Audio applications for Op Amps, Part II

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This is the second in a series of articles on single-supply audio circuits. The reader is encouraged to review the introductory material in the first article which concentrated on low-pass and high-pass filters. This article will concentrate on audio notch filter applications and on curve-fitting filters.

Audio Notch Filters

There are applications where a single audio frequency was not part of the original audio, is undesirable and annoying to the listener, and needs to be rejected. Frequencies on either side of the rejected frequency, however, contain useful audio content and if they are rejected as well there will be an equally annoying "hole" in the audio. Notch filters are used to reject very narrow frequency bands with minimal attenuation on either side of the notch, but compared to low-pass and high-pass filters they are hard to implement. Components that would only cause a slight ripple or *washout* in a low-pass or high-pass filter often have a dramatic effect on the depth of the notch; a slight mistuning of a low-pass or high-pass filter is inaudible, but mistuning a notch filter may cause it to miss the interfering frequency altogether. This section will give some hints for implementing a notch with a reasonable degree of confidence.

60-Hz Hum Filter

One of the most common problems with audio is the presence of 60 Hz. 60 Hz is one of the most prevalent interference sources in much of the world due to the fact it (with 50 Hz) are the frequencies used for ac power distribution.

Usually, 60-Hz hum is the result of poor grounding practices and it is far better to attack the 60 Hz problem at the source, rather than filter the audio. Nevertheless, there may be situations where the grounding of a particular system may not be accessible and in that case an add-on filter may be appropriate.

The 60-Hz hum filter shown (Fig. 1) is based on a Twin-T configuration. This topology is very effective, but it can be temperamental. The circuit response is very dependent on the absolute values of R1, R2, R3, C1, C2, and C3. All these tuning components should be 1% parts, but that may not be tight enough: They should all be taken from the same manufacturing lot when parts tend to have the same characteristics.

If the parts are matched properly performance can be very good, but mismatched components will seriously degrade the response. In the Twin-T configuration, R3 is half of R1 and R2 and the best way of making the resistance value for R3 is to use two of the resistors used for R1 and R2 in parallel. Similarly, taking two capacitors of the same value as C1 and C2 in parallel forms C3. This increases the component count of the circuit by two (one additional resistor and capacitor), but greatly benefits the matching -- because the designer can take easy steps to ensure that the parts are from the same batch (off the same reel -- out of the same box, etc.).

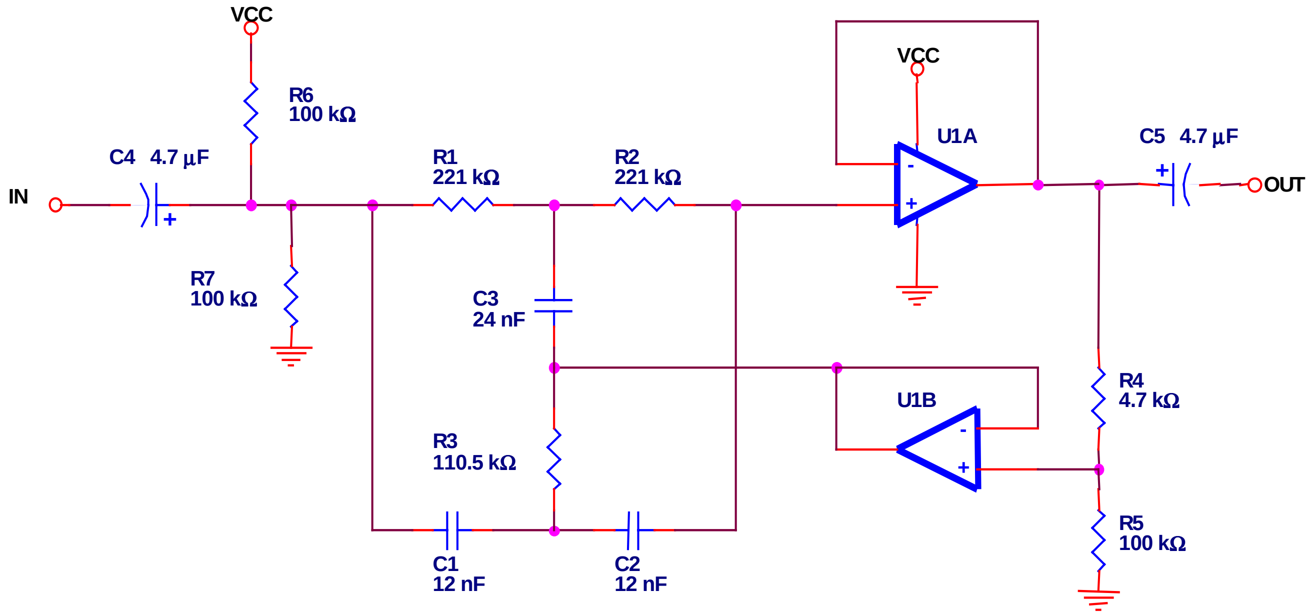

Fig. 1: 60-Hz Notch Filter

Note that no half-supply reference (see Part I) is required as R6 and R7 generate a half supply after the input capacitor.

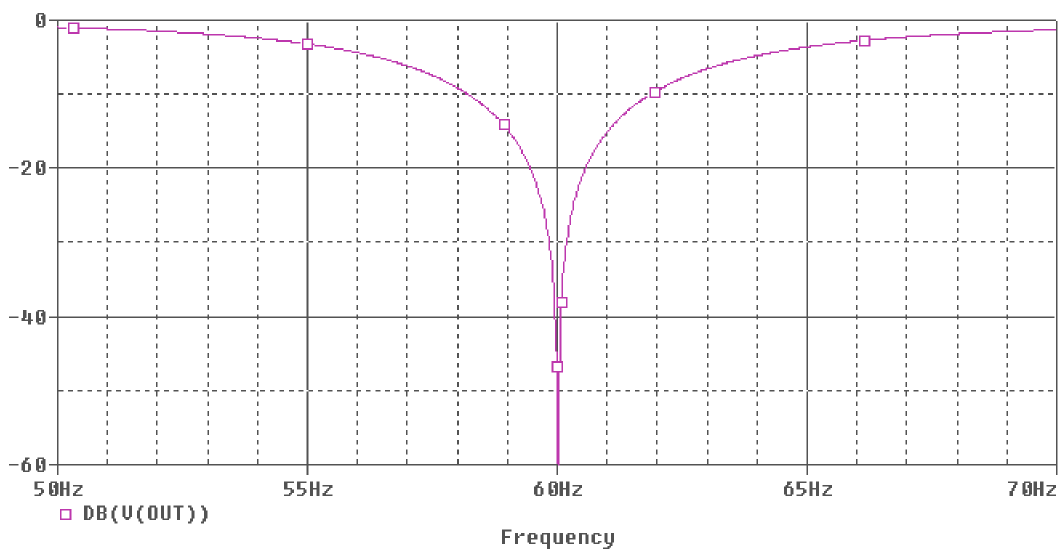

Fig. 2: Theoretical Response Of 60-Hz Notch Filter

The ideal values produce a theoretical notch as shown (Fig. 2.) If, however, the RC combination values are 1% off, the notch above will be shifted left or right 0.6 Hz (1%), and the 60-Hz rejection could be less than 20 dB.

