

Ultrasonic recording system without intrinsic limits

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Abstract

Today state-of-the-art bioacoustic research requires high-sample-rate, multi-channel, and often long-term recording systems. Commercial systems are very costly. This paper proposes and demonstrates an ultrasonic recording system (URS) design that is arbitrarily scalable. The system is modular and based on retail components and open source software/hardware. Each module has four microphones and modules can be combined to extend the coverage area, obtain higher spatial recording resolution and/or add recording redundancy. The system is designed to have no inherent scalability limits.

The system has been deployed in four different test settings. The first setup tests the system's ability to make medium-term recordings (1-2 minutes) with many microphones. The second setup tests the robustness of the system, being deployed throughout the Danish winter with only minor issues. The third setup integrates the system in a mobile robot as an echolocating guidance system, while the fourth setup demonstrates full-spectrum transducer calibration.

In most respects this system's hardware specification surpasses all competitors on the market at a quarter of the price. Tests demonstrate that large deployments are feasible and accurate ultrasonic measurements can be obtained.

1 Introduction

Today state-of-the-art bioacoustic research requires high-sample-rate, multi-channel, and often long-term recording systems. One application of long-term

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recordings has been when planning large structures, because there are increasing concerns about the impact of such structures on local bat populations. The concerns involve, for example, habitats, foraging grounds and migratory routes. Many methods of assessing impact are used: carcass counting post-construction [Fiedler; 2004(Barclay et al.; 2007], radar tracking [Bruderer and Popa-Lisseanu; 2005(Kunz et al.; 2007], infrared thermography tracking [Hristov et al.; 2008] and passive acoustical measurements, usually with one or more bat-detectors. Each of these methods has benefits and drawbacks.

When trying to survey a large area, none of these methods covers all of the following important aspects: 1) estimating how many bats have passed by 2) estimating what species have passed by 3) estimating their flight paths and 4) supporting automatic unsupervised long-term survey. Carcass counting is done too late and does not reveal the flight behavior, death rate and reaction to the structure. Radar tracking and stereo infrared thermography can give suitable data, at least for large species, but availability and price make them infeasible; multiple bat-detectors could, with some effort, be used to automatically count passing bats and estimate their species — but unless they are synchronized, flight-path extraction is difficult.

In principle, an acoustical array would provide all the information needed as bats echolocate when commuting or hunting. Bats hunting for insect prey out in the open emit high-intensity calls [Jensen and Miller; 1999(Surlykke and Kalko; 2008]. The multi-microphone array technique proposed exploits this fact. This gives the system an inherent bias towards loud bats; but windmills present the largest danger to bats out in the open hunting or migrating [Baerwald and Barclay; 2011(Barclay et al.; 2007] and since those bats are among the loudest this bias might not be a large issue.

The URS (Ultrasonic Recording System) market of today can only cover these requirements at high cost, and therefore we propose this new design. This new microphone array system is designed to fulfill the above needs at a fraction of the price of commercial or research alternatives.

1.1 Criteria for an ultrasonic recording system

An important goal of this work is that the system should be applicable to a wide range of experiments. The design chosen is therefore open-ended and the first prototype implementation is based on a versatile analog-to-digital and digital-to-analog platform. The design criteria we set up to ensure the system is useful for the anticipated applications and goals are listed below. The criteria are divided into three categories: *critical* criteria are considered essential — that is, if one of these is not fulfilled the system would not be useful; *important* criteria are features that the system might work without, but much emphasis has been put on achieving those goals; the *nice* criteria were implemented if convenient (Table 1).

<i>critical</i>	<ol style="list-style-type: none"> 1. Record ultrasound. 2. Easily adaptable to new experiments. 3. Record for long periods. 4. No inherent scalability limits. 5. Pre-trigger recording. 6. Data to estimate flight-paths.
<i>important</i>	<ol style="list-style-type: none"> 7. Real-time access to signal data from computer programs. 8. Use generic mass-market components leading to easy upgrade and low cost. 9. Use well defined protocols and standards to facilitate interoperability.
<i>nice</i>	<ol style="list-style-type: none"> 10. Platform agnostic. 11. Playback of ultrasound. 12. Save data in well-defined format. 13. Generate real-time event log of data.

Table 1: System requirements for an ultrasonic recording system. The criteria are ordered according to importance and placed into three priority groups: *critical*, *important*, and *nice*.

1.1.1 *Critical Criteria*

The system should be: capable of recording ultrasound for long periods and easily adaptable to new experiments. It should have no inherent scalability limits, which implies that all aspects of the system should be extensible i.e. more microphones, more storage, and more computers can be added with a linear increase in spatial coverage and performance.

Pre-trigger recording, i.e. recording of events happening n seconds prior to triggering requires that n seconds of recording are continuously stored in a round-robin or ring buffer. The data in the buffer is permanently stored upon a trigger event. This is essential for recording unpredictable events such as animal sounds.

In order to recover flight paths, a minimum of four synchronized channels is needed. This follows from the time delay of arrival equations [Madsen and Wahlberg; 2007(Spiesberger; 2004)] from which one recovers the 3D position of the emitting source.

1.1.2 *Important Criteria*

Real-time access to the data during long-term recordings enables the system to record only bits that satisfy pre-programmed criteria, e.g. sections identified by a matched filter bank of relevant bat calls.

The system should use generic mass-market components leading to easy upgrade and low cost. It should communicate with well-defined publicly available formats and standards to facilitate interoperability, both in direct inter-system communication and indirectly through stored data.

1.1.3 *Nice Criteria*

The design of the system should preferably be platform agnostic, so it can be implemented on the available hardware. It should be capable of ultrasound playback. It would be convenient if the system could generate a real-time event log of recorded data: interesting events should be time-stamped, analyzed, and recorded in a streamable log, which would facilitate real-time display and analysis of array recordings.

