

A LOW NOISE, LOW DISTORTION DESIGN FOR ANTIALIASING AND ANTI-IMAGING FILTERS

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Many customers have requested more information about the analog low-pass filters that appear in many of our PCM audio data sheets. They are used for antialiasing in front of ADCs or for smoothing on the output of DACs. The following bulletin is an excellent primer on the subject. —Ed.

In any digitizing system, antialiasing and anti-imaging filters are used to prevent the signal frequencies from “folding back” around the sample frequency and causing false (or alias) signals from appearing in the signal we are attempting to digitize. Very often, these filters must be very complex, high order analog filters in order to do their job effectively.

As sampling rates of converter systems have increased, however, oversampling may be used to reduce the filters’ stopband attenuation requirements⁽¹⁾⁽²⁾. In digital audio systems, 4x oversampling may be used, and it can be shown⁽³⁾ that for an antialiasing filter (which precedes the ADC), a simple sixth order filter may be used. For the output side, after the DAC, a simple third order filter may be used. Realizing these filters in a way that maintains extremely low noise and low distortion then becomes a challenge.

Compact disk player manufacturers began using a filter topology that was described many years ago—the Generalized Immittance Converter (GIC)⁽⁴⁾. This topology allows one to easily realize active filters beginning from a passive filter design. In addition, the GIC filter provides extremely low distortion and noise, at a reasonable cost. Compared with more familiar feedback filter techniques, such as Sallen & Key filter topologies, the GIC filter can be shown to have superior noise gain characteristics, making it particularly suitable for audio and DSP type applications⁽⁵⁾.

We use this type of filter on our demonstration fixtures for the PCM1750 and PCM1700, dual 18-bit ADC and DAC, respectively. When sending out schematics of these demonstration fixtures, very often the first question is, “What are those filters anyway?” Well, they’re GIC filters, and here’s how you design them and how they perform. Stepping through this design process will allow you to modify these designs for a different cutoff frequency for your particular application. A more detailed treatment of the theory behind these filters may be found in Huelsman and Allen⁽⁶⁾.

As stated above, for oversampling digital audio applications, third and sixth order filters are adequate. Thus, we may design our first GIC filter by designing a third order filter. The filter characteristic most desirable for sensitive DSP type applications is linear-phase. The linear-phase filter is sometimes called a Bessel (or Thomson) filter. The linear-phase filter has constant group delay. This means that the phase of the filter changes linearly with frequency, or that

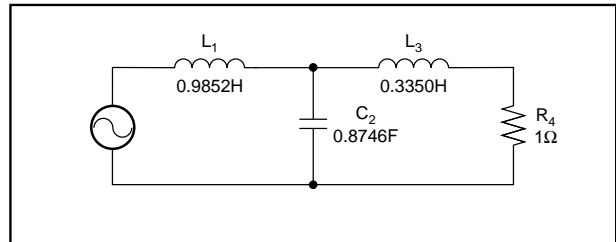


FIGURE 1. Passive Third Order, Linear-Phase, Low-Pass Filter Prototype.

the group delay is constant. These filters maintain phase information for sensitive DSP applications such as correlation, and preserve transient response. These characteristics are critical in audio applications as well, because they affect sound quality greatly.

Thus, we begin the design process by selecting a passive, third order linear-phase filter design that will be realized using this active approach. The passive design shown in Figure 1 is neither a Butterworth nor a Bessel response; it is something in between. The component values for this particular response, optimized for phase linearity and stopband attenuation, were found through exhaustive computer simulations and empirical analysis. Component values for standard Butterworth and Bessel responses may be found in standard filter tables, such as those available in Huelsman and Allen⁽⁷⁾. This circuit is then transformed to an active circuit by multiplying all circuit values by $1/s$, which changes all inductors to resistors, all resistors to capacitors, and all capacitors to Frequency Dependent Negative Resistors (FDNRs). These FDNRs have the characteristic impedance of

$$\frac{1}{s^2C}$$

and may be realized using the GIC circuit. Thus, L_1 becomes R_1 , C_2 becomes $1/s^2C_2$, L_3 becomes R_3 , and the terminating

