DOD 14.2 SBIR PHASE I PROPOSAL*
Title: Advanced Processing Algorithms for
Ad-Hoc Ground Based Counter-Fire System
Firm: Aquerre Technologies LLC
Proposal Number: N142-086-0025
Topic Number: N142-086

1 Identification and Significance of the Problem or Opportunity

The mission of the Ground Based Counter-Fire (GCFS) system is to provide the Point of Origin (POO), Point of Impact (POI) of rockets, artillery, mortars, and IEDs, as well as identify the type of firing platform. The current GCFS system utilizes multiple remote Listening Posts (LP) that include an acoustic array placed on the ground, meteorology sensors to measure the environmental conditions at the listening posts, and a radio to communicate information between the listening posts and the Command Post (CP). These remote listening posts must be emplaced, maintained, and repositioned by a team of Marines. This is costly in terms of manpower, maintenance, and training. According to the USMC, the current GCFS system requires a team of 10 Marines and one High Mobility Multipurpose Wheeled Vehicle (HMMWV) to operate and maintain the system, and the new Ad-Hoc GCFS would reduce the number of Marines to 3 and eliminate the need for a HMMWV.

The problem of ballistic event detection and localization with acoustic sensor networks has received significant attention from the R&D community, and several technologies have been developed and operational to date [1, 2, 3]. In this SBIR, we leverage recent advances in sensors, communication and computing technologies to support the implementation of advanced processing algorithms for the next generation Ad-Hoc GCFS. The main novelty in our approach is to exploit the capability of advanced sensor devices (*e.g.* MEMS microphones and inertial sensors) and communication protocols and hardware (*e.g.* power and bandwidth efficient wireless systems on chips) to support improved signal processing and ballistic event detection and localization capability using a flexible, robust *ad hoc* sensor network.

The Ad-Hoc GCFS will further benefit from increasingly accurate satellite and terrestrial positioning and timing systems, in some cases providing sub-centimeter position accuracy. Localization algorithms typically use relative time-delay of signal arrival measurements to determine the location of an acoustic source, hence inter-node time synchronization and position errors play a fundamental role in determining the performance of source localization algorithms. In addition, MEMS sensor technologies and low-power micro-processing and RF wireless communication technologies have advanced to a state of maturity such that compact, robust high-performance battery powered sensor-processor-radio units are now realizable within a practical budget.

A key feature of our proposed Ad-Hoc GCFS concept is the capability to exchange high fidelity sensor measurements including digital signals from the microphone arrays between LPs/RLPs (LP to LP) and the CP (LP to CP). This capability enables a broader range of signal processing options at the LPs and CP–including beamforming with *ad hoc* array data–while requiring increased bandwidth for the inter-node RF communication links. In this SBIR we propose to develop and evaluate algorithms for the Ad-Hoc GCFS using inter-node streaming audio to improve detection and localization performance. We plan to demonstrate that our algorithms can achieve improved detection and estimation performance within a reasonable processing delay deadline. We will further investigate the bandwidth-performance trade-off associated with sending compressed microphone data between nodes of the *ad hoc* sensor network.

In addition to digital audio signals from multiple reporting LPs/RLPs, our advanced system concept utilizes data pertaining to LP/RLP location, configuration, array orientation, local temperature and pressure, as well as other available data such as event report time, event bearing, and uncertainty surrounding event bearing. Existing GCFS methodologies (*e.g.* bearing crossing analysis) will serve as a baseline of available performance on top of which our advanced algorithms may be used to provide enhanced tactical information from the acoustic sensor network. Asynchronous reporting will be supported with an IP based core network and standard Ad-Hoc network management protocols will be used to address the challenge of RLPs dropping off or being added to the network as Marines use the vehicles (in our initial testbed system, the WiFi Ad-Hoc Protocol will be used).

Our Phase I work plan includes the development of an advanced GCFS algorithms simulation environment. Phase I.A includes further development of the simulation and algorithms software package, in addition to initial studies of algorithm performance with an experimental wireless sensor network testbed. Phase I.A testbed studies are expected to facilitate Phase II prototype development and ultimately the successful transfer of our technology to Navy systems.

The Phase I and Phase I.A Ad-Hoc GCFS Algorithm Development Tool (AGADT) will be delivered to the Government as part of this SBIR work program. In addition, the AGADT will be used demonstrate the simulated performance of the proposed algorithms throughout the program. Software development will be performed with GNU Linux and C/C++, using open software tools, such as Octave, where convenient. The proposed deliverables further include a (pre-Phase-II) study of algorithm performance with our Phase I.A testbed. The Phase I.A testbed will be assembled from COTS components, and in particular, we will use IEEE 802.11n (WiFi) protocol for its *ad hoc* radio connectivity and high-speed communication links.

2 Phase I Technical Objectives

The following is a set of objectives, paraphrased from the topic description, for which the proposed Ad-Hoc GCFS concept will be developed to accomplish:

- Utilize data from up to 10-20 total listening posts, composed of a mix of RLPs or static LPs.
- Determine the POO/POI and time associated with all events covered by the GCFS mission. Deconflicting multiple reported bearing angles from multiple Listening Posts and determining accurate POO/POI is viewed as a key technical challenge for this effort. Since acoustic sensor arrays can locate a sound event in bearing only and not the source location, correlation and deconfliction of many intersecting bearing angles to the correct point of origin is very challenging. This is compounded by asynchronous reporting of acoustic events (due to speed of sound) and effects of weather and terrain.
  - The GCFS will determine the POO of any indirect firing platform within range of an LP/RLP, with a 90% probability of successful event localization. In other words, if an indirect platform fires its weapon and is within range of an LP/RLP the GCFS will be able to locate it with at least 90% accuracy.
  - The GCFS will determine the POO and POI location to an accuracy of 2% of range between the closest listening post.
- Display the information on an overlay map for utilization by an operator.
- Store the POO/POI information.

The work plan and methods for achieving these objectives are detailed in the following section.

