BIAS STANDARDS FOR SIGNAL PROCESSING

AIMS, PROCESSES AND RECOMMENDATIONS
BIAS - Baltic Sea Information on the Acoustic Soundscape

The Baltic Sea is a semi-enclosed sea with nine bordering states. It consists of 8 sub-catchment areas (sub-basins) and a numerous of harbours. The shipping density is one of the highest in the world. It is estimated that about 2000 sizeable ships are at sea in the Baltic Sea at a given time. Besides shipping, existing and planned wind farms contribute to the ambient noise of these waters.

In September 2012, BIAS started. This project is funded by EU LIFE+ and has three main objectives. The first objective is to establish a regional implementation of Descriptor 11 of the Marine Strategy Framework Directive through underwater sound measurements throughout the Baltic Sea and the development of user-friendly tools for management of the Descriptor. The second objective is to establish regional standards and methodologies that will allow for cross-border handling of data and results, which is necessary for an efficient joint management. The third objective is to use the measurements to model the soundscape of the entire Baltic Sea.

BIAS will solve the major challenges when implementing Descriptor 11 in the Baltic Sea. In total 38 sensors were deployed throughout the Baltic Sea in 2014 to measure the noise levels during the entire year. The measurements were performed by following the standards documented in this report. Likewise were the data analysed using standardized signal processing routines. Results were subject to a quality control and finally stored in a common data-sharing platform.

Cover picture: Sound pressure level (one minute average) vs. distance to the closest ship at the BIAS location 12 (Bothnian Bay). The figures show that the noise of a ship closer than 5 – 10 km can be distinguished from the background noise. By Jukka Pajala.
The present signal processing instruction was drawn up within the scope of the research project „Baltic Sea Information on the Acoustic Soundscape“.

This report can be cited as follows:
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A. Introduction

The ambient noise levels in the oceans has increased noticeable over the past 50 years due to increased anthropogenic activities such as shipping and many types of offshore work. Since World War II, the average level of the ambient noise increased by 12-15 dB (Bjørnø, 1998). Sound can potentially have negative effects on marine organisms by disrupting vital behaviours or physiological processes, in extreme cases even leading to their death (e.g. Fernandez et al., 2005). The European Union therefore regards the introduction of sound energy as one of the threats to the marine environment that requires an EU wide cooperative action and regulation. This is a driver for the Marine Strategy Framework Directive (MSFD), adopted by the European Union in July 2008. The main goal of the Marine Directive is to achieve a Good Environmental Status (GES) of EU marine waters by 2020. With regards to underwater sound, Descriptor 11 of the MSFD states that GES is achieved when the introduction of energy, including sound, is at levels that do not adversely affect the marine environment. For implementing this, a monitoring programme has to be established observing the current level and any trend of ambient noise in European seas. Defining acoustic parameters and establishing standards facilitate regional marine environmental management of underwater sound and guarantee compatible and quality-assured data.

The international project “Baltic Sea Information on the Acoustic Soundscape” (BIAS) started in September 2012. This project is funded by the European Commission under the LIFE+ program and national co-funders to establish a regional implementation of Descriptor 11 of the MSFD for the Baltic Sea region. The BIAS sound measurements, conducted throughout the Baltic Sea, will provide a baseline of the currently prevailing ambient noise for future monitoring. The project also establishes regional standards and user-friendly methodologies that will allow for cross-border standardized measurements, data handling, signal processing procedures and the efficient management of Descriptor 11.

BIAS released a standard for noise measurements and data handling in 2014 and a revised version in 2015 (Verfuss et al., 2014; 2015). The current document entails a standard for digital signal processing, with the scope and the terms and definitions given in chapter B and chapter C. The document describes the aim (chapter D) and procedures of signal processing and the standards that were adopted within the BIAS project (chapter E), including quality assurance (chapter E.I), pre-processing (chapter E.II) as well as processing procedures (chapter E.III), and recommendations on how to present results (chapter E.IV). Acoustic data set characteristics and formats used in the BIAS projects are given in the Data management section (chapter E.V). Short descriptions of the applied Matlab scripts are given in the Appendix.

The aim of this report is to constitute the basis for a regional monitoring of human-introduced underwater noise in the Baltic Sea in accordance with MSFD descriptor 11 that can serve as a draft for a European Standard.

The BIAS Signal Processing Standard is prepared for Baltic Sea noise data and is based on the following previously published standards:


B. Scope
This document specifies the signal processing procedures that were adopted by all partners of the BIAS project (the BIAS beneficiaries). It is restricted to topics related to the analysis of the retrieved digitized sound data and therefore only deals with digital signal processing. It does not specify data acquisition procedures. Those are published in the “BIAS Standards for Noise measurements and Data Handling” (Verfuss et al., 2014; 2015). For each step relevant in the processing of the data, a general description will be given to what has been done in the BIAS project with some justification to why it has been done, followed by a detailed specification on the algorithms or procedures and recommendations given by the BIAS team.

C. Terms and definitions
This section defines the terms as used in the following text.

a. 1/3-octave frequency band
A frequency band with a bandwidth of one third of an octave. One octave is a doubling of frequency, and one third of an octave is a frequency ratio of $2^{1/3} \approx 1.26$ between the highest and the lowest frequency (adapted from Robinson et al., 2014).

b. 1/3-octave filter
A bandpass filter with a bandwidth of 1/3-octave.

c. ANSI
American National Standards Institute.
d. **Arithmetic mean**
The sum of several measurements divided by the number of measurements.

e. **Bandwidth**
The frequency range within which a recording system is sensitive.

