# An audio circuit collection, Part 2

**By Bruce Carter**
*Advanced Linear Products, Op Amp Applications*

## Introduction

This is the second in a series of articles on single-supply audio circuits. The reader is encouraged to review Part 1*, in the November 2000 issue of *Analog Applications Journal*, which concentrated on low-pass and high-pass filters. Part 2 concentrates on audio-notch-filter applications and curve-fitting filters.

## Audio notch filters

There are audio applications where a single frequency is undesirable and needs to be rejected. The frequency to be rejected is not part of the original audio and is annoying to the listener. Frequencies on either side of the rejected frequency, however, contain useful audio content. 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. Notch filters, compared to low-pass and high-pass filters, are hard to implement. Components that would cause only 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. Following are some hints about how to implement a notch with a reasonable degree of confidence.

*The original Part 1 release of this series had op amp polarity symbols reversed in several figures. Please download the corrected version at: www-s.ti.com/sc/techlit/slyt155

### 60-Hz hum filter

One of the most common problems with audio is the presence of a 60-Hz hum. Since 60 Hz is the frequency used for ac-power distribution, it is one of the most prevalent interference sources.

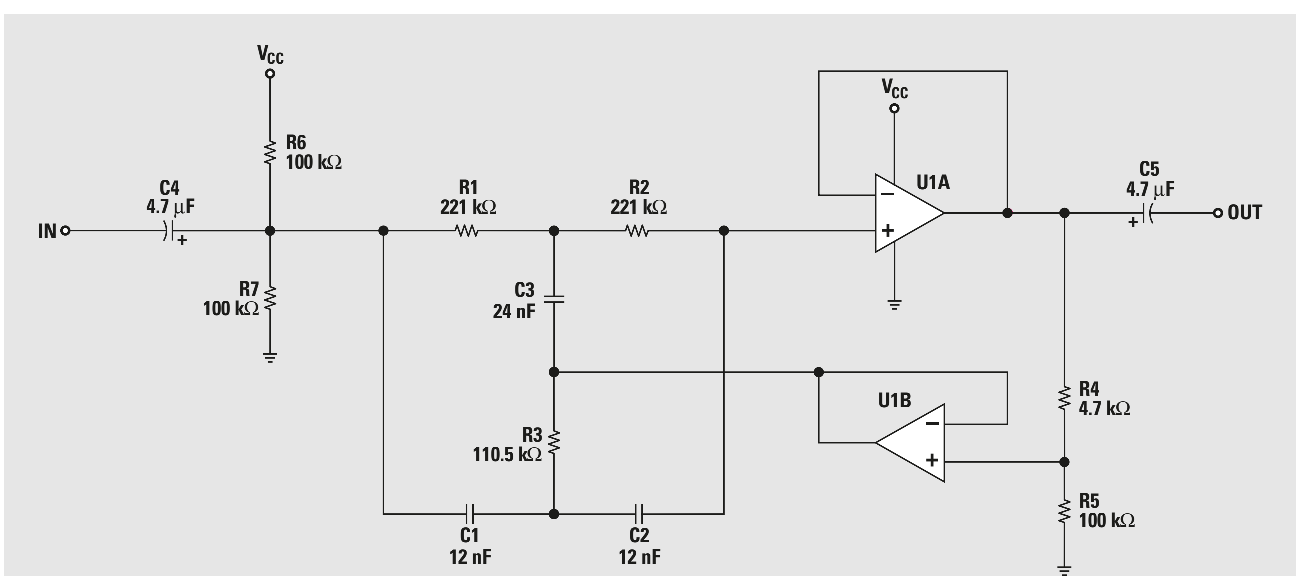
Usually, a 60-Hz hum is the result of poor grounding practices. It is better to attack the problem at the source than to filter the audio. Nevertheless, there may be situations where the grounding of a particular system may not be accessible. In that case, an add-on filter may be appropriate.

The 60-Hz hum filter shown in Figure 1 is based on twin-T configuration. This topology is very effective but can be temperamental. The circuit response is very dependent on the actual values of R1, R2, R3, C1, C2, and C3. All of these tuning components should be 1%, but that may not be enough. They should all be taken from the same lot, because parts manufactured at the same time tend to have the same characteristics.

If components are matched properly, performance can be very good. Mismatched components will seriously degrade the response. In the twin-T configuration, R3 is half the value of R1 and R2. 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 increases the

**Continued on next page**

**Figure 1. Configuration of a 60-Hz notch filter**



**Continued from previous page**

matching, because the designer can take easy steps to insure that the parts are from the same batch (off the same reel, out of the same box, etc.).