3 Statement of Work (including Subcontractors' Efforts)

The Statement of Work is organized as follows: Section 3.1 summarizes a schedule of major events including tasks planned, methods planned to achieve each objective or task, and the final products to be delivered in Phase I and Phase I.A (Phase I Option). Section 3.2 provides a detailed technical description of the preliminary channel and signal models, detection and estimation framework, and numerical results from a pre-Phase-I multiple gunshot detection problem. Section 3.2 contains proprietary data. Moreover the content of Section 3.2 is subject to further analysis and development throughout the Phase I program and beyond. We plan to work closely with the sponsor to make sure our ongoing development activities continue to address the specific needs of the sponsor and most importantly the needs of the end-user of the technology. Hence the Phase I/I.A algorithm development track will include ongoing feedback from the numerical evaluations, testbed experiments and sponsor feedback in order to facilitate development and transition of a successful GCFS technology.

Section 3.1.2 includes description of the Phase I.A Testbed, consisting of an experimental wireless acoustic sensor network hardware/software system assembled out of Commercial Off-The-Shelf (COTS) hardware, open software tools, and custom software designed by Aquerre Technologies. We are committed to working closely with the sponsor to ensure that our technology solutions yield a positive return for the Government in terms of fulfilling specific NAVY/MARCOR strategic needs. The use of Discretionary Technical Assistance (DTA) is not included in this proposal.

3.1 Schedule of Major Events

3.1.1 Phase I

The objective of the six month Phase I effort is to determine the technical feasibility of the proposed concept by means of analysis and numerical simulation. In Phase I, we propose to develop (a) a detailed mathematical model for the propagation channel and signals of interest for the application of ballistic events detection, classification and POO/POI localization with an acoustic sensor network, (b) advanced algorithms based on optimal signal detection and estimation theory and practice, and (c) a detailed computer simulation environment for testing the proposed algorithms against model assumptions. By the end of Phase I, we will have a complete specification of the system model and a specification of the initial algorithm designs, in addition to analysis/simulation of their performance. The final products delivered at the completion of Phase I are:

1. Detailed technical specification of the channel and signal models,
2. Detailed technical specification of the Phase I algorithms,
3. The Ad-Hoc GCFS Algorithms Development Tool (AGADT) (Release I) for simulating the Phase I system model and algorithm performance, including software documentation,
4. Phase I midterm progress report (delivered midway through Phase I), and
5. Phase I final report and road map.

Aquerre Technologies is committed to meeting all NAVY/MARCOR SBIR reporting and documentation schedules and requirements. Details of our Phase I/I.A approach are contained in the following sections.

Phase I Modeling Studies: Acoustic propagation channels can exhibit complex and nonlinear behavior. However, a common approach to system design is to assume the propagation channel from source to receiver is Linear and Time-Invariant (LTI). Linear System Theory is then applied to analyze and develop signal processing methods. We also start from the LTI assumption, in which case the channel to any given microphone can be viewed as a series of rays corresponding to source signal copies arriving via multiple reflected or Line-of-Sight (LOS) paths (a *multi-path channel*). The relative attenuation and delay of the channel multipath components varies from mic to mic. Advanced signal processing algorithms exploit relative timing and amplitude information from multiple receivers to obtain accurate characterizations of detected events.

In addition to standard LTI model assumption, our Phase I work plan includes the study and evaluation nonlinear acoustic propagation models. For example, shock wave propagation (*e.g.* shockwave from bullet or other supersonic object) exhibits time-dilation as the shockwave "pulse" travels outward from the moving projectile. Time-dilation is a nonlinear characteristic that cannot be modeled with the "shift-overlap-and-add" LTI propagation model. Our Phase I research will determine the complexity and performance tradeoff associated with the use of increasingly sophisticated trajectory and acoustic propagation models.

Phase I/I.A Simulation Tool: The AGADT (Ad-Hoc GCFS Algorithms Development Tool) for GCFS algorithms development and simulation will be implemented in a GNU Linux C/C++ development environment. Preliminary code development in support of this proposal was performed using the GNU Octave matrix algebra computational package. Our simulation tool developed in Phase I (and Phase I.A) will play a critical role in determining the performance of modeling and algorithm strategies as well as physical requirements of the acoustic sensor network hardware for Phase II and beyond. The simulation tool will further be capable of assisting system design choices such as as antenna array geometry, number and location of LPs, sensor data requirements such as timing and positioning accuracy, and power and bandwidth requirements to meet mission objectives. Dimensions of the required hardware resources are impacted by the performance demands of the proposed algorithms, and specification of Phase I revised hardware requirements will be supported by concurrent development of the simulation tool. The cost proposal to this submission includes the purchase of four quad-core Mac Mini's as material supplies to support the simulation studies throughout Phase I/I.A and beyond.

Our proposed signal abstraction relies on acoustic signatures of the events of interest and sources of interference, including rocket, artillery and mortar fire, IED blast fire, and platform noise *e.g.* HMMWV engine noise. The Ad-Hoc GCFS will address classification of event type by comparing to a large collection of waveform test data, specifically signals collected from measurements of real events and analytic models of signature waveforms. In developing the AGADT, we will address the issue of signature waveform data for the events covered by GCFS mission by utilizing a combination of (i) audio recordings obtained from sound effects vendors and (ii) synthesis of signature waveform data from physical models. The Phase I approach to waveform data will give us flexibility to make rapid progress in algorithms development using uncertain and quasi-real data. See Section 3.2.3 for additional information regarding the proposed Phase I signature waveform data methods.

Finally, the AGADT will allow us to make performance/complexity evaluations of the proposed algorithms. In particular, we seek to demonstrate the proposed algorithm's ability to determine the POO and POI of all events covered by the GCFS mission with performance characteristic specified in Section 2. We address the issue of de-conflicting multiple reported bearings from up to 10 to 20 LPs, by selectively polling some or all of the audio data across the *ad hoc* network and centralized signal processing of remote acoustic array data at the CP. Hence, while requiring more bandwidth and processing power at the CP, centralized processing of multiple remote audio signals enables a single location estimate with superior accuracy as compared to bearing crossing analysis based on multiple individual estimates from sensors in the field. In addition, this proposal includes current GCFS methods as a baseline upon which our advanced methods may be enabled for enhanced performance.

Phase I/I.A Detection and Estimation Framework: We emphasize a multi-detection and estimation framework based on multiple criteria of optimality, including Maximum Likelihood (ML), Bayesian, and Minimum Mean Squared Error (MMSE) methods of detection and estimation. The ML detector chooses the hypothesis that maximizes the probability of observing the measured data. ML offers the best possible performance for a given system model by choosing the most probable explanation (*e.g.* event type, origin and impact location) of signals measured by the network. Due to the exponential complexity of evaluating the likelihood of every possible configuration of system variables (event type, origin and impact), ML is prohibitively complex to implement directly. In 3.2.3, we provide the simulation results of a ML gunshot detector in which the location of the shooter(s) and angle of bullet trajectory are constrained to a small set of possible values.