The frequency range (in Hertz) obtained by subtracting the lower from the upper cut-off frequency.

f. **Broadband level**
The sound pressure level obtained over a wide frequency range with defined frequency limits.

g. **Centre frequency**
*Synonym to mid band frequency.*

The centre frequency (\(f_m\)) of a band equals the geometric mean of the lower (\(f_{\text{low}}\)) and upper (\(f_{\text{high}}\)) cut-off frequencies: 
\[
 f_m = \sqrt[3]{f_{\text{low}} f_{\text{high}}}.
\]
For a 1/3 octave band the cut-off frequencies are given as 
\[
 f_{\text{low}} = 2^{-\frac{1}{3}} f_m \quad \text{and} \quad f_{\text{high}} = 2^{\frac{1}{3}} f_m \quad \text{(amended from ANSI, 2004)}.
\]

h. **Clipping**
Clipping occurs when the recording system is saturated beyond its maximum sound pressure or voltage capacity. This results in underestimation of sound levels as well as spectral distortion.

i. **Cut-off frequency (lower / upper)**
*Synonymous to band edge frequency.*

Frequencies of the lower and upper edges of a bandpass filter such that the centre frequency is the geometric mean of the lower and upper cut-off frequencies (amended from ANSI, 2004).

j. **Discrete Fourier Transform**
The digital (sampled) Fourier transform over a finite number of samples (in contrast to the integral Fourier transform, which is applied to a continuous (analog) waveform). This algorithm involves long computations, but can be applied to signals of any length, in contrast to the Fast Fourier Transform (Bloomfield, 1976).

k. **Estimate**
The rule of method of estimation is called estimator, and the value to which it give rise in a particular is called the estimate (Poularikas, 1999).

l. **Fast Fourier Transform**
A class of fast algorithm implementing the discrete Fourier transform, that efficiently calculates the discrete Fourier transform from the sampled time waveform (Bloomfield, 1976, 2000).

m. **Fourier transform**
A mathematical operation, which transforms a repetitive signal into a sum of harmonic sine wave functions. Can be performed on continuous (analog) signals (integral formulation) and digital (sampled) signals (discrete formulation) (Bloomfield, 1976).
**n. Geometric mean**
The mean of n positive numbers obtained by taking the n\textsuperscript{th} root of the product of the numbers
\[ \bar{m} = \sqrt[n]{m_1 m_2 \ldots m_n}. \]

**o. IEC**
International Electrotechnical Commission.

**p. IEC/ISO band number**
Index of the 1/3-octave bands, as defined by IEC, ISO and ANSI. Centre frequency is given from band number x as: \( f_m = 10^{x/10} \), thus band numbers 0, 18 and 21 have centre frequencies of 1 Hz, 63 Hz and 125 Hz, respectively.

**q. ISO**
International Organization for Standardization.

**r. Measurand**
The quantity intended to be measured.

**s. Ring-test**
An inter-organization comparison of signal processing methods, being a tool to verify the correct practice in data analysis.

**t. Spectrum**
A quantity expressed as a function of frequency, either as a narrowband spectrum (e.g. 1 Hz bands) or as aggregated bands (e.g. 1/3-octave bands) (Robinson et al., 2014). Spectra can be derived by various methods, such as an FFT or filtering with a set of band-pass filters. A critical requirement of a proper spectrum is that the sum of the power in all bands should equal the total power of the signal.

**u. UTC**
Universal Time Coordinated.

**D. Aim of BIAS signal processing**
The main role of the BIAS signal processing is fourfold:

1. To control the quality of data,

2. To extract relevant estimates out of the raw data set,

3. To present the properties of data in an understandable way,

4. To provide results in an appropriate format for sound scape modelling.

Long-term acoustic recordings result in extremely large data sets that have to be processed as well as stored. It is thus necessary to automate the processing and to reduce data by calculating estimates that are relevant to and accessible for the end users.

Processing of data is not uniquely defined. There are a number of optional methods that can be employed. Depending on the method the statistical estimate may change, however. The resulting
value will be influenced by for example the window size used in estimating the spectrum or the averaging time length. The aim of the standard presented here is to make data from different studies and analyses comparable and compatible.

It is the end user that defines the precision needed in estimates, what ‘relevant estimates’ imply and to whom the presentation of data is aiming at. In the case of the MSFD descriptor 11 the relevant estimates are the annual averages of the 1/3-octave bands 63 and 125 Hz. BIAS intends to also estimate a number of additional parameters based on the measured data.

E. Standards on signal processing

I. Quality control of data

Quality assurance (QA) refers to the implementation of systematic activities so that the requirements ensuring reproducible results are fulfilled. For quality assurance, the analysis is to be done in a systematic way, following guidelines and standards. It entails monitoring of processes and feedback loops which confer error prevention and inspection processes. One example for a feedback loop is the ring-test. The ring-test is a quality assurance tool for the analysis routines of different organisations that should return similar results to ensure that the correct practice in data analysis is applied and results reproducible. The statistical evaluation and interpretation of the co-operators’ results is an important tool for assessing the quality and comparability of all participants’ analyses. To establish and ensure the required data quality it is essential to understand and define the purpose of the conducted study.

Conducting signal processing to high standards requires the identification of possible error sources, i.e. identifying which processes deliver uncertain results. Identifying those allow the definition of an uncertainty budget (a list of possible errors) connected to signal processing. Signal processing is however only one source of possible errors. One has to bear in mind that there are more error sources such as the measurand, instrumentation metrological performance, calibration, sampling, interface, user and environmental conditions.

Assessing uncertainties in signal processing is necessary to improve the quality of the measurements. For giving a measure of confidence to a calculated value, possible calculation errors should be determined and the likely significance of the effect on the result identified. The uncertainties connected to the results of the signal processing analysis can be expressed as an interval in which the error values will lie with a high probability. The process of systematically quantifying error estimates is known as uncertainty analysis. There are two general classes of uncertainty:

- **Random uncertainty**, which can be assessed by repeating calculations and examining the statistical spread in the results. In digital signal processing it is possible to make repeated calculations by using the same raw data. Random uncertainty is a measure of the precision in the calculations. High precision is obtained when the calculations are repeatable and with little dispersion in results,

- **Systematic uncertainty**, which represents a potential systematic bias in an analysis, for example caused by incorrect formula for the calculation or programming error. This category
of uncertainty cannot be assessed using repeated calculations, and must be evaluated by considering the potentially influencing factors on the calculation accuracy.

