An audio circuit collection, Part 1

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Advanced Analog Products, Op Amp Applications

Introduction

This is the first of two articles on audio circuits. New operational amplifiers from Texas Instruments have excellent audio performance and can be used in high-performance applications.

There have been many collections of op amp audio circuits in the past, but all of them focus on split-supply circuits. Often, the designer who has to operate a circuit from a single supply does not know how to perform the conversion. Single-supply operation requires a little more care than split-supply circuits. The designer should read and understand the introductory material.

Split supply vs. single supply

All op amps have two power pins. In most cases they are labeled $V_{CC}+$ and $V_{CC}-$. Sometimes, however, they are labeled V_{CC} and GND. This is an attempt on the part of the data sheet author to categorize the part as split-supply or single-supply, but it does not mean that the op amp has to be operated with the split or single supply shown by the data sheet. It may or may not be able to operate from different voltage rails. Consult the data sheet, especially the absolute maximum ratings and voltage swing specifications, before operating at anything other than the recommended power supply voltage(s).

Most analog designers know how to use op amps with a split power supply. In a split-power-supply system, the input and output are referenced to ground. The power supply consists of a positive supply and an equal and opposite negative supply. The most common values are ±15 V, but ±12 V and ±5 V are also used. The input and output voltages are centered on ground and swing both positive and negative to $V_{OM}\pm$, the maximum peak output voltage swing.

A single-supply circuit connects the op amp power pins to a positive voltage and ground. The positive voltage is connected to $V_{CC}+$, and ground is connected to $V_{CC}-$ or GND. A virtual ground, halfway between the positive supply voltage and ground, is the "ground" reference for the input and output voltages. Voltage swings above and below this virtual ground to $V_{OM}\pm$. Some newer op amps have different high- and low-voltage rails, which are specified in data sheets as V_{OH} and V_{OL}, respectively.

5 V is a common value for single supplies, but voltage rails are becoming lower, with 3 V and even lower voltages becoming common. Because of this, single-supply op amps are often "rail-to-rail" devices to avoid losing dynamic range. "Rail-to-rail" may or may not apply to both the input and output stages. Be aware that even though a device may be specified as "rail-to-rail," some specifications may degrade close to the rails. Be sure to consult the data sheet for complete specifications on both the inputs and the outputs. It is the designer's obligation to make sure that the voltage rails of the op amp do not degrade system performance.

Virtual ground

Single-supply operation requires the generation of a "virtual ground" at a voltage equal to VCC/2. The external circuit can be a voltage divider bypassed by a capacitor, a voltage divider buffered by an op amp, or preferably a power-supply splitter such as the Texas Instruments TLE2426. Figure 1 shows how to generate a half-supply reference if the designer insists on using an op amp.

R1 and R2 are equal values selected with power consumption vs. allowable noise in mind. C1 forms a low-pass filter to eliminate conducted noise on the voltage rail. R3 is a small (47-Ω) resistor that forms a low-pass filter with C2, eliminating some of the internally generated op amp noise. The value of C2 is limited by the drive capability of the op amp.

In the circuits in the figures that follow, the virtual ground is labeled "VCC/2." This voltage comes from either the TLE2426 rail-splitter or the circuit in Figure 1. If the latter is used, the overall number of op amps in the design is increased by one.

Passive components

The majority of the circuits given in this series have been designed with standard capacitor values and 5% resistors. Capacitors should be of good quality with 5% tolerance wherever possible. Component variations will affect the operation of these circuits, usually causing some degree of ripple or increased roll-off as the balance of Chebyshev and Butterworth characteristics is disturbed. These should be slight—almost imperceptible.