In Phase I we will explore the application of Bayesian estimation and MMSE beamforming as complexity reducing alternatives and complimenting methods to the ML based framework. *Bayesian estimation* involves averaging over unknown variables using models/estimates of their marginal probability distributions. *Beamforming* consists of focusing the microphone array data on specific points in space and scanning across a grid of such points in search of signature waveforms of interest. The Minimum Mean Square Error (MMSE) beamformer combines the diverse audio signals collected from multiple LPs in such a way that minimizes the mean squared error to signals transmitted from a given point in space. The Principle Investigator (P.I.) to this proposal studied the use of ML, Bayesian and MMSE methods, including receive beamforming with antenna arrays, for signal detection and estimation in multi-antenna Radio Frequency (RF) communication systems in his Doctoral Thesis at the University of California, Santa Barbara [4].

Phase I General Architecture of Sensors Needed to Meet the Requirements: Our proposed acoustic sensor network leverages inter-LP and CP data exchange of raw digital sound pressure signals to increase accuracy of the outputs of interest. By polling neighbor LPs the CP may obtain diverse measurements of events of interest. The CP benefits two-fold from multi-LP audio data: (i) enables signal combining methods that reduce effective noise and interference and (ii) yields geometry diversity, wherein a geographically distributed LP array may yield better measurement data for the purposes of event localization than an individual microphone array.

In order to obtain high-quality multi-channel buffered and streaming audio data from multiple LPs to a CP we initially assume a fairly wideband channelization requirement on the order of

$$(4 \text{ mics}) \times (40 \text{ kHz}) \times (24 \text{ bits/sample}) + (\text{low rate sensor data}) \approx 4 \text{ Mbits/s} \tag{1}$$

per LP/RLP. Equation (1) represents a coarse estimate of payload requirements for a hypothetical 4 element microphone array streaming at 40 kHz per channel and 24 bits per sample from one LP to a CP. Our estimate of 24 bits ADC is due to the wide dynamic range of signal amplitudes resulting from the variable proximity to source [2] and the sample rate requirement of 40 kHz is based on the Nyquest sampling rate for a typical MEMS microphone [5]. The estimated bit rate payload requirement (and other system requirements) of our proposed GCFS will be refined throughout the Phase I and Phase I.A segments by means of analysis, simulation and testbed development studies.

We further assume that extremely accurate location and timing data are available to the LPs using multi-constellation Global Navigation Satellite System (GNSS) receivers and corrective signal-processing with terrestrial and space based augmentation systems. Error ranges for civilian state-of-the art timing and positioning GNSS ICs are on the order of 10 picoseconds/1 centimeter using RTK (GPS base station reference receiver), and 1 nanosecond/1 meter without RTK (DGPS) [6]. Military timing and positioning ICs are assumed to have an accuracy comparable to the civilian system using RTK. The Phase I simulation tool will be used to understand the effect of location and timing uncertainty by adding pseudo-random noise to the location and time variables.

Other required sensor data includes: temperature and humidity sensor data (correct speed of sound [7]) and gyroscopic sensor data (pitch and roll orientation of the microphone array). Section 3.1.2 has an example commercial MEMS Gyroscopic IC with 3-D orientation outputs.

3.1.2 Phase I.A

Phase I.A (Phase I Option) is scheduled for the six months immediately following the completion of Phase I and preceding the start of Phase II. In Phase I.A, we will continue development of AGADT, in addition to development of an experimental wireless acoustic sensor network testbed. A key contribution of the Phase I.A segment will be the development of a custom hardware/software testbed assembled out of COTS components for studying the performance of Phase I algorithms and beyond. Using actual acoustic propagation channels (instead of channel models), with real microphone sensor arrays, GPS and environmental data will ensure that our methods are matured to a state of readiness for transfer to DOD systems.

The Phase I.A Testbed development studies will be conducted in a benign sound location/classification environment. In addition to advancing algorithm development towards transition to Navy/MARCOR systems, the Phase I.A Testbed will be used to identify specific hardware components needed to meet performance objectives. Moreover, we seek to ultimately develop a complete hardware/software solution available for commercial purchase.

The Phase I.A deliverables are as follows:

1. The AGADT Software Package and Supporting Documentation (Release I.A),
2. Pre-prototype COTS system design and development studies report,
3. Phase I.A midterm progress report, and
4. Phase I.A final report and Phase II proposal.

Sketch of Phase I.A Testbed: We plan to use dual-use the Mac Mini's purchased in Phase I as the main processing unit associated with each test LP/RLP. The Mac Mini comes with a WiFi (IEEE 802.11n) radio interface (2.4GHz/5GHz unlicensed spectrum), which can be configured to operate in Ad Hoc Network mode (Peer-to-Peer Network) with peak data rates up to 100 Mbps at a range of 20 meters or more. WiFi is a convenient standard for RF wireless development due to its widespread adoption and high data rates. Our future system might use a next generation wideband wireless standard, such as LTE-Advanced in unlicensed frequency spectrum. We plan to use USB 2.0 to attach sensor components such as microphone array, GPS receiver, temperature and humidity sensors, gyroscopic and possibly a chirp amplifier/speaker. Code development will be performed in the Debian Linux environment, using open libraries and open software whenever possible, such as the Debian `libiw-dev` (wireless development tools), GNU Plot (output rendering), GNU Octave (matrix algebra), GNU C/C++ and associated libraries. Proprietary libraries such as sensor API libraries will only be used as needed.

