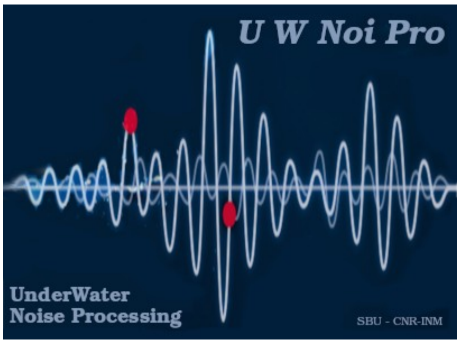

U W Noi Pro

UnderWater Noise Processing

USER MANUAL

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1. Revision history

Revision	Date	Description
5.05	8-mar-2024	First edition for standalone application
5.06	27-mar-2024	Changed to decidecade bands, added multichannel, minor fixes
5.06.1	5-apr-2024	Minor changes to text and format, added flowchart

2. Description

UWNoiPro – UnderWater Noise Processing – is a program for processing audio files in WAV format containing underwater sound recordings to compute any of the following output quantities used in standard analysis:

- Broadband Sound Pressure Level (SPL)
- Power Spectral Density (PSD), broadband (BB) and in decidecade bands (DD)

Processing is done in intervals (snapshots) of given time duration from a given start time (including beginning of file) to a given end time (including end of file). Results are presented as average and percentile plots computed over all snapshots. Current snapshot waveform is also showed during processing.

Input data are converted into absolute pressure in pascal using a user selected, frequency dependent calibration table. Processing is done on one or more input audio files contained in a user seleced input folder. Results are plotted in separate figures during processing, and may be saved to JPEG files. Numeric results may also be saved as text files. Program execution details may be logged to a text file. All output files are saved to a user selected output folder.

3. Input file requirements

Audio file(s): WAV format, single- or multi-channel (e.g. stereo), any sampling rate, any bit depth. Any file length may be processed compatible with system requirements, since file is only read piecewise one snapshot at a time and not altogether. Files in input folder whose names do not end with ".wav" or ".WAV" are ignored.

Calibration file: text format, one [frequency dB] calibration point per row, blank or tab separated. Lines beginning with any non-numeric character and blank lines are ignored. Minimum three points are required, at least partially covering the frequency range of interest. Values used for processing are linearly interpolated between available points, and extrapolated using last value as a constant. Values are meant to be frequency in Hz and sensitivity in dB re 1 V/µPa. Equivalent Scale Factor (SF) values in Pa may also be given instead of dB. The program detects either representation by reading first value: if negative, dB are assumed throughout, otherwise SF.

4. Setup

A graphical user interface (GUI, see above) pops up at program start, containing all parameters to be sent to processing functions for each input audio file. A description is given below following a top-bottom order which is the normal order for selecting options and parameters.

4.1. Processing options (not exclusive, may be selected in any order)

- Broadband SPL: sound pressure level in dB re 1 µPa as a function of time, one point per snapshot at midpoint times within each snapshot.
- Broadband (BB) PSD: power spectral density in dB re 1 µPa^2/Hz as a function of frequency over full band up to Nyquist frequency (0.5 * sample rate).
- Decidecade (DD) PSD: as above, computed in a series of decidecade frequency bands with nominal center frequencies selected between 1 Hz and 800 kHz.

Selecting DD, user can choose first and last band in a series, and can select boxplot as output plot instead of bullet plot. Selecting BB PSD, user can optionally add all lines for each snapshot in output plot (default is plot only average and percentiles), and add Knudsen curves for a range of sea states between SS0 and SS6.

4.2. Input / output file options

WAV files to be processed are read from input folder. Other files with extension else than ".wav" or ".WAV", if present, are ignored. No action is done if folder is empty.

Multichannel WAV files (e.g. stereo files) are read one channel at a time, and "(chN)" identifiers are appended to figure titles and output filenames.