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Figure 1. Half-supply generator
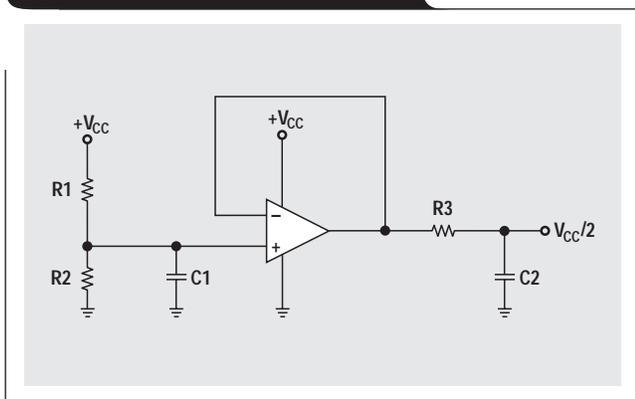
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Speech filter

Human speech most frequently occupies an audio spectrum of 300 Hz to 3 kHz. There is a requirement, especially in phones, to limit the frequency response to this range. Frequently, this function is performed with a DSP chip. DSP chips, however, require an anti-aliasing filter to reject high-frequency components. The anti-aliasing filter requires an op amp. Since there is already an op amp anyway, why not consider adding a second and doing the entire function with analog components? An additional op amp (plus one for the half-supply reference) can perform the filtering with no aliasing problems, freeing the DSP for other tasks.

Figure 2 shows a single-supply phone speech filter in two op amps, five capacitors, and four resistors. This circuit is designed to be low-power and compact, and is scalable for even lower power consumption.
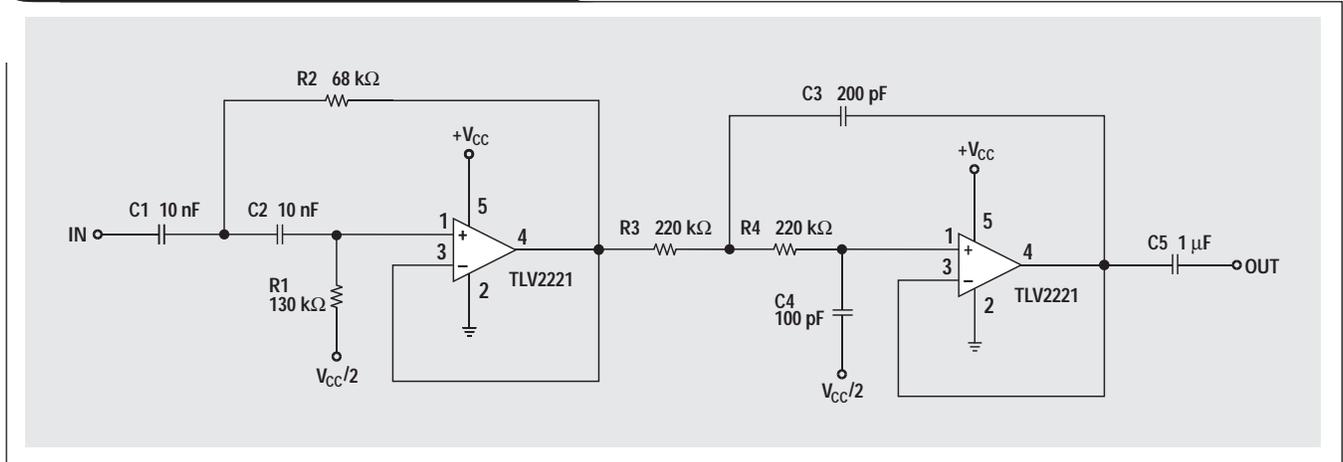
Figure 2. Single-supply phone speech filter

