lows that of **Figure 1**. Changing D_1 to a Schottky diode reduces the recovery-tail voltage that R_1 must dissipate. Add R_2 to draw a keep-alive current of 100 µA through Q_1 and Q_2, speeding turn-on. These keep-alive currents need not affect the timing. You can cancel out their effect with a slight reduction in the value of R_3. Fitting Q_2 and Q_4 with Schottky clamps D_2 and D_3, respectively, keeps the transistors out of saturation. These changes improve high-speed performance (**Figure 4**).

Although improved, the circuit still relies on D_1 for the final tail of recovery. To eliminate this problem, you can replace D_1 with a fourth transistor, Q_4 (**Figure 5**). Because transistors Q_1 and Q_2 are slightly conducting, a voltage one diode drop below that of supply V_1 is always present at their bases. You filter this voltage with R_5 and C_2 and provide it as a bias to the base of Q_4. This step keeps Q_4 nearer the threshold of conduction than would a diode to supply V_1. When source V_2 changes to a negative state, Q_4 is fully off and draws no current. When V_2 changes to a positive state, the emitter of Q_4 conducts at voltages above V_1 to catch the recovery transition, further reducing the recovery-tail amplitude.

BECAUSE Q_1 AND Q_2 ARE SLIGHTLY CONDUCTING, A VOLTAGE ONE DIODE DROP BELOW THAT OF SUPPLY V_1 IS ALWAYS PRESENT AT THEIR BASES.

R_6 may be used to limit Q_4's base current, but its omission is acceptable if source V_2 has sufficient output resistance. It may be destructive to apply source V_2 swings large enough to cause excess reverse voltage across the Q_4 base-emitter junction. Q_3 and Q_4 can share the same package. These additions further improve the pulse generator's high-speed performance (**Figure 6**). EDN

Implement an audio-frequency tilt-equalizer filter

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In the 1970s, Quad Ltd developed a "tilt" audio-tone control, which first appeared on the company's model 34 preamplifier. The tilt control tilts the frequency content of the audio signal by simultaneously boosting the treble and cutting the bass frequencies, or vice versa (**Figure 1**). Only one knob is needed to tilt the frequency response around a pivot frequency, F_P (**Figure 2**).

Quad Ltd never published a transfer function for the filter. You need a Spice simulation and many trial-and-error cycles to tune it to your desired response. By deriving the transfer function, you can easily select the component values. Surprisingly, the transfer function also shows how you can make the tilt response asymmetric, with different amounts of boost and cut. You begin deriving the transfer function by expressing the input versus the output as a function of dc-feedback resistor, R_F, and Z, the complex impedance of the RC branches:

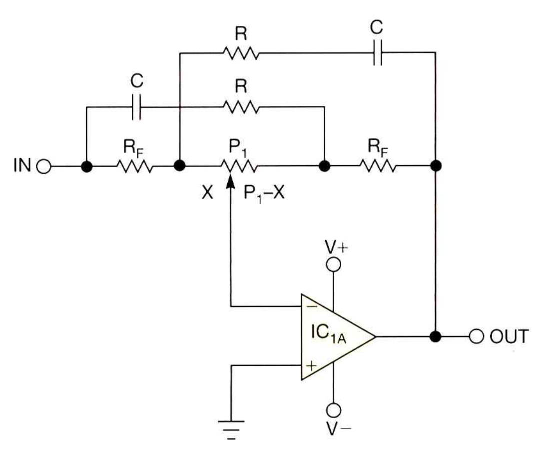

Figure 1 In a tilt audio-tone control, the tilt control tilts the frequency content of the audio signal by simultaneously boosting the treble and cutting the bass frequencies, or vice versa.

