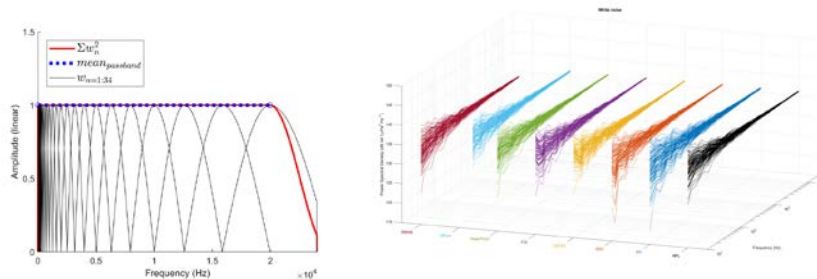**Joint Monitoring Programme for Ambient Noise North Sea**

2018 – 2020

Standard for Data Processing
of Measured Data

WP 3

Deliverable/Task: 3.3

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Cover pictures: Example plots showing (left) idealised filter band responses, and (right) plot of results from the benchmarking exercise for data analysis.

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Summary

Introduction

The aim of this project is to develop a framework for a fully operational joint monitoring programme for ambient noise in the North Sea. Output will be the tools necessary for managers, planners and other stakeholders to incorporate the effects of ambient noise in their assessment of the environmental status of the North Sea, and to evaluate measures to improve the environment.

Sounds are omnipresent in the underwater environment and can be produced by natural (waves, weather, animals) and anthropogenic (shipping, construction) sources. International concern increasingly focusses on the potential negative effects of anthropogenic underwater noise on sensitive marine fauna. Sound sources, sound transmission, and the distributions of vulnerable species in the North Sea are all transnational questions which must be tackled transnationally, as specifically required by the Marine Strategy Framework Directive.

The project will deliver an innovative combination of modelling and high quality measurements at sea for an operational joint monitoring programme for ambient noise in the North Sea. The use of consistent measurement standards and interpretation tools will enable marine managers, planners and other stakeholders internationally to identify, for the first time, where noise may adversely affect the North Sea. Next, we will explore the effectiveness of various options for reducing these environmental impacts through coordinated management measures across the North Sea basin.

This report is part of Deliverable 3.3 of Work Package 3 of project JOMOPANS.

The aim of WP3 is to develop underwater noise monitoring standards suitable for monitoring MSFD Indicator 11.2.1 in the North Sea Region as part of "JOMOPANS" joint monitoring project

At this moment, there are no international standards for monitoring and predicting ambient noise in the ocean. Such standards would require consensus on a number of topics:

(i)	terminology for describing the monitoring of underwater ambient noise;
(ii)	specification, performance requirements, calibration and deployment of the measurement equipment;
(iii)	analysis of the measured data obtained from monitoring;
(iv)	acoustic modelling of underwater sound field.

Within the project, there is a need to standardize these activities so that all partners use a common approach in order that data obtained within the project are comparable.

Scope of this document

Task 3.3 of WP3 is concerned with developing a standard for data analysis

The aim of this task is to obtain provide guidance on the processing of the measured acoustic data in a common manner.

Note that in the JOMOPANS project, the equipment used by the partners has not been standardised such that the data format from each measuring station is identical (as would be the case if every measuring station used the same model of acoustic recorder). Thus, the data analysis procedure has been made non-specific with regard to the data acquisition hardware.

This document refers to the JOMOPANS standards for Terminology with regard to definitions, and the standard for Equipment Performance, Calibration and Deployment with regard to hardware descriptions.

For the purpose of benchmarking the data analysis, files have been created using synthesized data with spectral known amplitudes that are known *a priori*. Two data files are made available in .WAV format. Instructions for use of the benchmarked data are provided in Annex D.

1 Data quality assurance and pre-processing

1.1 Introduction

It is strongly recommended that quality assurance checks are performed on the data before analysis of the data begins and (for selected data) while analysis is being conducted. This is to ensure that the data is of good quality and conforms to the requirements of the project before carrying out the data analysis. Some of the checks on data quality outlined below can only be undertaken after the data analysis is at least partially complete. The procedure described here should be read in combination with the other JOMOPANS standard procedures [4 -6], and the JOMOPANS Measurement Guidelines [7].

1.1.1 Exchange-format for underwater sound monitoring data

Storing continuous acoustic recordings requires storage and organization of large amounts of data. The data formats HDF5 and NetCDF are both well suited for this purpose and technically widely supported. These have been chosen for the JOMOPANS project. There is a high level of compatibility between both formats and an abundance of technical support for format conversion and write/read support. More details are provided in the "JOMOPANS sound data exchange format" [8].

1.2 Check for missing data and data consistency

The data coverage should be checked by inspection of the data files. Ideally, this should be accomplished by use of automated software routines.

Data coverage

To calculate the data coverage, the times stamps and durations of the actual recorded data must be compared to the planned recording times and durations. The coverage may be calculated as a percentage of the planned period of measurement.

- Check that correct number of data files exist for the duration of measurement period

- Check that the file sizes are as expected for the recording length and duty cycle

- Check that the recording durations (and start and stop times) are as expected

For each measurement station, the number of files and the size of each file will depend on the acquisition system used and the settings.

1.3 Removal of contaminated data

For autonomous recorders deployed for fixed periods, the data recorded *before* deployment *after* retrieval of the recorder shall be labelled and removed before analysis is undertaken.

In addition, data recorded *during* deployment and retrieval, and while the deployment vessel was in close proximity to recorder shall also be labelled and removed before analysis.

For hard-cabled systems, any recordings made during the installation or maintenance of the measuring system shall not be analysed. If a local sound source is deployed close to the measuring station for the purposes of providing an in-situ acoustic check on the system performance, the recordings during periods when this source is operational shall be labelled and excluded from the analysis.

1.4 Checks for clipping and distortion

If the measuring system has recorded signals which exceed the dynamic range of the system, the recordings will saturate and be "clipped" (the amplitude is truncated at the maximum value

of the ADC used to record the signal).

To determine the degree of clipping (and therefore the quality of data), the proportion of the recording where clipping has occurred must be quantified. Note that the clipping of the signal can at the positive or negative extremes of the signal.

The proportion of clipped data shall be calculated and presented as percentage of data samples within each recording period under analysis. For very long duration sound recordings, some clipping is likely if the measuring station has undergone a very close encounter with a passing vessel. It is recommended that the data file be labelled if the proportion of clipped samples within the analysis window exceeds 0.1%. (Clipping of less than 0.1% of the samples will have insignificant effect on the analysis).

1.5   Checking for spurious signals

If possible, it is beneficial to plot a times series of the data to inspect the data quality. This can only be done after at least some of the signal processing is completed (described in sections 2 and 3). The time series can take the form of a plot of the SPL in a specific third-octave band as a function of time, or a spectrogram of the data (either narrow-band or in the form of a third-octave band spectrogram). This may not be feasible for all the data due to the large data sets, but it is beneficial to undertake such analysis for a selection of the data to examine the data for spurious signals and artefacts. When examining the data in this way, undertake the following checks:

- Check for low frequency periodicities in the data (below 50 Hz) which correlate with tidal flow. This can be done by plotting the SPL in (for example) a very low frequency third-octave band (eg in the range 10 Hz to 25 Hz) alongside the local tidal variation. Strong correlation indicates that the acoustic data is likely to be contaminated by flow noise. A comparison of the SPL values at slack tide with those at full tidal flow will confirm this. Any data sequences thought to be contaminated by flow noise shall be labelled and consideration given to excluding those data from the analysis (this should apply only to frequencies below 50 Hz).

- Check for other periodic impulsive sounds in the data which may indicate spurious noise from moorings and rigging. This may be caused by abrasion of cables and ropes, collision of shackles and anchors (especially if the contact is metal on metal). Such artefacts do not represent true measures of the ambient sound field but are examples of platform self-noise (see JOMOPANS Terminology standard). Such platform-related signals can have a broad frequency content and are often difficult to identify in a time-series or spectrogram. One solution is to listen to a playback of a selection of the recorded data – the human brain is often a good classifier of sound signals. Other features to look for are periodic repetitions of the same signals; for example, the signals may correlate with tidal variation because they are worse during conditions of high tidal flow (perhaps because the moorings to undergo increased agitation).

- Check that the data is not limited by the noise-floor of the measurement instrumentation. This can be done by checking the signal level against the noise-floor of the instrumentation (this is the non-acoustic self-noise described in the JOMOPANS procedures for Terminology and for Equipment performance, calibration and Deployment, and is due to the electronic noise in the hydrophone, preamplifier and electronics). Alternatively, in post-processing it may become evident that lowest value of the measured data is limited (the distribution of values reaches a common "floor" which represents the self-noise of the measuring system). If the data appears to be limited by the noise-floor of the instrumentation, this must be noted for that data set. Note that the limitations imposed by the noise-floor cannot be relieved by increasing the gain of the amplification in the measuring chain (this increases the amplitude of both the signal and the "noise"). Instead, the noise-floor must be reduced by use of electronic components and instruments with lower electronic self-noise, or by use of a hydrophone with higher inherent acoustic sensitivity.

1.6   Software control

The software used for the analysis should be the subject of quality control to minimize errors in the analysis. Good practice in software quality control includes:

- A well-defied user specification, functional specification and operating instructions for users;

- Configuration control to provide traceability including version numbering for the software code, protection to avoid inadvertent changes being made, and documented records of modifications and improvements made to the code;

- Documented testing of the software, including re-testing when any modifications or improvements are made using benchmarked data sets

- Elimination (or at least minimization) of the opportunity for users to accidentally modify the raw input data or the output data – this includes avoiding any requirement for users to "cut and paste" data, for example between cells in a spreadsheet (where a simple mistake can cause significant errors)

- Output data files should include headers which provide traceability of the software used, and so would include the software title and version number that created the output file.

2   Data processing – conversion to sound pressure

The output of the acquisition system of the instrument measurement chain (hydrophone, preamplifier, ADC) is assumed to be a signal waveform consisting of a digitized time series expressed in digital counts and representing the signal detected by the hydrophone and recorded by the acquisition system. The data format is assumed to be a binary data file (such as a .WAV file, but other data formats may apply) containing the digitized signals of the hydrophone recordings obtained over a discrete time series, where t_i corresponds to the ith point in the time series.