Output folder collects results as JPEG files reproducing final plots, and TXT files containing corresponding numeric data in tabular form. Each filename contains the audio file name, date-time of creation, and an identifier of processing type. If selected, a logfile in text format is also created in the same folder, containing processing details, and optionally parameter setup. If a logfile is already present in output folder, further data are appended to it. Files are created in output folder only if corresponding options for JPEG, TXT, or logging are selected.

A valid calibration file must be selected to start processing: an error message box pops up to prompt user action to correct a missing or invalid file.

4.3. Defining time range, snapshot, FFT

Start and end time range for processing may be provided according to audio file duration. To process each audio file entirely, set Stop time = "inf". Note that a fixed stop time might cause an error if audio file duration is shorter.

Snapshot length defines the time interval over which processing is done. Snapshots may be consecutive (0 % overlap), overlapping (overlap > 0 %) and there may be a given skip time between each. For example, set 10 s length and 50 s skip to process only the first 10 s in a minute. Length, overlap and skip time all have an effect on the number of snapshots for a given time range.

FFT options include standard options for n. of points (in powers of 2), overlap, and windowing.

4.4. Averaging and percentiles

Averaging method (arithmetic mean or median) and percentiles (high from 50 % to 95 %, low from 5 % to 50 %, both in 5 % step) apply to statistics over total number of snapshots to produce average and percentiles curves plotted in final results. If boxplot option is selected, final plot only shows 25 % 75 % percentiles regardless of user selection.

4.5. Highpass filtering

Snapshots may be additionally highpass filtered prior to processing by entering a cutoff frequency in Hz > 0. Otherwise, only the DC component is removed. Filter characteristics are: 60 dB stopband attenuation, 0.1 dB passband ripple, transition band = 0.15 * cutoff f. Note that filtering may consideraby increase processing time especially for long snapshots.

4.6. Plot options

Titles and legends may be optionally applied to final plots. Titles include audio file names and major parameters. Options for drawing all BB PSD lines and Knudsen curves are described above in processing options. X and Y axis ranges may be set using fixed values in corresponding units or using autorange. Size of figures may be set to fixed width and height in pixels: in this case, beginning each run all figures are sized and positioned with a small tile offset near bottom left screen corner. During processing, or after it, user can freely move or resize figures across the screen. By selecting "Keep size & position", on next run all figure windows are kept in their current position with current size. When saving to JPEG files, each figure size is the same as the one on screen. In each figure window icons are provided for changing plot parameters interactively. Additionally, an “Info” window located near bottom left screen corner shows information on processing parameters that depend on current audio file characteristics. A variable pause may be added between plots to ease readability before each plot is refreshed with new data.

4.7. Logging

Time-tagged information may be logged to a text file saved in output folder, with options: Standard, Detailed, Debug. Standard option is adequate for most uses, while Detailed adds info on FFT, DD frequencies and intermediate results. Debug adds much more data and requires user action by pressing a key after each FFT plot, which is only recommended for testing on small datasets. The entire parameter setup in GUI may also be added to logfile. A single logfile is created for each folder, and all subsequent processing append further text to it.