A way around the problem is to put a small potentiometer (pot) in series with R3 (which should then be reduced one or two standard E-96 values.)

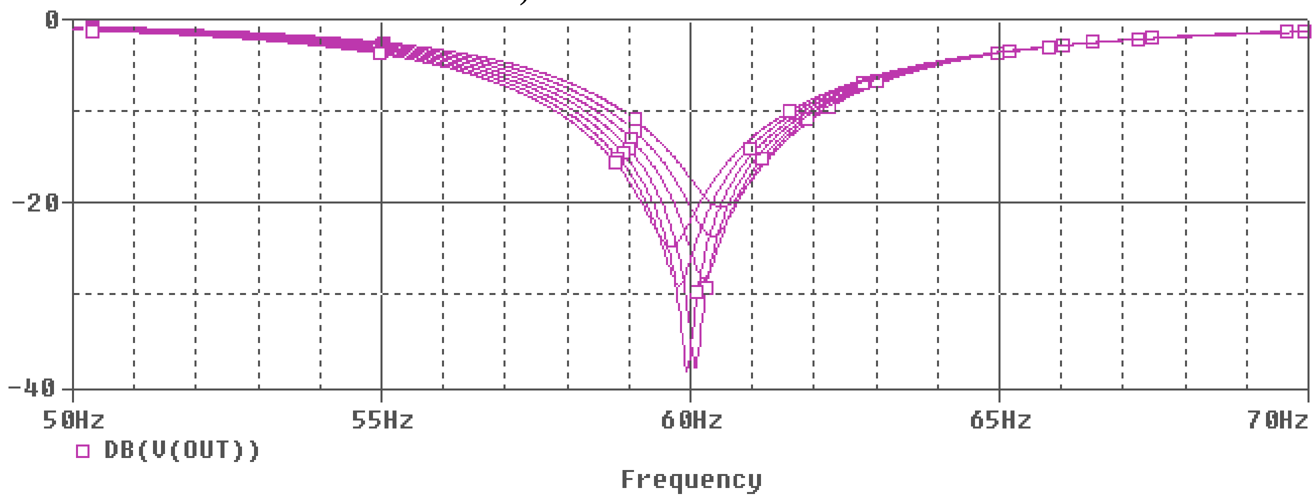

Fig. 3: Varying The Value Of R3 To Center The Notch

The graph (Fig. 3) shows the circuit response with R3 varied from 107 kΩ to 113 kΩ. Varying R3 even over this small range produces a tremendous variation in the depth of the notch as well as the Q of the circuit. In every case, though, more than 20 dB is achievable, with as much as 30 dB or 40 dB most of the time. If a deeper null is needed then R1 and R2 need to be adjustable as well, which can be achieved using a dual pot, to reduce the number of pots in the circuit to 2. If the capacitors are well matched (which is possible if they are from a single batch), 60 dB rejection is realizable.

50-Hz Hum Filter

The circuit (Fig. 1, again) can be modified to reject 50 Hz hum from the line frequency of ac power in Europe and other parts of the world:

- Change R1 and R2 to 42.4 kΩ.
- Change R3 to 21.2 kΩ (two 42.4 kΩ resistors in parallel).
- Change C1 and C2 to 75 nF
- Change C3 to 150 nF (two 75 nF capacitors in parallel).

The comments above regarding the 60 Hz notch filter apply to the 50 Hz version as well.

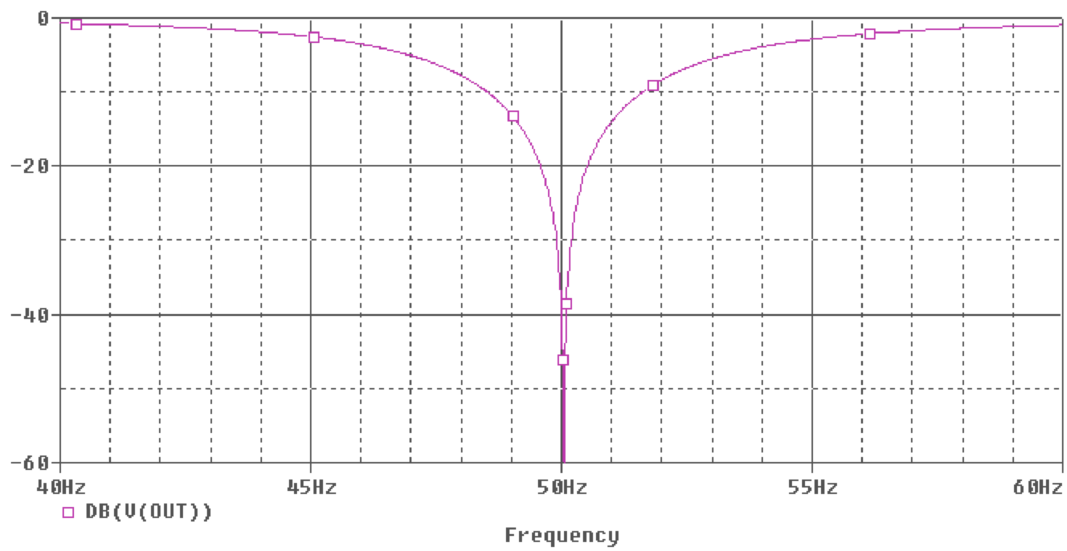

Fig. 4: Theoretical Response Of 50-Hz Hum Filter

The theoretical response of the filter is shown in Fig. 4. Standard E-24 capacitor values and standard E-96 resistor values do not produce a combination that is as close to perfect as in the 60-Hz case, so trimming may be even more necessary for a 50-Hz notch filter.

Medium-Wave Whistle Filter

For 10-kHz Channel Separation:

In North America, medium-wave (AM) stations are separated on 10-kHz channels from 540 to 1700 kHz. There is no regulation saying that a station cannot bleed over onto an adjacent channel or even a second adjacent channel -- which is nice for music. It means that AM frequency response is unlimited in comparison to FM, which severely rolls off above 15 kHz. Unfortunately, there will be interference from the adjacent channels on medium-wave, especially at night.

The audio modulation from adjacent channels is usually not a problem on strong local stations, but the carrier is. It shows up in the audio as a 10-kHz tone. This tone can be quite loud -- again especially at night -- even on local stations. For those who can hear it the pitch is extremely annoying. Making the problem even worse -- there are channels above and below most stations, and they are probably not exactly at 10 kHz: The FCC allows a tolerance of ±20 Hz from the assigned frequency, and that will make the two adjacent channels modulate each other and create beat frequencies, adding to the annoying aspect of the tone.