The data processing to convert the waveform data to sound pressure shall be conducted according to the following steps.

a)   Read a segment of the data for analysis consisting of $N = T f_S$ points where T is the analysis window duration in seconds and f_S is the sampling frequency used when generating the data file. Note that the analysis window for the JOMOPANS has a duration of 1 second.

b)   Confirm that these data show no signs of "clipping" (overloading the maximum allowed amplitude of the measurement chain). This will be evident from data samples which reach the maximum or minimum value of the ADC.

c)   If the sensitivity of the recording system is provided as a system sensitivity for the whole acquisition system (including hydrophone, preamplifier and ADC) in counts per pascal (units: Pa^{-1}), then skip step (d) and move directly to step (e). See the JOMOPANS procedure for Equipment Performance, Calibration and Deployment for details of the system sensitivity and calibration of the instrumentation.

d)   If the system sensitivity is provided in analogue units of V/Pa (or dB re 1 V/µPa), then the digitised signal must first be converted to a voltage signal before the sensitivity can be applied to the signal. In this case, convert the signal waveform to a representation of electrical voltage in volts, $V(t_i)$, by dividing by the sensitivity of the digitiser ADC (analogue to digital converter), where the digitizer sensitivity is the number of digital counts per volt (units: V^{-1}).

e)   If the sensitivity response of the instrument measurement chain is uniform in the frequency range of interest, then a frequency-independent sensitivity may be used (a single value may be applied across the entire frequency range). In this case, the signal voltage waveform in volts, $V(t_i)$, may be converted to a sound pressure waveform $p(t_i)$ in pascals by dividing by the system sensitivity, M_s in V/Pa (assuming the system does not introduce any phase delay):

$$p(t_i) \;=\; V(t_i) \;/\; M_s \tag{1}$$

f)   If the response characteristics of the instrument measurement chain are not uniform in the frequency range of interest, then an appropriate frequency-dependent sensitivity shall be applied. If frequency-dependent sensitivity values are available from a suitable calibration, the sound pressure frequency spectra $P(f_i)$ shall be calculated from the voltage spectra $V(f_i)$ (obtained by taking the Fourier transform of $V(t_i)$) by dividing by sensitivity frequency response (the frequency spectral representation of the modulus (magnitude only) of the system sensitivity), $M(f_i)$

$$P(f_i) \;=\; V(f_i) \;/\; M(f_i) \tag{2}$$

NOTE 1   The system sensitivity in Step e) accounts for the sensitivity of the hydrophone(s), gain of amplifiers, and insertion loss of filters. For digital systems, the calibration of the digitizer (analogue to digital converter) in Step d) may be incorporated into the system sensitivity, in which case Steps d) and e) are combined into one step and the digital system sensitivity is the number of digital counts per pascal (Pa^{-1}).

NOTE 2   The phase information in the signal can be distorted by a non-uniform frequency response in a measuring system or by filtering of the signal. However, for calculation of acoustic metrics which depend on the energy or power in the signal (for example, SPL), this is not a problem. A non-uniform phase response has a significant effect only on time-domain metrics such as peak sound pressure.

NOTE 3   The combination of time sampling increment, δt_i, and number of points acquired, N_p, determines the frequency increment, δf_i, of the spectrum of the measured signal according to the equation: $\delta f_i = 1/(\delta t_i\, N_p)$. For the analysis window duration chosen in JOMOPANS of 1 second, the frequency increment will be 1 Hz. Frequently, it is found that this frequency increment of the spectrum of the measured data differs from that supplied between consecutive points in the hydrophone calibration data. To ensure that calibration data are available at the appropriate frequency points, it may be necessary to interpolate values at the required spacing from the calibration data.

3   Calculation of sound pressure level in third-octave bands

3.1   Sound Pressure Level

The Sound Pressure Level (SPL) is defined in the JOMOPANS Terminology Standard and by ISO standards ISO 80000-8 and ISO 18405. It represents the mean-square sound pressure averaged over a specified time interval expressed as a level in decibels relative to a reference sound pressure value of 1 µPa. In the guidance below, it is assumed that the raw waveform data has already been converted to values of sound pressure (see Section 2).

The SPL may be calculated in the time domain or frequency domain.

3.1.1   Time domain calculation

For a discrete series of sound pressure values, the broadband Sound Pressure Level averaged over time T (in seconds), can be obtained in dB re 1 µPa from the digitized time series of the sound pressure, $p_i = p(t_i)$, recorded at a sampling rate f_S (in Hertz), where the reference sound pressure p_0 is 1 µPa, using a numerical computer implementation of

$$L_p = 10 \log_{10} \left(\frac{1}{N\,p_0^2} \sum_{i=1}^{N} p_i^2 \right)$$

where $N = T\,fs$.

Note that this calculation is performed on the time domain signal by calculating the average of the squares of the sound pressure values in the time series over the defined time period, and then converting to a level in decibels using a reference sound pressure of 1 µPa. Note that the bandwidth of the SPL calculated in this manner is the bandwidth of the time domain signal (and so depends on the bandwidth of the recording system and instrumentation). If any filtering has been applied to the time domain signal, either before or after the digitisation, this will determine the actual bandwidth of the specific signal. For example, if the broadband signal has been filtered into a number of channels representing specific frequency bands, the SPL values for each channel represent the represent the levels for that band.

3.1.2   Frequency domain calculation

By use of Parseval's theorem, the equivalent calculation in the frequency domain is given by:

$$L_p = 10 \log_{10} \left(\frac{1}{N^2\,p_0^2} \sum_{k=1}^{N} |P_k|^2 \right)$$

where P_k are the coefficients in the discrete Fourier transform (DFT) of the sound pressure time series p_i, and where k is the index of the coefficient. Therefore, the sound pressure level, L_p, in a specified frequency band can be obtained by calculating the power sum of the individual coefficients in the frequency band.

3.2   Time-bandwidth product and uncertainty

The time-bandwidth product is the product of the time duration of a signal and its spectral width in the frequency domain. This product influences the accuracy with which signal amplitudes can be determined in the frequency domain for a given signal duration in the time domain. When calculating the spectral amplitudes of signals, a greater signal duration in the time domain will be required for accurate representation of the low-frequency spectral amplitude than for higher frequencies. This degrades the accuracy with which the levels of the low-

frequency third-octave bands may be determined for the 1 second analysis window chosen for JOMOPANS, and this is true whether the third-octave band calculation is undertaken by summation of the power in individual Fourier coefficients (the individual frequency "bins"), or by use of a digital filter. A one second analysis window provides a one hertz resolution for the bins in the frequency domain, leading to very few bins contributing to the power in the very low frequency bands. For example, in the 10 Hz band only two full bins exist between the lower band edge of 8.9125 Hz and 11.22 Hz. A digital filter for this band is very narrow and requires more than one second to reach a steady-state response. Note that the ANSI/ASA S1.11 standard recommends a signal duration of 30 seconds for accurate amplitude representation in the 10 Hz third-octave band.

This means that for repeated determination of the power in the frequency bands, the standard deviation expected is inversely proportional to frequency, with the highest standard deviations for the lowest one third-octave bands. To illustrate, Figure 1 shows the results for analyzing a 1000 second sequence of random (white) noise by dividing the signal into 1000 one second analysis windows and calculating the one third-octave band spectral density levels. The error bars show the standard deviation obtained for each one third-octave band (each standard deviation being calculated from all 1000 sequences). Note that the error bars are asymmetric because of the transformation into decibels (they are symmetric when plotted as linear percentages of the mean-square signal amplitudes). Note also that aggregating the one second values into longer sequences and calculating the arithmetic mean would reduce the standard deviation (in the same way that taking a longer observation window), but longer sequences than 1 second will not be used in JOMOPANS.
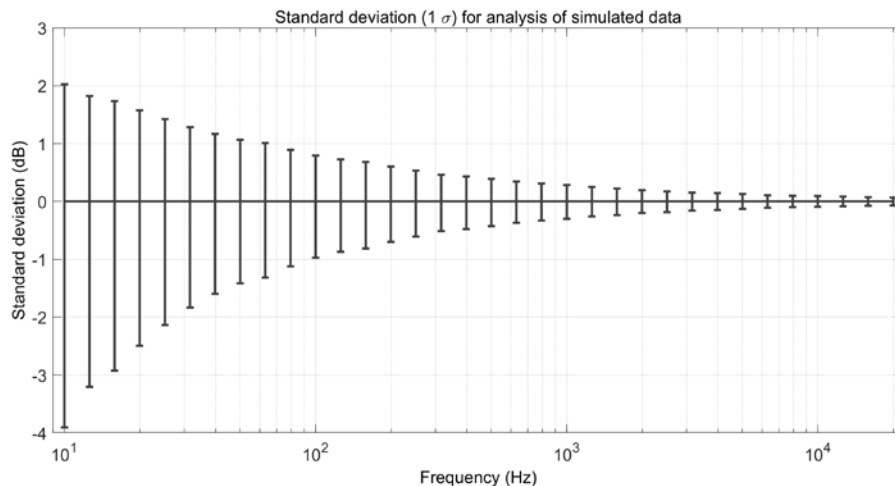

Figure 1 Results showing standard deviations from analyzing a 1000 second sequence of random (white) noise by dividing the signal into 1000 one second analysis windows and calculating the one third-octave band spectral density levels.

3.3   Calculation of third-octave band levels

3.3.1   One third octave bands

One third octave bands (base 10) shall be used for the frequency band calculations in JOMOPANS. These frequency bands are described in IEC 1260-1:2014 (these are consistent with ANSI/ASA S1.11-2014). The centre frequencies of the band f_c in hertz is given by:

$$f_c = 1000 . 10^{\frac{n}{10}}$$

where n is the band index and where the reference frequency is explicitly 1000 Hz. Upper and lower band frequencies are respectively half of one tenth of a decade above and below the centre frequency, namely:

$$f_{min} = f_c \, 10^{\frac{-1}{20}}$$

$$f_{max} = f_c \, 10^{\frac{1}{20}}$$

Table A1 in Annex A provides a listing of the one third octave frequency bands in the range from 10 Hz (n = -20) to 20 kHz (n = +13) calculated according to IEC 1260-1:2014.

3.3.2 Calculation in frequency domain using DFT

In this method, the DFT is calculated for the times series in the analysis window and the one third-octave bands powers are calculated by summing the amplitudes of the individual bins that lie within each one third-octave band (the band edge frequencies are specified in Table A1 in Annex A). This follows the equation given in section 3.1.2.

The method has the benefit of simplicity and computational speed. This method also provides unbiased estimates of the amplitudes in the one third-octave bands.

As with all methods, the method suffers from increased standard deviation at low frequencies for a limited duration analysis window due to poor frequency resolution.