Making sure that working routines are at a proper level and include personal training and quality assurance tools minimizes uncertainties.

Recommended quality assurance procedures and tools are the following:

- Control of computer system
- Testing of signal processing software
- Guidelines and standards
- Monitoring of processes
- Feedback loops

**a. Control of computer system**

One QA procedure is to nominate one person that is responsible for authorizing any software to be used. With the help of a sample data set it can be determined if values calculated by the software are correct. The controlling system for software should effective and auditable: each computer should have a log with installed hardware and software. Any new software or modifications to existing software, including new releases of commercial packages, has to be recorded in the log with first date of use. It is then possible to determine which version of the software was in use at any particular time should an error need to be traced.

It should be assured that:

- One has control of what software is loaded onto the analysis computer,
- All computer systems, including software, have been checked to ensure that they record and process data correctly,
- All software is secured against unauthorized changes,
- A log is used capturing software updates and software checks on the correct functioning after update,
- A procedure is in place for a regular backup of data stored on the computers.

**b. Testing of a signal processing software**

Software testing is a QA procedure and investigation conducted to provide users with information about the quality of the product. The testing procedure consists of test specification, test execution and test reporting.

When deciding on signal processing software it should be tested if it fits for one purpose, and that it is reliable, efficient and maintainable. It should be assured that:

- It meets the requirements that guided its design and development,
- It responds correctly to expected inputs,
- It performs its functions within an acceptable time,
- It achieves the result its users desire and it can be installed and run in its intended environments.
In BIAS, Matlab scripts were compiled specifically for the pre-processing and processing of BIAS-data, which could be used by each beneficiary, although the use of the scripts was not obligatory (see ring-test).

c. Guidelines and standards
Guidelines and standards are QA tools ensuring that procedures are done in a comparable way within a project team.

In BIAS, a working group was established for the development of the signal processing standards and any guidelines given therein, discussing and agreeing upon which standards to use for the digital signal processing of BIAS. The results are compiled and presented in the current document.

d. Monitoring of processes
Processes need to be monitored for enabling their traceability from the raw data to the results and vice versa. In BIAS, instructions and protocols were prepared tailored to the BIAS-needs to guaranty the conduction of procedures in a standardised way by all beneficiaries.

e. Feedback loops
Feedback loops enable the crosschecking of routines for a timely detection of errors. One feedback loop used in BIAS is a ring-test, for which identical sound samples, provided by the QA coordinator, were analysed by all beneficiaries. The sample data package included signal samples in WAV format, total hydrophone sensitivity or full-scale value and set gain. Additionally, software (Matlab script) for analysis was also included. Each BIAS beneficiary carried out the data analysis on the sample data using the method and software as used for the analysis of the BIAS data obtained at the measuring stations to secure that the results are comparable. They could choose to either use the common Matlab script developed for the BIAS analysis or to use their in-house software. It is important that the same method and software is used in the ring-test that later will be used for the BIAS raw data analysis. If other software than the common Matlab script was used then it had to be assured that the heading format was adapted to the in-house software as required.

Errors that can be identified with a ring test are:

- Application of wrong hydrophone sensitivity and/or instrument gain factors,
- Choosing the wrong signal channel when more than one channel was used,
- Different 1/3-octave filter configurations,
- Individual or unclear configuration of the file heading,
- Differences between software, software scripts or software versions used (these were small in BIAS, less than 1 dB).

II. Pre-processing
Pre-processing is a basic step in signal processing to prepare data for data analysis and to assess the quality of the data.

The pre-processing procedure in BIAS consists of a number of consecutive steps as follows:

- Measuring self-noise,
- Organisation of recorded data,
• Testing of data coverage,
• Testing of file size and file length of the recorded files,
• Testing of non-numerical values in the recorded data files,
• Testing of clipping.

a. Measuring self-noise
It is important to reassure that the self-noise of each recording system (including the anchoring system) does not exceed the ambient noise levels prevailing at the measuring station. This should be done by measuring and comparing the self-noise to the recorded noise levels.

b. Organisation of recorded data
For each station, the amount of data has to be determined, i.e. the number of files per month and station, and the corresponding file sizes. All data files for one station and a specific month are placed in a corresponding data folder. Data recorded before deployment after retrieval of the system are to be removed. Data recorded during deployment and retrieval, while the deployment ship was close to the sensor and contributed to the recorded ambient noise, are also to be removed.

c. Testing of data coverage
For calculating the data coverage, the time and duration of recorded data are compared relative to the planned recording time and duration. The coverage is given as percentage of period covered per period planned.

d. Testing of file size and file length
Pre-processing should entail a test of the file lengths. In the BIAS project a minimum of 15 minutes per hour recordings are required. The Matlab script bias_gaps.m (see Section G.I.d) notes the time difference between the recording length and the required minimum time of 15 minutes. This script tests the time length and the size of the recorded files by reading the header of the files. The test of the file length is done to assure that all files have the same length.

e. Testing of non-numerical values
Files with correctly recorded data contain only numerical data. In the script bias_mean.m (Section G.I.a) the quality of the data samples are checked for non-numerical values NaN (not-a-number) and Inf (too high value for a numerical representation). NaN is the IEEE arithmetic representation for Not-a-Number. A NaN is obtained as a result of not defined operations such as 0.0/0.0. Inf returns the IEEE arithmetic representation for positive infinity. Infinity is also produced by operations such as divisions by zero, (e. g. 1.0/0.0). If NaN and Inf are obtained in the data these values are omitted prior to processing. The presence of NaN or Inf in the raw data can lead to errors, which are difficult to trace, as the behaviour of processing functions when encountering NaN or Inf cannot always be predicted.

f. Testing of clipping
When received signals are of higher sound pressure than what can be picked up by the recording system, the system gets saturated. The corresponding digitised value will return the maximum sound pressure that the system is able to recorded, which is lower than the true value of the signal. This phenomenon is called clipping. For determining the quality of data, the amount of positive and negative clipping is to be quantified. The amount of clipping is calculated and presented as percentage of clipped data samples for a period of 20 s. The recommendation is to flag for clipping
when at least 0.1% of the samples per second of data is clipped. Clipping of less than 0.1% of the samples will affect the estimates of the analysis insignificantly. Information on the flagged data should be noted in the processing protocol for further consideration and tracking. Any further procedures decided upon with regards to the flagged data should be tracked carefully.

