High Performance Air Acoustic Detection and Classification Sensor

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ABSTRACT

Acoustic signals are a principal detection modality for unattended sensor systems. However, the performance of these systems is frequently suboptimal due to insufficient dynamic range in small systems or excess power consumption in larger systems. This paper discusses an approach to developing an unattended ground sensor (UGS) system that has the best features of both worlds. This system, developed by McQ Inc., has exceptional dynamic range (> 100 dB) while operating at power levels of 1.5-5 watts. The system also has a user definable signal parameter library and automated detection methodology that will be described.

Keywords: acoustic, signature, unattended ground sensor, remote sensor, high performance, automatic classification

1. INTRODUCTION

Acoustic signals are a principal detection modality for unattended ground sensor systems due to the nature and variety of available targets. The breadth of targets is unmatched in other modalities and includes speech, gun shots, explosive blasts, ground vehicles, generators, helicopters, and fixed-wing aircraft. Unfortunately, the performance of many acoustic sensor systems is frequently suboptimal due to insufficient dynamic range in smaller systems or excess power consumption in larger systems.

McQ Inc. has developed an unattended ground sensor (UGS) that features the best properties of both worlds. The sensor has an exceptional instantaneous dynamic range of over 100 dB, while operating at power levels below 1 Watt. The sensor may operate in multiple mission-driven modes. In monitoring mode, the sensor performs target detection and classification against user defined signal and target parameter libraries. In data collection mode, the sensor collects and stores acoustic signature data in nonvolatile memory for post-mission analysis. The sensor design and system architecture are described and the system performance is presented.

2. SENSOR DESIGN AND ARCHITECTURE

The primary goal of the sensor hardware development was to provide a flexible, modular, high-performance platform that could be used for a variety of applications, from high-precision data acquisition for signature collection to real-time target detection and classification for sensor applications. The implementation features modular components, as shown in Figure 1, including separate mechanical enclosures for each power supply, the data acquisition electronics and the signal processing section. These enclosures each provide electromagnetic shielding, electronic filtering, and isolation to help control electronic noise throughout the system. Connections between enclosures are made with coaxial cable to bulkhead SMA connectors or, for differential signals, triaxial cables to bulkhead triaxial connectors fitted to each enclosure. The major components of the design are the power supplies, the data acquisition system and the signal processing system.

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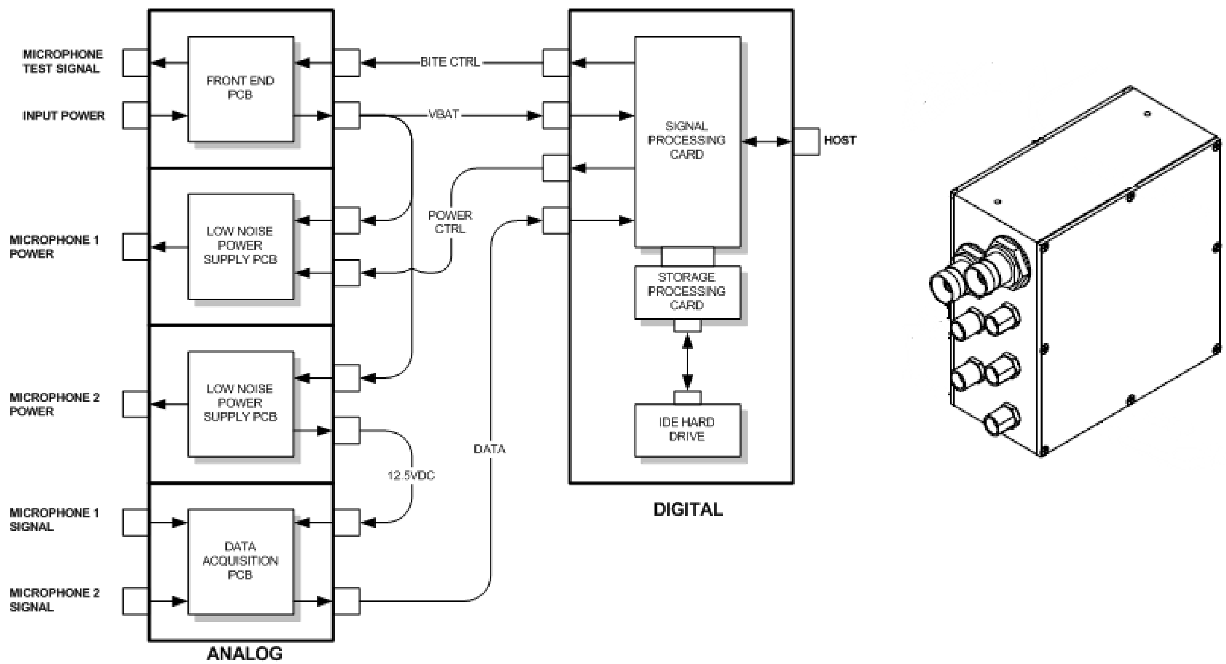


