Subsonic Filter for Phono preamps and Sub-Woofers

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Introduction

Frequencies below 20Hz are usually not able to be reproduced, and with the exception of synthesisers and pipe organs, are not a wanted part of the audio spectrum. This is especially troublesome with phono systems, since many of the vinyl discs you treasure (or wish to transcribe to CD) will be warped to some degree. Any warp in a vinyl disc will cause large outputs in the subsonic region, typically well below 20Hz.

For example, a 33 1/3 RPM album with a single warped section will create a signal in the pickup at 0.55 Hz (33.3 RPM / 60 = 0.555 Hz). This is a signal that will cause significant cone movement, but is undesirable in the extreme. Not only will vented subs be completely unable to handle such a signal linearly, but sealed subs will also be stressed. Large amounts of available power will be wasted trying to reproduce a signal that was never intended to be there in the first place.

To be effective, a subsonic filter has to be very steep - this allows all wanted frequencies to get through, and rejects those that will only cause problems.

At least one circuit that the author knows of uses a method of summing the channels below 140Hz, and although this is effective in removing the low frequency rumble (or sub-rumble in this case) component, it causes frequency response abberations that (IMO) are not acceptable. The subsonic frequencies generated by record warp are by nature out of phase. The mono component of a vinyl disc is lateral, whereas warp signals are vertical. Stereo signals are at 45° The summing method was examined in great detail before deciding that it should not be used if the overall frequency response of the disc is to be preserved.

Please note that PCBs will be available for this project, and they are expected to be ready by the end of April 2003.

Description

The circuit shown is completely conventional. The Q of the filters has been optimised to allow a higher input impedance than would otherwise be possible, with the final Q of the two filters being almost exactly 0.707 (i.e. a traditional Butterworth filter). Although in theory the tolerance of both resistors and capacitors *should* be 1% or better, in reality it is not that important. 1% metal film resistors are recommended (as always) but only for lowest noise, and capacitors are standard (i.e. 5% or 10%) tolerance. Yes, this *will* cause the response to deviate from that shown below (see Figure 2), but compared to other errors in the system (recording EQ, room LF node problems, etc.) these may be considered minor.

