tended period of calm weather and system noise increases when the sea ground is towed into them). When profiling on fresh water, it may be necessary to use a «floating» ground.

A major source of electrical noise is the key pulse line by which the seismic source is triggered. The electrical connection it forms between the source and receiving system should be eliminated by triggering through an optical isolator. This is especially important when using high-voltage sources that operate by electrical discharge, *i.e.* boomers or sparkers.

Electrical noise can also be caused by induction. Power cords and signal lines should be well separated; an easily remembered rule being «power lines on port side, signal lines on starboard side». All electrical lines should be as short as possible and any excess length should be looped in figure-8 patterns rather than being coiled. The so-called «french coil», a folded-over figure-8 widely used because it avoids kinks in long cables, is very effective. Radars, radios and other high-voltage equipment should not be operated during data acquisition unless absolutely necessary. If a microwave positioning system is used, special care should be taken that its console and all related cables are well removed from signal lines and the recording equipment. These precautions will not necessarily eliminate all electrical noise, but they will simplify the task of locating any remaining sources. All sources of electrical noise should be identi-

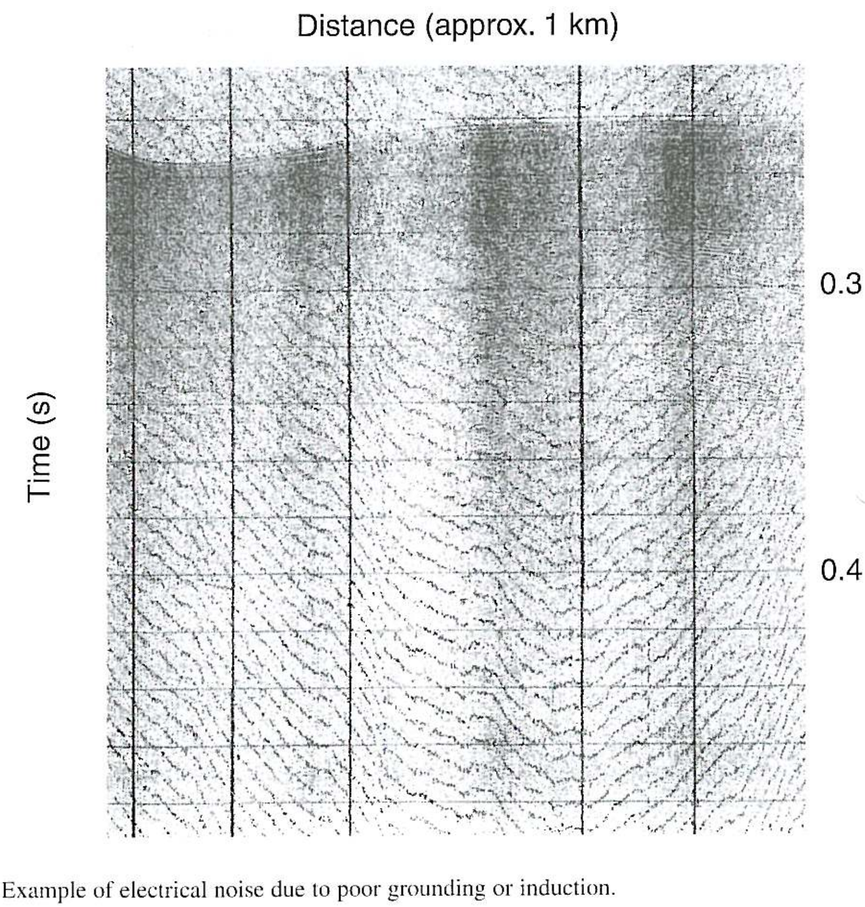


Fig. 12. Example of electrical noise due to poor grounding or induction.

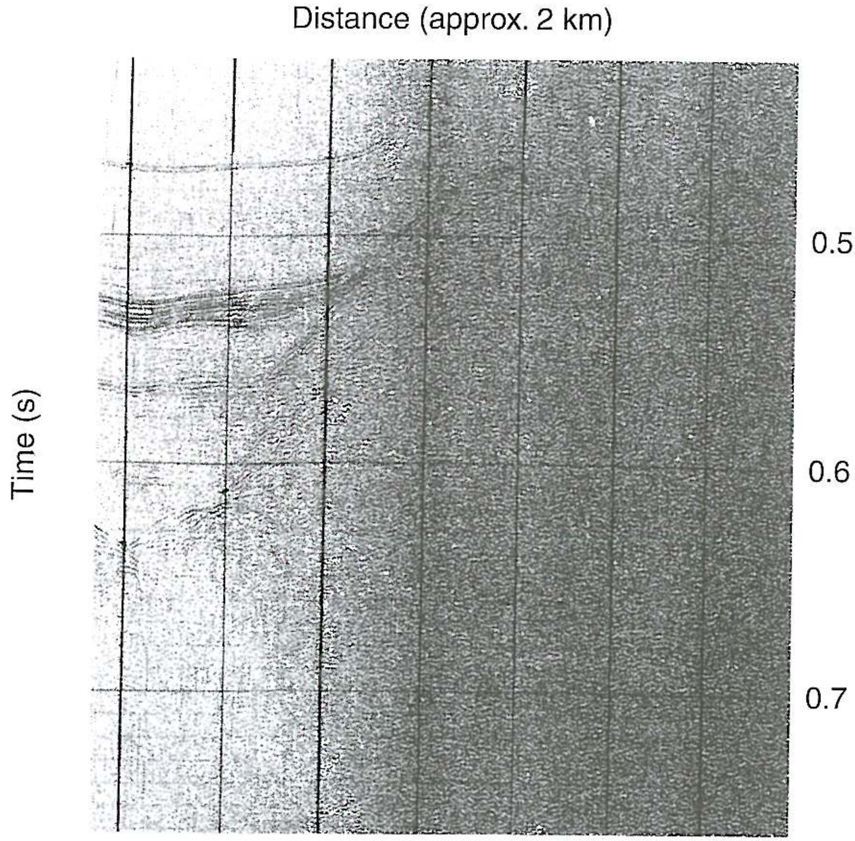


Fig. 13. Example of operational noise due to nearby ship traffic.

fied and their effects minimized while the vessel is still at dockside. A few hours spent there will be rewarded amply when profiling.

An example of electrical noise on a field record is shown in fig. 12. Notice the sinuous organization that occurs due to the noise being out of phase with the plotting cycle of the recorder. This organization is typical of electrical noise due to poor grounding or induction.

