C. 3) RECORDING SOUNDS. (AS)

I) Low budget, portable media.

Computer Soundboard, Marantz (Flash Memory recorder), or SONY DAT recording from Batdetector with digital memory.

First: Calibrate Computer Sound board. Check for **anti-aliazing filter**. Use function generator to produce pure tones (or a sweep) above and below the max. frequency allowed. Most computer sound boards still sample at 44.1 and /or 48 kHz sampling rate, but sound boards with up to 192 kHz sample rate are available.

Transfer function of recording system: Bat detector with digital memory and Computer SoundBoard.

a) **Sound:** Produce a high frequency sound sweep from ca. 10-100 kHz (Exact function generator, amplifier, electrostatic loudspeaker).

b) **Reference:** Record the sound with a ¼" high frequency microphone (G.R.A.S. 40BF, Brüel & Kjær 2804 amplifier at +40 dB) on on of the high frequency systems from part II of this practical. Remember to note the distance between speaker and microphone. Record a calibration tone to get dB SPL.

c) **Recording:** Record the same sound with the bat detector (Pettersson), play back the recorded signal at 10 times reduced speed from the bat detectors digital memory and record that signal in the computer. Analyse the signal stored on the computer by digitizing on the Wavebook. Compare spectra from b) and c).

d) **Direction/external microphone.** Try to turn the bat detector to different angles and repeat c) – or try using the bat detector with the external microphone, and repeat c).

A similar procedure is used to calibrate high frequency microphones:
To compare microphones to the "standard" B&K ¼" microphone, repeat b) using the unknown microphone (See also Microphone practical for audible sound). Refer to the B&K calibration chart.

Other small cheap Media.
There are lots of music storage media (Dictaphones, MP3 etc.) Most use some sort of Adaptive Transform Acoustic Coding to reduce the size of the data.

If you have one, you can test the reliability in different ways:

1. Repeat c) for the DAT using your device instead of Computer or Marantz.
2. Compare recordings of "easy" signals (pure tones or signals changing slowly in time or frequency) and more "difficult" signals (signals with abrupt changes in time or frequency)

Suggested signals:
Single pure tone HF and LF (C)
Two-tone signals, eg. 12000Hz and 12040 Hz and 12100Hz (within and across more filters of Minidisc: 128 filters from 11-22 kHz) (C)
White or pink noise (M)
Sweep from 1 to 15 kHz (C)
AM-modulated signal: 6.5+7.5 kHz (C)
White whale (beluga) signal (C)
Signals from arctiid moth (C)
White noise (M)

Reduced data formats use Adaptive Transform Acoustic Coding is based on psychoacoustical results on human hearing, and reduces the amount of data by taking into account the human audiogram, and forward, backward and simultanously masking (- no need to store quiet tones simultaneous with loud tones if they are close in frequency or in time). Wav-files on the contrary are not compressed but stores all sampled words.

The development in digital media is fast and new systems are smaller, cheaper and faster, so soon cheap systems may also be satisfactory for scientific recordings of high frequency sounds.
See for example the BIOACOUSTICS-L email discussion list managed by Cornell University. (To subscribe send an email to <listproc@cornell.edu> the following message: subscribe Bioacoustics-L yourrealname)

RECORDING SOUNDS.

II) High frequency systems: **A/D convert and store digitized signal on computer**
Systems: **IOTech Wavebook, 8 channels, max. 1 mHz**
Avisoft Ultragate, 4 channels, max. 750 kHz
National Instrument DaqCard, 1 channel, max. 500 kHz (I/O)

Analyse all signals in Adobe Audition / BatSound (open files as "all files, bin, 16 bit mono, and change to the correct sample rate in BatSound). Save as Powerpoint.

a) **Frequency.**
Record high frequency pure tones and broad band signals at high and low sampling rate (Wavebook, notepad in program)

Suggested signals (generated with PM5152 function generator):
50 kHz tone
Sweep from 10 to 100 kHz (adjust upper frequency in standby mode)
Suggested SR (Wavebook): 50 kHz and 250 kHz

b) **Antialiasing**
Produce a sweep and set a sample rate at the Wavebook that does **not** follow the Nyquist sampling theory (SR > 2 x max. freq. in signal).
Use filter to set different LP-frequencies. How close to the max. allowed frequency should the filter be set?

c) **Dynamic range and Overload.**
Measure the dynamic range of the Wavebook – record a signal that is close to the max. amplitude, and check how much you can decrease the amplitude and still find the signal.
Increase the amplitude of the output signal (from PM5152) to overload and record on Wavebook.
Use "legal" sampling rates in c).

Note:
For short unpredictable sounds like many animal generated sounds digital systems like the Wavebook are almost ideal. They allow for recordings sounds that just occurred with FIFO circulating buffer storing pretrigger signals for as long as the memory of the system allows.
Very long recordings on many channels, especially of high frequencies, will demand extreemely large storage capacity (harddisk space). However, harddisks are dropping rapidly in price and increasing in size.

C. 4) Diffraction (Practical 4)

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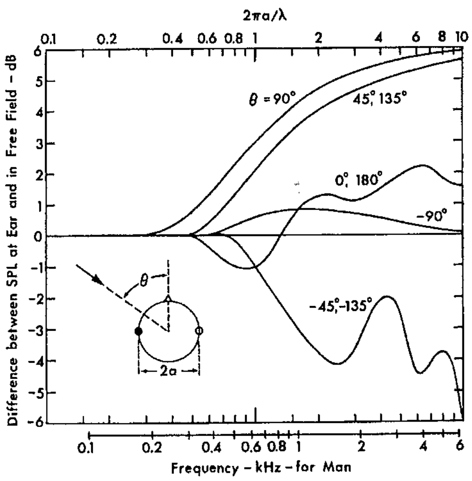
- Waves (light, sound) pass round obstacles that are small relative to the wavelength, as if the obstacle does not exist. Waves are partly reflected from large surfaces, and the reflected rays can be predicted from the geometry. The interaction between waves and middle-sized objects is much more complicated. In courses of elementary physics, some simple cases of diffraction (e.g., the end of a thin wall or a slit or hole in a thin wall) are often illustrated by means of Huygen's method of wavefront construction, but such examples do not give the students (or their teachers!) a realistic impression of the nature and effects of diffraction.

- The change of sound pressure (or light intensity) at the surface of a sphere hit by sound (or light) at various frequencies (that is, for various relations between the wavelength and the radius of the sphere) is shown in the following figure (note the frequency scale below for a (hypothetical) spherical human head):



But it is important to realize that the changes occur not only at the surface of the objects, but also in the spaces around the object. But how far away do changes occur? Which consequences do these effects have for the construction of acoustic setups?

In the practical we shall explore these problems. The equipment consists of a loudspeaker, various microphones and objects, and a Fast-Fourier-Transform frequency analyzer (the operation of which will be explained by the instructor). The participants have to design and carry out a number of experiments with this equipment and hopefully learn a lot about diffraction and other basic acoustical phenomena.

The outcome of the experiments should be described in brief with the help of drawings and the spectra that can be plotted from the FFT-analyzer.