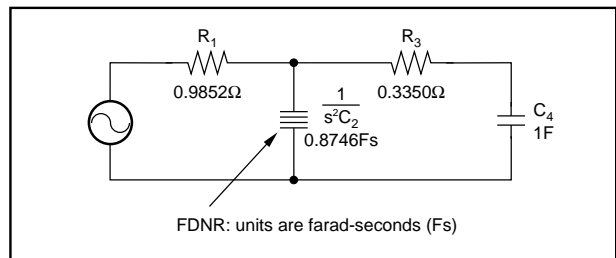


FIGURE 2. Filter of Figure 1 Transformed by Multiplying All Component Values by $1/s$.

resistor R_4 becomes C_4 , as shown in Figure 2.

The FDNR is then realized by the GIC circuit shown in Figure 3. The value of the FDNR is determined by

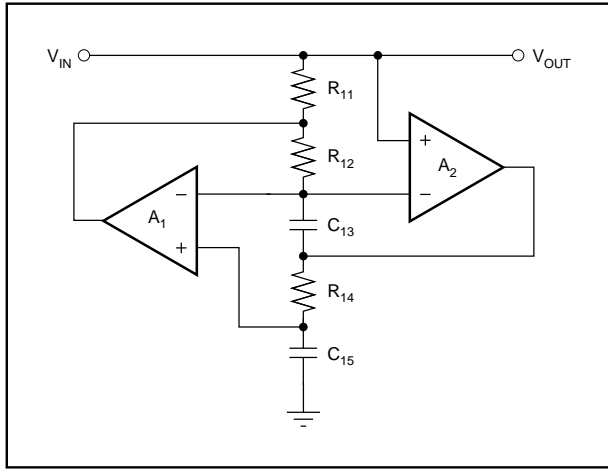


FIGURE 3. Frequency Dependent Negative Resistor (FDNR) Realized Using Generalized Immittance Converter (GIC).

$$D = (R_{12} \cdot R_{14} \cdot C_{13} \cdot C_{15})/R_{11}$$

Thus by setting $R_{11} = R_{12} = 1$ and $C_{13} = C_{15} = 1$, D is entirely determined by the value of R_{14} . For the FDNR of Figure 2, $R_{14} = 0.8746\Omega$.

The entire third order filter circuit is shown in Figure 4. This circuit now must be scaled in frequency to give the desired cutoff frequency, and then must be scaled in impedance to allow for the use of reasonable sized component values.

The filter circuits found in filter tables, such as that in Figure 1 and the active realization of this passive circuit (Figures 2 and 4), are designed for a cutoff frequency of $\omega = 1$ rad/s. To make the filter have the cutoff frequency we desire, we must scale it in frequency by the scaling factor

$$\Omega_N = 2\pi f_c$$

This scaling factor is applied to all frequency-determining components—capacitors in this case. The example filter will be designed for audio, so we might consider a cutoff frequency of 20kHz. However, linear-phase filters tend to roll-off very slowly, causing 1-2dB attenuation before the cutoff frequency; generally audio systems prefer to have their frequency response out to 20kHz to be within 0.1dB. The example filter then will have a cutoff frequency of 40kHz, commonly used in many of today's CD players. All capacitor values are divided by the frequency scaling factor, so