At least two methods may be used to improve the performance for the low frequency bands. Firstly, a correction can be made to the bandwidth to reflect the fact that the poor resolution in the DFT leads to incorrect values of the width of the band. This correction, C_{bw}, can be calculated from the actual bandwidth used in the summation, Δf, and the upper and lower edges of the band:

$$C_{bw} = -10 \log \left(\frac{\Delta f}{f_{max} - f_{min}} \right)$$

The above assumes that the power in the band can be corrected purely by the difference in bandwidth between the desired one third-octave band and the actual band generated by the summation of the DFT bins. However, this is only true if the signal power is invariant with frequency (a flat spectral amplitude). Values of C_{bw} are given in Annex A for all 34 one third-octave bands assuming a one second analysis window and one hertz frequency resolution.

A better method of increasing performance at low frequencies is to artificially increase the frequency resolution by zero-padding the data in the analysis window in the time series. This is achieved by adding zeroes to the data sequence before performing the DFT. Doubling the length of the sequence by adding zeroes increases the frequency resolution by a factor of two, providing 0.5 Hz resolution for a one second analysis window. Note that this does *not* provide additional real data, but instead effectively performs an interpolation of the existing data. The method has the advantage of making no assumption about a flat spectral amplitude. However, the process adds a small computational overhead (not significant if the padding factor is no greater than four).

3.3.3 Calculation in the frequency domain using digital filters

In this method, a digital filter is applied to the data, the filter being created in the frequency domain and producing outputs, which are the levels of the one third-octave bands. A filter is created for each of the one third-octave bands.

The choice of filter presents a trade-off between computation time and accuracy. To accurately represent a third-octave band requires a relatively high order filter which has an increased computational cost. Use of lower order filters can provide a more rapid calculation but can lead to bias in the pass-band amplitude, leading to degradation in accuracy.

For JOMOPANS, greater onus should be placed on accurate representation of signal power. For this it would be desirable that the sum of powers through all the filter bands be unity. This can be achieved with suitable choice of filter parameters, such as the order of the filter. However, if large data sets are to be analysed, computation time becomes a consideration for the purposes of filter selection. See Annex B for more details on typical filter performance.

3.3.4 Calculation using a third octave band filter bank

Filters can be implemented in either the time or frequency domain. One third-octave filters may be implemented as infinite impulse response filters in the time domain to provide a "real-time" analysis into one third-octave bands (this type of signal processing is often employed for commercial acoustic frequency analysers). The output of such a "filter bank" may be time domain signals for each of the one third-octave bands. If so, the data sequence for each channel may then be broken into short sequences (1 second duration for JOMOPANS) and the SPL in that band may be calculated in the time domain using the equation in Section 3.1.1.

Annex B shows some examples of filter banks and compares the performance with other methods. The examples shown are the frequency responses of 34 off one third-octave filters of order eight. These filters have an emphasis on running on multiple channels in real-time.

3.4 Comparison of methods and recommendations

For the sake of accuracy and computational simplicity and speed, the recommended method for calculation of one third-octave band levels for SPL in JOMOPANS is the method described in section 3.3.2.

In this method, the DFT is calculated for the times series in the analysis window and the one third-octave bands powers are calculated by summing the amplitudes of the individual bins that lie within each one third-octave band.

It is recommended that zero-padding (to four times the window length) is used to increase the frequency resolution (at least for frequency bands below 100 Hz). If this proves computationally prohibitive, the bandwidth correction factor may be employed as an alternative for these low frequency bands.

Annex B shows a comparison of the different methods described in section 3.3 with the methods implemented in Matlab, and in one case using LabView.

3.5 Method for simulating signal data

The aim is to generate a time series of sound pressure values lasting T seconds and recorded at a sampling frequency f_S Hertz such that the time series has the following properties:

i.   It mimics noise having a specified power spectrum;
ii. Reference values for the average power over third-octave frequency bands are known and can be used as the basis for verifying the values provided by different algorithms and software.

The time series is denoted by $p_i, i = 1, \ldots, N$, where $N = Tf_S$. The one-sided power spectral density of signal is required to be proportional to $1/f^{\alpha}$, in which case the choice $\alpha = 0$ corresponds to so-called white noise and the choice $\alpha = 1$ corresponds to so-called pink noise.

The basis of the approach is to generate the time series of sound pressure values by applying a digital filter to a time series of white noise values where that digital filter is constructed to provide the desired power spectrum. The form of the digital filter is then used to evaluate directly the one-sided power spectral density of the filtered time series and, subsequently, to use that density function to evaluate directly values for the average power over third-octave bands. Following [1, 2], the digital filter is defined by the difference equation

$$a_1 p_i = b_1 e_i - a_2 p_{i-1} - \cdots - a_q p_{i-q+1},$$

with coefficients

$$b_1 = 1, \qquad a_1 = 1, \qquad a_k = \left(k - 2 - \frac{\alpha}{2}\right)\frac{a_{k-1}}{k - 1}, \qquad k = 2, \ldots, q.$$

The parameter q is the order of the digital filter and controls how well the desired power spectrum is reproduced, with higher values of q providing better approximations to the power spectrum but necessitating more computation. The input to the filter is the time series e_i,i=1,…,N, of white noise given by making independent random draws from a zero-mean Gaussian distribution. A scaling is applied to the filtered time series, and to the filter coefficient b_1, to avoid the possibility of "clipping" when it is saved as an audio "wav" file. Note that for the case α=0, the filter coefficients for the digital filter reduce to

$$b_1 = 1, \qquad a_1 = 1, \qquad a_k = 0, \qquad k = 2, \dots, q,$$

so that the filtered time series is identical to the input time series of white noise.

The transfer function of the digital filter is defined by the filter coefficients as follows:

$$H(z) = \frac{b_1}{a_1 + a_2 z^{-1} + \cdots + a_q z^{-q+1}},$$

and the complex-valued frequency response $h(f)$ of the filter as the transfer function evaluated at $z = e^{j\omega}$, where $\omega = 2\pi f$ denotes angular frequency. If follows that the one-sided power spectral density is given by:

$$P(f) = \frac{2}{f_s} |h(f)|^2.$$

Finally, for the third-octave frequency band defined by lower frequency f_{min} and upper frequency f_{max}, the average power is given by

$$B = \frac{1}{f_{max} - f_{min}} \int_{f_{min}}^{f_{max}} P(f) \, df,$$

which is evaluated to a prescribed numerical accuracy using an adaptive quadrature rule [3].

4   Presentation of benchmarking data

As part of WP3 a benchmarking exercise was carried out for project partners to compute one third octave band power levels from a common synthetic data set. The intention was to check how robust the guidance in Section 3 was and to compare the levels reported by partners following their own implementation of the processing.

Project partners were given two files and instruction to compute the power spectra as outlined in 5, Annex D. The two files were generated with the same random seed and the two noise spectra were coloured white and pink. It is noted that this description of the noise-types in the files (i.e. 'white' or 'pink') was not given to participants at the time of circulating the simulated data in case the description somehow biased the outcome of the benchmarking. The method for generating the simulated reference data is given in Section 3.5. Partners returned the spectral levels for each second of the two 100 second files individually.
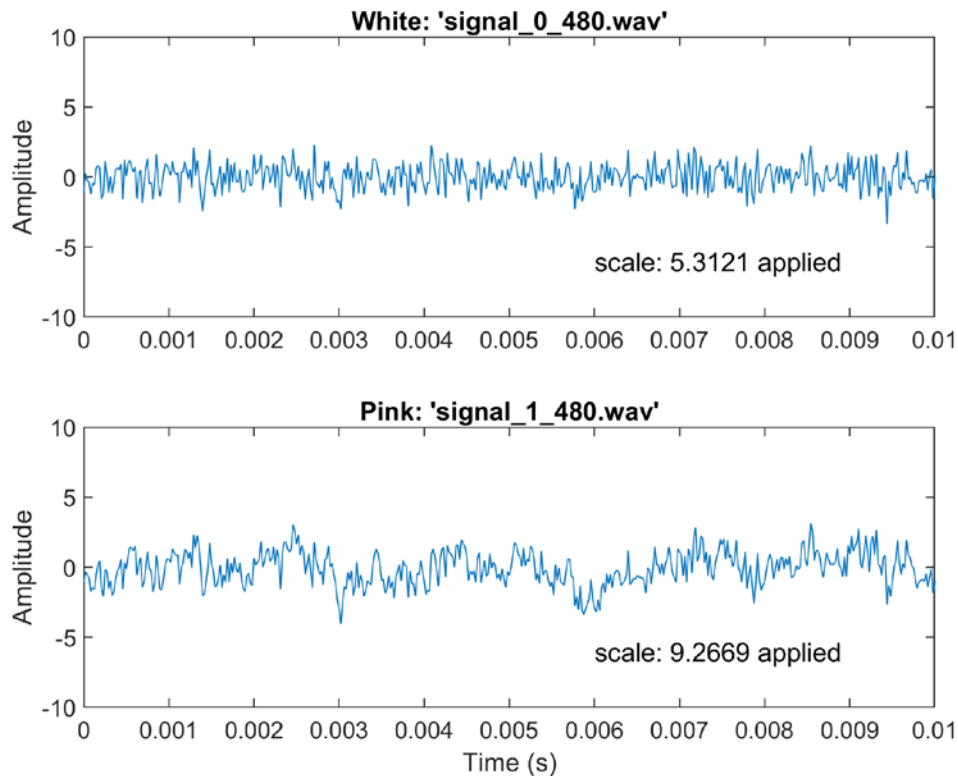

Figure 2 Time traces of the first 10ms of the two simulated signals; white noise (top) and pink noise (bottom)

For the purpose of plotting results, the calculated levels for each third octave filter-band were sorted into ascending levels for the individual one-second observations. Results are plotted in Figure 3 and 4 for the two noise types. The sound pressure spectral density level of the white noise is plotted in Figure 3. The sound pressure level plotted as a power spectrum is plotted in Figure 4. Each partner is plotted in a different colour. There are 100 lines representing the individually sorted one third octave bands for each second of signal. Theoretically, the spectral density of white noise should theoretically be flat, albeit greater variance seen at the lower frequency due to the one-second snapshot in the integration time.

The instance of pink noise is plotted in Figure 4. The sound pressure level is plotted as a power spectrum rather than spectral density (as in Figure 3) because the power spectrum of pink noise should theoretically be flat. Pink noise was an important test-case to include in the benchmarking because it better represents the practical ambient noise levels typically seen in the ocean. Again, greater variance is seen at lower frequencies due to the one-second integration time of the chosen snapshot length.