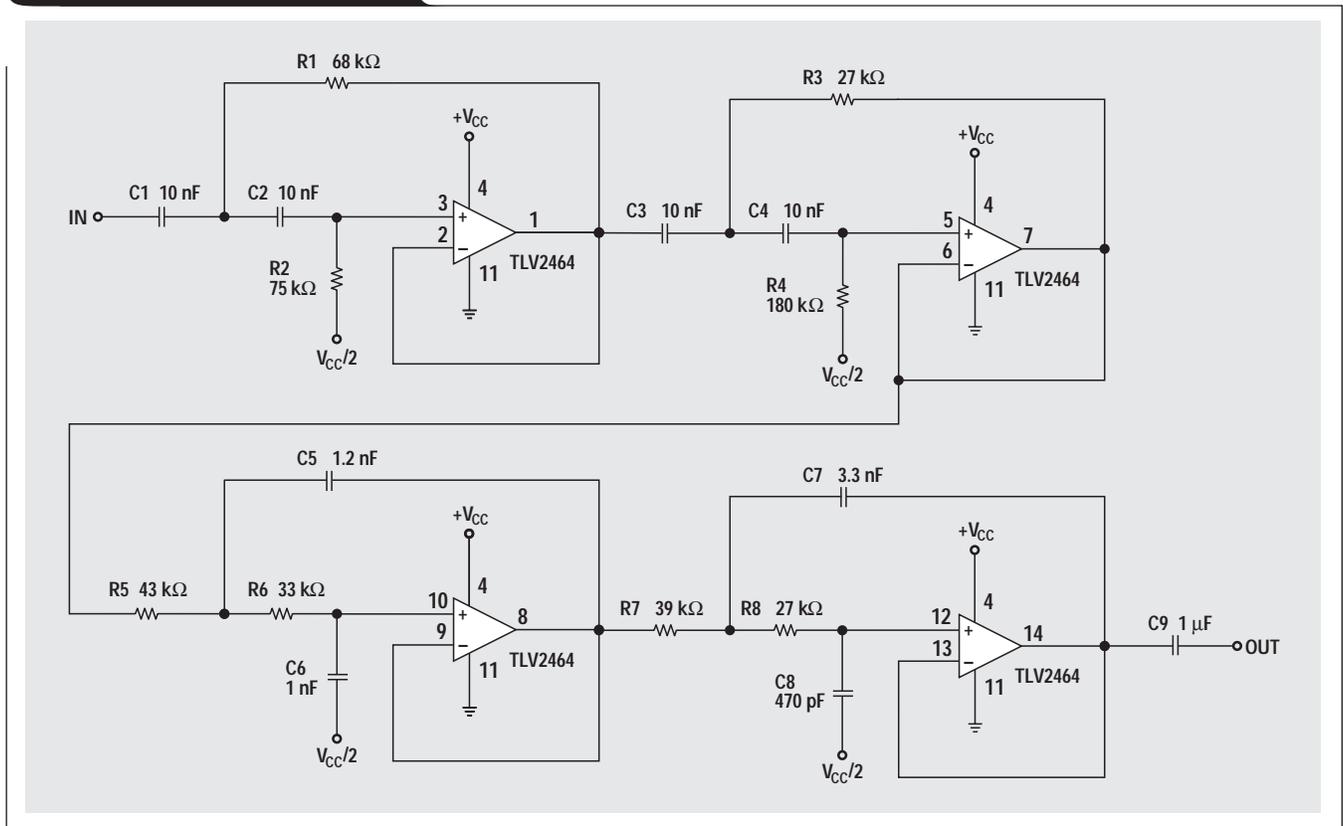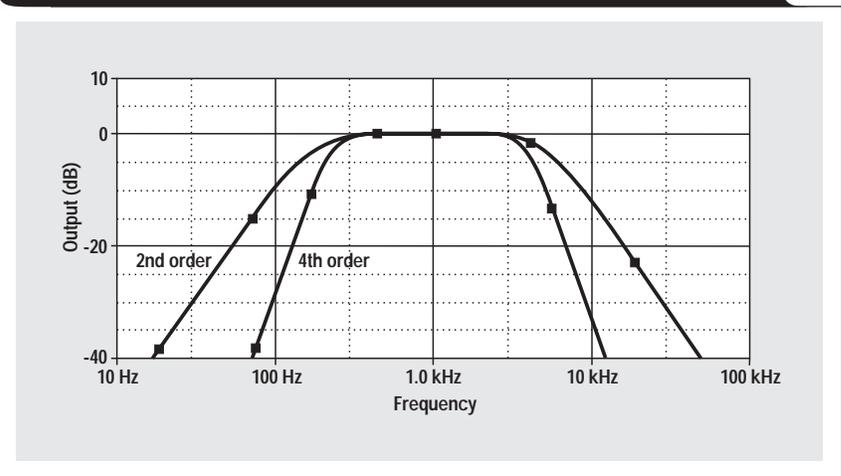**Figure 3. Second-order circuit**

