Elliott Sound Products

Project 99

Infrasonic Filter for Phono preamps, Sub-Woofers, PA Systems, Etc

Updated May 2021

Please Note: PCBs are available for this project. Click the image for details.

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 infrasonic region, typically well below 20Hz. While often referred to as 'subsonic', the correct terminology is 'infrasonic' ('subsonic' means slower than the speed of sound, 343m/s).

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, an infrasonic filter has to be very steep - this allows all wanted frequencies to get through, and rejects those that will only cause problems. Even 24dB/ octave is likely to be marginal, especially when driving a transformer load.

A steep infrasonic rolloff is essential is driving a distribution transformer - typically for 70V or 100V public address systems as used in offices, shopping centres, factories, etc. Any very low frequency signal that gets through the amplifier and saturates the transformer is likely to cause either amplifier failure, *gross* distortion, or commonly both. See High Voltage Audio Systems for the details. An infrasonic filter is absolutely essential in these systems, but even some amplifier manufacturers don't seem to appreciate the risks. With 70V and 100V public address systems, there is usually no reason to reproduce anything below 80Hz, even for background 'music'.

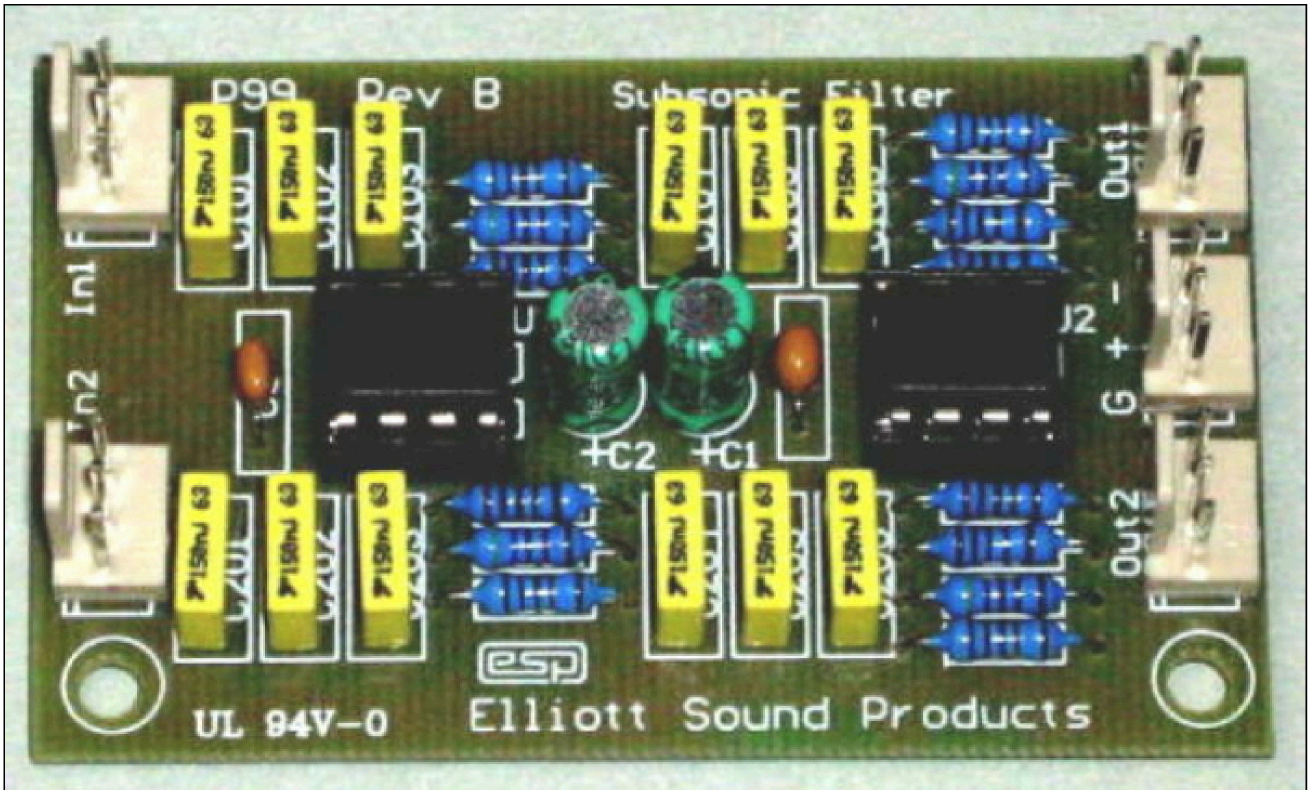


Photo of Completed P99 Revision-B Board

At least one published rumble filter circuit 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 aberrations that are unacceptable. The infrasonic 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 carefully before deciding that it should not be used if the overall frequency response of the disc is to be preserved. Summing also cannot be used with a mono signal, and that would limit the usefulness of the filter.

