Bode plot measurement using soundcards

Jesús Arias

7th November 2005

1 Introduction

This document is the description of a computer program intended for the measurement of the transfer function of audio circuits in the frequency domain. This is accomplished using the soundcard of the PC both for the generation of test signals and for the acquisition of data. Today soundcards typically exhibit sampling rates up to 48KHz and 16 bits of resolution. They are almost always based on sigma-delta analog-to-digital and digital-to-analog converters resulting in an impressive resolution and linearity, making them a potential high-performance laboratory tools.

The program described in this document is able to measure the Bode plot on the circuit under test and to plot it in the computer screen several times per second using nothing more than the audio soundcard integrated in the PC mainboard. The obtained results are worthy of an expensive audio network analyzer. This program runs on a PC under Linux using ALSA sound drivers with OSS API emulation. The sound card must support full-duplex operation.

Tested audio cards:

- SoundBlaster 16 Vibra (ISA PnP. 44.1KHz)
- Yamaha OPL3-SA (built-in, ISA PnP, 44.1KHz)
- Intel ICH5 chipset + Analog Devices codec (built-in, 48KHz)
- VIA 8237 + Realtek ALC650F (built-in, 48KHz)
- Sun Blade 2500 (built-in, Solaris, under development)

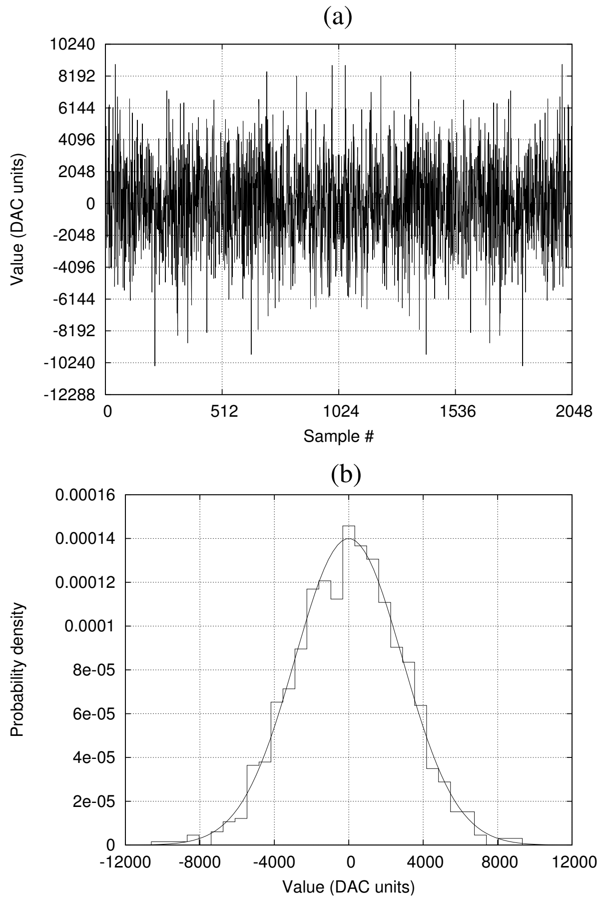

Figure 1: (a) Test signal. (b) Amplitude histogram and Gaussian fit (rms value: 2894).

2 Measurement fundamentals

Typically, Bode plots are measured by means of applying a sine-wave signal to the input of the circuit and measuring the output amplitude and phase. This has to be repeated for a lot of different frequencies resulting in an slow process. This long measurement time is mostly due to the sequential frequency scan and the big audio buffers used in soundcards. In the program "bodefft" a different approach was followed: we apply all the test frequencies simultaneously, and therefore, the measurement of the Bode plot is done "in parallel". Therefore, the time required for the measurements is orders of magnitude shorter.