It was intended that the results be presented in such a way as to determine the equivalence between the partners' various implementations and to identify any sources of discrepancy. The benchmarking process, and returned data from partners, provided a means to learn how the guidance was implemented and the subsequent effect on the way the results present. Some notes from inspecting the returned data were:

i.   Partners returned either spectral density or power spectra and the filter bandwidths are prerequisite to interchange between the two. Where partners didn't provide their one-third octave band bandwidths it should be noted that the values in the second column of Table 1 have been used to interchange between spectral densities and power spectra

ii.   Three of the partners omitted to use the correct scaling factor (the scale factor was intended to be different for both white and pink signal examples.

iii. Some partners omitted a few frequency bands from returned data. It was not clear why this was. In some cases, these were the lowest four third-octave bands (nominally 10, 12.5, 16 and 20 Hz as per Table A1). In other cases, these were the highest four third-octave bands (nominally 10, 12.5, 15 and 20 kHz)

iv.   The term 'Scale factor' alone was ambiguous for use in scaling the recorded data. Whereas some might have interpreted the 'scale' as the dynamic range on an Analogue to Digital Converter (ADC), the 'scale factor' here means the number the to multiply the record (in the range -1 to 1 in '*.wav' data) in order to obtain the relevant Volts at the ADC.

Figure 5 and 6 present the 25th and 75th percentiles for the white and pink colour signals. The results are presented across a frequency range where all partners gave a result. This graph help to quantitatively reveal the variation in the partners' results.

It would be worthy of noting that there is no correct answer in the result, only different presentation of the same data and interesting details seen between the behaviour of each partners' one-third octave band filtering. Interestingly each partner processed their data in an identifiably different way as can be seen in Figures 5, 6, 7 and 8. This can be seen from how observations of the spectra differ between partners.

FFI and Marine Scotland showed a larger range between their 25th and 75th percentiles for both white and pink signal types. This can be seen from the larger range spanned by the blue lines in Figures 5 and 6. RBINS marginally understated the values relative to the other partners, as seen by the lower values shown by the dark red line. This is more clearly visible for the sound pressure levels of the pink noise reported as spectrum.

CEFAS gave the lowest variation in the 40 Hz band as seen in Figures 7 and 8 between the 1st and 100th percentiles. All partners showed a similar trend in the reduction of variance with increasing frequency.

Finally, Figures 9 and 10 show the mean spectral density and pressure level. FOI and CEFAS report fluctuating means between 50 and 80 Hz for the white noise plots in Figure 9. Many partner observe a notch in the mean sound pressure level for the pink noise at 63 Hz in Figure 10. RBINS reported the lowest levels at low frequency.

