Proposals for the collection of multifrequency acoustic data

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Abstract:

Acoustic surveys are used to estimate the abundance and distribution of many fish species, and have been based traditionally on data collected at a single acoustic frequency. Although it has been known for some time that the use of additional frequencies can provide information on the nature of the acoustic target, the knowledge and technology required to combine the so-called “multifrequency data” in an appropriate manner has been limited. The use of several transducers of different frequencies is now common on board research vessels and fishing vessels, so multifrequency data are often collected. In order for these data to be combined appropriately, their physical and spatial characteristics from each frequency should be as similar as possible. We detail the requirements deemed necessary to collect multifrequency data in an appropriate manner. They can be stringent and may not always be achievable, so we also consider the consequences of combining acoustic data originating in transducers with varying degrees of spatial separation and with different beam widths.

Keywords: data collection, multifrequency acoustics, species identification
Introduction
Acoustic surveys are used extensively throughout the world to determine the abundance and
distribution of various types of marine and freshwater fauna and flora (Simmonds and MacLennan,
2005). Abundance is derived from density measurements that are the result of echo integration
(MacLennan 1990) from a single acoustic operating frequency. Although data from more than one
frequency were used by Cushing and Richardson (1955) to infer differences in scattering by frequency
from different fish species, it was only in the 1970s that multifrequency data were used in earnest to
identify scattering from various zooplankton size groups (Holli day, 1977; Greenlaw, 1979). This last
work formed the basis of much of the effort to quantify and identify zooplankton in the 1980s
(Greenlaw and Johnson 1983; Pieper and Holli day 1990), and coincided with concerted efforts to
understand the theoretical basis for scattering by such groups, with the development of scattering
models (e.g. Stanton 1989). These in turn led to practical techniques allowing for the discrimination of
larger taxa in acoustic-survey data, such as krill (Madureira et al., 1993) and, more recently, fish
(Kang et al., 2002; Korneliussen and Ona, 2002, 2003). Multifrequency acoustic data, therefore, have
the advantage of providing information on the nature of the [acoustic] target of interest such that some
discrimination by acoustic means may be possible (see Horne, 2000, for a review).

In most cases, acoustic-survey data are collected in a manner that is optimized for a single
frequency, and less consideration is given to combining acoustic data from multiple frequencies.
Although data manipulation and processing can allow data from many single frequencies to be
combined, e.g. compensating for different transducers spaced apart along the ship by shifting the
spatial reference of pings from one frequency to be aligned with the pings from another, optimal
multifrequency data cannot be achieved from a system if the input data are not collected properly.
Here, we propose methods for the collection of multifrequency acoustic data in a manner that is most
appropriate for subsequent analysis.

Multifrequency data may be collected in various ways on board research and fishing vessels. They
may be collected as if they were single-frequency data, i.e. with no intention of combining the
different frequencies, or they may be collected with the explicit intention of combining all frequencies.
Here, a common term for both these types of raw data is multiple single-frequency data. For these data
to be analysed appropriately, the physical and spatial characteristics of acoustic data should be as
similar as possible. Although direct comparability of data at different frequencies is impossible in all
respects, ideally data should be as well suited as possible to allow for a combination of frequencies at
a high spatial and/or temporal resolution. Acoustic data from several single frequencies are defined as
ideal in this context if they can be used to generate combined frequency data at the same resolution as
any one of the original single frequencies. The term “combined frequency data” refers to new artificial
data generated from several of the original single-acoustic frequencies. This requires comparable
physical measurements, carried out simultaneously from identical sampled volumes, limited only by the
effective range of the highest frequencies.

It is desirable to keep the spatial resolution of acoustic data as high as possible in order to resolve
scatterers, but it is also desirable to reduce acoustic variability to categorize the acoustic returns
precisely. These two requirements are contradictory, because the averaging used to reduce the
variability inherently also reduces the spatial resolution. Acoustic scattering has a stochastic nature, so
there is a need to average (e.g. via smoothing) many acoustic measurements. Smoothing inherently
reduces the spatial resolution of the acoustic measurements. Some of the natural stochastic variation is
reduced by the use of echosounders capable of rapid pinging and rapid sampling, by averaging
samples from the same small elementary volume, but still there may be some stochastic variations
attributable to radiation patterns, tilting, and the distribution of the scatterers in the measurement
volume. As it is not clear how much averaging is needed to remove the stochastic variation of the
measurements, it is reasonable, initially at least, to collect combined- frequency acoustic data at as
high a resolution as possible.

Several recommendations are made and examined here under two major headings specifying the
requirements of making data comparable: (i) physically, and (ii) spatially. Each proposal is numbered
sequentially across these two headings, and these are finally summarized as a prioritized set. In
practice, several of these proposals may not be achievable using current systems. When working with

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hull-mounted transducers on research or fishing vessels, it is particularly difficult to obtain spatially comparable data. Different transducers are often mounted separately on the hull and may be several metres apart, so that the ideal case of co-locating all transducers at the same point is far from being fulfilled. Transducer size, beam width, and selectable pulse duration are generally optimized for target detection at each frequency, rather than for a combined analysis. Following the proposals, we examine the errors in echogram processing when data are collected from transducers spaced apart, or from transducers with different beam widths. Data from such equipment are termed “compromised” multifrequency data.

Many of the example settings are given with reference to Simrad echosounders and acoustic transducers, because these are currently the most commonly used instruments in marine fisheries acoustics. However, these are by no means the only instruments available, and operators wishing to develop along the lines we propose should approach manufacturers of their particular devices to obtain analogous settings where appropriate.

Requirements to make data physically comparable
Measurements of acoustic scatter at one frequency from fish or plankton should be comparable between equipment made by any manufacturer, provided the measurements are also spatially comparable. Physical measurements refer to echosounder outputs such as $s_v$, $s_A$, target strength (TS) (and similar), and measurements that are used to calculate these, e.g. signal voltage, absorption, and sound speed. Absorption and sound speed are both components of range-dependent amplification used to calculate $s_v$, $s_A$, and TS. Absorption affects the physical measurements, whereas sound speed affects mainly spatial comparability, because it is seldom so wrong that it leads to significantly erroneous values of $s_v$, $s_A$, and TS. The range-dependent amplification is, therefore, treated under both the first and the final requirement.