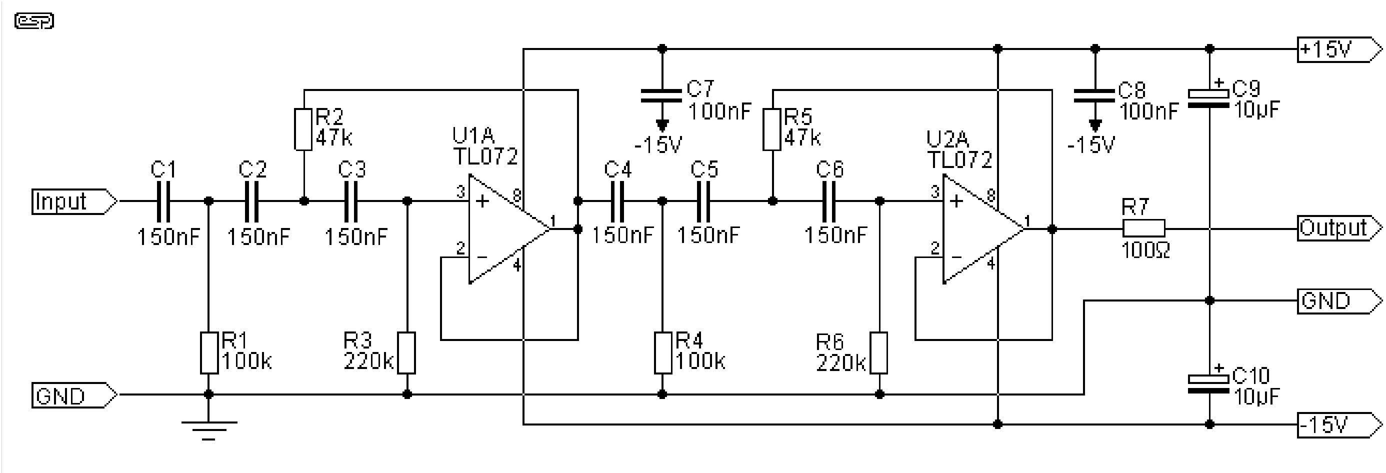


Figure 1 - Circuit Diagram of 1 Channel

The circuit of the filter is shown above. It is essentially a pair of cascaded 18dB/octave filters, giving an ultimate rolloff of 36dB/octave. The -3dB frequency is about 17Hz with the values shown. I do not suggest that you experiment with resistor or capacitor values unless you know *exactly* what you are doing, since any changes will affect the Q of the filters, and will cause either a "lump" in the passband response, or will roll off too gradually, resulting in a loss of bass.

Figure 2 shows the theoretical response of the filter. I say "theoretical", simply because it is unrealistic to expect any signal to be over 200dB down from the passband level. This is simply beyond the noise limits of any known device. Having said that, the attenuation of ultra-low frequencies is still very high indeed, and even a badly warped disc will cause very little (if any) subwoofer cone movement.