(a) (b)

Figure 1. (a) Sensor architecture illustrating system components, modular design and separate enclosures for each subsystem, shown by the heavy lines, (b) Example of mechanical enclosure

2.1 Power Supplies

Supplying power to various subsystems in the sensor is performed in a staged approach. The power supply front end, shown in Figure 1, provides reverse- and over-voltage protection and noise filtering. The resulting conditioned power is delivered to each subsystem where it is converted to voltages necessary for that enclosure only. At each interface and throughout each voltage conversion, noise suppression techniques are used to maintain the lowest possible system noise. This approach improves the overall power supply rejection ratio (PSRR) as well as the rejection of coupled noise within the system.

Within each enclosure, the conditioned power is converted to voltages used locally. The input power is filtered prior to the first stage regulator, which is generally a high-efficiency, low-noise switching regulator. Both fixed frequency PWM regulators and spread spectrum, variable PRF regulators were considered in the design, with the final decision for each enclosure driven by the impact of the noise characteristics on the electronics. The switching regulators are generally followed by linear regulation and additional filtering to further improve the local PSRR.

2.2 Data Acquisition Module

The data acquisition module, shown in Figure 2, is designed to amplify, filter and sample two differential analog signals and transmit the resulting digital data to the signal processing module. The differential signals are conditioned using balanced, active antialiasing filters to attenuate signals above the 16 kHz Nyquist limit. The goals of the analog design were to minimize crosstalk and pass band ripple, while maximizing the stop band attenuation. Performance achieved during module testing were a crosstalk of better than -115 dB and a ripple of less than 0.5 dB across the pass band and a stop band attenuation of over 80 dB.

The analog-to-digital converter (ADC) is a 24-bit delta-sigma converter with a specified dynamic range of 114 dB (or 19 effective bits). The ADC performs sampling, analog-to-digital conversion, and has additional internal antialiasing filters. The ADC generates 24-bit values for both inputs in a synchronous serial data stream at a sample rate of 32 kHz per channel. Key performance curves, illustrating the frequency response, two-tone dynamic range, channel crosstalk, and harmonic distortion, are shown in Figure 3. All performance data was measured on industry standard Audio Precision test equipment.

