

# Avisoft-RECORDER

Version 4.2

**Sound recording and playback software for  
Microsoft-Windows Vista / 7 / 8 / 8.1 / 10**

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# CONTENTS

LICENSE AGREEMENT .....	2
INTRODUCTION .....	7
INSTALLATION PROCEDURE.....	9
GETTING STARTED.....	11
BASIC CONCEPTS.....	13
THE MENU STRUCTURE.....	17
Monitoring .....	17
Start  .....	17
Stop  .....	17
Pause  .....	17
Start/Stop (toggle).....	17
Freeze display (toggle) .....	17
Split .wav file(s).....	17
Log file date format.....	18
LOG file comment.....	18
LOG file comment text modules .....	18
Transfer last file to SASLab  .....	19

Ring buffer.....	19
Autostart .....	20
Autostart 120s delayed .....	20
Shut Down / Sleep time.....	20
Daily alarm actions.....	21
Email Error Report settings.....	23
Enable monitoring watchdog .....	23
Trigger level calibration / Full-scale SPL range.....	23
Options .....	27
Configuration...  .....	27
Configuration management > .....	27
File naming .....	28
.wav file split trigger settings... .....	28
USV monitoring setup wizard.....	28
Bat call trigger filter settings.....	28
Automated call measurements settings .....	29
dXML metadata input settings .....	29
Weather station settings... .....	29
GPS settings.....	32
Control output settings.....	35
Spectrogram settings.....	36
Display .....	36
Waveform display  .....	36
Spectrogram display  .....	36
Energy display  .....	36
Grid .....	36
Large buttons.....	36
Reset buttons on stop.....	37
Autotransfer all channels to SASLab.....	37
Enable Multichannel mode .....	37
Disable 'WAVE_FORMAT_EXTENSIBLE' .....	37
Enable soundcards / USB audio interfaces / multichannel mode	38
Keyboard shortcuts and popup menu.....	38
LOG-Files .....	38
Play.....	38
Play  .....	38
Start .....	38

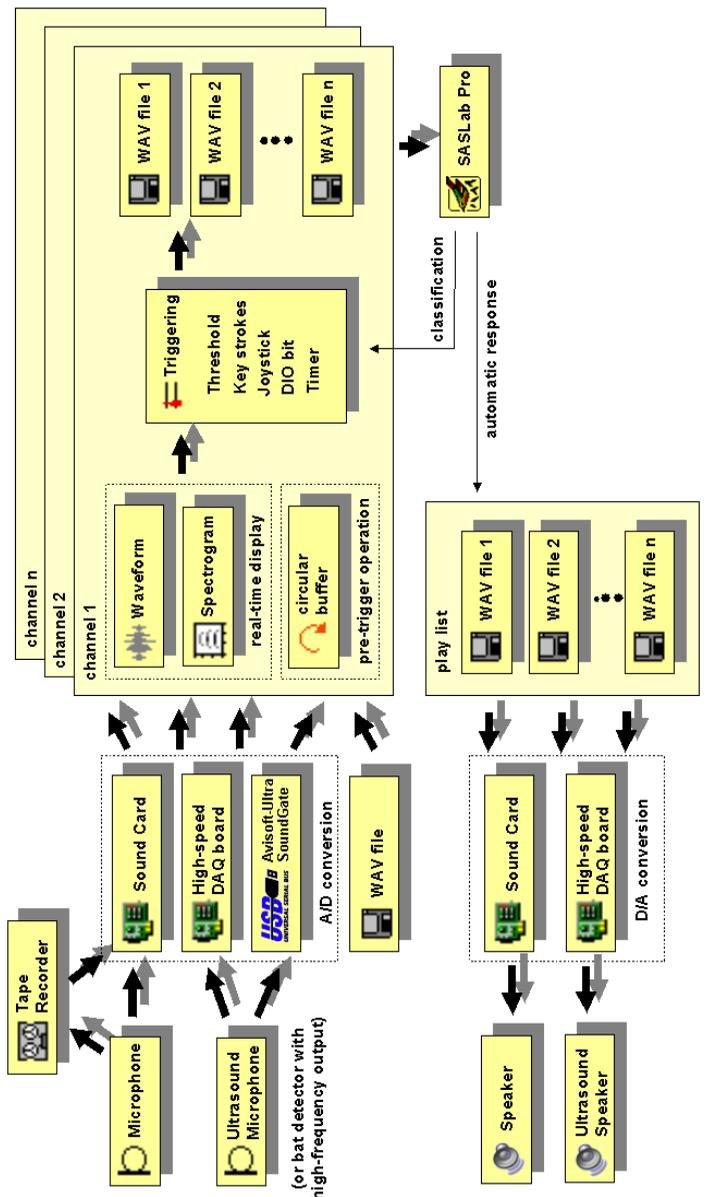
Stop 	38
Pause 	39
Loop 	39
Mute	39
Playback Volume	39
Playback Volume keyboard shortcuts	39
Settings	39
Device	41
File	41
LOG file	42
Reset LOG file	42
Playlist	42
Autostart	44
Show playlist entries on touch panel	44
Play playlist entry	44
<b>CONFIGURATION</b>	<b>45</b>
Channel Settings	46
Input Device Settings	58
Display Settings	59
Time Interval Definition	62
Monitor	64
Control Output settings	66
Advanced USGH Device Settings	69
<b>COMMAND LINE OPTIONS</b>	<b>70</b>
<b>DDE COMMAND INTERFACE</b>	<b>71</b>

<b>RECORDING AND ANALYSING RAT OR MOUSE ULTRASONIC VOCALIZATIONS FOR BEHAVIOURAL, NEUROPHYSIOLOGICAL AND PHARMACOLOGICAL STUDIES .....</b>	<b>73</b>
Getting started .....	73
USV Real-Time Monitoring .....	76
USV Real-Time Monitoring Setup dialog box .....	77
Batch mode for rat or mice drug testing .....	79
<b>BAT CALL TRIGGER FILTER .....</b>	<b>80</b>
<b>AUTOMATED CALL PARAMETER MEASUREMENTS .....</b>	<b>84</b>
Call Classification Settings .....	89
<b>DXML METADATA INPUT SETTINGS &amp; DATABASE FIELDS .....</b>	<b>90</b>
dXML metadata record .....	92

# Introduction

**Avisoft-RECORDER** is a versatile multichannel triggering harddisk recording system. It was designed for the special needs of bioacoustics, but it can also be used for any other audio signals. Possible applications are:

- Long-term monitoring and recording of acoustic events both in the laboratory and in the field.
- High-speed recording with sampling rates up to 1 MHz with various data acquisition devices (Avisoft-UltraSoundGate, National Instruments, MeasurementComputing, DataTranslation and ines).
- Playback with soundcards, UltraSoundGate Player and data acquisition boards by National Instruments.
- Pre-trigger recording of sporadic animal vocalizations in the field.
- Individual triggering and saving of single channels from multichannel soundcards or data acquisition boards.
- Multichannel spectrographic display.
- USV real-time monitoring for drug tests with mice or rats
- Recording front-end for **Avisoft-SASLab Pro**.



Principle of the RECORDER software in conjunction with various I/O devices

## Installation procedure

The RECORDER software can be installed from the supplied software installation CD or USB flash drive by launching the **setup.exe** installation program located in the subfolder **RECORDER** if you have purchased the universal RECORDER software (part #10301, 10201, 10201 or 20202) or **RECORDER USGH** if you have an UltraSoundGate that comes with the bundled RECORDER USGH software. The most recent software versions are available from the Avisoft website at [www.avisoft.com/download.htm](http://www.avisoft.com/download.htm).

The supplied USB Hardlock key should be plugged into computer after the installation has finished (not before). If you are using one of the Avisoft-UltraSoundGate devices, the Hardlock USB key is not required. In this case, the UltraSoundGate-specific software versions RECORDER USGH, (rec\_usgh.exe) or RECORDER USG (rec\_usg.exe) must be used.

In case the Avisoft-RECORDER should be used with 3<sup>rd</sup> party high-speed data-acquisition devices, the driver for the acquisition board must first be installed from the software installation CD provided by the device manufacturer.

There are various versions of the Avisoft-RECORDER software, each dedicated to a specific type of recording hardware:

**RECORDER (recorder.exe)** is the standard version, which supports soundcards and audio interfaces that comply with the Windows multimedia interface (MME, WDM, USB audio class driver).

**RECORDER 32 (recorder32.exe)** is equivalent to the above recorder.exe, but supports up to 32 channels.

**RECORDER USG (rec\_usg.exe)** supports the old Avisoft-UltraSoundGate models 116 and 416 manufactured before 2008 (with limited sample rate support on Windows 7/8/10). A Hardlock key is not required (*the UltraSoundGate device itself will act as a key*).

**RECORDER USGH (rec\_usgh.exe)** supports all Avisoft-UltraSoundGate devices under Windows Vista, 7, 8 and 10. A Hardlock key is not required (*the UltraSoundGate device itself will act as a key*). Note that it is also required to install the Avisoft-UltraSoundGate device driver (USGH.SYS) to run this software version.

**RECORDER USGH with NIDAQmx DIO support**

**(rec\_usgh\_nidaqdi.exe)** is equivalent to the above RECORDER USGH, except that it replaces the USG TRG input by a National Instruments DAQ DIO port. This software version requires that the NI-DAQmx driver has been installed.

**RECORDER NI-DAQ (rec\_ni.exe)** supports both soundcards and National Instruments E-series DAQ devices (Traditional NI-DAQ driver).

**RECORDER NI-DAQmx (rec\_nimx.exe)** supports both soundcards and National Instruments E, M and X-series DAQ devices with 16 bit resolution (NI-DAQmx driver).

**RECORDER CB (rec\_cb.exe)** supports both soundcards and Measurement Computing (formerly ComputerBoards) devices.

**RECORDER DT (rec\_dt.exe)** supports both soundcards and Data Translation (DT-Open Layers compatible) devices.

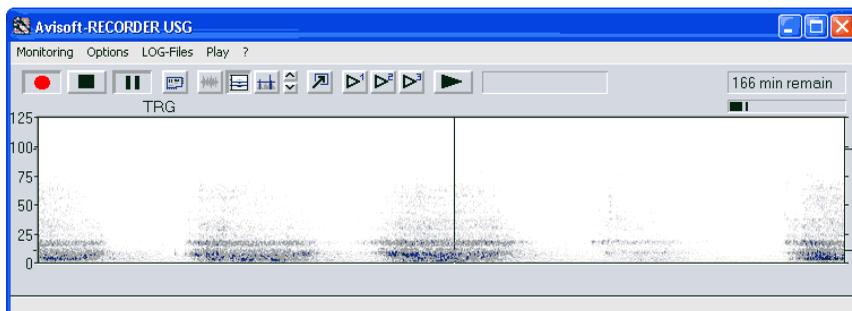
The device-specific versions recorder\_xxx are located in the *Avisoft Bioacoustics* program files directory. These programs will only run if the appropriate manufacturer-specific driver has been installed.

## Getting started

1. Install the soundcard, sound interface or data acquisition board that you intend to use with Avisoft-RECORDER.

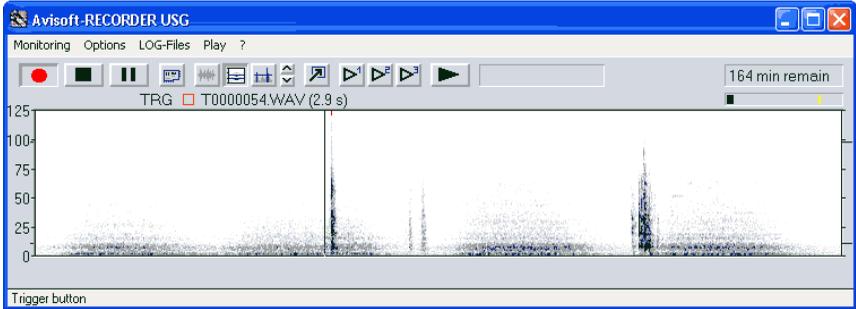
2. Configure *Avisoft-RECORDER* from the Options/Configuration...  dialog (for details see page 44). Please note that Avisoft RECORDER is based on a concept that is completely different from other common sound recording applications. Starting and stopping recording sessions can be controlled by a wide range of trigger events (simple keyboard or joystick button hits, exceeding a certain sound level threshold, timer, ...). In conventional recording applications a recording session can usually only be initiated by clicking at a button.

3. Start monitoring by the menu option Monitoring/Start  while saving of WAV-files is disabled (Stop button  is pressed).



4. Adjust the recording level of the soundcard by using the recording control and mixer software supplied by the soundcard manufacturer (or the software supplied by Windows). External sound interfaces usually have a physical recording level adjustment knob (e.g. Avisoft-UltraSoundGate). Adjust the recording level in such a way that there is no over-modulation (clipping). Clipping is indicated by red colour on the recording level bar, the waveform display and the spectrogram display. It is also important to prevent too soft recordings. Soft recording levels would waste the potentially available dynamic range of the recording hardware and might lead to poor signal-to-noise ratios.

5. Optimise the configuration settings, especially the trigger event settings by entering the Options/Configuration...  dialog again or by mouse tracking of thresholds and frequency ranges in the spectrogram and energy display. Once the triggering behaviour is working satisfying, enable saving of .WAV files by releasing the stop button .



6. Stop monitoring by the command Monitoring/Stop .

7. Check the acquired sound files that should have been saved in the base directory specified in the configuration dialog. Use *Avisoft-SASLab Pro* (version 3.5 or later) to navigate comfortably through these consecutive numbered WAV-files. Once a numbered WAV-file in the format T0000001.WAV has been opened in *Avisoft-SASLab Pro*, there will be two additional menu options for that purpose in the *File* menu: *Previous file*  and *Next file* . The menu option *Delete file* can be used to delete single files. All succeeding files will then be re-numbered.

## Basic concepts

Avisoft-RECORDER is a versatile software tool for various sound recording and playback tasks. This chapter describes its basic concepts.

While common sound recording/editing software applications usually require entering a filename for each newly acquired soundfile, Avisoft-RECORDER saves the soundfiles under subsequently numbered file names in the same way as in digital photo cameras or CompactFlash audio recorders. This automatic file naming concept enables the automated acquisition of sounds without any further user interaction.

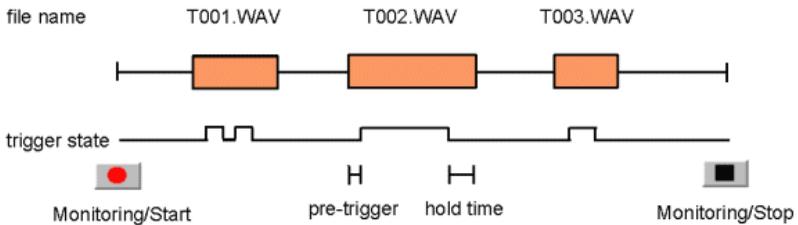
In contrast to other sound recording applications, Avisoft-RECORDER employs a basic monitoring mode in which the audio data are streamed continuously from the audio interface (sound card or data acquisition card) into a temporary circular data buffer. At the same time, the incoming sounds are visualized on the screen as oscillograms or spectrograms. At this stage, no data will be saved into any sound file and the process can run endlessly (for days or weeks). The actual recording procedure (saving the data stream into a sound file on the harddisk) is initiated by an internal logic trigger state. This trigger state can be controlled from various sources (listbox 'Trigger' on the configuration dialog box). Among these trigger sources are:

- instantaneous sound level of the incoming audio data (sound-activated)
- keyboard stroke
- joystick button hit
- interval timer
- time program (time table)
- digital input of the data acquisition board (or Avisoft-UltraSoundGate)
- permanent

The sound data will be written into the current numbered file as long as the logic trigger is active. If the trigger becomes inactive, the sound file will be closed and the next occurrence of a trigger will create a new (incremented) file.

The internal logic trigger signal is additionally modified by a pre-trigger interval and a hold time interval. The "Pre-trigger" parameter will shift the beginning of the sound file into the past. In this way it is possible to catch

sounds that occurred before activating the trigger (also known as 'pre-recording buffer'). Similarly, the end of the soundfile can be delayed by the specified 'Hold time':

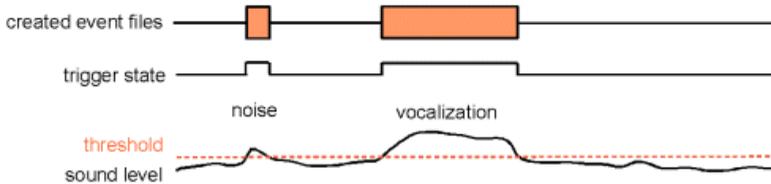


An exception from the standard triggered recording mode is the trigger type "permanent". In this mode of operation, the sound files will be created immediately after clicking at the red recording button. This behavior corresponds to common sound recording applications:

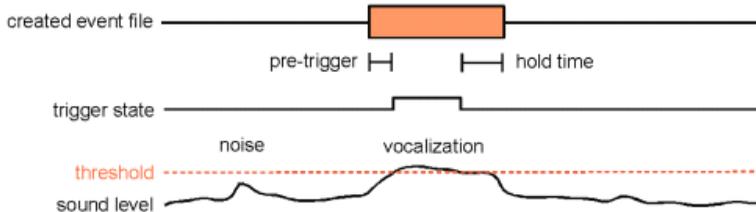


## Optimizing sound-activated recording

Under practical recording conditions it is often problematic to distinguish between environmental noise (e.g. cracks caused by the activity of the animals) and the desired vocalizations. So, it might happen that the trigger is also activated by that unwanted noise:



However, it is possible to minimize those false alarms by optimizing the settings of the Avisoft-RECORDER software. Assuming that the noise is softer than the vocalizations, the threshold should be set to a value that is well above the peak noise level. In order not to lose softer parts of the vocalizations at the beginning or end of a call or phrase, the pre-trigger and hold-time parameters should be increased accordingly:



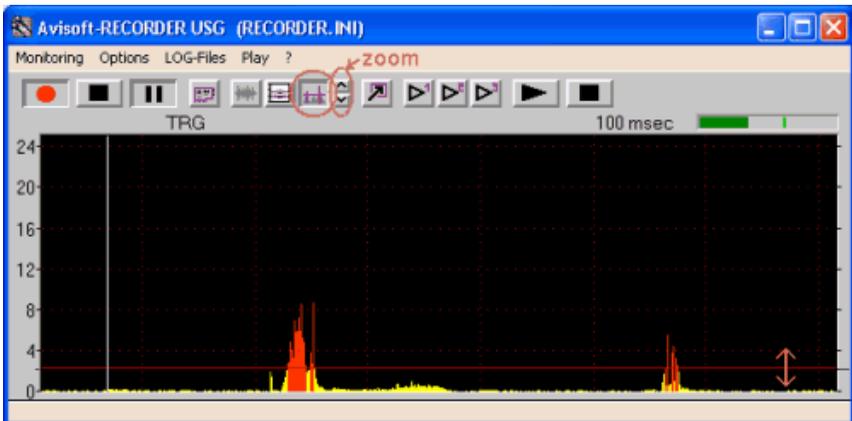
In case the vocalizations to be acquired have a harmonic structure (whistle-like sounds) and the disturbing noise has a broad-band structure, then it is possible to reject such broad-band noise by activating the Entropy threshold. In this way, even noise louder than the desired vocalizations would be rejected.