Echosounder systems should be operated such that the linear-wave equations apply
Most theory within fisheries acoustics is based on linear-wave equations, which is typified by the use of range compensation in fisheries acoustics, commonly known as time-varied gain (TVG) (MacLennan, 1990; Simmonds and MacLennan, 2005). However, although non-linear acoustic interactions are always present, they are much reduced compared with linear sound, particularly when the acoustic power output from the transducers is reduced. These non-linear interactions take place in the insonified water-column and depend on the acoustic intensity and the acoustic frequency (Pedersen, 2006; Tichy et al., 2003). In order to reduce the effect of non-linear interactions, the power output from a transducer should be selected at a level where the non-linearly generated sound is negligible compared with linearly generated sound. In order to achieve this in, for example, the case of an echosounder using a Simrad E838D transducer (38 kHz in Table 1), 15 kW m$^{-2}$ or less output power is sufficient: for 60% transducer efficiency, this gives 25 kW m$^2$ or less input power, which is obtained by setting the maximum input power on the echosounder to 2500 W. Maximum input power settings for other Simrad transducers are given in Table 1. Note that the higher the frequency, the lower the input power needs to be to avoid generation of non-linear effects.

In the examples given in Table 1, most transducers weight the power across the transducer face to reduce side-lobe levels, i.e. to ensure that the power enforced on the outer elements is less than that at the central elements. The use of too high a power level for the transmission of sound leads to significant generation of sound at higher frequencies, through the non-linear effects. This will appear in the transmitted sound at the original frequency as loss in the signal in addition to the losses expected from absorption and geometric loss. Some of this loss is compensated for during the calibration process (Pedersen, 2006), but generally it is advisable to reduce the power to levels indicated in Table 1.

All echosounder and transducer systems must be calibrated
Foote (1982, 1989) and Foote et al., (1987) described the generally accepted method of calibrating echosounders. The total error in the calibration method should be no more than 4%, i.e. the uncertainty in the measurements of $s_A$ (nautical-area-scattering coefficient) attributable to calibration is approximately 4% (see Table 7 of Foote et al., 1987, or Havforskningsinstituttet, 1994). The
In general, noise is all the unwanted signals, including transmitted sound backscattered from wind-generated bubbles. It is, however, difficult to separate free bubbles from swimbladders in small fish or...
bubbles generated for buoyancy by some types of plankton. A proper definition of noise is needed before developing a model to remove it. The definition of noise according to Korneliussen (2000) is “if the intended signal is defined as all transmitted sound backscattered onto the transducer surface, then noise is everything else.” Sound generated by ships, animals, collapsing bubbles, wind, or sea are noise in this case, as is instrument noise not associated with the transmission of sound. Under this definition, backscattered sound caused by unwanted electrical signals in the transmit part of the echosounder is not noise, nor is sound backscattered from bubbles. When acoustic data are corrected for noise, and the noise is uncorrelated with the backscattered signal from the targets, the maximum range of the acoustic data is limited by the sampling volume of the beam (Ona, 1987; Foote, 1991). The acoustic-sampling volume is the volume where all targets of interest, at all orientations, are acoustically visible in all parts of the sampled volume for the ranges used: it is species- and density-dependent. Foote (1991) described the statistical properties of the sampling volume.

Measurements should not be biased by noise

To be able to quantify and remove noise, the noise and intended signal should not be correlated. Noise can be quantified and removed from the measured signal using the methods described by Korneliussen (2000), which requires the collection of passive acoustic data, or by Nunnallee (1987). These methods require that the echosounder does not truncate measurements below a threshold, and that noise is not removed automatically by an internal algorithm. In the case of the Simrad EK500 the “noise margin” should be set to 0 dB; fortunately, the new Simrad EK60 has no noise-removal feature to worry about.

In the absence of passive acoustic data, the data needed to quantify noise may be selected according to the scheme suggested by Korneliussen (2004). Although backscatter from bubbles is not noise according to the definition above, bubbles associated with the wash from the hull of a vessel are undoubtedly unwanted components of the backscatter within fisheries acoustics. It is, therefore, suggested that acoustic transducers be mounted on the bottom of a protruding instrument keel (Ona and Traynor, 1990), such that they can be lowered below the bubble layer to reduce unwanted backscatter from bubbles created from the wash of a ship’s hull.

Noise should not reduce the acoustic sampling volume

This requirement is to ensure that noise does not influence the spatial comparability of the acoustic data. The TVG function compensates the acoustic measurements for range, and the calculations are based on a detection area A at range R. If noise exceeds the detection threshold, the area where the echosounder can detect targets is less than A, so the sampling volume is reduced. In general, the distance from the transducer at which data are considered valid should be reduced, rather than trying to correct data collected from a reduced sampling volume (see below). The data, therefore, should not be used beyond a range R at which noise starts significantly limiting the sampling volume of any target of interest (Ona, 1987; Foote, 1991).

The range R is where the sampling volume V starts being reduced, e.g. where volume backscatter is no longer proportional to R². Strictly speaking, the measured data can be corrected if the reduction of the sampling volume between ranges R and (R + ΔR) is known. Note, however, that the range R where the sampling volume V starts being reduced obviously depends on the TS. Therefore, if a measured volume contains different species, and/or different sizes of each species, each of those would need their own correction function. The compromise is to use a common range, R, for all targets. In rare cases it may be possible to estimate functions at each frequency to correct for the reduction in the acoustic sampling volume. This would be the case for acoustic data collected where there was only one target of interest, of essentially one size, e.g. Norwegian spring-spawning herring 33 cm long in Ofotfjord during most winters between 1988 and 2006.