2 Related work

Recording ultrasound requires hardware which is capable of sampling data at sufficiently high rates [Luke; 1999(Shannon; 1948)] but this generates large data volumes. Thus, the system should be able to handle large data volumes, while at the same time being versatile and rugged.

When recording ultrasound in the field there are limited options with regards to equipment. The system described here is compared to several commercial solutions and one university-developed solution called VoxNet (Allen et al. [2008]) with respect to the design criteria listed in section 1.1.

The existing commercial solutions can be divided into two categories: multi-channel triggered arrays, and single- or dual-channel long-term recording systems. The comparison of these systems and the one proposed in this paper is presented in Table 2.

The commercial solutions perform fairly well with regards to some of our criteria, but not all. Either they are too simple (like bat detectors) or, if they can deal with multiple channels, they are inflexible and become prohibitively expensive when scaling them up. The VoxNet system is both versatile and arbitrarily scalable, sharing some important design choices with the new system described here, but VoxNet could not record ultrasound without a major re-implementation and partial re-design.

2.1 Hardware

Within the **triggered** array category, a popular system among biologists is the Avisoft AD converter, e.g. the USG1216H [Avisoft Bioacoustics; 2012]. Similar systems from other companies, such as National Instruments, Tucker/Davis etc., are commonly in use. Such systems are used for obtaining many synchronized acoustic samples from many places at the same time, though usually for short periods. The AD converter is fed from an analog processing chain comprising analog microphones, filters, pre-amplifiers, and amplifiers — such as the Avisoft analog chain used in Gillam et al. [2007] and Dzal et al. [2011] or a chain based on G.R.A.S./B&K equipment as used in Sorainen and Rytönen [2002], Kirchner and Röschar [1999], and Surlykke and Kalko [2008]. The exact choice of system and optional hardware depends on the requirements and resources available.

There are now a number of different **long-term** recording systems, which are primarily used for surveys assessing bat activity at an area of interest, e.g. when planning to build large constructions. Systems evaluated include the SM2BAT+ from Wildlife Acoustics [2012], HD-P2 from TASCAM, a division of TEAC America, Inc. [2012], the Batcorder from ecoObs GmbH [2012] and detectors from Petterson Elektronik AB [2012]. These are single- or dual-channel systems which are often acoustically triggered and store their recordings on-board. They are usually rugged and can be deployed outdoors to capture data continuously for months.

The academic solution referred to is the VoxNet embedded sensing platform [Allen et al.; 2008] and its predecessor the ENSBox [Girod et al.; 2006a], used in bioacoustics by, for example, Ali et al. [2007] and Collier [2011]. VoxNet and ENSBox are based on networked nodes where each node records data and, if appropriate, propagates it back to a network master. Thus these two use a different design from any of the systems described above. For networking they rely on a mesh network topology, a type of networking where each node must not only handle its own network communication but also serve as a network relay for other nodes [Wu and Stojmenovic; 2004]. This means that as long as each node can “see” enough other nodes, it can communicate with *all* other nodes. Each node has on-board storage, a processing unit, and a wireless link to nearby nodes. The system scales very well, is flexible and cheap (easily adaptable to

	Record ultrasound (1)	Flexibility (2)	Long-term (3)	Scalability (4)	Post-trigger (5)	Minimum channels (6)	Real-time access (7)	Generic (8)	Interoperable (9)	Platform agnostic (10)	Emit ultrasound (11)	Event log (13)
triggered	✓	(✓)	✓	✓	✓	✓					✓	
long-term	✓		✓		✓							
VoxNet		✓	✓	✓		✓	✓	✓	✓	✓		
mclurs	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	

Table 2: Comparison of the different recording systems, with the criteria from Table 1 as keys in the table column (numbers refer to the corresponding point in Table 1). The compared systems are shortened thus: **triggered** refers to commercial triggered arrays, **long-term** to mono- or dual-channel long-term commercial recording systems, **VoxNet** is a flexible network system and **mclurs** is the proposed system described in this paper. Check marks in parenthesis, i.e. (✓), are qualified statements as follows. (**triggered**, flexibility): although technically possible, these systems are usually hard to apply to problems that they were not designed for. (**triggered**, scalability): these systems are in principle scalable — if multiple AD converters are connected with a common trigger distributed recording is possible; however the recording length would be limited by local storage capacity. This option soon becomes prohibitively expensive if a large number of channels are needed.

new problems). Long-term recording to local storage is possible, and recording can be triggered by analyzing the incoming data stream. The system includes its own stream programming language Newton et al. [2008]. Depending on the iteration of the system, it is based on a Slauson Single Board Computer or the smaller Mica Mote and a fair amount of custom-made electronics and protocols.

VoxNet’s design is modular and distributed, like the one presented here. However, it cannot record ultrasonic signals in its present form and therefore does not satisfy our critical criteria; its extensive dependence on custom components means that to adapt it for ultrasonic recording would be a major redesign.

2.2 Software

Recording acoustical data in real-time means handling continuous streams of data. Most of the systems mentioned above have developed their own proprietary solutions to handle the data streams, normally excluding any outside access. That is, if you have a concrete task involving a criterion you wish to apply to the real-time data (such as online classification, saving only some bits of the stream, triggering only when a matched filter hits, etc.) it is usually

impossible without involving the vendor of your product. Only the vendor’s accompanying software handles real-time streams, and the user is restricted to the built-in software solution.

This real-time restriction does not hold for the Voxnet system, where external real-time access was designed in from the beginning [Girod et al.; 2006ab(Newton et al.; 2008)].

3 Design

As guiding design principles, we have taken inspiration from Kelly Johnson (Skunk Works) and Douglas McIlroy (of Unix fame). Johnson said *always use the simplest solution possible* [Rich; 1995, p. 221 and 231] and McIlroy recommends (slightly modified) *creating components that do one thing and do it well* [McIlroy et al.; 1978].