The project as presented here can be used anywhere that you need a rapid rolloff to prevent infrasonic signals from causing havoc. As noted above, it's essential for 70V and 100V line PA systems, or anywhere that a transformer is driven from a power amplifier. It's also very useful with vented speaker enclosures, and prevents excessive cone excursion at frequencies below the box resonance. It can also be used with instrumentation/measurement systems where low frequency energy 'pollutes' the results. PCBs are available for this project, which makes it very easy to put together.

Description

The circuit shown is a conventional Sallen-Key filter, but some simplifications have been made so that the number of different value components is minimised. 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.

Although it is stated below that the input impedance of this filter should be less than 100 ohms, it may be directly connected to the Project 06 phono preamp. Testing shows that the overall frequency response is changed by less than 0.1dB at any frequency above 30Hz. Naturally, low frequency response is affected by the filter as it should be. Even with an input impedance as high as 10k, there is no significant deviation from the expected response curve, and only a tiny (about 0.2dB) loss of overall level.

The circuit of the filter is shown below. It is essentially a pair of cascaded 18dB/octave filters, giving an ultimate rolloff of 36dB/octave. The -3dB frequency is about 18Hz with the values shown. See the table below for different capacitor values you can use to obtain different rolloff frequencies.

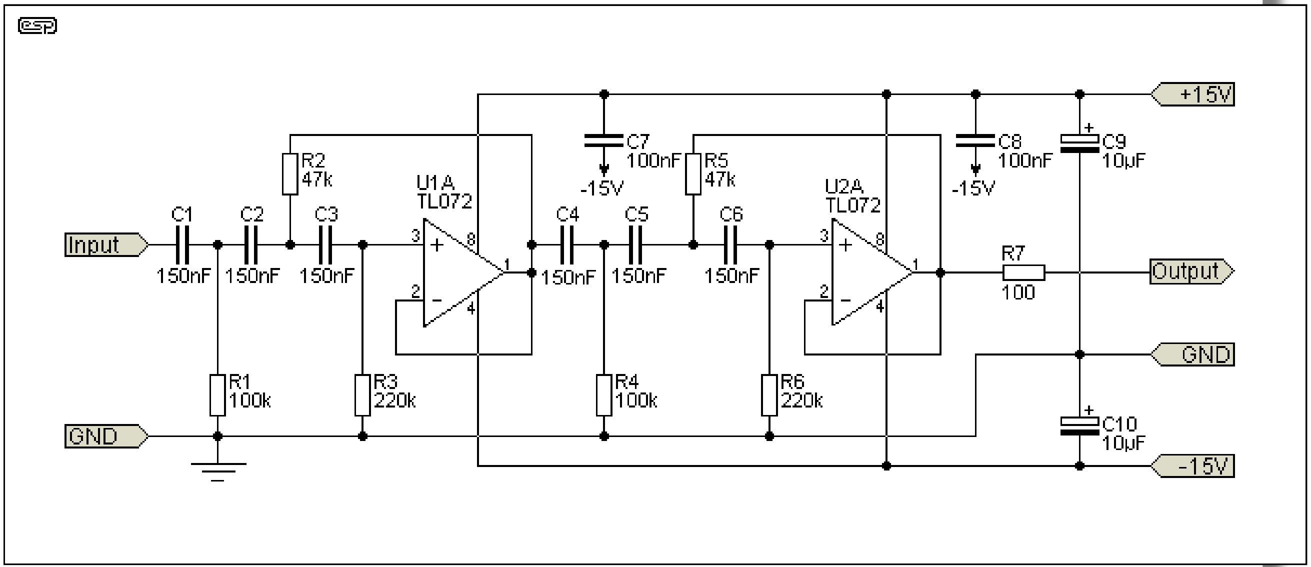


Figure 1 - Circuit Diagram Of One Channel

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 well over 100dB down from the passband level (in excess of -120dB at 1Hz). This is simply beyond the noise limits of audio equipment. 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.

