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Advanced Array Techniques for Unattended Ground Sensor Applications

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ABSTRACT

Sensor arrays offer opportunities to beamform, and time-frequency analyses offer additional insights to the wavefield data. Data collected while monitoring three different sources with unattended ground sensors in a 16-element, small-aperture (~ 5 meters) geophone array are used as examples of model-based seismic signal processing on actual geophone array data. The three sources monitored were:

(Source 01). A frequency-modulated chirp of an electromechanical shaker mounted on the floor of an underground bunker. Three 60-second time-windows corresponding to (a) 50 Hz to 55 Hz sweep, (b) 60 Hz to 70 Hz sweep, and (c) 80 Hz to 90 Hz sweep.

(Source 02). A single transient impact of a hammer striking the floor of the bunker. Twenty seconds of data (with the transient event approximately mid-point in the time window).

(Source 11). The transient event of a diesel generator turning on, including a few seconds before the “turn-on time” and a few seconds after the generator reaches “steady-state conditions”.

The high-frequency seismic array was positioned at the surface of the ground at a distance of 150 meters (North) of the underground bunker. Four Y-shaped subarrays (each with 2-meter apertures) in a Y-shaped pattern (with a 6-meter aperture) using a total of 16 3-component, high-frequency geophones were deployed. These 48 channels of seismic data were recorded at 6000 and 12000 samples per second on 16-bit data loggers. Representative examples of the data and analyses illustrate the results of this experiment.

Keywords: Multichannel, time-frequency, signal processing, array, estimation

1. INTRODUCTION

The data collected and analyzed for this high-frequency seismic array experiment is part of a larger data set collected for several different experiments conducted in 1995 for R & D investigations related to monitoring various underground bunkers. The objectives for this high-frequency array experiment will be presented, followed by some background information related to multichannel processing for detection, location and characterization. Next, some results of the seismic characterization of three different sources will be discussed; followed by a discussion of conclusions that have been arrived at from these studies.

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1.1. Objectives of the B2-300 Test 5: High-frequency Seismic Array Experiment

This high-frequency array was positioned at the surface of the ground at a distance of 150 meters (North) of an underground bunker. The bunker is designated as B2-300. Four Y-shaped subarrays (each with 2-meter apertures) in a Y-shaped pattern (with a 6-meter aperture) using a total of 16 3-component, high-frequency geophones were deployed. Fig. 1 shows the surveyed positions for the element locations for this array. These 48 channels of seismic data were recorded at 6000 and 12000 samples per second on 16-bit data loggers.

The four objectives for this array experiment were: (1) Measure the bandwidth¹ of the coherent seismic signals at a range of 150 meters for different sources. Range was selected based upon the minimum range for “far-field” observation condition and the maximum range to observe signals propagating through the heterogeneous “overburden” (alluvium and colluvium interbedded with tuff) and uncontaminated by seismic arrivals traveling through the higher velocity “bedrock” (represented by Paleozoic shales and carbonates at this particular test location in Nye County, NV). (2) Apply spatial a sampling² of “minimum-redundant” subarrays to provide robust broadband capabilities while maintaining maximum protection against spatial aliasing. This high-frequency array was designed to accommodate compressional seismic (or P-type) waves with frequency content up to 1000 Hz and Rayleigh surface (or groundroll-type) waves with frequency content up to 300 Hz (3) Use multichannel, coherent array beamforming techniques combined with model based signal processing to characterize the nonstationary source signals using time-frequency analyses; and (4) Evaluate the “in-phase” and “in-quadrature” combinations of the three orthogonal axes (vertical, radial, and transverse directions); and compare these results with the subset of “vertical components” only.