It is also helpful to attain a higher level of abstraction in design, making it possible to ignore underlying complexity; this necessitates that the possibilities and limitations of the component interfaces are well understood. Another desirable trait is that the design should be platform agnostic, so it can be implemented in any operating system and architecture.

A schematic of the design can be seen in Figure 1. The system consists of multiple ultrasonic microphone array recordings systems, modules 1 to n . In each module, the transducers (microphones) convert pressure changes impinging on them into a voltage signal, which is amplified and filtered. The signal is then digitized in an analog-to-digital converter (ADC or AD converter). Finally the signal is passed to a computer, which further processes, stores, and/or forwards it.

As illustrated in Figure 1, all modules can be considered as *one* array recording system provided they all have a common time reference. In this way we can obtain an arbitrarily scalable transducer recording system. As the figure also shows, this system can then be controlled through standard networking technologies, such as LAN (IEEE 802.3), Wi-Fi (IEEE 802.11), etc., by a computer, smartphone or other device supporting the control protocol. The system can be set up to trigger acoustically and data can be forwarded to a Network Attached Storage box, i.e. a network-accessible hard disk. When recording continuously, the network bandwidth will limit how many channels can operate simultaneously, so to achieve arbitrary scalability local storage must be used.

4 Implementation

4.1 Analog filter and amplification

Designing custom hardware is not a trivial task, and we have sought to limit the exercise and make the system as generic as possible. This entails simple open design utilizing cheap off-the-shelf and easily replaceable components.

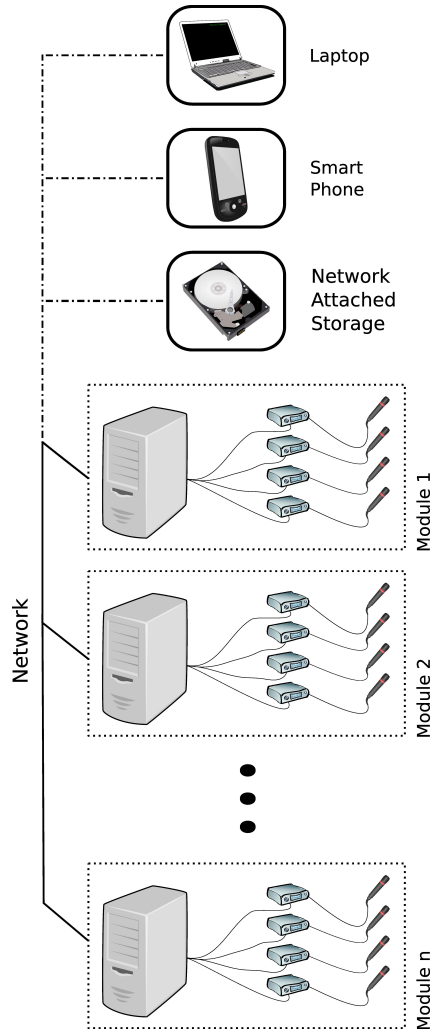


Figure 1: (color online) Block diagram of a generic modular ultrasonic recording system. Each module consists of a computer connected to 4 filter/amplifier boxes each with an ultrasonic microphone attached. Note that even though the modules have a limited number of microphones, no restrictions are put on the total number of modules. The analog voltage from microphones is filtered and amplified and AD converted into digital information which is then handled by a computer. The recording system can be controlled through standard network protocols.

Attaching ultrasonic sensors to a standard computer requires some specialized amplification and power supply; some customized hardware has therefore been developed. This is not needed for some high-end microphones, which are sold with filters, pre-amplifiers and amplifiers; though for these devices signal conditioning might still be needed to convert voltage levels to those supported by the analog-to-digital converter. We have emphasized making the system as open as possible, which is a well-understood concept in the software world. Open source hardware, on the other hand, is not so widespread although it is not a new phenomena, see e.g. Ackerman [2008].

Our custom-designed hardware includes bandpass filters and analog amplification for each channel [Brandt; 2012]. The analog bandpass filter for the system is a 2nd order Bessel biquad filter with 6 dB high-pass at 15 kHz and 6 dB low-pass at 150 kHz. The analog amplification on each input channel is programmable, which allows each channel to be attenuated or amplified individually from -20 dB to +60 dB. The hardware supports voltage signals up to $\pm 10V$ peak, so many different types of transducer are directly supported.

4.2 Microphone

The microphone used in our prototypes is the FG23329-PO7 from Knowles Acoustics with sensitivity, ≈ -73 dB re 1 VPa⁻¹ [Knowles Acoustics; 2012]. Originally designed for cell phone use, this microphone is usable for ultrasonic reception as it is sensitive up to approximately 140 kHz. It is also cheap (circa 40 USD) and small (2.54 mm in diameter with a 1 mm acoustic aperture), making it a near ideal choice given the criteria for the system.

As illustrated in Figure 2, 4 microphones are connected to the filter/amplifier box, using the connectors which have been used for all prototypes/versions of the analog part of the system. To make the system rugged we used IP68-certified connectors so that the joints are dust proof and submersible in water to below 1 m (Hirose Electric type LF07WBJ-6P and 6S). The microphone cables are commonly available shielded cable (Carol Brand type C0743A.21.10).

4.3 Analog to digital converter

For AD conversion we used a DAQ-2010 data acquisition board from Adlink Technology Inc. [Adlink Technology, Inc.; 2012], chosen for its hardware specification and relatively open driver software. It provides 4 channels with an analog-to-digital conversion rate of 2 MSample/s/channel at 14 bit/sample. In addition, it has 8 digital input channels and two 12 bit digital-to-analog channels with a maximum sample rate of 1 MHz/s/channel. It comes with proprietary binary Linux drivers.

