Audio Applications for Op Amps, Part I

By Bruce Carter

Advanced Analog Products, Op amp Applications

Texas Instruments, Incorporated

This is the first of three articles in a series on audio circuits with the focus on new operational amplifiers that have excellent audio performance and can be used in high performance applications.

There have been many earlier collections of op amp audio circuits but all of them use split supplies. Many times, 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.

Split Supplies 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 a split-supplies or single-supply part. It does not mean, however, that the op amp has to be operated that way. 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. The power supply consists of a positive supply and an equal and opposite negative supply. The most common value is ±15 V, but ±12 V and ±5 V are also used. The input and output voltages are centered and referenced to 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.

5 V is a common value for a single supply, but voltage rails are lowering, with 3 V and lower 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 $V_{CC}/2$. 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. Fig. 1 shows how to generate a half supply reference if the designer insists on doing this with an op amp.
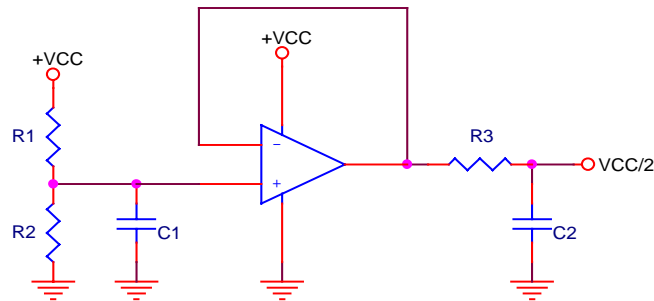
Fig. 1: Half-Supply Generator

R1 and R2 have 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 that follow, the virtual ground is labeled "VCC/2." This voltage comes from either the TLE 2426 rail-splitter or the circuit above. If the circuit above is used, the overall number of op amps in the design is increased by one.

Passive Components

The majority of the circuits given in these articles have been designed with standard capacitor values and 5% resistors. Capacitors should be good quality, 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.

Speech Filter

Human speech mostly occupies an audio spectrum of 300 Hz to 3 kHz. There is a requirement, especially in telephones, to limit the frequency response to this range. Frequently this function is performed with DSP chips, but they require an anti-aliasing filter to reject high frequency components. The anti-aliasing filter requires an op amp, and if there is 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.

Fig. 2 shows a single-supply phone speech filter using 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.
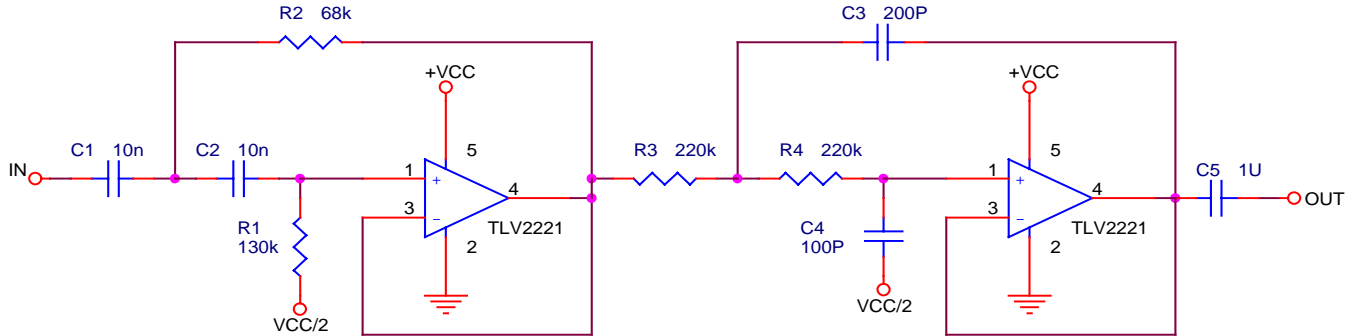