To eliminate the 10-kHz tone, a notch filter is needed that eliminates a narrow band around 10 kHz, while leaving other frequencies untouched. High-priced AM receivers (many years ago) used a high-Q LC filter, but tuning was so critical that it was of limited use. The op amp approach is extremely stable, and never requires additional adjustments once the initial center frequency is set. The Twin-T notch filter topology is used again, due to its ability to provide large attenuation with only two op amps. All of the comments about the 60-Hz notch filter apply to the 10-kHz notch filter. Even with simple tuning, however, the improvement should be dramatic.

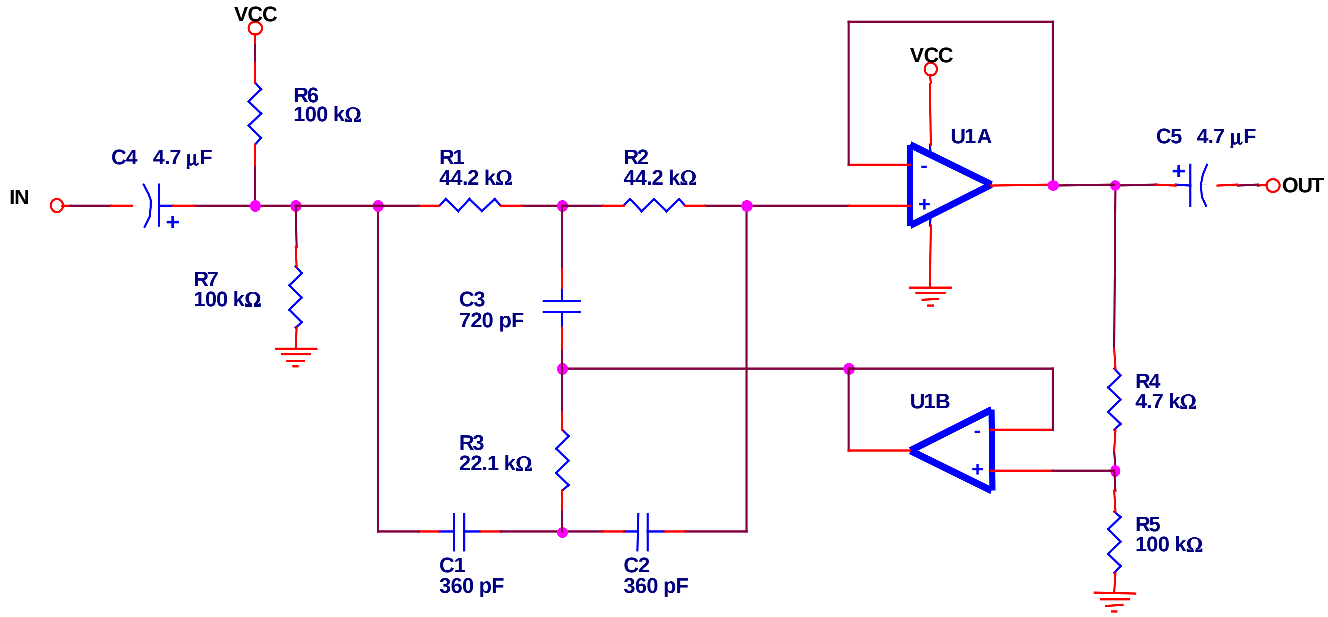

Fig. 5: Twin-T Notch Filter For 10 kHz

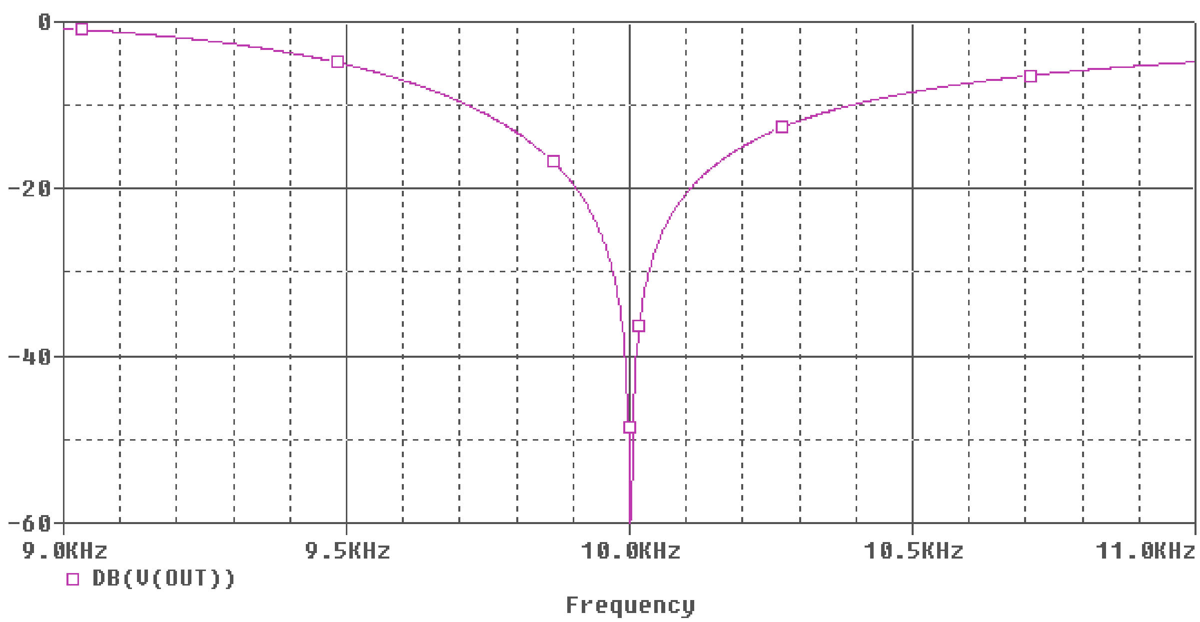

Fig. 6: Theoretical Response of 10-kHz Notch Filter

For 9-kHz Channel Separation:
In Europe and much of the rest of the world, the medium-wave uses channels separated by 9 kHz from 531 to 1611 kHz. This causes a 9-kHz tone in the received audio, instead of 10 kHz.