The benchmarking process showed that the guidance in Section 3 could be followed and the reported spectral density and power spectrum for the two cases could be largely processed consistently between partners, albeit variations arising between them as a result of the exact implementation.
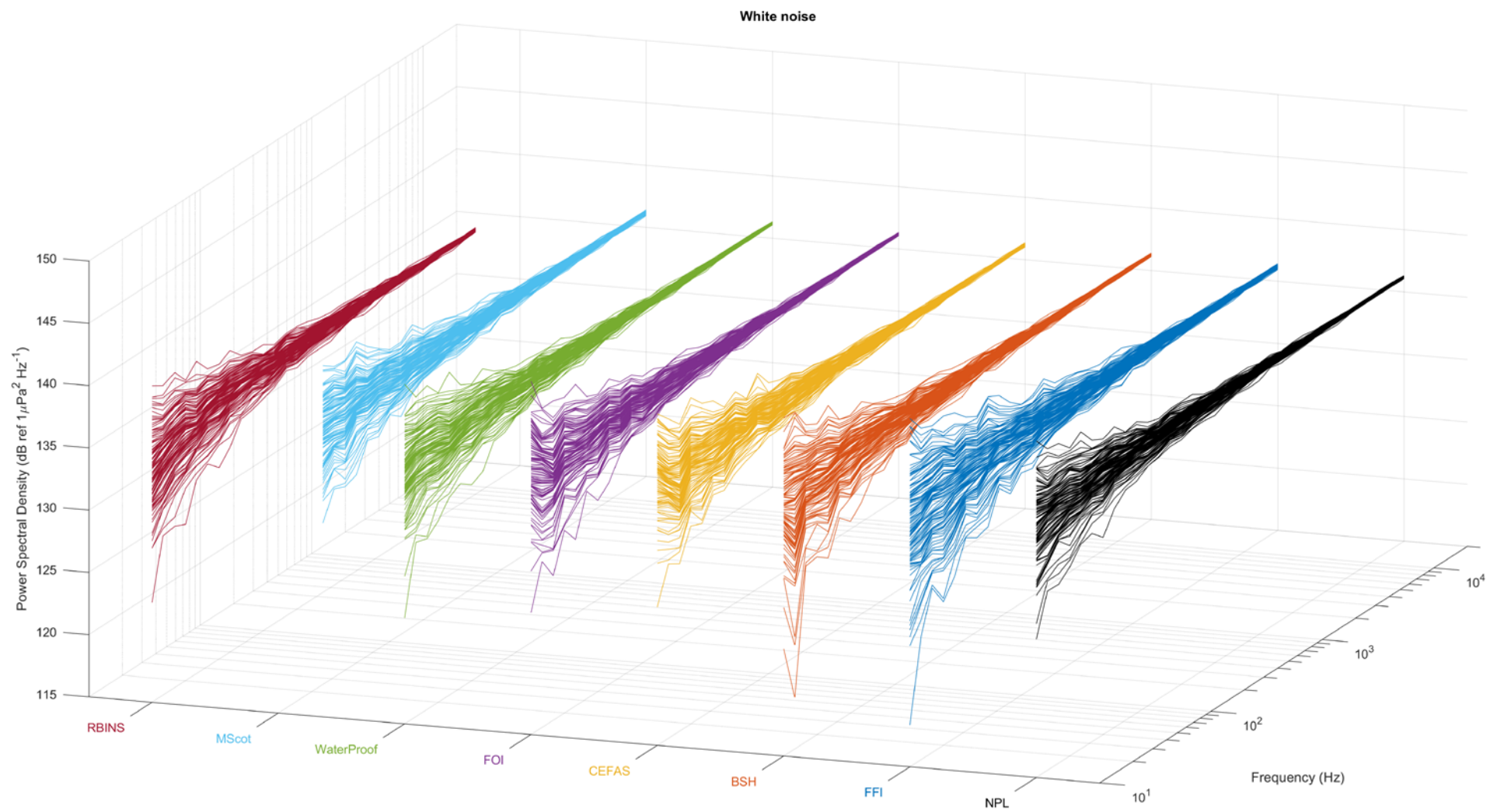
Figure 3 Statistical spread of reported percentiles in the 100 seconds of white noise as reported by eight partners
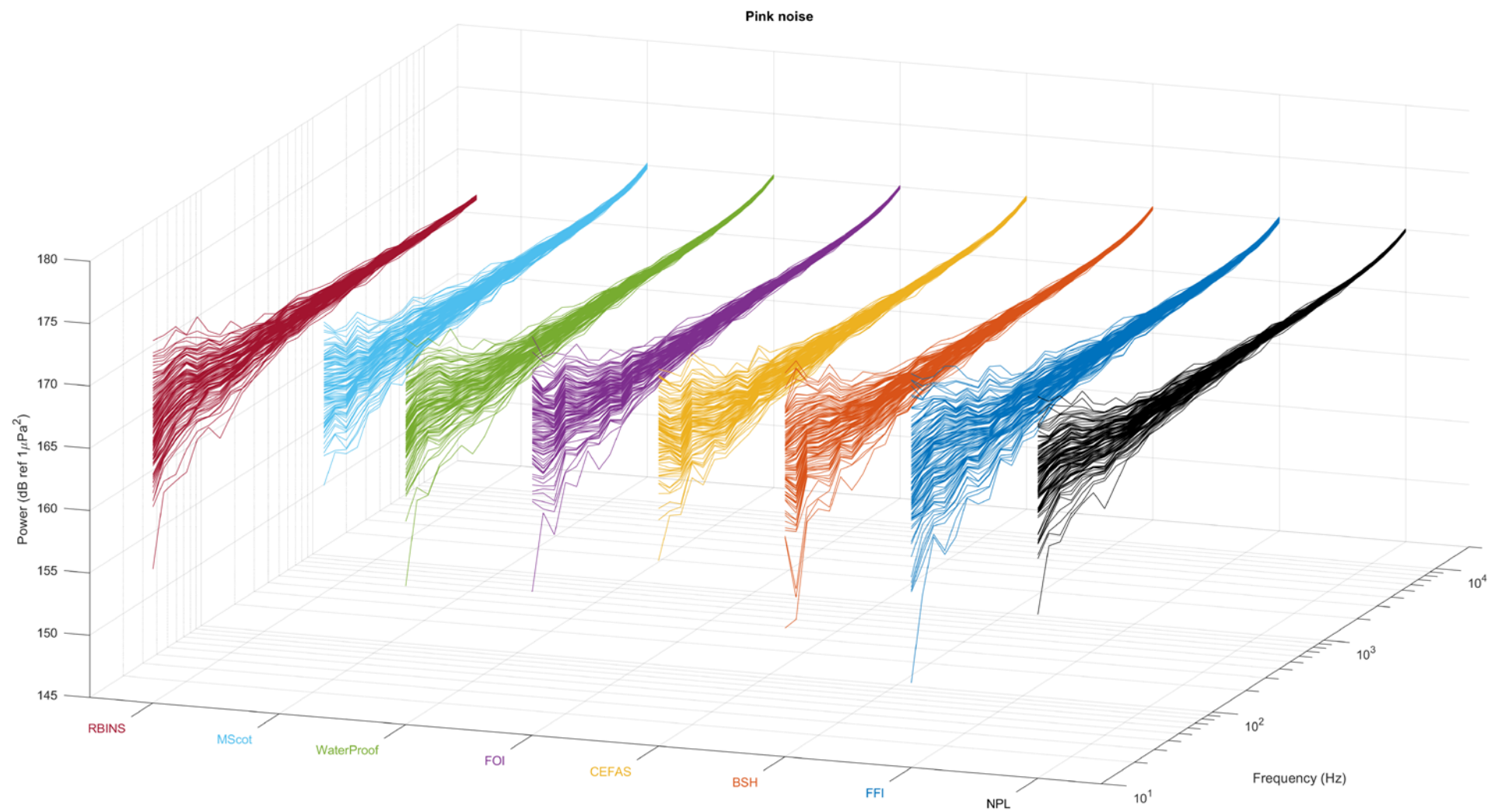
Figure 4 Statistical spread of reported percentiles in the 100 seconds of pink noise as reported by eight partners
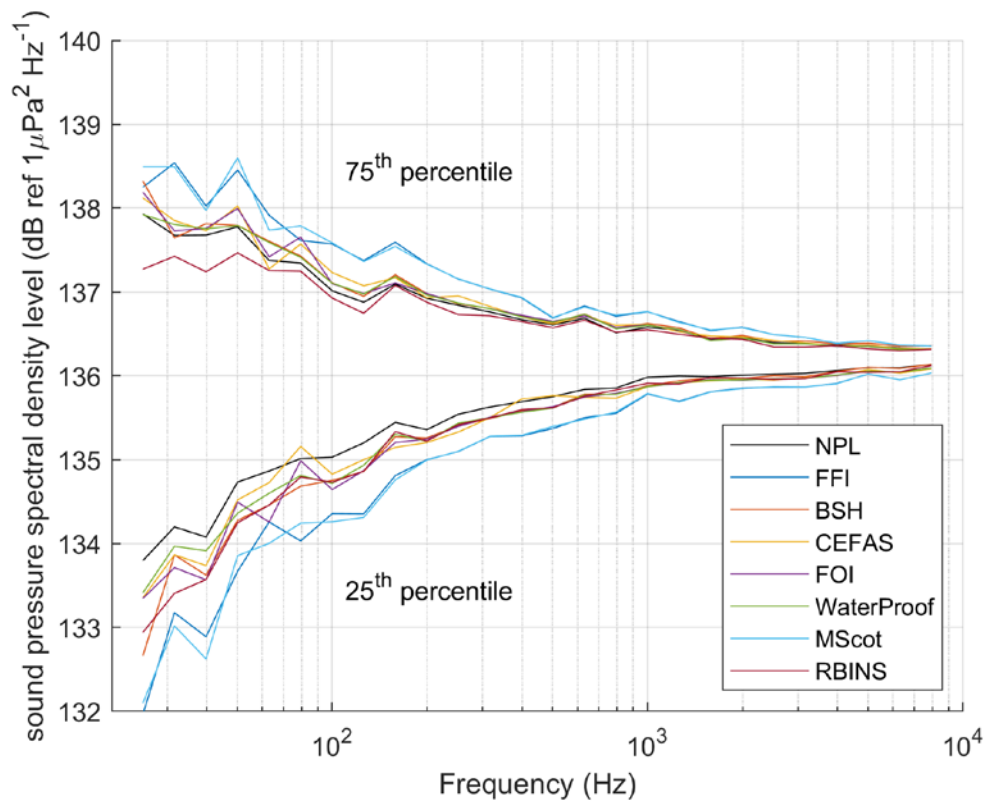
Figure 5 Sound pressure spectral density levels of **white noise** in one third-octave bands as reported by partners in the benchmarking process (showing the 25th and 75th percentiles)
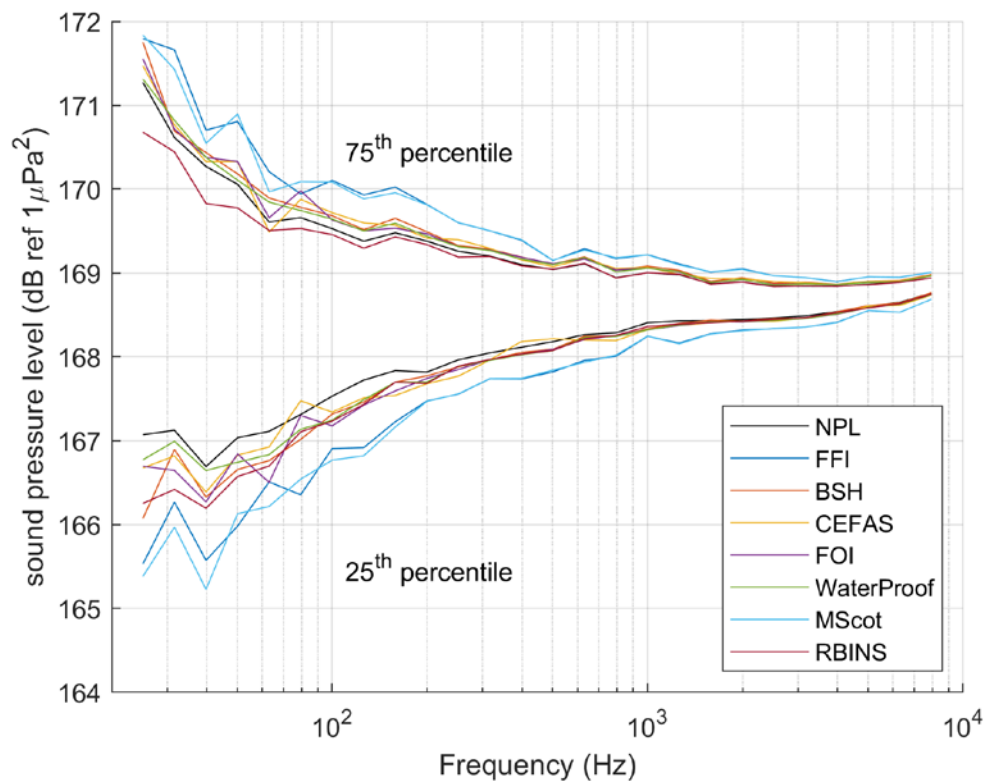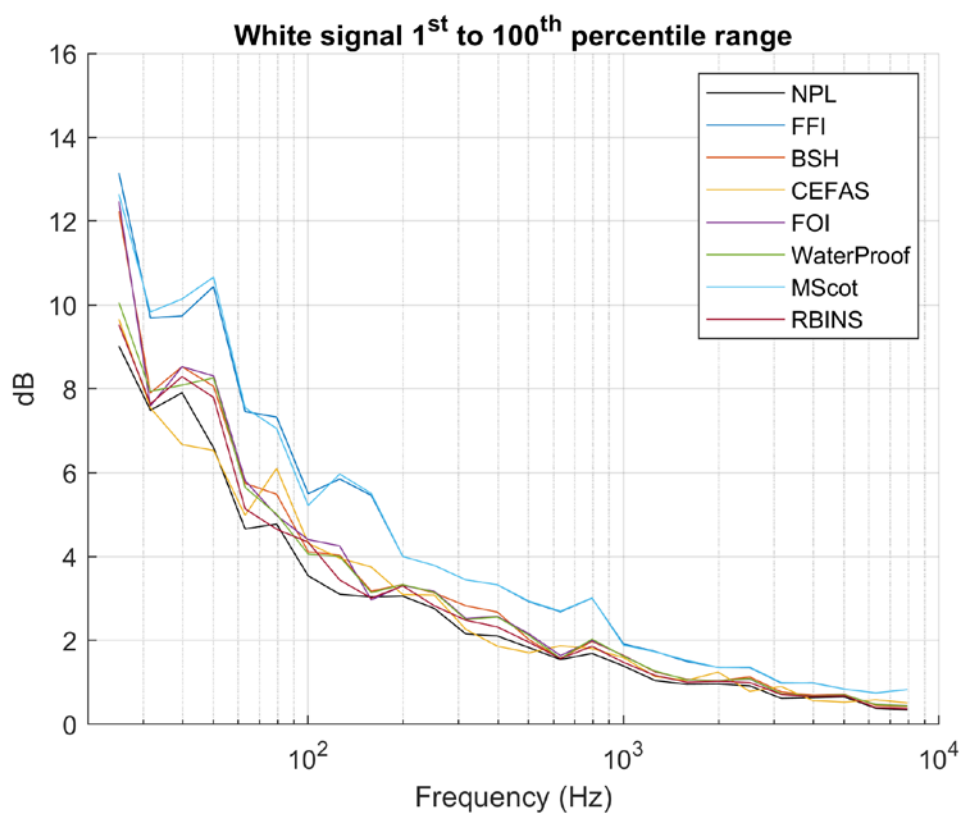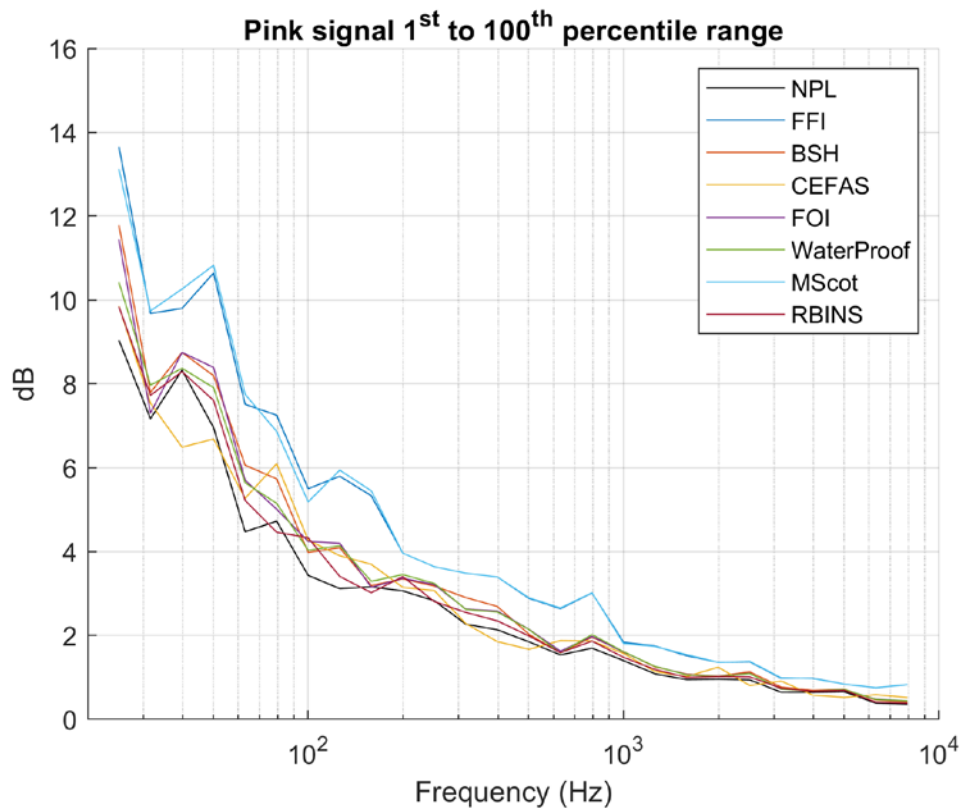

Figure 6 Sound pressure levels of **pink noise** in one third-octave bands as reported by partners in the benchmarking process (showing the 25th and 75th percentiles).