Fig. 2:Single-Supply Telephone Speech Filter

This circuit is only a second-order circuit and 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 (see Fig.3), although more complex, can be implemented in a single quad op amp (plus an external V_{CC} reference.)
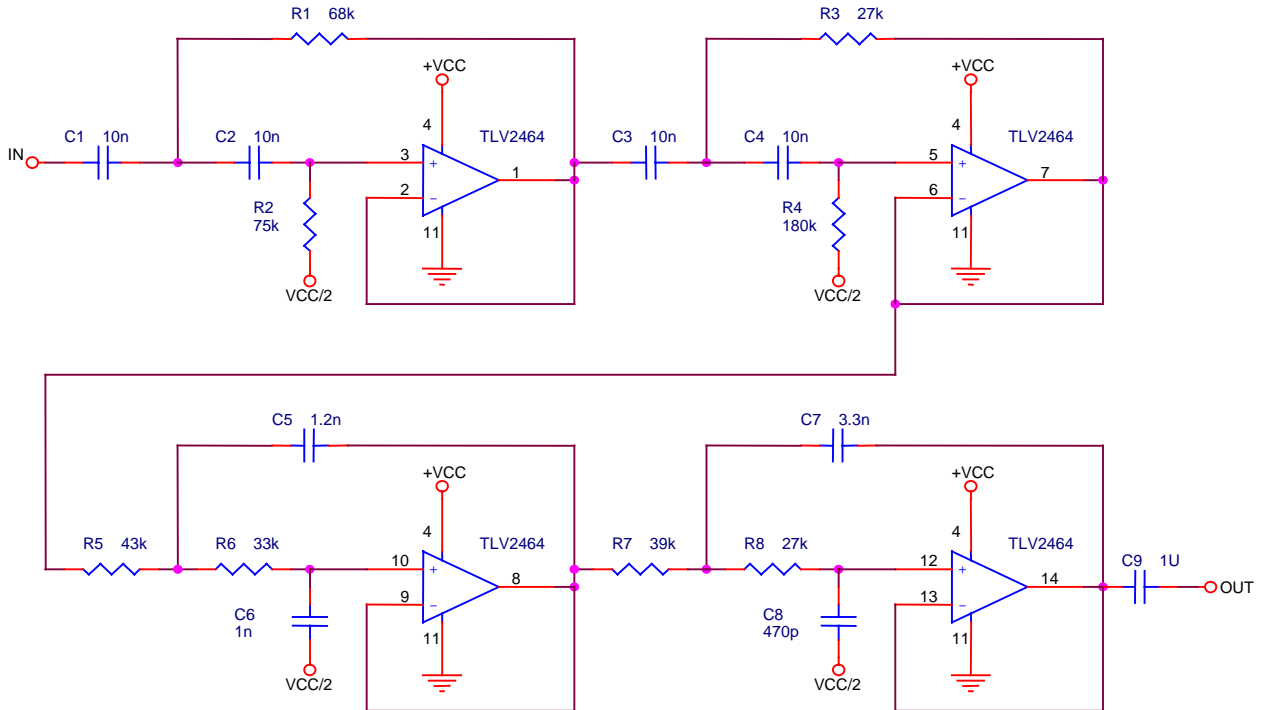
Fig.3: Fourth-Order Speech Filter
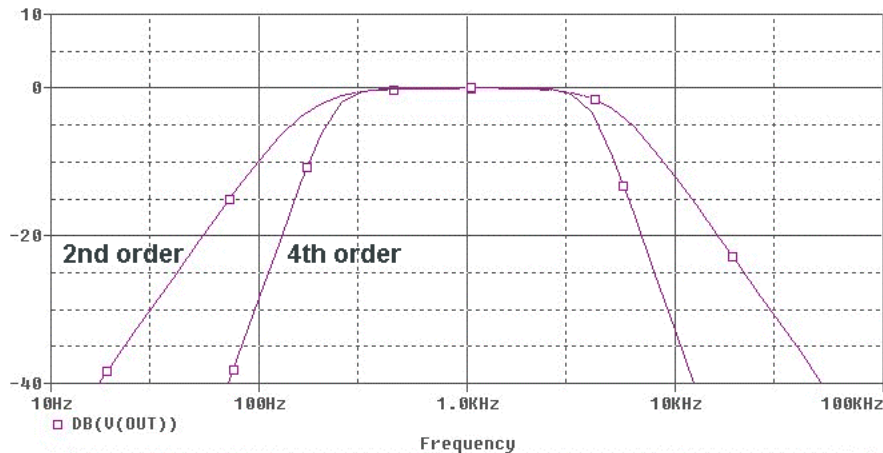
Fig.4: Second- And Fourth-Order Filter Responses

In the responses of the filters (see Fig. 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 direct 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, are large, heavy, and expensive; another disadvantage of such a network is that it is a first-order system. At high volume, destructive levels of audio can be passed to speakers not designed to handle a particular frequency range. If a speaker is incapable of moving in response to such stimulation, the only way the energy can be dissipated is in heat which can build up and burn out the voice coil.