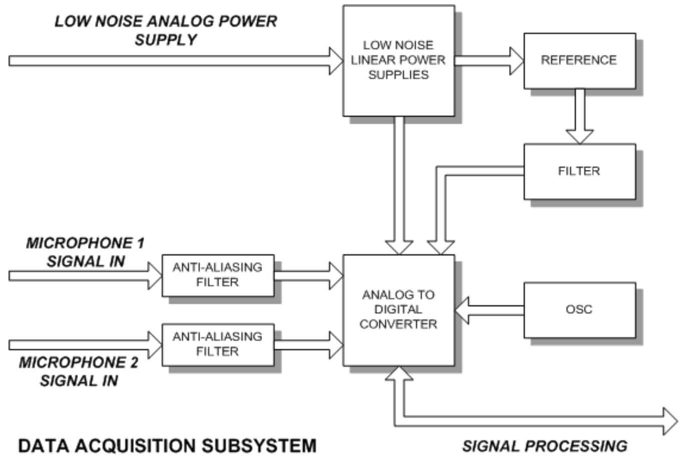


Figure 2. Data acquisition module architecture

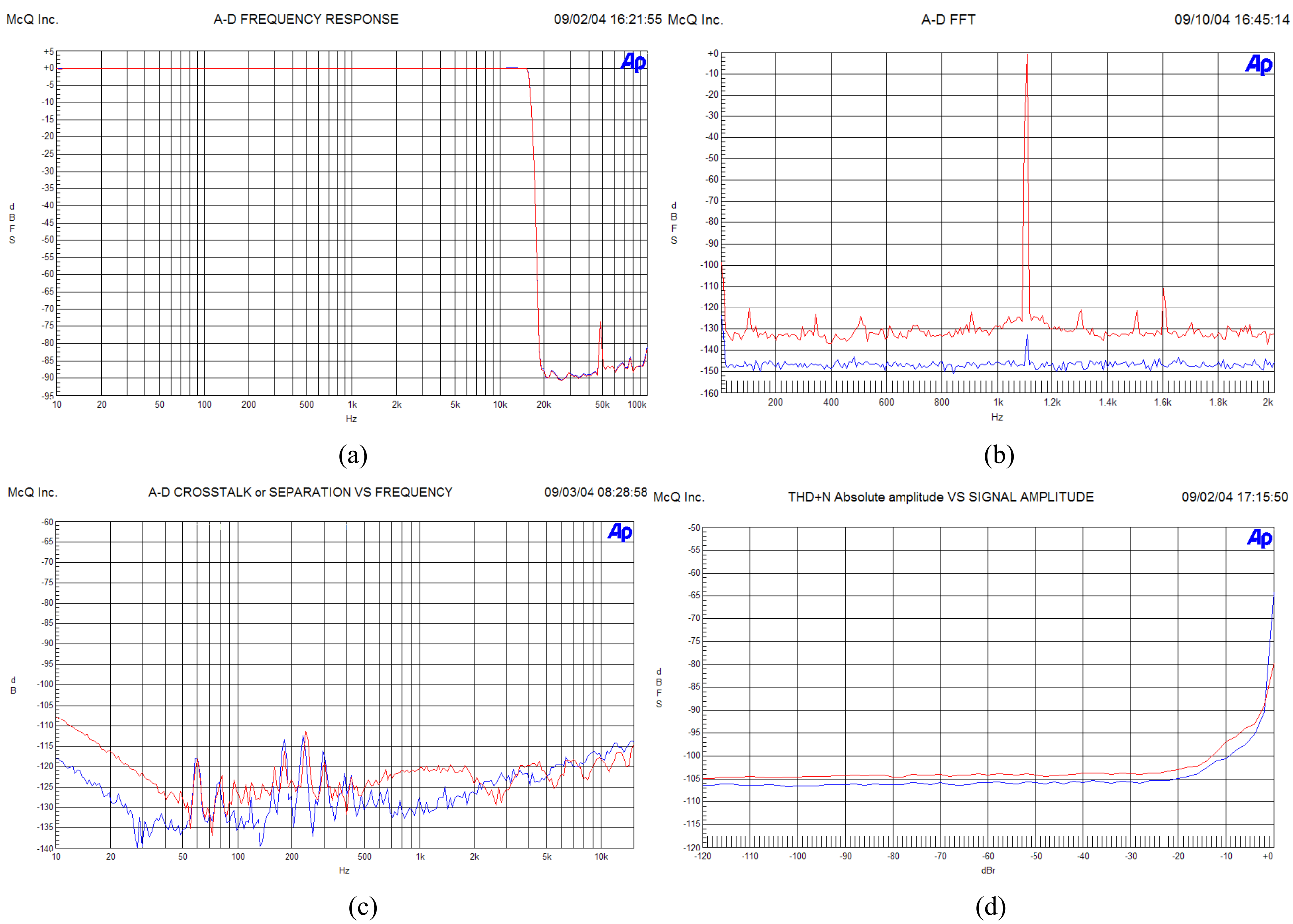


Figure 3. (a) ADC and analog front end frequency response with channel 1 in blue and channel 2 in red (input level = 1.862 V_{rms}, 240 frequency steps), (b) ADC and analog front end 2-tone dynamic range (Tone 1 = 1.1 kHz, Tone 2 = 1.6 kHz, Input level = 1.766 V_{rms}), (c) ADC and analog front end crosstalk (Input level = 1.852 V_{rms}), (d) Total harmonic distortion plus noise (variable signal amplitude) (color image in online version)

2.3 Signal Processing Module

The signal processing module receives digital acoustic data from the data acquisition module and analyzes the signals for the presence of targets. Upon detection of targets matching the user-defined signatures, the signal processing module may notify other systems via discrete signals or the RS232 serial interface. It may also automatically begin collection of the audio data onto resident compact flash memory or optional hard drive. Hard drive storage of the collected data allows over a week of uninterrupted signature collection.