## Adjusting the threshold level for sound-activated recording

The threshold level for the sound-activated trigger mode (Trigger = "level of this channel") can be adjusted either by entering the desired level numerically from the configuration dialog box:



or graphically by mouse dragging on the "energy in f-range" real-time display:



# The menu structure

## *Monitoring*

### **Start**

Starts the monitoring process according to the settings made on the *Configuration...* dialog box. The activated soundcards or data acquisition boards will start writing data into the internal circular buffer. The system is waiting for trigger events.

### **Stop**

Stops monitoring.

### **Pause**

The pause button disables recording even if there are active trigger events. The current state of the trigger events assigned to the active channels will be displayed. This mode is useful for optimising the trigger event settings.

### ***Start/Stop (toggle)***

This command starts monitoring. When executing this command again, the monitoring will stop. This toggle mode may be useful for quick operation through keyboard shortcuts. The keyboard shortcut for this command is the Enter key.

### ***Freeze display (toggle)***

This command freezes the current display for observation and manual measurements. The recording process will not be affected. Use the keyboard shortcut (Esc) in order not to delete the current display. Executing this command again will continue the running display.

### ***Split .wav file(s)***

This command closes the current .wav file and immediately starts a new one.

## **Log file date format**

Selects the date format in log files: MM/DD/YY, MM/DD/YYYY, YYYYMMDD, YYYY-MM-DD, DD.MM.YYYY or DD.MM.YY.

## **LOG file comment**

Launches a dialog box for entering commentaries into the LOG file. The **channel** combo-box allows to select the channel to which the comment should be assigned.

The option **all** will write the comment to all active channels.

The option **Keep text for next entry** will retain the comment text for future comment entries. Otherwise, the previous text will be deleted each time the comment dialog box is launched.

The option **Hide this dialog box when using text modules** will reject this dialog box if one of the **LOG file comment text module** commands has been executed. In this way, the commentary procedure can be accelerated. Pressing the Ctrl key while executing any of the LOG file comment text module commands will temporarily disable the hide option, which will allow editing the text modules afterwards.

The option **Create .wav file label** will additionally create a label that is directly embedded into the .wav file, which can be viewed in SASLab Pro. Due to the label constraints, the text is limited to 30 characters.

The option **Create automatic playback comment** will create a comment each time when a .wav file playback starts. In order to get this work, the option “create playback log file” on the command “Play”/“Settings...” must be activated in the instance that performs the playback.

## **LOG file comment text modules**

These commands provide a set of user-defined text modules for entering frequently used comments. The individual text modules can be edited by selecting the one of the 12 commands in this sub menu, entering the desired text and then clicking at the button **Save as text module!** The default keyboard shortcuts (F1...F12) for these pre-defined comments can be changed from the command **Options/Keyboard shortcuts and popup menu...**

## **Transfer last file to SASLab**

Transfers the last file saved by *Avisoft-RECORDER* to *Avisoft-SASLab Lite / Pro* for displaying and post processing. If SASLab Lite / Pro is not already running, The software will first try to launch SASLab Lite. If this application cannot be found, SASLab Pro will be launched.

### **Ring buffer...**

The ring buffer (circular buffer) functionality will permanently save the incoming data of each device into a fixed-sized ring buffer on disk, which allows reading the data by other third-party (custom-made) applications in real-time. The ring buffer contains at any time the most recent data (it is something like a flight recorder), which could also be useful for some other applications.

The option **enable ring buffer on disk** will activate the ring buffer functionality. The buffer size list box determines the duration of the buffer expressed in Msamples. The option **indicate oldest sample** will mark the position of the oldest sample in the buffer by inserting two full-scale samples (32767 and -32768).

The ring buffer resides in a common .wav file (**ring.wav**, located in the RECORDER base directly) that has an additional custom “**RING**” RIFF chunk, which itself has three sub chunks:

**last** : Represents the write pointer index (expressed in bytes) relative to the first data sample in the WAVE data chunk

**loop** : The number of write pointer loops since the start of the monitoring

**offs** : The file offset address of the waveform data in the WAVE data chunk expressed in bytes from the begin of the .wav file

**size** : The total size of the ring buffer in bytes

For convenience, the information stored in the RING chunk is also regularly written into the **ring.ini** file, which might be easier to access by custom applications. This .ini file could be easily read by the *GetPrivateProfileString()* API function.

When polling and reading the files *ring.wav* and *ring.ini* be prepared that they can occasionally be locked (*FileOpen* fails) due to the write access by the RECORDER software. The files must be opened with the *READ* access attribute only and should be closed immediately after reading the data.

## ***Autostart***

When this option is checked, Avisoft-RECORDER will enter the monitoring mode immediately after the application has been launched. If the Avisoft-RECORDER has also been added to the Startup folder of the Windows operating system, the monitoring will start automatically after booting the computer. This feature might be useful to simplify the operation of the RECORDER software in the field.

## ***Autostart 120s delayed***

This option is similar to the above Autostart option, except that the start of the monitoring process is delayed by 120 seconds from the start of the operating system. Use this option if there are problems with the normal Autostart option (Windows might be sometimes too busy within the first few minutes after boot to reliably process the audio data streams).

## ***Shut Down / Sleep time...***

This dialog box allows defining a daily automated shut down of the computer at a certain time (list box option “**shutdown**”) or a daily sleep period (list box option “**sleep**”) in order to save power on a battery-operated recording system.

The option “**enable**” activates the specified shut down time or sleep interval.

In case the (desktop) computer supports a Real Time Clock Alarm (RTC Alarm), it would be possible to switch on the computer for a certain time period every day. The start time has to be entered from the BIOS setup (‘RTC Alarm’ in the ‘Power Management’ section). The shut down time can be entered from this *Shut Down / Sleep Time* dialog box in the RECORDER software. Additionally, the above *Autostart* option must be activated. In this way it would be possible to run the RECORDER software on a battery-powered (desktop) computer in the field for a limited period of time (e.g. 30 minutes) each day over several days without any inspection.

Laptop or tablet computers usually do not offer an RTC Alarm option in their BIOS, which prevents that kind of RTC alarm / shutdown operation. Most computers can however be put into a power-saving sleep mode that can also extend the battery life. By selecting the list box option “**sleep**”, the daily sleep period can be entered. The computer will be put to sleep at the “**from**” time and will be waked up at the “**until**” time. To make this work, it

is important that the power management options of the computer have been adjusted properly. For instance, the automatic sleep mode offered by the operating system should be disabled (“Put the computer to sleep” should be set to “Never”) and the mandatory input of a “password on wakeup” should be disabled (Control Panel > System and Security > Power Options). In order to enable the wakeup mechanism, make sure that wake timers are allowed (Control Panel > System and Security > Power Options > Edit Plan Settings > Change advanced power settings > Sleep > Allow wake timers > On battery : Enable). In addition to that the corresponding BIOS ACPI option (S1) must be activated.

The **advanced** option allows defining more complex activation patterns by means of a time program. If this option is activated, one of 8 time programs can be selected from the list box and the individual activation intervals can be defined from the **Edit...** button. See the chapter on the time table functionality (page 62) for further details.

The option “**pause**” is similar to the *sleep* option except that it deactivates the monitoring mode rather than putting the computer to sleep.

### ***Daily alarm actions...***

The Daily Alarm Actions dialog box allows to define a number of actions that should be executed each day at the same time. That **time** can be defined in the **Daily Alarm** section at the top of the window. Use the **Test run now!** button to test run the current settings. The option **run the alarm actions additionally on monitoring start** will execute the selected alarm not only at the select time of day, but additionally each time when the monitoring is started.

### **Alarm Actions**

When the option **log file reset** is activated, additional temporary log files will be created (titled xxx\_RECENT.LOG or calls\_RECENT.txt) that will be reset each day at the defined clock time. The option **keep old log files** will keep these daily log files by adding the current date to the filename (xxx\_RECENT.LOG will be renamed to xxx yyyy-mm-dd.LOG).

So, if the alarm time is set to the beginning of the daily recording period, these additional log files will always contain the data of the most recent recording period, which can assist the system maintenance (the files can for instance be sent regularly as an email attachment via the

sendEmail.exe application through the “*run a command line application*” option).

The **force trigger** option will trigger the recording at the programmed alarm time regardless of the specific trigger configuration (threshold).

The **play test signal .wav file** option allows to play a preselected .wav file (containing a suited reference signal) at the alarm time. Use the **File...** button to select the .wav file and use the **Play!** button to test run the playback through the selected audio playback **Device**.

**activate the USG signal generator (TRG out : button1)** : This option will activate the TRG out signal on the UltraSoundGate xx16H devices (through the button1 trigger), which can be setup in such a way that a dedicated test signal generator is activated for the specified period of time.

**measure the reference signal level xxx time(s) for yyy s** : This option will launch the **Trigger Level Calibration** dialog box for a predefined period of time in order to automatically measure the current signal level at each active channel. The resulting signal level (in percent of the full-scale level as well as in dBFS) along with the current time will be saved into a log file titled **reference.txt** that is located in the general **base directory** along with the recorded .wav files. The measurements can be repeated a number of times (listbox 1,3,5,9,17,...,513). The program will then subsequently perform the selected number of measurements and will take the median value of this data set as the final result.

The related option **adjust the trigger level relative to the reference signal once the deviation exceeds xxx dB for yyy days** will additionally execute a trigger threshold calibration for the first channel in case the new trigger level would differ from the existing one by the specified distance for the specified number of days (the *calibrate!* button on the *Trigger Level Calibration* dialog box is automatically activated at the end of the measurement procedure). In order to get this mechanism work, it is required that the desired trigger level threshold has been defined from the command “Monitoring”/”Trigger level calibration...”.

The option **run a command line application** allows executing a user-defined command line program for specific tasks that are not supported within the Avisoft-RECORDER software. This could be for instance a program for sending emails such as the free *sendEmail* application (<http://caspien.dotconf.net/menu/Software/SendEmail>). The complete path to the program as well as the required command line parameters can be

entered in the associated input field. Use the **File...** button to browse for the executable file and use the **Run!** Button to test-run the application with the specified parameters. See also <http://www.avisoft.com/Automatically%20sending%20log%20files%20via%20email.pdf> for further details on this. The button **create template for sendEmail.exe !** provides a starting point for configuring this functionality.

The last option **Move the recorded files to external drive** allows copying/moving the recorded sound files and log files to a different mass storage device, such as an external hard drive. That way it would for instance be possible to record during the day/night onto the internal drive while the external drive is in sleep mode or powered off in order to minimize the power consumption of the recording system. At the programmed alarm time, the files could then be moved at once to the (large) external drive.

### ***Email Error Report settings...***

This command allows setting up the software in such a way that an email is sent each time when an error is reported on the RECORDER status window. The button “*create template for sendEmail.exe*” provides a template for the sendemail.exe command line tool (see also the description of the above *Daily Alarm Actions* tool). Use the “*Test Run!*” button to test the current settings. Among the errors that are reported are USB data transfer errors (USG), data transfer timeout (TMOUT) and data storage errors (HDD). It is also possible to execute any other command line or batch program in response to an error event.

### ***Enable monitoring watchdog***

If activated, this option will automatically resume the monitoring mode after recovering from a recording hardware failure, such as an interrupted USB connection. In order to make this possible, all error messages will be rejected, which means that this option should be deactivated while debugging.

### ***Trigger level calibration / Full-scale SPL range...***

The Trigger Level Calibration tool allows to calibrate the trigger level by using a known (or otherwise fixed) reference signal. This kind of calibration is required for quantitative monitoring applications (at wind turbines for instance) that require well-defined and repeatable trigger settings.

In addition to that this dialog box allows to determine or to manually enter the full-scale dB SPL range for each channel.

The Trigger Level Calibration tool adjusts the trigger level (threshold) proportionally to an externally provided reference signal. If the absolute dB SPL of the reference signal is known, then it is also possible to calibrate the trigger level directly in dB SPL.

**Trigger level (threshold)** : If the option “*relative to reference signal level*” is selected, enter here the desired trigger level in dB in relation to the rms level of the reference signal. 0dB would set the trigger level exactly to the level of the reference signal. -6dB would for instance set the level to half of the reference level. In case the **reference signal level** is has been entered into the corresponding edit field and the alternative option “*absolute dB SPL*” is selected, enter here the desired trigger threshold in dB SPL.

**take max signal level** : If activated, the maximum registered signal level will be used for the calibration (“max hold” functionality). To reset the current maximum value of a particular channel, click at the “X” button.

The option **x10** corresponds to the same option on the *Configuration* dialog box and should be activated in low-gain setups that require triggering on low signal levels (it will be activated automatically if the desired trigger level is low compared to the standard resolution).

**toggle button1 / ref** : Controls the reference signal generator of the disk microphone models through the XLR pin 5 of the UltraSoundGate 116Hnbm.

The trigger level calibration should be executed as follows:

1. Setup the recording system according to the specific needs of the application from Options/Configuration. In case the recording hardware introduces a significant DC offset, then it is recommend to activate the *high-pass filter* on the Configuration dialog box. In order to get a visual feedback on the current signal and trigger levels, the *Display Settings* option “energy in f-range” should be activated.
2. Start the Trigger Level calibration tool from **Monitoring/Trigger level calibration...**
3. Adjust the gain of the recording hardware according to the specific needs of the application (in order to prevent clipping, select the lowest

gain that still provides sufficient sensitivity or an acceptable inherent noise floor).

4. Apply the reference signal to the microphone that you intend to calibrate. The rms level of this reference signal should then be shown and updated in real-time both in % (of the full-scale level or the measurement range) and dBFS. In case the absolute reference signal level [dBSPL] has been specified, the current (temporary) full-scale SPL will also be displayed. This display could also be used to adjust the full-scale range of the recording hardware (by turning its gain control knob) to a specific level (in this case, it would be appropriate to set the trigger level (from the Configuration dialog box) to a fixed value, such as 5% for instance).
5. Once the real-time display of the reference signal level has been settled after about 5 seconds (the time constant of the internal rms calculator is about 1 second), click at the associated **calibrate!** button, which will calculate the final trigger level (expressed in % or dBFS) according to the current reference signal level and the specified (relative or absolute) trigger level setting. At the same time, this % trigger level will be copied into corresponding edit field on the Configuration dialog box and updated on the graphic real-time display. Use the **inc** and **dec** buttons to adjust the display range of the real-time display.
6. In there are several channels, repeat the procedure for each channel (go back to paragraph 3 or 4).
7. Click Ok to close the calibration dialog box.

The full-scale SPL range calibration will be executed each time when **calibrate!** button is being clicked. In case a trigger level calibration is not required or desired, the **temporary full-scale SPL** can be copied to the **full-scale SPL range** edit field by clicking at the **>** button. If the full-scale range is already known, this value can alternatively be entered manually into the **full-scale SPL range** field. Once the **full-scale SPL range** has been determined, the numeric amplitude values shown on the real-time spectrum and level meter display (Configuration > Display Settings > Show spectrum (dBFS/dBSPL vs frequency) will represent absolute values in dB SPL. In order to get accurate SPL level measurements it is important to select the level meter mode “rms level” rather than “peak level” (dBSPL values are usually based on the roots mean square). However, in situations where averaged rms values would underestimate the signal level (when the duty cycle is less than 100%), it is recommended to use the peak level option (the difference between the rms and peak level is 3 dB in continuous sine signals).

Once the **full-scale SPL range** values have been defined, they will be saved automatically into to a custom-specific .wav file header chunk (titled "SPL ") of each recorded .wav file, which will be read automatically by the Avisoft-SASLab Pro software in order to display absolute sound level measurements based on the calibration done in the RECORDER software.

## Options

### Configuration...

This command launches the set-up dialog box. See page 44 for details.

### Configuration management >

This popup menu provides commands for managing configurations:

**Open...** : Opens a previously saved configuration (.ini file).

**Save** : Saves the current configuration.

**SaveAs...** : Saves the current configuration into a new .ini file.

**Reset** : Resets all settings to their defaults.

### Save mode on exit >

#### Save current configuration automatically

If this option is checked, the current configuration will be saved automatically each time the software is closed down.

### Prompt

If this option is checked, the current configuration can optionally be saved each time the software is closed down.

**Presets** : Adjusts the configuration settings to various specific applications such as **Bat Monitoring** in the field, **USV** recording in laboratory rodents or **Bird Song** recording. The Bat Call Playback preset creates a playlist of the .wav files that are currently located in the folder *<user>/Documents/Avisoft Bioacoustics/Batcalls*.

By default, the configuration files (\*.ini) reside in the folder *<user>/Documents/Avisoft Bioacoustics/Configurations/RECORDER*.

Note that the current file numbers are also saved in these .ini files. So, take care of these numbers because this might lead to an undesired overwriting of previous sound files.

**Copy configuration file on Monitoring>Start** : If activated, this option will copy the current configuration file into the .wav file folder (into a subfolder titled "configuration YYYYMMDD HHMMSS") each time when the command Monitoring > Start is executed. In this way it is possible

to track down the settings that have been used for the individual monitoring sessions.

### ***File naming***

See page 48.

### ***.wav file split trigger settings...***

This dialog box allows defining a trigger signal that will split the current .wav file by closing the current one and immediately creating a new one. Select here the desired *trigger* that should initiate the splitting.

The *maximum file size* option corresponds to the same edit field on the *Configuration* dialog box, which will also split the current .wav once the defined time interval has elapsed.