In Phase I and I.A, certain hardware components may be over-dimensioned and our research and development efforts will include a focus on reducing the resource requirements of individual components (*e.g.* power and bandwidth requirements) while still providing the desired level system performance. A brief initial search yielded the following components with the capability to support our advanced GCFS concept:

- Trimble BD970 GNSS Receiver Module [6]: Multi-satellite tracking from multiple satellite navigation systems: GPS, Galileo, GLONASS, Compass, QZSS. Sub-centimeter accuracy using RTK. Positioning outputs at 50 Hz.
- STMicroelectronics L3G4IS MEMS motion sensor [8]: 3-axis digital output gyroscope, includes 8 bit temperature data output ($\pm.25°C$) and 16 bit yaw, pitch and roll data at approximately 100 Hz.
- STMicroelectronics MP45DT02 MEMS audio sensor omnidirectional digital (PDM) microphone [5]: Wideband response 50Hz - 15kHz, top port, surface mountable, compact, good for array applications.
- STMicroelectronics HTS221 Capacitive digital sensor for relative humidity and temperature [9]: 16-bit humidity and temperature output data at 12.5 Hz.
- Wolfson Microelectronics WM5102 Audio Hub CODEC with Voice Processor DSP [10]: Supports 6 channels ADC/PDM at multiple sample rates up to 192kHz at 24 bits precision.

3.1.3 Phase II and Beyond

Results from Phase I/I.A development activities will be used to guide specification of prototype algorithms during Phase II. Phase II prototype activities may include: testing on DOD systems and fabrication of a custom wireless sensor prototype. A complete Phase II proposal will be delivered as part of the Phase I.A work program.

Throughout this SBIR we will seek to actively participate in commercialization assistance programs such as the Navy Transition Assistance Program (TAP) (Phase II) and the DOD Commercialization Readiness Program (CRP) (Phase II.5).

3.2 Technical Approach

3.2.1 Signal Model

Suppose we a have a database of signature waveforms associated with a given set of ballistic events (*e.g.* artillery, mortar, rocket, IED, etc.) or other relevant event types (*e.g.* wind, platform, and other environmental noise). In the following, we denote such a database as

$$\mathcal{S} = \{s_k(t) : k = 1, \ldots, K\}.$$

Each element of $\mathcal{S}$ represents an acoustic sound pressure waveform characteristic of a ballistic event of detection. Any given event may be associated with multiple characteristic waveforms such as POO, In-Flight and POI signature waveforms.

Next, define

$$\Gamma = \{\gamma_q : q = 1, \ldots, Q\}$$

as the set of possible realizations of *individual* events, where each γ_q is comprised of an event type, POO, POI and time offset of the event. The variable Q represents the number of possible states per individual event. And, finally, let

$$\Theta = \{\theta_m : m = 1, \ldots, M\}$$

denote the coordinate location vectors of the M microphones of the sensor network.

The GCFS concept proposed in this paper jointly detects the event signature waveforms by combining the received signals across the microphone arrays. Depending on the realization of the pth event $\gamma'_p \in \Gamma$ and the system geometry Θ, the received signal at the mth microphone can be expressed as:

$$y_m(t) = \sum_{p=1}^{P} r_m(t; \gamma'_p, \Theta) + n_m(t), \quad 0 \leq t \leq T,$$

where P denotes the number of concurrent events in the time window $[0, T]$ and $n_m(t)$ denotes Additive White Gaussian Noise (AWGN). As detailed in Appendix A, the acoustic propagation channel is modeled as LTI and the output signal $r_m(t; \gamma'_p, \Theta)$ corresponding to the mth microphone and pth event is given by a superposition of convolution integrals whose inputs are the signature waveforms associated with the event type, as follows:

$$r_m(t; \gamma'_p, \Theta) = \sum_{j=1}^{N_p} \int h_{m,p,j}(\tau - t_p) s'_{p,j}(t - \tau)\, d\tau, \quad 0 \leq t \leq T,$$

where $s'_{p,j} \in \mathcal{S}$ denotes the $j$the signature component of the pth event, N_p denotes the number of signature waveforms associated with the pth event, $h_{m,p,j}(t)$ is the channel impulse response corresponding to microphone m, event p, and component signature waveform j, and t_p is the time-offset of the source event. The channel impulse response includes the time delay of propagation between the source and destination as well as any multipath components in the received signal.

In this proposal, we use the example of gunshot detection with acoustic arrays due to the ready availability of data and also because it has all the basic elements of the more general ballistic event signal model. In the preliminary (Pre-Phase-I) results that follow, detection of multiple possible concurrent gunshots is studied for a low-complexity ML detection scenario.

A typical acoustic waveform associated with a gunshot is depicted in Figure 1(a). Depending on the POO and POI in relation to the microphone element, the received signal will include the precursor shockwave (due to supersonic velocity of the bullet) and muzzle blast (POO) signature waveforms depicted for example in Figure 1(b). The shockwave shown is generated based on a physical model [11] and the muzzle blast waveform sample was obtained from an audio sound effects vendor (`blastwavefx.com`).

Digital Signal Model: The preceding continuous-time model can be expressed in discrete-time as follows: Let f_s denote the sampling frequency. We initially assume that the input is sampled past the Nyquest frequency of the microphone receiver bandwidth, *e.g.* $f_s = 40$kHz, with negligible quantization noise, *e.g.* 24 bits/sample. With these assumptions and a Riemann approximation to the channel convolution integral, the digital received signal at microphone m can be expressed as:

$$y_m[n] \approx \sum_{p=1}^{P} \sum_{j=1}^{N_p} \sum_{l=1}^{N_{ch}} h_{m,p,j}[l] s'_{p,j}[n - l] + n_m[n], \quad n = 1, \ldots, N_T$$

where

$$y_m[n] = y_m(n f_s^{-1}), \; n = 1, \ldots, N_w$$

CP. In light of these considerations, our Phase-I proof of concept will demonstrate the efficacy of *ad hoc* network beamforming methods for spatial isolation of signals of interest and the effect of signal compression on beamforming performance. The P.I. has research experience in the application of beamforming methods to Multiple-Access (MA) and interference suppression for mobile RF communication systems [17, 18].

4 Related Work

The problem of acoustic localization of ballistic events is well studied and the main novelty of our proposal lies in exploiting recent advances in sensors, communication and computing technology in support of advanced algorithms concepts and implementation. Estrin *et al.* (2003) [19] describe an *ad hoc* acoustic sensor network based on iPAQ PDAs, with RBS timing synchronization and an "audio server" running at each sensor that sends requested audio data samples to a centralized processor. Their system used offline post-processing using "approximate-ML" criterion to locate a speaker-emulated vehicle moving through the sensor field. Far-field localization was performed by centralized bearing crossing analysis of bearing estimates reported from multiple sensor clusters.