1.2. Background for multichannel wavefield analysis

Multichannel spatial and temporal filtering of seismic wavefields using sensor arrays are expected to offer special utility and may help characterize important aspects of observed signals. Modeling the wavefield as monochromatic plane waves propagating in a homogeneous half space is sometimes too simple. Propagating waves are neither plane nor monochromatic; and in the real-world the geology is not homogeneous. We must design the seismic arrays and experiments to maximize the signal-to-noise ratio (SNR) over a finite band of frequencies or wavenumbers and do this at several spatial locations³ that sample the variability of the wavefield and of the geologic conditions. Arrays have been used to enhance SNR, estimate direction of propagation, measure velocity of propagation, or characterize scattering of seismic signals. These benefits are obtained through two basic processes: suppression of uncorrelated random noise through averaging and modeling the coherent signals using matched filtering.

One fundamental concept (and constraint) is that the array is a sampled aperture that produces a more complicated diffraction pattern than that which is produced by a continuous aperture. Diffraction patterns for continuous apertures are often described in optics where there are one-dimensional “slits” or two-dimensional “holes”. These simple apertures are categorized as

¹Steinberg [Ref. 8, pp. 134-136] points out that one key factor controlling the design a sampled aperture array is the size of the “array parameter”, which he defines as the band large signal-to-noise+interference ratio (SNIR).

²Orfanidis [Ref. 7, p. 372] states that the MUSIC and ESPRIT (eigenvalue based array processing) can be applied to multiple, replicate, subarrays of arbitrary geometry; but that the mini order array processing techniques require equally spaced linear arrays. The “Y” is a non-redundant two-dimensional array pattern or design.

³Followill [Ref. 3, p. 108] shows that the time-bandwidth-SNR product is crucial to the against design criteria.

continuous because there is no discrete sampling (rather than discretely sampled apertures that we synthesize using multiple simple apertures) and the diffraction pattern of each simple continuous aperture, in turn, is categorized as having a main (or central) lobe and associated side lobes. From optics⁴ we find that the maximum peak side lobes resulting from the one-dimensional linear slits is 13 dB smaller than the main lobe; and that the maximum peak side lobes resulting from the two-dimensional circular hole is 18 dB smaller than the main lobe.

It is also well known in the literature that sampled apertures have an additional complication that is usually referred to as a “grating” lobe.⁵ There can be no grating lobes due to the diffraction of a single continuous aperture; neither for the continuous slit nor for the continuous hole. The grating lobe is a result of a regular pattern in the sampling of the aperture. The grating lobe is of the same strength as the main lobe. This grating lobe is due to “folding” or “aliasing” other wavenumbers into the base band.

Each sampling point of an aperture requires a data channel. The expense of collecting and analyzing the multichannel array data leads us to seek a sampling configuration⁶ that minimizes the number of samples to achieve a some maximum level of performance. This quickly leads us into trade-offs between desired performance of an array and the number and positioning of elements in an array.

1.3. Background for space-time processing and modeling

Other trade-offs must also be considered. These additional trade-offs involve partitioning the processing problem and selecting the orders⁷ for the models used in the signal processing.⁸ Partitioning may need to be done in the time domain, segmenting the data into piecewise stationary subsets and possibly prewhitening the subset. The data set provides several transient events in addition to samples of the ambient background noise. The models for these transient events differ for different sources; and the model for the background noise usually has a suite of parameters different from the transient events. Furthermore, the background noise may exhibit nonstationary⁹ properties. Pilot investigations¹⁰ of the character of the time series and the spectra were used to help plan the analyses presented below.

⁴Steinberg [Ref. 8, p. 62] shows that a continuous aperture has no grating lobes and that they are reduced (relative to the main lobe) by 13 dB for the linear slot aperture and by 18 dB for the circular aperture.

⁵There may be exceptions as Johnson and Dudgeon [Ref. 4, p. 93] indicate that grating lobe aliasing does not occur for randomized linear arrays.