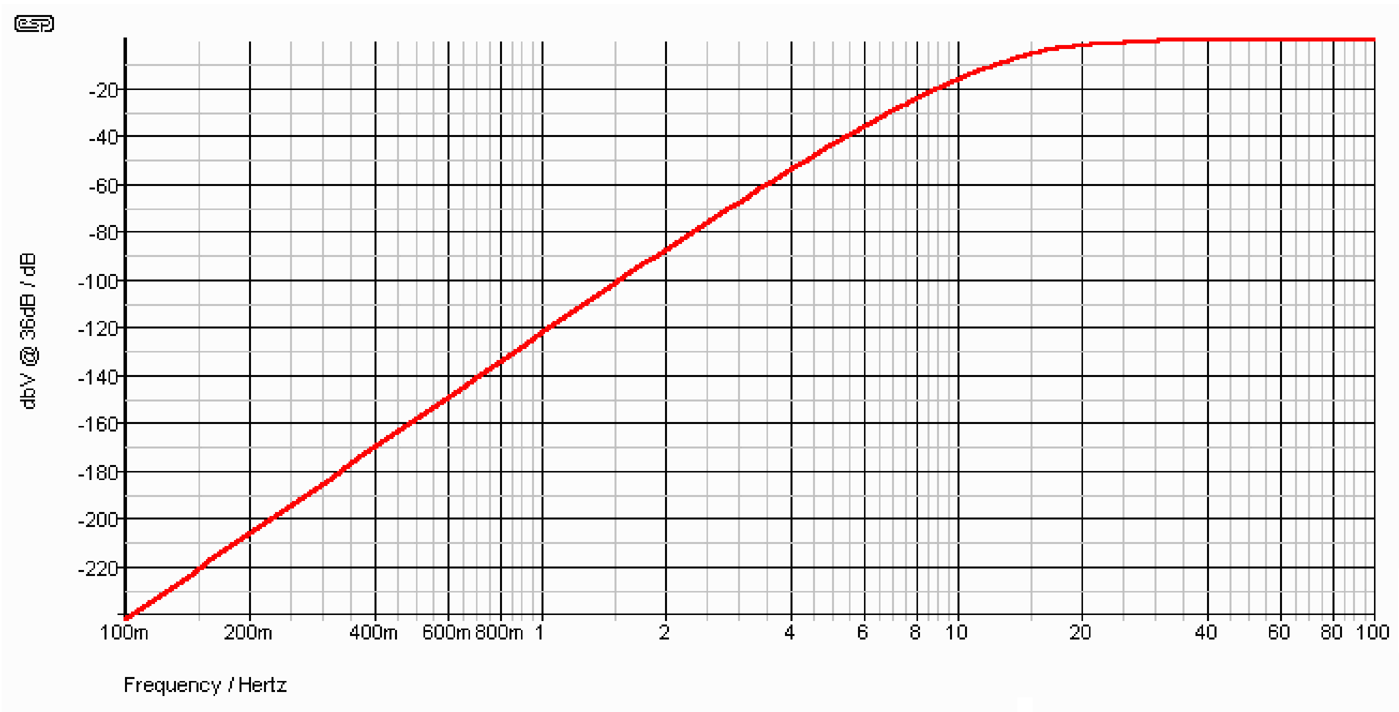


Figure 2 - Filter Frequency Response

As can be seen from the above, below 1Hz the overall response is better than 120dB below the passband level - nominally anything above 17Hz. There is no reason to try to better this, as it already exceeds the resolution of any digital format, and places all typical warp signals well below audibility or danger level for a sub-woofer.

