Audio Chopper Circuit and Configuration

This document remains in a "construction stage" !
If you need an analogue input with DC-coupling, use an external A/D converter for the PC's serial port instead !

Introduction

The 'chopper' is a simple piece of external hardware which basically reverses the polarity of an input signal before it is fed into the soundcard. Its main purpose is to process signals with DC components (for example battery voltages, signals from ELF receivers etc).

The problem with the input circuitry of most soundcards is, they use "AC" coupling (usually by means of one or more capacitors). To bypass this problem, the chopper described here turns DC signals into low frequency signals ***before feeding them into the soundcard***.

The analyzer program then puts the chopped pieces together again, using a ***synchronous rectifier*** which is completely software-based.

Thanks to Phil Beastall and Ken Tapping at the UKARANet for the idea.

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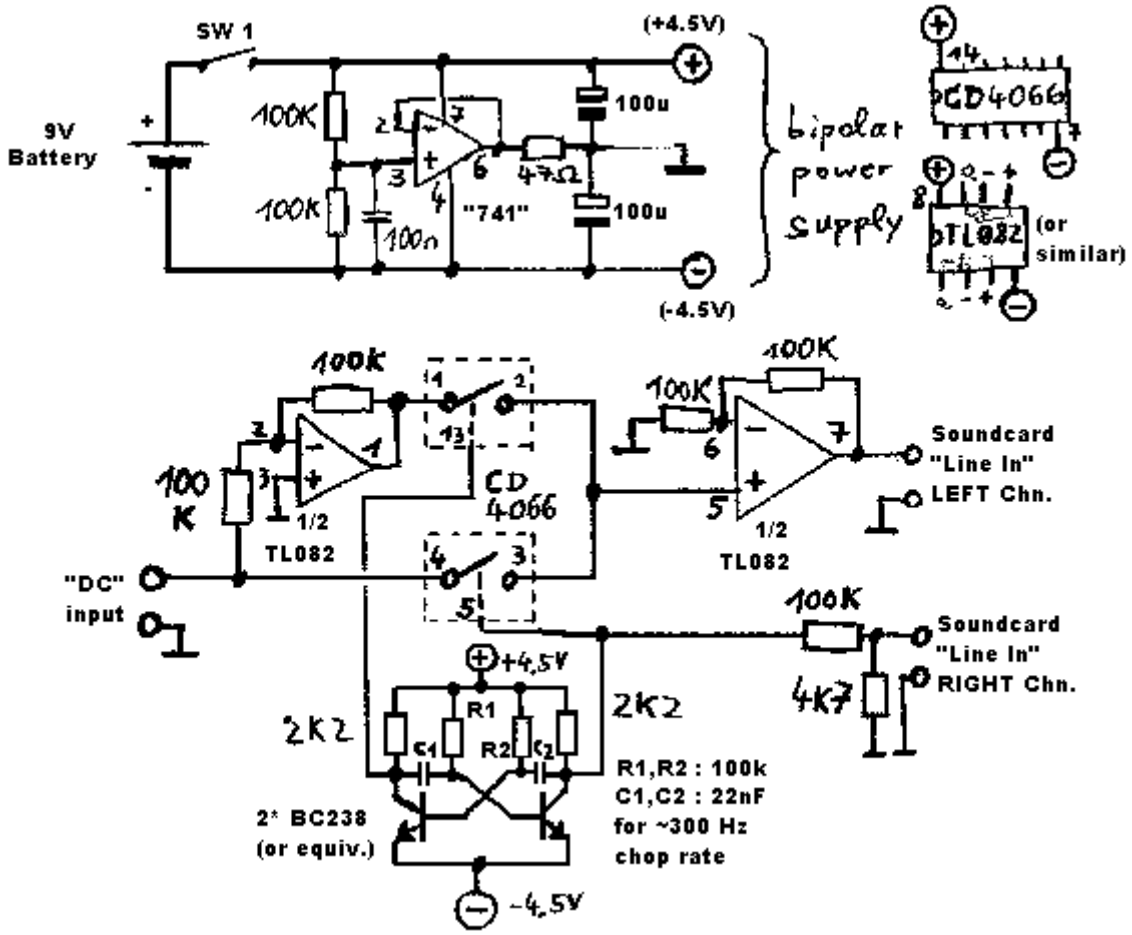
Schematic diagram

- under construction -

A crude sketch of my test circuit. Note: Here a simple astable multivibrator produces the copping clock, you may replace it with anything better (for example a crystal oscillator plus binary counter like the CD4060, which can also be used as an accurate clock source to calibrate the soundcard's sample rate).



A signal with a few hundred millivolts may be fed into the "DC input", either polarity. Note that the analog switch is not connected to ground, but to a positive and negative power supply. The same applies to the double OPAMP, like the TL082 used here. If you don't need battery-powered operation, leave the circuit around the "741" OPAMP away and use a simple dual power supply instead (+/- 5 V).

A small fraction of the square wave which drives the analog switches is also fed into the RIGHT input channel of the soundcard. This is important for the synchronous rectifier (which in fact is a subroutine in the software).

If you need a higher bandwidth, you may have to increase the chopping rate. The frequency of an astable multivibrator like the one shown above is:
fo = 1 / ((0.69*R1*C1) + (0.69*R2*C2)). Replace C1 and C2 (originally 22nF) with the desired value, for example 10nF if you want to process frequencies from 0 Hz up to 500 Hz.

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Settings for the audio chopper in Spectrum Lab

The settings for the audio chopper can be modified on the "Chopper" tab sheet in Spectrum Lab's configuration sreen. To get there, select "Options".."Audio Settings" from Spectrum Lab's main menu.

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On the "Chopper" tab are these controls (maybe already more):

Enable Chopper and synchronous rectifier
: This checkbox is used to turn the chopper and associated rectifier routines in Spectrum Lab on or off. If you don't need DC capability or don't have the required external hardware, turn it off.

Nominal chopping rate (Hz)
: Usually, you can leave the chopping rate on the default value which is about 100 Hz. If your soundcard's lower edge frequency is exceptionally bad (=high), use a higher value.
Note: If the option "produce chopping signal by software" is enabled, the true chopping frequency will be rounded to the nearest fraction of the ADC's sample rate for various reasons.

S/W produces chopping signal and outputs it to the DAC
: The chopping rate is produced via software, which guarantees that exactly the same values are used for the external chopping switch and the synchronous rectifier. To drive the chopping switch, a square wave is produced at the soundcard's output (occupying the RIGHT output channel of the soundcard if it runs in "stereo" mode... may be user-selectable in future).

dont produce but READ the chopping clock from the 2nd ADC channel
: For this mode, the soundcard must run in "stereo" mode which can be set in Spectrum Lab's "audio settings". The LEFT input is the chopped signal from a receiver (or similar), the RIGHT input is the chopping signal (hopefully a square wave) which drives the analogue switches in the chopper and also the synchronous rectifier which is implemented in software.

total delay of chopping signal between ADC and DAC
: This value is not measured in SECONDS, but in ADC CYCLES. You can use it to compensate the delay between the ADC input and the DAC output (which depends on the soundcard's hardware and the way how Windows treats the audio buffers), and also a little bit on the external chopper hardware.

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See also: Spectrum Lab's main index .

Last modified: 03/2002