US Army Research Laboratory (ARL) published results on an acoustic mortar and rocket detection and localization system developed at the request of Iraq-based units [2]. Up to four sites were operated in Iraq detecting and locating insurgent mortar and rocket fires as of 2004. The algorithm estimates location of event by analyzing the crossing of bearing estimates from multiple LPs. It further used manual identification of POO and POI with automatic methods in development. The paper notes that "the mortar launch signature is predominantly in the 100 Hz region, while the impact signature contains more energy and wider frequency content."

ARL also published results on detection and estimation of aeroacoustic shock waves generated by supersonic projectiles [11]. Their paper demonstrated the results of a reduced complexity wavelet-based detector that identifies the leading and trailing edges of the impulsive shock waveform generated by supersonic projectiles. They showed that the algorithm was effective for detecting simulated shock wave signatures of a 5.56 mm projectile at a miss distance of 220 meters in the presence of HMMWV engine noise at 1 meter, with sampling rates as low as 15 kHz. Their efficient wavelet-based approach is interesting as a possible means of searching for certain-types of events by focusing on their high-frequency shockwave characteristic.

In [1] (1996), the concept of a low-cost distributed acoustic counter-sniper system concept is developed. The system is comprised of omni-directional, tetrahedral arrays, and helmet mounted arrays that provide low data rate partially processed acoustic data and logistics support data to a command node by RF wireless links. The algorithms use shock wave and muzzle blast wave edge detection to determine the time delays at all the mics and Minimum Mean Absolute Error (MMAE) criterion to fit delay data with the bullet acoustic signature model. The trajectory model accounts for de-acceleration due to drag force on the bullet.

Finally, [20] develops a new interpretation of Capon criterion which can be used to construct beamforming combining vectors. Capon criterion is equivalent (up to scalar multiple) to the MMSE criterion for generating beamforming solutions. Our Phase I effort will compare beamforming strategies with other detection and estimation criterion, including ML based strategies and Bayesian inference methods. See also [18] for MMSE estimation in the context of beamforming arrays and [16] for methods of reduced-complexity implementations of ML detectors for RF communication receivers.

5 Relationship with Future Research or Research and Development

If the proposed design concept is successful this SBIR will result in the development and commercialization of an advanced acoustic battlefield awareness system, with a level of accuracy and sophistication not seen in previous systems. Our strong knowledge base in optimal detection and estimation theory and practice, algorithms development and transition to product, and our diverse network of relationships with academic and industry researchers in the field will drive this SBIR and future R&D opportunities related to this SBIR.

The Schedule of Major Events (in Section 3.1) is partitioned to facilitate a smooth ramp-up towards commercialization of the proposed product concept. The Phase I effort will play a crucial role in demonstrating initial proof-of-concept by means of mathematical analysis and numerical simulation. In Phase I.A, we continue development of the Ad-Hoc GCFS Algorithm Development Tool (AGADT), in addition to test studies of Phase I algorithms with an experimental COTS wireless acoustic sensor network. By the start of Phase II, we will have the second iteration of the algorithm specifications, simulation tool, and physical resource requirement specifications of the component sensor, signal processing, and communication hardware. The major milestones of the currently envisioned Phase II segment will be (i) specification of prototype Ad-Hoc GCFS algorithms using feedback based on advanced field testing and (ii) fabrication of a custom prototype wireless acoustic sensor chip. Our Phase II productization plan will be based on the results of Phase I and Phase I.A R&D feasibility and pre-prototype studies.

This SBIR is ITAR restricted and the Phase II work program of this SBIR application may require security clearance(s). We note that all personnel performing work on this contract are U.S. citizens and that the P.I. does not currently possess a security clearance. In the event that such clearances are necessary to conduct Phase II testing, we will work closely with the sponsor to ensure timely completion of the applicable processes, certifications and approvals.

6 Commercialization Strategy

The first and primary target application for the proposed technology is the next generation NAVY/MARCOR Ad-Hoc GCFS. Aquerre Technologies seeks to be the primary vendor of advanced signal processing algorithms and simulation tools for the next generation Ad-Hoc GCFS. Our custom solutions will be specifically tailored and maintained to satisfy ongoing NAVY/MARCOR strategic needs. A secondary commercialization objective will be to obtain patent rights if appropriate, and lastly we will seek to have a viable commercial product for purchase outside of the GCFS market.

According to Intel Corporation [21], the Digital Surveillance Security (DSS) market represents a $40 billion industry as of 2013, with retail, education and public sector (government/transit) leading growth. Although typically associated with video surveillance systems, acoustic surveillance represents an integral component of a complete digital surveillance solution, especially in cases where acoustic signatures can be detected at further ranges than visual signals, and for augmenting video surveillance systems with acoustic sensor awareness.

We claim that a complete acoustic sensor network solution for ballistic events detection can be brought to market within the typical time-frame and budget of a multi-phase SBIR program. If our proposal is successful, we would apply/participate in the DON TAP program and DOD CRP program. Our in-house R&D experience (see P.I. CV) prepares us for transitioning software/algorithms to product in Phase II. Additionally, we would seek external assistance for bringing a custom hardware prototype/manufactured solution where appropriate.

DOD SBIR Award searches reveal a competitive small business landscape in the defense technologies market. A title, keyword, and abstract award search for the terms “acoustic array” yielded the following companies with recent DOD SBIR funding:

- Adaptive Methods, Inc, Centreville, VA: Navy Phase I 2011 $79,966 (`www.adaptivemethods.com`)
- AEgis Technologies Group, Inc., Huntsville, AL: Army Phase I 2012 $99,952 (`www.aegistg.com`)
- Interdisciplinary Consulting Corporation, Gainesville, FL: Army Phase I 2012 $99,826 (`www.iconsultcorp.com`)
- NAVMAR APPLIED SCIENCES CORP., Warminster, PA: Navy Phase I 2007 $69,976 (`www.navmar.com`)
- PHYSICAL OPTICS CORPORATION, Torrance, CA: Army Phase II 2006 $729,990 (`www.poc.com`)
- PROGENY SYSTEMS CORP., Manassas, VA: Navy Phase I 2004 $99,299 (`www.progeny.net`)
- SeaLandAire Technologies, Inc., Jackson, MI: Navy Phase I 2010 $79,981 (`www.sealandaire.com`)
- Sherpa Solutions, LLC, Roswell, GA: Navy Phase I 2012 $80,000 (`www.sherpasolutions.net`)

- Softronics LImited, Cedar Rapids, IA: Navy Phase II 2014 $711,302 (`www.softronicsltd.com`)
- TECHNOLOGY FOCUS LLC, Covina, CA: Army Phase I 2005 $69,957 (`www.tecfocus.com`)

The competitive advantage of our Firm, Aquerre Technologies, lies in our core expertise in communication, control, and signal processing engineering systems. In this SBIR proposal, we exploit cutting edge communication and control technologies to support implementation of advanced algorithms and signal processing methods for the next generation Ad-Hoc GCFS.