Note in Figure 1 that no half-supply reference is required. R6 and R7 generate half supply after the input capacitor. The theoretical response of this filter is shown in Figure 2. The ideal values produce a notch as shown. If, however, the RC combination is 1% off, the notch will be shifted left or right by 0.6 Hz, and rejection of the 60-Hz frequency could be less than 20 dB.

A way around that problem is to put a small potentiometer in series with R3, which should be reduced one or two standard E-96 values. Figure 3 shows circuit response with R3 varied from 107 kΩ to 113 kΩ.

Varying R3 over even 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, and most of the time as much as 30 or 40 dB. If a deeper null is needed, R1 and R2 need to be adjustable as well. They can be adjusted in a dual potentiometer to reduce the number of potentiometers in the circuit to 2. If the capacitors are matched (which is possible if they are from a single batch), 60-dB rejection is possible.

The circuit can be modified to reject hum from the 50-Hz line frequency of ac power in Europe and other parts of the world, as follows:

- 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 previous comments regarding the 60-Hz notch filter apply to the 50-Hz version as well. Theoretical response of the 50-Hz notch filter is shown in Figure 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.

**Figure 2. Response of 60-Hz notch filter**

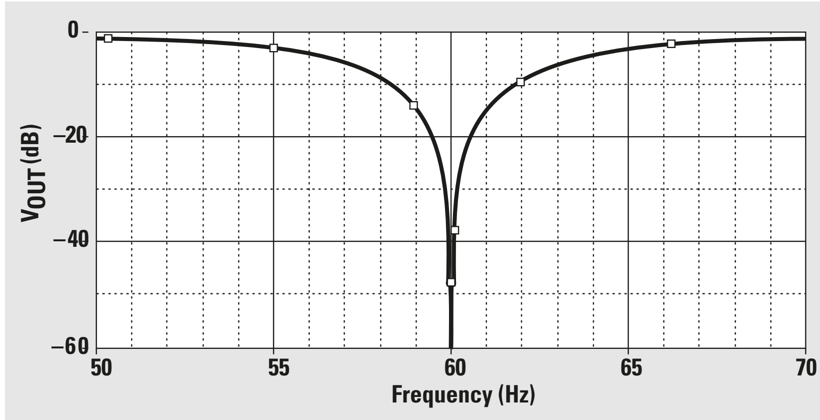



**Figure 3. Varying notch depth**

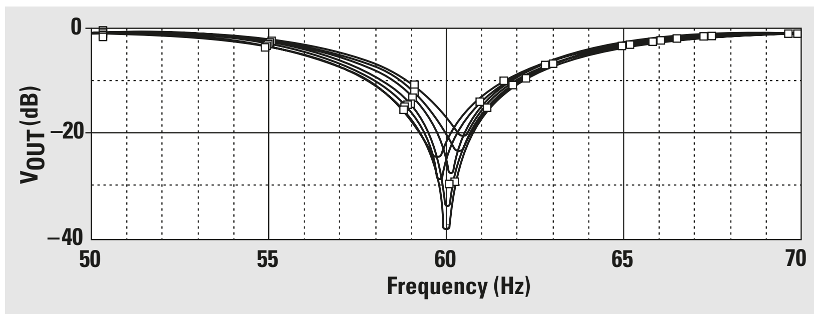
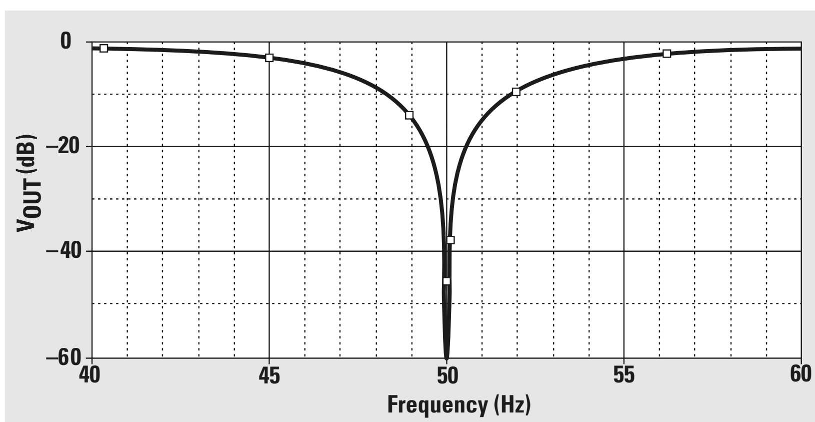