To reject the 9-kHz tone resulting from 9-kHz channel spacing:

- Change R1 and R2 to 45.3 kΩ
- Change R3 to two 45.3 kΩ resistors in parallel (22.65 kΩ)
- Change C1 and C2 to 390 pF
- Change C3 to two 390 pF capacitors in parallel (780 pF)

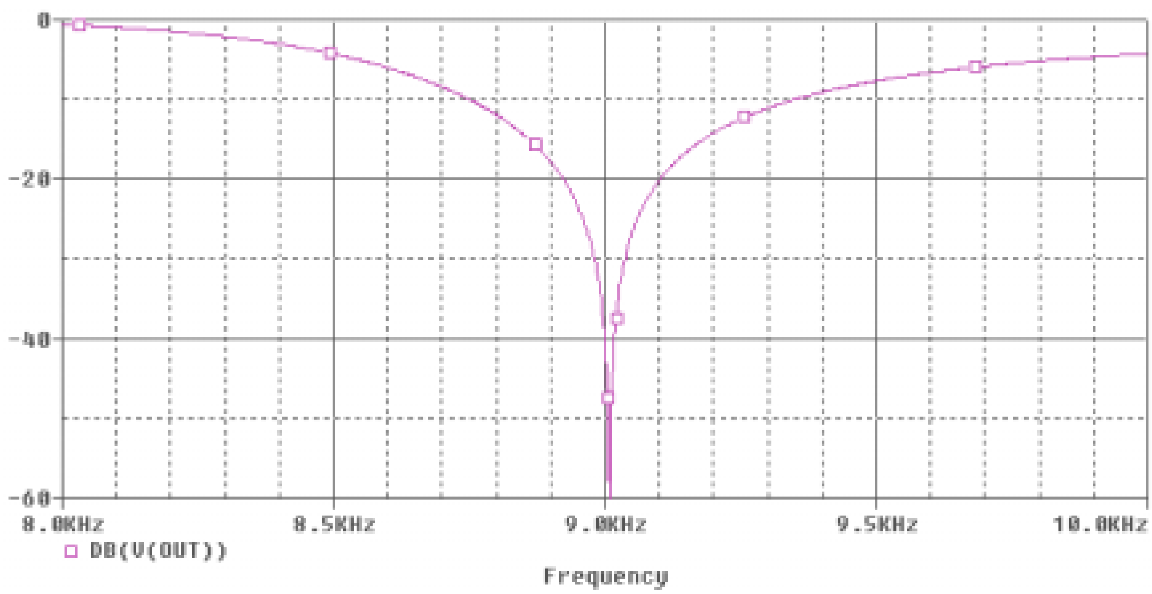

Fig. 7: Theoretical Response Of 9-kHz Notch Filter

For 5-kHz Channel Separation:
Much of the world relies on shortwave radio for news and entertainment. Shortwave radio is transmitted on several bands, with stations separated by only 5 kHz. To reject the 5-kHz tone resulting from 5-kHz channel spacing:

- Change R1 and R2 to 42.4 kΩ
- Change R3 to two 42.4 kΩ resistors in parallel (21.2 kΩ)
- Change C1 and C2 to 750 pF
- Change C3 to two 750 pF capacitors in parallel (1500 pF)

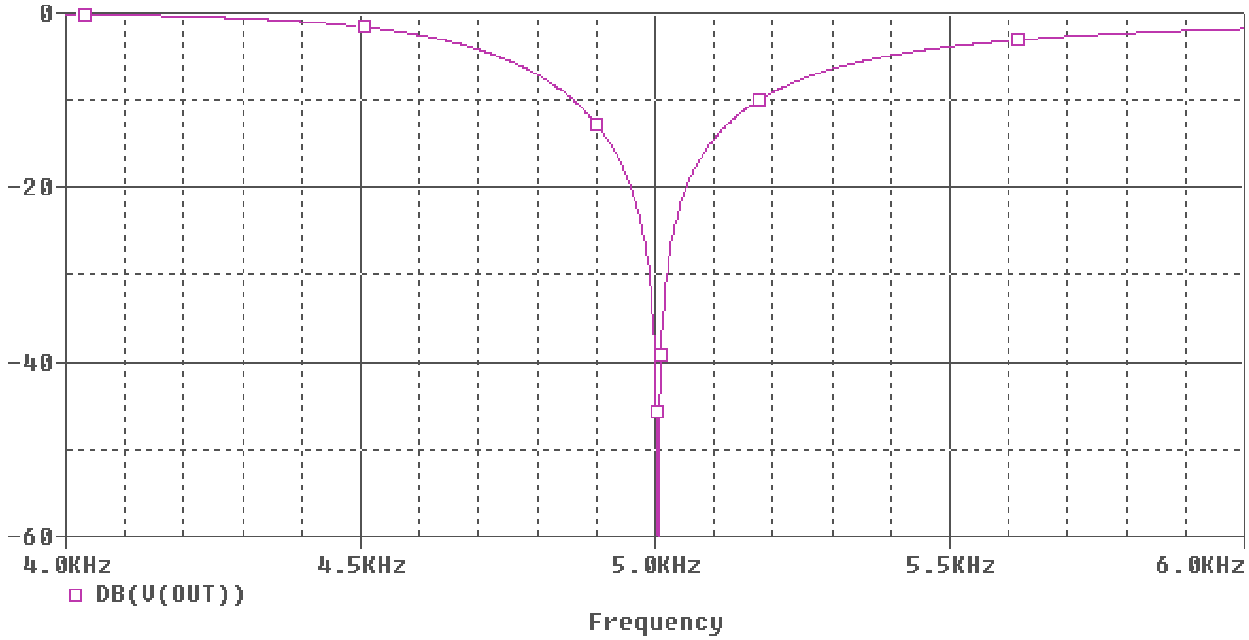

Fig. 8: Theoretical Response Of 5-kHz Notch Filter

This notch filter topology can be retuned to reject almost any audio frequency that poses a problem. Areas of the world served by both 10 kHz-spaced and 9 kHz-spaced medium wave may experience objectionable tones at any frequency from 1 kHz to 10 kHz and above.

Curve-Fitting Filters

Analog designers are often asked to design low-pass and high-pass filter stages for maximum rejection of frequencies that are out-of-band. This is not always the case, however: Sometimes the designer is asked for a circuit that will conform to a specified frequency response curve. This can be a challenging task, particularly if all the designer knows is that a single-pole filter rolls off 20 dB per decade, double-pole 40 dB per decade. How does the designer implement a different roll-off?

To begin with…it is not possible to get more out of a filter than it is designed to produce. A single-pole will give no more than 20 dB per decade -- and cannot be increased or decreased. More roll-off demands a double pole filter with 40 dB per decade. Obtaining different roll-off characteristics is done by allowing filters at closely-spaced frequencies to overlap.

One popular curve-fitting application is the RIAA equalization, which compensates for the pre-equalization applied to the audio on vinyl record albums during manufacture. Many newer pieces of audio gear have omitted the RIAA equalization circuit completely, assuming that the majority of users will not desire the function. In spite of the enormous popularity of audio CDs, there is still a dedicated group of audiophiles who have a large library of record albums -- titles that are not available on CDs, or are out of print.

