Taking a picture of underwater sounds

or

“How do I take a good underwater recording of whales and dolphins understanding what I am doing”

These pages were informally written to help those who are facing their first steps in the world of underwater bioacoustics, because, according to my boss "Even with very good equipment, just pressing the REC button is not enough to make good recordings...".
We will try to keep a general overview on issues related with field researches, well knowing that much more detailed and complete descriptions of each topic is available to the researcher on specific literature.
As we consider the internet as a widely-accessible updated resource, we end many chapters suggesting keywords the reader can use to put a seed in search engines, and find his own references.

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The environment

Underwater acoustic environment is really **not** isotropic.
Pressure (that changes with depth and other factors), temperature and salinity, considering the most influent factors, have severe effects on underwater propagation of sounds.
To predict the “quality” of propagation of undewater sounds is a demanding task, that requires in-situ measures (mainly bathy-thermography), good modeling of the surrounding waters and some software. Sad to say, this is something generally not available to bio-acousticians.

Sounds, that in an ideal isotropic medium would propagate spherically (acoustic pressure would decrease according to $4\pi r^2$, the surface of a sphere), change their “behaviour”, suffer refractions, some frequencies of the original signal may be selectively attenuated, and often, over distance, end concentrating in one or more “sound channels” if they exist.

Describing the general behaviour of travelling underwater sounds, we could say that they go, and stay, where the propagation speed is slow. Small differences and gradients in **density** (determined by pressure, temperature, salinity...) determine this stratification or layering of the sound distribution.
Sound, that travels at speeds around 1540 meters per second, remains “entrapped” at levels where one “slow” horizon is bordered by two “faster” horizons.
Here propagation can pass from spherical shape to a cylindrical one, suffering attenuation not according to a sphere surface anymore ($x / 4\pi r^2$) but to a cylinder surface: x / (height of the layer * $2\pi r$).

A fraction only of the acoustic energy crosses these layers and can therefore hardly be detected with sensors placed in positions other than the “sound channel” itself.
It goes without saying that sensors set in sound channels collect many more data than sensors set in other positions of the water column.

Acoustic intensities in the water have a different reference point than in the air. As these “zero” points are different, and deciBel is a measure unit referred to a “zero”, values in dBs in the water cannot be directly compared with those in the air.

Due to physical characteristics of the water (acoustic impedance), the same amount of power will produce in the water a level 35.5dBs lower than in the air.
At the same frequency, and due to the higher speed, wavelenghts in the water will be 5 times longer than in the air.

Underwater background noise distribution at sea is not linear. Lower frequencies have a much higher importance (about 110dB at 10Hz) than higher frequencies (35dB at 1000Hz). This noise is related to many causes, from seismic events, thermal effects and turbolence of moving water, in the low frequencies, to sea-state, ship’s traffic, marine mammals and precipitations in the high frequencies.

keywords: underwater acoustic environment, (underwater) sound propagation, sound channel, acoustic impedance, acoustic intensity, underwater noise

Sensors

As microphones are used to detect sounds in the air, **hydrophones** are used to detect sounds in the water. They are usually made of a piezoelectric sensor (a solid cristal or ceramic with the property of transforming pressure variations, what we perceive as *sound*, into electric current).
The electric signal, representing the sounds, is usually amplified within the hydrophone itself, by means of a small preamplifier, and then runs through the cable up to a second amplifier, to the recorder and/or to the researcher’s ears.
As single transducers, hydrophones are usually omnidirectional or nearly omnidirectional. This omnidirectionality could change according to the frequency examined.
Hydrophones are normally described with their specifications: frequency range, sensitivity, directionality, maximum operating depth and other fringes.