Figure 4 shows a block diagram of the signal processing module. Controlling the signal processing module, as well as most aspects of the sensor, is a high-performance, high-efficiency DSP. The DSP peripherals include external memories for expanded volatile and nonvolatile storage, an RS232 serial interface for command and control, a battery-backed real time clock for time keeping, a compact flash for intermediate signal data storage and a PC104 interface for access to hard drive storage.

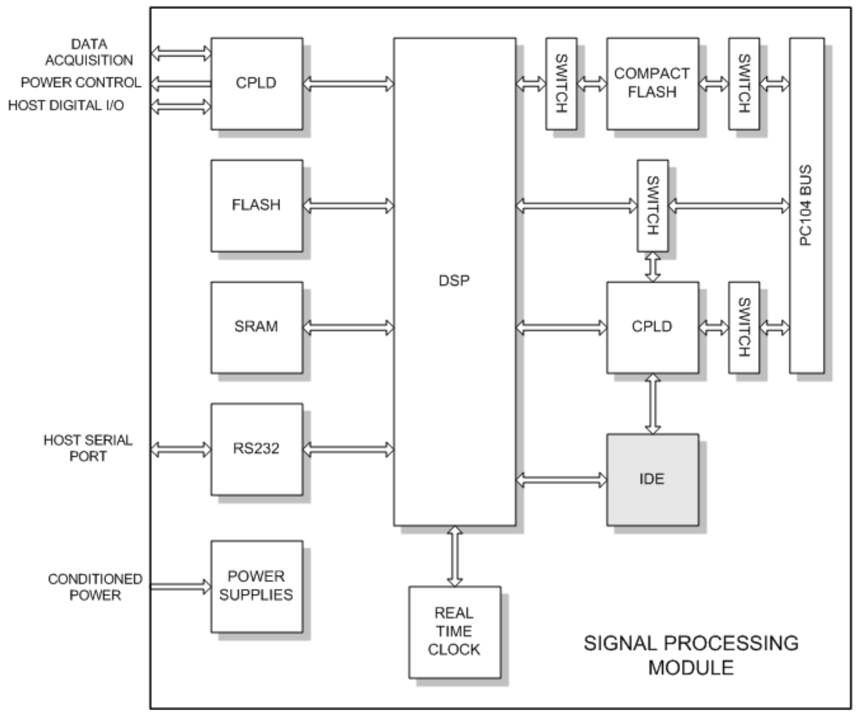


Figure 4. Signal processing module architecture

3. ENERGY CONSUMPTION

Energy consumption is a major consideration in unattended ground sensors due to the often difficult nature of sensor emplacement and recovery. In this sensor, the minimization of energy consumption was considered in all phases of the design. The hardware uses high efficiency power supplies and energy efficient electronic components. The software also uses a variety of techniques to conserve energy, such as adaptive peripheral power control and variable speed clocking. As a result, the sensor consumes less than 1 watt, running 32 kHz data collection in two channels or real-time acoustic signal processing algorithms.

4. TARGET SIGNATURE LIBRARIES

Target detection and classification is performed in real time using a novel user-defined database of target signatures. Each target is built as the combination of a set of user-defined signals and each signal is defined as one of three signal types: tone, transient, or pulse, with user-defined parameters. This approach allows the user to decompose the target's characteristics into its component signals, as described below.

4.1 Signal Libraries

Signals are defined as one of three types: tone, transient, or pulse, each with different parameters that describe the signal in objective measures that the signal processor can estimate. Each library, as shown in Figure 5, is detailed in the following sections.