⁶Steinberg [Ref. 8, pp. 71 ff.] uses the concept that the sampled aperture array is a digital equivalent of a continuous aperture and then explores the aperture synthesis, its limitations, and its various designs; particularly the impact that sampling has upon the patterns and levels of the main lobe and side lobes for spaced linear arrays and for unequally spaced (random, logarithmic, or thinned arrays).

⁷Clarkson [Ref. 2, p. 438] states that experience, trial-and-error, and subjective judgment are used to make a final decision about the model order to use.

⁸Candy [Ref. 1, pp. 200-203] shows that the Minimum-Variance Distortionless Response (MVDR) used by Capon as an array spectral estimator, is data adaptive and provides a smoothed, unbiased estimate of the spectral power, by averaging over all the higher-order maximum entropy model (MEM) estimates. Instead, we have chosen to use a biased autoregressive model because we are interested in characterizing mechanical resonances that should appear as sharp lines or poles.

⁹Johnson and Dudgeon [Ref. 4, p. 187] suggest spatial averaging for non-stationary signals.

¹⁰McDonough and Whalen [Ref. 6, pp. 339-381] discuss some approaches that may be suitable for signal detection in the presence of colored Gaussian noise; and they conclude that the singular value decomposition (SVD) technique may perform best for detection under these assumptions. They [Ref. 6, pp. 339-381] discuss estimation in non-white Gaussian noise and they then suggest that we should consider a two-stage process in which we first pre-whiten the noise and then apply a maximum likelihood technique for estimation. Orfanidis [Ref. 7, p. 54] points out that the time series data

Processing this space-time array data may be described as a moving-average (MA) auto-regressive (AR) modeling or filtering of the waveform data. We are searching for spectral lines produced by resonances of machinery that are best and efficiently modeled by the all-pole or auto-regressive techniques. On the other hand, the delay-and-sum operation of beamforming is a moving average filter. The MA process is equivalent to a Finite Impulse Response (FIR) spatial filtering of the wavefield data and the AR process is equivalent to an Infinite Impulse Response (IIR) filtering of the time-domain output or result of the FIR filter. Matched field filtering is actually an ARMA process which is usually modeled jointly; rather than sequentially as we have processed the data in this analysis. In either approach, models are adapted or adjusted to the data and there are tradeoffs in accuracy and resolution in characterizing the data that is being modeled.

One must select the order of the model (i.e., how many free parameters¹¹ will I allow in the model to represent the data?) and this has to be judged by the “goodness of fit” of the model to the data. There are several measures or criteria that are routinely used in modern signal processing. To name five of the more frequently used measures: the final prediction error (FPE), the Akaike information theoretic criterion (AIC), the minimum description length (MDL), the criterion-autoregressive transfer function (CAT), and the Zhao-Atlas-Marks/Cohen (ZAM) local maximum index. Each of these attempts to provide a measure of the “goodness of fit” versus the “number of parameters”. Intuition suggests that we should use the minimum number of parameters that will be adequate to describe or model the data.

If we allow one parameter for each data point we can fit the data “exactly”; but with a very complicated and complex model that provides little insight into the “process” being investigated and little confidence in the “values” being measured or estimated. The initial guess of the model order may be sequentially increased and the goodness of fit can be measured for each model order selected. Where to start and where to stop are, at best, guesses; and, at worst, entrapments into local minima that may be irrelevant. Experience modeling seismic spectra under widely varying conditions provides some initial guidance. For example, an estimate of number of AR parameters may be obtained by observing the number of spectral lines in a preliminary or pilot spectral estimate. This preliminary spectral estimate may be derived from modeling the spectra with a generous number of parameters. One such estimate of this generous number of poles may be taken as the ratio of the number of data divided by its natural logarithm.