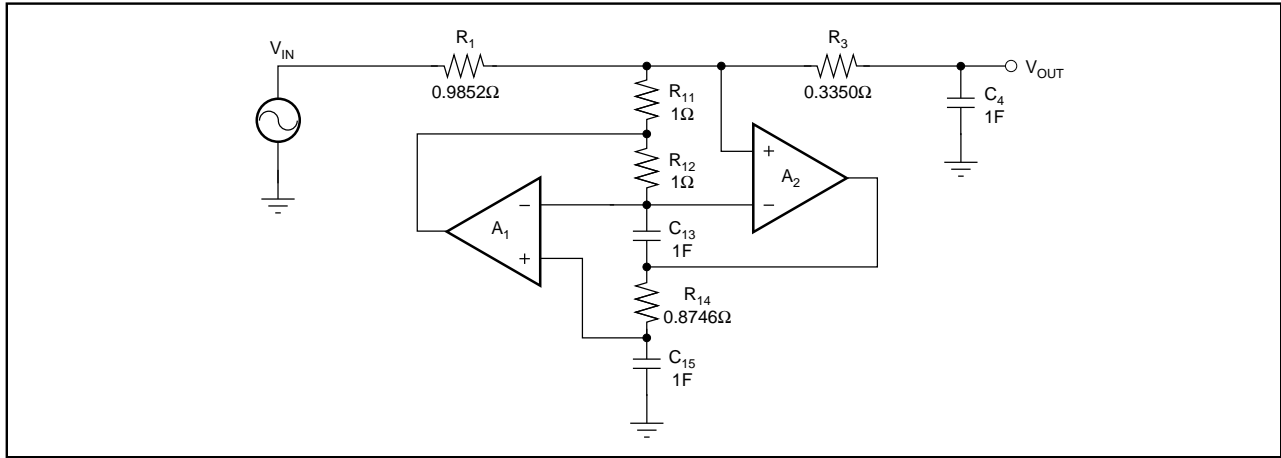


FIGURE 4. Third Order, Linear-Phase Realization of Circuit Shown in Figure 2.

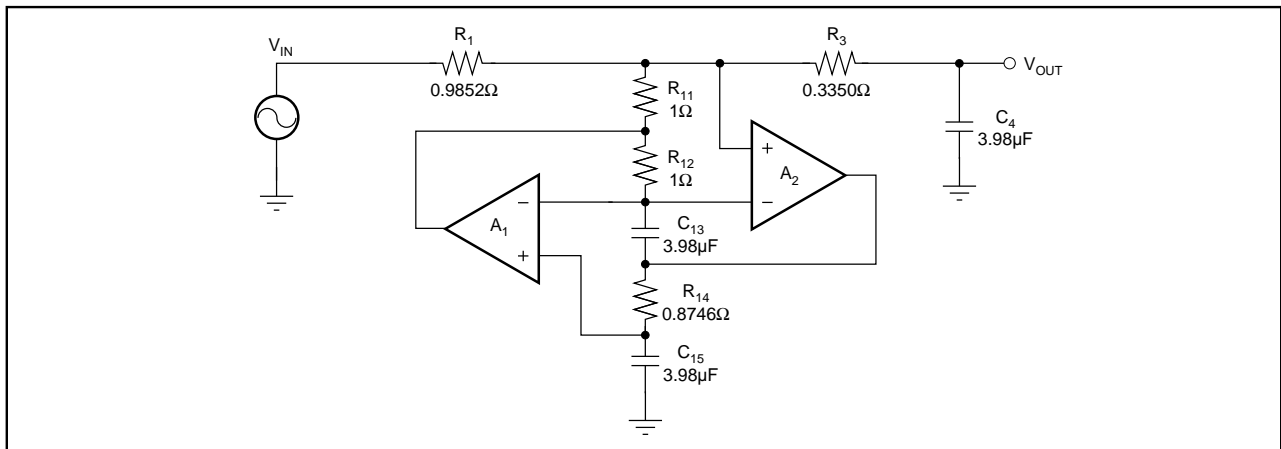


FIGURE 5. Circuit of Figure 4 Scaled to a 40kHz Cutoff Frequency.