Interference between frequencies should be insignificant

If the echosounder system, i.e. the echosounder electronics, acoustic transducers, and connection cables, at any single frequency is interfered with by a system operating at another frequency, the signal and noise are correlated, such that the algorithms known to remove noise cannot be used. The interference can be checked but, ultimately, a solution must be provided by the echosounder manufacturers to avoid interference between frequencies, e.g. by offering appropriate selection of
acoustic frequency, bandwidth, and transducer input power. Further, the electronics used at one frequency should not interfere with the electronics used at another frequency, but this is usually not a problem. A narrow bandwidth in the system will reduce the problem of acoustic interference, but will exacerbate other problems related to the pulse envelope and total-system delay. Measurements to date indicate that interference between echosounder systems is a minor problem, at least in the measurements of backscatter. However, strong targets may be detectable at a frequency, e.g. 18 kHz, for a system running in passive mode if there is an active system running at a frequency close by, e.g., 38 kHz. Note also that in the case of moderate to strong non-linear generation of sound at frequency $f_0$, there will be unwanted sound-components that will interfere with systems at frequencies $2f_0$ and $3f_0$ (harmonics).

The choice of frequencies should, therefore, be sufficiently different so as to avoid mutual interference. Moreover, care must be taken to avoid choices which are harmonics of each other (e.g. 200 and 400 kHz), because of the non-linear generation of sound. When 200 and 400 kHz transducers were used together on board FRV “G. O. Sars” (IMR vessel 3), the second harmonic of 200 kHz ($2 \times 200 \text{ kHz} = 400 \text{ kHz}$) generated so much sound, even with the input power recommended in Table 1, that the 400 kHz system could not be used. The latter is now being replaced with 333 kHz. Frequencies of odd multiples ($3, 5, 7, \ldots$) should also be avoided because of the linear generation of sound. The frequency sequence (in kHz) $18; 38; 70; 120; 200; 333; 555; 926; 1543; 2572; 120 \times 1.6667^n$ ($n = 7, \ldots$); is one of many possible options. This sequence gives a reasonable resolution for small targets, e.g. small zooplankton. The factor 1.6667 from 120 kHz is a convenient choice to select the next frequency, although there is nothing special about that number.

Requirements to make data spatially comparable

Figure 1 illustrates the problem associated with horizontal and vertical spatial overlap. Considering a cone as a simplified beam, two such beams of equal beam width $\theta$ irradiate two partly overlapping discs of equal size. At range $R$ from the transducers, the fraction horizontal overlap ($O_h$) of two beams with beam width $\theta$ is:

$$O_h = \left(\frac{2\alpha}{\pi}\right) - \frac{a \sin(\alpha)}{R \pi \tan(\frac{\theta}{2})},$$

where $\alpha = \cos^{-1}\left(\frac{d}{2R \tan(\frac{\theta}{2})}\right)$, $d$ is the separation distance between the transducer centres (m), $\theta$ the 3 dB beam width of the beams (rad), and $R$ is the distance (range) from the transducer face (m).

Figure 2 shows $O_h$ as a function of $R$ for beams of width $7^\circ$. Note that $O_h$ is not a measure of horizontal overlap between beams of different widths, because it is obviously meaningless to calculate mutual overlap for beams of different width. Only 56.6% of the backscatter measured within $11^\circ$ is, on average, also within $7^\circ$ of the same beam generated from a transducer radiating as a perfect circular piston, at all ranges. This is calculated from the two-way Bessel directivity functions of intensity multiplied by the ensonified area. For real beams, the level of the side lobes is less than the Bessel directivity, so 60% within $7^\circ$ may be a better estimate for real beams than 56.6%.

The fraction-vertical overlap ($O_v$) for pulses of equal duration and shape between data collected at two acoustic frequencies, with similar beam width, is defined as

$$O_v \equiv [1 - \text{abs}(\Delta v_1 - \Delta v_2) / \Delta z],$$

where $\Delta v_1$ and $\Delta v_2$ are the vertical offset distances attributable to total-system delays, and $\Delta z$ is the vertical resolution.
$O_v$ is increased either if $(\Delta v_1 - \Delta v_2)$ is decreased or if the vertical resolution is decreased by increasing $\Delta z$. $O_v$ can be improved if data are collected at a sufficiently high resolution provided the 3 dB beam widths are the same. Echosounder-pulse envelopes differ from an ideal square pulse, especially for narrow bandwidth and wider beams at low frequencies, e.g. 18 kHz. This makes the result of the vertical shifting of data at 18 kHz more uncertain. The correlation of vertically shifted data at 18 kHz relative to any of the other frequencies does not provide a significant improvement for sample data tested.

The fraction-spatial overlap ($O_s$) between the beams at different frequencies is defined as:

$$O_s = O_v O_h$$  \hfill (3)

There is no strict requirement with respect to the overlap required for the generation of combined frequency echograms, but a $O_s \geq 0.85$ seems reasonable. In the case of the 38 and 120 kHz transducers on the FRV “G. O. Sars”, as shown in Figure 3a (IMR vessel 2), where the transducers’ centres are 39.5 cm apart, $O_s$ of 0.85 is achieved at 28 m. For the 38 and 200 kHz transducers spaced 67.5 cm apart, $O_s = 0.85$ at 47 m. For methods involving division or multiplication of data at two frequencies, $O_s = 0.85$ gives an uncertainty of about 15% in the result, in addition to the measurement uncertainty.

$O_s$ can never be better than $O_h$. The transducer configuration for the newer vessel, FRV “G. O. Sars” (Figure 3b; IMR vessel 3), was designed specifically to enhance spatial overlap of the beams, and shows a significant improvement on the configuration in the previous vessel: an $O_s$ of 0.85 as calculated by equations (1) and (3) is achieved 13–34 m below the transducers, depending on which beams are compared.

Further considerations of non-ideal situations, where data do not overlap because of either transducer spacing or different beam widths, are considered later.

**Pulse lengths and pulse shapes should be identical at all frequencies**

The nominal pulse lengths become equal when the pulse durations are equal at all frequencies. Equal nominal pulse durations at all frequencies are, therefore, a necessary requirement. The requirement of equal pulse shape also requires equal bandwidth in the system, which is more difficult to achieve.