1.4. Background for using differential time-of-arrival measurements for direction and slowness

If we take the simple case of only two sensors or data channels (or samples of an aperture) we have the interferometer beam which has extensive literature discussing the application of a pair of elements on an interferometer beam to measure relative phase differences of a wave observed at the two ends of the interferometer. The output signals from sensors at the two ends of an interferometer may be compared by correlating their outputs. The variance of the measured relative phase difference between the two ends of an interferometer beam is inversely

shorter series that may be considered to be piecewise stationary; and [p. 54 ff.] points nth-order Markov and therefore the data may be transformed or high-pass filtered to produce exhibits properties that are closer to the built-in assumptions that the data are independent (iid).

¹¹Kailath [Ref. 5, pp. 115-119] suggests that we can use the prediction error as a measure of applying a modern spectral analysis to data that may be near-stationary.

proportional to the SNR measured at the two sensor outputs¹². Perhaps a more precise statement should define this SNR as the ratio of the quasi-coherent (desired) signal power to the quasi-noncoherent noise (or undesired signal) power. This variance of the measured relative phase differences is also known as the (phase) coherency between the two channels. By extending this concept to several channels, taken as a pair of channels at a time, the performance of an array may be evaluated as a multiple-beam interferometer. One way to analyze these “taken one pair at a time” relationships is to construct the covariance matrix of the multichannel array data.

Localization techniques include triangulation using multiple differential times of arrival, beamforming, and frequency-wavenumber analysis. The expected performance of the back azimuth estimation is dependent upon the mutual SNR between channels and upon the length of the interferometer beam. In an ideal world we would simply use a very long interferometer arm to produce an accurate estimate the direction of the back azimuth or the phase velocity of the propagating wave. In the real world, the coherency between sensors decreases as the distance between sensors increases. Another tradeoff must be made and the question becomes “How close must the two sensors be placed to provide the desired accuracies?”. The accuracy of back azimuth estimation also degrades as the two sensors are placed close to each other. One obvious reason that this back azimuth estimate degrades is that the length of the “triangulation arm” is too short. Another obvious reason that the back azimuth estimate degrades is that the undesired noise (as well as the signal) becomes very coherent between two co-located sensors; resulting in a degraded SNR for this measurement condition.

The extension to multichannel back azimuth estimates using several pairs of sensors is essentially the averaging of several “paired-sensor back azimuths”. The frequency-wavenumber processing to estimate back azimuth is done taking a single monochromatic frequency at a time over all sensors for a given time window of observation. For stationary signals, long observation time increases the effective SNR. For transients, the time window is fixed by the short duration of the transient event and the SNR improvement achieved by averaging can only be accomplished by averaging over frequencies. The beamforming approach to estimate back azimuth represents a search for the maximum SNR obtained by “coherently stacking” the data using weighted-delay-and-sum that would correspond to waves propagating across the array from various assumed directions and velocities.

1.5. Background for using multichannel data for seismic data and site characterization

The successes of these techniques for triangulation using differential times of arrival, beamforming, and frequency-wavenumber analysis in various seismic applications in known geologies are well documented in the literature. The applications of multichannel arrays in unattended ground sensor (UGS) scenarios, however, often do not afford a full characterization of local geology before the deployment. Furthermore, operational constraints may limit one’s ability to place sensors into standard configurations. The data gathered nevertheless retain some value for determining source position and for suppressing noise. In addition, the data may also be exploited to characterize the deviations of the geologic conditions from assumed homogeneity.

One might try to use comparisons of wavefield measurements taken at different spatial locations to estimate average¹³ site conditions as well as local deviations from the average.

¹² This simple but important result that the variance of the differential-time-of-arrival measurements in an array is inversely proportional to the ratio of the power in the coherent noise plus interference (SNIR) is shown by Followill [Ref. 3, p. 110].

¹³ Johnson and Dudgeon [Ref. 4, p. 229] discuss problems to be expected because of non-stationarity and suggest that Cholesky factorization of the spectra combined with averaging piecewise

This might be done with multiple subarrays. If the different subarrays each have identical spatial sampling patterns then the comparisons between the subarrays may be made directly. Under certain conditions, the subarrays may be invariant upon rotation and high-resolution eigenstructure array techniques may be applied. Furthermore, selecting subarray designs that possess special properties such as minimum redundancy in their sampling patterns; economy in the total number of sensors required may be obtained.