The Matlab scripts designed for the BIAS pre-processing (Section K.I) are used on data recorded with Loggerhead and Wildlife autonomous hydrophone sensors.

III. Processing

Processing of data constitutes the main analysis of the raw data with the aim of extracting the fundamental measures needed for modelling sound propagation and to sum up the statistics for the national responsible agencies and their reports to the European Commission. The Commission Decision 2010 prescribes the European member states to monitor underwater noise in the 1/3-octave bands around 63 and 125 Hz (Commission Decision 2010/477/EU, 2010).

1/3-octave band analysis is a well-established method to filter a signal with a bank of overlapping filters, all with the same Q ratio, which is the ratio between bandwidth and centre frequency. The specifications for such filter banks are well established in international standards (IEC 1995 (EN 61269), 1996 and ANSI, 2004). There are several reasons for choosing a 1/3-octave band analysis; most importantly, there are physiological justifications: Substantial experimental evidence from humans (Scharf, 1970) and other mammals (Fay, 1988) show that the mammalian ear can be well modelled by a filter bank consisting of overlapping filters, roughly 1/3-octave wide. The filter bandwidth is known as the critical bandwidth and can be measured experimentally by masking experiments. The detection threshold of a pure tone is measured while applying noise with increasing bandwidth around the tone. When the bandwidth of the noise is larger than the critical bandwidth, the detection threshold of the tone is constant, i.e. independent of the noise bandwidth. When the bandwidth of the noise is smaller than the critical bandwidth, the detection threshold of the tone decreases with decreasing noise bandwidth. The critical bandwidth is thus determined as the bandwidth where the threshold starts to decrease. A simpler, but indirect estimate of the filter bandwidth is accomplished through measuring the so-called critical ratio (Fletcher, 1940). The critical ratio is the ratio between the intensity of a pure tone signal and the spectrum density level of a broadband noise masker (which has a flat frequency spectrum) at the detection threshold of the masked tone. Because the noise is given in units of intensity per Hz, the critical ratio will automatically be given in units of a frequency. A substantial number of studies have been conducted where critical band and/or critical ratio were measured in different marine mammals. These studies include bottlenose dolphin (Au and Moore, 1990), harbour porpoise (Kastelein et al., 2009) and harbour seal (Southall et al., 2003) and are generally consistent with a constant-Q filter bank model, although with considerable variation. In actual measurements the bandwidths derived from critical ratio measurements are usually considerably smaller than those derived from direct measurements of the critical band. The critical bandwidth and the critical ratio values are not the same because central assumptions made by Fletcher (1940) in deriving the critical ratio are not met by the mammalian ear.

Thus a 1/3-octave filter bank appears to be a useful first approximation for modelling the ability of marine mammals to detect narrowband signals in noise.
1/3-octave filters have traditionally been measured using analogue filters but have now been almost entirely replaced by digital signal processing equivalents. The most direct way is to use a set of digital band-pass filters, but such filtering can be slow and difficult to implement, and great care has to be taken to make sure the filters are stable. The main problem with a digital filter bank is that a very large number of filter coefficients is required to cover the low frequencies, or alternatively, as done in the fitbank function for Matlab, recursively down-sample the signal. A much faster and simpler approach is to use filters based on a Fast Fourier Transformation (FFT). The filters are constructed to conform as close as possible to the IEC/ANSI standard for this type of filters.

Centre, lower and upper cut-off frequencies of the standard (IEC/ANSI) filters are given as:

\[
\begin{align*}
    f_m(x) &= f_r G^{(x-30)/3} \\
    f_{\text{low}}(x) &= f_m(x) G^{-1/6} \\
    f_{\text{high}}(x) &= f_m(x) G^{1/6}
\end{align*}
\]

where

- \( f_m \) is the centre frequency
- \( f_{\text{low}} \) is the lower cut-off frequency
- \( f_{\text{high}} \) is the upper cut-off frequency
- \( x \) is the IEC/ISO band number,
- \( f_r \) is the reference frequency (1,000 Hz)
- \( G \) is the octave ratio (base 10 system) with \( G = 10^{3/10} \)

Another common statistical quantity widely used in noise engineering is the percentile level. It is defined as the level \( L_N \) that is exceeded for \( N \) percent of the time interval considered. For example, \( L_1 \) is the level that is exceeded 1% of the time. The \( L_1 \) can be used as a measure for the maximum level. It is a more robust estimate than the single maximum in the data, since the absolute maximum is prone to heavy bias from singular events, such as rattling of the anchoring system or electric transient noise. Accordingly, \( L_{99} \) and \( L_{95} \) are used to describe the minimum level. \( L_{50} \) is the median level.