6.2. *Reducing operational noise*

The most common sources of operational noise are the survey vessel, other activities in the survey area, towing procedure and sea state. The easiest way to diagnose which sources are contributing to a particular problem is to listen to a speaker tapped into the hydrophone signal line. This allows the recognition power of the ear to be used for identifying sounds that would not be visually obvious when plotted. For example, the ear immediately distinguishes between mechanical clanking and the gurgling of water, but the two have a very similar appearance on a recording. Mechanical sounds indicate that either the survey vessel or nearby machinery is the source. When the sound of water is heard, either the towing of equipment or wave action is the source. When operational noise is low, the speaker is quiet except for the sound of the seismic source quickly followed by a de-

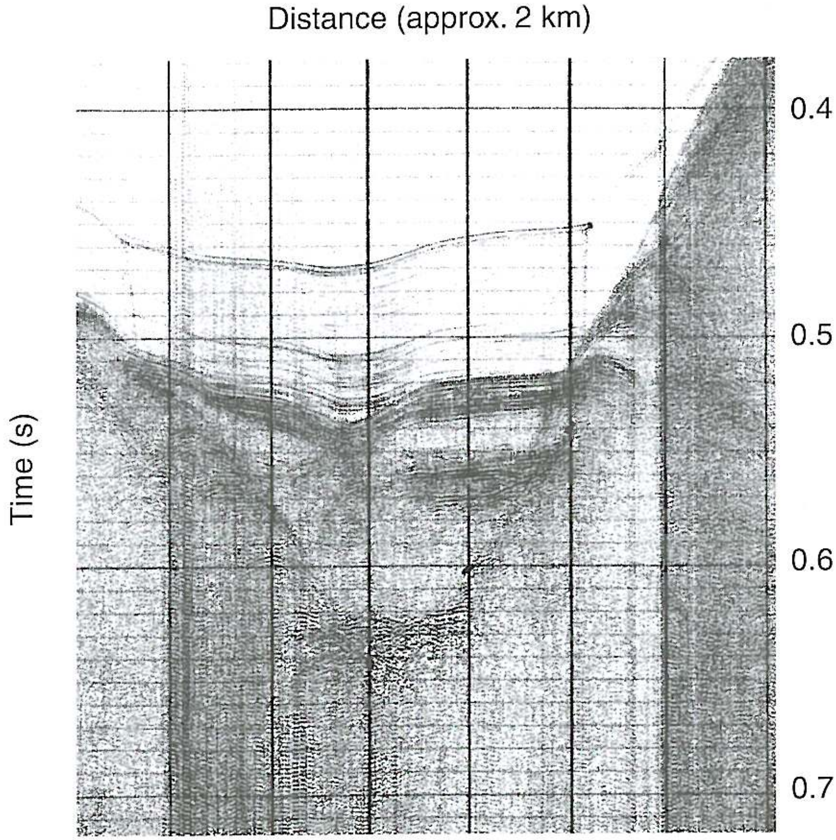


Fig. 14. Example of organized ship noise that might be mistaken for electrical noise.

creasing sequence of echoes. If the source is a low-power boomer, the sound is reminiscent of the sonar in old submarine movies. If the source is a large airgun, a roar is heard.

The best way to minimize the noise of the survey vessel is to choose a quiet ship. Older wooden work boats with slow turning engines and dry exhaust are usually very quiet if the shaft, propellor and rudder mounts are in reasonably good condition. Vessels with steel hulls and powerful hard-mounted engines tend to be noisy, especially at high engine speeds. Underwater exhaust is noisy on any vessel, particularly in combination with an outboard drive. Variable pitch propellers and bow thrusters also tend to be very noisy.

If a quiet vessel is not available, the next best solution is to tow the hydrophones at some distance. This is not always the simple solution it seems to be, especially in shallow water. Depending on the type of water bottom, the required tow distance may be found to be rather long because the noise of the vessel is repeatedly reflected from the bottom and the water surface, becomes trapped in the water layer and decreases slowly with distance. If the hydrophone is towed at a large distance, the source must be towed similarly to retain good source-receiver geometry. If the source is towed a long distance behind the vessel, it is usually necessary to attach floatation to its firing line. Further complications arise because long tow distances

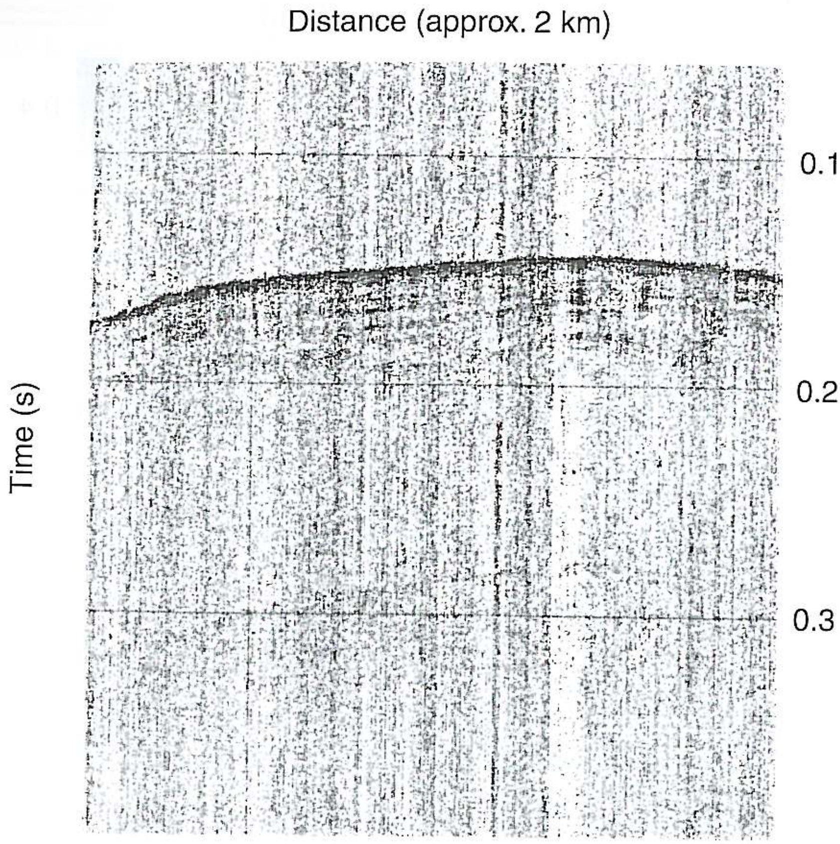


Fig. 15. Example of operational noise due to a stormy sea state.

hamper maneuverability and become a nuisance when working close to shore or in congested areas.

Noise due to other activities in the work area may be associated with the survey work or may be totally unrelated. That associated with the work is easiest to control. A common source is a support vessel approaching or accompanying the survey vessel. Support vessels should stand by at a far distance during data acquisition. Other sources, such as dredges, may have periodic interruptions for maintenance or location change. Data acquisition in the vicinity of these sources should be scheduled during such interruptions. If the source is traffic in a shipping lane, work must be done as opportunities arise. This problem can be quite severe. For example, the noise of a laboring steel-hulled tug has been observed to obliterate reflection data at a distance of several kilometers (fig. 13). Profiling was interrupted for more than two hours as it passed through the survey area. The noise in fig. 13 appears to be disorganized, but that is not necessarily the case. An example of organized ship noise is shown in fig. 14. Because of the organization it might be mistaken for electrical noise. The ear would less likely be confused.