**Figure 4. Response of a 50-Hz notch filter**

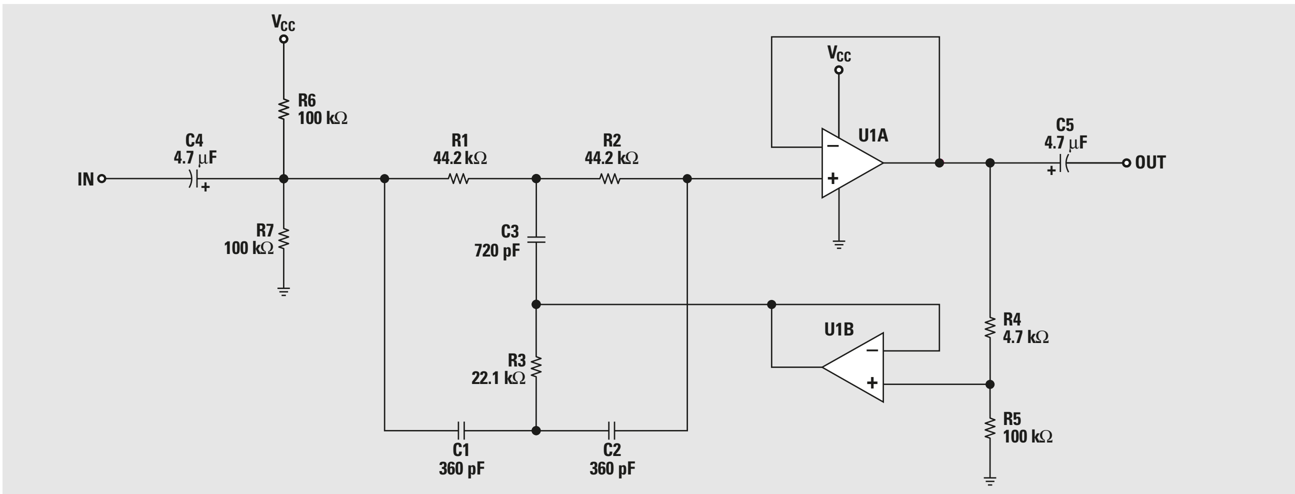



## Medium-wave whistle filter

In North America, medium-wave (AM) stations are separated by 10-kHz channels from 540 to 1700 kHz. AM frequency response is unlimited compared to FM, which is severely rolled off above 15 kHz. Unfortunately, there will be interference from the adjacent channels, 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—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 those used by most stations, and they are not exactly at 10 kHz. The FCC allows a tolerance of ±20 Hz from the assigned frequency, and that will make the two

**Figure 5. 10-kHz notch filter**




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. Many years ago, high-priced AM receivers used a high-Q LC filter, but tuning was so critical that it was of limited use. The op amp approach shown in Figure 5 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 attenuations 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. The response of this filter is shown in Figure 6.

In Europe and much of the rest of the world, the medium-wave transmission uses channels from 531 to 1611 kHz, separated by 9 kHz. This will cause a 9-kHz (instead of 10-kHz) tone in the received audio.

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).

The response of a well-tuned 9-kHz notch filter is shown in Figure 7.

Much of the world relies on short-wave radio stations for news and entertainment. Short-wave radio is transmitted on

**Continued on next page**

**Figure 6. Response of 10-kHz notch filter**

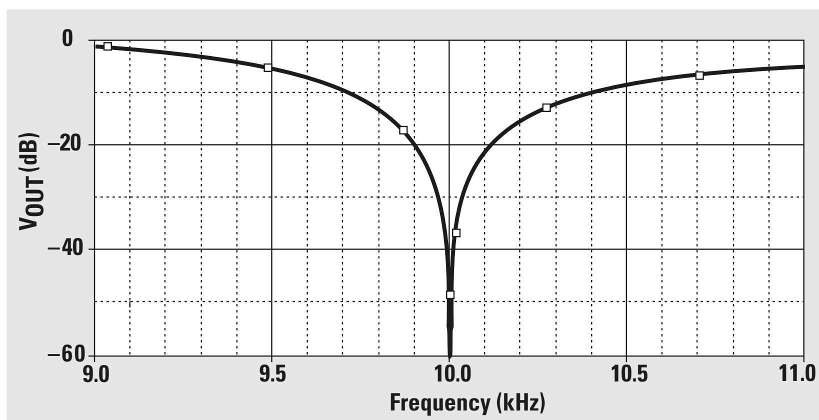
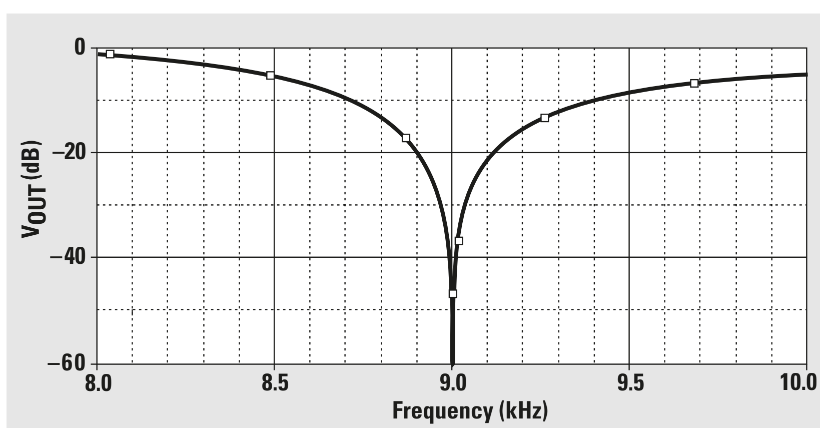