### ***USV monitoring setup wizard...***

This command will assist you in setting up the *USV real-time monitoring* tool (see page 76 for further details).

**trial duration** : Enter here the desired duration of the trial.

**number of channels** : If you have multichannel recording hardware, enter the desired number of channels here.

**run from .wav file** : Check this option if you want to repeat the USV measurement on a previously recorded .wav file in off-line mode. You will then be prompted for the desired .wav file.

**batch mode**: This option activates the batch mode for rat/mice drug testing. See page 79.

**detection & report settings Setup...**: Launches the *USV Real-Time Monitoring Setup* dialog box for adjusting the USV detection and report settings.

**recording device settings Setup...**: Launches the *Configuration...* dialog box.

Finally click at the **Start** button to start both the .wav file recording process and the real-time USV monitoring.

### ***Bat call trigger filter settings...***

See page 80.

## ***Automated call measurements settings...***

See page 84

## ***dXML metadata input settings***

See page 84.

## ***Weather station settings...***

The weather station functionality allows to read weather data from a weather station that is equipped with a suited computer interface. The current weather data are written directly into a custom .WAV file metadata chunk (“dXML”) that can later be viewed and analyzed in the Avisoft-SASLab Lite / Pro sound analysis software.

Currently, the ***Kestrel 4xxxx Weather Tracker*** series and ***Davis Instruments Vantage Pro2*** devices are supported. Additionally, a number of NMEA-based weather/GPS stations (Airmar WS-LBxxx / PBxxx, Navilock NL-120 / Wintec WGS-1000) are supported through the GPS receiver interface, whose weather data acquisition can be activated from the GPS settings dialog box (***Options/GPS settings.../read NMEA weather data sentences***). Once this GPS option has been activated, the **Model** selection on the *Weather station settings* dialog box is inhibited.

The weather data are stored into user-defined dXML database fields whose names must exactly match the dXML fields listed on this dialog box.

**Enable weather station communication:** Activate this option to use the weather data logging feature.

**Model:** Select here the weather station model that you intend to use:

- **Kestrel 4xxx Weather Meter** : Kestrel pocket weather meter with computer interface (external USB / serial interface or a 4xxxBT model with integrated Bluetooth interface is required)
- **Davis Vantage Pro2, Vue, Envoy** : Davis Instruments weather station with computer interface (WeatherLink data logger with USB or serial interface is required). Note that the USB data logger model must be configured to emulate a serial COM port (see the WeatherLink manual on how to ***Convert USB to Virtual Serial***)
- **Generic real-time log file** : This option will continuously read the weather data from a log file of a third-party weather data logging software, which will allow to import weather data from a large number of different weather station models. The location of the log

- file must be specified on the **log file** field.
- **Heavy Weather, KlimaLogg Pro:** La Crosse / TFA Dostmann weather stations with Heavy Weather or KlimaLogg software. In this mode of operation, the RECORDER software reads the current weather data from the *currdat.lst* or *KlimaLogg.dat* file that is regularly updated by the Heavy Weather / KlimaLogg software. The data file containing the current weather data is usually located on the folder *C:\ProgramData\*. Click at the button **HeavyW** or **KlimaLogg** to set the default filenames of these applications.

**Weather station communication settings:** Enter here the serial **COM port** settings for the weather station. The **Boud rate** must match the settings of the unit. Use the **Scan ports!** button to examine the settings automatically. The **time out** field determines after which time the weather data measurements are considered outdated when no new data are received from the weather station.

If the **Generic real-time log file** model option has been selected, the log file of the third –party weather station software must be specified in the field labeled **log file**. Use the ... button to browse for the file or drag it into the dialog box. In case the name of the current log file changes with time (from day to day), use wildcard (\*) characters to identify the files. The **delimiter** field defines how the individual data fields are separated. If there are space (0x20) or tabulator (0x09) characters required use the strings <SP> and <HT>. In order to identify the location of the weather data in the log file, it is necessary to define the index of the measurements. Enter these positions into the corresponding fields behind the measurement titles. The additional **position of the time stamp or line number** identifies the field that contains the time stamp or current number of the data set that always changes from one row to the next. Use the buttons on the **presets** section to retrieve template settings for a number of common weather station software packages:

**TEMPer** : HID TEMPer USB thermometer (<http://pcsensor.com>)

**WSDL** : Weather Station Data Logger for Oregon Scientific WMR-xxx (<http://wmx00.sourceforge.net>)

**Cumulus** : Cumulus for Oregon Scientific WMR-xxx and La Crosse 23xx (<http://sundaysoft.com>)

**WDisplay** : Weather Display (<http://www.weather-display.com>)

Note that the corresponding weather station software must be launched and its data logging mode must be activated in order to take it work.

**station** : Select here the desired station from which the temperature and

humidity measurements should be taken. When using the KlimaLOGG Pro software, select here the option “extra station 1”.

**station2** : Select here an additional (Davis) station that should acquire a second temperature (*temperature2*) and/or humidity (*humidity2*) value. This list box will only be enabled when the *temperature2* or *humidity2 measurements* options have been selected.

**measurements (dXML database fields)** : Select here the weather data that should be logged. On the Kestrel devices, the units we be determined by the settings made on the device. On the Vantage Pro2, it can be selected between metric and U.S. units (use buttons on the **set units** to section). Use the **Select all!** button to select all measurements that are available on the selected weather station model.

**Launch dXML setup dialog box** : Check this option in order to create the appropriate dXML database fields for saving the weather data into the individual .wav files. It will launch the **dXML metadata input settings & database fields** dialog box with the newly created weather data fields once the **OK** button is pressed. In case the required data fields have already been defined, this option can be deactivated.

**Weather log file** : Activate this option to save the weather data continuously into a separate log file (weather.xxx) that is located in the base directory that has been defined on the *Configuration* dialog box. The option **create a new log file each day** will append the current date to the filename, which will provide individual log files for each day. The **log file format** list box selects the desired format of the log file (ASCII .txt, .html or .xml). The sample interval determines the  
at which the data samples are stored. Note that this does not affect the internal update rate of the weather station unit.

As long as the RECORDER software receives current weather data from the weather station, the corresponding dXML field names will be displayed in **green** letters. If there are no current weather data for more than 5 seconds, the field names will change to **red**.

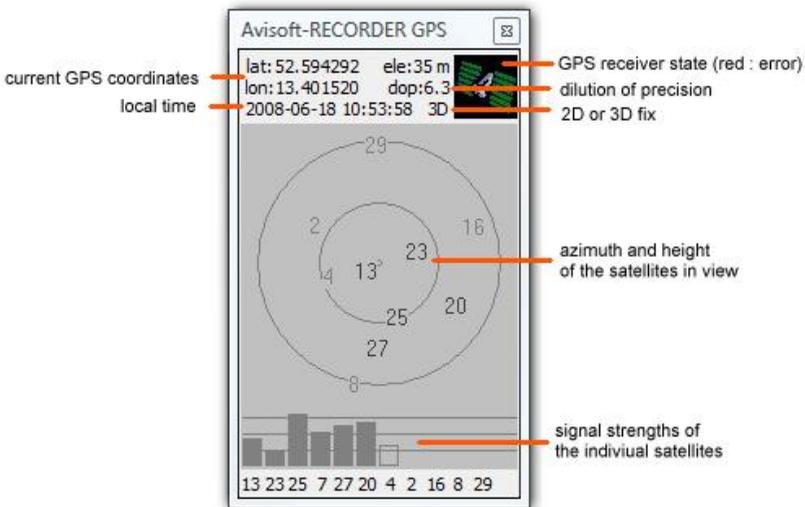
In case the **Create bext .wav file header chunk** option has been activated the from File Name Option dialog box, the weather data will also be saved into bext .wav file chunk.

## GPS settings...

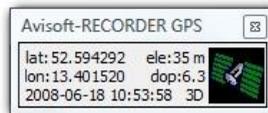
The GPS option allows to attach any GPS receiver complying with the NMEA 0183 protocol (either a Bluetooth GPS receiver, a USB GPS mouse or a handheld GPS unit) to the PC in order to simultaneously acquire the associate geographic locations of the recorded .wav files. The collected geographic coordinates are saved into the .wav file header, the ASCII-formatted RECORDER LOG file and optionally into common .gpx or .kml files that can be imported into GIS applications such as Google Earth.

The GPS Setting dialog box can alternatively be launched by right-clicking at the **Avisoft-RECORDER GPS** status window.

**enable GPS receiver communication** : Activate this option in order to enable the serial communication with the GPS receiver. If activated, the **Avisoft-RECORDER GPS** status window will be launched. This window displays the current GPS coordinates, the current date and the local time:



The *Avisoft-RECORDER GPS* status window can be minimized or enlarged by left-clicking at it:



If activated, the option **use the Windows Location API** option will retrieve the location data directly from Windows Location API rather than reading the information through a (virtual) COM port. This requires at least

Windows 7 and a GPS receiver that supports the Location API. Use this option on Windows 8 / 8.1 / 10 tablet computers with inbuilt GPS.

### ***GPS receiver communication settings***

The communication with the GPS receiver takes place through a (virtual) serial COM port of the PC, which must be identified here. Note that Bluetooth and USB-based GPS receivers also communicate with the PC via a virtual COM port (they therefore require to install an appropriate USB-to-Serial Com Port driver).

**COM port** : If the com port number is already known (for instance from the Windows Device Manager), then select here the port number to which the GPS receiver is connected.

**Boud rate** : Select here the boud rate at which the GPS receiver sends its data. This information can be found on the manual or specification sheet of the GPS receiver.

**Scan ports!** : If either the COM port number or the boud rate is unknown, use this button for detecting the proper COM port settings for the GPS receiver automatically. Depending on the number of available COM ports, this procedure can take up to several minutes.

**show detailed GPS satellites status** : This option activates the satellites status display that indicates the signal strengths of the currently used satellites. This display option can also be toggled by left-clicking at the ***Avisoft-RECORDER GPS*** status window.

**read NMEA weather data sentences:** This option will additionally parse NMEA weather data sentences that are provided by a number of GPS receivers. Currently supported are the Airmar/Furuno PBxxx/LBxxx WeatherStations (\$WIMDA) and Navilock NL-120GR / Wintec WSG-1000 (\$IISMD) GPS receivers. If this option has been activated, the ***Weather Station Settings*** dialog box will be launched afterwards, which allows to select the measurements to be logged. The ***dXML metadata input settings*** dialog will then activated the corresponding dXML date fields. In order to get both the weather and GPS data from the Airmar weather station devices, only the following NMEA sentences should be activated: GSA, GSV, MDA, RMC (GGA, which provides the elevation must be disabled in order to satisfy the bandwidth constraints).

### **.wav file position/tracklog while recording**

This option will additionally save the GPS positions of the recorded .wav files into separate log files.

**log file format** : Chose either the GPX Exchange Format (.gpx), the Google Earth Format (.kml) (also referred to as “Keyhole Markup Language”) or the ESRI Shapefile Format (.shp including .shx, .dbf, prj). The shapefile option will also save the optional dXML data set (including weather data).

**filename** : Enter here the filename of the recording log file. The path name for the log file is always the **Base directory** selected from the RECORDER Configuration dialog box.

**add date** : This option will append the current date to the filename.

**create individual log files for each .wav file** : If this option is activated, an individual .gpx/.kml log file will be created for each recorded .wav file. The log file name will then be derived from the corresponding .wav file name (same path and filename, except of the file type extension).

**.wav file tracklog** : This option will create a track representing the path of the .wav file if the distance moved from the start until the end of the sound file is longer than the threshold specified on the following option:

**Create track if distance > xxx m** : If the distance moved from the start until the end of a sound file is shorter than this threshold, no track entry will be created. Enter here a distance that is longer than the current GPS position variation in order to prevent generating undesired track entries in the log file while the microphone position is actually fixed.

**add a link to the .wav file** : If activated, this option will add a direct link (href="C:\Users\...\T00001.wav") to the .wav file into the waypoint data tag (only available for .kml files). An alternative pathname can be defined from the path name edit field. The local pathname of the recording computer will then be replaced by this string (it can also be an URL pointing to a website).

### **Continuous tracklog while monitoring**

This section allows to specify whether an additional log file should be created for complete monitoring sessions. Use this option for surveys in which the .wav files are being recorded in a trigger mode. In this way it is possible to document the path moved while no sound file was being recorded.

**log file format:** Select here the log file formats .gpx, .kml, .shp or .txt. The last option “NMEA 0183 Format (\*.txt) saves the original NMEA data stream as received from the GPS receiver. The shapefile option will also save the optional dXML data set (including weather data). Kml files can be

created either with the conventional <LineString> format (that does not include time stamps) or alternatively with the more recent **MultiTrack** format that is support in Google Earth 5.2 or later.

**filename** : Enter here the filename of the monitoring log file. The path name for the log file is always the **Base directory** selected from the RECORDER Configuration dialog box.

**add date** : This option will append the current date to the filename.

### **tracklog options**

**resolution xxx m**: The specified distance determines the resolution of the created tracklogs. Each time the current GPS position has changed at least by this amount, a new track point entry will be added to the log file. Smaller values will create larger log files.

**insert additional points at xxx sec intervals** : If activated, this option will add additional waypoints to the tracklog at the specified time interval. Use this option for an easier navigation through .wav files that have been recorded on the move (on the .wav file tracklog) or to create regular samples on the continuous tracklog file.

**Default!** : The default button resets all settings to their defaults.

In case the **Create bext .wav file header chunk** option has been activated the from File Name Option dialog box, the GPS position data data will also be saved into bext .wav file chunk.

### **Control output settings**

See page 66.

### **Create labels from digital input events**

This dialog box allows defining digital input events that should be saved a .wav file labels, which can later be viewed in SASLab Pro.

To activate this mode of operation, the option **enable creation of labels** must be activated.

**digital input** : Select here the digital input (or the TRG signal on RECORDER USGH) that should be used. The last option “DIO bits 0...7” will create individual labels for each bit.

**event**: Select here the slope or state of the digital input that should create the label. The options rising and falling slope will create single point labels, while the other two options low and high state will create section labels. The accuracy of the label time stamps is limited by the duration of the input device buffer.

## ***Spectrogram settings...***

This dialog box allows adjusting the real-time spectrogram display settings similar to the *Overview Display Parameters* dialog box in SASLab Lite/Pro.

### ***Display***

#### ***Waveform display***

The waveform of each active channel will be displayed.

#### ***Spectrogram display***

The spectrogram of each active channel will be displayed. In this display, the trigger event frequency interval can be modified graphically by mouse. Additionally, a cross-hair cursor allows to measure frequency and time. For time measurements left-click at the start point and drag the cursor to the end point.

#### ***Energy display***

The energy within the pre-defined trigger event frequency interval of each active channel will be displayed. The trigger event threshold can be changed graphically by mouse.

### ***Grid***

This option will add a grid to the above displays.

### ***Large buttons***

This option will enlarge the buttons in order to better support touchscreens.

## **Status display**

This option will launch a status window that show a number of general status information items (current *file number*, recent *number of files*, recent *duration*, *free storage*, *errors*, ...). At the programmed *Daily Alarm* time, the current content of this window will be saved into the logfile <base directory>/STATUS.TXT, which can be sent as an email via the sendEmail program through the *command Monitoring/Daily alarm actions...* Double-right click at the status window in order to reset the *recent files* and *error* entries.

## **Navigation**

These commands serve as keyboard shortcuts for various navigation commands.

### ***Reset buttons on stop***

This option automatically resets the trigger buttons1...3 and the 'channel buttons' when the monitoring is stopped. Otherwise, these buttons will keep their current state.

### ***Autotransfer all channels to SASLab***

If this option is activated, all active channels will be transferred to Avisoft-SASLab Pro. This option overrides the channel-specific option "Open in SASLab" on the Configuration dialog.

### ***Enable Multichannel mode***

In order to access all available channels in multi-channel soundcards (interleaved single data stream mode), this option must be activated

### ***Disable 'WAVE\_FORMAT\_EXTENSIBLE'***

There are different ways for accessing a soundcard. Avisoft-RECORDER uses the format WAVE\_FORMAT\_EXTENSIBLE with a PCM subtype by default. Some soundcard drivers may not be able to process that format correctly. So, if you are encountering unusual effects with a specific soundcard model (e.g. mystery clipping to 50 % or missing signals in 24 bit mode), then disable WAVE\_FORMAT\_EXTENSIBLE by checking this option. Avisoft-RECORDER will then use the simpler format WAVE\_FORMAT\_PCM instead.

## ***Enable soundcards / USB audio interfaces / multichannel mode***

Activate this option in order to access common audio interfaces for both recording and playback or to enable multichannel capabilities in conjunction with single-channel UltraSoundGate device. This option is only available on the Avisoft-RECORDER USGH version.

## ***Keyboard shortcuts and popup menu***

This command allows to define the keyboard shortcuts for all menus. Additionally, each command can be added to the popup menu, which is activated by right clicking on the application window. To modify a keyboard shortcut, first select the command on the list and then activate the desired shortcut on the bottom of the dialog box. In order to simplify the access to frequently used commands, it is recommended to use appropriate shortcuts or popup menu entries for those commands.

## ***LOG-Files***

The log-files of the channels will be displayed in the notepad application.

## ***Play***

***Play*** 

This menu provides various commands for playing .WAV files.

## ***Start***

Starts playback of the currently selected .WAV file. If no .WAV file has been chosen before, you will be prompted for a sound file name.

***Stop*** 

Stops a running playback session.

## **Pause**

Stops the current playback until the pause command is executed again. The playback will then continue at the file position at which the playback has been stopped.

## **Loop**

When this option is activated, the selected file will be played in a loop until the Stop command is executed.

## **Mute**

Sets the playback volume temporarily to zero until the command is executed again. This command can alternatively be controlled from the check box on the left hand side on the *Playback Volume* dialog box.

## **Playback Volume...**

This command displays a playback level meter and allows adjusting the playback volume. The horizontal slider and the **+** / **-** buttons modify a digital gain that is applied to the (digital) samples before they are transferred to the playback hardware. The default setting of 0 dB does not change the original sample amplitude. Note that depending on the dBFS level of the samples in the sound files, a gain setting larger than 0 dB may lead to digital clipping. The checkbox on the left hand side of the *Default* button will automatically launch the Playback Volume box once a sound file is being played.