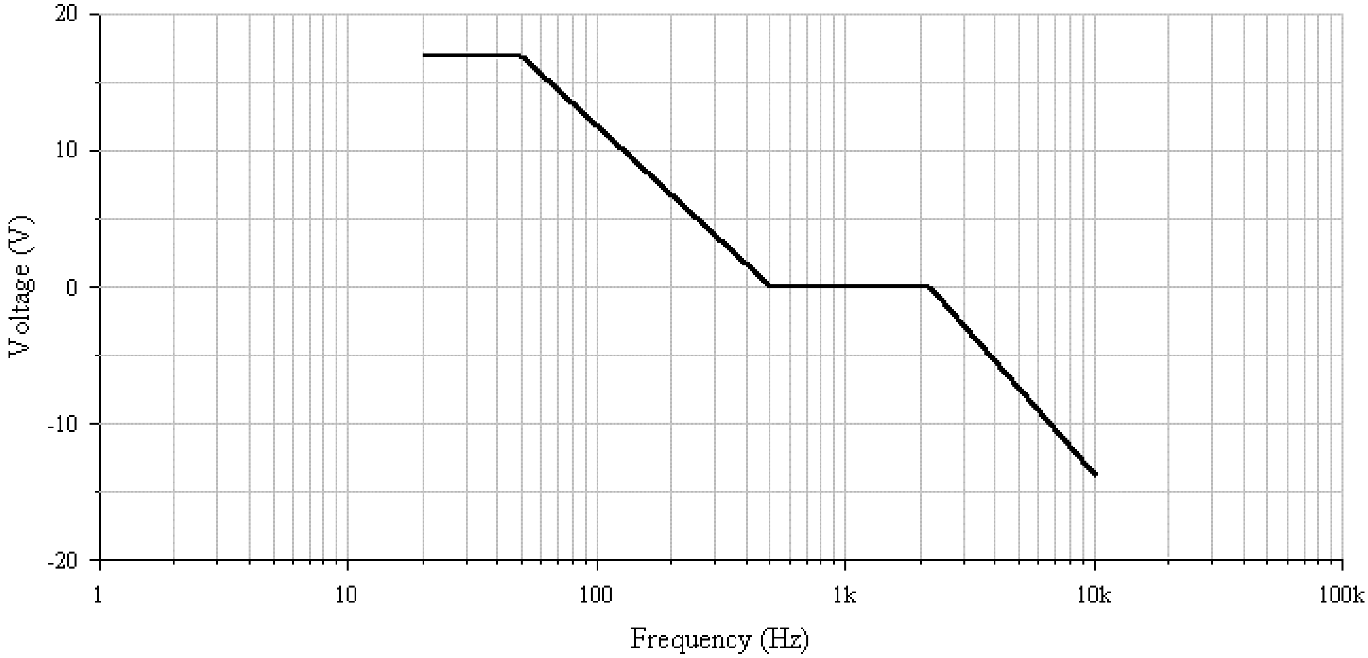

Fig. 9: The RIAA Equalization curve

The RIAA equalization curve has three breakpoints:

- +17 dB from 20 to 50 Hz
- 0 dB from 500 to 2120 Hz
- -13.7 dB at 10 kHz

RIAA equalization curves often include another breakpoint at 10 Hz to limit low-frequency "rumble" effects that could resonate with the turntable's tonearm. The standard input impedance in the circuits shown here is 47 kΩ. This impedance makes a convenient place to inject dc offset into single-supply circuits, so it is isolated from the phono cartridge by an input capacitor. The phono cartridge output level is assumed to be 12 mV.

Application circuits were evaluated from many sources in print and on the web. Many of these either did not work at all, did not easily translate to single-supply operation, or deviated markedly from the RIAA specification.

The first circuit topology presented here (see Fig. 10) was one of the most common, appearing in several sources. This circuit was tweaked manually to produce the closest possible conformance to the RIAA curve. A small additional gain resistor was sometimes added between the junction of R3 and C3 and the inverting input. It did not seem to be necessary, and this implementation contains the least number of passive components. There is even a web page that contains a Java-based calculator dedicated to this topology.