As the signals produced by Cetaceans (yes, this text will consider examples with Cetaceans...) range from a few Hertz to hundreds thousands Hertz, the frequency response of hydrophones used in Cetacean research normally goes well beyond human hearing, reaching, in many cases, 300 kHz.
Human hearing rolls down at 14-15-16 kHz depending on age and “hear abuse” (...attending heavy-metal concerts every week does not help your hearing..).
The frequency range is probably the most important parameter when choosing the hydrophone. The researcher must choose an hydrophone that is sensitive at the frequencies he is interested in. If one is interested in ultrasounds he must then choose a sensor whose sensitivity goes well beyond 20kHz. Some bibliographic research will help in finding the frequencies at which the target animals are acoustically active.
One apparent solution: let’s have the widest possible band!
It is not always true that wide-band hydrophones are better than narrow-banded. They usually cost more. Even much more. Their band (more frequency extension) can be paid with less sensitivity.

keywords: hydrophone, piezoelectric, microphone scheme

If one single hydrophone gives an idea about what, somewhere around, is happening, complex hydrophonic systems are often used to gain more spatial information and raise signal to noise ratios. They consist of multiple transducers (arrays of hydrophones) assembled to be more directional and sensitive. As far as *underwater bioacoustics* is concerned, there are two main kinds of setups: stationary mono/multi-hydrophonic configurations by which to control selected areas, and towed hydrophones/arrays to continuously detect sound during navigation.

Stationary hydrophones are used from steady platforms (from inflatables to large ships or fixed coast bases) and generally have good signal to noise ratios. The absence of hydrodynamic noise related to water flow (present in towed systems) usually offers good sound quality. If they are used from a moving platform (during surveys) they will require a good sampling protocol, as sounds present while going from station to station will be missed, and an almost still boat will be needed to have good performance of the sensor (time will be precious !!).

Towed arrays allow continuous surveying in large areas while cruising. The simplest towable hydrophone is made of a single sensor, usually with a long cable (more than 100 meters), designed to have a good hydrodynamic shape and an electronic attenuation in low frequencies. It is not, literally, an *array*, as you will need two or more hydrophones to have an *array*.
Simple arrays are made of two hydrophones, usually setup in an oil-filled pipe. The pipe helps minimizing the hydrodynamic noise. Arrays can give stereophonic signals, allowing to determine "angles" from which the sounds appear to arrive. Ears, when trained a little, are usually good in performing this task.

Arrays used for military purposes, i.e. to detect submarines at great distances, may be very efficient and expensive. They are often made by hundreds of sensors and they are connected to very powerful processing systems able to perform beamforming in real-time.
Beamforming is a technique based on the processing of signals detected by a series of sensors to produce a plot of received energy vs arrival direction; in other words, a beamformer allows to show the energy of received signals in a "degree *vs* time" plot. Thus it is possible to separate sources at different angles and to listen to a specific direction only.

Arrays for studying marine mammals' sounds are often based on a small series of sensors to keep low the complexity, the weight and the cost of the array. A small array can be deployed and retrieved by hand from a small platform such as a motorsailing boat.

What do I find when I ask for an hydrophone, and what is it good for ?

Hydrophones are usually made of a sensor (a cylinder or a ball) with a cable that comes out of it. Sometimes a waterproof connector is interposed between the sensor and the cable, or along the cable. At the other end there usually is a box, with switches, connectors and most of the times A BATTERY !

Connectors are used to connect (..it goes without saying...) to a recorder or other audio equipment (please check with manufacturer if the signal is compatible with a LINE IN or a MIC IN or both !).

A separate connector can be present for headphones (then a Volume knob should be present), another connector can be set for external power supply.

The sensor and the cable **must not be left in the sun** when out of the water. The sensor is quite a robust item, but direct sunlight, over a coating that usually is black, can produce high temperatures. And high temperatures can brake ceramics inside, brake inner cabling, brake external cable coating, alter physical characteristics.
It is a good practice to protect from direct sunlight the cable that goes from your recorder down to the water as well. An old T-shirt works good for this.
The cable is usually the most delicate part of both stationary and towed hydrophones

Waterproof connectors are waterproof when correctly tightened. You should rinse them, and the whole cable, in fresh water, after use. You should carefully clean everything before winter storing. Ask manufacturer about using de-oxidizers or silicon grease sprays in the connectors (they could be bad on your cable or connector).