A possible test signal containing all the frequencies with the same amplitude and phase is the impulse. This signal has one sample with the maximum possible value and all the other samples are 0. The impulse signal is typical in theoretical analysis, but is not very practical in experimental environments. The main problem with the impulse signal is the bad Peak-to-Average ratio (PAR): Even if one sample has a very high amplitude all the oth-

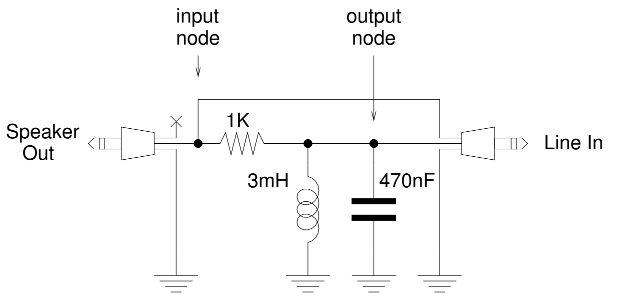

Figure 2: Experimental setup for the example circuit.

ers are zero and the resulting rms value of this signal is quite small. In a 2048 sample stream, the impulse can have a maximum value of 32768, resulting in an rms value of $32768/2048 = 16$ and a PAR of $32768/16 = 2048$ (66 dB!). With impulses it is not possible to apply high rms voltage values to the circuits under test without overloading the ADC, DAC or the circuit under test itself. A different signal must be used.

In our program the test signal includes all the frequencies with the same amplitude, but the phase of each tone is selected randomly between 0 or 180^o. Then, an inverse FFT is performed resulting in the waveform shown in Figure 1(a). This signal looks much like random noise, but a closer look reveals some symmetry. The amplitude distribution is shown in Figure 1(b). It is pretty much Gaussian with an rms value of 2894, a peak value of -10451and a PAR of 3.6 (11 dB).

The test signal is played continuously on the soundcard output. The signal at the circuit output is recorded and translated to the frequency domain using a FFT. The measured phases are rotated 180^o in the tones whose output phases were chosen as 180^o in the test signal. In this way we obtain the equivalent of an impulse signal test but with a much more convenient signal. The only remaining issues are related with calibration.

Almost every soundcard provide an stereo input. We can make a good use of the extra input channel for calibration connecting this input to the input of the circuit under test. In this way the voltage at both the input and output of the circuit are recorded together and the information form the input can be used to remove the effects of volume settings and the frequency response of the soundcard's output channel. The only requirement needed in order to use this calibration technique is a good matching between the L and R channels of the soundcard, and this requirement seems to be fulfilled in every card tested.

3 Example

In Figure 2 an example circuit is shown. It consist of an RLC circuit with a second-order band-pass transfer function. The output of the audio soundcard is connected to the input of the circuit under test. Only one of the two available channels is used. It is a good idea to use an amplified output instead of the Line Output. This is strongly recommended if the circuit under test exhibits a low input impedance. The voltage of the input node is connected to one of the two Line-In channels. The other channel is connected to the output of the circuit under test. In this way we can measure simultaneously the voltage present at both the input and output of the circuit.