On a stereo file, the (default) **ganged** option will adjust both channels simultaneously.

## **Playback Volume keyboard shortcuts**

These command are intended for adjusting the playback volume through keyboard shortcuts (Ctrl+Up Arrow, Ctrl+Down Arrow, ...)

## **Settings**

The Playback settings dialog box allows to set several options that will influence the playback feature.

## **the play button should play the :**

**selected file** : This option will play the selected file (from Play/File...), each time the play command is executed. This is the default setting.

**last recorded event file** : This alternate option will play the most recently recorded event file, each time the play command is executed. Please note, that the play command will fail as long as the current recording has not finished.

**automatic echo** : If this option is activated, each recorded sound event file will be played back automatically after the file has been closed.

**disable triggering while playing a file** : Activating this option will disable all triggers while a playback is running. This may prevent undesired feedback conditions.

**ext start trigger** : This list box provides external triggering options for playback. The selected option will either execute the Play/Start command (if the playlist window is not launched) or play a playlist entry if the playlist dialog box has been launched. Note that not all of the listed options are functional when the (recording) monitoring is not running. The “!” option will negate the state of the selected trigger source.

**ext mute control** : This list box provides external signals for controlling the command Play > Mute. The “!” option will negate the state of the selected trigger source.

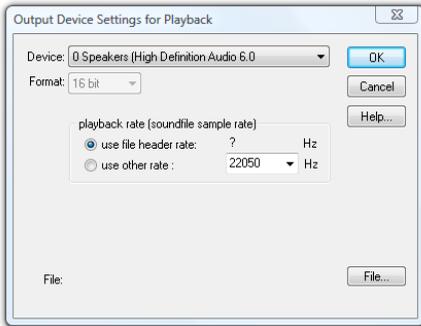
**ext pause control** : This list box provides external signals for controlling the command Play > Pause. The “!” option will negate the state of the selected trigger source.

**create playback log file** : If activated, a log file titled playback.log containing the time stamp and the file name of the played file will be created in the folder User/My Documents/Avisoft Bioacoustics.

**seamless loop / playlist mode** : If activated, the loop playback mode (*Play / Loop* option) and the playlist will be performed seamlessly without any gaps between the end and the start of the file(s). On the playlist, there will be no gaps between consecutive files if the “delay” option is not activated. In addition to that, the playlist files must all have the same format (sample rate, number of channels and so on).

## ***Device...***

This dialog box allows selecting the waveform output device and the playback rate of the .wav files to be played. The playback rate can be either the original sampling rate taken from the .WAV file header (option 'use file header rate') or a different rate specified here (option 'use other rate'). This feature can be used to modify the playback rate (slow-down or speed up) of the sound files.



For National Instruments high-speed DAQ boards, the data format for the DAQ output device can be specified. For E-series devices as the DAQCard 6062E, the option '12 bit int' should be used. This setting will automatically convert the 16 bit wav file data into the 12-bit format required by that card model (each sample will be divided by 16). The option '16 bit int' should be used if no format conversion is desired (in case you provide .wav files containing 12 bit data with integer values ranging from -2048 to 2047).

For UltraSoundGate Player devices, the RECORDER USGH software additionally provides a 'real-time oversampling' feature that allows to play sound files with sample rates that do not match the sample rate(s) supported by the UltraSoundGate Player hardware. See the UltraSoundGate Player manual for details.

The File button selects a .WAV file for playback.

## ***File...***

Selects a new .wav file from a FileOpen dialog box and starts playback immediately. Alternatively, files can be played back by drag and drop into the Avisoft-RECORDER application window.

### ***LOG file...***

This command launches the playback log file (playback.log). The playback log file records all played wav files along with the start time stamp.

### ***Reset LOG file***

Resets the playback log file (playback.log).

### ***Playlist...***

The play list command allows to playback sequences of pre-defined .wav files at once. Play list entries can simply be added by dragging sound files from the Windows folders into the Playlist panel. Play lists can be saved as ASCII text files for later re-use.

When invoking the Playlist command from the Play menu, a separate Playlist panel will appear. First you should add the desired .wav files to the list. This can be made either by drag-and-drop technique or by clicking at the **Append** button. The **Insert** button will insert the new file before the selected entry. The selected entry can be removed from the list by the **Delete** button. The **Delete all** button will remove all entries.

When using the drag & drop technique, the new files will be appended at the bottom of the list, when you release the mouse button over the area outside the file list (or below the last entry). By releasing the mouse button over an existing file on the list, that new file will be inserted before the existing one.

The currently selected list entry can be moved by using the **Up** and **Dn** buttons.

Complete play lists can be saved into ASCII files by clicking at the **Save** or **Save As...** button. It is recommended to use the .txt file extension. Subsequently, these files can be loaded by using the **Open...** button or simply by drag-and-drop into the playlist window. These .txt files may also be edited in an external text editor. The single sound files must be separated by CR/LF.

Playback of files on the list is started by clicking the play  button. Playback will start at the currently selected file. The stop  button will cancel the playback procedure.

With the option **stop after each file** activated, playback will stop after finishing each file. The next file on the list will be played once the play

button is clicked again. If this option is used for triggering playback of large numbers of files, the space bar may be used to initiate playback (the very first shot must be started by clicking at the play button).

The **loop mode** option will continue playback on the top of the list, each time the end of the list has been reached.

The option **randomize** will produce random playback sequences.

The option **delay** will introduce a break after each playback. Use the button **<-set** to define that delay. The appearing dialog box has an option labeled **random delay**. If this option is not activated, there will be a fixed delay between consecutive playbacks as defined in the field **t =**. Otherwise, there will be a random delay between playbacks. These random delays will be distributed between the two margins specified in the fields **t =**.

The **timetable** option activates a different mode of operation in which the sound files in the playlist will be played at certain (absolute) start times. These start times can be defined in a number of different formats:

**relative to start** : relative to the time when the play button is pressed

**absolute time** : absolute clock times (hh:mm:ss)

**relative to sunrise** :

**relative to sunset** :

On the last two options, the geographic coordinates (latitude and longitude) must be entered in decimal degrees.

The option **hh:mm:ss** selects the hh:mm:ss time format instead of seconds. To define the start times for the individual files in the playlist, the each individual file must be selected and the desired start time is entered from the edit files at the bottom of the file list.

The **absolute time, relative to sunrise / sunset** options will continue to play the selected files each day at the defined times until the stop button is pressed. Note that the time intervals between two consecutive files must be longer than the file duration. In the timetable mode, the elapsed time (or the current clock time) and the remaining delay until the next file will start are displayed.

The **gate** list box allows limiting the playback to certain time intervals that can be controlled either by (internal) time programs or external DIO (joystick) signals.

## ***Autostart***

When this option is checked, Avisoft-RECORDER will start to play the current play list immediately after the application has been launched. If the Avisoft-RECORDER has also been added to the Startup folder of the Windows operating system, the playback will start automatically after booting the computer. This feature might be used for unattended playback experiments.

## ***Show playlist entries on touch panel***

This option will launch a separate *Interactive Playback* window representing each file on the currently selected playlist as a large button, which facilitates interactive playback experiments on computers with touch screen. The window can be resized according to the personal preferences. The buttons are by default labeled by the index number of each file in the playlist. In order to customize the appearance of the individual buttons, it is possible to assign user-defined bitmap (.bmp) and text (.txt) sidecar files. These files must be located in the same folder and have the same filename as the corresponding .wav files. The bitmap graphics file would then replace the default number and the content of the .txt file would replace the default filename label. The playlist can be created or opened from the command Play/Playlist... or it can simply be dragged into the *Interactive Playback* window. Alternatively, the playlist could also be created in a simple text editor (each .wav file with its complete path name must be in a separate line that is terminate by newline control characters).

## ***Play playlist entry***

This submenu allows to play single play list entries randomly via keyboard shortcuts. To enable this mode of operation, a play list must have been previously saved onto disk and that play list must have been selected from the Play/Playlist dialog box (via the Open button.). The keyboard shortcuts (by default 1, 2, 3, ...) can be selected from Options/'Keyboard shortcuts and pop up menu...'

This command might be useful for interactive playback experiments, where specific sound files should be played back in response to the current behavior of the animal.

## Configuration

This dialog box allows adjusting all the parameters of hard disk recording and triggering.

**Channel Settings**

Device	Channel	Name	Active	Threshold	Range	Pre-trigger	Min	Hold
... (High Definition Aud	1	ch1	X	1.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch2	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch3	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch4	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch5	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch6	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch7	-	5.0	15,000-80,000	0.5	0.0	0.5
... (High Definition Aud	1	ch8	-	5.0	15,000-80,000	0.5	0.0	0.5

Device: 0 Microphone (High Definition Aud 6.0) Channel: left  Active

Name: ch1  voice notes Current file number: 2  Increment

Pre-trigger: 0.5 Hold tm: 0.5 s Duration > 0 s Syllable > 0 s Monitor...

Trigger: level of this channel  reject wind/rain ...  Open in SASLab  
 whistle tracking  Wait for SASLab

Trigger Event

Level: 1.04 %  Energy  Range: 15 - 80 kHz  Entropy <

Base directory: C:\Users\Raimund\Documents\Avisoft Bioacoustics\

**Input Device Settings**  uniform settings on all devices

Device: 0 Microphone (High Definition Aud 6.0)

Sampling rate: 192000 Hz Buffer: 0.200 s

Format: 24 bit Number of Buffers: 4

high-pass filter:

Gate with: nothing  !  
nothing  !

Save the active channels of each device into a single multichannel file (no separated triggering)

**Display Settings**

Display: spectrogram

Range: 5 % 250 kHz

750Hz FFT size: 256

x10 Overlap: 0 %

Enlarge y axis by: 1

Maximum file size: 10 min

Config. File

Open... Save Save As... Default

The Configuration dialog box is divided into three areas. In the top area, the channel-specific settings including trigger events can be specified. On the left bottom you will find the input device specific settings as the sampling rate and buffer properties. The display settings are located at the bottom on the right side.

## Channel Settings

There is a list of all 8 virtual channels that can be managed by Avisoft-RECORDER (multi-channel version) on the top of the dialog box. The single-channel version of Avisoft-RECORDER will hide that list, because there is only one channel available. Each of these 8 virtual channels can be assigned to one of the physical sound card channels. This is done by choosing the input device (soundcard or data acquisition device) from the **Device** combo box and selecting the desired channel of that device from the **Channel** combo box. Make sure that the driver of the soundcard and data acquisition device (e.g. by National Instruments Corporation) to be used for monitoring has been installed before.

The two last entries of the *Device* combo-box, the '**WAV File**' and '**WAV File Folder**' options, allows you to run the recording procedure from existing sound files located on disk. Select the desired file or folder by clicking at the **File...** button under *Input Device Settings* or alternatively drag the file or folder either into the Configuration dialog box or the RECORDER main window. Use the **Out dev...** button to configure the playback output device for the selected file. This option may be useful for finding the best trigger settings for specific (sporadic) signals or for filtering .wav files that have been recorded with third-party (bat detecting) equipment. Once the 'WAV File' or 'WAV File Folder' option has been selected, .wav files can also be opened and started by dragging them into the RECORDER program window. While the option 'WAV File' will process the selected file only, the option 'WAV File Folder' will process all the files in the selected folder and its subfolders if the option "**include subfolders**" is activated. The processing of .wav files and folders can also be started by drag&drop once the corresponding Device option has been selected. The *Device* option "**Playback**" allows visualizing and processing the waveforms that are currently played through the *Play* menu. Use the option "**automatically skip corrupted .wav files**" in order to automatically reject damaged .wav files. That way the batch processing procedure will not be interrupted by corrupted files.

The **Active** checkbox should be checked if you want to monitor that channel. If that checkbox is not checked, this channel will be passive and the software will only save the settings for a future activation. The *Active* checkbox is not available in the single-channel version (the single channel will always be active).

**Open in SASLab** If this checkbox is enabled, each acquired WAV-file of the selected channel will be transferred to Avisoft-SASLab Pro once the file

has been closed. To enable this mode of operation, *Avisoft-SASLab Pro* (version 3.5 or later) must have been started before. This option can only be activated for one single channel.

**Wait for SASLab** In order to prevent never-ending feedback loops, this option should be activated, when *Avisoft-SASLab* is running in the Automatic Response mode. If this option is activated, the corresponding option *Send acknowledge to Avisoft RECORDER* in the *SASLab Automatic Response Settings* dialog must be activated too. In that mode of operation, *Avisoft-RECORDER* will reject triggering until *SASLab* has reported, that playback in response to the current file is finished.

In the **Name** edit-box you should input a unique name for that channel. This edit-box is not available in the single-channel version.

The **voice notes** option identifies the selected channel as a supplementary voice note track that is associated to the first active (primary) channel. If activated, the path and file names of this channel will consist of the name of first active channel plus the appended string ‘\_notexxx’. In this way, the voice note files that belong to a specific (primary) .wav file can be easily identified. On the *RECORDER USGH*, it is required to enable the menu option ‘Options’/‘Enable soundcards / audio interfaces / multichannel mode’ in order to access the voice notes option and to select and adjust the settings of the sound card to which the voice microphone is attached. In order to save storage space, it is recommended to use a low sample rate of for instance 11025 Hz and a bit depth of 8 bits. If the trigger source of the voice notes channel is different from the primary channel, it is possible to create several short (sound activated) voice note files during longer lasting file of the primary channel. The created voice note files can be visualized and played in the *SASLab Pro* software (the option “File”/“FileOpen Settings...” > “display voice notes” must be enabled).

The **Range** or **Gain** list box is only available for high-speed data acquisition boards and allows selecting the input sensitivity of the channel. *ComputerBoards* and *ines* boards do not support different gains or ranges for different channels. Please note that not all gain values listed here are necessarily supported by the data acquisition board. See the technical documentation of that board to find out which gain settings are valid.

The **Current file number** editbox shows the current file number under which the next soundfile will be saved. This entry has no effect as long as the *Track Number* option from the *File Name Options* dialog has not been activated.

The **Increment** checkbox should be activated if the file numbers should be incremented after completing each event file. If that checkbox is not checked, incrementing of file numbers will not take place. So, each new event would overwrite the previous one. That option is useful if you only want to transfer single sound events to *Avisoft-SASLab Pro* and you do not want to archive all sound events.

It is also possible to increment the file number from *Avisoft-SASLab* using the classification and Automatic Response option. This remote controlled incrementing allows more sophisticated signal-specific filtering of sound events. To set up this mechanism, the **Open in SASLab** and **Wait for SASLab** should be activated. In *SASLab Pro* appropriate Automatic Parameter Measurements and Classification settings have to be made. The Automatic Response action should be set to *increment RECORDER file number*. With such an arrangement, only those files containing an element, described in the classification section of *SASLab*, will be saved.

The **Filenames** button launches the *File Name Options* dialog box that provides a few options for the sound file names under which the sound events will be saved. The **channel name** option will add the name of the channel to the filename instead of creating a separate folder for each channel. The **Filename prefix** is a fixed user-defined string that precedes the rest of the file name. If the option **add space** is checked, the prefix will always be separated from the rest of the file name by a space character. The **Date** and **Time** options insert the current date and time to the filename. If the option **Compact format** is activated, the date and time is formatted in a more compact form. The **GPS** option will add the current coordinates and weather data to the filename if the corresponding options have been activated. The **Track Number** option adds a continuous number to the filename. The **Digits** list box determines the number of digits in that number. Make sure that there are sufficient digits to represent all forthcoming sound events. If the *Date* and *Time* options have been activated, the track number can be reset each day at a certain clock time by selecting the desired **Reset track number at** option. If the *Time* option is not activated, the track number can only be reset at 00:00.

If the **Prompt up a filename editing dialog box** option is checked, the **Filename** of the acquired .wav file dialog box is launched on each end of a file and allows modifying the filename of the previously saved file. The option **Modify the prefix only** will limit the modification to the first part of the filename. The date or file number will remain unaffected. In order to accelerate inputting frequently-used text strings, the text modules defined from the main window commands *Monitoring/LOG File comment text modules* are available on both the *Text modules* menu (which are available

also through the keyboard shortcuts F1...F12) an the combo-box list. The option **Keep the modified prefix** keeps the entered prefix for the next file. Check the option **Disable this filename prompt mode** if you want to cancel the manual file naming.

The Default button provides the standard settings (Prefix = "T", Track Number with 7 digits). The structure of the currently selected file name is displayed at the bottom of the dialog box. The option **Prompt up a filename editing dialog box** will launch a dialog box that allows to modify the name of the previously saved file. Therefore, this option should only be activated when using a manual trigger mode. The current folder number can be edited from the **Folder number** field. The **Create subfolders for each monitoring session** option adds an additional folder layer. The **Folder prefix** field defines the fixed prefix of the folder names. The current folder number can be edited from the **Folder number** field. The option **reset track for each subfolder** will reset the track number to 0001 for each new subfolder.

The option **Create 'best' .wav file header chunk** creates an additional chunk in the .wav file header that complies with the broadcast wave format (BWF), which includes the creation date, the time reference from the start of the monitoring process and the GPS coordinates (if the GPS functionality is activated).

The option **Batch mode for rat/mice drug testing** will activate a special batch mode for processing large numbers of laboratory animals (see page 79).

The option **regularly update the .wav file header while recording** will make sure that the recorded .wav files are being saved properly even if the program fails to close them at their regular end due to a power failure for instance.

The option **add last playback filename and its start time** will add the name and start time of the last played .wav file to the filename, which can be useful for certain playback experiments. The playback filename and clock time is taken from the playback logfile.

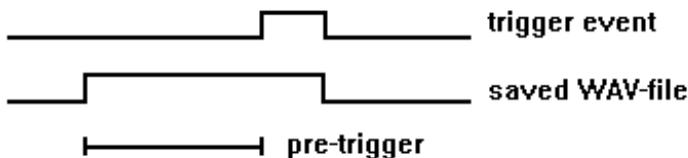
If either the *WAV File* or *WAV File Folder* option has been selected, the name of the original source .wav file can be added to the newly saved .wav files by activating the option '**add source filename**'. On the *WAV File Folder* device option, is possible to **copy the source files to the subfolders ACCEPTED or REJECTED**, depending on the occurrence

of at least one trigger event, which can facilitate for instance applying the Bat Call Filter to .wav files that have been automatically recorded by third-party bat monitoring equipment. This mechanism will also work with the Pause button activated. In case the procedure is repeated with different settings, the folders ACCEPTED and REJECTED must be manually deleted before running the procedure again.