“Microphone level” is lower than “Line level”. A Microphone output connected to a Line input usually results in a very weak or undetectable signal. If one person has Mic outputs and Line inputs only (and really needs to connect them !), he should use a preamplifier to pull up the signal levels.

Some recorders have a **filter switch** on Mic inputs. We talk about filters separately.

Most recorders have a **recording level** knob. Levels must be set carefully, and should stay around 0dB with an average signal. A *peak* indicator is often present, and should really seldom light itself if levels are correctly set.

If you want to connect **two headphones** to a single hydrophone, using a Y splitter, you should use the same brand and model of headphones. Using different models can produce unpredictable results in your ears, but can hardly brake anything.

If an **external power supply** is requested USE AN EXTERNAL BATTERY, NOT MAINS. Batteries are by far the best power source you can use for this kind of audio device. The battery you are using should be disconnected from any equipment other than your hydrophone. Using a 220/110 V power supply, an inverter, batteries connected to the boat or anything else, could produce unpredictable noises in your system, ground loops, 50 / 60 Hz buzzes, or simply misleading noises at variable levels and frequencies.

Filtering a signal

Filtering is bad, filtering is good, filtering is necessary, everything filters.

"Any medium through which a signal passes can be regarded as a filter"
(http://www-ccrma.stanford.edu/~jos/filters/What_is_Filter.html).

According to this, a sound propagating in the water is first filtered by the water itself.

To filter means to attenuate (most of the times attenuate only, not cancel !) a part, in frequency, of the original signal. Filtering is frequently used to reduce the weight (intensity) of part of the spectrum, when this part is more intense than the part of the spectrum the researcher is interested in. Not only. Filtering is necessary in digital recorders to limit the acquisition upper band of the system. (check *aliasing* and *anti-aliasing filtering* in literature)

Filters can be, in general, high-pass, low-pass, band-pass and band-stop or notch. They do what they are named after: an high-pass lets frequencies over its value pass, a low-pass does the opposite, a band-pass lets a band between two frequecies pass while a band-stopper or notch attenuates a specific band.

Filters have a frequency value at which they start operate (frequency of intervention), and a typical steepness. The steepness is the amount of attenuation the filter is giving to the signal along the frequency axis.

Example

> A low-pass filter is declared to attenuate signal 6dB per octave starting at 1000Hz. Remebering that one octave is the space that separates one frequency from its double (or half) and that 6dB is the space that separates one intensity from its double (or half), the reported filter will operate on the signal attenuating the intensity by half for every doubling of the frequency.

Filters with a 6dB attenuation are called "one-pole filters", or "first order filters ". Two poles filters, or second order filters operate a 12dB attenuation, and so on.

Example.

> The spectrum (the distribution of intensity versus frequency) of undewater noise shows that low frequencies (less than 50Hz) are very intense. This noise is mostly created by the movement of water masses and is everywhere as it propagates very well. Its level grows lowering the frequency.
> Let's imagine a researcher who uses an ideal hydrophone that equally and correctly transforms any frequency into its electric image. This hydrophone would correctly work with intensities going from complete silence up to a certain amount of pressure or dBs. We can name this feature of the hydrophone "dynamic range". If our dynamic range is 100, the noise at very low frequencies could be 70.
> This would mean that only 30 units of his dynamic range could be used to rapresent signals in the same frequency band of the background noise. It would be very easy to fill these 30 units and reach the maximum dynamic range of the sensor. Once the dynamic range is filled, the sensor would not correctly rapresent ANY frequency. Not even the high ones.
> This is why most of the hydrophones have a built-in high-pass filter that attenuates low frequencies.

Filtering has nothing magic in it. You will **never** be able to have a recording cleaned from unwanted noise if this noise is in the same frequencies you are interested in. If in a recording, for example, you are looking for whistles that range from 5kHz to 12kHz, you can use a band-pass filter that acts on these two limits, reducing signals outside these limits, but the noise within these limits can not be reduced without reducing the signals themselves.

keywords: audio filter, signal filtering, anti-aliasing filter

Signal storing

Signals can be recorded on analog and digital devices. These devices store the electric image of the signal as converted prom the environment by the transducer. A good recorder must record signals without alterations matching their dynamic and frequency ranges and preserving their features. **Analog tape recorders**, both compact cassette and open-reel recorders, degrade the signals they record by adding hiss, distortion, frequency response alterations, speed variations (*wow and flutter*), print-through effects, and drop-outs.