A pulse duration of 1.0 ms is sufficient for the pulse envelope to stabilize in 18 kHz echosounder systems with common bandwidths. Such a pulse duration should, therefore, be used across all frequencies; shorter pulse durations would be sufficient if 18 kHz data are not used. Older equipment may require manufacturer modifications: in the case of the Simrad EK500, the same adaptation (PROM) that delivers 2-cm samples across all frequencies can be configured to deliver 1.0-ms pulse durations. If the special EK500 PROM is not available, a short pulse duration should be used for 12–27 kHz, medium for 38–70 kHz, and long for 120 kHz and above. In the case of the new Simrad EK60, it is possible to set the pulse duration to 1.0 ms for all frequencies. Wide bandwidth, 10% of centre frequency, is recommended for the Simrad EK500 for 70 kHz and below, and narrow bandwidth, 1% of centre frequency, for 120 kHz and above. The bandwidths for the EK60 are calculated by the system. When using 1-ms CW pulses, these are: 1.6 kHz at 18 kHz, 2.4 kHz at 38 kHz, 2.9 kHz at 70 kHz, 3.0 kHz at 120 kHz, 3.0 kHz at 200 kHz, and 3.1 kHz at 364 kHz.

**Individual pings should be identifiable in the data files at all times**

This requirement ensures that simultaneous pings of different frequency can be identified and compared. It is insufficient to count pings in a datafile, because pings are occasionally lost, and the ping rate may be different when several echosounders are used simultaneously. Time should be registered when the echosounder is triggered to transmit, and should be stored with a resolution sufficiently high to avoid two pings at the same frequency being registered at the same time. A time resolution of 0.01 s is sufficient for such purposes; a resolution of 1.0 s is not.

**Acoustic sampling volumes should be similar at all frequencies for comparable ranges to the scatterers**

Targets of interest should be visible acoustically in all parts of the sampled volume for the ranges used (Foote, 1991). Provided there is insignificant noise, this implies similar half-power beam widths and
that all transducers should have the same centre, including identical transducer depth, and the same
acoustic axis for the transducers.

This point is to achieve maximum horizontal overlap between the beams, but this is generally not
possible. At best, transducers with similar beam widths should be mounted at the same depth and with
the same acoustic-axis orientation. The smallest transducers should be placed in the middle in order to
reduce the average distance between them. Standardizing the 3-dB beam widths to approximately 7° is
a reasonable compromise between long range and wide beam width, to cover a large volume. For
commercial low-frequency transducers, for example 18 kHz, generated beams of 11° may be the
smallest achievable, and hence closest to 7°. The transducer faces should be adjusted to give the same
orientation of the acoustic axis of all transducers if this cannot be done electronically: the acoustic axis
is expected to be very close to a vertical straight line. Horizontal distance between the transducers will
result in errors that are discussed further below. The effect of the horizontal offsets is reduced with
increasing range from the transducers because of the conical shape of the beams. Therefore, the
fraction-horizontal overlap (\(O_h\)) increases with depth.

**Transmission of pulses should be simultaneous at all frequencies**

In order to sample as similar a volume as possible, pulses from individual transducers (frequencies)
should be transmitted at exactly the same time and incorporate appropriate system delays (to achieve
maximum vertical overlap). For equal bandwidth in the systems, there will be no differences in the
total-system delays (see above). However, in practice, the systems at different frequencies will have
different bandwidths, and therefore, also different total-system delays, which have to be compensated
for in some way. Total-system filtering causes vertical offsets. Increasing the difference in total-
system delay increases the \(O_v\), and a reduction in vertical resolution reduces it. If the data samples are
collected with a sufficiently high vertical resolution and the vertical shift is known, the samples can
simply be shifted vertically. If data are not collected with a sufficiently high resolution, the effect
could be reduced somewhat by smoothing the data with weights shifted vertically.

**Synchronization**

Pulse transmission is properly synchronized within each Simrad EK500 echosounder, which can
accommodate a maximum of three frequencies. When operating more than one EK500, the slowest of
the utilized sounders, the EK500a, should trigger the fastest, the EK500b, or preferably they should be
connected to an external trigger unit. In the latter case, it may be difficult to use the common bottom
depth-dependent ping-rate.

Using an external time source as input to all EK500s will synchronize time, but if data are logged
with time in all EK500s synchronized continuously, e.g. by GPS time, experience has shown some
strange side effects, such as the wrong time or a time-jump on one of the echosounders. This effect is
avoided by manually setting the time once to, say, GPS time, then switching back to the now corrected
internal EK500 clock. Setting the EK500 to satellite time is done by setting the parameter “\(/UTILITY
\)MENU/External Clock=Serial” to set the time, then “\(/UTILITY MENU/External Clock=Off”.

In the Simrad EK60, which can incorporate up to seven transducers (frequencies) simultaneously,
pulse transmission is properly synchronized for all transducers (frequencies).

**System delays should be incorporated**

Calculation of the delays from the echosounder’s internal-trigger pulse are straightforward provided
the electronic characteristics of components of the system are known. It is not recommended to
compensate for the total-system delay until theoretical delays are verified by measurements.

Ona et al. (1996) measured the delays with a standard version of the Simrad EK500 software.
Measurements and calculations were consistent, and showed that the total-system delay (in s) at
frequency f (Hz) when non-composite\(^1\) transducers were used were:

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\(^1\) The term “composite transducer” refers to the way the acoustic transducer is designed. The ceramic is cut
into several thin rods, e.g. 2 mm × 2 mm cross-section, where the length of the rods defines the main acoustic-
resonance frequency. Several rods are glued together in a regular pattern. By doing so, most of the unwanted
resonance modes cannot be excited, while the main resonance can still be used. The removal of unwanted modes
increases the frequencies where the transducer can be used. The use of glue between the rods also increases the
Total-system delay for wide bandwidth (10% of the centre frequency): 14.8/f (s);
Total-system delay for narrow bandwidth (1% of the centre frequency): 44.6/f (s).

The vertical shift for wide bandwidth is then close to 1480(14.8/f)/2 (m) for the EK500, which for 38 kHz is 29 cm. In the expression, 1480 m s\(^{-1}\) is the sound speed, and the division by 2 is to account for two-way transmission. Depths associated with the measurements of MVBS in the Simrad EK500 are not corrected in the echosounder-output data.