Figure 3 shows a second-order circuit. Nearby out-of-band signals, such as 60-Hz hum, are not rejected very well. This may be acceptable in cellular telephone headsets, but may not be for large switchboard consoles. A fourth-order speech filter, although more complex, can be implemented in a single quad op amp (with an external VCC/2 reference).

The response of the second- and fourth-order speech filters is shown in Figure 4. The 60-Hz rejection of the fourth-order filter is greater than 40 dB, while that of the second-order filter is about 15 dB. Both filters have been designed to have an imperceptible 0.5-dB roll-off at 300 Hz and 3 kHz.

Crossover filter

Inside any multiple-speaker cabinet is an array of inductors and capacitors that directs different frequency ranges to each speaker. The inductors and capacitors, however, have to handle the full output power of the power amplifier. Inductors for low frequencies in particular tend to be large, heavy, and expensive.

Another disadvantage of the inductor and capacitor crossover network is that it is a first-order network. At high volume, destructive levels of audio can be transferred to speakers not designed to handle a given frequency range. If a speaker is incapable of moving in response to stimulation, the only way the energy can be dissipated is in heat. The heat can build up and burn out the voice coil.

Figure 4. Response of second- and fourth-order speech filters

A number of audiophiles are beginning to talk about the virtues of bi-amplification, or even multiple amplification. In this technique, the crossover network is applied to the audio source before amplification instead of after it. Each speaker in the cabinet is then driven by a separate amplifier stage that is optimized for the speaker.

The primary reason for bi-amplification or multiple amplification is that human hearing is not equally sensitive to all frequencies. The human ear is relatively insensitive to low and high frequencies, but audio amplifiers are designed to have flat response (constant power) across

Figure 5. Crossover network
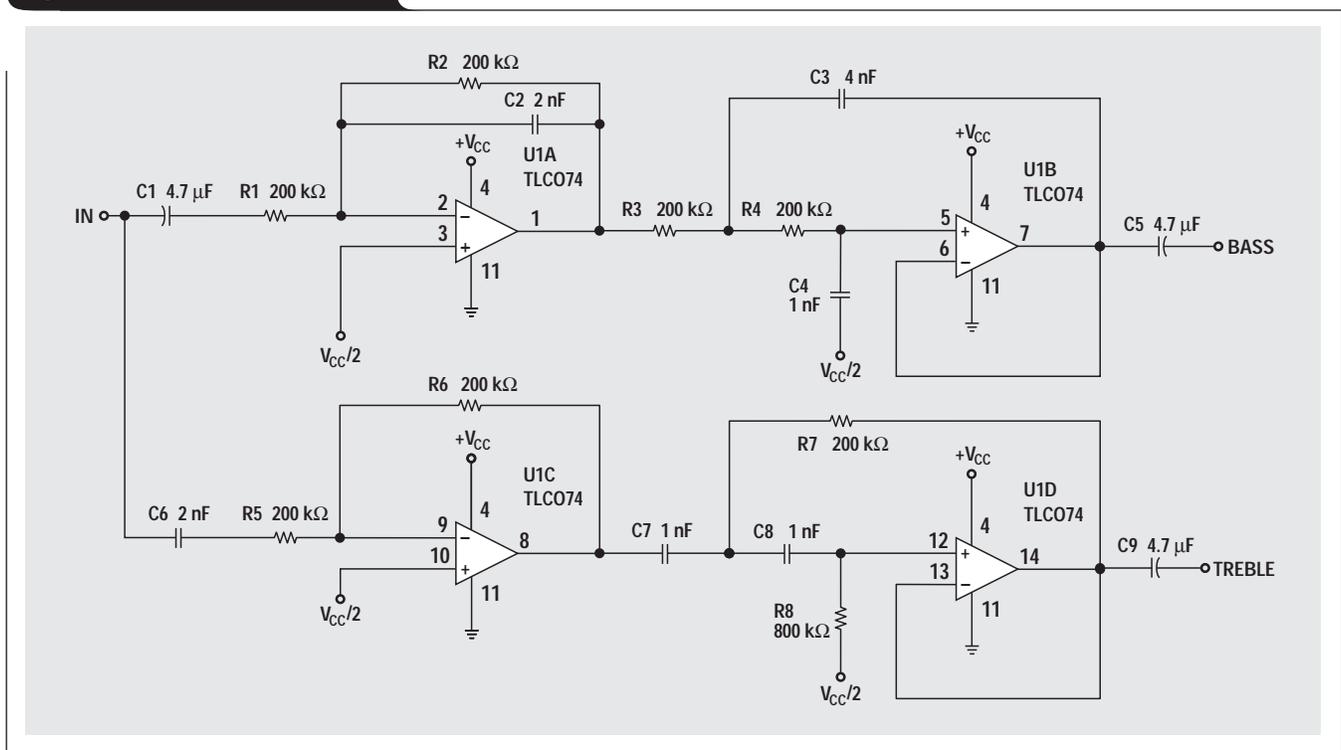
the audio band. The result is that the listener uses tone controls or graphic equalizers to compensate for the human hearing curve to make the sound pleasing. This compensates to some degree for the differential power requirements, but most tone controls are limited to ±20 dB—not nearly enough to make up for human hearing sensitivity at really low or really high frequencies. Tone control characteristics are linear; even if they were devised with more gain, they still would not follow the human hearing curve. Graphic equalizers are limited to discrete frequency values and produce an unpleasant degree of ripple when several adjacent controls are turned up.