In order to achieve a greater wavenumber bandwidth, convolving the subarray pattern with a fractal scaled pattern with larger dimensions will retain this economy of sensors required. In addition to the economy of sensors, each subarray provides some enhancement of the SNR in a bandwidth of wavenumbers. This increased SNR provides an associated increase in the coherency of signals between subarrays. These subarrays may be then taken as the elements at the ends of longer interferometer beams (for signals with higher SNR's) to estimate back azimuth and slowness properties of the wavefield. This provides a fractal scaling of the dimensions of the geologic heterogeneities. The smaller wavenumbers (which correspond to longer wavelengths and lower frequencies) provide less resolution of in the estimates of the back azimuth and slowness; but are less perturbed by the localized heterogeneities. Residuals of the higher wavenumber estimates relative to the lower wavenumber estimates provide a measure of the local heterogeneity at each subarray.

This observed variability in the spatial properties of the wavefield must be taken into account when evaluating the significance of different spectral signatures from the bunker. Also, the average properties would be considered to be the characteristics most portable from site to site. Moreover, the variabilities at the site relate directly to the array design; both with respect to numbers of sensors needed in each subarray and with respect to the geometrical pattern needed to obtain a specified level of performance against desired objectives for the array. The Y-pattern array convolved with a fractal scaled Y-pattern is reasonably robust¹⁴ against small perturbations in element mislocations and in small scale geologic heterogeneities. This is achieved through spatial averaging and by near-uniform sampling of the coarray¹⁵ space. (The coarray is the autocorrelation of the array aperture function: the sampled coarray is the correlation of the sampled aperture function. Fig. 2 shows the coarray pattern for this high-frequency array configuration used in these experiments. The two-dimensional Fourier transform of the coarray is the wavenumber spectral response of the array.)

The "Y-Y" array design with a fractal scaling of 3 between the "large Y" of the array and the "small Y" subarrays and with a total aperture of 6 meters provides 16 channels of vertical seismic geophone data. These data were beamformed to obtain an estimate of the optimum velocity or slowness to use in the following analysis. A value of 1500 meters per second obtained by this process has been used in these analyses.

2. MEASUREMENTS

will help condition the autocovariance matrix and improve its Toeplitz character. Clarkso shows that multiple "snapshots" in space or time may be averaged to reduce the variance of parameters; and demonstrates that, although the variance and bias of covariance estimates with increase of lag, averaging improves the estimates and should be used.

¹⁴Followill [Ref. 3, pp. 112-114] demonstrates that the "Y-Y" design for 16 elements is significant perturbations of array response due to errors in array sensor element position in a square array or a 16-element Prime-Ring (such as the NORESS, ARCESS, FINESS, GERESS, or IS concentric circles array design.

¹⁵Johnson and Dudgeon [Ref. 4, p. 94] discuss the importance of the pattern of the correlated coarray; [Ref. 4, pp. 94-103] present some illustrations of sampled apertures and their correlation and [Ref. 4, pp. 103-105] discuss filled arrays, sparse arrays, and random arrays.

We select a small subset of the various sources and experiments to illustrate the procedures and results. Initially, we performed a quick look and preliminary analysis of the high-frequency data set through trend removal, pilot spectral analysis, time-frequency spectrogram analysis of all the data. From these preliminary analyses we excised subsets of data to subject to a more detailed analysis.