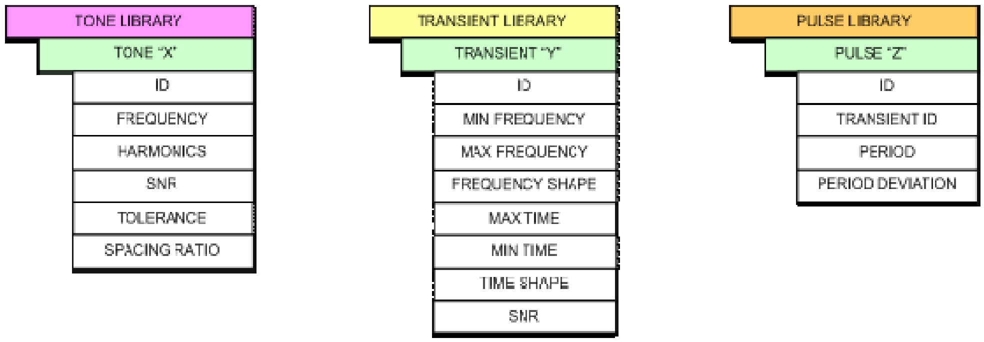


Figure 5. Signal Library Organization

4.1.1 Tone Signal Library

The tone signal library contains a set of entries that describe continuous tone signals. Each signal is defined by the following parameters: a unique tone identification number, center frequency, number of harmonics, signal-to-noise ratio, a tolerance for frequency shifting about the center frequency, and a spacing ratio. These signal definitions are used as discriminants to determine when a monotone signal is present.

- **Frequency** – The frequency is the nominal center frequency of the tone to be detected. The range of valid values for the frequency covers the frequency response of the sensor, 5-16000 Hertz.
- **Harmonics** – The number of harmonics is the expected number of harmonics of the detected tone that are concurrently detected. The sensor can associate up to 20 harmonics with a given tone.
- **Signal-to-Noise Ratio –** The SNR is the expected signal-to-noise ratio of the detected tone. The sensor calculates the signal-to-noise ratio as $\mathbf{10 \cdot \log_{10}} \frac{A_{Tone}}{A_{Noise}}$ in the power spectral density, where A_{noise} is an estimate of the noise around the tone (in the frequency domain).
- **Frequency Tolerance –** The frequency tolerance specifies an allowance for the deviation in the center frequency of the detected tone. The specification of a tolerance allows the sensor to reliably associate tones that vary statistically among individual units of a specific target.

4.1.2 Transient Signal Library

A transient signal library contains a set of entries that describe transient signals. Each signal is defined by the following parameters: a unique transient identification number, a minimum and maximum frequency, a frequency-domain shape, a minimum and maximum time, a time-domain shape, and a signal-to-noise ratio.

- **Minimum Frequency –** The minimum frequency is the lower limit of the frequency band over which the pulse is to be detected, covering the frequency response of the sensor, 5-16000 Hertz. The minimum frequency should always be less than the maximum frequency. The frequency band (from minimum frequency to maximum frequency) should be specified to cover at least several times the pulse signal bandwidth, but should exclude nearby signals when possible to accommodate the expected bandwidth of the signal plus a guard band surrounding the signal.
- **Maximum Frequency –** The maximum frequency is the upper limit of the frequency band over which the pulse is to be detected, covering the frequency response of the sensor, 5-16000 Hertz. Guidelines noted for the minimum frequency also apply to the maximum frequency.
- **Frequency-Domain Shape –** The frequency-domain shape is used to describe the expected shape of the power spectrum of the transient (i.e., amplitude vs. frequency). Shapes are chosen from a predefined set and a shape

matching algorithm computes the degree of correlation to the chosen shape.

- **Minimum Time –** The minimum time specifies the minimum duration of the transient event. This improves the rejection of false alarms due to noise within the detection bandwidth.
- **Maximum Time –** The maximum time specifies the maximum duration of the transient event. This allows the discrimination of transients versus continuous or fading tones within the detection bandwidth.
- **Time-Domain Shape –** The time domain shape is used to describe the expected shape of the band-limited amplitude over time of the transient (i.e., amplitude vs. time). Shapes are chosen from a predefined set and a shape matching algorithm computes the degree of correlation to the chosen shape.
- **Signal-to-Noise Ratio –** The SNR is the expected signal-to-noise ratio of the detected transient. The sensor calculates the signal-to-noise ratio as $\mathbf{10 \cdot \log_{10} \frac{A_{Peak}}{A_{Noise}}}$ in the power spectral density, where A_{peak} is the maximum amplitude in the specified transient bandwidth and A_{noise} is an estimate of the noise in the band, excluding the peak.