$$\frac{V_O}{V_I}=\frac{X\times(Z-R_F)-Z\times(P_1+R_F)}{X\times(Z-R_F)+R_F\times(Z+P_1)},$$

where X indicates the wiper position of potentiometer P_1 and the values of the resistors and capacitors define Z:

$$Z=R+\frac{1}{i\times2\times\pi\times F\times C}.$$

The frequency response in **Figure 2** is for the extreme wiper positions, where X=0 or P_1. All of the other responses, with 0 less than X and X less than P_1, lie between those curves. To get the frequency responses in decibels, multiply the log of the absolute value of the transfer function by 20: $20\log(|T_F|)$. To get a log/log scale on the graph, substitute 10^F for F on the X axis. Pivot frequency F_P depends on component value, including the setting of potentiometer P_1, as it sweeps between an X value of 0 and P_1, where R_F must be greater than R:

$$F_P=\frac{\sqrt{(P_1+2\times R_F)}}{2\times\pi\times C\times\sqrt{(R_F-R)}\times\sqrt{(P_1\times(R+R_F)+2\times R\times R_F)}}.$$

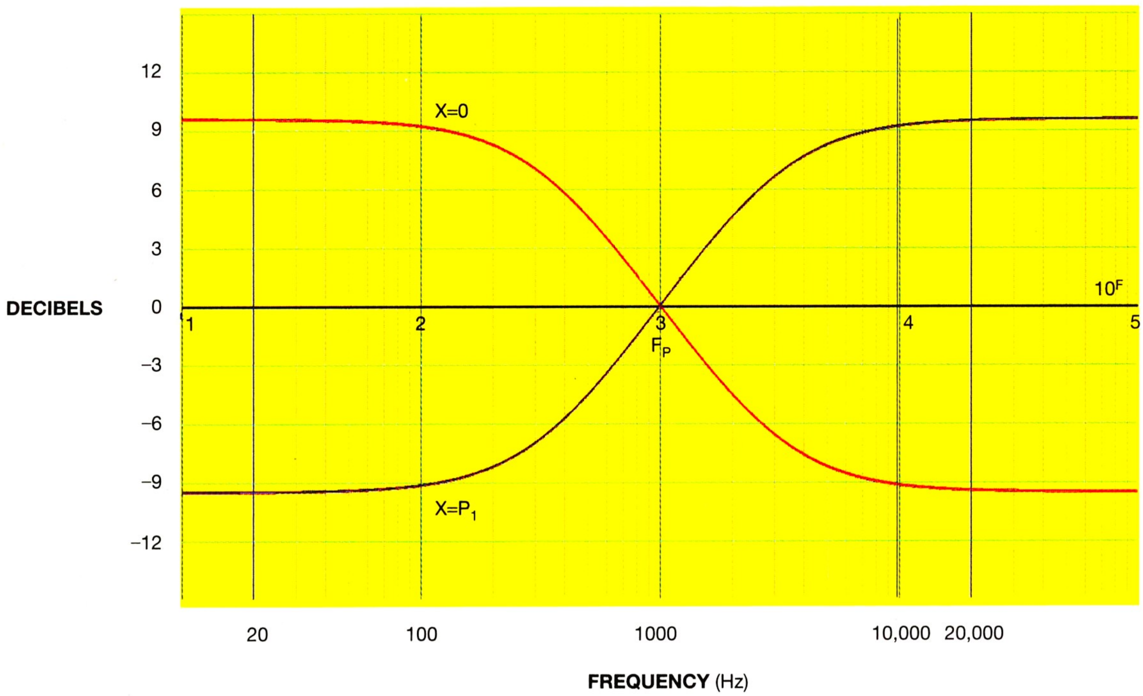

Figure 2 This frequency response is for the extreme wiper positions, where X=0 or P_1. All of the other responses, with 0 less than X and X less than P_1, lie between these curves.