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

The primary reason for bi- or multiple-amplification is that human hearing is not equally sensitive to all frequencies. The human ear is relatively insensitive at low and high frequencies. Yet audio amplifiers are designed to have flat response (constant power) across the audio band. The result is that listeners use tone controls or graphic equalizers to compensate for the human hearing curve and 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. Also their characteristics are linear and even if tone controls with more gain were devised, they would still 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 at 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 this 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. The best amplifier for midrange frequencies is a relatively low power, very low distortion class "A" amplifier. As little as 10 W 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 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 mid- 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 (see Fig. 5) routes low (bass) frequencies to a woofer, and midrange and high frequencies (treble) to the tweeter, a common application as many speaker cabinets contain those two.
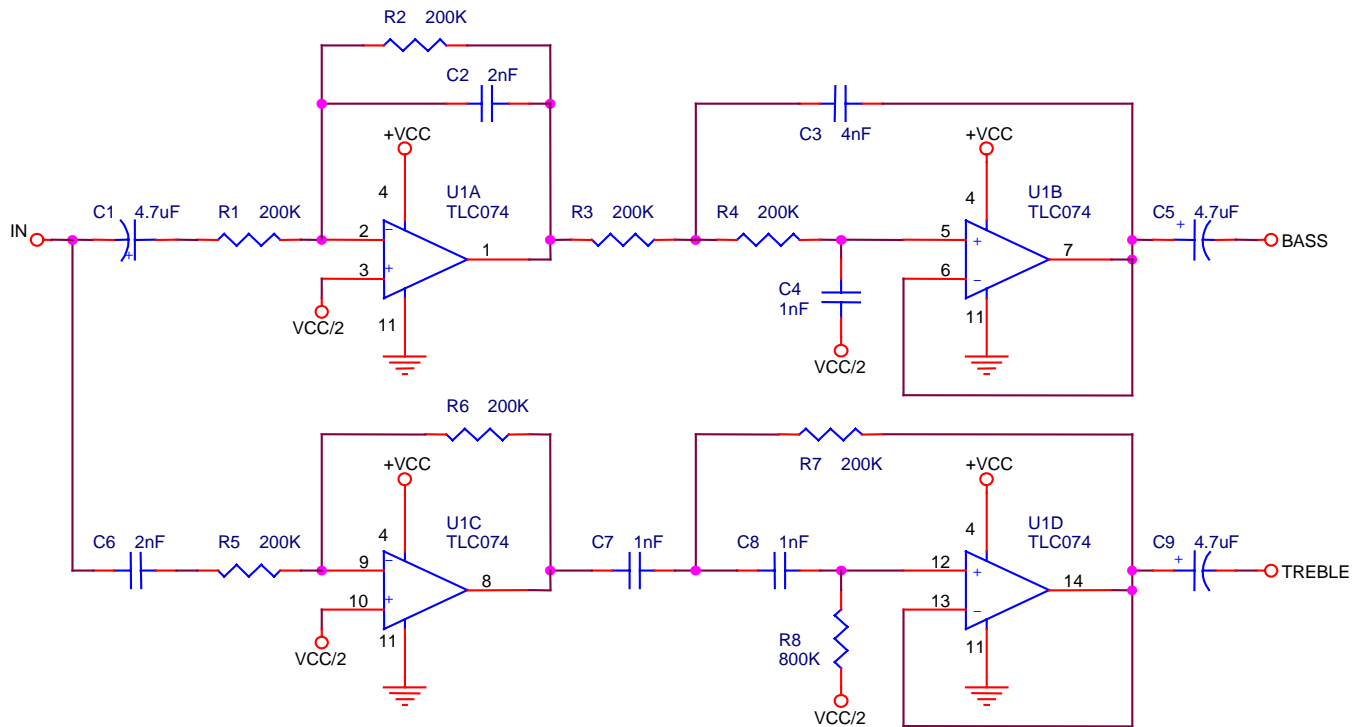
**Fig. 5: A Bass/Treble Crossover Implementation For Bi-Amplification**

The crossover frequency has been selected to be 400 Hz, 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 and the op amp sections can be interchanged, of course. There are only two values of capacitor, and one value of resistor! Three 4.7 µF electrolytic capacitors are used for coupling -- they are sufficiently large to ensure 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 be connecting four 200k resistors in series.