**Figure 7. Response of 9-kHz notch filter**



**Continued from previous page**

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).

The response of a well-tuned 5-kHz notch filter is shown in Figure 8.

This notch filter topology can be re-tuned 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 stations 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 to design 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; and a double-pole filter, 40 dB per decade. How does the designer implement a different roll-off?

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 (see Figure 9), which compensates for equalization applied to vinyl record albums during manufacture. Designers of 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 are still dedicated audiophiles who have a large library of record albums—titles that are not available on CDs or are out of print.

RIAA has the following response:

- 17 dB from 20 to 50 Hz,
- 0 dB from 500 to 2120 Hz, and
- –13.7 dB at 10 kHz.

**Figure 8. Response of 5-kHz notch filter**

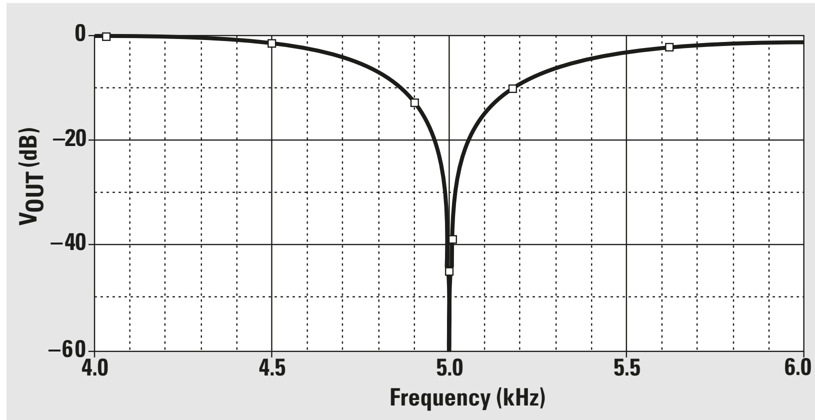



**Figure 9. The RIAA equalization curve**

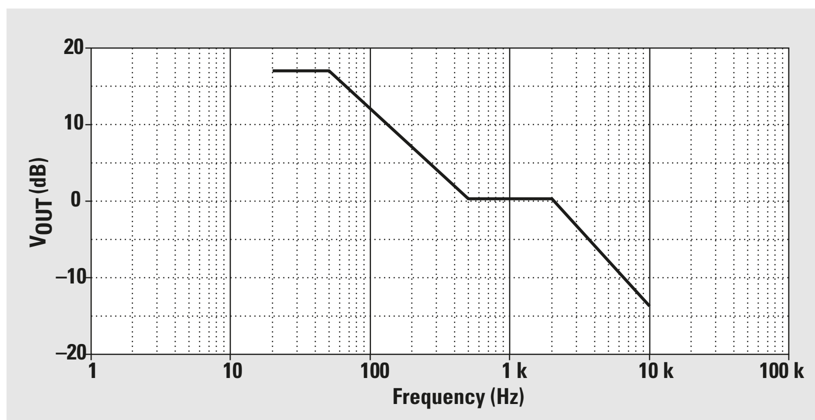



**Figure 10. RIAA equalization circuit**

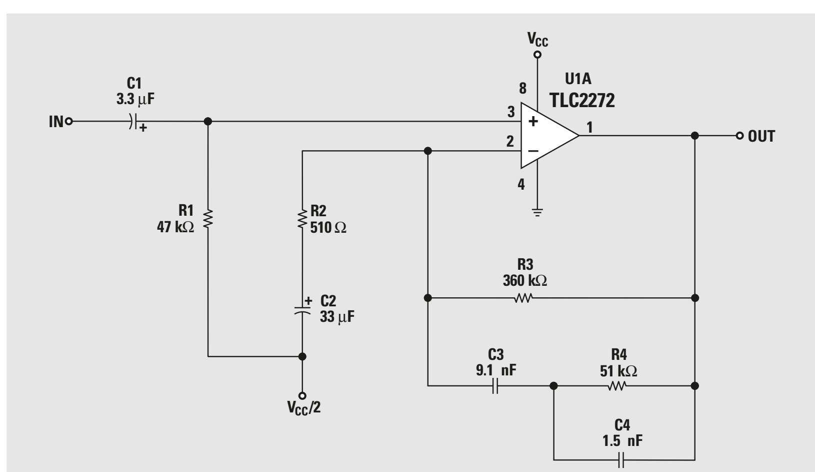
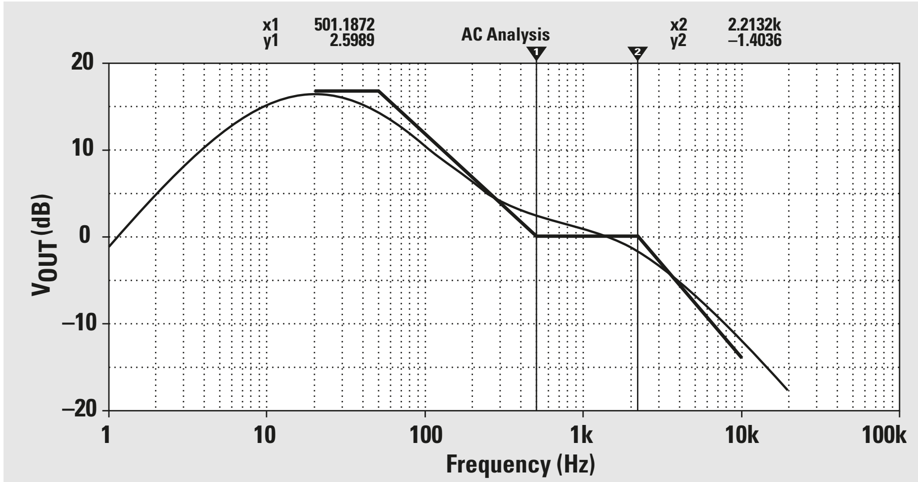


Figure 11. RIAA equalizer resonse




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

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

The circuit topology presented in Figure 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 smallest number of passive components. There is even a Web page that contains a Java-based calculator dedicated to this topology (see www.vwlowen.demon.co.uk/java/riaa.htm).

The implementation of the circuit in Figure 10 yields the curve shown in Figure 11. Several things are troublesome with this topology:

- No matter how much the circuit is optimized, the section from 500 Hz to 2120 Hz is not simulated well. The first-order breakpoints that are possible with the single op amp create only a slight ripple on the characteristic curve in Figure 11. These breakpoints require a second-order filter. This is very near the region where human hearing is the most sensitive and errors will be the most audible. The musical content immediately below 1 kHz will be too loud, and that immediately above 1 kHz will be too soft. Aesthetically, this will make the sound "muddy," lacking brilliance and tonal clarity.
- C2 is a large capacitance value that 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. Further, the difficulty of getting precision values of electrolytic capacitors will lead to wide variations in response—both of the amplitude and the low-frequency roll-off breakpoint.
- Fine-tuning this circuit is difficult; virtually all components interact.

The procedure for tuning is:

1. Set the low-frequency gain (LFG) with R2 and R3:

$$LFG = \frac{R3}{100 \times R2} = 16.97 \text{ dB}$$

2. Set the mid-frequency gain (MFG) with R4:

$$MFG = \frac{R4}{100 \times R2} = 0 \text{ dB}$$

3. Set the low-frequency roll-off (LFR) with C2:

$$LFR = \frac{1}{2\pi \times R2 \times C2} = 9.46 \text{ Hz}$$

4. Set the low-frequency breakpoint (LFB) with C3:

$$LFB = \frac{1}{2\pi \times R3 \times C3} = 48.6 \text{ Hz}$$

5. The mid-frequency breakpoint (MFB) is already determined by the values of R4 and C3:

$$MFB = \frac{1}{2\pi \times R4 \times C3} = 342 \text{ Hz}$$

6. Set the high-frequency breakpoint (HFB) with C4:

$$HFB = \frac{1}{2\pi \times R4 \times C4} = 2080 \text{ Hz}$$

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 of the curve (Figure 11). The mid-frequency breakpoint is constrained to 342 Hz when it should be 500 Hz.

Continued on next page

**Continued from previous page**

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

- Set the low-frequency roll-off with R1 and C1:

$$LFR = \frac{1}{2\pi \times R1 \times C1} = 10.3\ \text{Hz}$$

- Set the low-frequency gain with R3 and R2:

$$LFG = \frac{R3}{R2} = 16.9\ \text{dB}$$

- Set the low-frequency breakpoint with C2:

$$LFB = \frac{1}{2\pi \times R3 \times C2} = 48.2\ \text{Hz}$$

- Set the high-frequency breakpoint with R4 and C3:

$$HFB = \frac{1}{2\pi \times R4 \times C3} = 723\ \text{Hz}$$

The response of this circuit is shown in Figure 13.

