SPM STUDENT DESIGN PROJECT SERIES

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University:

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SUMMARY

Project Title: Acoustic detection and localization of impulsive events in urban environments

Motivation: Our project's motivation involved the study of reverberations phenomenon in acoustical signal processing applications; to record reverberated raw data occurring to varying degrees in real world environments, to attempt to remove its effects with a single or multiple microphone setup, and to use what we learn in helping to create a system that detects and localizes impulsive events. In summary, the goal was to build an electronic system that detects and identifies sounds, gunshots for example, in urban environments (*e.g.* densely populated cities *etc.*) and then tracks the sound source with minimal false positive detection rates. This can have many applications *e.g.* in law enforcement. In the majority of acoustical applications which we came across while discussing this project with our mentors and whilst doing our own literature survey, we found that the effect of reverberations is usually ignored. This, in our opinion, is a naive approximation that severely limits the efficacy of many works and applications. In an increasingly digital age where algorithms must identify useful information from data automatically in real time; be it speech or a specific event like an ambulance siren or a gunshot, the need for reverberations to be studied and compensated for becomes apparent especially for autonomic systems. Furthermore, the verification of results can best be performed if tied in with a specific real world application, which in our case was a gunshot detection and subsequent localization (detecting where the original sound source was located) system.

Project Goals: It was thus due to the highly experimental nature of the project that many acoustical signal and information processing best practices and experience came to light which we wanted to share. The goals of our project were:

- The building and testing of reliable sound recording test-beds for the following two reasons:
  1. Reverberation analysis and detection in a single channel: a portable audio recording setup that accurately gives raw sound data without the non-linear components of recording equipment.
  2. De-reverberation and localization of the sound source by analyzing the results of multiple microphones;
- To develop a high-speed data acquisition system that is able to record from multiple sources simultaneously was needed.
- The identification of appropriate ‘impulsive' acoustic events that allow the studying of the acoustical channel and most closely resemble the sound signature of a gunshot. This involves testing different methods, materials, and techniques of attempting to generate 'gunshots' (without actually firing guns of course).
- The collection of experimental data via recordings in different locations of these impulsive events and the construction of a database of clean, reverberated, and outsider signals for future study.
- The implementation of correlation algorithms that perform signal and statistical analysis on the database of signals and determines the degree to which reverberations affect signal characteristics.
- The implementation of angle-of-arrival calculations that localizes a source by computing certain loci from known microphone array geometries. As an extension, this system would then control a motorized ‘Rig’ (turret) that would then turn towards the source and fire toy bullets at it. This of course would be the fun part of the project.

Learned Engineering Practices and Experience:

In the test bed creation stage, multiple considerations had to be kept in mind. For single channel analysis a portable system that can record in a maximum number of environments had to be developed. In acoustical applications, the specific non-linear elements of recording equipment can always affect the reliability of results by creating distortion of output waveforms. Thus the system should be able to record a single event using microphones of different types and companies so that by numerical comparison the raw data can be obtained.

The microphones we used were the Blue Yeti Studio microphone, inbuilt computer microphones, and small microphone 'kits' available from the 'Adafruit' website. These represent an equipment variety which ranges from very affordable to the high-end. Comparison of data from this range would allow us to measure the impulse responses of the acoustical channel between the source and the microphone, provided of course that distortions occurring from non-linearity in the source (be it a loudspeaker or some other) and the detector could be separated. The microphones were arranged in the tetrahedral shape that ensured spatial diversity of between the channels.

For multi-channel input to be collected a reliable high-speed acquisition system that fulfilled Nyquists' criterion (to at least five times the minimum limit, as per normal engineering conventions) and recorded simultaneously over multiple channels was needed. This meant implementing a system that operated at a sampling frequency of 100 KHz (to effectively cover the entire acoustic range) per channel and did this without synchronization problems that multiplexed Analog-to-Digital Converters suffer. The system would then either dump this acquired data onto a computer for later processing or would perform computations on these signals in real time.

This system was built using available National Instruments hardware, which primarily included the Compact Reconfigurable Input-Output (cRIO) setup and associated Input and Output modules. The on-board FPGA unit was programmed to minimize any latency in the data collection and processing. This would the control unit of the setup; acquiring multiple channels of sound and processing these signals for signal identification and localization.

An important practice that must be followed in an experimental setup such as this is the careful tagging and organization of data. As a significant amount recording had to be done, the recorded signals must maintain descriptive tags that describe not only what is being recorded, but also the location, orientation, and disposition of both the source and the detector (ambient conditions). A MATLAB algorithm would thus record for 10 seconds, then display the audio profile graph for viewing and, if the impulsive event was successfully recorded, would save the recording with a pertinent descriptive name.

However, another issue that plagues outdoor sound recording is the existence of background noise from passing cars and pedestrians, or even moderate wind movement. Recordings thus had to be collected as soon as winds died down and we found we could maximally utilize these 'quiet windows' by recording multiple impulsive events in the same 10-second window. An algorithm was then developed that identified these multiple impulsive events in a single recording (based on signal energy) and subsequently classified them. This optimizes data collection without sacrificing accuracy or precision.

This brings us to the identification of our 'gunshots,' or identification of impulsive events. We attempted approximating gunshots with balloons, smashing different material together, and firing toy guns, then comparing (via energy and window-normalized correlation) the recorded waveforms with that of an actual gunshot recorded at a firing range. Our results showed, via process of elimination, that the source best suited to our requirement were the toy guns. This, then, was our 'source.' Once the source was identified, it was necessary to model the medium that the sound travels through. To characterize this 'channel,' and it's impulse response (*H(s)*), we studied and employed the use of Dirac software (from the Acoustical Engineering firm) and a logarithmic sine-sweep analysis (first developed by Prof. Angelo Farina) to determine what the impulse response of an ideal channel should look like. The choice of this analysis method was straight-forward; of the four different methods used when identifying a channel's impulse response; the Maximum Length Sequence (MLS), the Inverse Repeated Sequence (IRS), Time Stretched pulses, and SineSweep (where a sinusoid of temporally increasing frequency, both linearly and exponentially, is applied via accurate speakers), the SineSweep method proves best for dealing with 'slightly nonlinear' systems (the loudspeaker or toy gun) that are coupled with linear systems. Studying a very small acoustical channel where the source and receiver were only 2 meters apart, we approximated that this was had a large Signal to Noise Ratio (SNR) margin, and was free from significant attenuation, reverberations, or echo. From a Signals and Systems perspective, this response should resemble a Dirac delta function as sound should travel these 2 meters without any material effect. Using different methods we obtained the time domain impulse response *h(t)* of this 2-meter long channel and it very closely resembled our toy gun signal. Thus, through a practice of employing many different methods we were fairly confident that our chosen toy gun source was a very good choice as an impulsive event signal. By being fairly certain of this choice, we are able to confidently make many design choices as well as useful conclusions from our data, without having to worry as to whether or not the things we are observing are due to reverberations or a less than ideal ambiguous source signal. Finally, once the source was identified, the channel characterized and the acquisition setup completed, we used National Instruments' hardware and LABVIEW to build a toy 'turret' that tracked source signals. The incumbent sound wave would be detected by the tetrahedral microphone array, the electro-acoustical signals processed by cRIO, and an array of motors would aim a toy 'nerf' gun at the source (essentially, a servo motor was made). Due to the FPGA hardware used, the system's 'lag' was minimal and the setup was used both as an entertaining toy and a research tool to further knowledge about acoustically reverberant systems and channels.