Tow noise is decreased by «fairing» all towed equipment and cables to reduce the «strumming» caused by vortex shedding. Hydrophones should be extremely fair and not oscillate under tow. Linear hydrophone arrays should not be coiled tightly for storage or shipment. If they cannot be kept straight, they should be looped loosely in a figure-8 pattern and then laid straight in a warm place for as long as possible before being used. Sometimes the only available array is one that has been coiled for a considerable period of time. Such an array usually retains a helical shape under tow so that the individual hydrophones are located at different depths and the signals from them are not in phase. Furthermore, such an array usually «cork screws» through the water so that the total signal varies from shot to shot. In less severe cases the problem can be solved by attaching a drogue or large diameter rope to the tail of the array so that it pulls the array taut at profiling speed. In more severe cases, or when the profiling speed is low, a coiled array can be straightened by taping it to a straight length of wood. If the cross-sectional area of the wood is small it will have little effect on the response of the array, especially if the tape is located between individual hyrophones to avoid interfering with their responses.

Noise due to sea state is best avoided by scheduling work during periods of good weather. If work must be done in choppy conditions, the noise may be reduced by deploying the hydrophones in the lee of the survey vessel or, if they must be astern, by towing them deeper. Any increase in tow depth should be as little as necessary so that the loss of resolution due to ghost reflections is minimized. An example of sea-state noise is shown in fig. 15. Notice the streaky appearance. It is typical.

7. Recording the data

Field data is usually recorded both on paper and on an electronic medium in either analogue or digital format. The paper recording is annotated in the field and referred to later, during analysis of the data. Recording is done on an electronic medium, either analogue or digital, so that the data can be processed and/or plotted at different scales during analysis.

Since the two types of recordings have different uses, the hydrophone signal is usually split with the part going to the paper recorder being «conditioned», *i.e.* amplified and filtered, to produce the best image and the part going to the electronic recorder being retained in relatively pristine form, *i.e.* no application of uncalibrated gain and minimal frequency filtering.

Automatic Gain Control (AGC) should not be used prior to electronic recording because its effect is not recoverable and can mask amplitude information.

Frequency filtering prior to electronic recording should be used sparingly. If filter slopes are selectable, it is better to use gentle slopes, *i.e.* 6 or 12 decibels per octave, rather than steep slopes, *i.e.* greater than 18 decibels per octave. The steeper a filter slope is, the more the wavelet is lengthened by a phenomenon known as «filter ring» which reduces resolution. Filter ring is particularly severe in the case of steep low-cut filters. Notch filters should be avoided entirely because they produce distortion so severe that successfully following it with other processing is difficult.

It is best to use frequency filtering only on the paper record and apply no filters to the electronically recorded data.

7.1. *Recording on paper*

Paper recordings are usually one of two types; «wiggle trace», *i.e.* fig. 7 or «variable density», *i.e.* fig. 1. If the trace amplitudes are represented as oscillating curves, it is said to be a wiggle-trace recording. If the trace amplitudes are represented in a grey-scale format, it is said to be a variable-density recording. Either can be very helpful when choosing instrumental settings such as gain levels.

If the trace length is relatively short, it is usually sufficient to use constant gain. If the length is great enough that the amplitudes of deeper reflections are too small, a linearly increasing Time-Varying Gain (TVG) will improve the resolution of deeper reflections while avoiding too much amplification on shallow reflections. Automatic Gain Control (AGC) is sometimes useful for display purposes but it should be used with care because it masks changes in reflection amplitude that can be a valuable source of information.

Setting gain levels is largely a matter of trial and error because signal strength can vary greatly over a survey area. Geologic variations can cause reflections to be strong in some places and weak in others, the amount of variance seldom being known prior to acquiring the data. Therefore it is usual to run a test profile over representative portions of the area to aid in choosing appropriate gain settings.

If the paper recorder has contrast and threshold controls, start the test profile with the contrast control at a medium setting and the threshold as low as possible. Then begin increasing the gain until the desired reflections are visible.

After the gain has been chosen, spend some time «tweeking» the settings to improve the appearance of the paper record. The situation is a bit like focusing a camera, when close to «in focus» slight adjustments can produce big improvements. Reduce the contrast to as little as necessary to avoid a «grainy» appearance. Increase the threshold slightly to see if the background noise can be reduced without losing the weaker reflections.

Do not get carried away with changing instrument settings, however. Once a good paper record is obtained, leave the settings alone unless it becomes absolutely necessary to change them. If possible, entire profiles or even entire surveys should be recorded at the same settings. This allows changes in reflector amplitude to retain significance during data analysis.

7.2. *Analogue recording on magnetic tape*

The traditional way of recording analogue data for subsequent electronical reproduction is to record it on magnetic tape. Frequency Modulation (FM) recording is better than Amplitude Modulation (AM). The bandwidth of a good FM tape recorder can be greater than 40 kHz, which is sufficient for pneumatic sources and large sparkers but not for the broadest band sources such as boomers and waterguns.

Tape recorded signals are usually conditioned either by constant gain or TVG. Care should be taken that the gain applied is not so much that the tape is saturated. That would cause the signal to be «clipped» and information would be irretrievably lost. If TVG is used, it should be well calibrated so that its effects can be accounted for during data analysis. The use of AGC is strongly discouraged.

7.3. *Digital recording*

The amplitude and phase of seismic signals often represent significant information. Subsequent processing can access this information only if the data are recorded digitally, and then only if the digitization has been done properly.

There are two principal parameters to consider when digitizing a seismic signal; the digital resolution, *i.e.* the number of bits used to represent the analogue amplitude, and the sampling rate, *i.e.* the number of digital samples per unit time. Both affect the fidelity to which the analogue signal is represented. Increasing either or both serves to improve the fidelity, but also increases the digital volume to be stored. The size of the data volume was a major concern when digital storage was expensive, but storage media are now so cheap that large volumes are no longer a problem.

When digitizing a signal, the analogue gain is chosen such that the analogue signal voltages occupy most of the operational range without exceeding that range. Because the reflectivity of the sea floor can change significantly, unexpectedly doubling or quadrupling the analogue signal strength, it is conventional practice to choose the gain level so that the upper one or two bits of resolution normally are unused. This acts as a safety factor to allow reasonable increases in analogue signal strength without clipping the digital signal.

At the time of this writing, 8-, 12- and 16-bit Analogue-to-Digital (A-D) converters are common with one bit required for sign (positive or negative amplitude). Inclusion of the safety factor has the effect of reducing these to 5-, 9- and 13-bit converters. For this reason, A-D converters less than 16-bit are not really useful for practical seismic applications. Since reflection seismograms typically have higher amplitudes at shorter record times and significantly lower amplitudes at longer record times, it is often advisable to apply a TVG before digitization in order to avoid clipping the early arrivals but still record the later arrivals with reasonable digital resolution. Since it is necessary to account for the amount of gain during processing, the TVG must be well calibrated or recorded as a separate channel.