The circuit will be the starting point for simulation of the RIAA curve. The response from 500 to 2120 Hz should be flat at 0 dB. This first-order circuit is 1.8 dB too high at 500 Hz, and 2.4 dB too low at 2120 Hz. Selecting the HFB at 723 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 2120-Hz portion to second-order. A unity-gain Sallen-Key stage is selected, as shown in Figure 14.

R4, R5, C3, and C4 control the 2120-Hz breakpoint. The response of the circuit changes to that shown in Figure 15. The 2120-Hz response has improved from 2.4-dB to 0.8-dB deviation from the curve. Unfortunately, there is less interaction with the 50-Hz low-pass filter; and the 500-Hz response is now 2 dB, instead of 1.8 dB, above ideal. 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, as shown in Figure 16.

**Figure 12. Dual op amp implementation of equalizer**

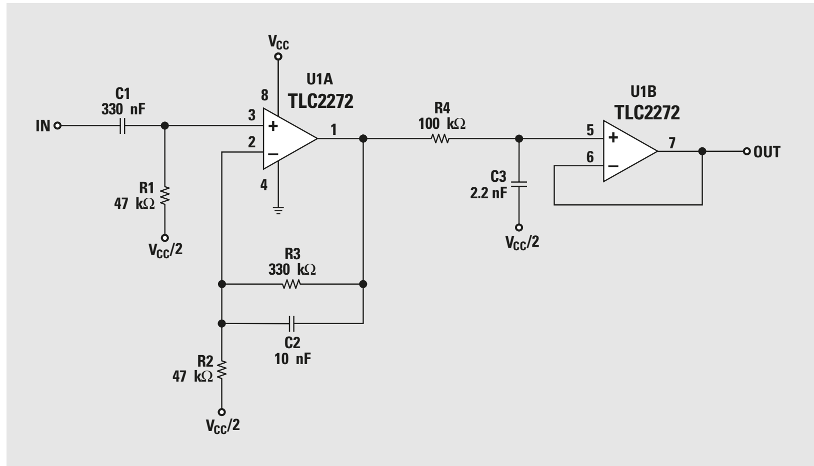



**Figure 13. Response of split op amp circuit**

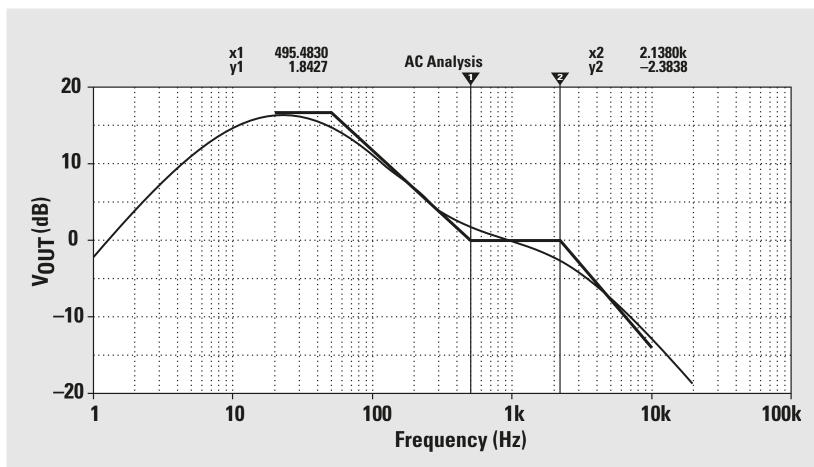
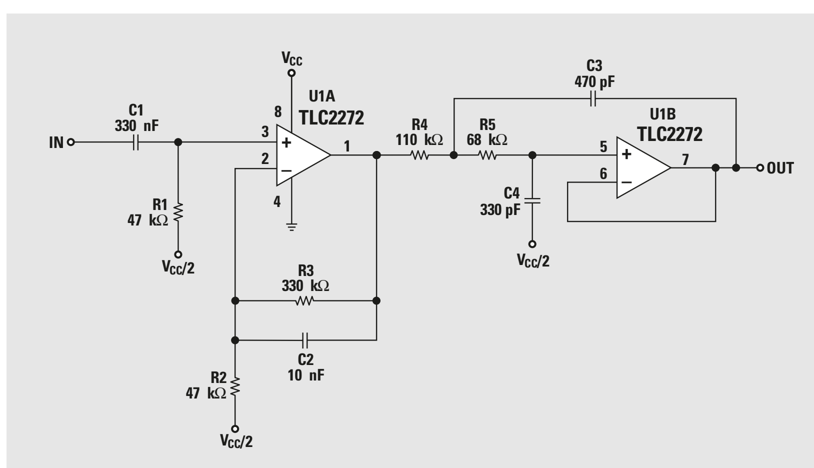



**Figure 14. Unity-gain Sallen-Key equalizer**



This circuit topology is very flexible. Most of the RIAA breakpoints are independently adjustable, as follows:

- R1 and C1 set the LFR as before.
- U1A, R2, and 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 2120-Hz low-pass filter begins to affect the response.
- R8, R9, R10, C6, C7, and U1D form the 2120-Hz low-pass filter as before, but the input resistor has been split into a summing resistor.