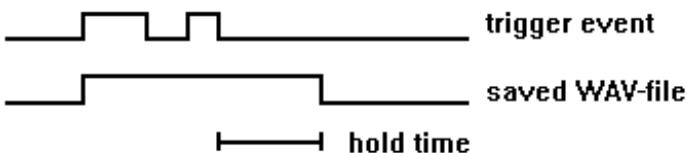
The **CtrlOut...** button provides various mechanisms for exporting the current trigger (or file saving) state. See page 66 for details.

The **USV...** button launches the USV Real-Time Monitoring Setup dialog box (see page 77for details).

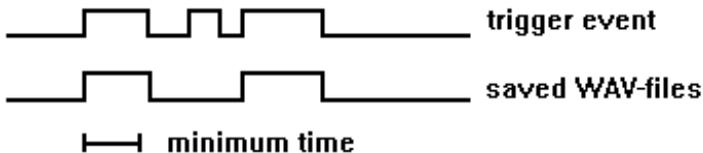
The **Pre-trigger** edit-box allows you to input the desired pre-trigger duration. This is the duration of sound that will be saved before the trigger event occurs. Keep this value as small as possible because large pre-trigger will require a large amount of RAM memory (depending on the sampling rate). Another reason for keeping the pre-trigger small are the possible gaps in the hard disk recording process when the PC is not able to save the entire pre-trigger data before the internal circular buffer is saturated. The internal temporal resolution of the pre-trigger is limited by the **Buffer time** specified in the **Input Device Settings** section.



The **Hold tm** edit-box specifies the amount of hold time that will elapse after the last trigger event occurred before the sound file of that event will be closed. A hold time greater than zero will prevent that several consecutive short trigger events will be saved into separated files, which might be undesired because related sound elements would be cut up. The internal temporal resolution of the hold time is limited by the **Buffer time** specified in the **Input Device Settings** section.



The edit box **Duration >** allows specifying a minimum duration for a trigger event. If the duration of the recorded sound file is shorter than this parameter, the sound event file will be skipped and overwritten by the next one. This mechanism may be useful to reject short sound events from which is known that they are out of interest. The internal temporal resolution of the minimum duration parameter is limited by the **Buffer** time specified in the **Input Device Settings** section. The **Duration >** threshold is applied to the size of the recorded sound file excluding the initial *pre-trigger* and the final *hold time*. However, gaps within the acquired sound event that are shorter than the *hold time* setting will be ignored and will therefore not reject the sound file. The default setting for the **Duration >** parameter is 0.



The edit box **Syllable >** allows specifying a minimum duration for the syllables within the recorded sound file. If there is no syllable (a single sound element that exceeds the amplitude threshold) within the recorded sound file that is larger than specified minimum syllable duration, then the sound event file will be skipped and overwritten by the next one. In contrast to the above **Duration >** threshold setting, the **Syllable >** threshold does not take into account the *hold time*. This means that any silent interval within a vocalization will reset the internal syllable duration counter. The internal resolution of the **Syllable >** threshold is only limited by the **FFT size** setting and not by the **Buffer** duration as in the **Duration >** threshold mechanism.

This syllable duration threshold mechanism might help to reject unwanted short calls, while songs that consist of longer syllables will be recorded. The default setting for the **Syllable >** parameter is 0.

The **Monitor** button launches a dialog box for activating various acoustic monitoring and measurement functions. See page 62 for details.

The **Trigger** combo-box defines the type of event used to start and stop (trigger) this channel. The default trigger is the **level of this channel**, which means that the level of the same channel will be used for triggering.

Alternatively, a different channel (**level of channelx**) can be used. This allows starting several channels simultaneously by the same trigger event of a particular channel. The **whistle tracking** option replaces the default level trigger by a special whistle tracking mechanism.

There is a set of other trigger events that can be used for triggering recording into WAV files:

**any channel (OR)** Any of the levels of the active channels will activate the trigger. The trigger events of all active channels will be linked by logic OR. This trigger option is useful for microphone array applications, where any of the microphones should be able to trigger a recording for all channels (use the option “Save the active channels of each device into a single multichannel file (no separate triggering)”).

**any key** Any key pressed on the keyboard will activate the trigger.

**key 1 ... key 8** A user-defined keyboard key will be used for triggering. The desired key can be selected from a list box. Note that those keys that have already been defined as keyboard shortcuts (command “Options”/“Keyboard shortcuts and popup menu...”) will not function as a trigger.

**button1 ... button3** One of the buttons  ...  on the top of the Avisoft-RECORDER window will be used for triggering. Button3 will be set automatically each time a playback is initiated (for recording responses in playback experiments).

**Play button** The play button will be used for triggering. This option may be useful for recording responses in playback experiments. In such an application, an appropriate hold time should be used.

**channel button** The play button  associated to the channel will be used for triggering.

**permanent (unlimited)** This option represents a permanent ‘trigger’ that is always active. This means that the recording will start once you have clicked at the recording button (while the pause button is not active) and will continue until you click at the stop button.

**permanent (limited)** This option represents a permanent ‘trigger’ that becomes active immediately after clicking at the red start button and continues until the defined period of time has elapsed.

**interval timer** A trigger event is generated at regular time intervals. Use the Hold time field to set the duration of each shot.

**time program 1...8** The channel is recorded as long as any of the pre-defined time intervals is valid. The **Edit...** button allows to input the desired time intervals for the selected time program (there are 8 independent time programs available). See page 62 for details.

**DIO bit 0...7** The digital IN bits of a data acquisition unit is used as trigger event. By default, the trigger will be activated when selected bit is at the low state (active low logic). In the NI-DAQmx version of the software, the desired device and port numbers can be activated and defined from the **NI-DAQmx DI** edit field at the bottom of the dialog box (the default setting is "Dev1/port0").

**left-click / touch window area** A trigger event is generated by left-clicking (or tapping) the program window area below the button bar.

**permanent with pretrigger** This option is similar to the option "permanent (unlimited)" except that it will function with the pretrigger (pre-recording).

**UltraSoundGate DI** The digital input of the Avisoft-UltraSoundGate x16x device is used as trigger event.

**call class other than "?"** A trigger event is generated once any of the call classes defined on the command "*Options/Automated call parameter settings... > Define call classes...*" has been detected. In order to capture the corresponding call, the pre-trigger duration should be selected appropriately.

**.wav file playback** A trigger event is generated each time when a .wav file playback is started. This mechanism will work across instances. In order to get this work, the option "create playback log file" on the command "Play"/"Settings..." must be activated in the instance that performs the playback. Note that the "Hold tm" must be set to a value that covers the duration of the playback.

**joystick b1/b5 ... joystick b4/b8** One of the joystick buttons will be used for triggering. If the joystick control mechanism is not working properly, check the Game Controller settings on the Windows Control Panel (Start/System Control/Game Controller/Extended). Make sure that the desired joystick is selected as the preferred device.

**joystick north, south, west, east** One of the directions pressed on the joystick will be used for triggering.

Some of these trigger events (any key, F-keys and joystick buttons) can be used in a **Toggle** mode. If the **Toggle** checkbox is activated, each hit of the key or button will change the state of the event. The first hit will switch the event on and the second hit will switch it off. If the toggle mode is disabled, the event is active as long the key or button is pressed.

The trigger sources **DIO bit 0...7, UltraSoundGate DIN and joystick b1...b4** can be used in inversed logic (active high) by activating the **!** option. The **TC** option allows to trigger on a specific timecode stamp (requires either a LANC or SMPTE timecode signal fed into a UltraSoundGate DIN input). The time stamp string is expected in the format hh:mm:ss:ff.

### **Trigger Event**

This section defines the behavior of the trigger event generated by this channel. The trigger event is a binary signal that is derived from the FFT-spectrum of the analog channel. Either the maximum amplitude of a spectral component within a frequency interval or the amount of energy within that frequency interval is compared with a threshold. If the incoming signal exceeds that threshold, the internal binary trigger event signal will become active. So, those analog trigger events may be used for a sound-activated recording mode (see **Trigger** above).

If the **Energy** checkbox is checked, the **Level** will be compared with the amount of energy within the specified frequency Range. If the **Energy** checkbox is not checked, the **Level** will be compared with the maximum amplitude of a spectral component within the specified frequency range. The **Energy** checkbox mechanism allows to distinguish between signals whose energy is spread over a wide frequency interval (noisy signals) and signals whose energy is concentrated into a narrow frequency range (pure tones, whistles).

The **Range** edit-boxes represent the frequency boundaries of the frequency interval to be observed. Any signal components outside that frequency range will be ignored for generating the trigger event. This allows excluding background noise and other signals that should not start a recording. This frequency discrimination is done by FFT, which has a limited frequency resolution depending on the sampling rate and **FFT-length** (see below). The current **Resolution** is displayed behind the **Range** edit-boxes. That **Resolution** limits the specified values for the frequency **Range**. If no frequency selective trigger is desired, the **Range** option can be disabled. Especially in configurations with many channels or high sampling rates, the disabled Range option may save computing power, because the FFT is not computed. In order to prevent computing

the FFT, additionally the display modes *spectrogram* and *energy in f-range* must not be activated.

The **Entropy** < option allows to further limit the potential trigger events in order to prevent triggering for noise signals. If this option is activated and the amplitude threshold described above is exceeded, an additional 'purity' parameter is computed from the current FFT spectrum. Only if this instantaneous entropy parameter is below the specified threshold, the trigger event will be valid. Otherwise, the trigger event is ignored. The entropy parameter is expressed in percent. A pure (sine) tone would theoretically produce a value of 0%. A white noise signal would have 100%. Due to the properties of the windowed FFT spectra, pure tones will have entropy values significantly above 0%. Setting the entropy threshold to a value of about 35% will exclude noisy signals from triggering. The **energy in f-range** display shows the instantaneous entropy and the associated threshold in green. Depending on this real-time display, the entropy threshold may be modified in order to get the desired behavior.

The option **reject wind/rain** can help reducing undesired triggering by more or less stationary broad-band noise signals, such as wind or rain. If a broad-band signal (which is identified by calculating the Wiener entropy of the instantaneous spectra) exhibits a predefined duty cycle threshold and lasts longer than a predefined time interval (defined by an averaging time constant), the sound-activated trigger (*level of this channel*) will be temporarily inhibited for a user-defined hold time. The parameter settings associated with this mechanism can be adjusted from the **Reject Wind/Rain Noise** dialog box that can be launched from the ... button:

The **Amplitude level threshold** corresponds to the **Level** threshold of the **Trigger Event** and determines the sensitivity of the broad-band noise detector. For proper operation, this threshold should be lower than the Level threshold for the Trigger Event. The Wiener **Entropy threshold** is applied to the instantaneous spectra once the **Amplitude level threshold** is exceeded. If the entropy of the current signal exceeds this threshold, the signal is internally labeled as a potential noise signal. The state of this intermediate signal is averaged by the specified **Averaging time constant**, which provides a measure of the instantaneous duty cycle of the noise signal. If this measure exceeds the **Duty cycle threshold**, the trigger will be deactivated for the specified **Hold time**. These settings can be tested for instance by using a previously recorded .wav file (Device = 'WAV file') that contains the undesired noise signal. During monitoring, a

lower case **w** indicates that the trigger is currently inhibited due to the wind/rain noise rejection mechanism.

The entropy and its threshold can be visualized by activating the option '**energy in f-range**' on the **Display** list box. The green dots then represent the instantaneous values of the entropy and the horizontal solid green line is the threshold that can be modified by mouse dragging.

In case the option **whistle tracking** has been activated, the default level trigger will be replaced by a special whistle tracking mechanism that only looks at the continuity of the peak frequencies (or frequency modulation) on the real-time spectrogram. This mode of operation can be useful for detecting soft whistle-like vocalizations that are almost buried in the background noise.

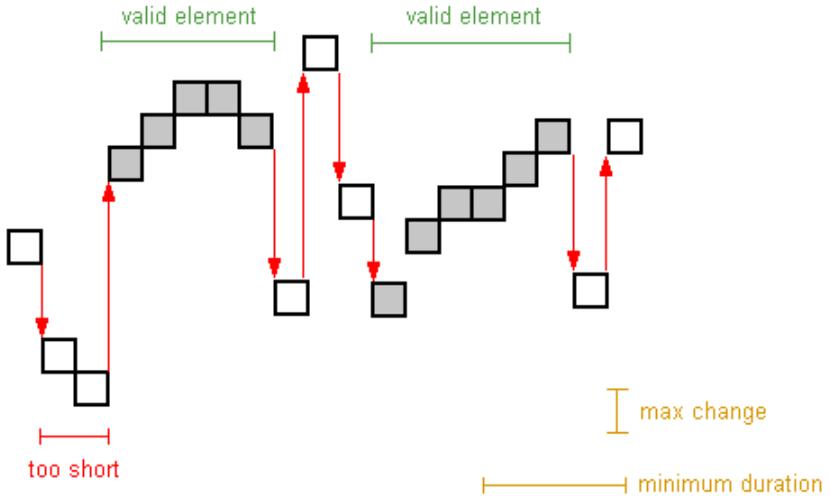
The amount of frequency modulation from one time bin to the next that should be tolerated within a valid vocalization can be defined from the edit field **max change**. It is expressed in spectrogram pixels and therefore depends on the currently selected FFT size and sample rate. The **min duration** setting rejects potential whistles that are shorter than this specified duration. In order to prevent false detections (caused by noise), this value should be as large as possible, but shorter than the typical USVs you are looking for. In order to enable the proper operation of the whistle tracking algorithm, this duration must be at least three spectrogram time bins long (one spectrogram time bin corresponds to FFT size/sample rate). There is another **hold time** parameter that determines how closely-spaced calls or calls with short silent gaps are treated. This hold time setting can be edited only from either the bat call filter setup dialog box or from the USV real-time monitoring setup dialog box.

The option **slope** allows to enter a range (between min and max) for the tolerated frequency change. Positive values correspond to rising and negative values represent falling frequencies. The resulting frequency slope range (expressed in kHz/ms) is displayed at the right margin of the dialog box. The resulting slope limits also depend on the currently selected FFT size and overlap. The option **& level** will additionally limit the triggering to sound events whose amplitude level exceeds the *Level* threshold. The *Level* threshold can be edited by temporarily disabling the *whistle tracking* option or by dragging its graphic representation on the "*energy in f-range*" display mode. The slope option can be used to reliably distinguish short downward-modulated bat echolocation calls from other noise signals such as rain drops or broad-band insect buzzes (use the command

Options/Configuration management/Presets/Bat Monitoring with bat call filter to get suited parameter settings for this application).

In order to exclude any undesired noise components outside of the frequency range of the expected vocalizations, the frequency **Range** should be limited accordingly.

Successfully detected whistles will be indicated on the spectrogram display by red horizontal lines at the top of the display.



Principle of the whistle-tracking mechanism

The **Base directory** edit-box specifies the hard-drive and directory, where the acquired WAV-files for each channel will be saved. The directory can either be selected from the browse button (...) behind the edit box or by direct input of the folder name. Alternatively, the directory can also be selected by drag&drop into the Configuration dialog box. The subdirectories for the WAV-files of each channel will be allocated in this **Base directory**.

The **Copy** button will copy the current settings from the selected channel (and the input device) to all other activated channels. This might help to accelerate setting up complex multi-channel configurations.

## ***Input Device Settings***

This section allows defining the sampling rate and buffer settings for the input devices. The parameters displayed apply to the currently selected **Device**, unless the option ***uniform settings on all devices*** is activated. So, to watch and edit the parameters of all input devices you need to subsequently select each of the devices.

**Sampling rate** This is the sampling frequency of the selected device.

**Diff** This option activates the differential input mode for ines data acquisition boards. For National Instruments boards use the *Measurement & Automation Explorer* to configure the input modes.

**Format** Depending on the type of soundcard different bit-depths can be selected (8, 16 or 24). For high-speed data acquisition boards, the binary data format of the analog-to-digital converter has to be selected here. This will ensure that 12 bit data are shifted into the 12 most significant bits of the 16 bit data words of the .WAV file. Some A/D converters (e.g. ComputerBoards devices) supply a special offset format for signed integers. If the appropriate format is selected, the numbers are converted into the standard integer format, which is required for common 16 bit .WAV files. For the DAQCard 6062E and ines DAQ cards, the option '12 bit int' should be selected. Select '32 bit int' for the NI DSA devices (44xx).

**Buffer** In order to ensure a gap-free monitoring and recording, it is necessary to provide several data buffers of a certain size to the device driver of the soundcard or audio interface. The size is expressed in seconds. This time should be long enough to bridge all other tasks of the PC while monitoring. Larger values will result in a save but more discontinuous operation of the on-line display.

**Number of buffers** This is the number of buffers used for the data streaming mechanism from the device driver. Large numbers will result in a more save operation but larger memory requirements.

**Slave** This option is only available in the RECORDER USGH software version and activates the slave mode for UltraSoundGate XX16H devices.

**Settings...** This button (only available in the RECORDER USGH software version) launches the *Advanced USGH Device Settings* dialog box for configuring the UltraSoundGate XX16H devices. See page 27 for details.

**File...** If the File input device is selected, this button allows selecting the .wav file to be used as input signal.

**Out dev...** If the 'WAV File' input device has been selected, this button allows you to set up the output sound device for playback during processing the selected sound file. The appearing dialog box presents a list of all available output devices. The playback rate can either be taken from the *.wav file header* (default) or *another rate* specified here. Selecting the *other rate* option will speed-up or slow down the original file playback speed. However, the frequency scaling for spectrograms and the trigger settings will still be based on the original sampling rate in the *.wav* file header. This option may be useful to adapt the sampling rate to the capabilities of the selected output (playback) device. If the playback during processing the *.WAV* file is not possible (due to the number of channels or a high sample rate) or not desired (in order to shorten the procedure), the last *Device* entry 'no output – process as fast as possible' can be selected. In this mode of operation, the *.wav* file will be processed at the maximum possible speed (depending on the selected *Buffer* duration). The processing speed can be increased by selecting a higher *Buffer* duration (0.5 or 1.0 s).