Digital recorders get rid of these problems. Within the dynamic range and the frequency limits, due to the number of bits and sampling frequency they use, they record and reproduce signals with great accuracy, low noise, good frequency response, and no speed variations.

DAT (Digital Audio Tape) standard is based on 16 bits of resolution and a sampling frequency of 48000 samples/second to allow, respectively, about 90 dB of dynamic range and a frequency response of 10Hz-22kHz. Some DAT recorders allow two additional modalities: 16 bit at 44100 s/s (frequency response up to 21kHz) to allow a direct mastering of CDs and 12 bit at 32000 s/s (frequency response up to 15kHz) to double the recording duration of the tape.

DAT recorder delivers a sound quality slighty better than CD in a small, easy to use and easy to store, long duration, tape-based format. And, since DAT is a standard, it is easy to exchange tapes with others. Also, digital transfer to and from a computer allows duplication, editing and analysis in an entirely digital domain. Digital transfer from DATs to computers is allowed by specific boards with digital I/O capabilities.

...not even DAT is anyway perfect! ...Both recorders and tapes require care and tape duplication is a safe measure to preserve valuable recordings.

Some new products for portable direct digital recording are today available. We are talking about MP3 players that can record sounds on an hard disk as ordinary WAV files or compressed files. They are very portable, better than a standard tape cassette but not as good as a DAT, battery consuming (this is a great problem), and not that expensive. The Italian distributor of Creative Labs has offered one of their JukeBox (Nomad 2) for a trial. It proved to be acoustically, when recording, not as good as we hoped. Now a third generation improved product is available, and on its 20GByte hard disk more than 30 hours of uncompressed sound can be recorded with a much better quality. Products from other brands are available but not tested by us. These products all have a digital interface to transfer your files to a computer.

Long-term storage is quite a thrilling topic, and while digital media when even partially damaged is often totally unrecoverable, analog systems generally still offer more stability over time. With much lower quality.

Recordist must be aware of the requirements of their analysis and of the limitations due to their entire recording chain (acoustic transducer + cables + recorder + tapes). While it is important to choose components carefully for their specific features, it is essential that their combination results in an optimally performing system. Even one single low quality component only, when added to a superb recording chain, will condition signals with its features, lowering the whole recording to its standard.

> **DAT recorders** deal mainly with audible signals: their frequency response allows very good recordings from low frequency signals, as low as 10 Hz, up to 22 kHz. Unfortunately, they can't be used to record ultrasounds: for frequencies higher than 20 kHz specific and expensive instrumentation recorders are required.
>
> DAT recorders with doubled speed and 96k s/s were developed to record and play two channels with 40 kHz bandwidth; other DAT models offer four channels with 20 kHz bandwidth or 8 channels with 10 kHz bandwidth.
>
> We have to consider that consumer DAT recorders are disappearing as they are replaced by MiniDiscs and MP3 recorders. Professionals concerned with high quality sound recording deal with expensive professional DAT recorders and emerging technologies such us optical disks (CDs and others), solid state recording, hard-disk recording.

Laptop recording
Recording on a PC, either desktop or laptop, may have great advantages. Quite inexpensive sound boards for desktops are now available with 192k s/s to provide more than 80 kHz of useful bandwidth while dedicated instrumentation acquisition boards can sample up to 500k s/s, giving a band of more than 200kHz.
For laptop use, USB and FireWire sound devices allow up to 8 channels at 96k s/s. A 30 GB hard disk can record 45 hours with DAT quality (16 bit, stereo @ 48kHz); larger disks can allow to record for weeks.