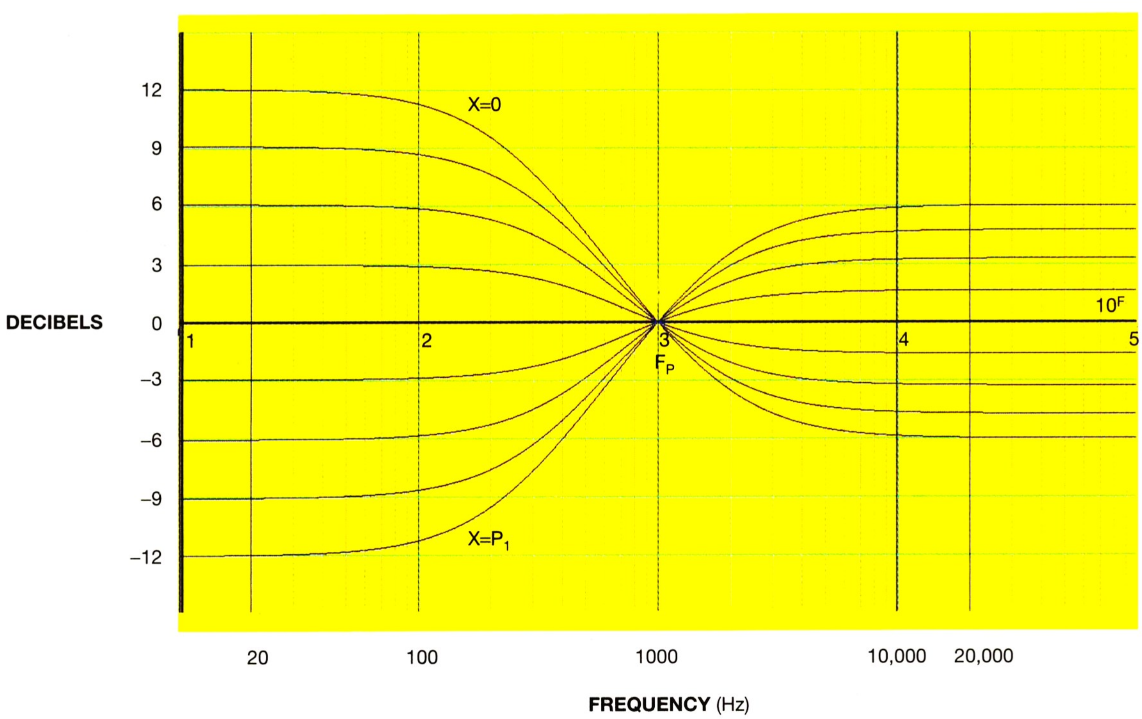

Figure 3 R_F is 16.66 kΩ, R is 7.14 kΩ, and C is 12.24 nF.

To calculate component values, you first define the maximum low-boost asymptote as M_L, when the frequency goes to 0 Hz and the potentiometer's value is also 0Ω. You then define the maximum high-boost asymptote as M_H, when the input frequency goes to infinity, and set the potentiometer to its maximum value. This step gives the component values for R_F, R, and C:

$$R_F=\frac{P_1}{M_L-1};$$

$$R=\frac{P_1}{M_H\times M_L-1};$$

$$C=\left\{\left[(M_L-1)\times\sqrt{(M_L+1)}\times(M_H\times M_L-1)^{3/2}\right]\times\sqrt{\left(\frac{M_H-1}{(M_L-1)\times(M_H\times M_L-1)}\right)}\right\}\Big/$$

$$2\times\pi\times M_L\times P_1\times F_P\times(M_H-1)\times\sqrt{(M_H+1)}.$$

For the **equations** to work, M_L-1 and $(M_H\times M_L-1)$ must be greater than 0. You can choose any reasonable value of potentiometer P_1. For example, select a P_1 value of 50 kΩ, a desired pivot frequency of 1 kHz, a maximum low-frequency boost of 4, and a maximum high-frequency boost of 2. The **equations** yield an R_F of 16.66 kΩ, an R of 7.14 kΩ, and a C of 12.24 nF (**Figure 3**).

You take 20 times the log of M_L to get the response in decibels, so an M_L of 4 is the 12-dB maximum low-frequency boost, and an M_H of 2 represents the 6-dB maximum high-frequency boost. When you normalize the resistor and capacitor values to standard values, you get only a minor error in your desired response. By defining the variables M_L and M_H, you can make tilt equalizers that have an asymmetric response between boost and attenuation.