Human hearing is not sensitive to low frequencies, so more power is required to reproduce them at a level that can be heard. A high-power class "B" amplifier can drive a large bass woofer. Crossover notch distortion from the class "B" topology is inaudible at these frequencies, and the efficiency of the amplifier allows it to generate a lot of power with relatively little heat.

Hearing is most sensitive in the midrange frequencies, for which the best amplifier is a relatively low-power, very low-distortion, class "A" amplifier. As little as 10 watts can produce deafeningly loud audio in this frequency range.

But what about high frequencies? There are purists who would insist that high frequencies should be amplified the same way as low, so that the human ear could discern them as well. While this is technically true, there are some reasons why it is not desirable:

- The energy required to accelerate a speaker cone to a given displacement at 20 kHz is 1000 times that required to accelerate a speaker cone to that displacement at 20 Hz. There are some piezo- and ceramic-type tweeters that can produce high output levels, but they require correspondingly high amounts of energy to drive. These output levels are enough to shatter glass and eardrums.

- The spectral content of almost all music is weighted with a "pink" characteristic. Simply stated, there is much more middle- and low-frequency content than high-frequency content.

- Because there is not much high-frequency spectral content but a constant level of white noise throughout the spectrum, a lot of amplification in this range will increase audio perception of noise at high frequencies.

The crossover network shown in Figure 5 routes low (bass) frequencies to a woofer, and midrange and high frequencies (treble) to a tweeter. This is a very common application, because many speaker cabinets contain only a woofer and tweeter.

A crossover frequency of 400 Hz has been selected, which should suffice for the majority of applications. The filter sections are third-order, which will minimize energy to the wrong speakers.

This circuit was designed to be very easy to build. The op amp sections can be interchanged, of course. There are only two capacitor values and one resistor value! Three 4.7-μF electrolytic capacitors are used for decoupling; they are sufficiently large to insure that they have no effect on the frequencies of interest. The 2-nF and 4-nF capacitors can be formed by connecting 1-nF capacitors in parallel. R8, the 800-kΩ resistor, can be made by connecting four 200-kΩ resistors in series.