A subwoofer section (Fig. 6) can be added to the crossover network above to enhance subsonic frequencies.
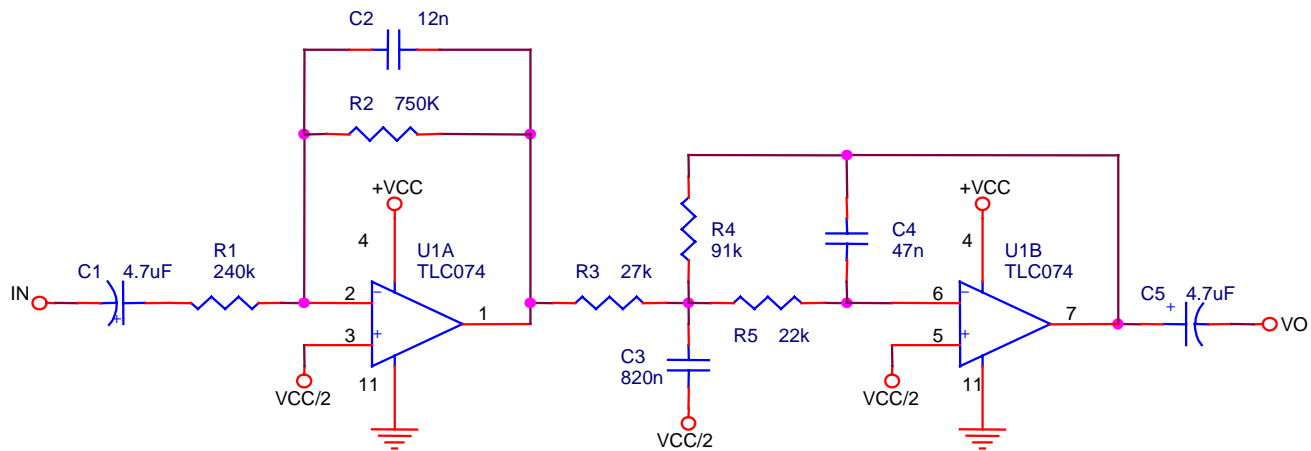
Fig.6: Subwoofer Section That Can Be Added To Fig. 5's Crossover

This circuit is a true subwoofer circuit which will not work with 6 or 8 inch "subwoofers." It is for 15 to 18 inch woofers in good infinite baffle, bass reflex, or folded horn enclosure, driven by an amplifier with at least a hundred 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 and even most audio CDs do not go below 20 Hz. This will prove most useful for home theater applications, which do have subsonic audio content. Examples are the dinosaur stomp in "Jurassic Park," or the "Earthquake."
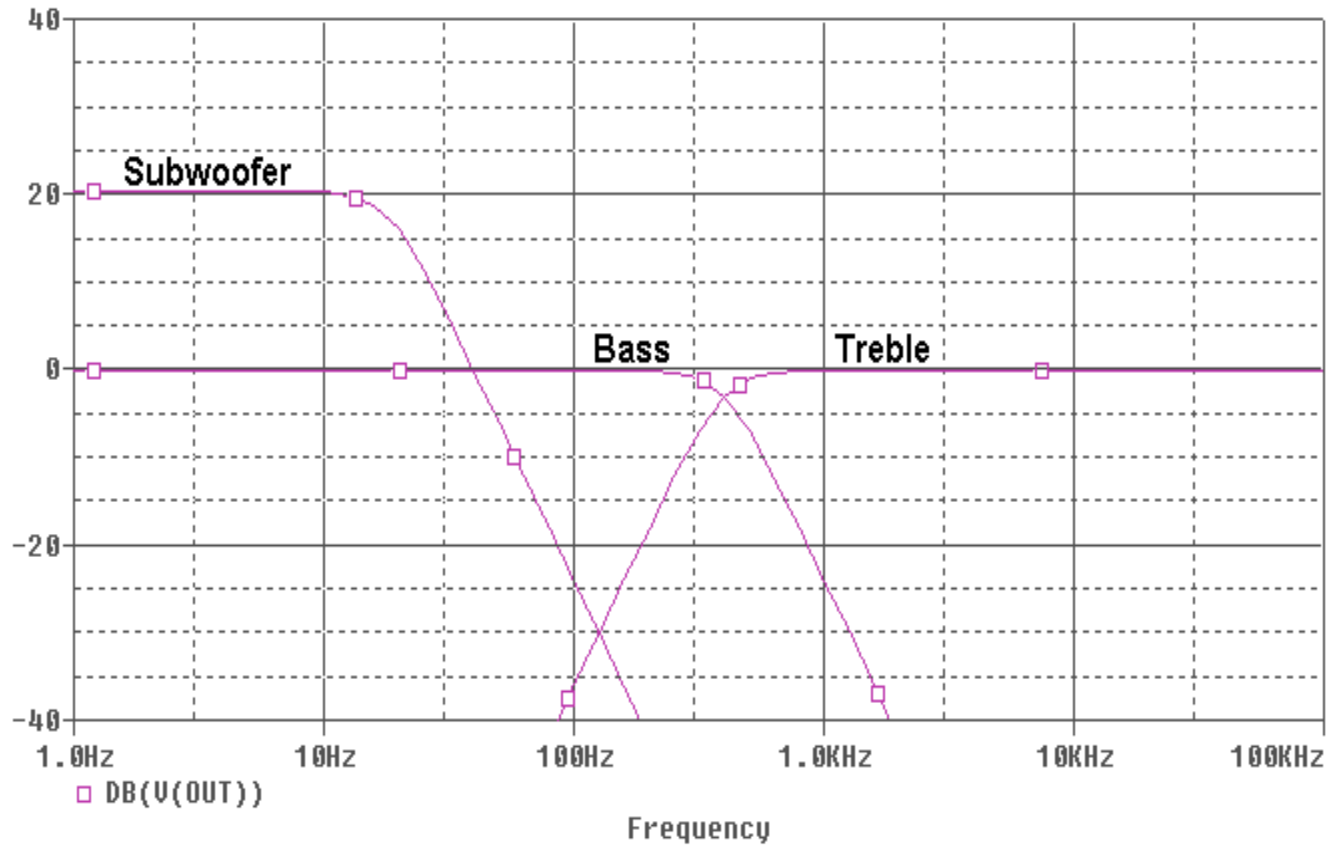
Fig.7: Combined Responses Of Crossover and Subwoofer Circuits