4.4 System computer

The smallest computer that provides the required PCI port for the AD board is a Single Board Computer with the Mini-ITX form factor (motherboard size of

17x17 cm) defined by VIA Technologies [Via Technologies, Inc.; 2001]. Figure 2 shows the first prototype. From the node computer the customized hardware is connected to the AD board through an off-the-shelf cable.

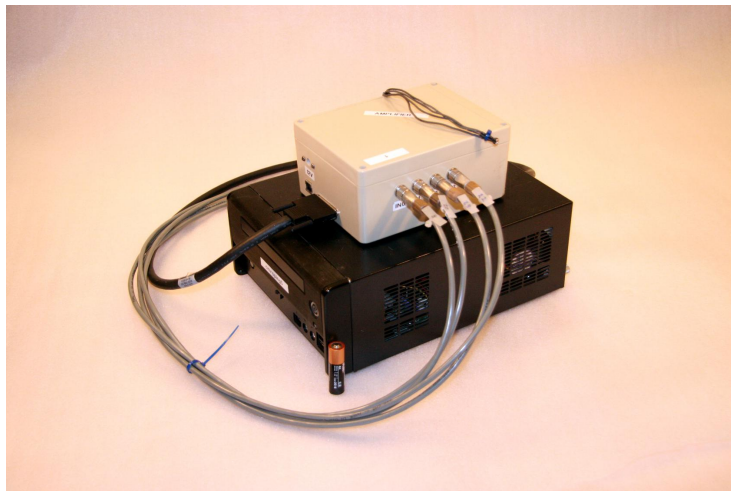


Figure 2: (color online) First prototype, AA batteries show the scale. The computer and AD converter are in the black box, whilst the custom filter and amplifier hardware is in the light box. Four Knowles microphones are bundled and placed on top of the filter/amplifier box.

5 Results

The system has been deployed in four different setups which demonstrate its capabilities, such as high number of channels, long-term recordings, synchronized playback/recording and operation under harsh conditions — recording right before the wet season in Panama with well above 90% relative humidity or recording during the Danish winter with sometimes freezing temperatures and winds as high as 20 ms^{-1} .

We deployed this prototype with several different transducers in the different setups and we took specific measures to deal with the large amount of data generated. In one setup we also attached a weather station to the prototype.

5.1 Multi-channel pre-triggered recording

The system was deployed to record *Noctilio leporinus* and *Micronycteris microtis*, flying in two outdoor flight rooms at the Smithsonian Tropical Research Institute’s field station on Barro Colorado Island, Panama in March 2010. Three modules were used, so a total of 12 channels were available. Figure 3 shows the spectrogram of a 1 s long clip taken out of a 1 minute *M. microtis* recording.

10 microphones were used; the eleventh channel recorded a synchronization signal for off-line synchronization with high speed video recordings.

N. leporinus and *M. microtis* have very different hunting behaviors. *N. leporinus* is a trawling bat, while *M. microtis* scans leaves for prey. Thus recording these two bat species poses different demands on the system. *N. leporinus* searches for prey on or over smooth water surfaces and once detected swoops down and catches the prey. To capture a whole hunting sequence requires a recording length of about 2–3 seconds with a 2 seconds pre-trigger time to be sure to capture the sequence from the very beginning. *M. microtis* can scan leaves for up to 2 minutes before finally taking the prey. This posed no problem for this system as it was a matter of increasing the pre-trigger buffer length to a couple of minutes.

5.2 Robust long-term multi-channel recording

Weatherproofing the electronics was achieved by mounting it inside a rain and dust proof IP65-rated box (Fibox type EK-W). The Knowles microphones used are weather resilient; pointing them downwards with a water guide gives them some water protection.

A four channel system was deployed outdoors from summer 2010 to summer 2011. There were only minor disruptions, and only one failing microphone, even though the system remained outdoors in Denmark throughout the winter. When the system is weatherproofed as described above, it is indeed seen to be very robust.

Long-term recording with the system was demonstrated using the same setup. The system was deployed for more than three months and recorded for approximately two of those months. Operational failures were due to accidental disconnection from the power grid by university ground staff.

One of the interesting recordings from the long-term setup was captured during the night between Sept. 26 and Sept. 27 2011. Figure 4 shows 2 seconds of processed data from the 2nd channel of this recording, 1 second long pre- and post-trigger buffers. The recording shows high frequency echolocation calls with main energy around 55 kHz interspersed with low frequency (down to ca. 20 kHz) social calls of a *Pipistrellus pygmaeus* [Barlow and Jones; 1997].

Long-term recordings generate large amounts of data, e.g. for a node with 4 channels sampling at 500 kHz:

$$4 \text{ channels} \cdot 2 \frac{\text{bytes}}{\text{channel}} \cdot 500\text{K} \frac{\text{samples}}{\text{seconds}} \cdot 10 \text{ hours} = 144 \text{ GB}$$

Analyzing such an amount of data manually would be very time-consuming, so automated methods are needed. We used a simple recording paradigm to reduce the amount of stored data: the system was set to a wakeup/shutdown cycle where it started recording at 8 PM and stopped at 6 AM. This timing was experimentally determined to cover the hunting period of the local bats.

When recording, only signals which exceed a triggering threshold are stored. The threshold was set at 5 times the noise floor of the topmost microphone.