The use of TGV becomes unnecessary as the digital resolution increases. The current tendency in high-resolution work is toward 24-bit systems (which are, effectively, 21-bit systems allowing for sign and safety factor). Some digital streamers used for oil exploration have 32-bit resolution and 64-bit streamers are being discussed. If an infinite digital resolution were possible, analogue gain would be entirely unnecessary.

The second principal digitization parameter, sampling rate, is traditionally chosen according to a few conventional rules-of-thumb. Recently there has been a paradigm shift in this regard, however, so it is necessary to discuss sampling rates in more detail than would have been necessary a few years ago.

8. Bandwidth, digitizing rate and seismic resolution

It is a tenet of conventional seismic wisdom that seismic signals have limited band widths, *i.e.* that there is no power above some finite «cutoff» frequency. The consequences of this assumption dominate the practice of digital seismology because the minimum rate at which an analogue signal should be sampled is twice the cutoff frequency (Shannon, 1949). After digitization, one-half the sampling rate is the highest frequency that the digital data can represent uniquely. It is called the «Nyquist» frequency of those data.

Theory states that only signals of limited bandwidth may be digitized without loss of resolution (Shannon, 1949; Slepian, 1976). The Heisenberg uncertainty principal requires that signals of limited bandwidth (such as sinusoids) have infinite duration and that signals of limited duration (such as pulses) have infinite bandwidth. The duration of seismic wavelets is limited because they are «causal», *i.e.* they do not exist prior to their onset times. Thus their bandwidth is infinite (McGee, 1998).

This presents a conundrum in practice because Shannon (1949) requires an infinite sampling rate if resolution is not to be lost. The implication is that at least some resolution is always lost when seismic signals are digitized, but the loss may be minimized by sampling as rapidly as possible. When a lower rate is desired, perhaps to limit data volume, a rate must be chosen that is adequate to provide the geologic information required.

9. Choosing an adequate sampling rate

In 1991, an experiment to determine how often signals generated by common marine seismic sources should be sampled was carried out on Lake Windermere in Cumbria, U.K. (McGee *et al.*, 1992). Under quiet conditions, the signatures of several commercial seismic sources were digitized at 204 000 samples per second. Spectral analyses confirmed that none of the signatures was obviously band limited. In fact, the power spectra of two commonly used sources, a small watergun and a low-energy boomer, were found to have exceptionally broad, smooth shapes. This is illustrated in fig. 16a,b where a 105-Joule boomer signature and its power spectrum are shown. The spectrum decays slowly without any obvious cutoff frequency. In situations such as this, Slepian (1976) suggests choosing a small power level at which to define an «effective» bandwidth. If a level of 40 db below maximum power (1% of maximum amplitude) is chosen, fig. 16b exhibits an effective bandwidth well above 30 kHz. Thus, by Shannon (1949), the digitizing rate for the 105-Joule boomer source should be *at least* 60 000 samples per second.

It is often argued that fast sampling rates are unnecessary because high frequencies are absorbed during propagation. This ignores the fact that a sampling rate is adequate only insofar as it is conducive to retrieving the desired geologic information from the seismogram. Retrieval of high-resolution information is accomplished by digital processing. Therefore a sampling rate should be chosen that is adequate for the intended processing procedures.

For reasons of computational efficiency, many procedures are implemented in the frequency

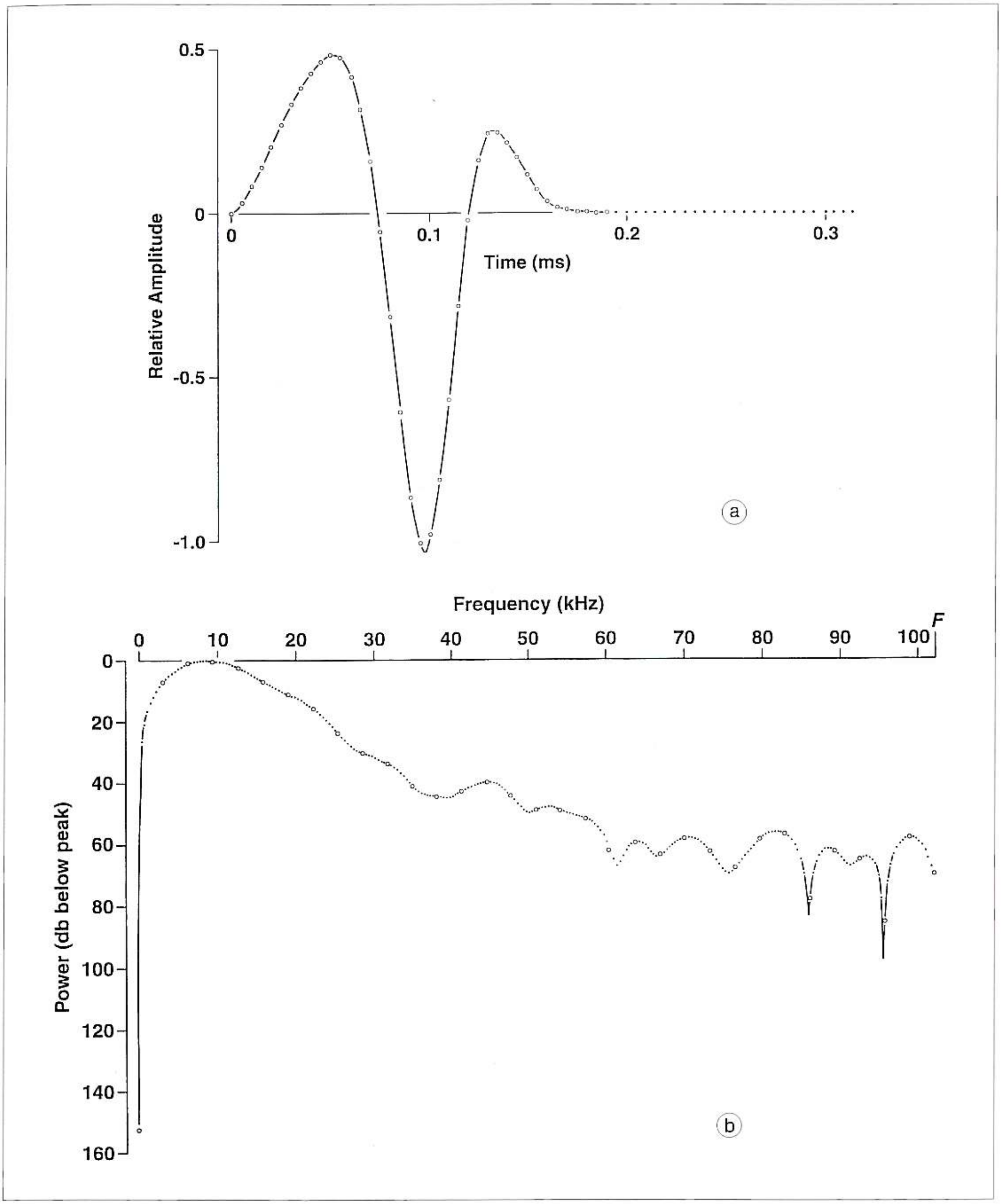
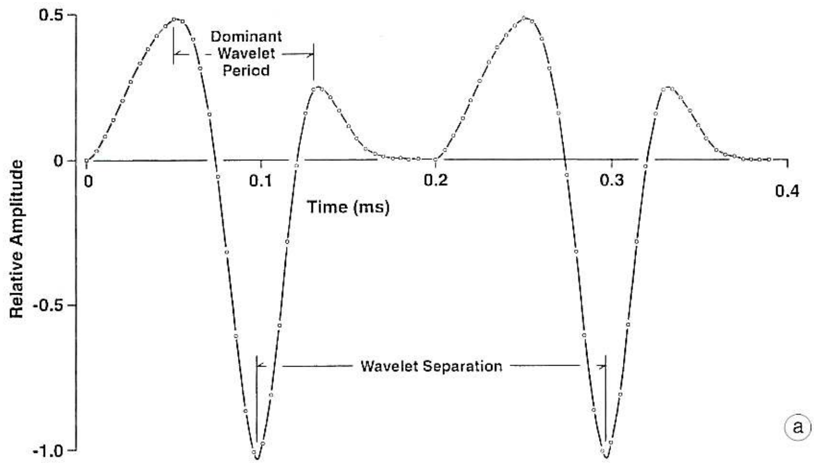