A subwoofer section can be added to the crossover network in Figure 5 to enhance subsonic frequencies (see Figure 6).

Figure 6 shows a true subwoofer circuit. It will not work with 6- or 8-inch "subwoofers." It is for 15- to 18-inch woofers in a good infinite-baffle, bass-reflex, or folded-horn enclosure, driven by an amplifier with at least 100 watts. Most of the gain is below the range of human hearing; and these frequencies, when used in recorded material, are designed to be felt, not heard. The filter is designed to give 20 dB of gain to 13 Hz, rolling off to unity gain at about 40 Hz. There is no broadcast material in the United States that extends below 50 Hz; even most audio CDs do not go below 20 Hz. This will prove most useful for home theater applications, which play material that does have subsonic audio content. Examples are "Earthquake" and the dinosaur stomp in "Jurassic Park."

The combined response of the three circuits is shown in Figure 7. One active crossover network will be required per channel, with the exception of the subwoofer crossover.

Figure 6. Subwoofer circuit
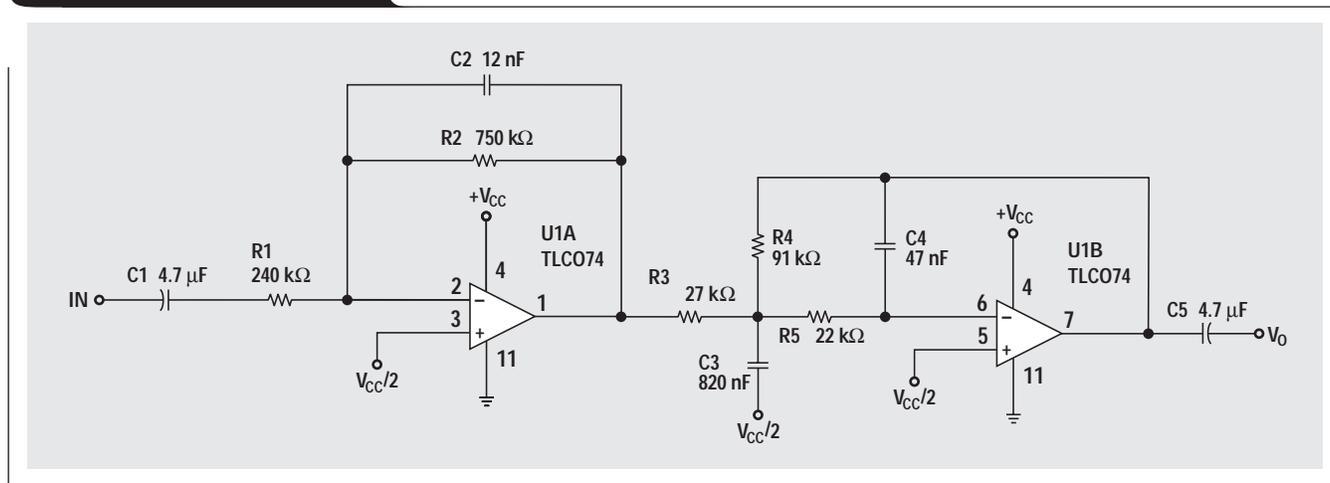
There is no stereo separation of low bass frequencies, and either channel (or both) can be used to drive the subwoofer circuit (sum into an inverting input with a second C1 and R1).

Tone control

One rather unusual op amp circuit is the tone control circuit shown in Figure 8. It bears some superficial resemblance to the twin T circuit configuration, but it is not a twin T topology. It is actually a hybrid of one-pole, low-pass, and high-pass circuits with gain and attenuation.

The midrange frequency for the tone adjustments is 1 kHz. It gives about ±20 dB of boost and cut for bass and treble. The circuit is a minimum-component solution that limits cost. This circuit, unlike other similar circuits, uses linear instead of logarithmic pots. Two different potentiometer values are unavoidable, but the capacitors are the same value except for the coupling capacitor. The ideal capacitor value is 0.016 µF, which is an E-24 value; so the more common E-12 value of 0.015 µF is used instead. Even that value is a bit odd, but it is easier to find an oddball capacitor value than an oddball potentiometer value.