Some words about the DCC and the MiniDisc
These two digital media were developed to provide the consumer market with digital performance at low-cost. Both of them offer some very attractive features: the DCC (now discontinued) offered the ability to play old compact cassettes while the MD offers random access typical of all disc-based systems. They sound fine, with low noise and undetectable *wow* and *flutter*, but unfortunately are both based on sound compression algorithms which degrade the sound they record. Sound compression algorithms are based on the statistical features of music and human hearing: they discard all non-audible sound details.
For these reasons, even if they appear good, they are not suited to record sounds whose features don't correspond to the models on which compression algorithms are based.

Instrumentation recorders
Instrumentation recorders are typically suited to record signals whose frequencies are lower or higher than those audible by man. These instruments often allow recording many independent channels at a time (multi-channel recorders) and have several tape speeds to be selected in relation to the recorded frequencies: higher speeds for record higher frequencies. They can be both analog or digital. To record frequencies up to 100 kHz, analog recorders run at tape speeds up to 76 cm/s. Instrumentation recorders designed to record ultrasounds are very expensive and not well suited for field use; thus, cheaper devices to detect and possibly record ultrasound were developed to study echolocation in bats.
Instrumentation recorders are now replaced by dedicated or general purpose PC systems equipped with data acquisition interfaces and large hard disks. PC based systems can acquire and record signals from 0 Hz to many MHz.

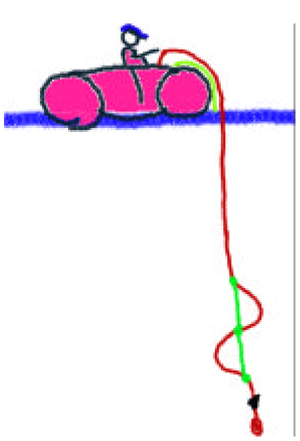
keywords: DAT recorders, digital audio recorders, AD interface, audio recorder

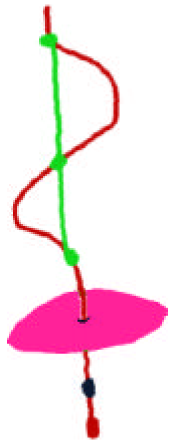
Using a simple single hydrophone

Using a single simple hydrophone can be a very exciting experience. Human ears are not made to detect underwater sounds, and even if it is true that if one person is swimming nearby a clicking sperm whale he will hear the clicks, most of the times the detected sounds are something new to our ears.
Once that the researcher has read these pages, understanding that nothing is *really* that simple and that many variables are acting on the propagating sound and on the equipment, he is ready to put the hydrophone in the water.

What will he do ?
The hydrophone cable will usually be shaked by the current, if any current is present. It will transmit noise and vibrations, and so must be somehow isolated from surface movements.
How ? using 1 metre of a simple elastic band (green) and a weight (black). The cable must be kept 50% longer than the elastic. The elastic should be fastened to the cable in three points (a couple of meters before the hydrophone), keeping the cable loose and S shaped. The weight (much better an

hydrodynamic shape) must be fixed short before the hydrophone. To give the hydrophone an higher inertia in the water, a 20cm disk (pink) can be fixed along the cable where the weight is fixed as well.
Again, isolate the cable from the boat using something soft, and protect the cable from direct sunlight.





Try to listen at different depths, going down to the thermocline bottom (sometimes more than 40 or 50 meters). As explained before, sound propagation can be very tricky and hard to predict.
If you are using a recording system always connect your headphones at the end of your recording chain. Sounds could be very good at the beginning, but could be distorted somewhere along the chain.

Which frequencies are interesting ?
If an excellent hydrophone with a 150kHz band is connected to a DAT, signals up to 22kHz only will be recorded. If connected to a digital 96kHz sampling system, signals up to 46kHz will be correctly recorded. If connected to a 500kHz sampling system, with a 200kHz low-pass filtering (to avoid aliasing), signals up to 150kHz will be excellently recorded, frequencies from 150kHz to 200kHz will be recorded according to the typical hydrophone attenuation, and frequencies over 200kHz will be recorded according to the hydrophone attenuation added to the filtering attenuation. Frequencies near 250kHz and over should then be so low in intensity that aliasing should then not occur at all.
This example is reported to show that the hydrophone itself is not enough to define the instrumental limits, and the worst system of all is, not to be forgotten, are our ears, limited to 16-18 kHz.