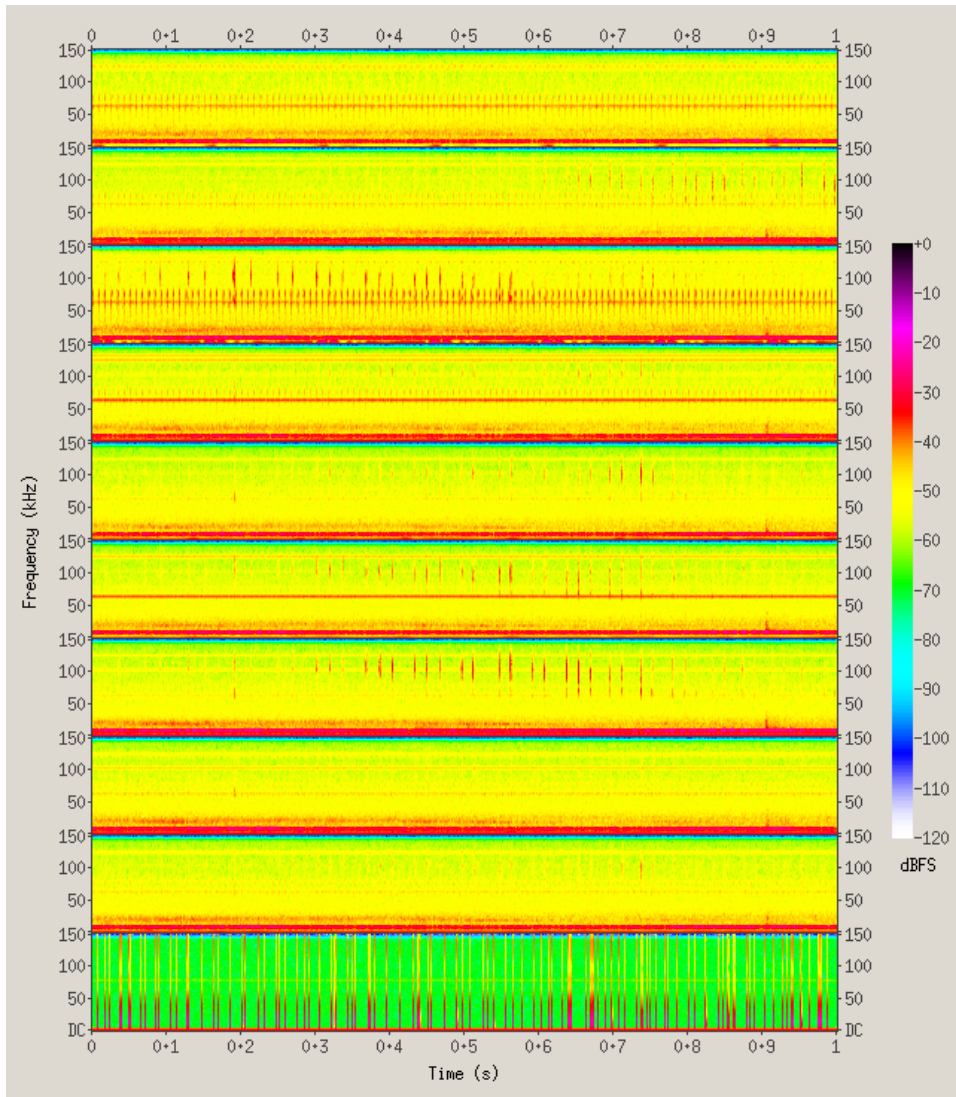


Figure 3: Multi-module recording done in Panama in March of 2010 (session 9 recording 21 offset 8.5 seconds). This recording is part of a *Micronycteris microtis* scanning sequence. The calls are the steep sweeps from circa 120 kHz to 70 kHz. Notice how it is only recorded on some microphones as it moves about. The recording consists of 11 synchronous channels. The 11th channel recorded a synchronization signal.

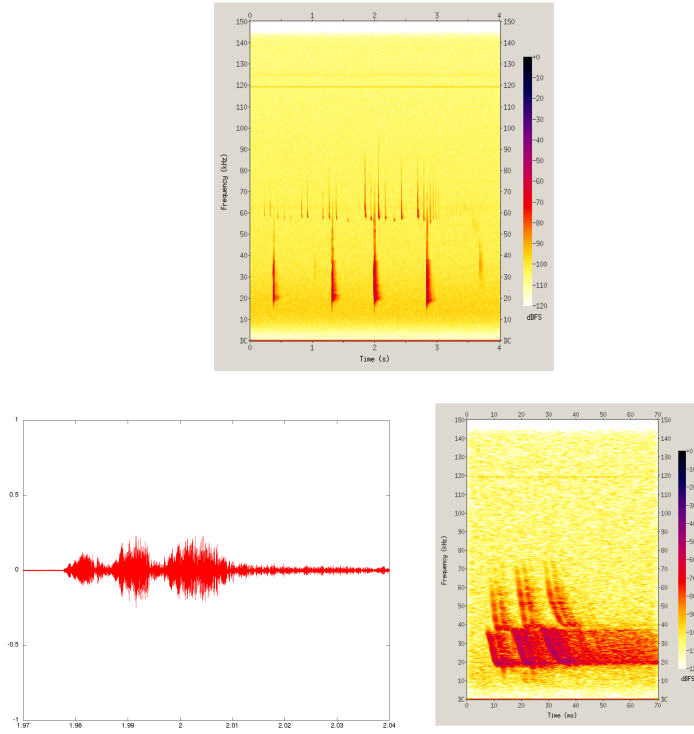


Figure 4: (color online) These plots show one of the recordings made with an outdoor setup on the roof of a University garage (Sep. 26, 2011). The top figure is a spectrogram of the whole recording (Hanning window with 4096 points), showing echolocation calls interspersed with social calls. Bottom figures show a small segment of the recording, from 7.5 ms before 2 s to 10 ms after, left is the amplitude plot and right shows the spectrogram (Hanning window with 512 points). Both spectrograms have been digitally low-pass filtered with f_c at 150 kHz. Note that there are significant reflections within the three chirps, which probably come from the rippled roof of the garage.