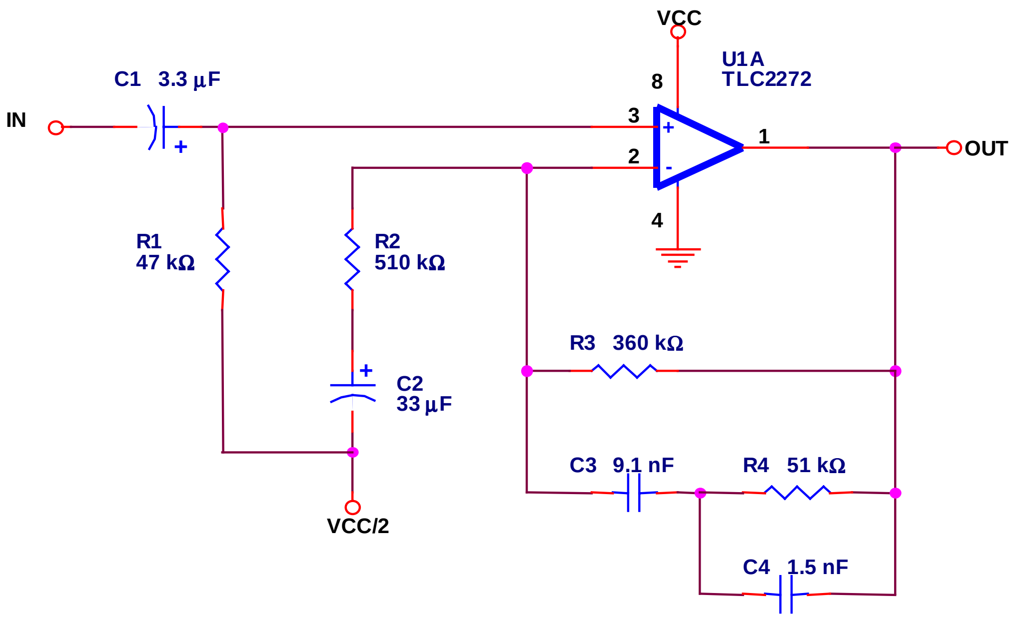

Fig. 10: RIAA Equalization Circuit From Commonly-Used Topology

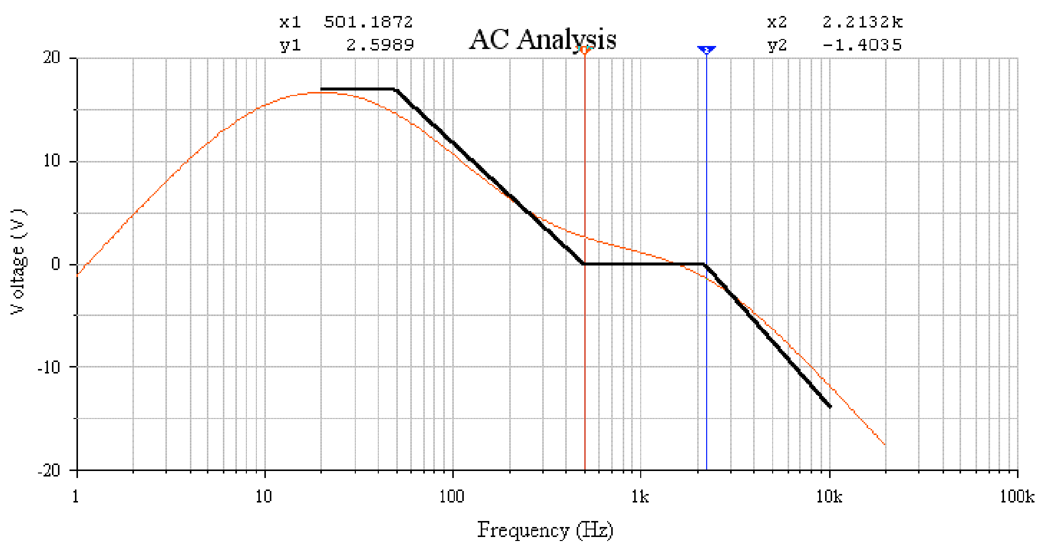

Fig. 11: Frequency/Voltage Characteristics Of Fig. 10 Topology

Several things are troublesome with this topology: No matter how much the circuit is optimized, the section from 500 Hz to 2.12 kHz is not simulated well. The first-order breakpoints that are possible with the single op amp only create a slight ripple on the characteristic curve (Fig. 11.) These breakpoints cry out for a second-order filter. This is very near the region where human hearing is most sensitive, so errors here will be the most audible. The musical content immediately below 1 kHz will be too loud, and the musical content immediately above 1 kHz will be too soft. Aesthetically, this will make the sound “muddy” and it will lack brilliance and tonal clarity.

C2 has a large capacitance value, which happens to be in the highest gain network in the circuit. Power-on transients will cause large, unexpected voltage swings -- possibly overloading the input to the next stage. They could also create loud, possibly destructive transients in loudspeakers. It is also hard to get precision values in electrolytic capacitors, which will lead to wide variations in response -- both in amplitude and the low-frequency roll-off breakpoint.

Fine-tuning this circuit is difficult with virtually all the components interacting.

The procedure for tuning is:

Set the low-frequency gain with R2 and R4: $LFG = \frac{R2}{100 \times R4} = 16.97dB$

Set the mid-frequency gain with R3: $MFG = \frac{R3}{100 \times R4} = 0dB$

Set the low-frequency roll-off with C4: $LFR = \frac{1}{2\pi \times R4 \times C4} = 9.46Hz$

Set the low-frequency breakpoint with C2: $LFB = \frac{1}{2\pi \times R2 \times C2} = 48.6Hz$

{The mid-frequency breakpoint is already determined by the values of R3 and C2: $MFB = \frac{1}{2\pi \times R3 \times C2} = 342Hz$ }

Set the high-frequency breakpoint with C3: $HFB = \frac{1}{2\pi \times R3 \times C3} = 2080Hz$

These steps must be followed in order. The initial selection of R2 determines the other components. It is unfortunate that there is no control over the mid-frequency breakpoint, which probably accounts for the error in the response in the curve above. The mid-frequency breakpoint is constrained to 342 Hz, when it should be 500 Hz.

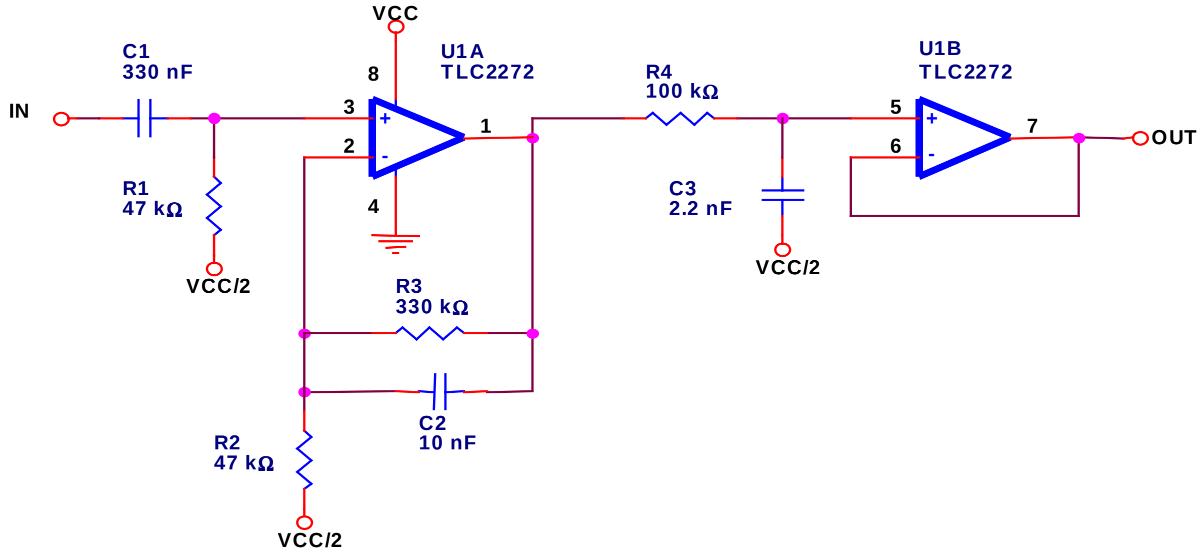

Fig. 12: RIAA Equalization Implementation Split Into Two Op Amps

Fine-tuning can be improved by splitting the implementation into two op amps (see Fig.12):

Set the low-frequency roll-off with R1 and C1: $LFR = \dfrac{1}{2\pi \times R1 \times C1} = 10.3Hz$

Set the low-frequency gain with R3 and R2: $LFG = \dfrac{R3}{R2} = 16.9dB$

Set the low-frequency breakpoint with C2: $LFB = \dfrac{1}{2\pi \times R3 \times C2} = 48.2Hz$