Arrays of hydrophones

As spatial information is lost using a single hydrophone, "dipole" arrays are used to retain basic information about spatial distribution of sound sources. Two or more hydrophones can be set in geometrically defined positions and can be used to determine the direction from where a sound is arriving. This can be done statically, using two or more stationary hydrophones, or can be done dinamically, using towed arrays of hydrophones.

These setups have extensively been used by military underwater surveillance systems, and are now available to scientific teams with medium budgets.
While who is writing has almost no experience with stationary arrays, towed dipoles has exensively been used by the team he works with. An efficient array is easy to deploy, and can be operated from

non specifically equipped vessels. A depth-meter, connected to the pressure transducer in the array, displays the operating depth in real-time so that it is possible to strike a known "propagation channel" and to evaluate the timing of surface reflected echoes. The whole instrument setup, including a laptop computer and additional equipment and interfaces, is powered by batteries and photovoltaic panels.

Sound analysis briefly

Introduction to sound analysis

Sound analysis allows to display the features of acoustic signals graphically, and, thus, to understand and measure their structure and to correlate it to observed species, behaviours and situations. Spectrographic representation of animal voices has been widely used since the first analogical analysis instruments were developed for military acoustic research.

The transformation of signals in the digital domain allows a new approach in the management of the data, thus easing operations of filing and analysis in connection with both the listening and the real-time display of the signals.

A number of analysis techniques are available; usually, they are based on dedicated digital systems or are carried out with general purpose computers equipped with suitable analog-to-digital conversion devices and specific Digital Signal Processing (DSP) software.
The simplest graphical displays are the **oscillogram**, which shows the waveform of the signal, and the envelope, which shows the amplitude of the signal in regard to time.
The most significant analysis is the **spectral** one, as it shows the signal frequency composition: instantaneous spectrum (frequency-amplitude plane) shows frequency components of a short segment, while the representation of more spectra, computed on consecutive segments, shows the evolution in time of its frequency structure. This is achieved by showing the spectra in an ordered time series, representing them, for instance in a three dimensional space (frequency-amplitude-time).
The most effective and easy to understand display is the representation on the frequency-time plane, with the component intensity coded through a colour scale. This kind of analysis is usually called spectrogram, or SonaGram™ since it was first realized by the Kay SonaGraph™. It is largergly used to analyze animal sounds as well as human voice.
Since spectrographic analysis, actually based on the windowed FFT (Fast Fourier Transform), is unsuited to analyze some non-stationary signals due to the uncertainty principle, several other processing techniques (zero-crossing, wavelet, Wigner-Ville) have been developed to resolve the frequency-time structure of complex signals or to accomplish particular tasks.
Using graphic representations, one researcher can easily compare signals to find similarities, to classify them according to their morphology, related behaviours, supposed meanings or individual emittors.

Equipment for sound analysis

Several instruments on the market allow to analyze animal sounds; most of them can acquire and store a signal segment to be later processed, analyzed, displayed and/or printed in one or more graphical formats. Basic hardware must include Analog-to-Digital (AD) and Digital-to-Analog (DA) converters with at least 16 bit resolution, selectable sampling frequencies up to 50000 s/s to allow analysis up to 22 kHz, and sharp low-pass filters to avoid aliasing.
More advanced and expensive equipment should include fast CPUs and highly optimized software, fast AD/DA boards to acquire ultrasounds, a DSP board to speed-up some intensive computation, a digital I/O board to directly connect an external digital device such as a DAT recorder.

Many excellent AD/DA boards are now available to provide up to 192k s/s sampling with 24 bits of accuracy. As hard disk can now store more than 120GB, PC recording can be an effective alternative to stand-alone audio or instrumentation recorders.
A number of programs running under Windows allow to record, edit and play-back sound files, although only few of them allow to visualize in detail their acoustic structure and can be effectively used for bioacoustic research.