Figure 7 Range (max-min) of sound pressure spectral density level reported for the **white noise**

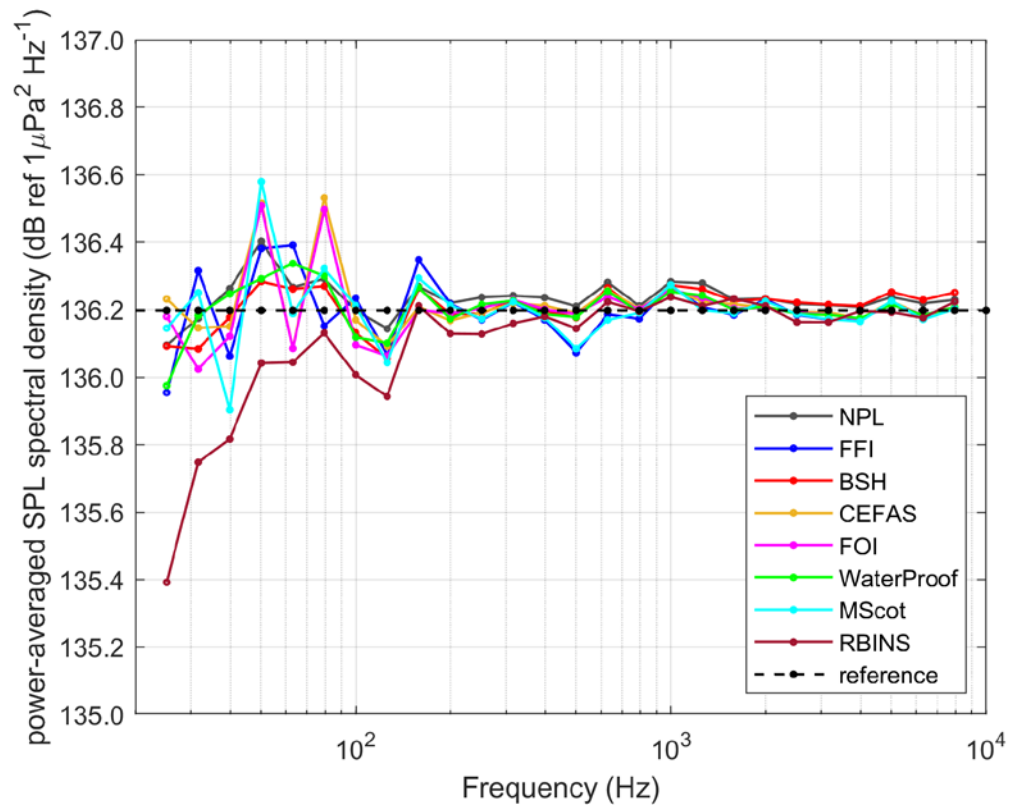Figure 8 Range (max-min) of sound pressure level reported for the **pink noise**

Figure 9 Arithmetic mean (power average of squared pressures) for the 100 one-second instances of white noise.
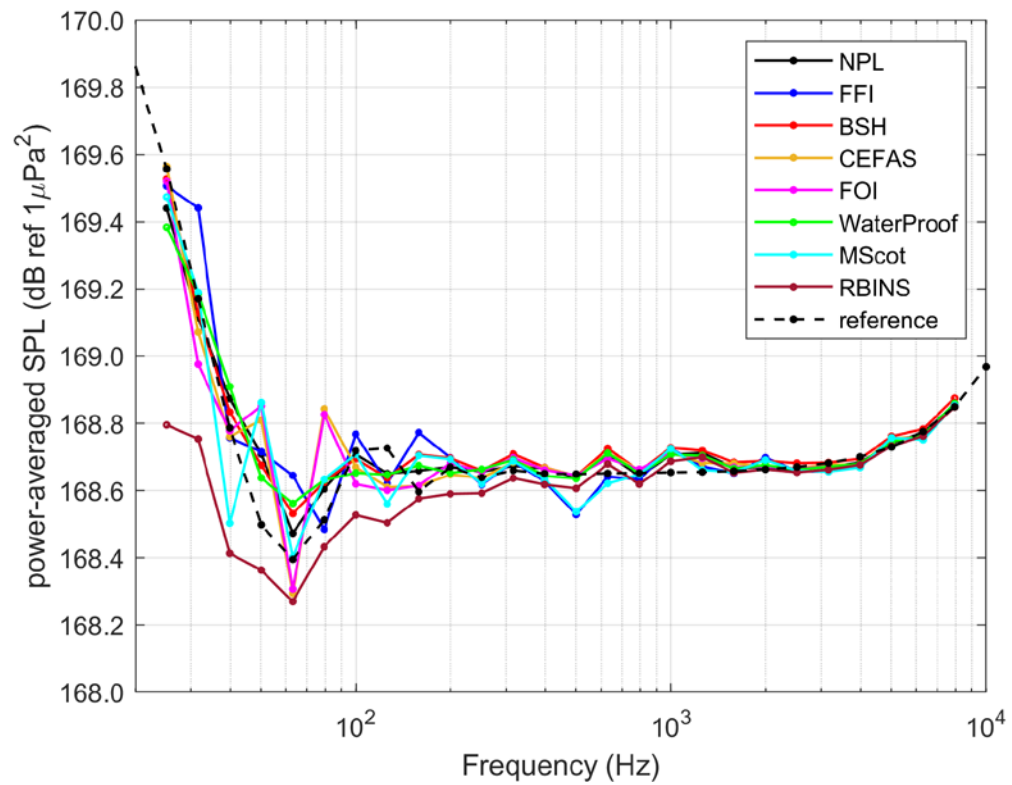

Figure 10 Arithmetic mean (power average of squared pressures) for the 100 one-second instances of pink noise.

Freq (Hz)	Bwidth (Hz)	NPL	FFI	BSH	CEFAS	FOI	WaterP	MScot	RBINS
25.1	5.8	9.0	13.1	12.2	9.7	12.5	10.1	12.6	9.5
31.6	7.3	7.5	9.7	7.9	7.5	7.6	7.9	9.8	7.6
39.8	9.2	7.9	9.7	8.5	6.7	8.5	8.1	10.1	8.3
50.1	11.6	6.6	10.4	8.1	6.5	8.3	8.3	10.7	7.8
63.1	14.6	4.7	7.5	5.7	5.0	5.8	5.7	7.5	5.1
79.4	18.3	4.8	7.3	5.5	6.1	5.0	5.0	7.1	4.7
100.0	23.1	3.5	5.5	4.1	4.3	4.4	4.1	5.2	4.3
125.9	29.1	3.1	5.8	4.0	4.0	4.3	4.0	6.0	3.4
158.5	36.6	3.0	5.5	3.2	3.8	3.0	3.1	5.5	3.0
199.5	46.0	3.1	4.0	3.3	3.1	3.3	3.3	4.0	3.3
251.2	58.0	2.8	3.8	3.1	3.1	3.2	3.2	3.8	2.8
316.2	73.0	2.2	3.4	2.8	2.3	2.5	2.5	3.4	2.5
398.1	91.9	2.1	3.3	2.7	1.9	2.6	2.6	3.3	2.3
501.2	115.7	1.8	2.9	2.0	1.7	2.2	2.1	2.9	2.0
631.0	145.6	1.5	2.7	1.6	1.9	1.6	1.6	2.7	1.6
794.3	183.3	1.7	3.0	2.0	1.8	2.0	2.0	3.0	1.9
1000.0	230.8	1.4	1.9	1.6	1.6	1.6	1.6	1.9	1.5
1258.9	290.5	1.0	1.7	1.2	1.1	1.3	1.3	1.7	1.2
1584.9	365.7	1.0	1.5	1.1	1.0	1.1	1.1	1.5	1.0
1995.3	460.4	1.0	1.4	1.0	1.2	1.0	1.0	1.4	1.0
2511.9	579.7	0.9	1.4	1.1	0.8	1.1	1.1	1.4	1.0
3162.3	729.8	0.6	1.0	0.8	0.9	0.7	0.7	1.0	0.7
3981.1	918.7	0.6	1.0	0.7	0.6	0.7	0.7	1.0	0.7
5011.9	1156.6	0.7	0.8	0.7	0.5	0.7	0.7	0.8	0.7
6309.6	1456.0	0.4	0.7	0.5	0.6	0.5	0.5	0.7	0.4
7943.3	1833.1	0.3	0.8	0.4	0.5	0.4	0.4	0.8	0.4

Table 1 Range of variation for one third octave bands reported by partners in white noise (file 'signal_0_480.wav'). Values in italics are decibels.

Freq (Hz)	Bwidth (Hz)	NPL	FFI	BSH	CEFAS	FOI	WaterP	MScot	RBINS
25.1	5.8	9.0	13.7	11.8	9.9	11.4	10.4	13.1	9.8
31.6	7.3	7.2	9.7	7.8	7.5	7.3	8.0	9.7	7.7
39.8	9.2	8.3	9.8	8.7	6.5	8.7	8.4	10.3	8.3
50.1	11.6	7.0	10.6	8.2	6.7	8.4	7.9	10.8	7.6
63.1	14.6	4.5	7.5	6.1	5.3	5.7	5.6	7.7	5.2
79.4	18.3	4.7	7.3	5.7	6.1	5.0	5.1	6.9	4.5
100.0	23.1	3.4	5.5	4.0	4.3	4.2	4.0	5.2	4.3
125.9	29.1	3.1	5.8	4.1	3.9	4.2	4.1	5.9	3.4
158.5	36.6	3.2	5.3	3.2	3.7	3.1	3.3	5.4	3.0
199.5	46.0	3.1	4.0	3.3	3.1	3.4	3.4	4.0	3.4
251.2	58.0	2.8	3.6	3.2	3.1	3.2	3.2	3.6	2.8
316.2	73.0	2.3	3.5	2.9	2.3	2.6	2.6	3.5	2.5
398.1	91.9	2.1	3.4	2.7	1.9	2.6	2.6	3.4	2.3
501.2	115.7	1.8	2.9	2.0	1.7	2.1	2.1	2.9	2.0
631.0	145.6	1.5	2.6	1.6	1.9	1.6	1.6	2.7	1.6
794.3	183.3	1.7	3.0	2.0	1.9	2.0	2.0	3.0	1.9
1000.0	230.8	1.4	1.8	1.6	1.6	1.6	1.6	1.8	1.5
1258.9	290.5	1.1	1.7	1.2	1.1	1.3	1.3	1.7	1.2
1584.9	365.7	0.9	1.5	1.1	1.0	1.1	1.1	1.5	1.0
1995.3	460.4	1.0	1.4	1.0	1.2	1.0	1.0	1.4	1.0
2511.9	579.7	0.9	1.4	1.1	0.8	1.1	1.1	1.4	1.0
3162.3	729.8	0.6	1.0	0.8	0.9	0.7	0.7	1.0	0.7
3981.1	918.7	0.6	1.0	0.7	0.6	0.7	0.7	1.0	0.7
5011.9	1156.6	0.7	0.8	0.7	0.5	0.7	0.7	0.8	0.7
6309.6	1456.0	0.4	0.8	0.5	0.6	0.5	0.5	0.8	0.4
7943.3	1833.1	0.4	0.8	0.4	0.5	0.4	0.4	0.8	0.4

Table 2 Range of variation for one third octave bands reported by partners in pink noise (file 'signal_1_480.wav'). Values in italics are decibels.

