A Monitoring System for the Detection, Identification, and Localization of Acoustical Transient Events in National Parks

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Introduction and Problems Statement:

The natural sounds program of the national park service (NPS) was established to help parks manage sounds in a way that balances park access with the expectations of park visitors and the protection of park resources. This program addresses acoustical issues raised by congress, NPS management policies, and NPS director's orders. The program also provides technical assistance to parks in the form of acoustic monitoring, data processing, park planning support, and comparative analyses of acoustic environments throughout the national park system. This research project seeks to improve acoustic monitoring efforts that provide a scientific basis for assessing the current status of acoustic resources. In particular, these efforts help to identify trends in resource conditions, quantify impacts from other actions, assess consistency with park management objectives and standards, and provide a basis for management decisions regarding desired future conditions.

Presently, acoustic monitoring stations have been deployed in various backcountry sites in several national parks (some for over a decade) recording data that can be used for eventual characterization of the soundscapes of these areas. These monitoring stations capture their acoustic environments using a single microphone and record data in two different formats. Due to the extremely large quantity of data, automated methods are needed for detection, identification, and localization of sources of intrinsic and extrinsic sound, since manual inspection of this entire database would be a tedious and impractical task. However these are challenging tasks owing to the presence of competing sources of interference, highly variable operating and environmental conditions, and limitations on the storage capacity of deployed monitoring stations. Some specific tasks in this research focuses on in order realize a system capable of automatic characterization of national park soundscapes include:

a) Development of algorithms and software that automatically estimate natural ambient sound levels, removing the effects of transportation and other noise sources. This includes exploration of new methods of characterizing natural ambient sound levels, possibly separating biological, transient physical, and background baseline levels.

b) Investigation of methods that can automatically identify various types of transportation sound sources. This may include options for estimating the speed of the vehicle, the slant range of its closest point of approach, and possibly its altitude (for aircraft).

c) Investigation of options for separating sound source signatures, to parse recordings of composites of sounds into representations of what each source would sound like in isolation.

d) Develop a public display system that can indicate the noise level generated by passing vehicles, with a flashing indication when the noise level exceeds current NPS regulations (like a radar trailer).

e) Deploy new environmental sound monitoring systems at NPS park units to assess their performance, reliability, and ease of use, while collecting data that can support Air Tour Management Plans, resource management plans, and park technical assistance requests.

Approach:

As can be seen, there is large variety of tasks to accomplish for this research, and therefore the approaches taken will be just as varied. Regarding the development of algorithms for the detection and identification of sources of acoustical transient events, several approaches have already been developed that are unique to each type of data format. The first format is single channel audio (.WAV or .MP3 files) that consistently captures the actual waveforms emitted by the sources. The second format represents the captured acoustic information using a series of 33-dimensional vectors, where each element of a vector is the sound pressure level obtained by integrating the energy over a different 1/3 octave frequency subband (33 total) in a one second time window. For raw audio data, transient events are detected by using a new method [1] that searches for a basis that permits a sparse representation of the signal that contains the transient signatures. An efficient algorithm is used to compare the cost, as measured by the l_1-norm, of representing the signal in each basis from a library of orthogonal bases. This method relies on the fact that the basis that gives the sparest representation of the signal will contain subsets of vectors that span the subspaces that the signatures generated by the sources lie in. Once sources are detected, extracting features for classification is a relatively simple task since the best basis can be used to temporally segment the signal so that a single feature vector can be extracted to represent each source.

Transient events are detected using 1/3 octave data vectors by employing a Page's test [2] for dependent observations implemented with Hidden Markov Models (HMM) [3]. A Page test uses the likelihood functions of the joint observations under signal present and signal absent hypotheses to update a detection statistic and has min-max optimality for average run length among all sequential detection schemes. When the 1/3 octave data vectors are quantized, a discrete HMM can be used to find the likelihood of observing a given joint observation sequence under each hypothesis. Furthermore, HMM’s allow this likelihood to be updated through the use of a forward variable, leading to efficient implementation of the Page test and the ability to perform real-time detection of transient events.