The fundamental requirement of the BIAS-processing is to provide mean sound pressure levels in the 1/3-octave bands centred on 63 Hz and 125 Hz as required by the MSFD. The final version of the Report of the Technical Subgroup on Underwater Noise and other forms of energy (Van der Graaf et al., 2012) points out that ambient noise peaks at higher frequency levels than these two frequencies bands, especially in shallow water, such as the Baltic Sea. The Technical Subgroup (TSG Noise) therefore suggests that higher frequency bands should also be measured and analysed, as they may prove to be valuable in future. The BIAS team shares this view and decided to additionally process the 2,000 Hz 1/3-octave band. Furthermore, the broadband sound pressure levels are determined.
BIAS SIGNAL PROCESSING

Processing of the sound data includes:

- Filtering data,
- Analysing data.

Processing should be performed only on quality-controlled and pre-processed data, so that the output has sufficient quality for any modelling and reporting. In reality, however, there may be a need for additional quality control of the output to reveal irregularities not captured by the pre-processing.

**a. Filtering data**

The extraction of the energy in the three 1/3-octave bands is made with an FFT-filter. This filter calculates the sum of the energy from frequency bins covered by the frequency range of the 1/3-octave band. The spectrum is derived by calculating the Discrete Fourier Transform (DFT) of the signal. The filters used in BIAS are bands number 18, 21 and 33. Band number 18 and 21 are the 1/3-octave bands centred at 63 Hz and 125 Hz, respectively, as required by the MSFD, while number 33 is centred at 2000 Hz (Table 1). Several adjacent FFT-bins can be summed to approximate a 1/3-octave band filter. The bandwidth will however not match exactly, as there is a finite number of bins per FFT-filter. An adjustment term is added to correct for this.

\[
BW_{\text{correction}} = -10 \log_{10} \left( \frac{BW_{\text{actual}}}{f_{\text{high}} - f_{\text{low}}} \right),
\]

where

- \( BW_{\text{correction}} \) is the correction factor to add to the sum across bins,
- \( BW_{\text{actual}} \) is the actual bandwidth value.

1/3-octave filters realised as FFT-filters are very steep, much steeper than traditional hardware filters, but still within the IEC/ANSI specifications. Filters at higher frequencies fit better to the standard whereas the discrepancy becomes substantial at the lowest frequencies.

With regards to the filtering, the BIAS recommendation is to derive the DFT with a nominal bandwidth of 1 Hz. This is a compromise between time resolution and the possibility of implementing FFT-based band-pass filters with reasonable shape even at low frequencies. Using a segment size of 1 s for the DFT, the nominal bandwidth of the DFT-bins becomes 1 Hz.

<table>
<thead>
<tr>
<th>IEC/ANSI band no.</th>
<th>Name of band (Hz)</th>
<th>( f_{\text{low}} ) (Hz)</th>
<th>( f_{m} ) (Hz)</th>
<th>( f_{\text{high}} ) (Hz)</th>
<th>Number of 1 Hz bands</th>
<th>Bandwidth correction (dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>18</td>
<td>63</td>
<td>56.2</td>
<td>63.1</td>
<td>70.8</td>
<td>14</td>
<td>0.17</td>
</tr>
<tr>
<td>21</td>
<td>125</td>
<td>112.2</td>
<td>125.9</td>
<td>141.3</td>
<td>29</td>
<td>0.1</td>
</tr>
<tr>
<td>33</td>
<td>2000</td>
<td>1778</td>
<td>1995</td>
<td>2239</td>
<td>460</td>
<td>0.01</td>
</tr>
</tbody>
</table>

Table 1  Information on the 1/3-octave band filters used in BIAS. \( f_{\text{low}} \) and \( f_{\text{high}} \) are the lower and higher cutoff frequencies, respectively. \( f_{m} \) is the centre frequency. ‘Bandwidth correction’ is explained in the text.
b. Analysing data
The BIAS calculation of the average is done in a number of consecutive steps. The basis for these steps is the pre-processed data of recorded time series from each BIAS-beneficiary, with different sampling rates depending on type of sensor used by the beneficiary. In the first step, an FFT-analysis is done over consecutive 1-second periods, giving amplitude spectra with a 1 Hz resolution. In the second step, the sound pressure levels (SPL) as defined in Verfuss et al. (2014, 2015) are calculated in the required 1/3-octave bands over 1 second. The 1-s averages are then further processed to averages over 20 seconds.

Estimates such as hourly, daily and monthly averages are calculated based on the 20-s averages, even though these are not required by the MSFD descriptor 11. Finally, the annual mean is also derived using the 20-s averages. In BIAS monthly and annual arithmetic means and the percentile levels are established for 63, 125 and 2,000 Hz as well as for the frequency band from 10 to 10,000 Hz.

For spatial modelling the BIAS-recommendation is to use the 20-second averaged data. This is a compromise between a sufficiently long time interval to obtain good estimates of the mean and an interval short enough for the noise level from nearby ships to remain approximately constant within each segment.

IV. Presentation of results
This section shows examples for common or useful representations of ambient noise data for a standardised and uniform presentation of noise measurement results. Formatting aspects of graphical representation of data, such as choosing a meaningful axis range, caption size, whether or not to include a grid, or where to place the legend, will not be given here.

For the data obtained in the BIAS project the BIAS beneficiary agreed on the following graphical presentations of the noise measurements:

- Level versus time,
- 1/3-octave spectrum,
- Level statistics: level histogram, cumulative distribution, percentile levels.

The graphs are based on sound pressure levels that were computed in steps of 1 second.

The examples shown here, in particular those from the BIAS position 31 (Fehmarnbelt) show rather high sound pressure level values compared to measurements from other places in the Baltic. This reflects the high ambient noise levels caused by a very high shipping density in this western Baltic area.

a. Level versus time
In Level versus time plots, the sound pressure level (SPL; defined in Verfuss et al., 2014, 2015) is usually presented in a classical "strip chart" diagram. The simplest option for a level versus time plot is to plot the 1-second levels, i.e. an SPL with an integration time over 1 second, the outcome of the first analysis step without further processing. An example for the 1/3-octave levels for the 63 Hz and 125 Hz bands as well as the broadband level over the whole recording bandwidth is given in Figure 1.
The broadband level is useful for a plausibility check of the data, as the 63 Hz and 125 Hz levels can never exceed the overall level.