**high-pass filter** If this option is activated, all channels of the selected device will be digitally high-pass filtered at the specified cutoff frequency. The number of taps of this FIR filter can be selected from the **Taps** list box. The number of taps determines the steepness of the filter. Note that this additional filter processing will increase the processor load. High sample rates, many activated channels and a large number of taps can lead to data loss (gaps in the data stream) due to the limited computing power. In such a case the number of taps should be reduced or the filter option must be deactivated.

Caution: In case the high-pass filter is applied to an UltraSoundGate channel, the state of the DIN input would not be preserved as long as the **Trigger** option is not set to **UltraSoundGate DIN** or the **Keep DIN** option on the Advanced USGH Device Settings dialog box is not checked. So, in order to preserve the DIN state it might be necessary to check the **Keep DIN** option.

## ***Display Settings***

During monitoring, each of the active channels can be displayed graphically. There are the following display options:

**no display** Use this option if you need to minimize the processor load in time-critical configurations (the graphic displays usually occupy a fraction of the available computing power).

**waveform**  Waveform display. Display range [%] can be edited. Over-modulation is indicated by a color change of the waveform display from yellow to red.

**spectrogram**  Spectrogram display with a graphical representation of the trigger event frequency ranges that can be edited interactively by mouse tracking. Display contrast [%] and display range [kHz] can be edited. The contrast can also be modified by pressing the Up and Down keys or by clicking at the buttons  and .

**energy in f-range**  Energy or amplitude display within the frequency range defined in the **Trigger Event** section of each channel. The threshold **Level** specified for each trigger event channel is represented graphically and can be edited interactively by mouse tracking.

**FFT size** This is the FFT-length used for both displaying of spectrograms and computing trigger events. This parameter determines the frequency resolution of the spectrogram display and the frequency-related trigger events. The **x10** option boost the FFT magnitudes by factor 10 and calculates the FFT in float rather than integer arithmetic, which provides a better trigger resolution in low-level signals (at the expense of a slightly increased processor load).

**Overlap** The FFT overlap setting determines the temporal resolution of the real-time spectrogram.

**Enlarge y axis** This listbox allows selecting a magnification factor for the y (frequency) axis of the real-time spectrogram display. Enlarging the frequency axis might be certainly useful when using the RECORDER software on laptops or UMPCs with small display panels.

**more** The more button launches a separate **Spectrogram Settings** dialog box that provides additional options such as **Window** type and **Color Palette**. The option “**Show spectrum (dBFS vs frequency)**” at the bottom of the box will activate a real-time spectrum display window showing the mean and peak spectrum along with the peak frequency and amplitude. Amplitude and frequency can be measured using the mouse cursor. The combo box titled “**peak hold duration**” determines how long the peak amplitude will be held. During the monitoring, right-clicking at the spectrum display will reset the peak hold values. The **level meter mode** list box allows switching between the **peak level** and the **rms level** calculation method on the recording level meter display. The default **peak level** option

allows monitoring the dynamic range usage and helps to recognize clipping, while the *rms level* option facilitates precise level measurements.

### **Maximum file size**

This option allows limiting the file size to a certain duration (specified in minutes). When this time has elapsed, the current file is closed, the file counter is incremented and a new file is created. There will be no gaps between these subsequent files. This mechanism will ensure that file sizes remain low even when recording over several hours. This option also overcomes the file size limitation in .wav files (max. 2 GBytes).

### **Gate with**

These two list boxes allow activating up to two additional trigger criteria (linked by logical AND) that apply to all active channels. By checking the options *!* behind these list boxes, that additional logic signal will be inverted (logic 'NOT'). If no additional criterion is desired, select **nothing**. The options **time program** will limit the sound-activated trigger operation to a few pre-defined periods of time. This might be useful if recordings should be made only for several hours (e.g. from 4am to 7am) each day. The **Intervals...** buttons launch the dialog box for defining the time intervals.

### **Save the active channels of each device into a single multichannel file (no separated triggering)**

If this option is activated, all channels of a multi-channel device will be saved into a single (interleaved) multi-channel .wav file. In that case the channels can of course not be triggered separately. The parameters for these channels can only be edited from the first channel of that device in the list. Threshold trigger events can also be activated from the first channel only, unless the trigger option "any channel (OR)" is selected. The created multichannel .wav files could afterwards still be split into a set of single-channel (mono) files in Avisoft-SASLab Pro (command *Tools/Batch processing... > Split multichannel files into mono files*).

### **Show timecode**

This option activates the LANC and SMPTE timecode decoding mechanism that detects timecode signals on the UltraSoundGate DIN inputs (only available on RECORDER USG/USGH).

### **enable USG TRG button | left-click as alternative trigger source**

If activated, this option will alternatively trigger all active channels either by pressing the UltraSoundGate trigger button (only available on RECORDER USGH) or by left-clicking (or tapping) on the program window.

The **Config. File** section provides commands for managing configuration files. See also page 27.

The **Default** button will set all parameters to their default settings.

## Time Interval Definition

This dialog box allows to define time intervals that can be used as a trigger signals or as a gate for other trigger options.

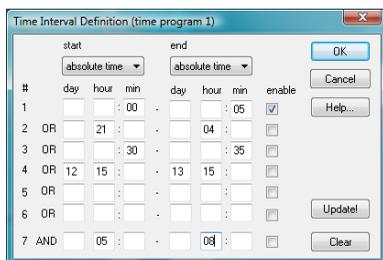
A time interval is defined by its **start** and **end** time (day, hour : min). Up to six periods (**#1...6**) can be specified (processed by logic OR). Additionally, those six intervals can be further limited by a seventh time interval that can further limit the resulting active time intervals (logic AND).

Only those periods that have been activated by the enable check box will be applied. It is not necessary to fill out the day entries except the time interval should become active only for one or more specific days in a month. If the day is not specified, the time interval will be valid for each day. If the hour is not specified, the time interval will be valid for each hour. The **Clear** button will remove all time entries.

By default, the time intervals are referenced to the (absolute) local time (option **absolute time**). Alternatively, the intervals can be referenced to either sunrise (rel. **to sunrise**) or sunset (**rel. to sunset**) of a predefined geographic location (**latitude, longitude**).

The **Update!** button reads the entered settings and checks them logically.

The following example shows four absolute time intervals, where only the first is activated. The fifth interval has been enabled in order to limit recording to the morning:



Interval #1 will be active each hour xx:00 to xx:05 am. The activated **AND** entry #7 will further limit the intervals to the three hours between 5 and 8 am. So, the RECORDER would record three times each day : 05:00 - 05.05, 06:00 - 06.05 and 07:00 - 07.05.

Interval #2 would be active each day from 9:00 pm to 4:00 am on the next day.

Interval #3 would be active each hour from xx:30 to xx:35

Interval #4 would be active each month from the 12th 3 pm to the 13th 3 pm.

If either the **rel. to sunrise** or **rel. to sunset** option has been selected, the absolute start and end times will be determined by the local sunrise or sunset time. In order to make this work, it is required to enter the geographic coordinates into the edit fields **latitude** and **longitude**. The coordinates are expected in decimal degrees, where positive values correspond to North and East and negative values to South and West. The time table entries #1...#6 will then define a time offset relative to sunrise or sunset. If no time offset is desired, a '0' must be entered in either the **hour** or **min** fields:

#	day	hour	min	day	hour	min	enable
1		+0	:00		+0	:00	<input checked="" type="checkbox"/>
2	OR		:		:	:	<input type="checkbox"/>
3	OR		:		:	:	<input type="checkbox"/>
4	OR		:		:	:	<input type="checkbox"/>
5	OR		:		:	:	<input type="checkbox"/>
6	OR		:		:	:	<input type="checkbox"/>
7	AND		:		:	:	<input type="checkbox"/>

sunset: 16:57      sunrise: 06:54  
latitude: 52      longitude: 12      (decimal degrees, WGS84 datum)

The following example defines a time interval that starts one hour before sunset and ends one hour after sunrise:

#	day	hour	min	day	hour	min	enable
1		-1	:00		+1	:00	<input checked="" type="checkbox"/>
2	OR		:		:	:	<input type="checkbox"/>
3	OR		:		:	:	<input type="checkbox"/>
4	OR		:		:	:	<input type="checkbox"/>
5	OR		:		:	:	<input type="checkbox"/>
6	OR		:		:	:	<input type="checkbox"/>
7	AND		:		:	:	<input type="checkbox"/>

sunset: 16:57      sunrise: 06:54  
latitude: 52      longitude: 12      (decimal degrees, WGS84 datum)

It is also possible to switch between absolute and relative time specification views, which can simplify the setup procedure. For correct operation of the time table functionality, it is required that the time zone and clock time of the PC has been set properly (Windows Control Panel > Clock, Language and Region).

## **Monitor**

This dialog box allows to set up several acoustic monitoring features.

The **acoustic monitor** will output the incoming sound to a soundcard output (speaker or headphone):

The list box **mode** provides the following options:

**Acoustic monitoring off** : The acoustic monitoring is switched off.

**1:1 monitoring (output=input)** : The input signal is passed unmodified to the output device. Use this option for monitoring audible (low-frequency) signals.

**broad-band HF monitoring** : The entire input signal is decimated and the resulting (under-sampled) signal is passed to the output device. The decimation ratio can be selected from the **decimation** list-box. The option **auto** will automatically select the best ratio for the currently selected input channel sampling rate. This mode will convert inaudible high-frequency signals into audible low-frequency signals. It is a broad band monitoring technique using the aliasing effect. The original frequency of the input signal cannot be determined from the output signal.

**envelope tracking** : The incoming signal is rectified and averaged in order to detect the envelope. This kind of monitoring can be used for listening to ultrasonic echolocation clicks (sonar) emitted by cetaceans. The time constant of the averaging process can be entered into the corresponding edit field. The time constant should be significantly smaller than the shortest interval between the incoming clicks. Larger values may reduce the low-frequency noise. The envelope tracking mechanism requires that the clicks to be detected have significantly steep slopes. Otherwise, the monitor output would be too soft or completely inaudible. The heterodyne monitor would be more appropriate for signals having softer slopes.

**selective heterodyne HF monitoring** : A limited frequency interval of the input signal is shifted into the audible frequency domain. The list box

**bandwidth+-** defines the bandwidth of the frequency interval to be monitored. The edit box **center freq.** defines the center frequency of the downshifted frequency interval. This center frequency and the bandwidth will be visualized on the real-time spectrogram display by horizontal green lines. The center frequency can be tuned in real-time by mouse or by pressing the left and right arrow keys. This mode emulates the functionality of a heterodyne bat detector. It is a narrow-band monitoring technique that only outputs the selected frequency interval. In case of limited computing power (in conjunction with high sampling rates and (or) many channels), this option should not be activated, because it will occupy a relatively high amount of the available processor capacity.

The list box **output device** defines the sound output device that should output the monitor signal. The list box **channel** defines the desired output channel of the soundcard. The options **left** and **right** will output the signal to the specified channel only (the opposite channel will be quiet).

The **digital gain** option can digitally amplify the digital samples that are being sent to the output audio device. Use a gain setting higher than 0 dB in order to amplify soft input signals. Note that a gain setting higher than 0 dB can potentially introduce clipping to the output signal.

The **delay** option allows to set an additional delay to the output signal. The minimum delay is limited by the sound card data handling and the selected **Buffer** size from the configuration dialog. Small buffer sizes will decrease the minimum possible delay. Usually, the delay should be set to the minimum available value. Lager delay settings may be selected for special applications requiring echo generation.

The **input rate** represents the currently selected sampling rate of the sound input device. The **output rate** represents the sampling rate of the output device for the currently selected monitoring mode.

The **measurements** section provides options for real-time frequency measurements on the incoming sounds. These measurements will be updated as long as the amplitude threshold is exceeded within the specified frequency interval. This amplitude threshold and the frequency interval is defined in the *Trigger Event* section of the Configuration dialog (or graphically by mouse from the energy and spectrogram display). For correct measurements, it is essential to adjust the threshold properly.

**instantaneous / end frequency** : The instantaneous peak frequency will be measured. For signals with long duty cycles (e.g. bat echolocation

calls), the displayed value represents the peak frequency at the end of the most recent pulse.

**frequency at max amplitude** : The frequency at the maximum amplitude of each sound event exceeding the threshold will be displayed. This mode will only be useful for relatively short sound pulses with long duty cycles (long silent intervals between sound elements, for instance bat echolocation calls).

The section **voice messages** provides options for issuing spoken comments on the input channel. The voice output of these options is fed into the preferred audio output device (selected from the Windows system control panel) :

**overload** : In case of overload, an overload message will be issued.

**start/stop** : The start and stop of each sound event file will be announced.

**frequency** : The frequency as defined under **measurements** will be issued. The option **kHz only** will limit the precision of the issued number to integer kHz values.

The voice messages are constructed from .wav file voice pieces located in the *Messages* directory of the Avisoft folder. So, these voice messages may be replaced by your own files. All .wav files must have the same file format (sampling rate, bit-depth and channel count).

## ***Control Output settings***

This dialog provides various mechanisms for exporting the current trigger (or file saving) state. The selected options will apply for all activated channels. If you need to distinguish between various channels, then you should activate the **append channel number** option. These options may be used for controlling external devices (e.g. video cameras or feeding devices in feedback experiments) or reporting trigger events (e.g. via e-mail).

### **activation mode**

**trigger state (TRG)** : The control signal is derived from the internal trigger state.

**file saving state** : The control signal is connected with the sound file generation (a control signal will be issued on the begin and on the end of each sound file)

**action** : This list box provides the available commands that will be executed on the begin and / or the end of a trigger or file:

**none** : no command is executed

**run .exe** : a command line application (.exe file) is executed

**DDE XTYP\_POKE** : a DDE poke command is issued

**DDE XTYP\_EXECUTE** : a DDE execute command is issued

**Play soundfile** : a .wav file is played back (select desired audio output device from the command Play/Device...)

**start** : specifies the command line application, sound file or the DDE command that is executed (played) on the start of an event

**end** : specifies the command line application, sound file or the DDE command that is executed (played) on the end of an event

The DDE command must be defined by specifying the application, the topic and item that should receive the command and finally the data that should be sent to the application. These strings should be separated by “,” characters.

**append current filename** : The current sound file name will be added.

**append channel number** : The number of the channel that has caused the command is added.

For the DDE options, the above **append** options will be added to the data section (separated by a space character). For the **run .exe** option, the appended data will be passed over as command line parameters.

**NI-DAQmx DO**: If this option is activated, the states of the activated channels will be passed to the specified digital out port (DO) of the installed DAQ device (only available with National Instruments DAQ devices with Avisoft RECORDER NI-DAQ or NI-DAQmx)). Up to 8 channels are mapped to the 8 bits of the digital out port. Channel 1 corresponds to P0.0 (... channel 8 to P0.7). Note that the specified digital port (bit 0...7) will be entirely configured as an out port once this option has been checked. It is therefore not possible to use the selected DO port here also as a trigger input DI on the Configuration dialog box.

**TRG out** : If the **TRG out mode** option on the *Advanced USGH Device Settings* dialog box has been activated and if you have a multichannel UltraSoundGate unit (416H, 816H or 1216H with the firmware version 1.1

or higher), this list box allows to select the desired channel whose trigger or file saving state should be routed to the TRG socket. The **any trigger** option will allow to output any of the internal trigger signals on the TRG socket.

**Active low** : If activated, the digital output signal at the (NI-DAQ) DIO out and (USG) TRG out will be active low (0V for the active state).

## **Advanced USGH Device Settings**

This dialog box provides various options for the UltraSoundGate XX16H devices:

**Enable master/slave mode for synchronizing several devices** If you use more than one UltraSoundGate XX16H device, it is possible to synchronize their A/D converter sampling clock signals by connecting them through the SYNC sockets and activating this option. In such a set-up, one device acts as the clock master and the other devices listen to the clock signal as slaves.

**Activate slave mode for this device** In the *master/slave mode*, one device acts as the master and the others as slaves. So, this option must be disabled on the unit that should act as the master and must be enabled on all other slave units (all USG XX16H devices can act either as slave or master).

**Enable low power mode** The A/D converter chips can optionally be operated in a power-saving mode that would increase the battery life in mobile laptop-based systems. The low-power mode slightly degrades the dynamic range performance. This option should also be activated if the models USG 816H and 1216H are bus-powered from a single USB port only. Otherwise, the supply voltage might drop below a critical value, which can prevent the proper operation of the A/D converters.

**Enable band-pass mode** At a few sample rates (50, 62.5, 75, 150, 187.5, 250 and 375 kHz), the A/C converters can be configured for a special band-pass mode in which the analog input bandwidth ranges from  $f_s/2$  to  $f_s$  instead of the normal mode that goes from 0 to  $f_s/2$ . This option can be advantageous for monitoring applications that require minimal .WAV files sizes. Note that the frequency scale of the resulting .WAV files will be reversed.

**Turn on the polarization voltage** This option activates the internal 200V polarization voltage generator of the UltraSoundGate XX16H devices and must be activated when using the CM16/CMPA microphones.

**Enable automatic DC offset compensation** If activated, this option removes potential DC offset voltages from the input signal. The **Keep DIN** option will keep the least significant bit (the DIN track) even if the DC offset compensation is activated.

**Show DIN** will visualize the state of the DIN input on the real time display as a blue trace.

**Ignore GetOverlappedResult error** By default, the *GetOverlappedResult* error message will stop the monitoring/recording process. **If** this option is activated, the monitoring procedure will be immediately restarted, which is desired in long-term monitoring applications.

**TRG out mode** This option allows to change the I/O direction of the TRG socket. If activated, the TRG socket will be configured as a digital output that carries the state of the internal trigger or file saving state as selected from the **Control Output settings** dialog box (*Ctrl Out...* button). In this case, the *USG TRG button* trigger source options will not work. The *TRG out mode* option is only available on units that have the firmware version 1.1 or higher.

**Enable safe multichannel data transfer mode** Under some circumstances it might happen that the PC is unable to reliably transfer multichannel data streams over the USB interface without gaps. This can become a serious problem when the sample rates and the number of channels are high and if the latency of the Windows operating system is poor (due to insufficient processing power or misbehaving device drivers). So, if you encounter any problems while recording more than one channel (sporadically mixed-up channels), this option should be activated. The RECORDER software will then correct those false channel associations. The latency of the PC can be checked by using the DPC Latency Checker application by Thesycon GmbH ([www.thesycon.de](http://www.thesycon.de)).

