uSampler
Project Number AR1778

In the 80’s, when I was a child, one of my firsts experiences with computer programming was a program called “type-a-tune” from the Atari 130XE owner’s manual. This program assigned musical notes to each key from 1 to 0 of the keyboard. Thus two of my main interests get together. Now I have the oportunity of join computers and music through this project.

A sampler is a device that can record and store audio signals adding effects like time stretching, distorsion, phaser, changes in pitch, etc. Firsts samplers were analog, they used pre-recorded tape loops asigned to a key in a keyboard but now the advance in analog to digital convertion and the develop of digital signal processing techniques and the miniaturization able us to perform all the task in a single chip.
The main characteristic of the uSampler (micro-sampler) is that with one sample of a musical instrument we can have all the scale through application of digital signal processing algorithms saving storage capacity in such a way that we can exploit the features of the microcontroller powerful processor. Besides, this Philips chip has large storage capacity and the necessary peripherals to acomplish communication with computers and humans.

Digital sampling

Digital sampling is the process of take samples of sound from musical instruments, voice, etc. at a given sample rate, typical sample rates are 44100 Hz (music) and 8 Khz (voice). This samples are encoded in PCM format to further processing.
Of course we never must forget the Nyquist-Shannon sampling theorem to avoid aliasing.

Hardware

The MCB2130 from Keil consitutes the main system. This board has a speaker which is used to verify the effects added to the sound sampled. The serial interface is used to debugging purpose and to input the commands from the keyboard through a terminal emulator (see figure 1). Also can be used without a PC through a PS/2 connector added to the board.

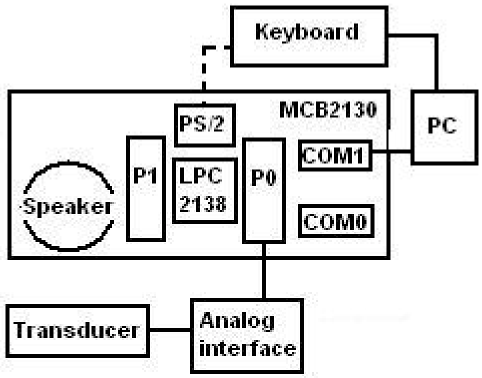

Figure 1 –Blocks of the system.

As de audio input I have used a microphone from a telephone hands free system together with an amplification and adaptation stage (see fig1), the output is connected to AIN0 through the pin P0.27.
Two operational amplifiers provide the signal adaptation, the first one, provide amplification to the signal of 220mV p-p generated by the signal transducer (a microphone electret) and the preamplifier, in this case a hands free telephone system. The second stage translate the signal 1.5V over to avoid the negative part and thus adapt it to the A/D input.
I used the LMC6084 and LMC6484 but it can be done with any general purpouse operational amplifier like LM324 .
The 1.5 votage comes from a votage regulator, it can be any voltage regulator or voltage reference IC.

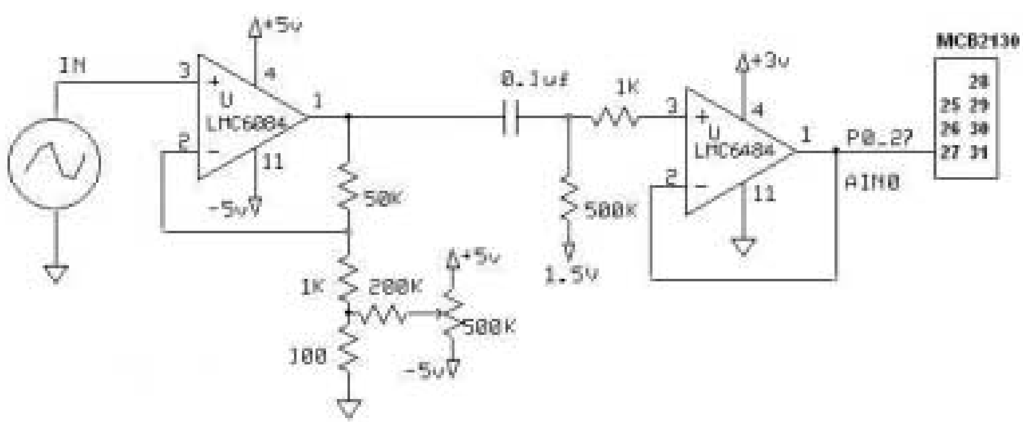

Figure 2 – Athe first operational amplifier should provide enough amplification and the second shift the negative part by 1.5V. The converter only accept 3.0V inputs.

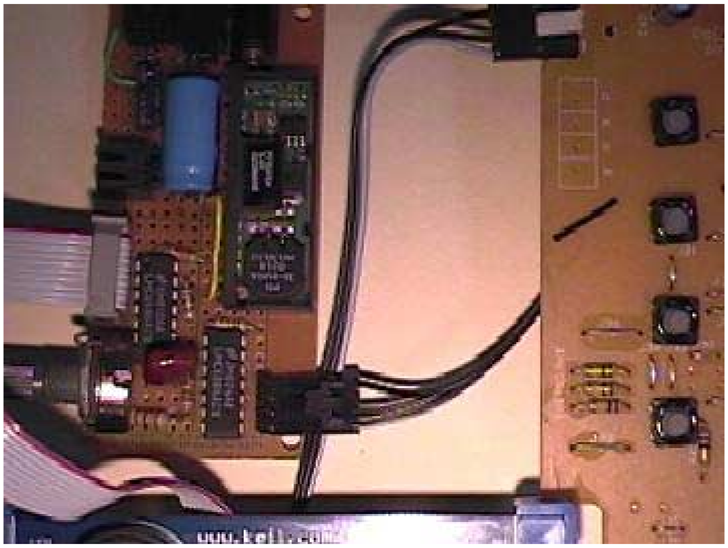

photo1. shows the analog interface.

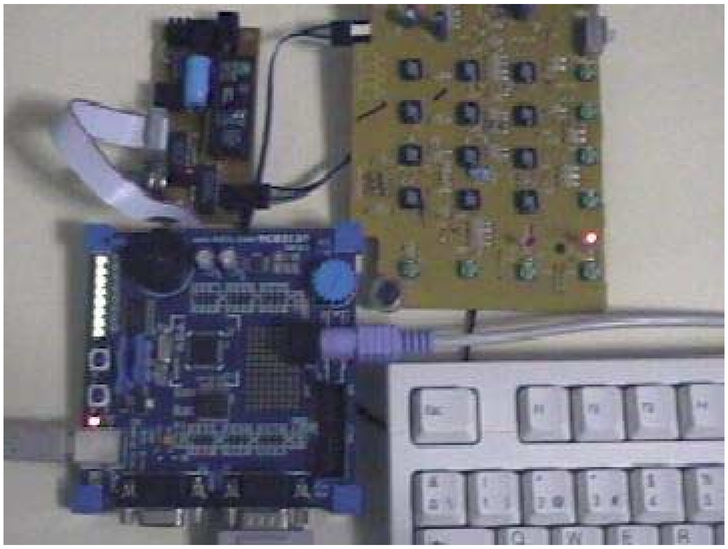

photo2: complete project