Figure 2 shows the 20-second-average sound pressure levels of the data shown in Figure 1, meaning that they are slightly more processed.

In BIAS it was decided that the time scale should refer to the start of an averaging interval, which was implemented in Figure 2: A data point at e.g. 13:05 represents an averaging interval from 13:05:00 to 13:05:59.

Figure 1. Twenty minutes recording from the BIAS station 31, Fehmarnbelt with a ship passing by very closely at 13:04. The curves show 1-second sound pressure levels of the first analysis step without further processing.

Figure 2. 20-second-average levels (20 seconds Leq) computed from the data shown in Figure 1.
b. **1/3-octave spectrum**

A more comprehensive description of the noise measurements requires the calculation of the frequency spectrum. For underwater ambient noise 1/3-octave bands is usually an adequate frequency resolution. An alternative spectral representation, which will not be further discussed in the current document, is the narrowband spectrum (usually in 1 Hz intervals). Such a presentation is more suitable when tonal features of a sound signal shall be analysed in detail, e.g. in acoustic ship signatures or wind turbine operating noise.

Examples of 1/3-octave spectra are shown in Figure 3. The maximum frequency that can be displayed in a spectrum depends on the frequency response of the recording system and the sampling frequency used. For the Wildlife Acoustics SM2M with a sampling frequency of 32 kHz, for example, the highest usable 1/3-octave band is 12.5 kHz.

![Figure 3. 1/3-octave spectrum (60 seconds Leq) for two points in time from Figure 2: Triangles show the close approach of a ship at 13:04, and open circles show the spectrum ten minutes later.](image)

c. **Level histogram and cumulative distribution**

A method to obtain statistical information from noise recordings is to plot a histogram, a distribution of the basic 1-second levels (Figure 4). If two or more histograms shall be displayed in the same diagram, straight lines (Figure 5) may be more comprehensible than the classical bar graph style. In both figures, the values are normalized.

At times, a cumulative distribution is shown instead of a histogram (Figure 6), representing the percentage of measurement values that are lower or equal to the value given on the y-axis. However, little additional information can be obtained from it compared to the level histogram. Features such as the skewness or a secondary maximum, which may well be seen in a histogram, are difficult to recognize in a cumulative depiction.
d. Percentile level
Percentile levels can be combined with a level versus time diagram and also with a spectrum display. Possible representations are shown in Figure 7 and Figure 8. As before, the graphs are based on 1-second averages of the sound pressure level.

Figure 4. Level histograms of 1 second values for 63 Hz 1/3-octave band and for the overall level, computed from three months of recording (duty cycle is 20 minutes of recording per hour).

Figure 5. A line graph is more comprehensible than the bar graph style of Figure 4., if two or more distributions are to be shown in the same diagram.
Figure 6. 100 - Cumulative percentage of the data shown in Figure 1.

Figure 7. Daily percentile levels (i.e. interval length = 86400 s) of the 125 Hz 1/3-octave band at position 31, Fehmarnbelt, for January through March 2014.
V. Data Management

The quality of the data management is of high importance since it will directly influence the management of the results and in particular the modelling tasks.

Acoustic data, once processed, needs to be shared in order to provide a common dataset to be used for modelling. Since the modelling is mostly done at a regional scale, the data management rules will ensure uniformity and universality of the representation and the storage. A special care should be taken to the followings:

- The overall volume of acoustic data can rapidly be very high. It is recommended to handle data in chunks not longer than one-month periods;
- Non-acoustic parameters such as instrument gain, sample frequency or hydrophone depth, should be reported with a maximum of care. Automatic numerical transcription tools should be developed since:
  o The manual transcription of these parameters from paper-forms filed at-sea is a potential source of errors;
  o The manual transcription of these parameters in the processing tools is a potential source of errors.
- The quality check of the processed data should be done using all sorts of data of opportunity. For instance, the depth of the hydrophone should be compared to the bathymetric map to verify the consistency of the values reported.

For BIAS, the following common guidelines and formats are defined which will ensure access and handling of the processed data in a uniformed and universal manner. Therefore, this section addresses:
• The procedure to deliver the files in a data sharing platform;
• The common conventions to be adopted;
• The format of the files that will content the processed data to be shared;
• The content of the files.

a. Delivery procedure

The delivery procedure (Figure 9) is specified for maximizing the level of efficiency when multiple organization exchange, process and exploit the processed acoustic files. The procedure can be implemented for any period of time. In the BIAS project, it has been decided that each period is equal to a calendar month. The procedure includes a quality assessment phase which objective is to insure the consistency of the data set and to avoid errors before the dataset is made available.

Two levels of responsibility have been defined (Figure 9):

• The beneficiary level corresponds to organizations that are responsible for implementing the measurements and the signal processing. Beneficiaries are responsible for archiving the data issued from their measurement stations, and to split the data set into groups for each period of time (in BIAS, each calendar month). The processing of the raw data is performed for each period separately. Header information are updated accordingly to insure that meta-data associated to each data set is correct. Some quality check should be performed in order to avoid anomalies in the processed files produces.

• The quality manager level corresponds to a unique organization that is responsible for quality check of all the processed data. The quality manager will gather the data from each contributors or beneficiaries, and will perform the following quality verifications in order to ensure:
  o That data have been processed for each measurement station;
  o That the file names are compliant with specification defined in section E.V.c;
  o That the processed data is consistent with the deployment periods for each measurement station;
  o That all the data are present for the given period;
  o That all expected 1/3-octave bands have been processed;
  o That all files are not empty or suspiciously small;
  o Random\(^2\) validation of measurement station positions;
  o Random check of the header;
  o Random check of the format.