Fig. 7 shows the combined responses of the circuits in Figs. 5 and 6.

One active crossover network will be required per channel, with the exception of the subwoofer crossover as there is no stereo separation of low bass frequencies, and either channel (or both) can be used to drive the subwoofer circuit (sum the two channels into an inverting input with a second C1 and R1).
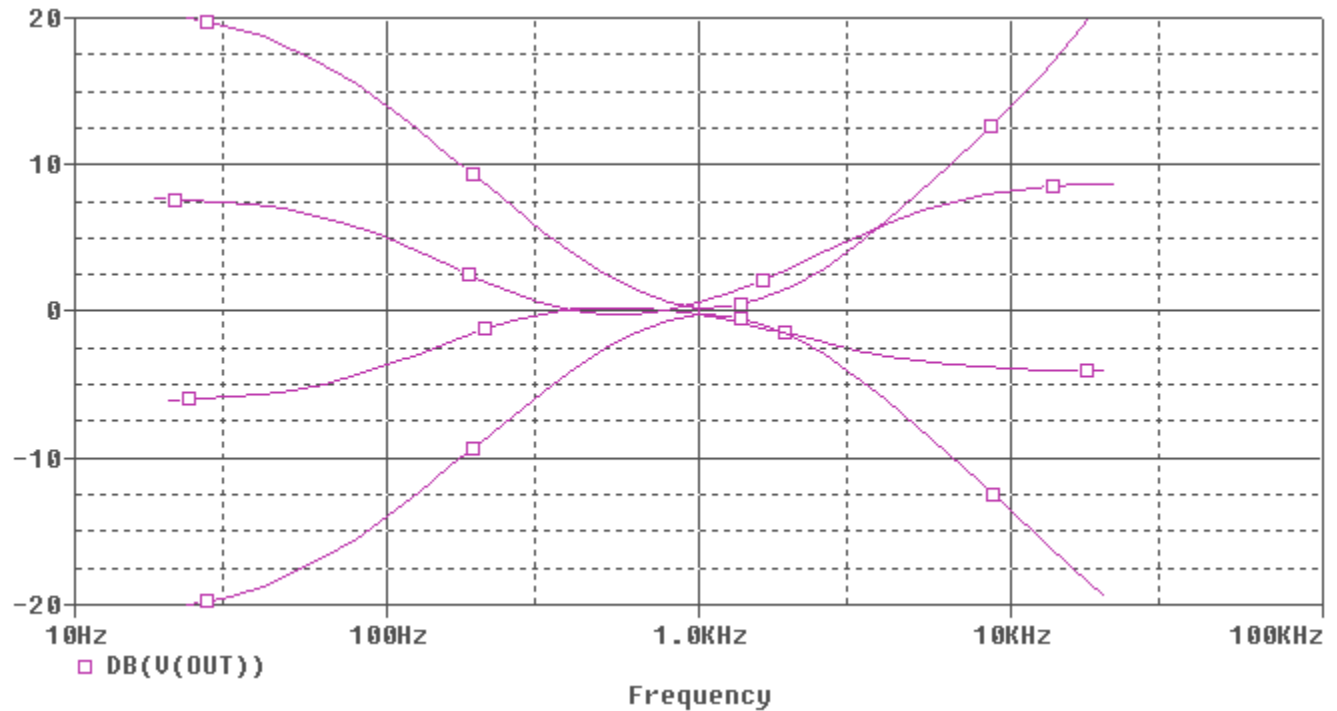
Tone Control