Fig. 16a,b. Axial far-field signature and power spectrum of a 105-Joule boomer source sampled at 204 000 samples per second: a) analogue signature approximated by the solid curve, digital samples indicated by circles; b) power spectrum, circles indicating a 64-point FFT and dots a 512-point FFT. The Nyquist frequency (F) is 102 kHz.



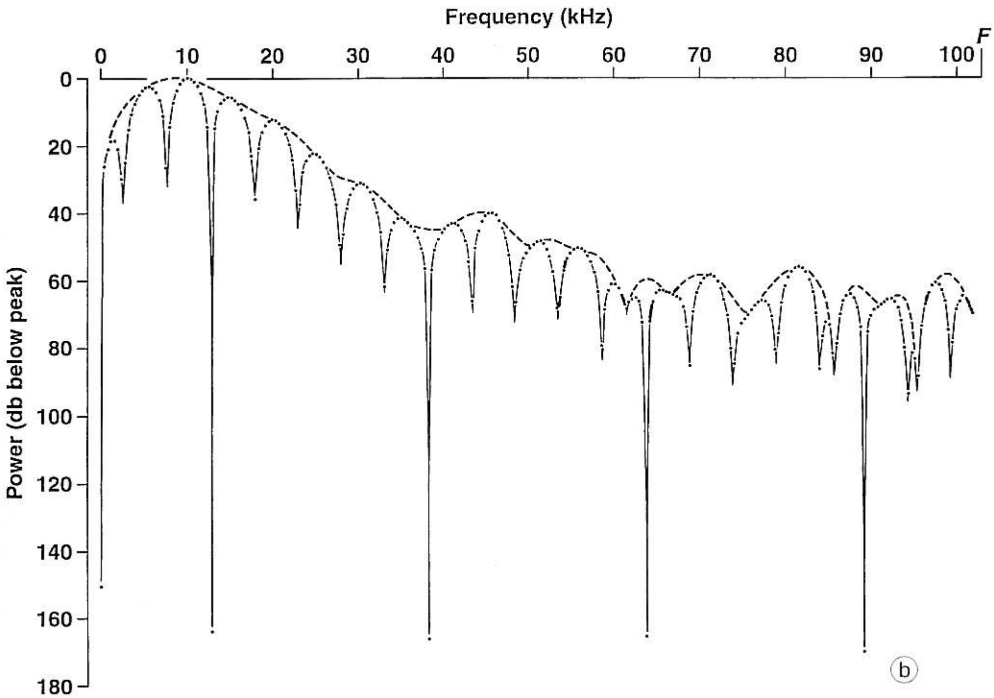


Fig. 17a,b. Simple «seismogram» and its power spectrum: a) «seismogram» constructed by adding a time-shifted version of the signature in fig.16a,b to itself; b) power spectrum obtained by applying a 512-point FFT (solid curves and dots). The spectrum in fig. 16a,b is shown dashed for comparison. The Nyquist frequency (F) is 102 kHz.

domain. The way geologic information is represented in the frequency domain is illustrated in fig. 17a,b by a simple «seismogram» and its power spectrum. In the time domain (fig. 17a), the «seismogram» consists of two identical copies of the boomer wavelet in fig.16a separated by an interval of time. The relative magnitude and polarity of the wavelets and the length of time between them is the «geologic» information to be retrieved. In the frequency domain (fig. 17b), that information is coded into a pattern of destructive interference (solid line) which is modulated by the power spectrum in fig. 16b (dashed line). The pattern appears as a set of spectral notches whose spacing is determined by the interval between the wavelets and their relative polarity. Since the wavelets are of the same polarity, the notches occur at odd-integer multiples, or harmonics, of a base frequency whose period is twice the separation interval. (If the wavelets were of opposite polarity, notches would occur at all harmonics of a base frequency with period equal to the interval). Relative wavelet magnitude is coded into the depths of the notches. Every fifth notch being deeper than the others results from the separation interval being almost exactly 2.5 times the dominant wavelet period (fig. 17a).

The spectrum of a real seismogram would display the superposition of a number of interference patterns, one for each pair of wavelets present in the seismogram. The geologic information would be retrieved by a computational process that, in essence, identifies the base frequency of each pair, determines whether its pattern consists of only odd or all harmonics, and compares the depths of the notches. The accuracy of the computations increases as more harmonics of the base frequencies are included in the process. For this reason, the Nyquist frequency (and thus sampling rate) should be higher than that necessary merely to represent the source signature.

The rule-of-thumb adopted by some physical acousticians is that 8 to 10 sample intervals should occur within the dominant period of the signal (J. Hendrix, personal communication). The boomer wavelet in the above illustrations has 17 sample intervals within its dominant period. Thus the cited rule-of-thumb suggests that 100 000 samples per second would have been adequate for processing seismograms generated by the 105-Joule boomer. This agrees with the rate calculated for deconvolving the same source signature by Verbeek and McGee (1995).

10. Oversampling

When there are no good reasons to restrict sampling to a minimally adequate rate, the concept of «oversampling» (Alesis Corporation, 1996; reproduced herein as Appendix B) offers good reasons for sampling much faster. Oversampling exploits the speed of modern A-D converters and the vastness of modern storage to define Nyquist frequencies so high that significant signal power at higher frequencies would be very improbable.