\(^2\) Random validation should be done on a minimum of 5 files per measurement period.
b. Common Conventions

The processed acoustic dataset are made of the acoustic data itself, but also of descriptive data such as time of measurement and position. Those data shall be released according to the following convention.

i. Acoustic quantity to be delivered for the modelling

The acoustic quantity to be delivered for the modelling is 20-second average sound pressure levels.

ii. Time convention

All time indications should use the UTC format.

iii. Position convention

All latitude and longitude are to be expressed in decimal degrees. Positive latitudes are towards North. Positive longitudes are towards East.

c. Format

This section describes the format of the files containing the processed acoustic data. Files are to be given as ASCII files. The standard C-language convention is used to describe the format for numbers, as illustrated in Table 2.
### Table 2. Formats of numbers in standard C-language convention.

<table>
<thead>
<tr>
<th>Format</th>
<th>Explanation</th>
<th>Example</th>
</tr>
</thead>
<tbody>
<tr>
<td>%03d</td>
<td>Integer coded with 3 digits. If the value is less than 100, force the number to be padded with 0s.</td>
<td>074</td>
</tr>
<tr>
<td>%05d</td>
<td>Integer coded with 5 digits. If the value is less than 10000, force the number to be padded with 0s.</td>
<td>01984</td>
</tr>
<tr>
<td>%8.2f</td>
<td>Floating point written with maximum 8 characters and with 2 decimals digits printed after the decimal point.</td>
<td>534.24</td>
</tr>
<tr>
<td>%11.8f</td>
<td>Floating point written with maximum 11 characters and with 8 decimals digits printed after the decimal point.</td>
<td>98.98747263</td>
</tr>
<tr>
<td>%5.1f</td>
<td>Floating point written with maximum 5 characters and with 1 decimal digit printed after the decimal point.</td>
<td>234.6</td>
</tr>
<tr>
<td>%10.8f</td>
<td>Floating point written with maximum 10 characters and with 8 decimals digits printed after the decimal point.</td>
<td>9.98747263</td>
</tr>
</tbody>
</table>

**i. Content of the processed acoustic files**

Each file has to contain the values of only one single bandwidth of the processed acoustic data. The amount of data contained in each file is not limited. It is however recommended to restrict the file length to one month of data.

**ii. Names of the processed acoustic files**

Each file is to be named using the following convention:

`StationIdSSS_BBBFFHz_StartYYYYMMDDHHMMSS_EndYYYYMMDDHHMMSS.asci`

Where:

- SSS is the Station Id, format: %03d;
- BBB is a keyword describing the bandwidth. It shall be ‘ThirdOctave’ or ‘BroadBand’ according to the nature of the acoustic data that is stored in the file;
- FFF is the rounded central frequency of the bandwidth when BBB is not ‘BroadBand’; format: %05d;
- Start designate the time of the first acoustic data contained in the file;
- End design the time of the last acoustic data contained in the file;
- YYYYYMMDDHHMMSS: Year, Month, Day, Hour, Minute, Second

**d. Content of the processed acoustic files**

Each file is made of a header and a table. The header of the file gives information about the measurement. The header will allow the user of the file to get the context of the data that is present in the table of the file. The following section describes how to format the header and the table.
i. Header

The header of the file

<table>
<thead>
<tr>
<th>Organization name</th>
<th>% Issued by</th>
<th>String</th>
</tr>
</thead>
<tbody>
<tr>
<td>StationId</td>
<td>% Station Id</td>
<td>Integer %03d</td>
</tr>
<tr>
<td>StationName</td>
<td>% Name of the Station</td>
<td>String</td>
</tr>
<tr>
<td>LoggerId</td>
<td>% Data Logger unit Id</td>
<td>String</td>
</tr>
<tr>
<td>DataType</td>
<td>% Type of acoustic data</td>
<td>‘SPL’ or ‘SEL’</td>
</tr>
<tr>
<td>dBUnit</td>
<td>% Units of the acoustic data</td>
<td>String</td>
</tr>
<tr>
<td>BandType</td>
<td>% ThirdOctave/ Broadband</td>
<td>String</td>
</tr>
<tr>
<td>Fc</td>
<td>% Central frequency of the band</td>
<td>Float %8.2f</td>
</tr>
<tr>
<td>Fmin</td>
<td>% Minimum frequency of the band</td>
<td>Float %8.2f</td>
</tr>
<tr>
<td>Fmax</td>
<td>% Maximum frequency of the band</td>
<td>Float %8.2f</td>
</tr>
<tr>
<td>Window</td>
<td>% Processing window (sec)</td>
<td>Integer</td>
</tr>
<tr>
<td>Latitude</td>
<td>% Lat Signed decimal degrees, WGS84</td>
<td>Float %11.8f</td>
</tr>
<tr>
<td>Longitude</td>
<td>%Lon Signed Decimal degrees, WGS84</td>
<td>Float %11.8f</td>
</tr>
<tr>
<td>L</td>
<td>% Height above bottom (m)</td>
<td>Float %5.1f</td>
</tr>
<tr>
<td>H</td>
<td>% Water depth (m)</td>
<td>Float %5.1f</td>
</tr>
<tr>
<td>Hdate</td>
<td>% Date of water depth measurement (UTC)</td>
<td>String YYYYMMDD</td>
</tr>
<tr>
<td>Issued</td>
<td>% Date of issue</td>
<td>String YYYYMMDD</td>
</tr>
<tr>
<td>ProgramName</td>
<td>% Processing program used</td>
<td>String</td>
</tr>
<tr>
<td>Sync</td>
<td>% Synchronisation Date of Data Logger</td>
<td>String YYYYMMDD</td>
</tr>
<tr>
<td>Drift</td>
<td>% Logger Drift (s/day)</td>
<td>Float %10.8f</td>
</tr>
<tr>
<td>N</td>
<td>% Number of lines in the table</td>
<td>Integer</td>
</tr>
</tbody>
</table>

ii. Table

Each row has to include columns with the following format:

- 1st column: Date – String - YYYYMMDDHHMMSS
- 2nd column: Minimum Level (%6.2f): for M-second values, the minimum level is defined as the minimum out of the M values of 1-second processed data, with M = Window (integer);
- 3rd column: Mean Level (%6.2f): for M-second values, the mean level is defined as the geometric mean of the M values of 1-second processed data;
- 4th column: Maximum Level (%6.2f): for M-second values, the maximum level is defined as the maximum out of the M values of 1-second processed data;
- 5th column: Standard deviation (%6.2f): for M-second values, the standard deviation is defined as the standard deviation of the N values of 1-second processed data;
- 6th column: Ratio of positive clipping samples in the processed window to the number of samples in the processed window (%10.8f) – Number between 0 and 1.
- 7th column: Ratio of Negative clipping samples in the processed window to the number of samples in the processed window (%10.8f) – Number between 0 and 1.

Each column is separated by a TAB.

iii. Case of no acoustic data for a station

In the case that a station / recorder has no recorded data or no valid data after processing, for any reason, all the processed files must still be produced and the filename must agree with format
described in Section c.ii. In case of no-data, the file must contain only the text **no data** in the first line, no other text.

**iv. Case of one value is unknown in the header**
In the case of a header value is unknown the field must be replaced by a dash.

**e. File template**

**i. File name**
StationId031_ThirdOctave00063Hz_Start2014010100000000_End20140104211240.asc

**ii. File content**

<table>
<thead>
<tr>
<th>FOI</th>
<th>% Issued by</th>
</tr>
</thead>
<tbody>
<tr>
<td>031</td>
<td>% Station Id</td>
</tr>
<tr>
<td>Tour Eiffel</td>
<td>% Name of the Station</td>
</tr>
<tr>
<td>SM2M-08</td>
<td>% Data Logger unit Id</td>
</tr>
<tr>
<td>SPL</td>
<td>% Type of acoustic data</td>
</tr>
<tr>
<td>dB ref. 1µPa:</td>
<td>% Units of the acoustic data</td>
</tr>
<tr>
<td>ThirdOctave</td>
<td>% ThirdOctave/Broadband</td>
</tr>
<tr>
<td>63.00</td>
<td>% Central frequency of the band</td>
</tr>
<tr>
<td>56.23</td>
<td>% Minimal frequency of the band</td>
</tr>
<tr>
<td>70.79</td>
<td>% Maximal frequency of the band</td>
</tr>
<tr>
<td>20</td>
<td>% Processing window (sec)</td>
</tr>
<tr>
<td>48.858093</td>
<td>% Latitude decimal degree WGS84-2.294694</td>
</tr>
<tr>
<td>-2.294694</td>
<td>% Longitude decimal degree WGS84 3.6</td>
</tr>
<tr>
<td>3.6</td>
<td>% Height above bottom (m)</td>
</tr>
<tr>
<td>53.4</td>
<td>% Water depth (m)</td>
</tr>
<tr>
<td>20131213121355</td>
<td>% Date of water depth measurement (UTC)</td>
</tr>
<tr>
<td>20140507</td>
<td>% Date of issue</td>
</tr>
<tr>
<td>leif_program (1.0).m</td>
<td>% Processing program used</td>
</tr>
<tr>
<td>20131214</td>
<td>% Sync Date of data logger</td>
</tr>
<tr>
<td>0.205284</td>
<td>% Drift (s/day)</td>
</tr>
<tr>
<td>7</td>
<td>% Number of lines in the table</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>% TimeUTC</th>
<th>Min</th>
<th>Mean</th>
<th>Max</th>
<th>Std</th>
</tr>
</thead>
<tbody>
<tr>
<td>20140101000000</td>
<td>100.82</td>
<td>102.50</td>
<td>105.01</td>
<td>1.41</td>
</tr>
<tr>
<td>20140101000020</td>
<td>102.21</td>
<td>104.33</td>
<td>106.98</td>
<td>1.33</td>
</tr>
<tr>
<td>20140101000040</td>
<td>99.89</td>
<td>100.01</td>
<td>101.02</td>
<td>0.22</td>
</tr>
<tr>
<td>201401010000100</td>
<td>98.76</td>
<td>100.01</td>
<td>101.02</td>
<td>0.22</td>
</tr>
<tr>
<td>201401010000120</td>
<td>102.21</td>
<td>104.33</td>
<td>106.98</td>
<td>1.33</td>
</tr>
<tr>
<td>20140101000120</td>
<td>99.89</td>
<td>100.01</td>
<td>101.02</td>
<td>0.22</td>
</tr>
<tr>
<td>201401010001240</td>
<td>100.82</td>
<td>102.50</td>
<td>105.01</td>
<td>1.41</td>
</tr>
</tbody>
</table>

**iii. Data storage**
The data storage has to be compliant with specifications meeting the general requirements for storing data on a data platform.
F. References


BIAS SIGNAL PROCESSING


G. Appendix

I. Short description of Matlab scripts for BIAS

a. Bias_mean.m
This script estimates the total power in 1/3-octave band 63, 125, 2,000 Hz and in the broad band 10 – 10,000 Hz by integration in the spectral domain for each 1 sec window and saves the results in a struct named s. Input are DSG or SM2M files.

b. Bias_output.m
This script prepares the data for output ASCII files in 20 sec arithmetic averages and estimate the arithmetic mean, median, standard deviation, max and min values and the percentage of clipping in both negative and positive direction in each 20 sec block.

c. Bias_plot.m
This script makes some plots of BIAS variables such as arithmetic mean vs. time. Also make cumulative plots in corresponding with TSG group recommendations.

d. Bias_gaps.m
This script can be used for pre-processing of DSG or SM2M data in wav files. Checking the file sizes and gaps between consecutive files. This script also checks gaps within files for checking of missing data samples.