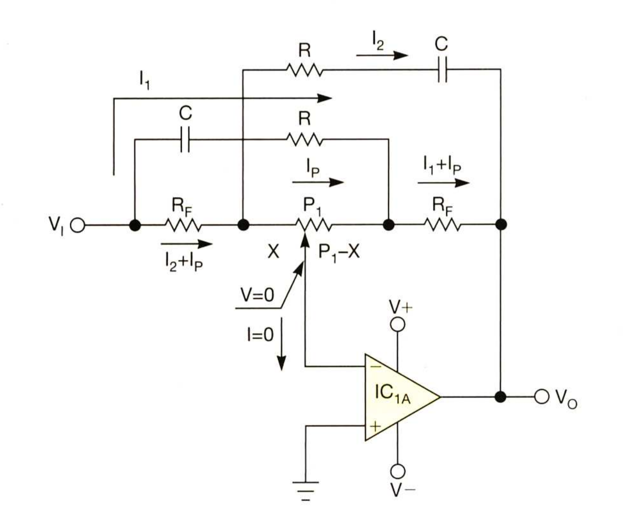

Figure 4 Voltages V_I, V_O, and V are all referred to ground.

THE GOAL IS TO FIND V_O/V_I; YOU NEED NOT SOLVE ALL OF THE UNKNOWNS.

A detailed derivation of the transfer function is included here. You begin by defining voltages V_I, V_O, and V, all referred to ground (**Figure 4**). In this case, I_1, I_2, and I_P are the minimal number of unknown currents. Because an op amp servos the output to keep the input pins at the same voltage, the potentiometer wiper is at 0V, a virtual ground. Further assume the infinite input impedance of the op-amp input pins so that the current at the inverting pin is 0A. V_I and V_O are unknown, letting you write a set of **equations** for the conditions:

$$V_I=I_1Z+(I_1+I_P)R_F+V_O$$
[Loop V_I→Z (input)→R_F (output)→V_O];
$$V_I=(I_2+I_P)R_F+I_2Z+V_O$$
[Loop V_I→R_F (input)→Z (output)→V_O];
$$V_I=(I_2+I_P)R_F+I_PX+0$$
[Loop V_I→R_F (input)→X(P_1 wiper)→virtual ground];
$$0=I_P(P_1-X)+(I_P+I_1)R_F+V_O$$
[Loop virtual ground→P_1–X(P_1 wiper)→R_F (output) →V_O];
$$V_I=(I_2+I_P)R_F+I_PP_1+(I_1+I_P)R_F+V_O$$
[Loop V_I→RF (input)→P_1→R_F (output)→V_O].

Remember that Zis the complex impedance of the RC branches. Now rearrange the **equations**:

$$V_I=I_1(R_F+Z)+I_PR_F+V_O;$$
$$V_I=I_2(R_F+Z)+I_PR_F+V_O;$$
$$V_I=I_2R_F+I_P(X+R_F);$$
$$V_O=I_PX-I_1R_F-I_P(P_1+R_F);$$
$$V_I=I_1R_F+I_2R_F+I_P(P_1+2R_F)+V_O.$$

From the first and second **equations** you can deduce that I_1 equals I_2. You can now substitute into the last three **equations** and rearrange them to get the final set:

$$V_I=I_1(R_F+Z)+I_PR_F+V_O;$$
$$V_I=2I_1R_F+I_P(P_1+2R_F)V_O;$$
$$I_1=(I_P(X-P_1-R_F)-V_O)/R_F;$$
$$V_I=2I_1R_F+I_P(P_1+2R_F)+V_O.$$

The goal is to find V_O/V_I; you need not solve all of the unknowns. If you substitute I_1 from the third **equation** above into the second **equation**, you can find I_P. You then substitute this I_P into the fourth **equation** and find the ratio of V_O/V_I, yielding the first **equation** in this Design Idea. This result is congruent with the actual numerical value of the examples in **Reference 1**. EDN

REFERENCE

1 Moy, Chu, "Designing a Pocket Equalizer for Headphone Listening," *HeadWize*, 2002, http://bit.ly/vveL7z.