2.1. Three-component analysis to search for characteristics of fundamental mode Rayleigh wave

The fundamental mode of the Rayleigh wave propagates with a particle motion that describes a retrograde ellipse in the vertical plane through the source and sensors. This implies that the vertical component is out of phase with the radial horizontal component by ninety degrees. The Hilbert transform produces a ninety-degree phase shift and may be used to transform the vertical component so that the transformed vertical geophone data will be in phase with the radial geophone data. We were unable to identify any Rayleigh wave signals in the frequency bands between 40 Hz and 400 Hz; and in the slowness range determined by the reciprocals of 750 meters per second and 250 meters per second. Based upon the negative results for this analysis we focused our remaining attention solely on the recorder channels containing the vertical component data.

2.2. Sources and time windows selected

We selected three different sources that represent the wide spectrum of available sources. These three sources will be described, then the signal processing used on each source, and then the plotted results.

The first source is a mechanical vibration produced by an electromechanical shaker using a controlled sweep of frequency from 45 Hz to 1000 Hz over a time interval of 16 minutes. The second source is transient impacts of a calibrated hammer (routinely used for modeling and analyzing responses of structures) striking the floor of the underground bunker. The third source presented in this report is a diesel generator (used as an auxiliary power source for the bunker). The data from the diesel generator is representative of the changes in signals as the diesel generator is turned on and reaching steady-state conditions.

The data for these experiments was collected on Digital Audio Tape recorders using 16-bit (90 dB) digitizers and sample rates of either 6000 or 12000 samples per second. These data were processed by (a) removing the trends from the time series, (b) bandlimiting and downsampling the data to 2000 samples per second, (c) delay-and-sum the data to construct a beamform corresponding source at a back azimuth of 180 degrees and a velocity of 1500 meters per second, (d) calculating auto-regressive spectral models of this beamed data, (e) apply an adaptive noise canceler to remove narrow spectral lines produced by the recording instrumentation, (f) calculate short-time-window maximum entropy method¹⁶ spectral models of the noise-canceled beam for sliding (50% overlapped sections) windows through the data, and (g) computing both the rms power in each short-time window and the likelihood ratio (presented as histograms in the plots) of the model coefficients in each short-time window.

2.3. Presentation of the data and signal processing results

Figures 3, 4, 5, 6, and 7 show the signal processing results for three different sources inside the bunker. These five figures have four parts (i.e., a, b, c, and d) each of signal processing and analysis results. These figures will be described, by source and time window, then the

¹⁶Candy [Ref. 1, p. 353] shows that, by using forward and backward projection, the maximum spectra do not exhibit sidelobes due to windowing of the data.

descriptions of the details will follow. There are five figures, a different one for each time window. The first three figures (Figures 3, 4, and 5) are short selections of the electromechanical shaker to illustrate about 10 seconds of data for an interval where the frequency-modulated (FM) chirp begins at frequencies of 50 Hz, 60 Hz and 80 Hz. The fourth figure (Figure 6) illustrates about 10 seconds of data that contains a transient impulse of the hammer impacting the floor. The hammer impact occurs nearly midway through the time window. This hammer impact is barely visible, and only on the spectrogram. The fifth and final figure (Figure 7) illustrates about 20 seconds of data that covers the time interval from just prior to the diesel generator starting up and continuing through the steady-state phase as the generator reaches equilibrium conditions.

Part (a) of each figure is the raw data for each of the 16 channels of vertical geophone data. Part (b) is the beamformed result using the 16 vertical channels of raw data. Part (c) is the modeled spectrum of the beamformed data, before and after adaptive noise cancellation to remove recording instrumentation noise. Part (d) is the spectrogram, the rms, and the ("indexed")¹⁷ histogram of model parameters for each short-time window estimate of the spectra.