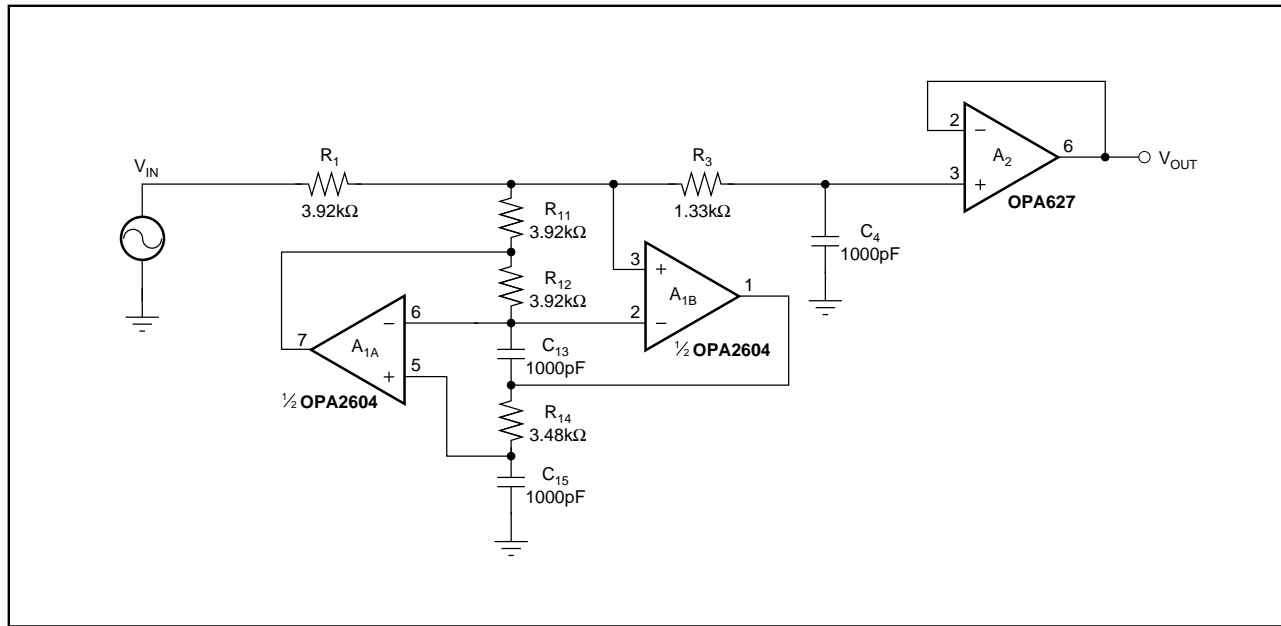


FIGURE 6. Circuit of Figure 5 Scaled in Impedance (note use of buffer amplifier to reduce output impedance of the filter).

$$C_{13} = C_{15} = C_4 = 3.98\mu\text{F}.$$

The filter (Figure 5) could now be built, but the large capacitor values and low resistance values could pose practical problems. To alleviate this, the impedances of the circuit are scaled by an impedance scale factor:

$$Z_n = \frac{\text{Present C value}}{\text{Desired C value}}$$

By choosing the desired C value as 1000pF, $Z_n = 3.97 \times 10^3$. This impedance scaling factor then is multiplied by all resistor values to find the new resistor values, and divides all the capacitor values, taking them from the present values to the desired capacitance.

The final filter design is shown in Figure 6. Since the output impedance of this filter is relatively high, it's a good idea to buffer the output using an op amp voltage follower. Amplitude and phase response of this filter is shown in Figure 7a. Figure 7b is a closer look at the amplitude response in the passband—the frequency response is flat well within 0.1dB out to 20kHz.

Figure 7c is a plot of the frequency response of the filter (solid line) and the filter's deviation from linear phase (dotted line). Note the phase scale; the phase response is well within 0.1° of linear phase in the 1kHz-20kHz region, where the ear is most sensitive to phase distortion.

Figure 7d is a plot of the total harmonic distortion plus noise (THD + N) of this filter versus frequency. At about -108dB , this would be suitable for digital systems with true 18-bit converter performance!

To make a sixth order filter, you can repeat the design process above from a passive realization and directly implement a filter. This implementation is very sensitive to the gain-bandwidth product (GBW) match of all of the op amps used, however; for a 40kHz cutoff frequency, an op amp

with extremely high GBW would be required. An example of a sixth order, 40kHz Butterworth filter realized in this fashion is shown in Figure 8, but its frequency response (Figure 9) is less than hoped for due to the GBW limitations described above.

A simpler solution is to cascade two of the third order sections designed above. This cascaded design (Figure 10) works equally well for most applications.

Figure 11 (a-d) shows the performance of this cascaded filter design. Note that the phase linearity and THD + N are still excellent using this approach.