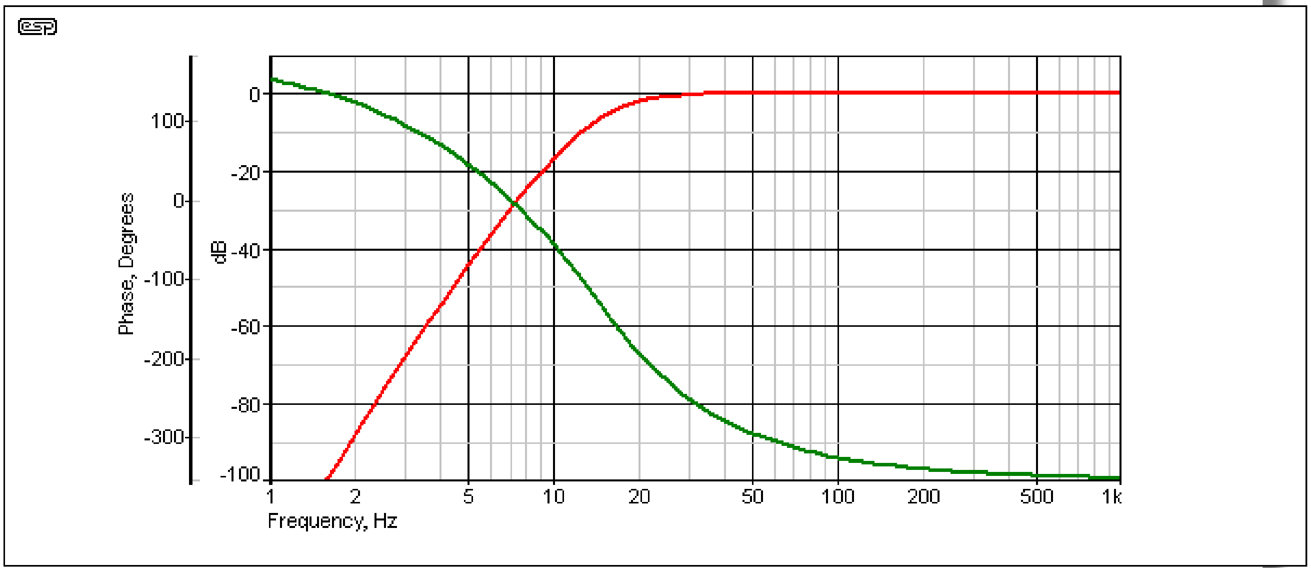


Figure 2 - Filter's Frequency (Red) and Phase (Green) Response

As can be seen from the above, below 2Hz the overall response is better than 90dB below the passband level - nominally anything below 17Hz effectively disappears. 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.

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 may be unacceptable phase variations in the low frequency regions. Ideally, 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 affected. However, it is probable that there will be no audible anomalies even if the P99 *is* installed in the subwoofer signal path.

If the full range signal is going to be passed through the filter, 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 an infrasonic filter is not reserved for vinyl discs - many CD recordings also contain infrasonic 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 Freq.**		**Capacitance**	**-3dB Freq.**
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.

If you want the response to be a little steeper than normal, R3 and R6 can be increased. At 270k, there is a tiny increase before rolloff (0.2dB at 35Hz), and the low frequency limit (-3dB) is reduced slightly, to 15.6Hz instead of 18.1Hz (with 150nF caps). This has the same effect with other cap values, and you can use the above table and simply reduce the -3dB frequency by a factor of 0.86. For example, using 120nF caps for C1...C6, the frequency will be reduced from 22.7Hz to 19.5Hz. The difference is easily measured with a simulator, but will not be audible.

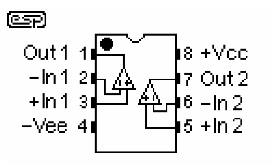
If you want to experiment with the resistor values feel free, but unless you can simulate the response you'll find that it's quite difficult to measure. The results can also be very unpredictable unless you are aware of all the interactions of component values with Sallen-Key filters. For example, if R2 and R5 are reduced to 22k, the -3dB frequency (relative to 100Hz) is barely affected, but there's a 3dB boost centred on 28Hz (using 150nF caps). This can be used to augment the extreme low bass response if desired.