The following are examples of ways of presenting the measured data after analysis.

4.1 SPL versus time

4.2 One third-octave band levels

4.3 Histogram of SPLs and cumulative distributions

4.4 Percentiles

Annex A   One third of octave band frequencies

One third of octave bands (base 10) shall be used for the frequency band calculations in JOMOPANS. These frequency bands are described in IEC 1260-1:2014 (these are consistent with ANSI/ASA S1.11-2014). The centre frequencies f_c are

$$f_c = (1\ kHz)\ 10^{\frac{n}{10}}$$

where n is the band index. Upper and lower band frequencies are respectively half of one tenth of decade above and below the centre frequency, namely:

$$f_{min} = f_c\ 10^{\frac{-1}{20}}$$
$$f_{max} = f_c\ 10^{\frac{1}{20}}$$

Table A1 (below) shows the one third octave frequency bands with centre frequencies from 10 Hz (n = -20) to 20 kHz (n = +13) calculated according to IEC 1260-1:2014.

Table A1 One third octave frequency bands

Band index	Lower bound	Centre frequency	Upper bound	Nominal centre frequency
n	f_{min}/Hz	f_c/Hz	f_{max}/Hz	$f_c, norm$/Hz
-20	8.9125	10	11.22	10
-19	11.22	12.589	14.125	12.5
-18	14.125	15.849	17.783	16
-17	17.783	19.953	22.387	20
-16	22.387	25.119	28.184	25
-15	28.184	31.623	35.481	31.5
-14	35.481	39.811	44.668	40
-13	44.668	50.119	56.234	50
-12	56.234	63.096	70.795	63
-11	70.795	79.433	89.125	80
-10	89.125	100	112.2	100
-9	112.2	125.89	141.25	125
-8	141.25	158.49	177.83	160
-7	177.83	199.53	223.87	200
-6	223.87	251.19	281.84	250
-5	281.84	316.23	354.81	315
-4	354.81	398.11	446.68	400
-3	446.68	501.19	562.34	500
-2	562.34	630.96	707.95	630
-1	707.95	794.33	891.25	800
0	891.25	1000	1122.0	1000
1	1122.0	1258.9	1412.5	1250
2	1412.5	1584.9	1778.3	1600
3	1778.3	1995.3	2238.7	2000
4	2238.7	2511.9	2818.4	2500
5	2818.4	3162.3	3548.1	3150
6	3548.1	3981.1	4466.8	4000
7	4466.8	5011.9	5623.4	5000
8	5623.4	6309.6	7079.5	6300
9	7079.5	7943.3	8912.5	8000
10	8912.5	10000	11220	10000
11	11220	12589	14125	12500
12	14125	15849	17783	15000
13	17783	19953	22387	20000

Table A2. Values of bandwidth correction factor for all 34 one third-octave bands assuming a one second analysis window and one hertz frequency resolution

Band index	Centre frequency	Nominal centre frequency	Correction (dB)
n	f_c/Hz	$f_c, norm$/Hz	
-20	10	10	-1.14
-19	12.589	12.5	-0.14
-18	15.849	16	0.86

-17	19.953	20	-0.36
-16	25.119	25	-0.15
-15	31.623	31.5	0.18
-14	39.811	40	0.09
-13	50.119	50	-0.16
-12	63.096	63	0.17
-11	79.433	80	-0.16
-10	100	100	0.01
-9	125.89	125	0.01
-8	158.49	160	0.07
-7	199.53	200	0.00
-6	251.19	250	0.00
-5	316.23	315	0.00
-4	398.11	400	-0.01
-3	501.19	500	-0.01
-2	630.96	630	0.02
-1	794.33	800	-0.02
0	1000	1000	0.00
1	1258.9	1250	0.01
2	1584.9	1600	0.00
3	1995.3	2000	0.00
4	2511.9	2500	0.00
5	3162.3	3150	0.00
6	3981.1	4000	0.00
7	5011.9	5000	0.00
8	6309.6	6300	0.00
9	7943.3	8000	0.00
10	10000	10000	0.00
11	12589	12500	0.00
12	15849	15000	0.00
13	19953	20000	0.00

Annex B   Comparison of methods for calculation of one third-octave band levels

Annex B.1 Methods to process noise data

Five different methods of processing noise data to produce power spectrum density in the one third octave bands of interest were compared in order to help the selection of the most suitable ones. The methods are:

- DFT with frequency bin summation
- DFT zero padded
- DFT with frequency bin summation using the "bandpower" function in MATLAB (Signal Processing Toolbox)
- Filter bank created in the MATLAB Audio Systems Toolbox (filter M)
- Filter bank created in LABVIEW (filter L)

DFT method

Assuming the noise data in one second is x, the power within the jth one third-octave band with the frequencies within the low and high end of the band [f_l f_h] was calculated using FFT as

$$P_j = \sum_{n=n\_low}^{n\_high} |S(n)|^2 \qquad (1)$$

where

$$S = fft(x)$$

S is the amplitude of the signal in frequency domain, and n_low is the lowest frequency number above the low end of the one third-octave band, $n_{\_low} \geq f_l$, and n_high is the highest frequency number below the high end of the one third-octave band frequency, $n_{\_high} \leq f_{h,}$. The power spectrum density can be obtained by dividing Eq (1) with corresponding bandwidth of the one third-octave frequency.

This method is very simple to implement to calculate the power within the band, but it introduces larger errors at the lower end of the one third-octave frequencies as the numbers of frequencies with these bands are small. The effects of small number of used frequencies can be reduced with zero-padding the time signal so that the numbers of frequencies with all the one third-octave bands are increased, resulting in smaller errors The number of zeros determines the errors, the large the number the smaller the error. However, the processing time will also increase with the number of zeros.

Bandpower method

Bandpower is a command in MATLAB. It takes the signal data sampled at a given frequency, and low and high end of the band as three input parameters. It produces results very close to the FFT method at the frequencies above 1 kHz, but introduces a large bias at the low end of the frequency. Zero-padding does not work properly as the power level changes with different length of zeros.

one third-octave filter banks in MATLAB (filter M)

This one third-octave filter bank can be designed in the Audio tool box of the MATLAB according to the specification parameters, such as the centre frequency of the band, the sampling frequency and the order.
It filters the time domain signal in parallel for all the one third-octave bands. One advantage of the filter is that it can be applied to time data first to produce output data from all the one third-octave bands before applied processing time window to calculate the required metrics. This will reduce the bias in the mean of processed data at the low frequency considerably.
However, there is a small bias of 0.1 dB in the output level with the filter in comparison with the other methods. This cause of this bias is unknown at this stage.

one third-octave filter banks in LABVIEW (filter L)

The filter banks by LABVIEW (filter L) was also tested against the other methods mentioned above. It has a comparable mean as the zero-padded FFT, but without the bias at the high frequency end as the filter M.

Annex B.2 Results of comparison

The test was carried out with 100 seconds of data sampled at 48 kHz. The data was divided into 100 one second sequences for analysis by DFT, zero padded DFT, DFT bandpower and the filter L. No windows are applied in the processing. The frequency resolution of DFT and the filter L was 1 Hz. The results are shown in Figure B1 (below).
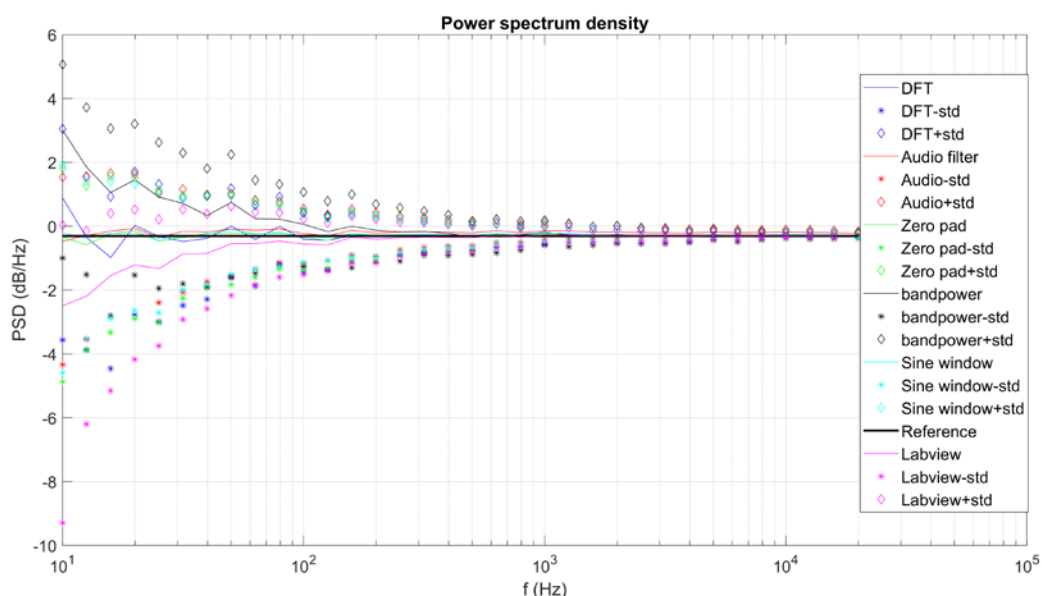

Figure B1 Comparison of results for power spectral density for different analysis methods showing the arithmetic mean of the analysis of 100 one-second sequences (lines) and the standard deviations (symbols).

Annex B.3 Third octave filter behaviour

The choice of filter presents a trade-off between computation time and accuracy.

Filters can be implemented in either the time or frequency domain. A particular onus is placed on accurate representation of power. For this it would be desirable that the sum of powers through the filter bands be unity. However, if large data sets are present, then computation time may also be a consideration for the purposes of filter selection.

Figure B2 shows one third-octave bands implemented as infinite impulse response filters in the time domain[1]. The examples shown are the frequency responses of 34 off one third-octave filters of order eight. These filters have an emphasis on running on multiple channels in real-time.

Figure B3 shows one third-octave bands implemented as sine windows applied in the frequency domain. These can be applied to the amplitude spectrum after using the fast Fourier transform on the received signal. This filtering technique has the advantage of more accurately producing unity gain across the full passband. However, this is at marginally more computational cost than the technique in Figure 1.