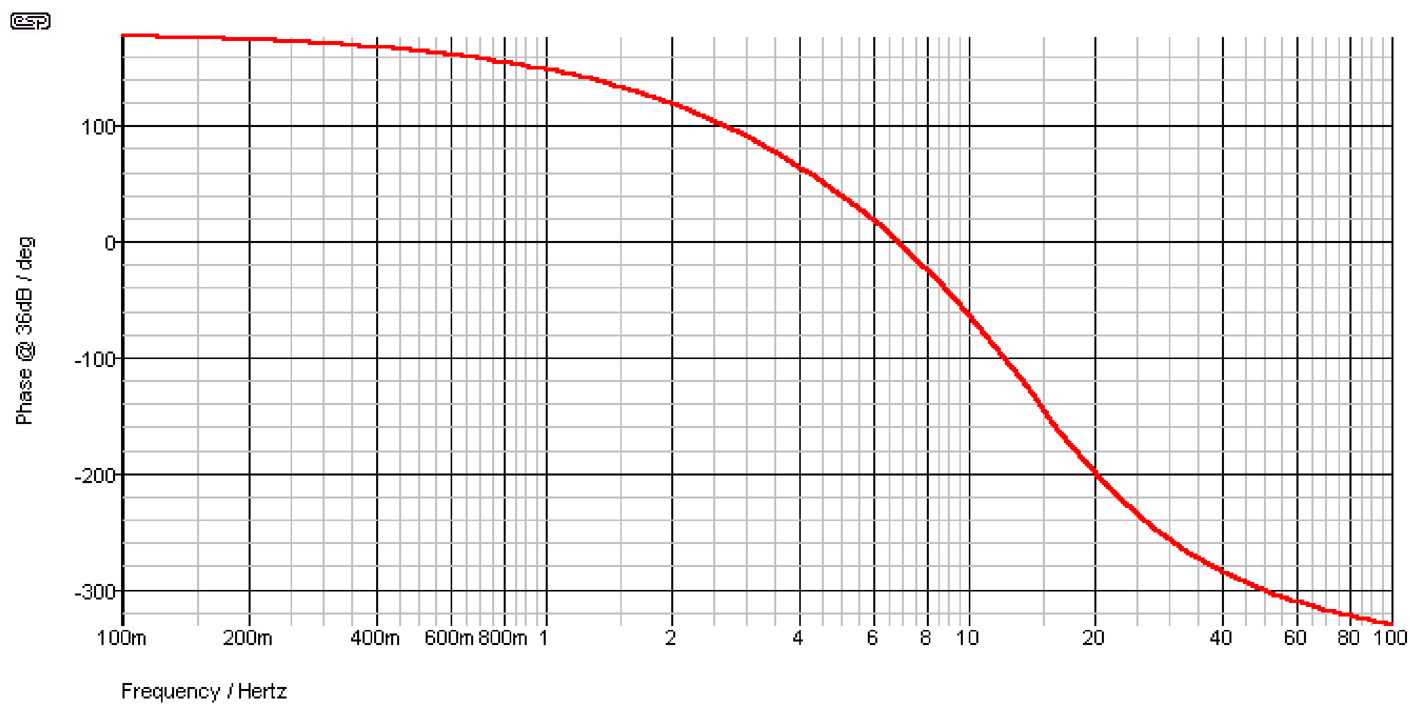


Figure 3 - Filter Phase Response

The phase response is as one would expect for any filter, but it is important to note that unless the full-range signal is filtered, there will be unacceptable phase variations in the low frequency regions. This filter should not be used in series with the sub-woofer amp, as the phase relationship between the main speakers and sub-woofer will be badly affected.

Because of this, it is recommended that high quality opamps be used to prevent noise or distortion in the main signal. If desired, a switch may be used to bypass the circuit when not in use. The use of a subsonic filter is not reserved for vinyl discs - many CD recordings also contain subsonic energy as well, either deliberately or by accident!

To change frequency, change *only* the capacitors. The following table gives a range of values and frequencies that should suit any application. These are for C1, C2, C3, C4, C5 and C6 and all must be the same value ...

Capacitance	**-3dB Frequency**	**Capacitance**	**-3dB Frequency**
220nF	12.4 Hz	56nF	48.5 Hz
180nF	15.1 Hz	47nF	57.8 Hz
150nF	18.1 Hz	39nF	69.8 Hz
120nF	22.7 Hz	33nF	82.3 Hz
100nF	27.2 Hz	27nF	100 Hz
82nF	33.2 Hz	22nF	123 Hz
68nF	40.0 Hz	18nF	151 Hz

Table 1 - Capacitance vs. Frequency

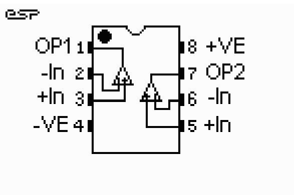
The range shown above obviously caters for frequencies well outside normal subwoofer range, but they are included as there may be other uses for the filter other than only for subwoofers. There are countless applications for very steep filters in control systems and other analogue applications, so there is no reason to restrict use to audio only.

Construction

Although construction is not critical, the usual precautions needed with any opamp circuit should be followed. Pay particular attention to bypassing, and do not omit the power supply ground connection. Naturally, I recommend that you use the PCB, as it makes a somewhat tedious wiring exercise very simple. You may (as always) use better opamps than the TL072 dual versions suggested, and the most important parameter is noise. Since the opamps are wired as unity gain buffers, upper frequency response will be well extended to beyond audibility.

Only a single channel is shown in Figure 1, the second channel uses the remaining opamp in each of the dual packages. It is imperative that this circuit is driven from a low impedance. The actual input impedance is greater than 47k at all frequencies, but the source impedance should be no more than 100 ohms or so.

Typically, the filter would be used at the output of your phono preamp. Subsonic frequencies are uncommon from other signal sources (but can and do exist!), but if you wish to use the circuit shown in series with your sub-woofer, then you must be aware of the possible effects of the phase response of the filter (see above for details).



The standard pinout for a dual opamp is shown on the left. If the opamps are installed backwards, they will almost certainly fail, so be careful.

The suggested TL072 opamps will be quite satisfactory for most work, but if you prefer to use ultra low noise or wide bandwidth devices, that choice is yours.

Testing

Connect to a suitable power supply - remember that the supply earth (ground) must be connected! When powering up for the first time, use 100 ohm to 560 ohm "safety" resisors in series with each supply to limit the current if you have made a mistake in the wiring.

The opamp DC output voltages should be nearly zero. Testing the frequency response will not be possible unless you have a signal generator (PC based ones are fine), and an AC millivoltmeter. Response above 20Hz should be essentially flat (there will be a very small peak at around 30Hz - less than 0.2dB), and at 10 Hz, the response should be at least -15dB. If you can measure down to 5Hz (or less), then the response should follow the graph in Figure 2 very closely.