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 ideally be no more than 100 ohms or so (although as noted above, even as high as 10k will cause few problems).

Typically, the filter would be used at the output of your phono preamp. Infrasonic 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. Suitable opamps include NE5532, OPA2134, LM4562, LME45710, NJM2068 etc. You may also be able to use the LM833, but they can be prone to oscillation, especially if you use an IC socket.

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' resistors 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.

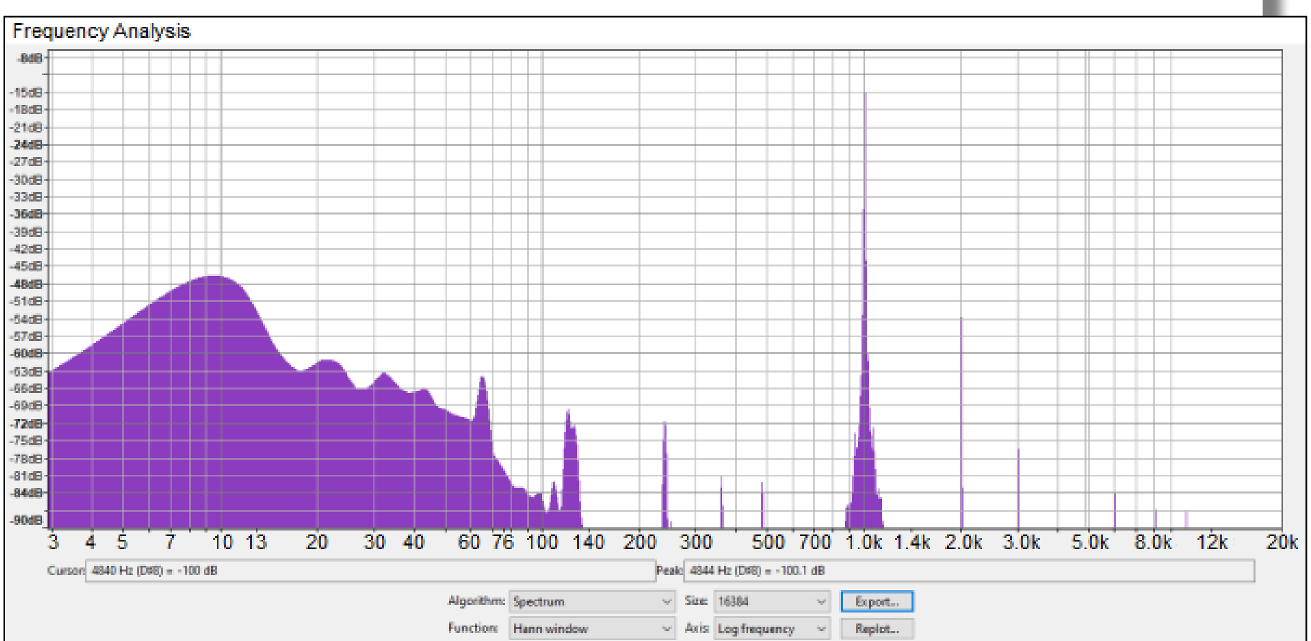


Figure 3 - Response From LP Without Filter [1]

To give you an idea just how much low frequency noise you may get from a turntable, the above was captured from a test disc with a newly refurbished unit, being belt drive with a DC motor, and with an Ortofon Blue moving magnet cartridge. The LF noise ('rumble') is clearly evident, peaking at 10Hz and around 32dB below the signal. The 1kHz test tone and its harmonics are visible, with the second harmonic being the most prominent. The phono preamp used was the ESP Project 06. It should be fairly obvious that using the high-pass filter described here will be beneficial.

[1] The above capture was kindly supplied by Bob Davis.

Main Index

Projects Index

Change Log: Page Created and Copyright © Rod Elliott, 04 Mar 2003./ Updated 12 Apr 05 - added note on suitability with P06./ 10 Mar 03 - added capacitance table./ 22 Apr 2003 - added phase graph and PCB availability./ 10 May 08 - added photo of board./ 08 Aug 08 - replaced response and phase graphs./ 12 Jan 09 - Revision-B board released./ May 2021 - added Figure 3 plus text.