Similar calculations for the total-system delay have also been done for the Simrad EK60 (H. Nes, Kongsberg AS, pers. comm.), but calculations have not been as well verified as those for the EK500. The system delays attributable to the digital filters in the EK60 are zero. The theoretical total-system delay at the frequency f (Hz) for the Simrad EK60 consists of the delay of the hardware and of the transducer:

- System delay for Simrad EK60 hardware (GPT): 4.5/f (s);
- Delay attributable to non-composite transducer (Q-factor = 4): 2.5/f (s);
- Delay attributable to composite transducer (Q-factor = 2.5): 1.5/f (s)

Currently, the Simrad transducers at 70 kHz and above are composite. These give a vertical shift for EK60 close to 1480(7.0/f)/2 (m), which for 38 kHz is ~14 cm. Table 2 shows the vertical offsets attributable to system delays in the EK500 and the EK60 for common settings and transducers, as calculated from the formulae above at different frequencies relative to 38 kHz.

Correct sound speed and absorption coefficient should be applied

Sound speed and acoustic absorption can be calculated from the formulae of Francois and Garrison (1982a, b). Both the sound speed and the absorption change with changing salinity, temperature, and depth (pressure). When salinity, temperature, and depth are known, the formulae give an accurate value for sound speed. Therefore, it would be necessary to calculate new values continuously for sound speed and absorption to obtain the correct acoustic measurements. This in turn raises the possibility of erroneously inserting inappropriate values of sound speed and absorption into the echosounder.

The suggested solution is as follows: a conductivity–temperature–depth (CTD) probe needs to be employed in the survey area at the beginning of each survey to provide the required data. It is probably sufficient to use the same sound-speed profile and the same frequency-dependent absorption throughout the whole survey, and probably also good enough to use the same sound speed throughout the water column. If a depth-specific sound speed is used, the CTD profile used to calculate the sound-speed profile should follow the acoustic data. The CTD profile used to calculate sound speed and absorption should, naturally, be taken in the survey area. It is, for example, poor practice to use the CTD profile taken at the calibration site to calculate sound speed and absorption in the survey area if the two areas are far apart or different oceanographically.

Compromised data – use of data that are not comparable in all respects

The ideal situation where two or more transducer axes overlap perfectly or have exactly the same beam width is virtually impossible to fulfil. What follows, therefore, is a discussion on studies conducted to estimate the error in multifrequency analyses when the data have been compromised with regard to beam overlap: specifically, studies were carried out for transducers installed on IFREMER’s FRV “Thalassa” from 12 kHz to 200 kHz, with beam widths from 7° to 16° and distances between transducer bandwidth, although as a positive side-effect of the transducer design. A non-composite transducer is composed of one or a few elements, where the length of the elements defines the main acoustic-resonance frequency. The cross-section of the elements could be, for example, circular with diameter 80 mm, which makes unwanted resonances more likely than with composite transducers.
transducer centres ranging from 0.4 m to 2 m. Transducers with axes spaced some distance apart and
with different beam widths are likely to provide data with the most common compromised condition.
In common with the new “G. O. Sars”, the transducers of FRV “Thalassa” have been rearranged to
have them as close as possible. All other requirements (identical pulse length and shape, simultaneous
pulse transmission, etc.) are assumed to be fulfilled. Further, what follows concerns only echotraces
obtained from large multiple targets such as fish schools, not single echoes.

If we define a $O_s$ of 0.85 as a reasonable value for comparing data, this definition relies on the
simple 3 dB definition of beam widths and does not cover all echoes contributing to the signal in the
sample volume at a certain threshold. The proportion of the beam truly occupied by the school during
the process of detection as it passes through the acoustic beam is not taken into account.

As the global process of school detection and generation of an echotrace is quite complex,
instrument error was estimated approximately using simulations of echotraces of fish schools (see
Diner, 2001). The simulations presented here rely on a simple backscattering model of fish, which has
been manipulated according to an updated method described in Diner (2007). Simulations were
conducted using several schools of different dimension and MVBS, at various depths, detected by
different commonly used nominal beam widths: 7°, 11° and 16°. The images obtained were
processed with different thresholds. In these studies two main analyses can be performed on the school
echotraces:

1. **Global MVBS.** In this case, the echogram from each frequency is processed separately and
   comparisons are made on whole echotraces at each frequency (i.e. average MVBS), extracted from
each school. Based on simulations of two identical transducer directivities at different locations,
the instrument error is estimated as the difference between the MVBS values calculated for the
echotraces of the same school detected by two different beams of the same frequency.

2. **Ping-to-ping.** In this case, the data from two channels are combined on a ping-to-ping basis to
generate a new synthetic or virtual echogram. Echotrace descriptors are then extracted from the
virtual echograms. The instrument error is equivalent to the mean difference in VBSs from the
school between the two beams at the same frequency.

Potential instrument errors are induced by the athwartship or alongship distance between
transducers, and by differences in beam width. The errors do not affect all types of multifrequency
analyses:

- the "global MVBS' is affected by the athwartship distance and the difference in beam width;
- the precision of the ping-to-ping analysis depends also on the alongship distance.

When considering the variation in school length and depth, and the various transducer positions
and beam widths that are possible, a large number of permutations are possible, and they are quite
complex to summarize. To illustrate the potential errors, the simulations were based on the detection
of geometrically homogeneous ellipsoid schools through the directivity function of an acoustic beam.
This complex process can be simplified by normalizing the results according to a realistic geometric
hypothesis of the dependence of the error on the derived parameter $N_{bi}$, the dimension of the echotrace
relative to the beam width (Figure 4):

$$N_{bi} = \frac{L_i}{2D_i \tan(B_i/2)}$$

where $B_i = 0.44 \times \theta \times (dST_i)^{0.45}$ as a valid approximation of the inverse Bessel function for directivity
in the considered part of the beam, $\theta$ is the nominal transducer beam width (7° for example), $dST$ is the
difference between the MVBS of the school echotrace ($S_{vi}$) and the threshold, and $L_i$ and $D_i$ are the
length and mean depth, respectively, of the school echotrace. Note that the beam width is estimated
using the detection angle, $B_i$, not the nominal angle of the transducer (Diner, 2001).
**Athwartship distance errors**

In multifrequency analyses, the athwartship distance between transducers could have an effect on the results obtained because, for example, small schools could occupy the entire part of one beam and a smaller part of another (Figure 5). During the school-detection process with a single beam, the width of the schools detected is unknown. For convenience, therefore, it was assumed that school width and length were equivalent. The fact that only a part of the beam is occupied and that only the edge of the target is detected, causes cumulative attenuation effects resulting in, for instance, a drastic reduction in the MVBS of the school detected through this beam. The phenomenon is quite complex because the whole process of detecting the school must be considered, i.e. all successive school echoes from the start to the end of detection, in addition to considering that the school is not on the beam’s central axis.