Oversampling is well suited to seismic signals because the power of seismic wavelets decays naturally as frequency increases. Therefore it is not difficult to select a Nyquist frequency that is several octaves above the highest frequency of interest, *i.e.* above the upper limit of the effective bandwidth. This promotes increased resolution both by avoiding losses associated with severe high-cut filtering prior to digitization and by improving signal coherency during digital processing.

11. Storage media

In light of the high volumes of data generated by the practice of oversampling, something should be said of suitable storage media.

Magnetic tape is able to store very large volumes of data for very little initial cost, but it has a major disadvantage which makes the true cost of its use much greater than initially supposed. The disadvantage is that seismic traces on magnetic tape must be accessed sequentially. Access time is similar to the time required to make the field recording. Thus, whenever processing data on magnetic tape, an enormous amount of time is spent searching the tape. This compounds processing costs greatly.

A fast way to record digital data is on a large-capacity hard disk. This allows random access to traces and reduces processing time greatly.

Hard disks are not reliable for long-term storage, however. For long-term storage, data on hard disks should be copied onto a more permanent media such as CD-ROMs which are cheap and relatively durable.

12. High-resolution digital signal processing

There are many techniques used to process seismic data digitally, most of them developed for deep exploration. Many exploration algorithms rely on assumptions that are not necessarily valid in high-resolution contexts, however. Discussed herein are a number of processes often applied to high-resolution single-channel data, some successfully and others not.

12.1. *Frequency filtering*

Frequency filtering is perhaps the most common of all processes applied to seismic data and is often used with little consideration of its negative effect on resolution. The resolution, or information-carrying capacity, of a signal is measured by its entropy (Shannon, 1948). Entropy is greater for signals that have smoother and more gently sloping power spectra. Filtering always reduces entropy to some extent (Khinchin, 1957). Filters that impose unnatural shapes on spectra, especially notch filters, do the most damage. In an extreme case of filtering so severe that the filtered signal is devoid of power over a continuous range of frequencies, all resolution would be lost.

The resolving power of any recording instrument is limited by its high- and low-frequency characteristics, but the resolution that an instrument is capable of providing can be maximized by avoiding frequency filtering as much as possible. As already discussed, high-cut filtering can be minimized by the practice of oversampling (Appendix B).

12.2. *Detrending*

Often, low-cut filtering can be avoided entirely by using the process of detrending (McGee, 1998). The trend of a seismic trace is a slowly varying voltage produced by some non-seismic process in the environment such as wave action. The seismic energy produces a more rapidly varying voltage that oscillates about the trend. If a low-cut filter would be used to attenuate the trend it would distort the low-frequency portion of the seismic spectrum and thereby degrade the results of subsequent processing. Instead of filtering, detrending models the slowly varying trend and subtracts it from the trace, thereby leaving the low-frequency portion of the seismic spectrum relatively undisturbed.

The detrending process is illustrated in fig. 18a-c. The input seismic trace containing a trend is shown in fig. 18a, the model of the trend in fig. 18b and the output in fig. 18c after subtraction of the model from the input. Figure 19a,b shows wiggle-trace plots that illustrate the application of detrending to a sequence of traces; (a) input and (b) output.

12.3. *Correcting for wavefront divergence*

Since seismic sources are of finite size, the wavefronts they generate expand as the energy propagates. This is called wavefront divergence and produces a reduction of the amount of energy per unit wavefront area. The result is that the signal strength decays as the distance propagated increases. The reduction in signal strength must be accounted for if true reflection amplitudes are to be to determined.

If the signal wavelength is large compared to the size of the source and the speed of propagation is constant, the divergence is spherical and the decay is inversely proportional to the distance propagated. The correction is then a linearly increasing factor equal to reflection time multiplied by the propagation speed.

It is usually assumed that the divergence is approximately spherical but it is possible to generate wavelengths smaller than the size of the source. In such a case, the radiation pattern of the source would not be spherical. Rather, its shape would be a function of frequency and the divergence correction would be vastly more complicated, *i.e.* it would have to be accomplished in the frequency domain.

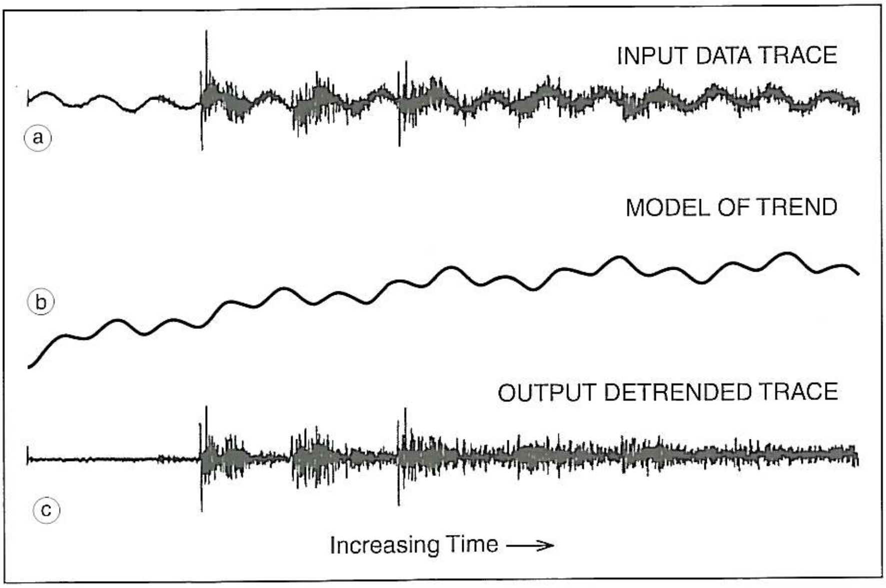


Fig. 18a-c. Illustration of the detrending process: a) input seismic trace containing a slowly varying trend; b) model of the trend; c) output trace after subtraction of the trend model.

This has been investigated by Verbeek (1992) for the case of boomer sources. In theory, a boomer should exhibit the radiation pattern of a piston, *i.e.* dependent on frequency and direction of propagation. This was found to be supported by field measurements. Verbeek concluded that the divergence of a boomer wavefront could be assumed to be approximately spherical only for propagation along the axis of the boomer plate. For this reason, spherical divergence corrections are valid for boomer data only if the source/receiver offset is short compared to the water depth and the reflectors are relatively flat lying.

Figure 20 shows short-offset boomer field data that were detrended and corrected for spherical divergence in order to produce the processed profile shown in fig. 7.

12.4. *Signature processing*

Digital processing can improve resolution significantly if a far-field source signature is recorded for each shot. Three types of signature processing are discussed herein: deterministic deconvolution, phase conjugation and matched filtering. Whichever one is used, it should be applied before correcting for wavefront divergence because the time-dependent divergence corrections distort the shape of the source signature.

The three processes are similar in that they all remove the phase of the source signature from the complex spectrum of the seismogram. This «compresses» each reflected wavelet into a short pulse that is symmetrical about a central lobe. It is this compression that provides the increase in resolution. The central lobe may be a peak or a trough, depending on the polarity of the reflection. With proper scaling, the amplitude of the central lobe can be used to estimate the reflection coefficient of the reflector.

