transient SPLs, it was subjected to 4kHz sinusoidal tonebursts, varying in duration from 1000ms down to 0.25ms. IEC 61672-1, 5.9 defines tolerance limits in terms of dBA readings relative to the steady state 4kHz reading for type 2 devices. As these are relative measurements and do not rely on the use of the SLM as a reference, the type 2 tolerance limits as documented in IEC 61672-1 will be used.

As shown in Table II, the DUT met all IEC 61672-1, 5.9 criteria for 4kHz toneburst response.

6.8. Urban audio reproduction

To further assess the DUT's ability to capture meaningful SPL data, a 15min urban audio recording was replayed a total of 4 times under anechoic conditions with the SLM and DUT microphone mounted directly adjacent to each other on-axis to the speaker. Correlation analysis was carried out on the resultant averaged SPL time histories from the SLM and DUT. The correlation coefficient (R^2) was calculated between the entire dBA (fast time weighting) time history for each device. The total R^2 value for this 15min urban signal was **0.9723** ($p \leq 0.0001$). The mean difference between the SLM and DUT time history values was **0.4dB**.

7. Future work

The full IEC 61672-1 standard includes specifications for parameters including: device directivity, high level thresholds and environmental variations, which require the full housing of the device to be incorporated. The final prototype will be tested against the extended set of requirements, including a long term exterior comparison against a type 1 SLM. Other factors such as the location of the sensor will be investigated as the majority of potential deployment locations are in close proximity to building facades.

Noise observed on the output from the analog MEMS board is caused in part by parasitic noise from the power supply unit (PSU). This can cause measurement inaccuracies at particular frequencies where the noise is prevalent. The next iteration of the sensor's microphone solution will be an entirely digital design, utilizing a digital MEMS microphone (includes a built in ADC) and a USB audio CODEC enabling it to connect directly to the sensors computing device. The vastly improved power supply rejection (PSR) values and reduced EM/RF interference of the digital MEMS microphones over their analog counterparts should result in a much lower noise floor and an increase in dynamic range. The elimination of this noise will also result in an improved ability to capture clean audio signals for further in-situ processing and analysis.

8. CONCLUSIONS

Based on this preliminary testing phase, the analog MEMS microphone solution can produce SPL data of high quality. Its adherence to the type 2 specifications for the tests undertaken is promising for its future use in a low cost environmental acoustic sensor. Further environmental testing is needed to quantify the effects of temperature and humidity on the devices response. The main limiting factor of its noise floor means it cannot effectively operate in ambient conditions of <30dBA, however, this level would rarely be observed in the urban sound environment of NYC.

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