4.1.3 Pulse Signal Library

A pulse signal library contains a set of entries that describe pulsed signals. Each pulse is defined as a sequence of transients and contains the following parameters: a unique pulse identification number, a transient identifier, a pulse period, and a period deviation.

- **Transient Identifier** – The transient identifier specifies the transient in the transient library upon which this pulse is based. The pulse analyzer evaluates the periodicity of repeated transient detections to determine whether a pulse train is present.
- **Period** – The pulse period defines the expected pulse repetition interval for the specified pulse.
- **Period Deviation** – The pulse period deviation specifies a time window around the nominal pulse repetition interval into which the specified pulse must fit.

4.1.4 Target Library

A target library, as illustrated in Figure 6, is a set of entries that describe targets. Each target is defined by a set of associated signals, collected from the individual tone, pulse, and transient signal libraries. Each target entry consists of a unique target identification number, the total number of tone, pulse, and transient signals associated with the target, a minimum target report threshold, a minimum target record threshold, and a list of signals associated with the target. Each signal in the signal list contains the unique index of each signal, a signal type field, the minimum threshold for detection of each signal, and two pulse parameter fields.

- **Target Report Threshold** – The target report threshold is a confidence level above which this target would be reported. While the target confidence remains below this level, the sensor would only continue to monitor the associated signals.
- **Target Record Threshold** – The target record threshold is a confidence level above which the sensor would begin to record full-bandwidth signature data to nonvolatile storage. The sensor would continue operating in this mode for a specified period of time or until the nonvolatile storage is filled.
- **Signal Detection Threshold** – The signal detection threshold represents the required confidence that the specified signal was detected. Since this is a parameter of the signal only in association with this target, the signal may be reused for multiple targets with differing required confidences.
- **Minimum Pulses** – The minimum pulses parameter defines the minimum number of pulses that must be detected to result in a detection event for the associated pulse train. These need not be contiguous detections.
- **Maximum Missed Pulses** – The maximum missed pulses parameter defines the maximum number of pulses that may be absent from a train of pulses (e.g., with MP=10 and MMP=1, the sensor would be required to detect 10 out of 11 pulses to consider the pulse train detected).

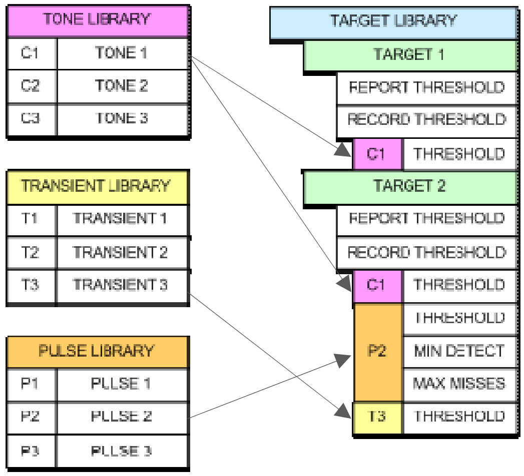
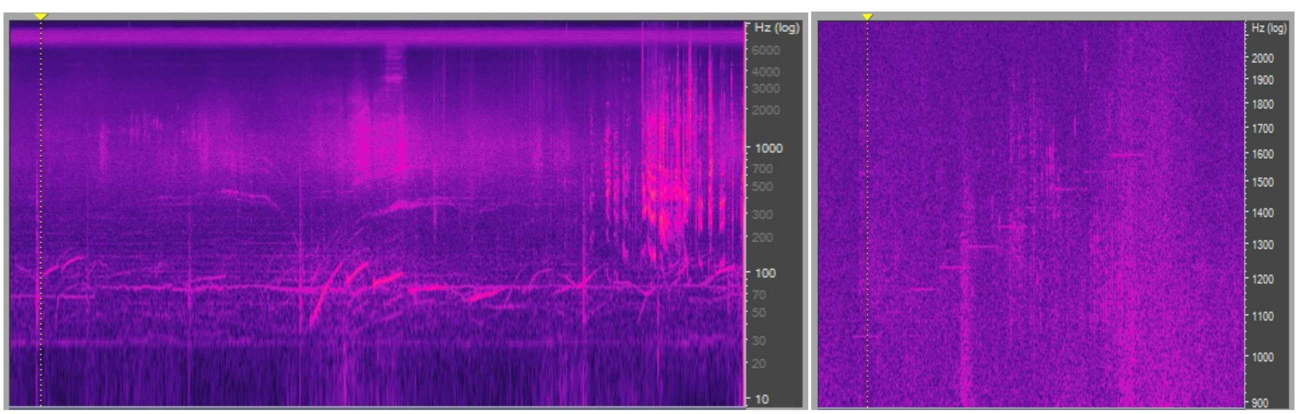