The three processes differ in the amplitude spectra of their outputs. Deterministic deconvolution divides the amplitude spectrum of the seismogram by that of the source signature, thus outputting an amplitude spectrum of bandwidth

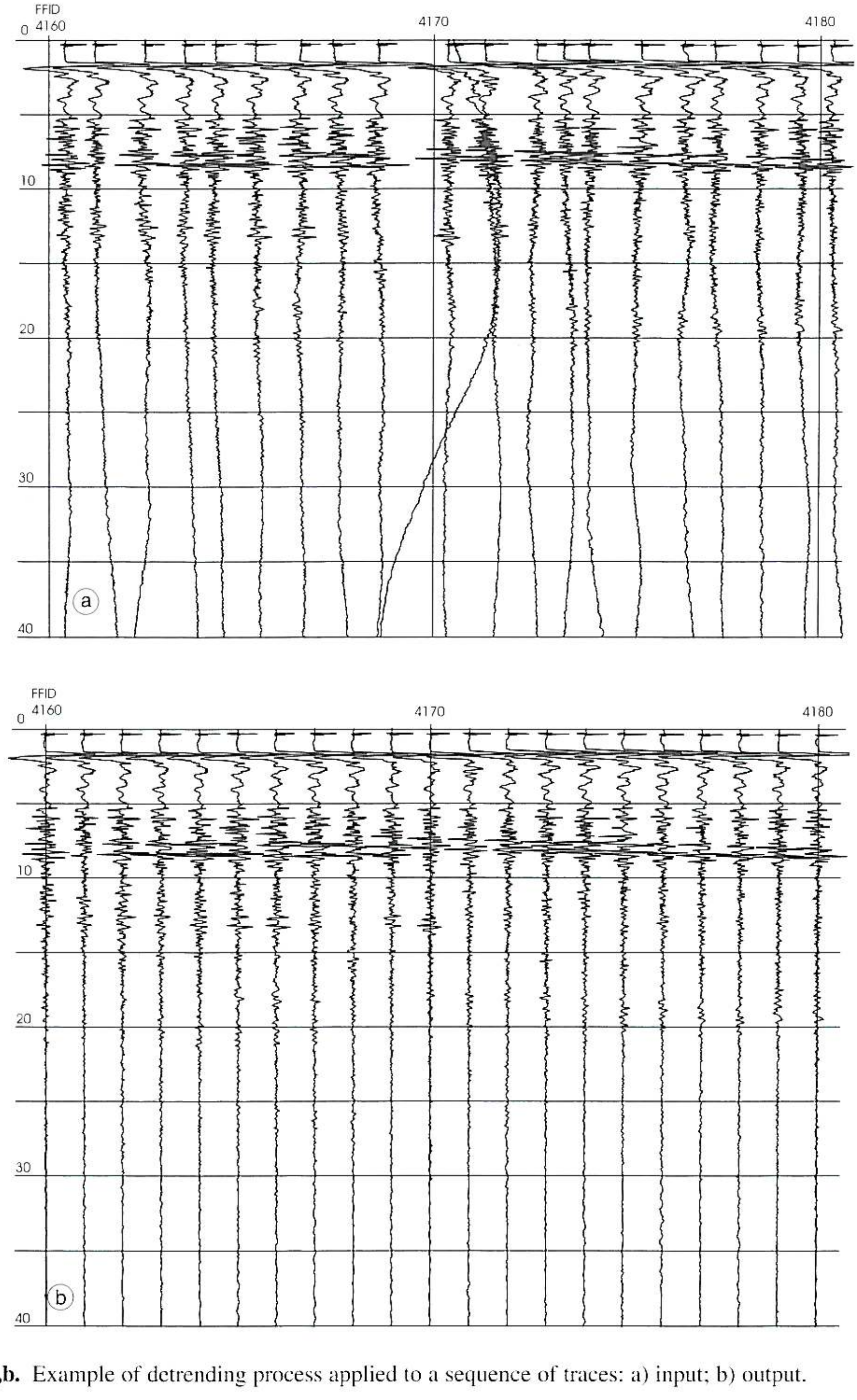


Fig. 19a,b. Example of detrending process applied to a sequence of traces: a) input; b) output.

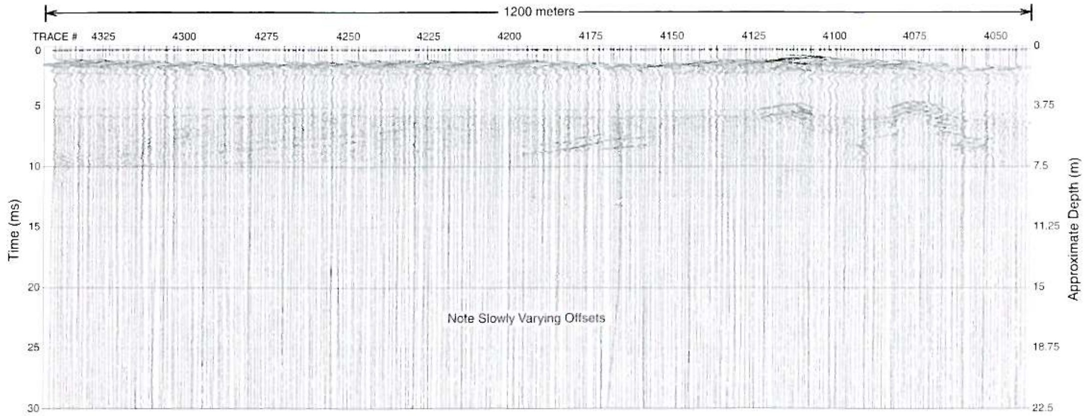


Fig. 20. Raw field data acquired using a 105-Joule boomer source with constant gain and digitizing at 200 000 samples per second. These data were detrended and spherically corrected to produce the profile shown in fig. 7.

broader than that of the seismogram. The amplitude spectrum of the phase conjugation output is the same as that of the seismogram. Matched filtering multiplies the amplitude of the seismogram by that of the source signature, thus outputting an amplitude spectrum narrower in bandwidth than the seismogram.

If the sampling rate is sufficiently high, resolution is proportional to bandwidth (Rabiner and Gold, 1975), thus deterministic deconvolution should provide the greatest resolution and matched filtering the least. Deconvolution is notoriously sensitive to noise, however, while phase conjugation is fairly robust in the presence of noise and matched filtering is even more so. Thus, in practice, phase conjugation often produces preferable results when the noise level is moderate and matched filtering when the noise is severe.

The application of source signature processing to single-channel seismic profiles is discussed by Partouche elsewhere (2000, this volume).

12.5. *Common mid-point stacking*

Multi-offset profiling for the purpose of summing common mid-point data traces is widely used for petroleum exploration. The summation, known CMP stacking, is done after correcting the traces for time delays, called «normal moveout», associated with increasing source/receiver offset distances and varying speeds of propagation.