3. RESULTS

We may summarize the results of our processing and analysis of these high-frequency array data as:

- (1) Maximum coherent bandwidth of the seismic signals (propagating to a range of 150 meters through alluvium, colluvium and tuff) appears to be limited to less than 300 Hz for each of the three different sources characterized in this study.
- (2) Spectral shaping or pre-whitening of the data (both signals and noise) by the use of a first-order differencing operator was indicated by the pilot signal analyses. This suggests that the data could be treated as 1st-order Markov-type in future signal modeling and analysis.
- (3) No "in-quadrature" (i.e., Hilbert transform) relationships between the vertical and radial sensor components were observed, indicating that the dominant signal types observed in this experiment were of the compressional seismic or P-waves rather than of the Rayleigh waves.
- (4) Multichannel, coherent array beamforming techniques were successful for frequency bands with signal-to-noise ratio (SNR) greater than unity. This (limiting value of unity SNR) corresponds to a value of the magnitude of the complex coherency of about 50% between channels. Greater time-bandwidth products and greater numbers of subarrays would provide better performance; but this would only be applicable for stationary signals, not for the nonstationary transients investigated in this experiment.
- (5) Model based signal processing of short time windows characterized nonstationary source signals using a time-frequency analysis. Maximum model orders for initial estimates were predetermined (a priori) by the number of samples in the analysis timewindow. Final model order number was determined (a posteriori) by the number of "peaks in the estimated spectrum". The typical a posteriori model order was less than one-fourth the model order estimated a priori.
- (6) Potential pathologies that might be expected from unknown geologic heterogeneities may

¹⁷ Clarkson [Ref. 2, p. 282] suggests "importance sampling" based upon maximum likelihood various spectral lines.

be minimized by the array design and by the use of subarray sampling of multiple spatial locations.

4. CONCLUSIONS

Time-frequency analysis of vertical-axis geophone data that had been beamformed toward the source (a) recovered the quasi-stationary “chirp” signals from an electromechanical shaker, (b) detected an obscure transient impact, and (c) characterized the non-stationary “turn-on” of a diesel generator.

Likelihood ratio tests (presented as histograms in Figs. 3-7) indicate that only the lower frequencies (i.e., frequencies less than 250 Hz) had adequate signal-to-noise ratios to be statistically significant in assigning model parameters to the signals. The initial array design was targeted for 1000 Hz bandwidth to provide an opportunity to evaluate maximum coherent signal bandwidth.

Multiple subarrays, sampling different spatial locations, proved to be robust with respect to local geological heterogeneities and with respect to minor perturbations of the “as deployed” spatial positions of the individual sensor locations. Variations of an individual sensor position by values less than one-tenth the average distance of that sensor to its nearest neighbors produced array response variations less than 1 dB.

Our results indicate that a larger aperture array (perhaps four or five times the 6-meter aperture of this high-frequency array) would provide utility for monitoring the transient events observed in this experiment. (One of the smaller configurations of a proposed aircraft-delivered subarray with an aperture of 25 or 30 meters would be consistent with this final conclusion.)

Finally, we would recommend deploying at least four subarrays at a range of about 150 meters from the bunker, with one subarray in each quadrant or azimuth about the underground facility to provide monitoring the bunker.

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Figure 1. Surveyed spatial positions for high-frequency array elements

Figure 2. Correlation sample points for high-frequency array

Figure 3 a. 16 vertical channels of raw data

Figure 3 c. Spectra of beamformed data

Figure 3 b. Beamformed data

Figure 3 d. Spectrogram, rms power and histogram

Figure 4 a. 16 vertical channels of raw data

Figure 4 c. Spectra of beamformed data

Figure 4 b. Beamformed data

Figure 4 d. Spectrogram, rms power and histogram

Figure 5 a. 16 vertical channels of raw data

Figure 5 c. Spectra of beamformed data

Figure 5 b. Beamformed data, rms power and histogram

Figure 5 d. Spectrogram, rms power and histogram

Figure 6 a. 16 vertical channels of raw data

Figure 6 c. Spectra of beamformed data

Figure 6 b. Beamformed data

Figure 6 d. Spectrogram, rms power and histogram

Figure 7 a. 16 vertical channels of raw data

Figure 7 c. Spectra of beamformed data

Figure 7 b. Beamformed data

Figure 7 d. Spectrogram, rms power and histogram