To trigger a recording, the threshold had to be exceeded for at least half a millisecond; at that moment the system stores the previous second's buffered data and continues recording until 1 second of silence occurs. More advanced triggers are being developed, for example acoustic profiles for a matched filter bank, which can then be used as acoustical triggers.

5.3 The echolocating robot

As the system readily supports synchronized playback and recording it lends itself to other applications. One of these was a sensor capable of echolocation implemented on a robot (Figure 5). The robot uses a Polaroid speaker and two Knowles microphones embedded in a 1.5 scale 3D print of a *Myotis daubentonii* head; the microphones are placed 1 cm inside each ear canal. The sensor successfully guided the robot to objects lying on a floor. Efforts are currently being made not only to find but also to identify these objects. This application demonstrates the flexibility of this AD/DA platform.

5.4 Calibration setup

The system has additionally been tested as a tool for calibrating microphones. Synchronization of input and output using the DAQ-2010 lends itself readily to linked emission and recording, which is not only useful for echolocation as described above but also makes averaging of data from many emission-recording cycles trivial. The frequency response of the emitter is recorded with a calibrated microphone in a reasonably anechoic place. By replacing the calibrated microphone with the microphone that needs to be calibrated, we can automatically acquire data to compute the transfer function of the microphone needing calibration.

The recording system will work with input from any transducer delivering voltages of ± 10 V peak, e.g. ultrasonic microphones from Knowles Acoustics, Avisoft, B&K, and G.R.A.S.

The analog output channels have a maximum voltage range of ± 10 V and have been used to drive a number of different speakers. The output impedance was chosen to be 50Ω to be compatible with the input for our custom-designed power amplifier for Senscomp's Series 7000 (22 mm) Polaroid transducer [SensComp; 2012]. The digital-to-analog converter has also been tested using a regular dynamic speaker and a ScanSpeak ultrasonic speaker from Avisoft.

5.5 Easy upgrade

As a demonstration of the easy upgrade path offered by this design, we produced a second prototype which is portable with low power consumption. This prototype is based on the open source/hardware AD converter USB-DUXFAST [Incite Technology Ltd.; 2012] and the DreamPlug computer [Globalscale Technologies, Inc.; 2012ab] which together offer a platform with very low power consumption and good technical specifications.

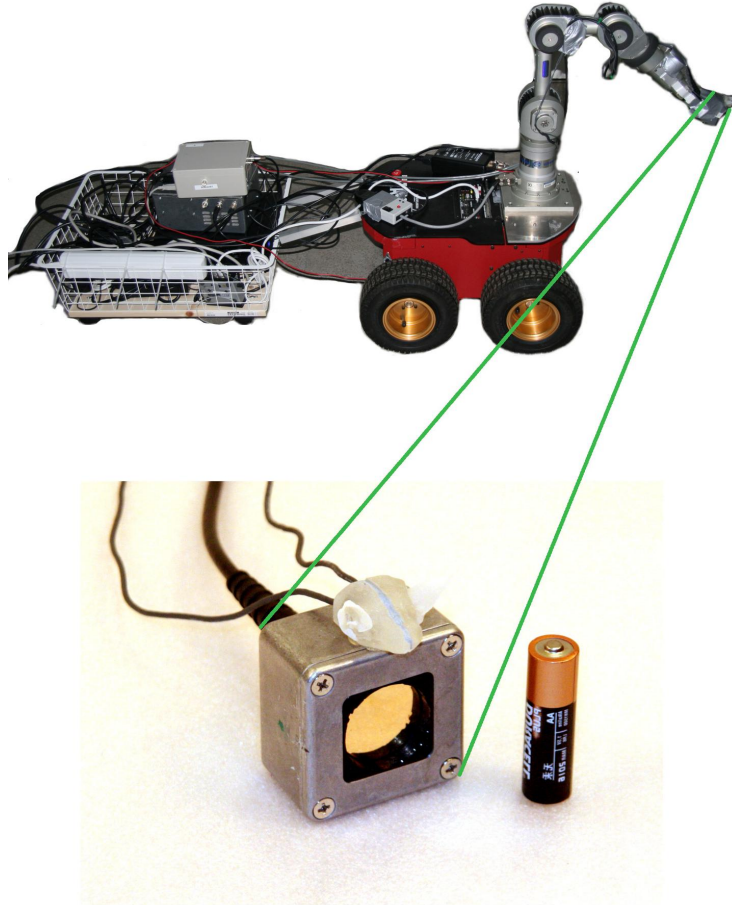


Figure 5: (color online) Echolocating robot, a demonstration of synchronized playback and recording. The robot consists of two robots joined together — a Neuronics Katana 450 robot arm with 4 degrees of freedom while the mobile robot is a Pioneer 3-AT robot from Adept Technology Inc. A blowup of the binaural echolocation sensor can be seen below, scale is shown with an AA battery.

This second prototype can be seen in Figure 6. All custom-built components from the first implementation were reused with small modifications. The only substantive modification is a change to the connector cable between the custom analog box and the AD converter, necessitated by the different termination connector on the USB-DUXFAST.

This prototype was optimized for portability, low-power usage, and many channels, thus making it useful as a multi-channel recorder for use in the field, or as a module in a distributed ultrasonic recording system. There are, however, some limitations with this prototype compared to the first one: the maximum sample rate is 750 kHz with 4 channels instead of 2 MHz; the bit depth is 12 bit compared to 14 bit; and the USB-DUXFAST does not include a digital-to-analog converter so playback is not possible. None of these limitations are significant for the ultrasonic recording application.

Table 3 compares the two prototypes and a commercial system based on the Avisoft UltraSoundGate type 1216H AD converter.