Prototypes for the new multi-channel acoustic monitoring station and public display systems mentioned above are also currently being designed and built. The new monitoring system provides the ability to continuously monitor the audible environment, similar to weather stations. Each unit (or node) consists of five microphones, high fidelity analog channels and a FPGA processing board to allow on-board generation of one-third octave sound pressure levels from all channels for man-made and natural sources. This system supports forming a sparse wireless sensor network of several nodes via the high performance radio links. The system will also have the capability of sending daily feature summaries or immediate alerts wirelessly to the park headquarters. The multi-channel processing ability of the system allows for separation, classification and localization of sound sources, capabilities that were lacking in the existing systems. Additionally, the new system will be of substantially lower cost and more compact, hence allowing for large scale deployment in national parks. The system can be easily integrated into the National Ecological Observatory Network (NEON).

Current Results:

Figure 1 shows an example of applying the best basis detection method to an audio segment that consists of the signatures of an airplane embedded in natural background noise and was recorded by one of the deployed acoustic monitoring stations. As can be seen, despite the fact that the amplitude of the source increases relatively slowly, and is not significantly higher than the background noise, the best basis method has successfully identified the interval containing just the signatures of the source. That is to say, by listening to the audio segment between the red lines, the entire portion of the audio segment containing the signatures of the plane would be heard, without hearing significant additional quiescent periods on both ends of the segment. This method has been applied to a database of several audio segments containing transient events and was able to successfully detect every event. Performance of this method on audio segments containing a high level of background noise and weak transient signatures still needs to be explored.

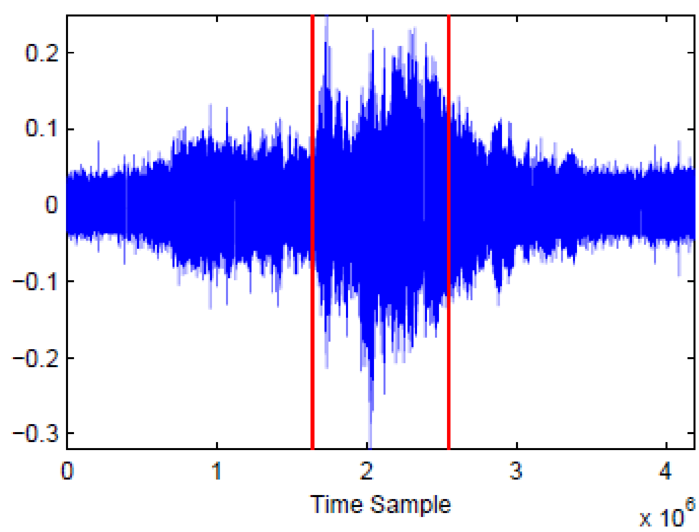

Figure 1: Time series of an audio segment containing signatures of an airplane along with intervals associated with its best basis.

Figure 2 shows the results of applying the above-mentioned Page's test transient detector implemented using HMM's to a sequence of 1/3 octave data vectors. The top portion of the window shows a plot of the 1/3 octave data vectors with time on the horizontal axis and frequency (vector element) on the vertical access. Below the data plot is a detection strip that is black for time segments the Page detector has declared a transient event and white for non-transient periods. As can be seen from visual inspection, the transient events (which can easily be identified in the data plot) are all detected successfully. Additionally, this system has proven to be relatively robust to interference such a weather effects and bird song, whose signatures can be seen in the higher frequency range. This is due to the fact that these sources of interference were accounted for in the HMM used to model the segments where the transient signal is absent.