Simulations were carried out for four athwartship separation distances: 0.40, 0.70 m, 1.0 m, and 2.0 m. For an athwartship transducer distance of 0.40 m, instrument errors for MVBS remain low, mostly <0.25 dB regardless of $d_{ST}$, school depth, or relative size of the school. Therefore, this is a good value to aim for when installing transducers. Differences in MVBS from the simulations with different athwartship offsets allowed for the definition of empirical minimum $N_{bi}$ values under the hypothesis that they are related to a given acceptable error in MVBS ($E$), school depth ($D_i$), and $d_{ST}$ factor. For an athwartship distance $e_n$ (in m):

$$e_{0.70}: N_{bi} \geq \frac{20}{D_i} + \frac{8}{E_{0.05}} - 0.03d_{ST} - 6.8$$

$$e_{1.0}: N_{bi} \geq \frac{20}{D_i} + \frac{8}{E_{0.05}} - 0.03d_{ST} - 6.6$$

$$e_{2.0}: N_{bi} \geq \frac{20}{D_i} + \frac{8}{E_{0.05}} - 0.05d_{ST} - 6.3$$

These relations are approximations of the influence of depth of the school and the threshold, and allow for rapid selection of schools, which can be processed with a potential instrument error that does not disrupt further multifrequency analyses. Figure 6 gives a general representation of these limit $N_{bi}$ values. At shallow depths, e.g. 15 m, with a $d_{ST}$ of 10 dB, the $N_{bi}$ limits are high, especially for large athwartship distances. Generally, the $N_{bi}$ limit decreases when $d_{ST}$ increases, i.e. when the processing threshold decreases.

**Alongship distance errors**

When transducers are spaced apart longitudinally by several metres, instrument errors are induced at the start of detection, because one beam detects a school some pings before another and, conversely, loses the school before the other, at the end of detection (Figure 7). If the transducers are not too far apart (<6 m, for example), the echotraces of the same school, detected by the two beams are, in most cases, similar, and the global MVBS is not subject to a large instrument error. This is not the case for ping-to-ping analysis. During the phases of the start or end of school detection, there is a regular variation of the level of the received signal from one ping to another. This signal level, related to the proportion of the beam width occupied by the fish, increases until the beam is fully occupied. It then remains constant for some pings, the centre of the school or school kernel, and finally decreases as the proportion of the occupied beam lowers, until the end of school detection (Figure 8). If a comparison between frequencies is made, e.g. MVBS$_{F1}$ – MVBS$_{F2}$, which is equivalent to the ratio of echo intensities $I_{F1}/I_{F2}$, the ratio would be high towards the end of detecting the school because $I_{F2}$ is lower than $I_{F1}$, and vice versa at the start of detection.

In order to determine the extent of this phenomenon, different school sizes, depths, and MVBS were simulated. Errors attributable to alongship separation distances of 0.5, 1.0, 1.5 and 2.0 m were calculated (with a ping interval of 0.5 m). For an alongship separation distance of 0.5 m and a $d_{ST}$ value of 5 dB, the instrument errors were <0.5 dB whatever the school size or depth. For an alongship separation distance of 2.0 m and a $d_{ST}$ of 5 dB, the school depth must be >25 m in order to reduce the
instrument error to an acceptable level. With a $dST$ of 10 dB or more, the errors remain high whatever
the school size or depth.

One solution to this problem is to compensate for this alongship distance in term of ping numbers,
i.e. to shift the frequency analysis by a number of pings equivalent to the distance. However, when
vessel speed is fast (e.g. 10 knots) and ping rate low (1 ping s$^{-1}$), the distance between pings can be
greater than the transducer alongship distance (~5 m in the example cited). Therefore, such a ping shift
is unable to compensate for the separation distance. High ping rates and/or reduced vessel speed would
give better results.

**Errors of beam width**

The underestimation of a school’s MVBS is related to different parameters, but critically to the
relative school and beam-width dimensions (Diner, 2001). Using different beam widths leads to the
underestimation of various parameters, resulting in errors in the multifrequency analysis in the global
MVBS approach. Shifted detection as a consequence of different beam widths generates problems
analogous to the alongship separation of transducers in the case of a ping-to-ping analysis. In order to
investigate this problem, simulations of school detection were carried out at different depths using four
different beam widths: $7^\circ$, $8^\circ$, $11^\circ$, and $16^\circ$. In each case, the difference between the echotrace MVBS
was calculated for the same school detected using two beam angles, $\theta_1$ and $\theta_2$: $\Delta$MVBS($\theta_1$, $\theta_2$) =

$$[\text{MVBS}_{\theta_1} - \text{MVBS}_{\theta_2}].$$

When a school is detected by a vertical beam, its MVBS is systematically underestimated. This
underestimate increases as the horizontal dimensions of the school become smaller relative to the
beam width (i.e. low $N_{bi}$ values). When a school is detected by two frequencies with the same beam
width, the two underestimates are similar and do not affect the result of the multifrequency analysis. In
the case of two different beam widths, the difference in the MVBS underestimate for the two
directivities must be determined: this difference will be equivalent to the instrument error in a
multifrequency analysis (global MVBS).