The plots in Figure 9 show the response of the circuit with the pots at the extremes and at the 1/4 and 3/4 positions. The middle position, although not shown, is flat to within a few millidecibels. The compromises involved in reducing circuit cost and in using linear potentiometers lead to some slight nonlinearities. The 1/4 and 3/4 positions are not exactly 10 and –10 dB, meaning that the pots are most sensitive towards the end of their travel. This may be preferable to the listener, giving a fine adjustment near the middle of the potentiometers and more rapid adjustment near the extreme positions. The center frequency shifts slightly, but this should be inaudible. The frequencies nearer the midrange are adjusted more rapidly than the frequency extremes, which also may be more desirable to the listener. A tone control is not a precision audio circuit, and therefore the listener may prefer these compromises.

References

1. *Audio Circuits Using the NE5532/54*, Philips Semiconductor, Oct. 1984.
2. *Audio Radio Handbook*, National Semiconductor, 1980.
3. *Op Amp Circuit Collection*, National Semiconductor AN-031.

Related Web sites

www.ti.com/sc/docs/products/analog/ *device***.html**

Replace *device* with tlc074, tlc2272, tle2426, tlv2221, or tlv2464

www.ti.com/sc/docs/products/msp/amp_comp/ default.htm

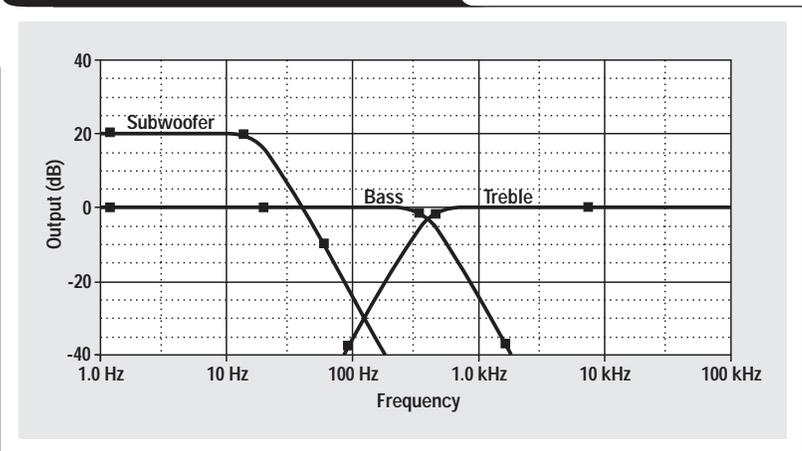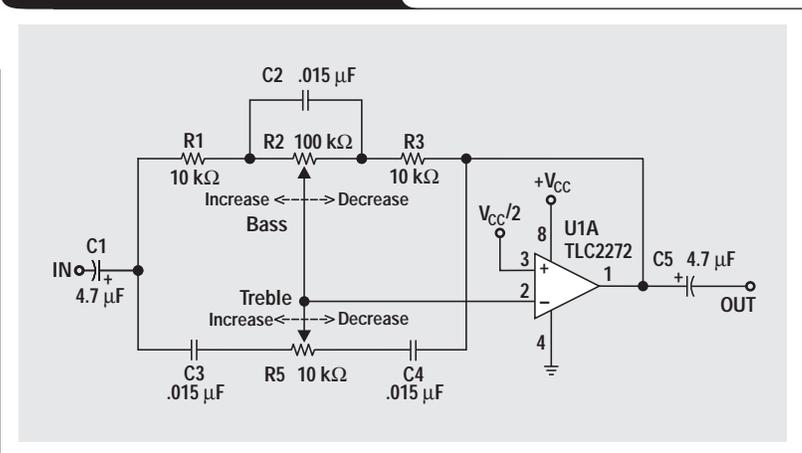### Figure 7. Combined filter response

Figure 8. Tone control circuit

Figure 9. Circuit response with pots at the extremes