## Command line options

The following command line options are supported:

### **/CFG=**

Sets a specific configuration file.

Example: C:\SASLAB\RECORDER.EXE /CFG=config1.ini

### **/DIR=**

Sets the base directory. This directory overrides the directory specified in the configuration file.

Example: C:\SASLAB\RECORDER.EXE /DIR=C:\RECORDS

## **/DAT**

Adds the current date to the base directory path. If the /DAT option is used, the /DIR option must be specified too. Please note, that the /DIR option must precede the /DAT option.

Example: C:\SASLAB\RECORDER.EXE /DIR=C:\RECORDS /DAT

## **/AUT**

Starts recording at program start.

Example: C:\SASLAB\RECORDER.EXE /AUT

# **DDE Command Interface**

The RECORDER software accepts the following Dynamic Data Exchange (DDE) commands (`XTYP_EXECUTE`):

**play filename** : plays the specified .wav file

**playsub starttime endtime filename** : plays a subsection of the specified .wav file. The start and end time must be specified in seconds.

**play\_stop** : stops the current playback

**play\_pause on, play\_pause off** : represents the command Play/Pause

**play\_mute on, play\_mute off** : represents the command Play/Mute

**playlist\_start** : starts the playlist (Play/Playlist > play button)

**playlist\_stop** : stops the playlist (Play/Playlist > stop button)

**config filename** : loads the specified .ini configuration file

**start** : executes the command Monitoring/Start

**stop** : executes the command Monitoring/Stop

**split** : executes the command Monitor/Split .wav file(s)

**wavfilefolder pathname** : executes the command Monitoring/Start with the *Device* option *WAV File Folder*

**button1 on, button1 off** : activates or deactivates the trigger button1

**button2 on, button2 off** : activates or deactivates the trigger button2

**button3 on, button3 off** : activates or deactivates the trigger button3

**comment text** : executes the command Monitoring/LOG file comment

The application name is "RECORDER" and the topic is "main". If there is more than one RECORDER instance, the additional instances are referenced to by "RECORDER2", "RECORDER3" and so on.

DDE commands can be easily sent through the third-party command line application CMCDDE.EXE

([http://www.avisoft.com/DDE/RECORDER/RECORDER\\_DDE.zip](http://www.avisoft.com/DDE/RECORDER/RECORDER_DDE.zip)).

This is a sample employing the command line application:  
 CMCDDDE.EXE RECORDER main "play C:\data\soundfile.wav"

Sample batch files using the above CMCDDDE.EXE command line program can be found on the software installation media (folder *Thirdparty/DDE/RECORDER* or online at [http://www.avisoft.com/DDE/RECORDER/RECORDER\\_DDE.zip](http://www.avisoft.com/DDE/RECORDER/RECORDER_DDE.zip)

The software also accepts similar XTYP\_POKE DDE transfers. The available items are as follows:

Item	Data	Corresponding command
play	<filename>	Play/File...
play_stop		Play/Stop
config	<filename>	Options/Configuration management/Open...
start		Monitoring/Start
stop		Monitoring/Stop
split		Monitoring/Split .wav file(s)
wavfilefolder	<pathname>	Monitoring/Start, <small>Device = WAV File Folder</small>
button1	on <or> off	Trigger button1
button2	on <or> off	Trigger button2
button3	on <or> off	Trigger button3

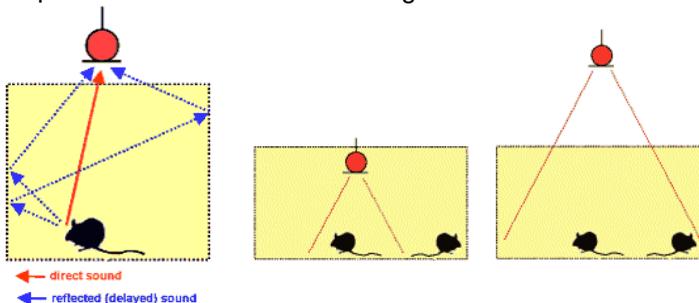
In order to allow a client application to check whether a .wav file is currently being played, the RECORDER software supports the DDE transaction XTYP\_REQUEST (data type: CF\_TEXT) on the item "play". The returned string is either "playing <filename>" or "not playing"

Item	Data
play	playing <filename> <or> not playing

# Recording and analysing rat or mouse ultrasonic vocalizations for behavioural, neurophysiological and pharmacological studies

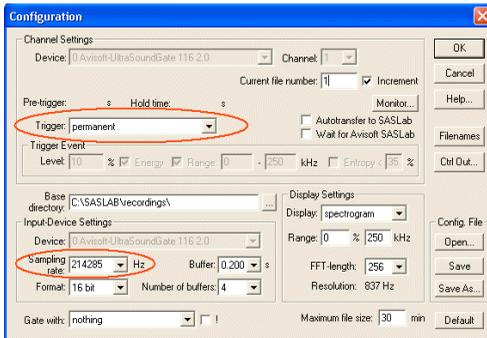
## Getting started

1. **Installing the Avisoft-RECORDER and SASLab Pro software** from the supplied CD ROM. At this time, the USB Hardlock dongle should still not be connected to the computer. If you have purchased the UltraSoundGate hardware, activate the option “**RECORDER USG/USGH for the Avisoft UltraSoundGate hardware**”.
2. **Connecting the Avisoft-UltraSoundGate unit** to an unused USB 2.0 port of the computer. Windows should automatically detect the UltraSoundGate device and install its device driver. Once the device installation has completed and the amber LED labeled "REC" on the UltraSoundGate device starts to flash, the RECORDER USG software can be launched (Start > Programs > Avisoft Bioacoustics> RECORDER USGH).
3. **Arranging the ultrasound microphone** close to the cage and connecting it to the UltraSoundGate device via the supplied extension cable. Note that larger acoustically reflective surfaces (such as the walls of a Plexiglas cage) surrounding the microphone and/or the animals may introduce unwanted reverberation. Also, keep in mind that the receiving angle of the microphone is relatively small at higher frequencies. Therefore, the distance between the cage base and the microphone should be adopted to the dimension of the cage base.

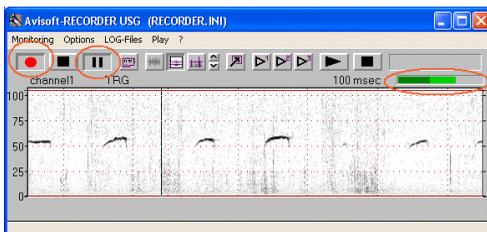


**4. Setting up the Avisoft-RECORDER USG software from the command "Options"/"Configuration..."**

Select the desired sample rate and trigger mode. The sample rate should be slightly higher than twice the maximum frequency of the USVs that should be analyzed. So, for a maximum signal frequency of 100 kHz, a sample rate of 214 or 250 kHz would be appropriate. For continuous recording sessions over longer periods of time (e.g. 30 minutes), the "Trigger" option "permanent" should be selected. In case there are only sporadic vocalizations that need to be monitored over several hours or days, the trigger option "level of this channel" should be used.

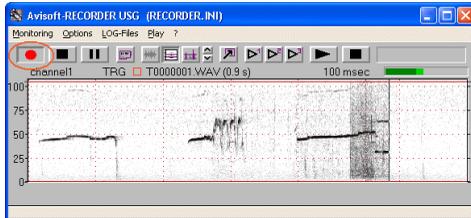


**5. Adjusting the recording level either from the level meter on the UltraSoundGate device or from the level displays in the RECORDER software. To do this from the RECORDER software, first click at the Pause button and then at the red Record button and adjust the red gain control knob on the UltraSoundGate unit until the level meter indicates an appropriate recording level. Any overmodulation (indicated by red color on both the level meter and the real time spectrogram display) should be avoided.**



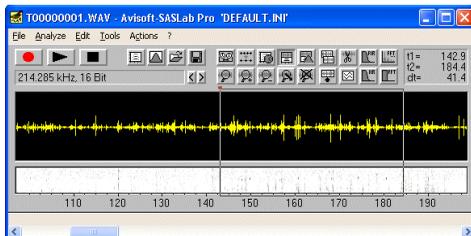
**6. Starting a recording session by clicking at the red start button (the pause button is not activated). The incoming sounds are now being recorded into the displayed .wav file (T0000001.WAV),**

which is located in the base directory that has been specified in the Configuration dialog box (by default *<user>/My Documents/Avisoft Bioacoustics*).

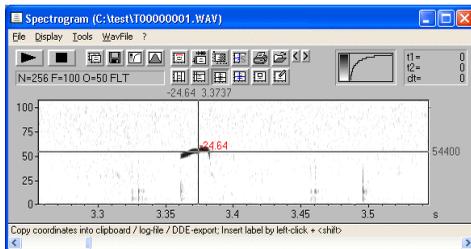


The currently running recording session can be finished by clicking at the Stop button (or by executing the command Monitoring/Stop). Click at the Start button again for initiating a new recording session (the file name will be incremented automatically).

7. **Viewing the recorded .wav files** with Avisoft-SASLab Pro. The recorded sound files can be opened in SASLab Pro from the command File/Open... or alternatively by drag&drop.



To see a detailed spectrogram display of a portion of the entire sound file, mark the desired subsection and create a spectrogram by executing the command "Analyze"/"Create Spectrogram".



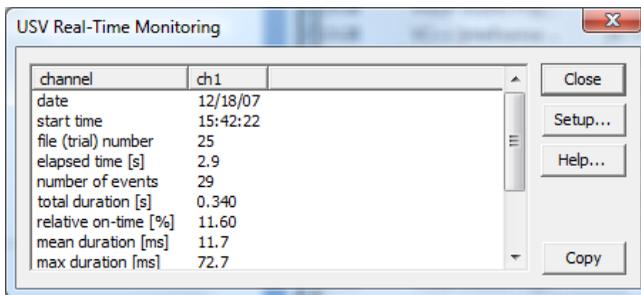
Various measurement cursors can be used to read out both time and frequency parameters.

## USV Real-Time Monitoring

The USV real-time monitoring tool provides instant results while recording ultrasonic vocalizations for behavioral or drug tests with mice or rats.

The implemented USV detection algorithm has been adapted to the poor signal-to noise ratios that are usually present in practical USV measuring setups. However, there should be only broad-band noise structures such the thermal noise of the microphone or the noise caused by the movement of the animals on the substrate. Any continuous harmonic noise components such as electrical interferences from computer monitors or switching power adaptors can prevent the proper operation of the USV detection mechanism.

In a typical USV test, the USV's are recorded continuously over a fixed period of time (for instance 10 minutes). This can be achieved by selecting the RECORDER trigger option 'permanent (limited)' from the configuration dialog box. The USV real-time monitoring option can be activated from the 'USV...' button on the configuration dialog box, which launches the associated USV monitoring setup box. Alternatively, the required settings can be made more easily from the menu command '**Options'/'USV monitoring setup wizard....'** (see page 48).



While recording the .wav file, the detected USVs will be indicated graphically on the real-time spectrogram display and the selected descriptive statistics items will be updated in real-time. When the trial has finished, the resulting .wav file can be found in the folder C:/User/Documents/Avisoft Bioacoustics/. The table containing the final descriptive statistics report is saved as a TAB-delimited ASCII file in the same folder under the same file name as the .wav file, but with the extension string 'usv\_stat.txt'. Additionally, the **Copy** button can be used to copy the ASCII-formatted report into the clipboard and then paste it into an Excel spread sheet for instance.

Note that the number and durations of the detected USVs will depend on the selected **USV detection settings**. In order to improve the results, it is possible to repeat the USV scan procedure with different settings on the original .wav file again (wizard option 'run from .wav file'). A more detailed analysis and any necessary editing can be done in the separately available Avisoft-SASLab Pro sound analysis software.

## **USV Real-Time Monitoring Setup dialog box**

**enable USV monitoring:** This option activates the USV monitoring tool. If activated, an USV Real-Time Monitoring report window will be launched once the monitoring is started from the command *Monitoring/Start* .

### **USV detection settings**

The implemented USV detection algorithm searches for steady whistle-like sounds having a relatively stable (peak) frequency course without rapid frequency modulations. The amount of frequency modulation that should be tolerated can be defined from the edit field *max frequency change*. It is expressed in spectrogram pixels and therefore depends on the currently selected FFT size and sample rate. The *min whistle duration* setting rejects potential whistles that are shorter than the specified duration.

**accept only monotonic structures :** If activated, only monotonic peak frequency shapes will activate the trigger, which will improve the efficiency of the algorithm in detecting whistle-like calls.

**max frequency change:** This is the maximum change of the peak frequency (expressed in spectrogram pixels or frequency bins) from one spectrogram time bin to the next that is allowed within a valid vocalization. Small values (for instance '1') will only detect slowly frequency-modulated vocalizations and provide the best immunity against undesired noise signals. Higher values (for instance '10') will tolerate more intensive frequency modulations, but involve a higher risk to interpret undesired noise as a USV.

**limit peak search to:** In order to exclude disturbing noise components outside of the expected frequency range of the USVs, the frequency range that should be processed by the USV detector can be limited here. If the option **limit peak search to** is not checked, the peak search is done on the full bandwidth signal, but the final USV detection is still limited to the frequency range specified here. If however the specified frequency interval

is relatively narrow, this option should not be activated because that would increase the false positive detection rate.

**min whistle duration:** The *min whistle duration* setting rejects potential whistles that are shorter than this specified duration. In order to prevent false detections (caused by noise), this value should be as large as possible (but shorter than the typical USVs you are looking for). In order to enable the proper operation of the whistle tracking algorithm, this duration must be at least three spectrogram time bins long (one spectrogram time bin corresponds to FFT size/sample rate).

**hold time:** The hold time parameter determines how closely spaced USVs should be treated. If the interval between two primarily detected vocalizations is shorter than this limit, the two will be reported as a single, but longer USV.

**min total duration:** This limit is similar to the *min whistle duration* parameter, but it is not used for the primary whistle detection. Instead, initially detected USVs that are shorter than this limit will be rejected. In case this additional limit is not required, it should be set to zero.

## report

The report section allows to select the descriptive statistics items that should appear on the *USV Real-Time Monitoring* report window:

**date:** The date of the trial (or that of the .wav file).

**start time:** The start time of the trial (or that of the .wav file).

**file (trial) number:** The file number of the current trial (or the filename in off-line mode).

**elapsed time:** The elapsed time from the start of the trial in seconds.

**number of events:** The number of detected USVs.

**total duration:** The total duration of all detected USVs in seconds.

**relative on-time:** The ratio of the total duration to the elapsed time in %.

**mean duration:** The mean duration of all detected USVs in milliseconds.

**max duration:** The maximum duration of all detected USVs in milliseconds.

**min duration:** The minimum duration of all detected USVs in milliseconds.

**variance of durations:** The variance of the duration of all detected USVs in milliseconds<sup>2</sup>.

**standard deviation of durations:** The standard deviation of the durations of all detected USVs in milliseconds.

**save the detected USV's into .wav file section labels** This option will create section labels for each detected USV, which are embedded into the .wav file.

**Default!** The default button sets all items to their defaults.

### ***Batch mode for rat or mice drug testing***

This batch mode supports processing large numbers of laboratory animals in drug testing procedures. The folders and filenames representing the animal ID numbers will be incremented automatically on each new batch. Once activated, a dedicated ***Rat/Mice Batch Processing*** dialog box will be launched each time a recording is started:

The ***Batch subfolder*** section allows to specify the subfolder name into which the data of the new batch should be saved. This section is only available if the option ***Create subfolders for each monitoring session*** on the *File Name Options* dialog box has been activated. If this option is not activated, the data of all batches will be saved directly into the base directory.

**Batch name:** This is the fixed prefix of the folder name.

**Batch No:** The current folder number. The ***increment*** option will increment the Batch number automatically on each new batch. The ***Reset*** button sets the batch number to 1.

The ***Filename*** section allows to edit the filename settings.

**Filename prefix :** Enter here the fixed prefix of the filenames (for instance "rat").

**first Rat ID No:** Enter here the ID number of the first animal. The ID number will be incremented for each channel (box) and each new batch. The ***Reset*** button resets the animal ID number to 1.

The ***Filenames*** list indicated the current filenames along with the associated channel names in brackets.

Deactivating the ***batch mode*** option will switch off the batch processing mode.

The recording process for the new batch is started by clicking at the ***Start*** button. In case the recommended trigger option ***permanent (limited)*** has been activated for all active channels, the ***Rat/Mice Batch Processing*** dialog box will be launched again immediately after finishing the previous batch. The ***Cancel*** button will end this loop.

## Bat call trigger filter

The bat call filter functionality can help optimizing false triggering on undesired environmental noise signals in passive bat monitoring applications.

The filter mechanism is primarily based on a whistle tracking algorithm that tries to identify more or less pure-tone signals based on their spectral continuity (the amount and direction of frequency modulation) that last for at least a certain minimum duration. This kind of trigger mechanism will work best for relatively long calls with little frequency modulation (low sweep rate, CF calls emitted by bats flying in open spaces). In order to cover also shorter and more intensively frequency-modulated calls (high sweep rate, FM calls emitted by bats flying in cluttered habitats), the software allows to define two different sets of limits (**FM** and **CF**) for the sweep rates and minimum durations: one for short FM calls and another for longer CF calls.

Additional criterions such as the *frequency range*, *high/low frequency magnitude ratio* and the *magnitude threshold* can further limit the trigger events in order to exclude undesired sound events from triggering.

On the *Configuration* dialog box, the bat call filter option is available on the **Trigger** type “level of this channel” and can be configured from the **bat call filter** section on the right-hand side of the *Trigger Event* section. Alternatively, the **Bat Call Trigger Filter Settings** dialog box can be launched from the menu “**Options**”/”**Bat call filter trigger settings...**” (shortcut Ctrl+B).

Bat Call Trigger Filter Settings

enable bat call filter

accept only monotonic structures

OK Cancel Help... Default

FM sweep rate and duration limits

min sweep rate max sweep rate min duration max dur.