An algorithm to determine a correction in school descriptors (Diner, 2001) was used to investigate
this. The underestimate in school MVBS in relation to $N_{bi}$ is given by

$$dSV = \frac{2.56}{N_{bi} - 1}. \tag{5}$$

The difference in MVBS for two different beam widths is then

$$\Delta$MVBS($\theta_1$, $\theta_2$) = dSV$_{\theta_1} - dSV_{\theta_2} = 2.56 \left[ (N_{bi\theta_1} - 1)/(N_{bi\theta_2} - 1) \right]. \tag{6}$$

By combining the results of simulation, relationships between $N_{b7}$ and other nominal beam widths,
i.e. $N_{b8}$, $N_{b11}$, and $N_{b16}$ were determined. The difference in MVBS coefficient can then be expressed
in relation to $N_{b7}$ by adjusting the coefficients of Equation (6) as follows:

$$\Delta$MVBS($7,8$) = 2.56 $\left[ (N_{b7\theta_1} - 1)/(0.87 N_{b7\theta_2} - 0.87) \right]$$

$$\Delta$MVBS($7,11$) = 2.56 $\left[ (N_{b7\theta_1} - 1)/(0.59 N_{b7\theta_2} - 0.57) \right]$$

$$\Delta$MVBS($7,16$) = 2.56 $\left[ (N_{b7\theta_1} - 1)/(0.46 N_{b7\theta_2} - 0.41) \right]$$

The potential errors for a range of $N_{bi}$ values are given in Figure 9. In general, there are small errors
for $7^\circ$/$8^\circ$, in most cases <0.5 dB. For $7^\circ$/$11^\circ$, or worse $7^\circ$/$16^\circ$, unless the schools are very large ($N_{b7}$ >
7), large errors are obtained, hindering comparison of the data obtained with a $7^\circ$ nominal beam width
(e.g. 38, 120, or 200 kHz) and an $11^\circ$ (18 kHz) or $16^\circ$ (12 kHz) beam width.

A possible solution in such cases is to limit the analysis to data from the school kernel, the portion
of the detected school when the beams of the two frequencies are fully occupied by fish. Some pings
at the start and end of school detection should, therefore, be removed from the analysis. This number
(of pings) is calculated, taking into account the larger beam width, but using the real detection angle, $B_i$:

$$B = 0.44 \times \theta \times (dST)^{0.45}. \quad (7)$$

The relevant distance at the start and end of school detection, $L_{pg}$, is then

$$L_{pg} = 2D_t \tan(B_i/2). \quad (8)$$

If the vessel speed is $V_s$ (in m s$^{-1}$) and the ping rate $P_g$ (in s), the total number of pings to be removed is

$$n_{pg} = \frac{2D_t \tan(B_i/2)}{V_s P_g}. \quad (9)$$

Some examples of numerical applications are given below:

- $\theta = 11^\circ$, $dST = 20$ dB $\Rightarrow B_i = 18.6^\circ$,
  $D_t = 150$ m, $V_s = 5$ m s$^{-1}$, $P_g = 0.5$ s, $n_{pg} = 20$;

- $\theta = 11^\circ$, $dST = 20$ dB $\Rightarrow B_i = 18.6^\circ$,
  $D_t = 50$ m, $V_s = 2.5$ m s$^{-1}$, $P_g = 0.5$ s, $n_{pg} = 13$;

- $\theta = 11^\circ$, $dST = 10$ dB $\Rightarrow B_i = 13.6^\circ$,
  $D_t = 150$ m, $V_s = 5$ m s$^{-1}$, $P_g = 0.75$ s, $n_{pg} = 12$.

**Discussion**

Our motivation in writing this paper was to improve automatic acoustic species identification at high spatial resolution, and therefore, the analysis of echotraces from biological targets, particularly fish schools small in size. At this stage, more effort needs to be allocated to improving echosounder systems and transducer platforms where they are used for multifrequency observations. In order to limit the instrument error induced by having transducers at different locations, the number of candidate schools that can be subject to multifrequency analysis will be limited, and this limits the number of small schools that can be considered. The suggestions for transducer arrangement (below) are directed towards the design of new research vessels, or for the rearrangement of transducers on existing research vessels. Preliminary results of this paper were used when FRV “G. O. Sars” was designed (Figure 3b), when the transducers of FRV “Thalassa”, FRV “Dr Fritjof Nansen”, FRV “Scotia”, and FRV “Johan Hjort” were rearranged, and when the commercial FV “Libas” was designed. Designers of towed vehicles have also benefited from some of the ideas in preliminary versions of this paper.

Care must be taken, however, with regard to the relative size of a school compared with the beam width, because for small schools the instrument errors based on athwartship offsets and different beam widths may be greater than any differences attributable to their natural frequency response. The minimum range for the methods described are limited by the requirement that $Q_t > 0.85$, and the maximum range is limited by the effective range of the highest frequencies which are, typically, 150–200 m for a 200-kHz system aiming to detect weak targets from vessel-mounted transducers. Most of the water column over the continental shelf may, therefore, be investigated at full survey speed. However, deeper fish and weak targets such as zooplankton must be investigated either by a combination of lower frequencies, or from towed vehicles equipped with similar instrumentation. Calibration of multiple transducers over the pressure range then becomes a fresh challenge (Ona and Svellingen, 1999).

To conclude, in order to collect multifrequency data of as high quality as possible, the actions below are proposed in order of priority.
1) Select multiple frequencies that avoid harmonic interference. One possible selection of frequencies is 18, 38, 70, 120, 200, 333, 555, 926, 1543, and 2572 kHz;

2) Choose transducers with similar, if not identical, beam widths;

3) Mount all transducers as close together as possible, with the smallest transducers in the centre;

4) Synchronize the transmission from the different transducers;

5) Ensure that each ping of each frequency is time-stamped at a resolution of at least 0.01 s;

6) Select the transducer power output at levels which preclude significant losses from non-linear effects, e.g. 250 W for 120 kHz, 110 W for 200 kHz;

7) Operate all frequencies with the same pulse duration, ping rate, and digitized sample length;

8) Collect data at the lowest possible threshold, e.g., –120 dB or less;

9) Calibrate all frequencies at least once per survey, ideally twice (at the start and end);

10) Apply appropriate sound speed and absorption coefficients;

11) Remove noise where appropriate, e.g. at greater depths at higher frequencies.