In theory, stacking can be done in such a way that the signal-to-noise ratio of the data is improved; *i.e.* signal is enhanced by in-phase summation while noise is attenuated by out-of-phase summation. In practice, the signal can be seriously degraded if the corrections are not sufficiently precise to align individual traces to within a fraction of the dominant signal wavelength. To a large extent, that precision depends on the accuracy to which the locations of the source and each receiver can be determined.

The dominant wavelengths of high-resolution signals are short, *i.e.* less than a meter. Thus the locations of boomer sources and receivers must be determined within a few tens of centimeters if the signal is to stack in phase. Such a degree of accuracy is almost impossible to achieve while profiling on water.

Another problem that affects the stacking of marine shallow waterborne signals is the directivity of some high-resolution sources. If the power emitted by a source varies with direction and frequency, traces at various offsets must be weighted in both the spacial domain and the frequency domain prior to being stacked. This is not impossible but is computationally intensive to the point of being, for most intents and purposes, impractical. The boomer source, widely used in shallow seismic profiling, is a good example of a strongly directional source (Verbeek, 1992). When operated under broadband conditions, it should not be expected to provide good stacked results without frequency-dependent weighting.

It is commonly claimed that stacking may be used to attenuate coherent noise such as water-layer multiples. This is a valid claim only if the moveout of the noise is significantly different from that of the signal. That is rarely the case for high-resolution data because propagation speeds in unconsolidated sediments are comparable to the speed in water. Thus primary reflections from within shallow sediments exhibit about the same amount of moveout as do water-layer multiples and stacking is not effective for discriminating between them.

13. Conclusions

Some theory and various practical experiences concerning high-resolution seismic profiling on water have been presented. Topics discussed have ranged from the most basic to the most advanced. It is hoped that there is something of value for everyone who would like to acquire high-resolution seismic profiles on water, regardless of their level of expertise.

Appendix A. Basic concepts of frequency-domain representation.

Continuous-time functions that are not periodic are transformed to the frequency domain by means of the Fourier Integral (Wiener, 1933). In order to determine whether a function is periodic or not, it is necessary to know its values over all time. This is never possible in practice, of course, and continuous functions which are known over only a finite interval are assumed to be periodic, one period being the length of the known interval. Except in fortunate cases where this assumption is valid, it is always a source of error. For periodic continuous-time functions, the Fourier Integral may be replaced by the Fourier Series.

Transformations in continous time are mainly conceptual exercises used for visualizing theoretically correct results. It is discrete, or digitized, functions that are most often encountered in practice. A discrete function is a sequence of values, or digital samples, located at particular points of the interval over which the function is known. Discrete functions are transformed by the Discrete Fourier Transform (DFT) which is closely related to the Fourier Series and, like it, assumes periodicity at the length of the known interval.

Whether continuous or discrete, the set of values in the time domain is transformed to a set of values in the frequency domain, said to be «frequency components» of the time function. Each component corresponds to a point in a plane called the «complex plane». Collectively, these points are called the «complex Fourier spectrum» of the signal. The location of each point is specified by a pair of numbers. In rectangular coordinates, one number is called the «real» part of the component and the other the «imaginary» part as shown in fig. A1.

It is also common to use polar coordinates to locate the points. The distance from the origin, called the «amplitude» of the component (fig. A1), is related to the real and imaginary parts by

$$(\text{amplitude})^2 = (\text{real})^2 + (\text{imaginary})^2.$$

The square of the amplitude is called the «power» of the component. The counterclockwise angle from the real axis to the line passing through the origin and the point is called the «phase» of the component (fig. A1) and is related to the real and imaginary parts by the trigonometric inverse tangent function, *i.e.*

$$\text{phase} = \text{arctangent}\ (\text{imaginary}/\text{real}).$$

These constitute the amplitude, power and phase spectra of the signal.

If the time samples are spaced at equal intervals and the number of them is not prime, computational effort may be reduced by using the Fast Fourier Transform (FFT). If the number of samples is equal to a power of 2, transformation may be done using a particularly efficient algorithm called the radix 2 FFT. It outputs «real»,

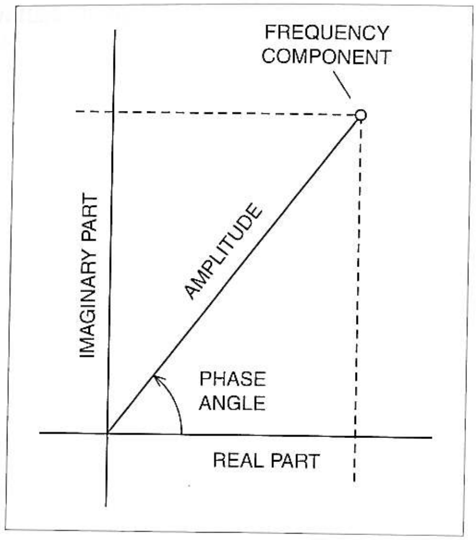


Fig. A1. The location of a frequency component in the complex plane illustrating real/imaginary rectangular coordinates and amplitude/phase polar coordinates.

«imaginary» pairs equal in number to the input samples. Output pairs are equally spaced from zero frequency to one interval less than twice the Nyquist frequency. (By the assumed periodicity, the pair corresponding to twice Nyquist is the same as the pair corresponding to zero). For real-valued input samples, the real parts are symmetric about Nyquist and the imaginary parts antisymmetric about it. Similarly, amplitude and power spectra are symmetric and phase spectra antisymmetric. The DFT and FFT are discussed extensively by Rabiner and Gold (1975).

Appendix B. What is oversampling?

The basic role of oversampling is to move the filtering requirements of A/D or D/A converters from the analog to the digital domain, where they can be more efficient (lower noise, flatter frequency response, no phase shift, etc.). It all comes down to the basic Nyquist theorem: when you are sampling (A/D), you cannot have any frequencies above half of your sampling rate, or else you get alaising (hence the term, anti-aliasing filters). In older A/D designs, an analog «brick-wall» (very steep) filter would be placed before the converter to keep all frequencies above 20 kHz (in most cases) out of the A/D converter. In an oversampling A/D converter, the sampling rate is much higher (64 times, for example). This means that the Nyquist frequency is now also higher, which means that the analog filter can have a much more gentle slope. If in a 48 kHz sampling system the A/D is a 64 times oversampling part, then the effective Nyquist frequency is 1.536 mHz. Since we do not care about anything above 20 kHz, the analog filter can start gently rolling off at 20 kHz, and be cutting significantly at 1.536 mHz (since it is over six octaves away). The A/D converter then has a digital filter that removes all frequencies above 20 kHz and reduces the sample rate (decimates) back down to 48 kHz. This is a somewhat simplified version, but hopefully you get the idea. A similar idea is used for oversampling D/As, but in reverse. The sample rate is increased by some multiple, and a digital filter removes everything above 20 kHz. The analog filter then only has to remove frequencies above the Nyquist of the new sample rate, simplifying its design.

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