7 Key Personnel

7.1 Noah B. Jacobsen, Principal Investigator

Education

- Ph.D., 2005, Electrical and Computer Engineering, University of California, Santa Barbara.
  - Specialization: Communication, Control and Signal Processing.
- M.S., 2002, Electrical and Computer Engineering, University of California, Santa Barbara.
- B.S., 2000, Electrical Engineering, Cornell University.

Professional Experience

- Aquerre Technologies LLC: Founding Member, Principal Scientist (May 2013 – present).
  - Research and development contracting and consulting services.
- Dex One: Sr. Operations Research Scientist (Sept. 2012 – March 2013).
  - Advertising technology predictive analytics.
- Columbia University: Adjunct Professor (Spring Semester 2012).
  - Linear Systems Theory, Dept. of Electrical Engineering.
- Alcatel-Lucent, Bell Labs: R&D Engineer (July 2006 – Oct. 2011).
  - Error control codes, cooperative relay codes, physical layer communications research, standardization, and development. Includes experience transitioning algorithms to product teams.
- Polytechnic Institute of New York University: Adjunct Professor (Fall Semester 2010).
  - Probability Theory, Dept. of Electrical and Computer Engineering.
- University of California, Santa Barbara: Post-Doctoral Researcher (Sept. 2005 – June 2006).
  - Cognitive radio networks, communication theory.

Conferences

- Attendee, 2014 National SBIR/STTR Conference and Short Course "How To Develop An Acceptable Accounting System", Washington, DC, Jun. 16–18, 2014.

Certifications

- California Basic Educational Skills Test (CBEST) Completed, December 2013.

Patents

- N. Jacobsen and R. Soni, "Method and system for encoding data using rate-compatible irregular LDPC codes based on edge growth and parity splitting", U.S. Patent No. 7,966,548, June 2007.

Achievements

- Session Chair, "Wireless Networks and Communications," 43rd Conference on Information Sciences and Systems (CISS), March 18-20, 2009.
- 3GPP2 Ultra Mobile Broadband (UMB) Air Interface Specification: Recognition of Contribution, LDPC Ad Hoc Group, 2007.
- National Science Foundation (NSF) and Japan Society for the Promotion of Science (JSPS) East Asia Summer Institutes Fellowship, Yokohama National University, Japan, 2003.

- Microelectronics Innovation and Computer Research Opportunities Scholarship, University of California, Santa Barbara, 2000–2001.
- Theodore C. Ohart Scholarship in Engineering, Cornell University, 1999–2000.
- Cornell University College of Engineering Cooperative Education Program, with Floyd R. Newman Laboratory of Nuclear Studies, Cornell University, 1998–1999.

7.2 R&D Engineer

Upon award of contract, we will hire an R&D engineer (part-time) to work specifically on this SBIR program. We will seek a diverse applicant with both relevant software algorithms experience as well as hardware interface experience. The P.I. has numerous academic contacts that could potentially be leveraged for recruiting talent. In particular, we will hire a U.S. citizen for the position associated with this program.

8 Foreign Citizens

This topic is ITAR restricted. All personnel performing work on this project are U.S. citizens.

9 Facilities/Equipment

The primary business office of Aquerre Technologies LLC is currently located at 1445 Colby Ave #3, Los Angeles, CA 90025. All facilities, equipment, and data management supporting the Phase I segment of this SBIR program are to be based out of our primary office. If this proposal is funded, we may seek to re-locate the primary address. Any changes to our location to would be subject to sponsor notification/approval. Further, we will work closely with the sponsor to ensure that our facilities situation is adequate for program requirements throughout all Phases of the program.

10 Subcontractors/Consultants

Neither sub-contractors nor consultants are included in the Phase I/I.A proposal.

11 Prior, Current or Pending Support of Similar Proposals or Awards

Aquerre Technologies LLC has no prior, current, or pending support for the work proposed for this SBIR program.

12 Discretionary Technical Assistance

The use of Discretionary Technical Assistance (DTA) is not included in this proposal.

Appendices

A Mathematical Model of the Acoustic Propagation Channel

The acoustic propagation channel model used in this paper is based on a linear systems model, similar to the approach taken in [22]. In particular, we start from the assumption that the propagation medium from source to receiver is Linear, Time- (and Space-Shift) Invariant (LTI) and therefore the channel output at a given location and time is given by a convolution of the input with the space-time channel impulse response [23].

In the following, we use the wave function corresponding to lossless, free-space acoustic transmission [24] to derive the channel impulse response $h(r,t)$ as a function of receiver distance r and time t. The channel output signal corresponding to an arbitrary input $x(r,t)$ is then given by [23]:

$$y(r,t) = \int_{-\infty}^{\infty} \int_{0}^{\infty} x(r-\rho, t-\tau) h(\rho, \tau)\, d\rho\, d\tau \tag{4}$$

The channel input $x(r,t)$ and output $y(r,t)$ of the channel are assumed to be acoustic pressure-deviation signals. Acoustic pressure deviations are produced by elastic displacement of particles in the medium in response to an input displacement. The signal measured by a human ear or microphone is proportional to the pressure deviation signal relative to ambient pressure at the sensor.

From Serway [24], the particle displacement signal $s_0(t)$ of a plane wave (acoustic source transmitting energy in a single direction) in a lossless medium induces the following steady-state displacement signal (wave function):

$$s(r,t) = s_0 \left(t - \frac{r}{c} \right), \tag{5}$$

where r is the distance along the direction of propagation, c is the speed of sound in the medium and t is time. In this Appendix, after analyzing the impulse response corresponding to plane wave propagation, we then treat the case of isotropic acoustic sources.