REFERENCES

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- (6) L.P. Huelsman, P.E. Allen, *Introduction to the Theory and Design of Active Filters*, McGraw-Hill, New York, 1980.
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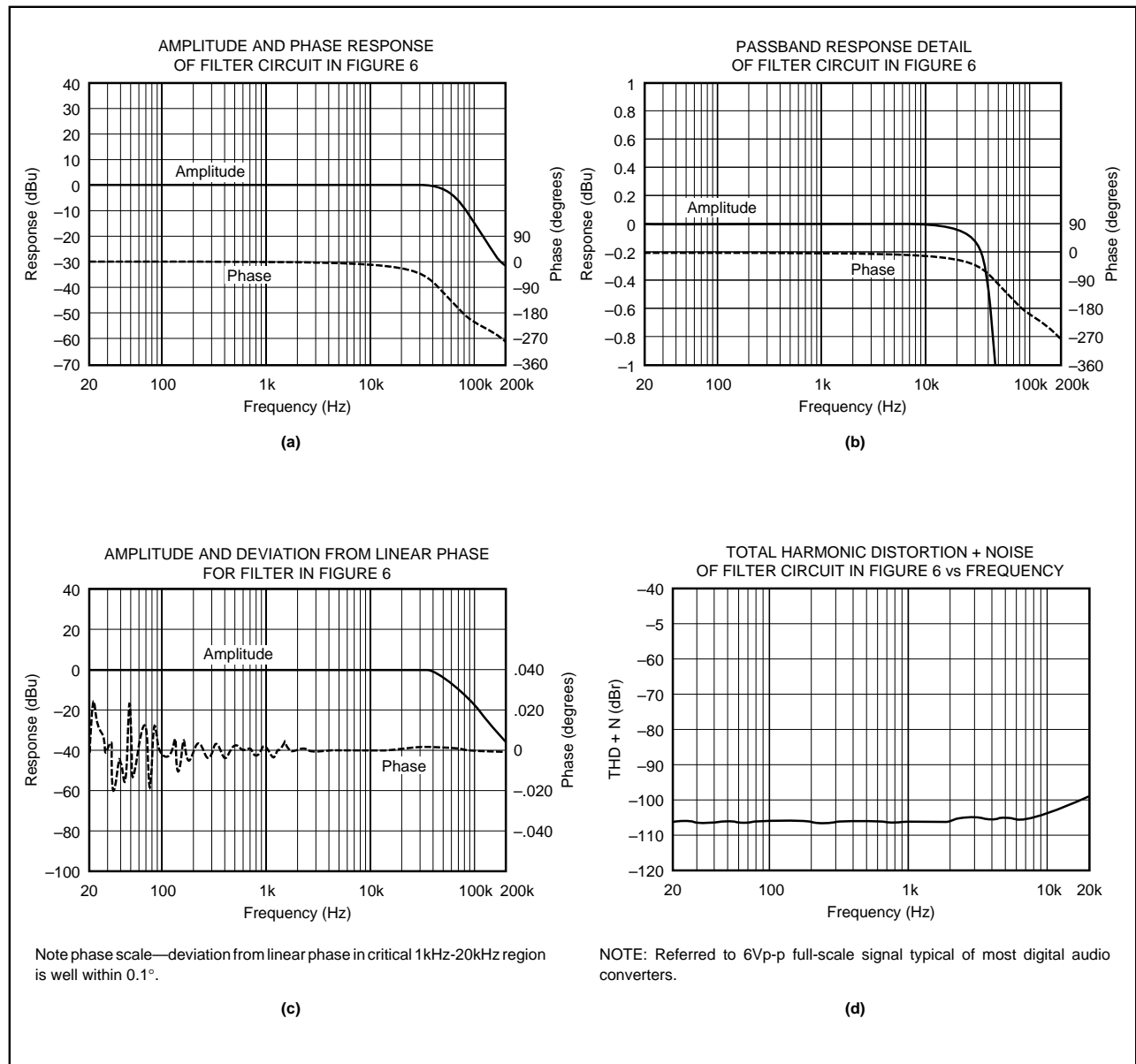


FIGURE 7. Performance Details of Figure 6 Circuit.