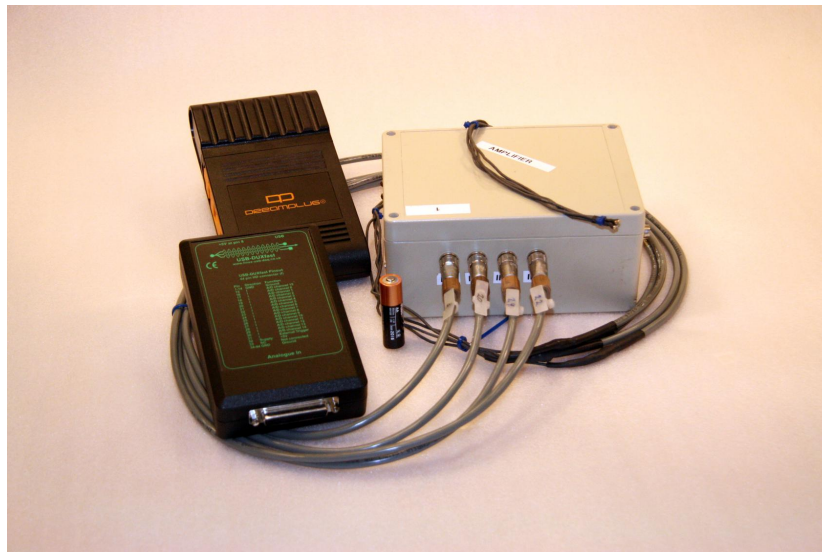


Figure 6: (color online) Portable, low-power prototype, scale shown with AA batteries. The computer is in the black and yellow box, while the AD converter is in the black and green box. The interconnection between the AD converter and filter/amplifier box still needs to be adjusted.

5.6 Logging other data

Each module includes a standard retail Single Board Computer, which will accept input from most available retail transducer hardware. As a demonstration of this, we sampled data from a weather station, the wireless Fine Offset WH1090 [Fine Offset Electronics; 2012], using the open source software

What (per ch.)	Avisoft	Proto. 1	Proto. 2
Sample rate	750 kHz	2 MHz	750 kHz
Bits per sample	12 bits	14 bits	12 bits
Variable gain	0-40 dB	-20-60 dB	-20-60 dB
Volume	240 cm ³	2200 cm ³	140 cm ³
Weight	260 g	1000 g	170 g
Power consumption	5.4 W	10.4 W	2.8 W
Price	\$2200	\$950	\$600

Table 3: Three array systems are compared based on their per-channel parameters: a system based on an Avisoft AD converter which needs a computer running Microsoft Windows, the first prototype based on the Adlink AD converter and the second prototype based on the USB-DUXFAST AD converter. Power consumption calculation of the Avisoft array includes a laptop using 20 W.

`wvsr` [Pendec et al.; 2012]. The recorded data includes time-stamped weather measurements such as wind speed, wind direction, precipitation, air pressure, temperature, and relative humidity, and is time-synchronized with the acoustic recordings.

Recording local weather conditions together with acoustical data enables us to correlate various climatic parameters with bat activity as well as providing the basic information (temperature, pressure and relative humidity) needed to calculate the local speed of sound. The system also allows easy integration of other types of data, for example light intensity, cloud coverage, wind turbine blade speed, etc.

5.7 Advanced use of the system

Using a Unix-inspired operating system on the modules' local computers makes available a large set of advanced high-level tools and paradigms. As an example, consider operating system pipes, which abstract data handover between programs and readily handle infinite data streams in a sensible way. Using such a paradigm it becomes possible to separate acquisition, analysis, multiplexing, de-multiplexing, and forwarding of the streams.

Below is a short working code example of what this might look like. The vertical bar (`|`) indicates a pipe, which links the output of one program to the input of another, running independently in parallel. The example uses linked programs to detect signals, to convert recorded data into Microsoft Wave PCM [Microsoft, Inc.; 1991] format, and to forward channel 1 and 3 to a different computer for further analysis. What follows is an actual working program chain; long command lines are split, for clarity, and the splits are marked with `\` tokens:

```
$ adac -r 500e3 \
| tee raw_data \
| signal_detect \
```

```
| sox -r 500e3 -c4 -t s16 - -t wavpcm remix 1 3 \  
| ssh -e none OTHER_HOST \  
    'further_analysis > outfile'
```

The chain works like this:

1. **adac**, a custom program, takes one argument, namely the sample rate. It acquires samples from the AD converter at the specified rate, 500 thousand samples per second per channel in this case, and emits them on its output stream.
2. **tee**, a standard GNU coreutils program, is an input replicator. It saves one copy of the data stream to a file named **raw_data**, whilst sending a second copy forward to the next linked program.
3. **signal_detect**, a custom program, conducts **local** real-time analysis; this is where matched filters and other data selection criteria might be applied, and where threshold-triggering is implemented when that is used. Data judged interesting by this program is passed along the chain; uninteresting data is dropped (but a copy remains in the **raw_data** file from the previous step).
4. **sox** is a powerful audio processing program with more than 20+ years of usage and active development. Here it converts channels 1 and 3 of the (interesting) raw data to Wave PCM format. Note that **sox** needs to be told the sample rate (**-r 500e3**), number of channels in the input stream (**-c4**) and sample format (**-t s16**) i.e. little-endian signed 16 bit. This information is encoded in the Wave PCM file format header.
5. **ssh**, a standard communication program, connects to the remote computer **OTHER_HOST** and forwards the Wave PCM-encoded data stream to a custom *remote* analysis program **further_analysis**.

The complete command chain is built up piecewise, offering flexibility and choice to the user, while allowing rather complex analysis to be accomplished with a chain of individually-simple steps.