The steady-state pressure deviation signal is related to the steady-state particle displacement signal Equation (5) by the following partial differential equation [24]:

$$p(r,t) = -\frac{\rho c^2}{2\pi} \frac{\partial}{\partial r} s(r,t), \tag{6}$$

where ρ is the density of the medium.

To derive the channel impulse response, we first consider an ideal sinusoidal input displacement signal of the form:

$$s_0(t) = s_m \cos\left(\omega t + \phi_0\right), \tag{7}$$

where s_m denotes the maximum particle displacement, ω is the angular frequency and ϕ_0 is the phase of the input displacement signal.

Substituting the sinusoidal input above in the formula for the pressure deviation wave function Equation (6) yields:

$$p(r,t) = -p_m \sin\left(\omega\left(t - \frac{r}{c}\right) + \phi_0\right), \tag{8}$$

where

$$p_m = \frac{\rho c \omega}{2\pi} s_m$$

is the maximum pressure deviation of the source.

The source input pressure-deviation signal, $p_0(t) = p(0,t)$, corresponds to the wave-function Equation (6) evaluated at $r = 0$. For a sinusoidal input displacement Equation (7), we have:

$$p_0(t) = p(0,t) = -p_m \sin\left(\omega t + \phi_0\right). \tag{9}$$

The average energy of the input pressure signal $p_0(t)$ corresponding to sinusoidal displacement is:

$$\begin{aligned}
E_0 &= \frac{1}{T}\int_0^T \left(p_0(t)\right)^2\, d\tau \\
&= \frac{p_m^2}{T}\int_0^T \left(\sin(\omega\tau+\phi_0)\right)^2\, d\tau \\
&= \frac{p_m^2}{2T}\int_0^T \left(1-\cos(2\omega\tau+2\phi_0)\right)\, d\tau \\
&= \frac{p_m^2}{2T}\left(T-\frac{1}{2\omega}\sin(2\omega T+2\phi_0)+\frac{1}{2\omega}\sin(2\phi_0)\right) \\
&= \frac{p_m^2}{2}\left(1-\frac{1}{4\pi}\sin(4\pi+2\phi_0))+\frac{1}{4\pi}\sin(2\phi_0)\right) \\
&= \frac{p_m^2}{2}, \qquad (10)
\end{aligned}$$

where $T = 2\pi\omega^{-1}$ is the period. Hence Equation (9) can be re-written as:

$$p_0(t) = \Im\left\{z_0(t)\right\}, \qquad (11)$$

where

$$z_0(t) = -\sqrt{2E_0}\exp\left(j\omega t + j\phi_0\right), \qquad (12)$$

and $\Im\{x\}$ denotes the imaginary part of x.

Let

$$x(r,t) = \delta(r)p_0(t) \qquad (13)$$

denote the channel input pressure-deviation signal as a function of radius and time. The notation $\delta(x)$ is used to represent the Dirac delta function, which is an abstraction for an ideal impulse signal (unbounded at a single point and zero elsewhere). By the assumption of linearity and space-shift invariance, the response to an arbitrary input can described by the composition of a continuum of responses to point-source inputs and therefore the channel impulse response can be derived using point-source assumption.

With the point-source input Equation (13), the channel convolution integral Equation (4) becomes:

$$\begin{aligned}
y(r,t) &= \int_{-\infty}^{\infty} p_0(t-\tau)\int_0^{\infty}\delta(r-\rho)h(\rho,\tau)\,d\rho\,d\tau \\
&= \int_{-\infty}^{\infty} p_0(t-\tau)h(r,\tau)\,d\tau
\end{aligned}$$

Substituting Equation (11) into the above equation yields:

$$\begin{aligned}
y(r,t) &= \int_{-\infty}^{\infty}\Im\left\{z_0(t-\tau)\right\}h(r,\tau)\,d\tau \\
&= \Im\left\{\int_{-\infty}^{\infty} z_0(t-\tau)h(r,\tau)\,d\tau\right\} \qquad (14) \\
&= \Im\left\{z_0(t)\int_{-\infty}^{\infty}\exp(-j\omega\tau)h(r,\tau)\,d\tau\right\} \\
&= \Im\left\{z_0(t)\int_{-\infty}^{\infty}\exp(-j\omega\tau)\left[(2\pi)^{-1}\int_{-\infty}^{\infty}H(r,\Omega)\exp(j\Omega t)\,d\Omega\right]\,d\tau\right\} \\
&= \Im\left\{z_0(t)\int_{-\infty}^{\infty}H(r,\Omega)\left[(2\pi)^{-1}\int_{-\infty}^{\infty}\exp(-j\omega\tau))\exp(j\Omega\tau)\,d\tau\right]\,d\Omega\right\} \qquad (15) \\
&= \Im\left\{z_0(t)\int_{-\infty}^{\infty}H(r,\Omega)\delta(\Omega-\omega)\,d\Omega\right\} \\
&= \Im\left\{z_0(t)H(r,\omega)\right\}, \qquad (16)
\end{aligned}$$

where $H(r, \Omega)$ denotes the Fourier Transform of the channel impulse response with respect to the time variable, and $z_0(t)$ is defined in Equation (12). In the above, Equation (14) is valid because the channel impulse response is a real-valued signal, and equations Equations (15)-(16) hold in the generalized calculus of impulsive signals.

Re-writing Equation (8) as:

$$p(r,t) = \Im\left\{z_0(t)\exp\left(-j\frac{\omega r}{c}\right)\right\},$$

and interpreting it as the channel output due to the input signal Equation (13), we find that:

$$H(r,\Omega) = \exp\left(-j\frac{\Omega r}{c}\right).$$

upon comparison to Equation (16).

Since

$$h(r,t) = \frac{1}{2\pi}\int_{-\infty}^{\infty} H(r,\Omega)\exp\left(j\Omega t\right)\, d\Omega,$$

the impulse response $h_p(r,t)$ corresponding to plane wave propagation in an ideal elastic medium is given by:

$$h_p(r,t) = \delta\left(t - \frac{r}{c}\right).$$

We next develop the channel impulse response for isotropic radiated acoustic sources. Path loss models based on isotropic radiation are standard for acoustic and electromagnetic antenna-based source signals. In particular, we assume that an ideal isotropic acoustic source radiates energy equally in all directions at the speed of sound in the medium. For an isotropic source in lossless media, the source energy E_0 at time zero is uniformly distributed across the surface of a sphere with radius $r = ct$ at time t. Thus, the energy per unit area at radius r is given by $E_r = E_0/(4\pi r^2)$. Letting $p_m(r)$ denote the maximum pressure deviation at radius r, we have $E_r = p_m(r)^2/2$ from Equation (10). Hence,

$$\begin{aligned} p_m(r) &= \sqrt{2E_r} \\ &= \frac{1}{r}\sqrt{\frac{E_0}{2\pi}}, \end{aligned}$$

showing that the amplitude of the pressure-deviation signal resulting from an ideal isotropic source decays inversely in r.