The overall response of the filter is shown in Figure 17.

The 500-Hz response is above the ideal curve by 0.8 dB, and the 2120-Hz response is below the ideal curve by –1.3 dB. This circuit is about the best that can be created without many more op amps and complex design techniques. It should produce very aesthetically pleasing sound reproduction.

**Figure 15. Response of the Sallen-Key stage**

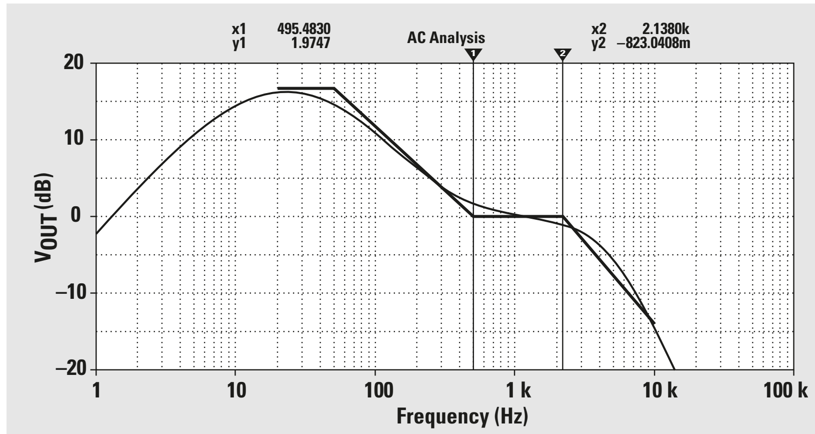
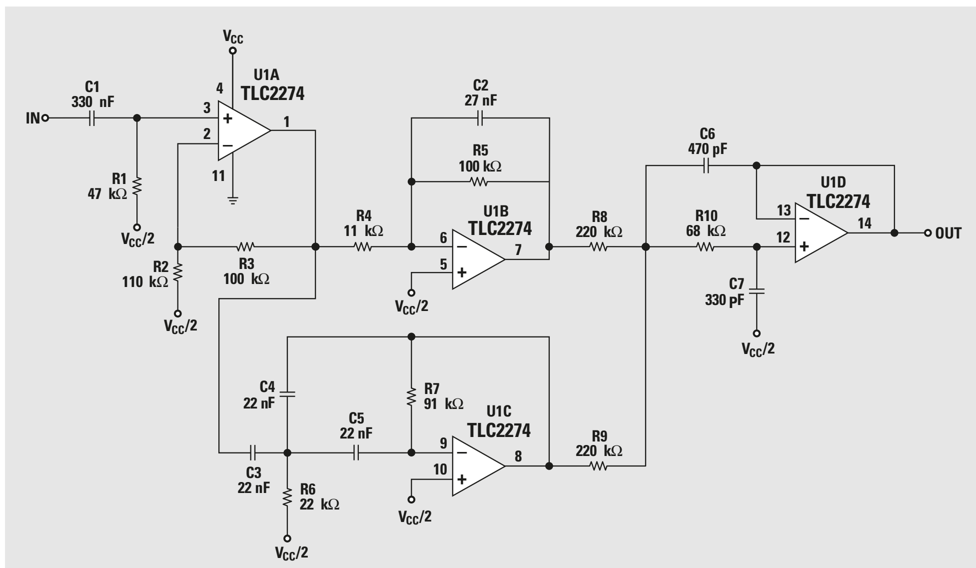



**Figure 16. Filter with four op amps**




## References

1. "Audio Circuits Using the NE5532/34," Philips Semiconductor (October 1984).
2. *Audio Radio Handbook*, National Semiconductor (1980).
3. "Op Amp Circuit Collection," National Semiconductor AN-031 (1978).