Figure 6. Target Library Organization

5. SYSTEM PERFORMANCE

The sensor was coupled with an ACO Pacific MK224 electret microphone and model 4012 preamp to optimize the sensitivity and dynamic range capabilities of the system. The microphone sensitivity, as reported on the calibration chart provided by ACO Pacific and verified using a Simpson calibrated sound level source, is 51.3 mV per Pascal. The output of the microphone and preamp is conditioned using a differential amplifier/filter for input to the ADC. The gain of the amplifier was set to match the measured dynamic range of the analog to digital converter. This provides the best sensitivity while maintaining the best available dynamic range. The range from the system noise to the start of measurable distortion was measured to be 115 dB.

The RMS noise floor of the microphone and analog board combination is equivalent to 1.2 μPascal per root hertz. Ignoring the very complicated real world acoustic propagation limitations and assuming only geometric spreading and atmospheric absorption, system noise limiting, and the 12 dB SNR required for detection; this sensitivity could provide detection of normal human voice (50 dB_{SPL}) at a distance of 500 meters. This sensitivity is generally 10's of dB below the ambient noise in most locations. The limiting factor in all of our field testing has been environmental noise and not the system noise. In areas around Fredericksburg, Virginia, even during the quietest times, the noise floor was limited by wind, cicadas, jets, distant vehicles, and other environmental noises, as illustrated in Figure 7.



(a) (b)

Figure 7. (a) Sensor-collected spectrogram illustrating complexity of "quiet" environmental testing, (b) Section of same sensor-collected spectrogram zoomed in time and frequency to show detected test signal at low SNR (for color image see online version of manuscript)

The system was tested for its ability to detect signals quickly. Two files containing test tones were generated at frequencies within the passband of the system. The first file contained an intermittent tone that lasted for 5 seconds and then quiet for 5 seconds. Each succeeding 5 second tone was at a level that was 6 dB louder than the previous tone. This file had eleven 5 second long tones providing a total source dynamic range of 66 dB. The starting level could be adjusted to the range of interest. This provided an automatic method of determining the source level required for the system to detect a five second tone.

The second test file generated a continuous tone that changed frequency every 10 seconds. The constant amplitude signal started at the lowest frequency and stepped in 10 even increments across the test frequency band. This was done to ensure that the system could detect all the frequencies queued in the library and that it would omit those not of interest.

Tests were conducted in several places around Fredericksburg, mostly in parks accessible to the public. Using the above described test tones with a 70 dB_{SPL} maximum level, detection ranges were achieved between 50 meters in a noisy park with lawn mowers in the distance and several hundred meters in a quieter park with a breeze, cicadas, and birds. Under optimum conditions, detection ranges in excess of 500 meters have been reported.

6. CONCLUSION

McQ Inc. has developed an unattended ground sensor (UGS) that features high dynamic range and high sensitivity. The sensor is designed for low energy consumption to enable battery-powered operation in remote locations. Sensor capabilities include multiple mission-driven modes of operation. A monitoring mode allows the sensor to perform target detection and classification against user defined signal and target parameter libraries. A data collection mode allows the sensor to collect and store acoustic signature data in nonvolatile memory for post-mission analysis.