6 Discussion

The proposed system design has been tested in various settings, and the results show that it is also applicable for doing large-scale recording. The framework is based on an open-ended design which has allowed the development of two markedly different systems without changing either the basic design or most of the software components. Another hallmark of the current design is the ease with which the configuration of prototype 1 was changed from, for example, multichannel recording of short events like the 1–2 second echolocation sequence of a trawling *Noctilio leporinus*, to long-term recordings of bat activity over a

whole season and finally to function as an input/output system deployed for two different purposes, robot echolocation and microphone calibration.

The ultrasonic recording system we propose is designed to achieve an unprecedented flexibility: as the interface is based on widely used open web standards, few limits are put on what type of equipment can communicate with the system. For example any system which has a web browser installed will be able to control and get feedback from the recording system — which implies, for example, that the system can be controlled by smartphones allowing flexible, wireless, universal access to the data. This has been tested by successfully running a terminal program on an Android smartphone. Simple preview, analysis, and control interfacing can be achieved by installing GNU Octave on the same type of smartphone; alternatively, more sophisticated analysis could be carried out on the module’s local computer and the smartphone be used only for display and control.

6.1 Benefits of the openness

Due to the openness of the system, there are no limits or restrictions to possible modifications and additions. Any user interface could potentially be used to control the system. The stream format is publicly available, so anyone can implement and integrate a user interface that would work with the ultrasonic recording system. Below we outline some of the perspectives in such an open, modular, scalable, and cheap system.

When scaling the system to arbitrary size, keeping track of individual channels or modules of the system becomes a book-keeping challenge. However, with an open framework everything is potentially programmable, and the system can be extended to deal automatically with this task. Electronically tagging microphones, filters and computers is straightforward and would allow system software to keep track of which components were in use in what combinations.

One useful possibility is the automatic determination of microphone array geometry, which is essential for computation of flight paths and is a major contributor to the effort of deploying a large-scale array. Using a small, loud impulsive sound source — such as an electrical spark — the geometry of an already-installed array can be calculated using time-of-flight computations to determine the positions of the (perhaps many) microphones while simultaneously calculating the positions of the sparks. Such software could be integrated into the array module local computers, making the array almost self-calibrating.

As shown by the tests we performed, the versatility of the system was sufficient to enable application to all of the experiments we intended. But an extra advantage of the open design is that it allows others to modify, remove or add components to the system and apply it straightforwardly to many other tasks which we have not yet anticipated.

6.2 Calibrated recording with interchangeable microphones

A second benefit of the possibility of electronically-tagged system components accrues if the tags can store acoustic calibration data. Each microphone could carry its own calibration information, as could each analog channel in each module. Plugging a microphone into a channel could then cause the automatic application of compensating (digital) signal processing so that a calibrated recording transfer function was realised independent of which microphone and channel were actually used. By this means, relatively inexpensive microphones might achieve the same recording tasks as expensive, stable, calibrated microphones.

A further question implied by this possibility is the stability of the microphone with aging and environmental conditions. The functionality of the Knowles microphones after being subjected to the Danish outdoor environment has been verified, but the stability of their transfer function over time and with use still needs to be determined. In particular, it is not known whether (or how much) the transfer function varies with temperature, humidity, and wind. Tagging microphones with their calibration data would go some way toward answering this question too.

6.3 Other possible applications

There is a number of other applications that this system could handle; some possibilities are outlined below.

Automatic flight tracking can be integrated in several different ways. In principle it should be sufficient to have 4 microphones placed in a non-planar arrangement, from which full 3D source position can be calculated. A tetrahedral structure's spherical symmetry offers position accuracy independent of direction, which may be important for environmental monitoring where it is unclear from which directions animals may approach. Adding multiple 4 microphone sets would improve position estimation for distant targets.

As noted by Guarato and Hallam [2009], the call of a flying bat can be reconstructed with reasonable accuracy if the bat's position can be determined, its emission directivity pattern is known (or can be approximated) and it emits its broadband call towards the center of a suitable array of calibrated microphones. As the proposed system is an arbitrarily scalable array, it would be possible to place a sufficient number of calibrated microphones in the environment to allow call reconstruction.

7 Conclusion

We have demonstrated a modular ultrasonic recording system design that is useful, inexpensive and versatile, and fulfills all our criteria for use in tasks involving recording or playback of ultrasonic signals for animals or animats.

The implemented system has been tested in a variety of application settings, and is able to record and emit ultrasound at sufficient rates and process the resulting data volumes. The use of open design allows for an easy upgrade path,

distributed deployment, long-term recording and emission, low-cost implementations based on retail components and open source software/hardware.

This system allows exploration of new areas of the ultrasonic soundscape. Multi-channel long-term surveys become possible, making it feasible to obtain more accurate assessments of presence and activity of bats or other sound-emitting animals. The flexibility of the system may also lead to applications not yet anticipated.

When building tools for academic exploration, open source software and hardware provide transparency and ease of design sharing, thus increasing the possible proliferation of successful ideas.

Acknowledgments

Part of this project was funded by the EU 7th Framework Programme (FP7), through the ChiRoPing project (ICT-2007-1 STREP project 215370). A special thanks goes to the OSS community which has brought about many good and useful things, many of which have been used in this project.

We also thank the Smithsonian Tropical Research Institute for allowing us to use their facilities on Barro Colorado Island, and also the biologists who helped us with many of the recordings.

We thank Ali Shekarchi who assisted with the debugging and use of the echolocator and microphone calibration setups.

A How to build a copy of this system

Taking the steps listed below would duplicate one of the described modules (not necessarily in order)

- Fetch schematics and gerber files from Brandt [2012] and send them to a PCB production company.
- Order components on the Bill of Materials list.
- Solder the filter/amplifier.
- Order AD board and PC.
- Order microphones, microphone cables, microphone plugs.
- Assemble the device.
- Install GNU/Linux on the PC.
- Install the proprietary driver from Adlink.
- Fetch and install the `adac` software from Andreassen [2012].

The authors are happy to assist if issues arise.

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