Set the HFB with R4 and C3: $HFB = \dfrac{1}{2\pi \times R4 \times C3} = 723Hz$

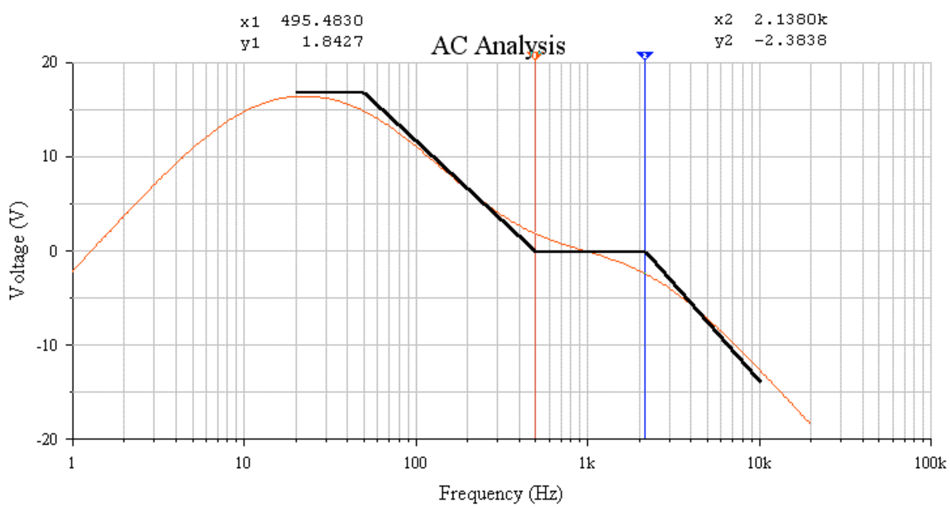

Fig. 13: Response Of the Revised Topology

The circuit (Fig. 12) will be the starting point for circuit a simulation of the RIAA curve. The response from 500 to 2.12 kHz should be flat at 0 dB but this first-order circuit is 1.8 dB too high at 500 Hz and 2.4 dB low at 2.12 kHz. Selecting the HFB at 723 Hz is a trick that shifts the response at 1 kHz down to 0 dB. This is a fairly drastic change, though. The first step in improving the RIAA characteristic is to change the 2.12 kHz portion to second order. A unity-gain Sallen-Key stage is selected.

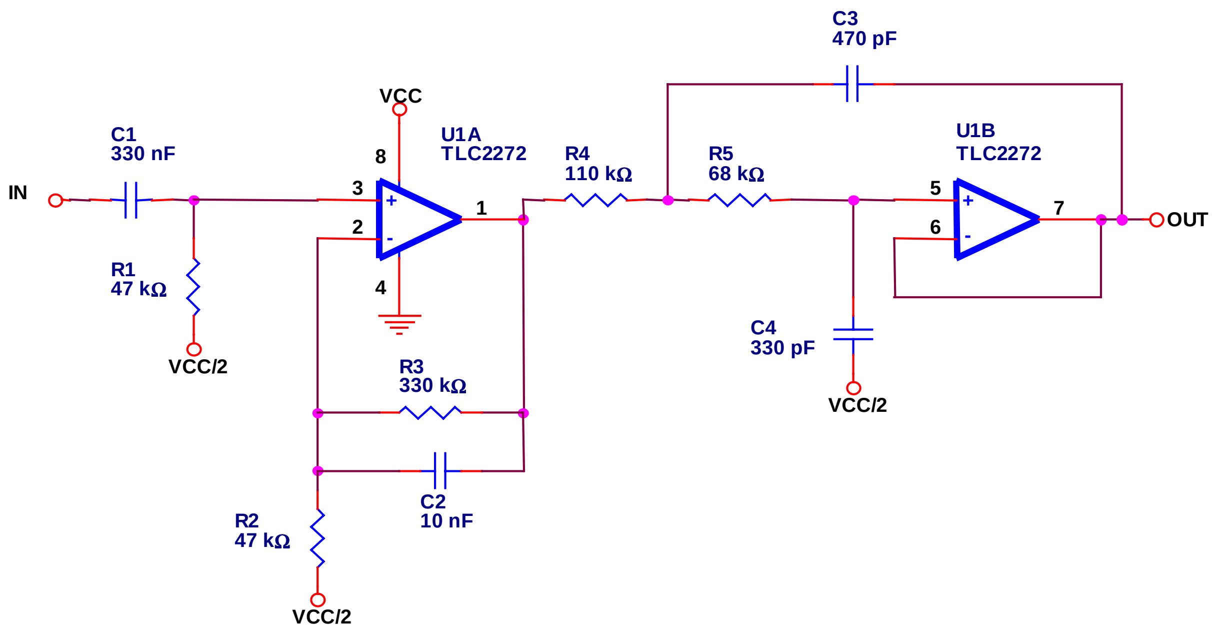

Fig. 14: Unity-Gain Sallen-Key Stage Is Selected For 2.12 kHz Section

Here, R4, R5, C3, and C4 control the 2.12 kHz breakpoint.

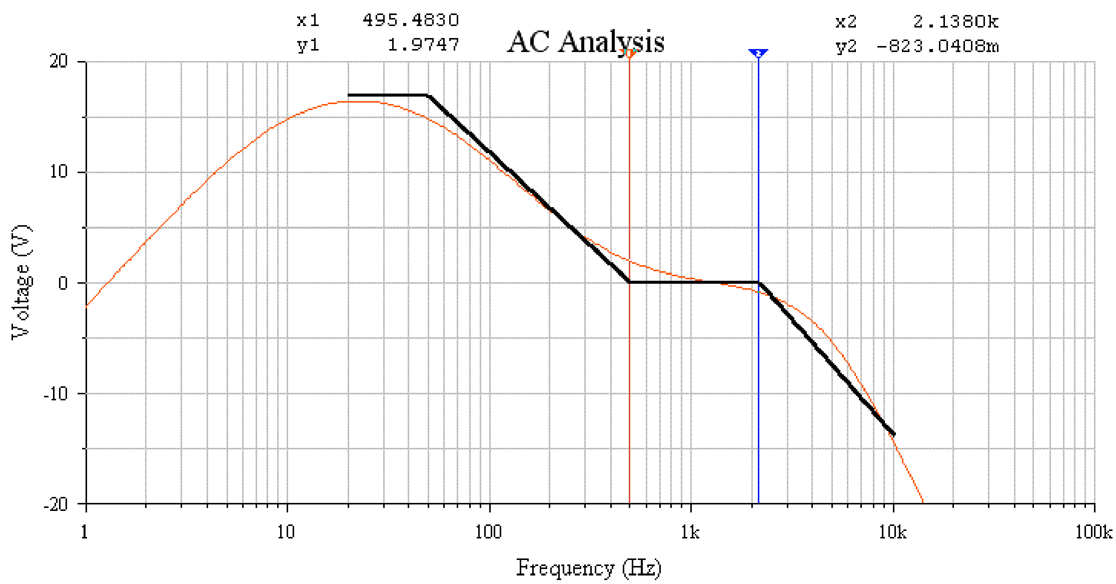

Fig. 15: Response Of The Modified Circuit

The 2.12 kHz response has improved from a 2.4 dB deviation off the curve to 0.3 dB. Unfortunately, there is less interaction with the 50 Hz low-pass filter, and the 500 Hz response is now 2 dB above ideal, instead of 1.8 dB. Clearly, another second-order filter is required. Accomplishing this requires a change in first-order stage topology, and an increase in complexity to four op amps.