FM : -20 -76.3 kHz/ms -1 -3.8 kHz/ms 1 10 ms

CF : -3 -11.4 kHz/ms 2 7.6 kHz/ms 4 40 ms

Spectrogram settings

FFT size: 256 Overlap: 75 Window: FlatTop

resolution : 0.256 ms x 0.977 kHz

frequency range: 15 - 250 kHz  limit peak search to this range

high/low frequency magnitude ratio > 6 dB fc = 15 kHz

magnitude threshold : -66 dBFS hold time = 50 ms

entropy threshold : 35 %

apply these settings also to all other channels

The **Bat Call Trigger Filter Settings** dialog box includes the following elements:

**enable bat call filter** : This option activates the bat call filter mode.

**accept only monotonic structures** : If activated, only monotonic peak frequency shapes will activate the trigger, which will improve the efficiency of the algorithm in detecting whistle-like calls.

**FM sweep rate and duration limits** : This section defines the limits for the maximum frequency modulation from one time bin to the next and the minimum durations for both the FM and CF call types. The sweep rates are expressed in frequency bins of the spectrogram. The resulting sweep rate (kHz/ms) depends on the *sampling rate*, the *FFT size* and the *Overlap* settings and is displayed on the right of each frequency bin edit box.

**min sweep rate**: This is the minimum sweep rate limit. Negative values represent falling frequency slopes.

**max sweep rate**: This is the maximum sweep rate limit. Negative values represent falling frequency slopes.

**min duration**: This is the minimum duration of the sweep, expressed in milliseconds. Note that the final internal resolution of this setting is limited by the temporal resolution of the real-time spectrogram.

**max dur.** : If activated, the potential bat calls will additionally compared with the specified maximum durations. If a signal exceeds the defined maximum duration limit, the signal will be rejected. Use this option to for excluding unwanted signals that look similar to bat calls but have a longer duration

**Spectrogram settings:**

The real-time spectrogram settings heavily influence the above sweep rates and their resolution.

**FFT size**: This setting is identical to the corresponding parameter on the Configuration dialog box.

**Overlap**: This setting is identical to the corresponding parameter on the Configuration dialog box.

The resulting time and frequency resolution of the spectrogram is indicated ms x kHz.

**frequency range** : This setting is identical to the corresponding parameter on the *Configuration* dialog box and determines the frequency interval that is processed.

**limit peak search to this range** : If activated, the peak search of the tracking algorithm will be limited to the selected *frequency range*. Deactivating this option (= peak search on the full spectrum) will lower the risk to falsely recognize broad-band noise signals such as rain drops or wind noise as valid bat calls.

**high/low frequency magnitude ratio** : If activated, the program will calculate an additional spectral parameter that can help to reject unwanted noise. It separates the frequency scale of the FFT spectrum at the specified frequency **fc** into two sections, determines the peak magnitude for both sections and then calculates the ratio between the two. A detected signal will then only be regarded as a valid bat call if the calculated ratio exceeds the specified threshold (expressed in dB).

**magnitude threshold** : If activated, this option will additionally reject low-level signals. The threshold corresponds to the *Level* threshold on the *Configuration* dialog box but is expressed in dBFS. This (red colored) threshold can be adjusted graphically by selecting the *Display* option “*energy in f-range*”.

**entropy threshold** : If activated, this option will additionally reject noise signals exhibiting a minimum Wiener entropy value above the specified threshold. This (green colored) threshold can be adjusted graphically by selecting the *Display* option “*energy in f-range*”.

**Hold time** : This hold time determines how close calls or calls with a short silent section in them are treated by the detection algorithm: If the silent interval between two consecutive calls is shorter than the defined hold-time, then the two calls will be treated a single call.

**apply these settings to all other channels** : If activated, this option will copy the settings for the selected channel also to all other channels.

**Default** : Sets all parameters to their defaults. Alternatively, these defaults can also be set from the command “Options”/“Configuration management”/“Presets”/“Bat Monitoring with bat call filter”.

Due to the complexity of the settings, it is recommended to start with the *Default* settings and perhaps disable the options “high/low frequency magnitude ratio” and “magnitude ratio”. The best settings for a specific set

of bat call echolocation calls can be found by testing and optimizing the various settings by trying it with .wav files that have been previously recorded. This mode of operation can be activated by selecting the "Device" "WAV file" on the RECORDER Configuration dialog box instead of the physical recording device.

Successfully detected bat calls are indicated by red markers at the top of the real-time spectrogram display.

While it is in most cases relatively easy to safely detect relatively long CF or QCF calls, it can be a challenge to distinguish very short FM calls by Myotis bat species (which are by the way not intended for long-distance echolocation by animals themselves) from other broad-band noise signals. It might therefore be that the detection of these short FM calls does not work very well. Under these circumstances it is recommended to use the normal threshold-based trigger mechanism instead (bat call filter and whistle tracking options disabled).

# Automated Call Parameter Measurements

The Automated Call Parameter Measurements tool can measure various call parameters both in real-time and off-line. It is however only guaranteed to work properly in off-line mode (Device = “WAV File” or “WAV File Folder”). In the real-time mode the results will depend on the rate of the incoming calls to be measured (high rates can lead to discontinuous processing and occasional loss of data).

The individual calls are identified either by the Trigger Event threshold **level** or any of the whistle-tracking trigger options (including the USV real-time monitoring and the Bat Call Trigger Filter). In order to get reliable measurements, it is therefore important to adjust the trigger setting for the intended application properly. The trigger settings can be optimized by using the Energy display (Options/Energy display), which indicates the call amplitude in relation to the current threshold level. In the whistle tracking mode, the detected calls are marked by red labels.

In addition to the amplitude threshold setting (or frequency contour criteria) the **hold time** setting (accessible from Options > Bat call trigger filter settings) will affect the way how calls are being detected. As long as there are short gaps within a call (or a series of close elements) that are shorter than the hold time setting, such short gaps will be ignored and all the closely spaced elements will be interpreted as a single call.

The option **enable call parameter measurements** activates the call measurement mode.

## Measurements

**interval** : Time interval from the start of the previous call to the start of the current call. This parameter will be rejected if the interval is longer than the group interval specified behind the option *indicate call groups that are separated by call intervals* >

**duration** : Duration of the call from start to end

**peak amplitude** : The peak amplitude (expressed in dBFS) of the call taken from the spectrogram or the peak-to-peak amplitude taken from the waveform in case the time domain option **tm** is activated.

**energy** : The energy of the call taken either from the spectrogram or from the waveform in case the time domain option **tm** is activated. If the *tm*

option is not activated, the energy reading will only represent a rough estimate of the energy of the dominant frequency component.

**minimum frequency** : Minimum (peak) frequency across the entire duration of the call.

**maximum frequency** : Maximum (peak) frequency across the entire duration of the call.

**mean frequency** : Mean (peak) frequency across the entire duration of the call.

**bandwidth (max-min)** : Call bandwidth calculated from the difference between the above maximum and minimum frequencies.

**Reject if amplitude < xxx dBFS** : If activated, soft intervals within a call (having an amplitude below the defined threshold) will be excluded from the minimum/maximum/mean frequency calculation. That way, erroneous readings can be prevented.

**start frequency** : Peak frequency at the start of the call.

**end frequency** : Peak frequency at the end of the call.

**best frequency** : Peak frequency at the location of the maximum instantaneous amplitude (*peak ampl.*) of the call.

frequency modulation

**minimum slope** : Minimum slope across the entire duration of the call, expressed in Hz/s or kHz/ms depending on the *show time in ms / kHz* settings.

**maximum slope** : Maximum slope across the entire duration of the call, expressed in Hz/s or kHz/ms depending on the *show time in ms / kHz* settings.

**time bins** : The instantaneous slope is calculated from the difference of the peak frequencies between consecutive spectrogram time bins. The *time bins* listbox determines the distance between the two time bins that are used for calculating the slope. The accuracy of the frequency modulation can be improved by increasing the number of time bins. Note that the resolution of the slope measurements is limited by the frequency resolution of the spectrogram and the number of time bins.

**curvature** : The average curvature of the call expressed in kHz. It is calculated by averaging the differences of the interpolated call frequencies (from start to end frequency) and the actual instantaneous frequencies at each time bin. In fm-cf calls, the curvature parameter is negative, while cf-fm calls exhibit a positive curvature.

**Trace**: The trace option will save the entire peak frequency course of the individual call. The trace interval (which depends on the sample rate, FFT size and overlap) will be added to the log file in order to reconstruct the original time scale. Note that the individual frequency samples will only be

saved into the log files and not displayed on the real-time display window.  
**max xxx bins = yyy ms** : This entry limits the maximum number frequency samples that are logged for each call. If this limit is exceeded due to the long duration of a call, then the saved trace is limited to this sample count.

**add relative time stamps of the above measurements** : This option will add the relative time of the above measurements relative to the start of the call (except for those parameters that apply to the entire call or specific locations, such as the mean frequency, bandwidth, start and end frequency)

**classification** : This option activates the call classification module that can be configured from the button **define call classes**. Calls that do not meet the criteria of any of the defined classes will be assigned to the class “?” The option **allow multiple classes** will assign the class names of all classes that meet the defined criteria to the call. Otherwise, only the first class that meets the criteria will be assigned to the call.

**BMU project extension for European bats** : Activates a special display style on the classification results table.

The total number of calls that have been assigned to each class will be displayed in real-time. If the option **indicate call groups that are separated by call intervals > xxx s** has also been activated, the most frequent class within the group will be assigned to that group and the total number of groups assigned to each class will also be displayed in real-time (on the column titled “groups”). The class count statistics will also be saved into log files whose filenames consist of the name of the underlying call log file and the suffix “\_stat”.

The option “**append class names to .wav filename**” will append the detected class names to the name of the .wav file. If there is more than one call class in the file, then all these class names will be added to the filename. If however the option “only single class” is active, then only the most abundant class name will be appended to the filename. The option “suffix” will insert the class name as a suffix. If this option is not checked, it will be inserted as a prefix.

**reject calls that are shorter than xxxx ms** : Detected calls that are shorter than this limit will be excluded from parameter measurements and will not appear in the log files.

**show time in ms** : If activated, time parameters will be provided in milliseconds instead of seconds.

**show frequency in kHz** : If activated, frequency parameters will be provided in kHz instead of Hz.

## **LOG file options**

**log file format** : The format of the log file can be either ASCII (.txt), HTML (.html) or XML (.xml)

**max listbox entries** : The results of the measurements will be displayed in a separate window in real-time. This list box determines how many of the recently detected calls saved in the window. More entries will occupy more RAM and may slow-down the program. If the limit is reached, the oldest entries will be removed from the list.

**add time stamp** : Adds the date and time of the detected calls. The date format can be selected from the menu **Monitoring/Log file date format**.

**ms** : Creates a time stamp at milliseconds resolution that is based on the number of samples acquired since the start of the monitoring. Otherwise, the time stamp is derived from the PC clock.

**add time offset relative the start of the file** :

**add .wav file name** : Adds the name of the current .wav file.

**include full path** : Adds the full path name to the file name.

**GPS coordinates** : Adds the current GPS coordinates (latitude, longitude and elevation) to the log file, if the GPS receiver communication mode has been activated from the command Option/GPS Settings....

## **limit the measurements to the periods when**

**.wav files are being saved** : If activated, the measurements will only be taken as long as a .wav file is being recorded. Note that calls that occur during the pretrigger period cannot be included.

**pause button is off** : If activated, the measurements will only be taken as long as the pause button is released.

**gates are open** : If activated, the measurements will only be taken as long as the selected "Gate with" criteria selected from the Configuration dialog box are met.

**create a new log file each day** : If activated, the current date will be added to the log file names, which means that a new file will be created on each new day. The related option **reset at the daily alarm time** will switch the log files at the selected *Daily Alarm time* (command Monitoring / Daily alarm actions...) rather than at midnight.

**create a new log file for each monitoring session** : If activated, a running number representing the individual monitoring session will be

added to the log file names. This means that a new log file will be created each time when the command *Monitoring/Start* is executed.

**create a new log file for each .wav file** : If activated, the log file name will represent the name of the currently recorded .wav file.

**indicate call groups that are separated by call intervals > xxx s** : This option can identify groups of calls that are identified by a call interval limit (elapsed time from the start of a call to the start of the next call) . A running group identification number identifies the detected groups. This feature can be used for instance for separating individual passing bats in long-term bat monitoring applications.

**save detected calls into .wav file section labels** : If activated, the occurrence of the detected calls within the recorded .wav file is indicated by automatically created section labels that can be afterwards viewed in the Avisoft-SASLab Pro software. The related option “**add call number to label**” will additionally include the call number to the label. When the software is processing previously recorded .wav files (*Device = WAV File* or *WAV File Folder*), then the labels will be saved into a separate .lbl file whose label entries are referenced to the original source .wav file. The labels in the .lbl files can be accessed in SASLab Pro by selecting the “auto save” option “into extra .lbl file” on the command *Tools/Labels/Label settings...*

**Counters** : There are a number of counters for log files, individual calls and call groups that are being incremented during the operation of the software. These counters can be reset to zero by clicking at the **Reset!** button.

## ***Call Classification Settings***

The selected measurement parameters that are being taken for each detected call can additionally be used to classify these calls into different categories, such as call types or species groups. The implemented classification mechanism is based on a simple comparison of the call parameters with a set of thresholds.

The list on the Call Classification Settings dialog box represents the defined call classes. Each row of that list represents a class that is identified by its **class name**.

A new class can be added to the list by clicking at the **new** button (there will be already an empty class named new on the very first launch of the dialog box). After adding a new class, its default name (new) should be renamed from the **class name** edit field. The selected class can be deleted by the **delete** button.

The call parameter thresholds of the activated parameters are listed at the bottom of dialog box and can be edited in the corresponding edit fields. The left row titled **from** defines the lower limits for each parameter and the right row represents the upper limits. A given call with its individual set of parameters is assigned to a specific class when all specified criteria are met (logic AND). If a parameter is not required for the definition of the class, then the corresponding fields can be left empty, which will ignore that parameter. It is also possible to define different sets of parameter thresholds with the same class name, which will create a logic OR link.

The created class list will by default be saved into the configuration file `.../Documents/Avisoft Bioacoustics/Configuration/RECORDER/call_classes.cli` when the OK button is clicked. Use the **SaveAs...** button to save the settings under a different name. The **Open...** button allows to open the settings from a previously saved `.cli` file. Alternatively, `.cli` files can be opened by drag&drop. The **Import...** button also opens a previously saved `.cli` file but additionally activates those parameters that were active when the `.cli` was saved.

The **<** and **>** button can be used to navigate through the list of classes. The **Reset!** button removes the entries of the currently selected class.

## dXML metadata input settings & database fields

The dXML metadata functionality allows to embed user-defined (XML-formatted) database fields into the recorded .wav files, which can subsequently be accessed from Avisoft-SASLab Pro.

The metadata input mode is activated from the command *Options / dXML metadata input settings*, where also the structure of the database fields is defined:

**enable dXML metadata input** : This option activates the metadata input mode. If activated, the **dXML metadata record** form will be launched.

**individual records for each channel** : If activated, this option will allow to enter individual data records for each channel.

**create sidecar files** . If activated, the dXML data will additionally saved into a sidecar .xml file having the same name as the associated .wav file.

The table below defines the individual dXML database fields. Each data field is represented as an individual row.

**field name** : Each data field must be given a unique field name, which cannot include special characters such as < or >.

**type**: A field can have one of the following data type attributes: *text*, *bool*, *integer*, *float*, *date* or *time*. The format of these data types is as follows:

text : abc123

bool : 0, 1

int : 123

float : 1.23

date: yyyy-mm-dd

time: yyyy-mm-dd hh:mm:ss

The date and time data types will automatically enumerate the current date/time in their corresponding combo boxes.

**size**: Defines the maximum size of the text, integer and float types.

**valid entries or pick list**: When the file type text is selected, this edit field allows to define a set of strings that are allowed to enter or that will appear as a pick list. The individual strings must be separated by commas (","). A list of valid strings (such as species names) can be imported by clicking at the **imp** button.

**validation:** If activated, only the specified list of strings will be accepted. Otherwise, the defined list of strings will be displayed as a pick list.

**imp :** Imports a list of strings from a text file into the **valid entries or pick list** field. The individual string in the .txt file must be separated by CR/LF control characters. The external .ext file can also be read by drag&drop (the cursor must be placed into the desired field before).

**lock :** If activated, this option will lock the input field (it will be grayed). Use this option to prevent unintentional modifications on the specific field.

A new field can be inserted by clicking at the **Add** button and the selected field can be removed by the **Remove** button. The selected field (marked by the check box in the row titled **select**) can be moved by using the **up** and **dn** buttons.

The **OK** button will save the defined data base structure into a separate .xml file (whose name is displayed at the bottom of the dialog button). The **Open** button allows to open one of the previously saved .xml database structure files. The **SaveAs** button will save the database structure under a new filename. The Reset button will reset all fields.

**reset after save:** If activated, this option will execute the **Reset** command once the **Save** command has been executed. This option should only be used in conjunction with the manual triggering mode where the metadata is entered after the corresponding .wav has been finished.

**Save on left click:** If activated, the **Save** command can also be executed by left-clicking anywhere on the client area of the *dXML metadata record* window.

**Wait for input:** If activated, the subsequent trigger events will be temporarily inhibited as soon as a trigger event occurs. At the same time the freeze display state will be activated. After executing the **Save** command the, software will resume to normal operation.

### **history**

**show:** If activated, this option will memorize all entered data and it will be possible to select these previous data from a pick list.

**reset:** Resets the history data.

In order to display the created data records properly in Avisoft-SASLab Pro, the same .xml data base structure file should be selected there (command Metadata / dXML database records / Define database structure... > Open...).

## ***dXML metadata record***

Once the option “*enable dXML metadata input*” from the command *Options / dXML metadata input settings* has been activated, this dXML metadata record input form will be launched. Depending on the settings, the content of the individual fields can be picked from a list and the entered strings will be checked according to the selected data type and validation options.

If the content of this form satisfies the pre-defined data type and validation constraints (indicated by green “v” characters), it will be saved into the corresponding .wav file when it is closed.

The menu of the *dXML metadata record* form provides the following commands:

**Text modules** : This command allows to pick text modules that have been defined from the main window command *Monitoring/Log file command text modules > Save as text module*.

**Setup**: Launches the *dXML metadata input settings & database fields* dialog box

**Reset**: Empties all data fields

**Save**: Saves the current record into the previously recorded .wav file. Otherwise, any changes made in the "dXML metadata record" panel will only affect the subsequently saved .wav files. The Save command can alternatively be executed by clicking somewhere on the client area of the dialog box.

[www.avisoft.com/downloads](http://www.avisoft.com/downloads)