The pressure deviation wave function resulting from an isotropic source in a lossless medium is analogous to Equation (8), but with the appropriate amplitude scaling factor:

$$\begin{aligned} p(r,t) &= -p_m(r)\sin\left(\omega\left(t - \frac{r}{c}\right) + \phi_0\right) \\ &= \Im\left\{z_0(t)\frac{1}{r}\sqrt{\frac{1}{4\pi}}\exp\left(-j\frac{\omega r}{c}\right)\right\}. \end{aligned} \tag{17}$$

Upon comparing equations Equation (16) and Equation (17), we find:

$$H(r,\Omega) = \frac{1}{r}\sqrt{\frac{1}{4\pi}}\exp\left(-j\frac{\Omega r}{c}\right).$$

Thus the channel impulse response due to an ideal isotropic point-source in a lossless acoustic medium is given by:

$$h(r,t) = \frac{1}{r}\sqrt{\frac{1}{4\pi}}\delta\left(t - \frac{r}{c}\right),$$

where r is distance to the source, t is time, and c is the speed of sound.

References

[1] G.L. Duckworth, D.C. Gilbert, and J.E. Barger. Acoustic counter-sniper system. In *Proc. SPIE: Command, Control, Communications, and Intelligence Systems for Law Enforcement*, volume 2938, Nov. 1996.

[2] S. Tenney, B. Mays, D. Hillis, D. Tran-Luu, J. Houser, and C. Reiff. Acoustic mortar localization system results from OIF. Technical report, US Army Research Laboratory, Adelphi, MD, Dec. 2004.

[3] H.E. Bass. Advances in technology to support battlefield acoustic sensing. Technical report, National Centre for Physical Acoustics, University of Mississippi, Nov. 2007.

[4] N. Jacobsen. *Bayesian approaches to noncoherent communication: From Shannon theory to practical architectures*. PhD thesis, 2005.

[5] ST Microelectonics. *MP45DT02 MEMS audio sensor omnidirectional digital microphone*. Technical Data Sheet, `http://www.st.com/st-web-ui/static/active/en/resource/technical/document/datasheet/DM00025467.pdf`.

[6] Trimble Navigation. *BD970 GNSS Reciever Module*. Technical Data Sheet, `http://www.dpie.com/datasheets/gps/Trimble-BD970-Datasheet-Document-611506.pdf`.

[7] D. Bohn. Environmental effects on the speed of sound. *J. Audio Eng. Soc.*, 36(4), Apr. 1988.

[8] ST Microelectonics. *A3G4250D MEMS motion sensor: 3-axis digital output gyroscope*. Technical Data Sheet, `http://www.st.com/st-web-ui/static/active/en/resource/technical/document/datasheet/DM00047823.pdf`.

[9] ST Microelectonics. *Capacitive digital sensor for relative humidity and temperature*. Technical Data Sheet, `http://www.st.com/web/en/resource/technical/document/datasheet/DM00116291.pdf`.

[10] Wolfson Microelectronics. *WM5102 Audio Hub CODEC with Voice Processor DSP*. Technical Data Sheet, `http://www.wolfsonmicro.com/documents/uploads/data_sheets/en/WM5102.pdf`.

[11] B.M. Sadler, T. Pham, and L.C. Sadler. Optimal and wavelet-based shock wave detection and estimation. *J. Acoustical Soc. of America*, 1998.

[12] W.B. Davenport. *Probability and Random Processes*. McGraw-Hill, 1970.

[13] H.V. Poor. *An Introduction to Signal Detection and Estimation*. Springer, 2nd edition, 1994.

[14] R.C. Maher. Modeling and signal processing of acoustic gunshot recordings. In *Proc. IEEE Signal Processing Society 12th DSP Workshop*, pages 257–261, Sept. 2006.

[15] G. Barriac, N. Jacobsen, and U. Madhow. Beyond BAD: A parallel arbitration framework for low-complexity equalization. In *Proc. 39th Annual Allerton Conf. on Communication, Control, and Computing*, Oct. 2001.

[16] N. Jacobsen and U. Madhow. Coded noncoherent communication with amplitude/phase modulation: From Shannon theory to practical turbo architectures. *IEEE Trans. Commun.*, 56(12), Dec. 2008.

[17] N. Jacobsen, G. Barriac, and U. Madhow. Noncoherent eigenbeamforming for a wideband cellular uplink. In *Proc. IEEE Intern. Symp. on Inform. Theory (ISIT)*, Chicago, IL, USA, June 2004.

[18] N. Jacobsen, G. Barriac, and U. Madhow. Noncoherent eigenbeamforming and interference suppression for outdoor OFDM systems. *IEEE Trans. Commun.*, 56(6), June 2008.

[19] J.C. Chen, L. Yip, J. Elson, H. Wang, D. Maniezzo, R.E. Hudson, K. Yao, and D. Estrin. Coherent acoustic array processing and localization on wireless sensor networks. *Proc. IEEE*, 91(8):1154–1162, Aug. 2003.

[20] T.L. Marzetta. A new interpretation for Capon's maximum likelihood method of frequency-wavenumber spectral estimation. *IEEE Trans. Acoust., Speech, Signal Processing*, 31(2):445–449, Apr. 1983.

[21] Digital security and surveillance market overview. Market brief, Intel Corporation, Nov. 2013. Retrieved from: `http://www.intel.com/content/dam/www/public/us/en/documents/solution-briefs/digital-security-surveillance-market-segment-overview-brief.pdf`.

[22] J.P. Powers. Acoustic propagation modeling using MATLAB. Interim Report, Naval Postgraduate School, Sept. 1993.

[23] T. Kailath. *Linear Systems*. Prentice-Hall, 1980.

[24] R.A. Serway. *Physics for Scientists and Engineers*. Saunders, 4th edition, 1982.