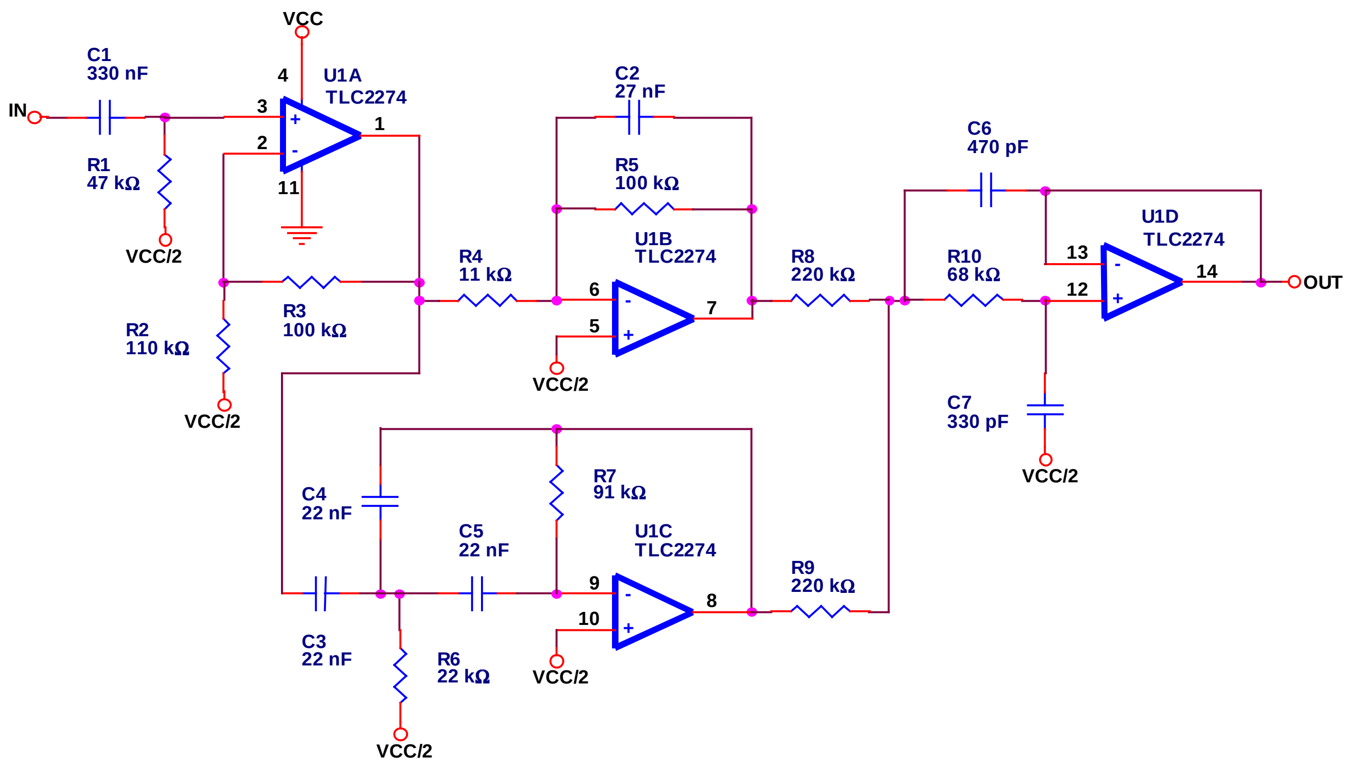

Fig. 16: Adding Another Second-Order Filter With Input Stage Changes

This circuit topology is very flexible -- most of the RIAA breakpoints are independently adjustable:

- R1 and C1 set the LFR as before.
- U1A, R2, R3 control the overall gain of the circuit.
- R4 and R5 control the LFG.
- R5 and C2 control the 50 Hz LFB.
- C3, C4, C5, R6, R7, and U1C form a 500 Hz high-pass filter that reverses the effect of the 50 Hz low-pass filter and flattens the response through 1 kHz until the 2.12 kHz LPF filter begins to affect the response.
- R8, R9, R10, C6, C7, U1D form the 2.12 kHz low pass filter as before, but the input resistor has been split into a summing resistor.

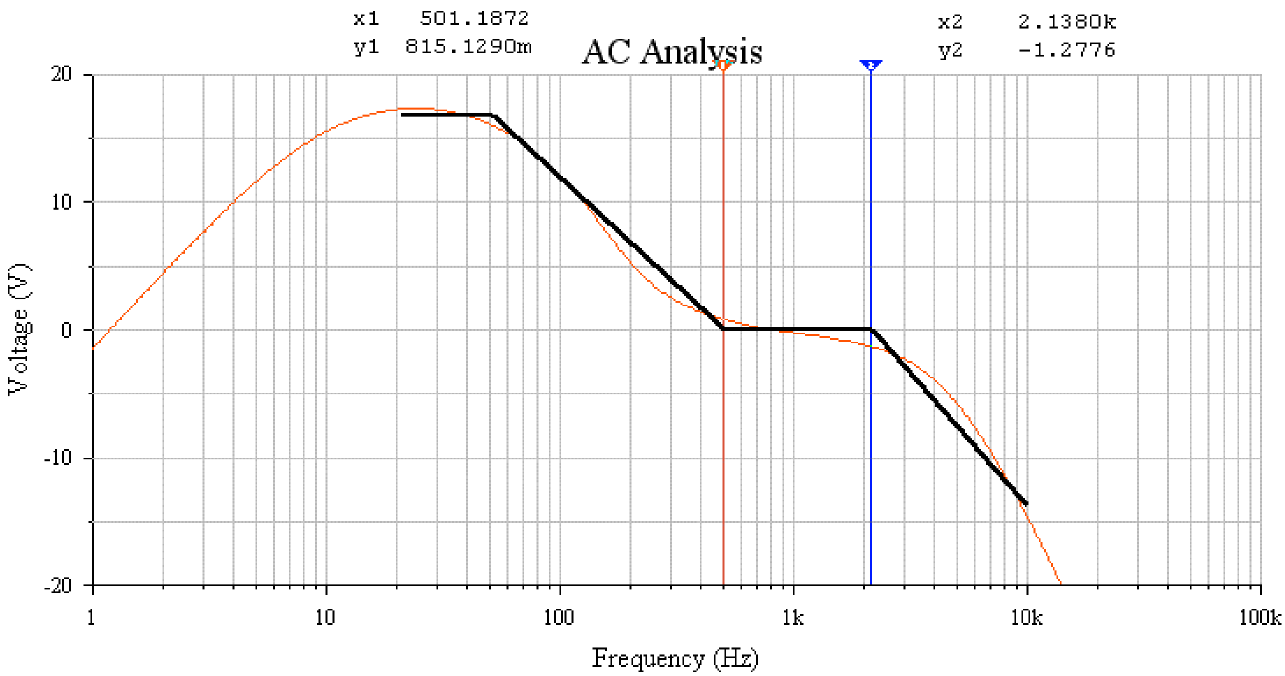

Fig. 17: Overall Response Of The Circuit in Fig. 16

The 500 Hz response is above the ideal curve by 0.8 dB, and the 2.12 kHz response is below the ideal curve by –1.3 dB. This circuit is about the best that can be achieved without using many more op amps and complex design techniques. It should produce very aesthetically-pleasing sound reproduction.