## Related Web sites

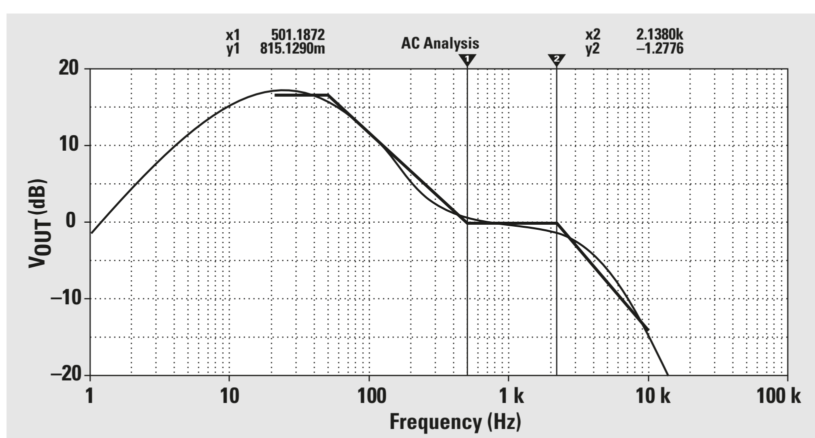
**www-s.ti.com/sc/techlit/slyt023**

**amplifier.ti.com**

**www.ti.com/sc/device/tlc2274**

**www.vwlowen.demon.co.uk/java/riaa.htm**

**Figure 17. Response of filter with four op amps**



## IMPORTANT NOTICE



Following are URLs where you can obtain information on other Texas Instruments products and application solutions:

| **Products** | |
|---|---|
| Amplifiers | **amplifier.ti.com** |
| Data Converters | **dataconverter.ti.com** |
| DSP | **dsp.ti.com** |
| Interface | **interface.ti.com** |
| Logic | **logic.ti.com** |
| Power Mgmt | **power.ti.com** |
| Microcontrollers | **microcontroller.ti.com** |

| **Applications** | |
|---|---|
| Audio | **www.ti.com/audio** |
| Automotive | **www.ti.com/automotive** |
| Broadband | **www.ti.com/broadband** |
| Digital control | **www.ti.com/digitalcontrol** |
| Military | **www.ti.com/military** |
| Optical Networking | **www.ti.com/opticalnetwork** |
| Security | **www.ti.com/security** |
| Telephony | **www.ti.com/telephony** |
| Video & Imaging | **www.ti.com/video** |
| Wireless | **www.ti.com/wireless** |

# *TI Worldwide Technical Support*

## *Internet*

**TI Semiconductor Product Information Center Home Page**
support.ti.com

**TI Semiconductor KnowledgeBase Home Page**
support.ti.com/sc/knowledgebase

## *Product Information Centers*

**Americas**

| | | | |
|---|---|---|---|
| Phone | +1(972) 644-5580 | Fax | +1(972) 927-6377 |
| Internet/Email | support.ti.com/sc/pic/americas.htm | | |

**Europe, Middle East, and Africa**

| | | | |
|---|---|---|---|
| Phone | | | |
| Belgium (English) | +32 (0) 27 45 54 32 | Netherlands (English) | +31 (0) 546 87 95 45 |
| Finland (English) | +358 (0) 9 25173948 | Russia | +7 (0) 95 7850415 |
| France | +33 (0) 1 30 70 11 64 | Spain | +34 902 35 40 28 |
| Germany | +49 (0) 8161 80 33 11 | Sweden (English) | +46 (0) 8587 555 22 |
| Israel (English) | 1800 949 0107 | United Kingdom | +44 (0) 1604 66 33 99 |
| Italy | 800 79 11 37 | | |
| Fax | +(49) (0) 8161 80 2045 | | |
| Internet | support.ti.com/sc/pic/euro.htm | | |

**Japan**

| | | | |
|---|---|---|---|
| Fax | | | |
| International | +81-3-3344-5317 | Domestic | 0120-81-0036 |
| Internet/Email | | | |
| International | support.ti.com/sc/pic/japan.htm | | |
| Domestic | www.tij.co.jp/pic | | |

**Asia**

| | | | |
|---|---|---|---|
| Phone | | | |
| International | +886-2-23786800 | | |
| Domestic | Toll-Free Number | | Toll-Free Number |
| Australia | 1-800-999-084 | New Zealand | 0800-446-934 |
| China | 800-820-8682 | Philippines | 1-800-765-7404 |
| Hong Kong | 800-96-5941 | Singapore | 800-886-1028 |
| Indonesia | 001-803-8861-1006 | Taiwan | 0800-006800 |
| Korea | 080-551-2804 | Thailand | 001-800-886-0010 |
| Malaysia | 1-800-80-3973 | | |
| Fax | 886-2-2378-6808 | Email | tiasia@ti.com |
| Internet | support.ti.com/sc/pic/asia.htm | | ti-china@ti.com |

**C011905**




Mailing Address: Texas Instruments
Post Office Box 655303
Dallas, Texas 75265