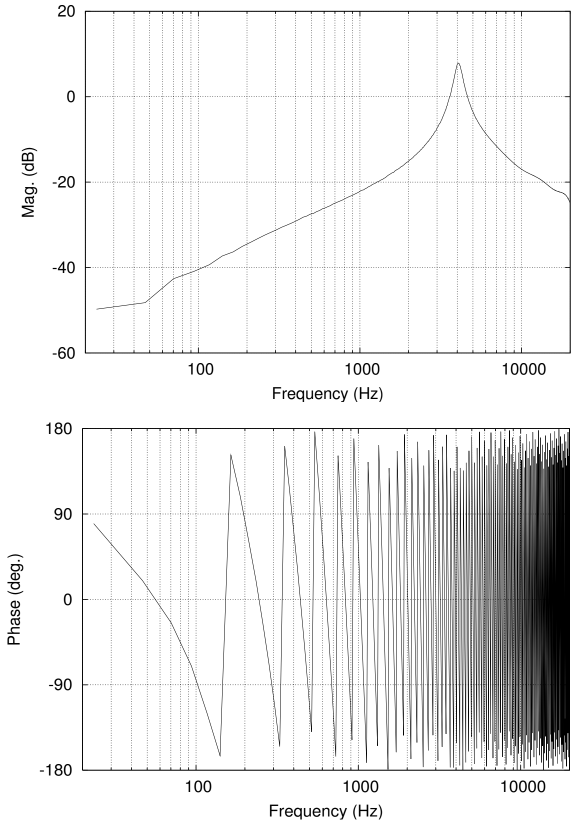

Figure 3: Measured Bode plot before delay adjust and channel calibration.

Before running the "bodefft" program a proper audio mixer setup have to be performed. A program like "aumix" is used for this purpose. We have to select the Line channel for recording instead of the microphone and to adjust the input gain and output volume level. Then we can start the "bodefft" program. A Bode plot is displayed on the X11 screen. We can exchange the roles of the two input channels using the "x" key. After performing these steps a plot like that presented in Figure 3 is obtained.

The Magnitude plot looks promising, but the phase shows a fast decay that, together to the circular nature on phase angles, looks like a sawtooth waveform. This is due to the delay between the input and output converters in the soundcard. This delay can be corrected interactively using the "d", "f" and "g" keys. After doing this the new

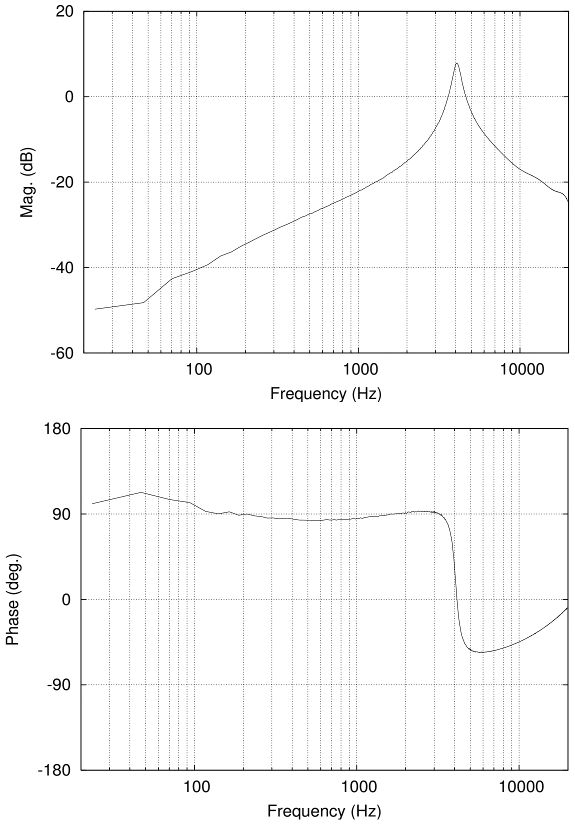

Figure 4: Measured Bode plot after delay adjust. No channel calibration.