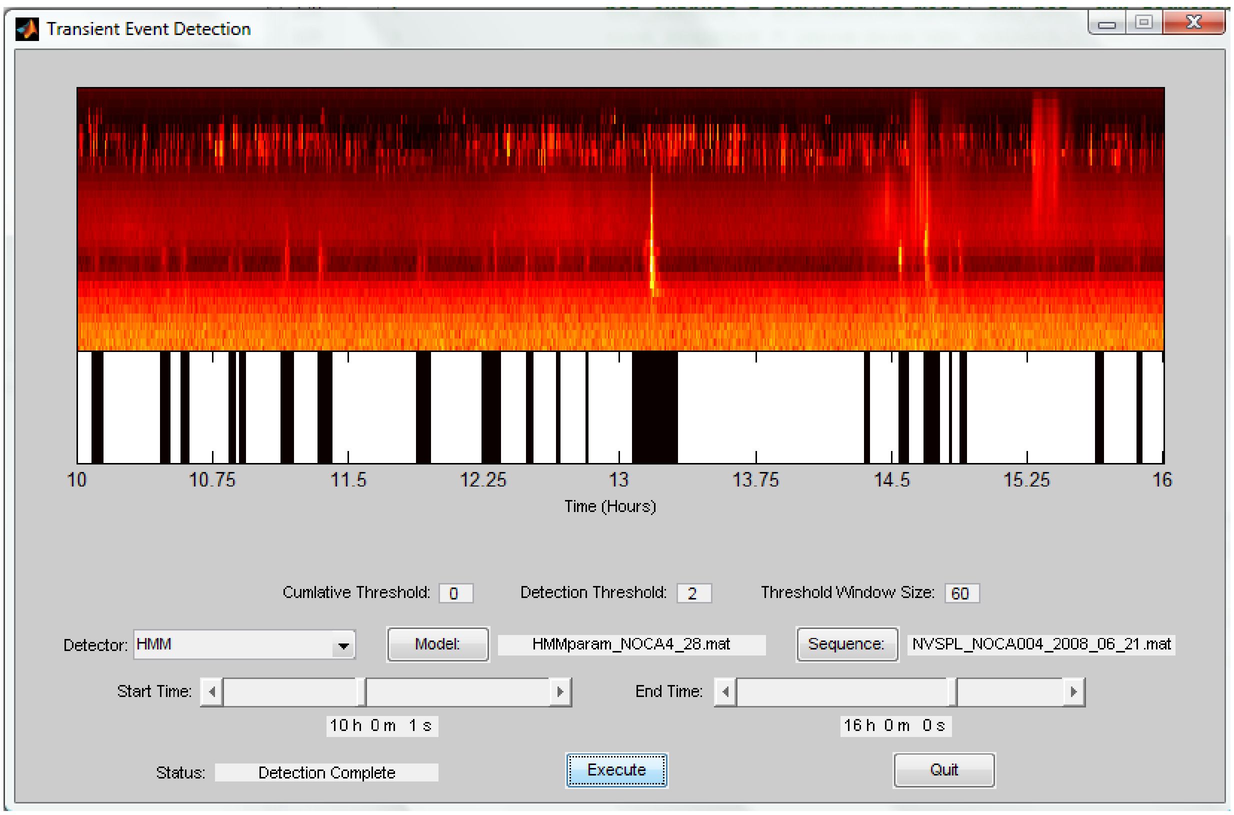

Figure 2: GUI displaying results of applying the Page's test transient detector implemented using HMMs to a sequence of 1/3 octave data vectors.

References:

[1] R. Coifman and M. Wickerhauser, "Entropy-Based Algorithms for Best Basis Selection," *IEEE Trans. on Information Theory*, vol. 38, no. 2, pp. 713–718, 1992.

[2] B. Chen and P. Willett, "Detection of Hidden Markov Model Transient Signals," *IEEE Trans. on Aerospace and Electronic Systems*, vol. 36, no. 4, pp. 1253–1268, 2000.

[3] Rabiner, L., "A Tutorial on Hidden Markov Models and Selected Applications in Speech Recognition," *Proc. of the IEEE*, vol. 77, no. 2, Feb. 1989.