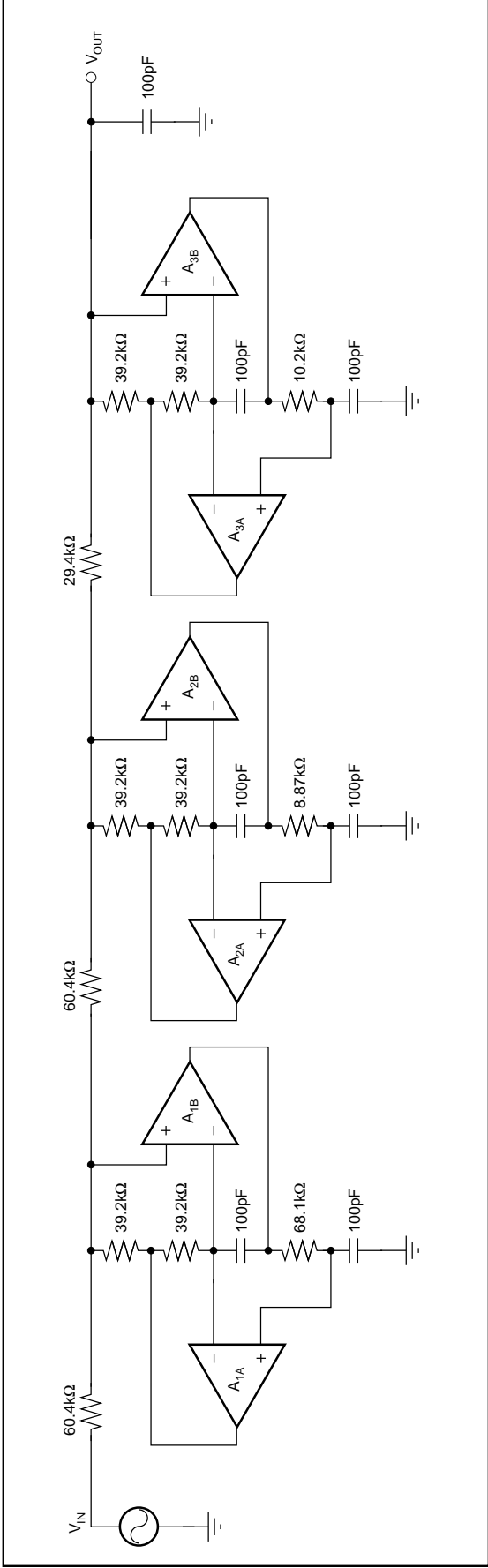


FIGURE 8. Sixth Order Butterworth Filter Realized by Method Outlined in Text (actual circuit would require output buffer amplifier to lower output impedance).

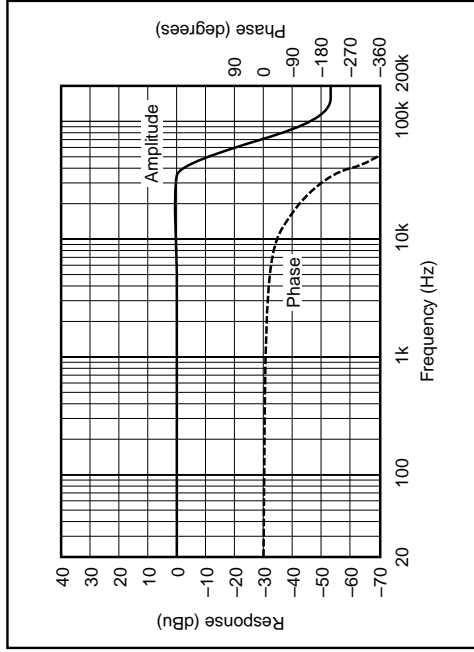


FIGURE 9. Amplitude (solid line) and Phase (dotted line) Response of Filter Circuit in Figure 8. (Note flattening of stopband response near 150kHz due to inadequate GBW of operational amplifiers used.)