DAT to PC connection

Most high quality PCI sound boards now include digital I/O capabilities to transfer a recording made with a DAT to the PC digitally, without any further conversion. A digital connection allows to transfer the recorded materials to the PC and back without any further degradation related with additional - and not needed - AD and DA conversions. In this way the quality of the recording is completely preserved.
The most advanced boards have both electrical (SPDIF and AES/EBU) and optical (TosLink) I/O; some also have an ADAT interface for connecting 8 channels ADAT peripherals.
Other than on PCI boards for desktop PCs, digital I/O is also available for notebooks through many USB audio interfaces and the recently introduced FireWire audio interfaces.

It is important to carefully verify the digital I/O capabilities of the selected board: some cheap devices don't perform direct digital transfer of data but they do perform a real-time sampling rate conversion (SRC) in the digital domain. This might be sometime useful, for example to downsample to 44.1kHz a 48kHz DAT recording for CD mastering purposes. Unfortunately, in these devices, sample rate conversion may occur even if transferring a 48kHz DAT recording to a 48kHz sound file!! This unnecessary data conversion introduces artifacts which corrupt the quality of the original data.
Often technical documentation does not provide information about this problem and only an accurate test can show this kind of behaviour.

Software for sound analysis - Windows

The software we developed and use as an example (freely distributed in a basic version at http://www.nauta-rcs.it) is named ***SEA wave – sound emission analyzer***, runs in win98/Me/NT/2000/XP and uses any installed windows-compliant sound device, including digital I/O boards and 96/192 kHz sound boards.

> The full version includes analysis, recording and display tools, with real-time spectrogram and cepstrogram, spectral averaging, frequency tracking, event counting, scheduled recording, etc..
> Other features include: 1 or 2 channels, real-time spectrogram display, real-time cepstrogram display, wrap-around or scrolling display, wide control on all analysis parameters, frequency-time cursor while in real-time mode, FFT size up to 16k points, frequency zoom capabilities with real-time spanning, frequency tracking, file analysis, file play with real-time display, drag & drop file play, play list management with real-time file concatenation, file recording with real-time display, display up to 1600x1200 pixels, screen save in .bmp format. Sound files and spectrograms are saved in a standard format to allow further processing with other software.
> To fullfill the requirements of acoustic surveys, recording capabilities have been expanded by adding scheduled recording capabilities, on event recording, gps position logging, and a continuous survey mode to record a file each hour until all the available disks space is filled.

Depending on the installed sound acquisition devices, analog I/O is allowed in the audio frequency range and/or in the ultrasonic range up to 48 and 96kHz (96 and 192 kHz sampling respectively). Digital I/O by means of PCI boards, USB devices or FireWire devices is also possible to provide direct transfer from DAT recorders to the PC.
An advanced version, named ***SEAdaq***, has been developed to support very fast sampling rates for studying ultrasounds extending to more than 150 kHz. This version uses National Instruments DAQ boards with sampling rates up to 500k samples/second. DAQ devices include PCI boards, PCMCIA cards and FireWire data acquisition peripherals.

High resolution real-time capabilities, typically available in much more expensive instruments, are very useful in field experiments to monitor the acoustic activities of the emitting subjects (immediate correlation among observed behaviours and emitted/received signals) and to optimise the instrumental setup (minimisation of noise, transducer placement, verification of the recording chain). These facilities allow to immediate evaluation of the results without waiting later analysis.
A portable version based on a notebook can be easily moved across laboratories or used in on-field applications, for example those requiring real-time visualisation and recording of acoustic events. Real-time capabilities and the continuous recording to disk with GPS position logging are particularly suitable for the continuous monitoring of underwater environment and for carrying out wide area acoustic surveys. By using a GIS it is then possible to map survey tracks and to plot the detected acoustic events on a map.

The classification of received acoustic events, either biological and artificial sounds, is still a challenge. A trained observer is normally able to correctly classify basic sound categories in real-time by join listening with spectrogram observation; though this is a very demanding task and requires skilled operators.
Doing the job 24/24h for long periods may be prohibitive. This is why it is required to develop reliable sound classification algorithms for working in real-time or for post processing long recordings. Such algorithms have been developed for very specific uses and can't be applied to generic tasks yet.