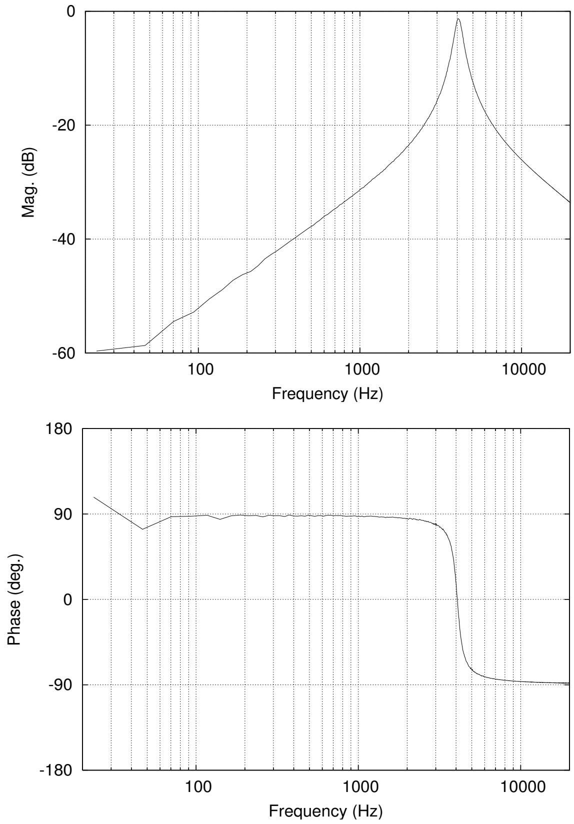

Figure 5: Measured Bode plot after channel calibration.

Bode plot looks like the one shown in Figure .

Now, the 180^o phase shift due to the LC resonance is clearly seen, but the plot is still not perfect. We can see a gain higher than 0dB while the circuit under test shows no amplification. We can also see some small ripples in the magnitude plot and a residual phase change at high frequencies. In order to correct these effects a calibration of the output channel have to be performed. In this calibration the voltage present at the input of the circuit is measured and then its magnitude and phase is used to correct that measured at the output. We can calibrate the output channel using the “c” key. After doing this, the final Bode plot of Figure 5 is obtained. Here no artifacts are shown. The measured dynamic range is about 60dB, even in the spite of some power line noise.

4 Measuring impedances

The Bode plots recorded using the program “bodefft” can be used to derive the value of complex impedances. Another program is included for this purpose (impedance.c). The circuit under test must include a series resistor of a known value (Figure 6). The impedance is derived as follows:

$$Z(f) = \left[\left(\frac{1}{H(f)} - 1\right)\frac{1}{R} - \frac{1}{Z_{IN}(f)}\right]^{-1}$$

Where $H(f)$ is the measured transfer function, R is the series resistor and $Z_{IN}(f)$ is the measured input impedance of the soundcard. $H(f)$, $Z_{IN}(f)$ and $Z(f)$ are complex values.

The value of R must be selected depending on the expected impedance. Best results are obtained when $R$is in the same order of magnitude than the measured

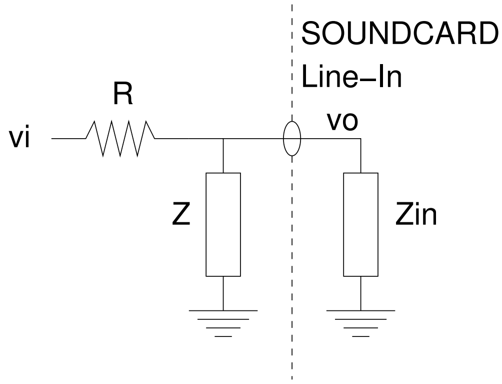

Figure 6: Setup for impedance Measurements.

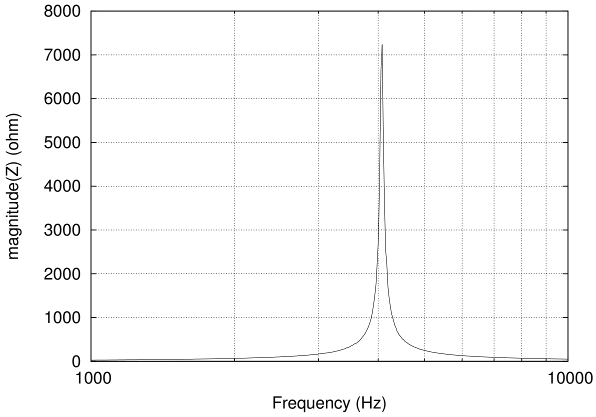

Figure 7: Measured impedance of the LC tank of the example circuit.

impedance. Z_{IN} can be measured by just removing the impedance Z prior to measurement and deriving its value. Z_{IN} is mainly resistive and about $33k\Omega$.

In Figure 7 the derived impedance of the LC tank of Figure 2 is shown. From this plot we can obtain a Q value of about 85 for the inductor at 4KHz.