One rather unusual op amp circuit is the tone control circuit (see Fig.8.) It bears some superficial resemblance to the "Twin-T" circuit configuration, but it is actually a hybrid of a one-pole low-pass and high-pass circuit with gain and attenuation.
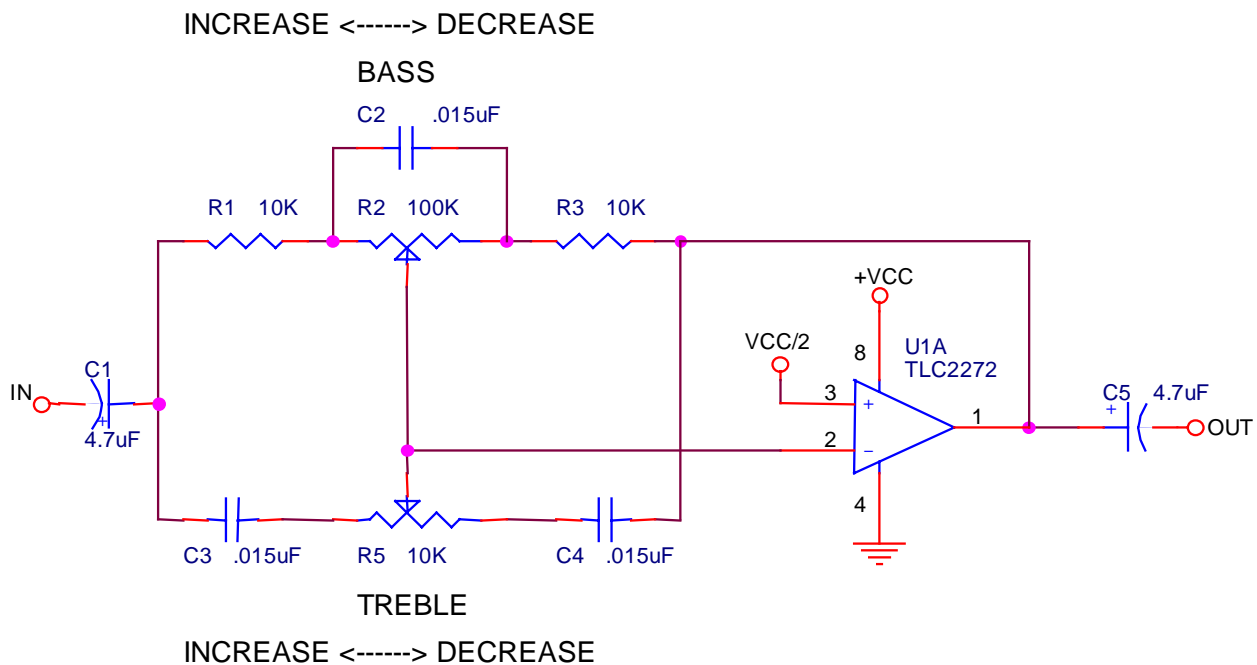
Fig. 8: Hybrid Tone Control Circuit

The mid range 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, seeking to limit cost. This circuit, unlike other similar circuits, uses linear pots instead of logarithmic. The two different potentiometer values are unavoidable, but the capacitors are the same value except for the coupling capacitor. The ideal value of capacitor 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 it is an oddball potentiometer value.

Fig. 9: Responses of Tone Control At Pot Extremes, 25% and 75%

The plots (Fig. 9) show the response of the circuit with the pots at the extremes, and at ¼ and ¾ positions. The mid position, although not shown, is flat to within a few thousandths of a dB. The compromises involved in cost reducing the circuit and using linear potentiometers lead to some slight non-linearities. The ¼ and ¾ positions are not exactly +10 and –10dB, meaning that the pots are most sensitive towards the end of their travel; this may actually 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 also 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.