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[1] Mathworks MATLAB 2017b, Audio Systems Toolbox
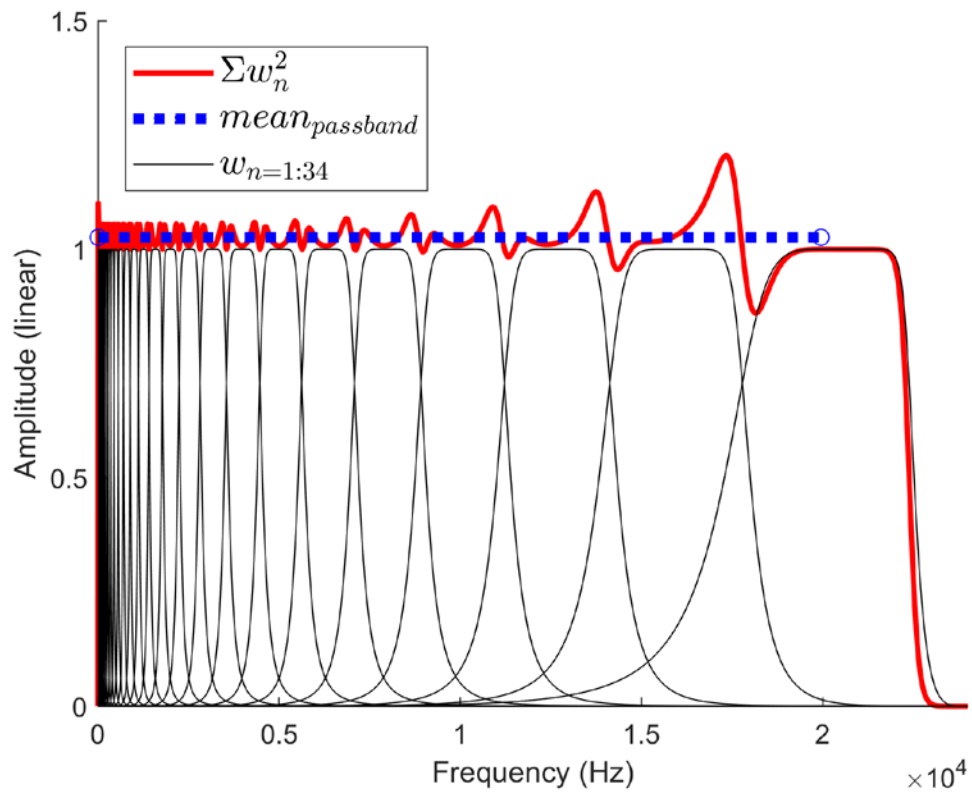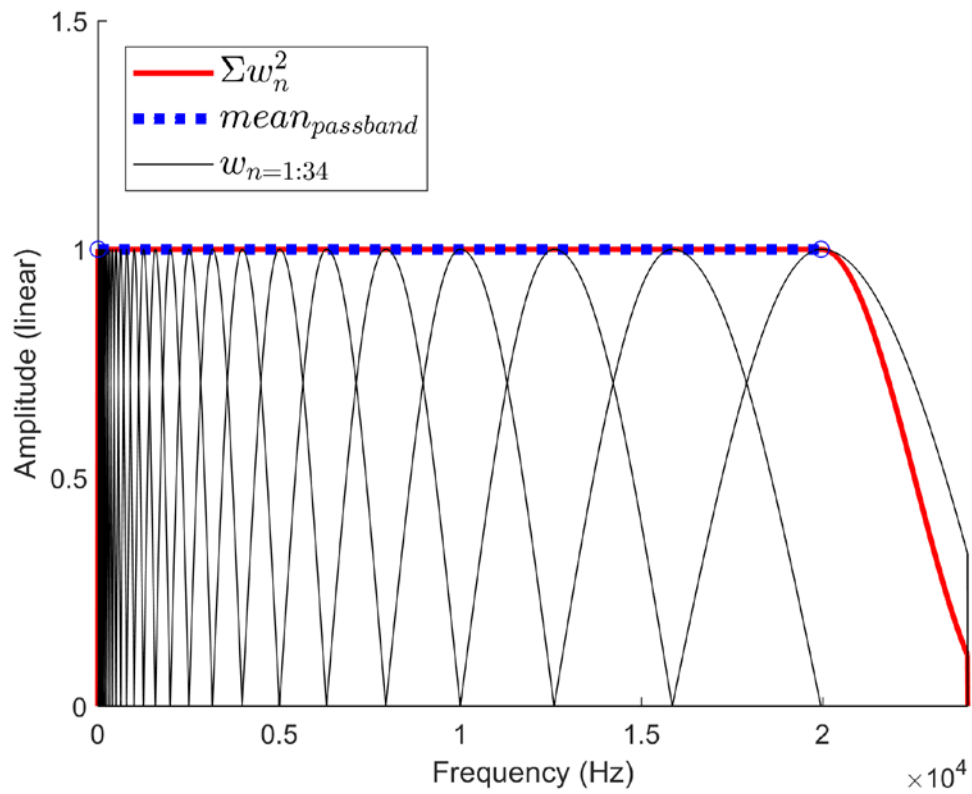
Figure B2

Figure B3

Annex C   Examples of spurious signals in measured data

The plots below show one third-octave band spectrograms which illustrate the presence of spurious signals due to the influence of factors such as platform noise and flow noise.

Figure C1 and C2 show third-octave band spectrograms of acoustic measurements in UK waters showing periodic features present due to flow noise and other influences.
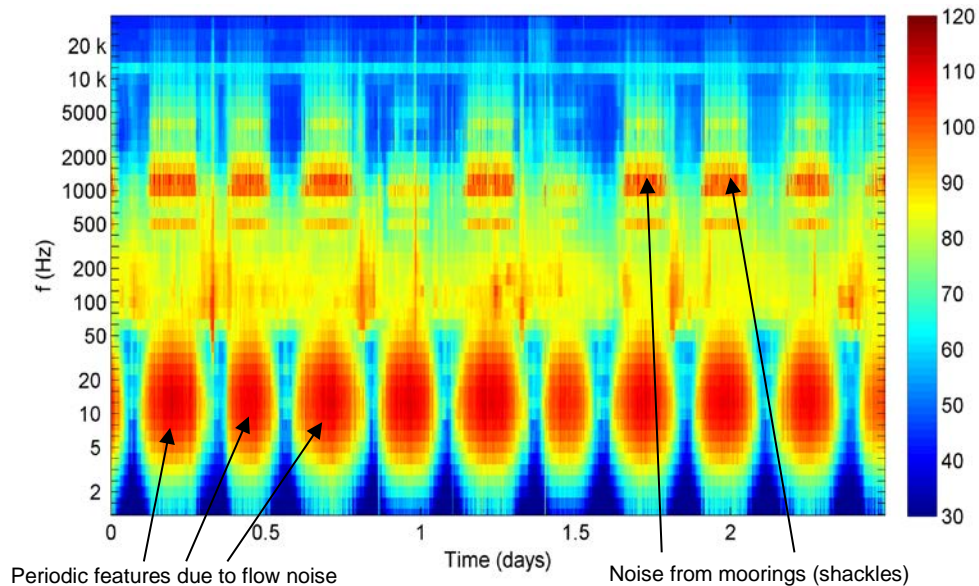

Figure C1 Spectrogram (third-octave band) of acoustic measurements of duration several days in UK waters. Note the periodic features present due to flow noise and other influences.
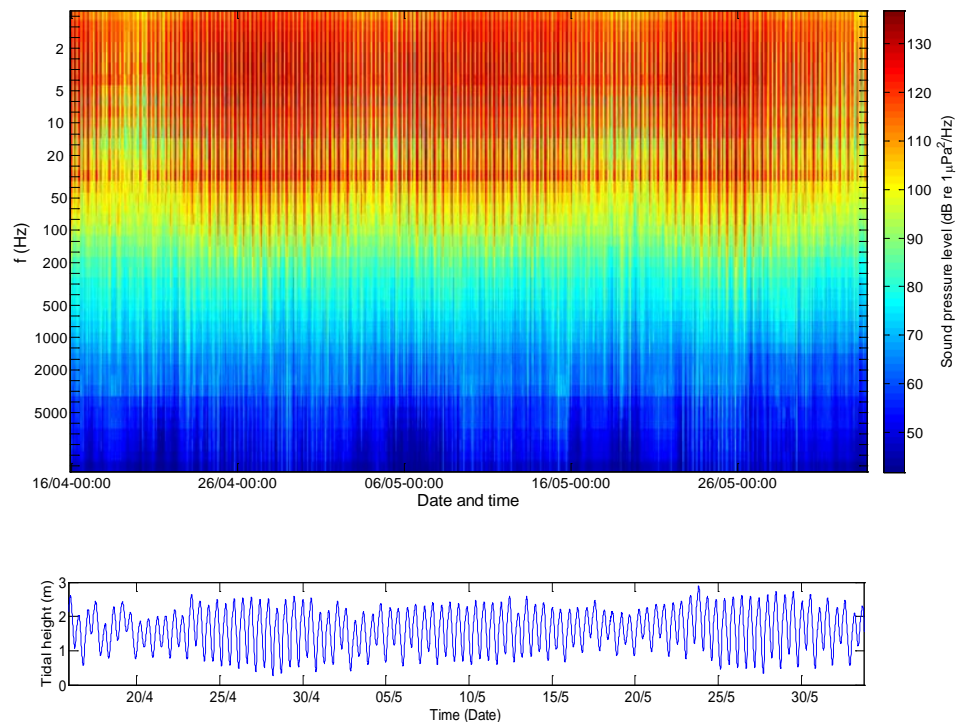

Figure C2 Spectrogram (third-octave band) of acoustic measurements of duration 15 days in UK waters and the tidal variation for the same location. Note the correlation in the levels below 50 Hz.

Annex D   Benchmarked data for comparison and validation of software

For the purpose of benchmarking the data analysis, files have been created using synthesized data with spectral known amplitudes that are known *a priori*. Two data files are made available in .WAV format are:

    signal_0_480.wav

    signal_1_480.wav

Both data files were created using a sampling rate of 48 kHz.
Both data files consist of data of duration 100 seconds.

The scale factors for the data files to be used to covert to voltage are:

    signal_0_480.wav   5.3121

    signal_1_480.wav   9.2669

The overall system sensitivity is -180 dB re 1 V/μPa (equivalent to 1000 μV/Pa). The sensitivity may be assumed to be invariant with frequency.

To perform the analysis on the files: read in the data in one second windows and for each data window calculate the SPL in each of the 34 one third-octave bands identified in Annex A. For each file, the results should be 100 SPL values for each of the 34 one third-octave bands. Return the processed data to NPL (for example in the form of a spreadsheet).

For the analysis, each partner should implement and use their analysis method of choice, but when choosing the method to use, consider the recommendations provided in Section 3.4 of this procedure.

5 References

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