5. Execution

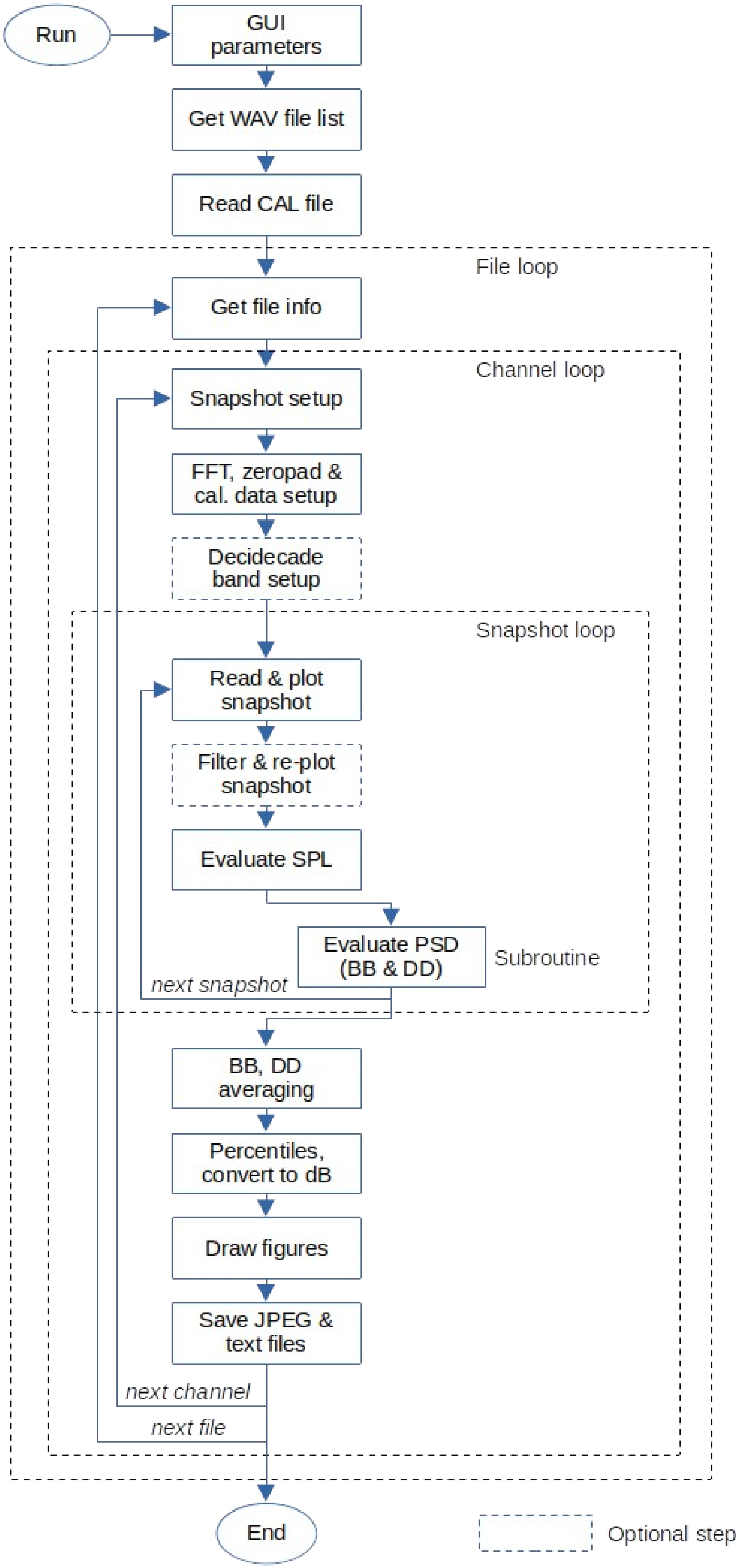
Once all options have been selected, press “Run” to start processing. From this point on, no further action is required until all files are processed. Visual and text indicators describe exit status: both refer to last error encountered during processing. In case of errors related to global parameters, execution stops before processing first audio file. If errors are encountered in one specific audio file in a set, execution stops but valid results may still be available from previous files.

Once ended, processing may be done again changing parameter setup as needed. Output will be on different files provided that next execution ends at a different time in [hh-mm] format, that is at least one minute afterwards: otherwise, output files will be overwritten.

To repeat processing with same setup on a different set of audio files, just change input folder or its contents. Output folder can be the same or it may change: note that there will be one logfile per output folder, with information on all output files in that folder.

To exit program, click the upper right corner icon in GUI window titlebar. Upon exit, all figure windows are also closed.

6. Flowchart



Run
GUI parameters
Get WAV file list
Read CAL file
File loop
Get file info
Channel loop
Snapshot setup
FFT, zeropad & cal. data setup
Decidecade band setup
Snapshot loop
Read & plot snapshot
Filter & re-plot snapshot
Evaluate SPL
Evaluate PSD (BB & DD)
Subroutine
next snapshot
BB, DD averaging
Percentiles, convert to dB
Draw figures
Save JPEG & text files
next channel
next file
End
Optional step