Where data have been compromised through one of the above criteria not being met, one or more of the following steps may alleviate some of the problems:

- Use as low a noise threshold as possible when processing the raw data;
- Average data over multiple samples to obtain equivalent sampling volumes;
- Compensate for alongship separation of transducers by shifting pings equivalent to the separation distance;
- Compensate for transducers with different beam widths by limiting multifrequency analyses to the school kernel.

**Acknowledgements**

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**References**


**Figure legends**

**Figure 1.** Schematic diagram of some of the spatial problems for the generation of high-resolution, combined-frequency echograms from data at two arbitrary, not co-located transducers, Transducer 1 and Transducer 2. Partial overlap, or at best horizontal offset, is the normal situation. The effect of horizontal offset decreases with increasing depth, whereas the effect of vertical offset remains. Reduction of the vertical resolution may reduce the problem of vertical offset.

**Figure 2.** Fraction-horizontal overlap ($O_h$) of two beams with similar 3-dB beam widths plotted as function of depth range (R) below the transducer for several distances ($d$) between two transducer centres.

**Figure 3.** Example of placement of transducers on the instrument keels of two research vessels. (a) FRV “G. O. Sars” (IMR vessel 2) (top view). (b) FRV “G. O. Sars” (IMR vessel 3) (top view).
Figure 4. School lengths as a function of $N_{bi}$, i.e. the dimension of the echotrace relative to the beam width, for different school depths and a fixed difference between school MVBS and threshold ($dST$) value of 15 dB. Each curve shows $N_{bi}$ at a fixed depth $D$.

Figure 5. Schematic showing detection of a school by two different transducers spaced athwartship by a distance $e$, in the vertical (left) and horizontal planes (right). On the right panel, circles labelled A indicate when the beam is on the border of the school, and the school is not detected, circles labelled B indicate cases where the centre of the beam is on the border of the school where the beam is partially occupied by the school, and circles labelled C indicate cases where the school occupies the whole beam.

Figure 6. $N_{bi}$ limit values, i.e. the dimension of the echotrace relative to the beam-width limit values, as a function of school depth, calculated by empirical relationships, for an error of 0.5 dB, difference between school MVBS and threshold ($dST$) of 20 dB (continuous lines) or 10 dB (dotted lines), and athwartship distances $e$ of 0.7 m (red lines with diamonds), 1.0 m (blue lines with squares), or 2.0 m (green lines with circles).

Figure 7. Detection of a school by two beams, F1 and F2, spaced a distance apart alongship, in five successive transmissions.

Figure 8. Successive ping amplitudes for the mean depth of a 20-m-long simulated school (dotted line) located 25 m deep (simulated signal level in mV). Solid lines indicate the ratio of the signal level as detected by transducers separated by alongship distances of 0.5 m (green), 1.0 m (blue), and 2.0 m (red).

Figure 9. Potential errors, in relation to $N_{bi,7°}$, i.e. the dimension of the echotrace relative to the beam width, induced by using different nominal beam widths: $θ = 8°$, 11°, and 16°, compared with 7°.

Running headings

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Proposals for the collection of multifrequency acoustic data

Table 1. Parameters and recommended maximum input power for common sizes of Simrad transducer.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Maximum input power per frequency (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>18</td>
</tr>
<tr>
<td>Approximate transducer area ($10^{-3} \text{m}^2$)</td>
<td>200</td>
</tr>
<tr>
<td>Approximate 3-dB beam width (°)</td>
<td>11</td>
</tr>
<tr>
<td>Recommended maximum input power for 60% electro-acoustic efficiency (W)</td>
<td>5000</td>
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</table>

* Transducer area estimated by authors.

Table 2. Difference in vertical offset in EK500 and EK60 for common settings and transducers at different frequencies relative to 38 kHz. $\Delta 18$ is the difference between offsets at 18 kHz and 38 kHz.

<table>
<thead>
<tr>
<th>Echosounder</th>
<th>$\Delta 18$ (cm)</th>
<th>$\Delta 38$ (cm)</th>
<th>$\Delta 70$ (cm)</th>
<th>$\Delta 120$ (cm)</th>
<th>$\Delta 200$ (cm)</th>
<th>$\Delta 333$ (cm)</th>
</tr>
</thead>
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<tr>
<td>EK500</td>
<td>32*</td>
<td>0*</td>
<td>-0.5†</td>
<td>-12.0†</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EK60</td>
<td>15</td>
<td>0</td>
<td>-7‡</td>
<td>-10‡</td>
<td>-11‡</td>
<td>-12‡</td>
</tr>
</tbody>
</table>

EK500 “WIDE” filter
† EK500 “NARROW” filter
‡ Composite transducer
Original data  

Overlap of combined data

1. Original resolution

Potential problems in combined data
- Exact match (Ideal situation)
- Partial overlap (Normal situation)
- Total miss (Worst case)

2. Vertical resolution reduced

3. Vertical and horizontal resolution reduced

Vertical inter-frequency offset

Horizontal inter-frequency offset

Transducer 1

Transducer 2

Ping

Depth
a) Transducer mounting on protruding keel on RV “G. O. Sars” (IMR vessel 2) (top view)

b) Transducer mounting on protruding keel of RV “G.O. Sars” (IMR vessel 3) (top view)
Fig. 4: Graph showing the relationship between echotrace length (m) and Ezhore (Nbi) for different distances (D). The graph includes lines for D=25 m, D=50 m, D=100 m, and D=200 m.
Fig. 6
![Graph showing the change in MVBS (ΔMVBS) with Ping number.

- ΔMVBS+60dB
- ΔMVBS_0,5m_20dB
- ΔMVBS_1m_20dB
- ΔMVBS_2m_20dB

Ping number range: 0 to 50

ΔMVBS (dB) range: 0 to 25

Fig. 8]
Fig. 9.

\[ \Delta \text{MVBS}(7, \theta) \text{ (dB)} \]

\[ N_{bi, 7^\circ} \]