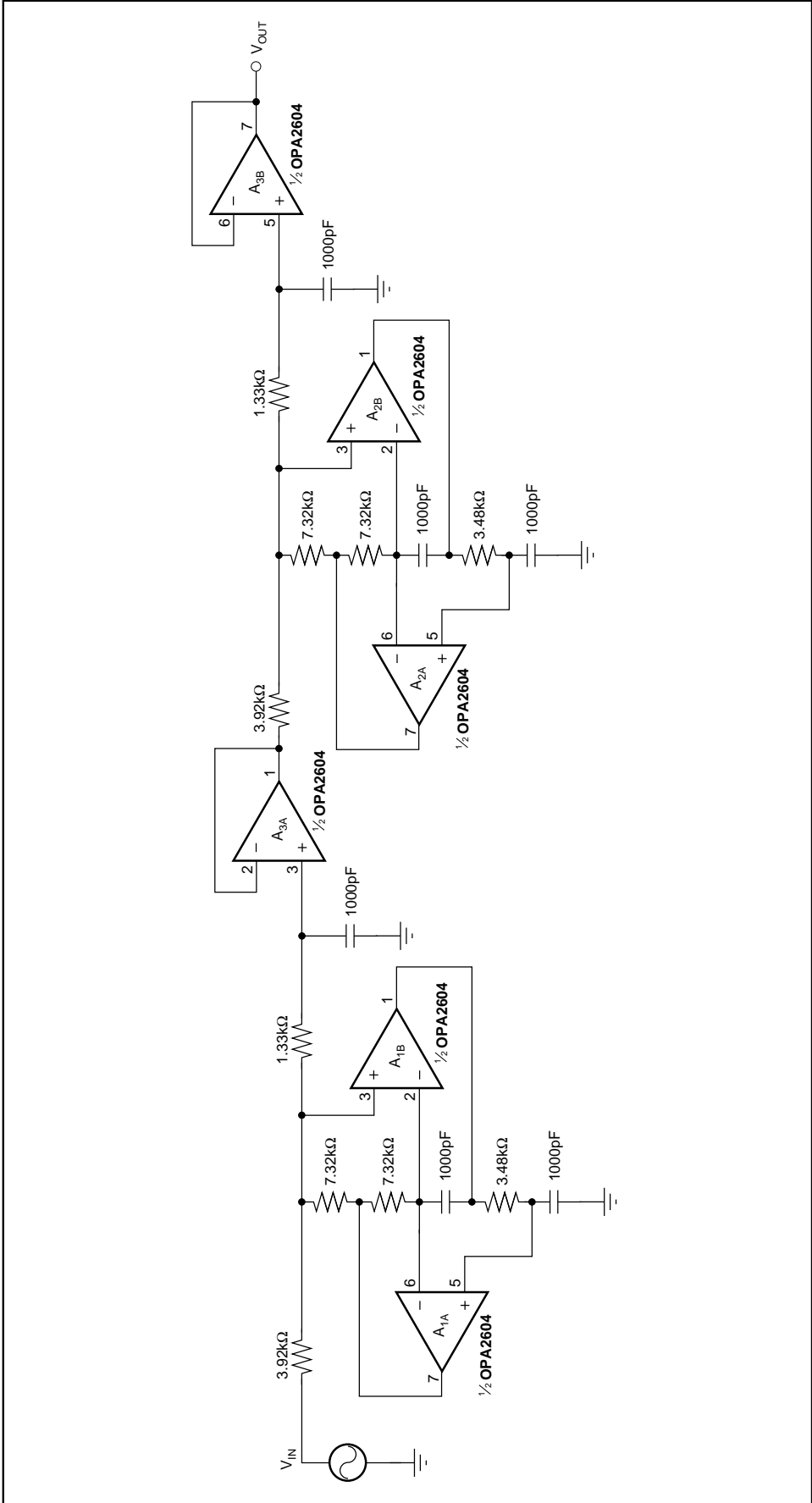


FIGURE 10. Sixth Order Linear-Phase Filter Made by Cascading Two Third Order Filters.

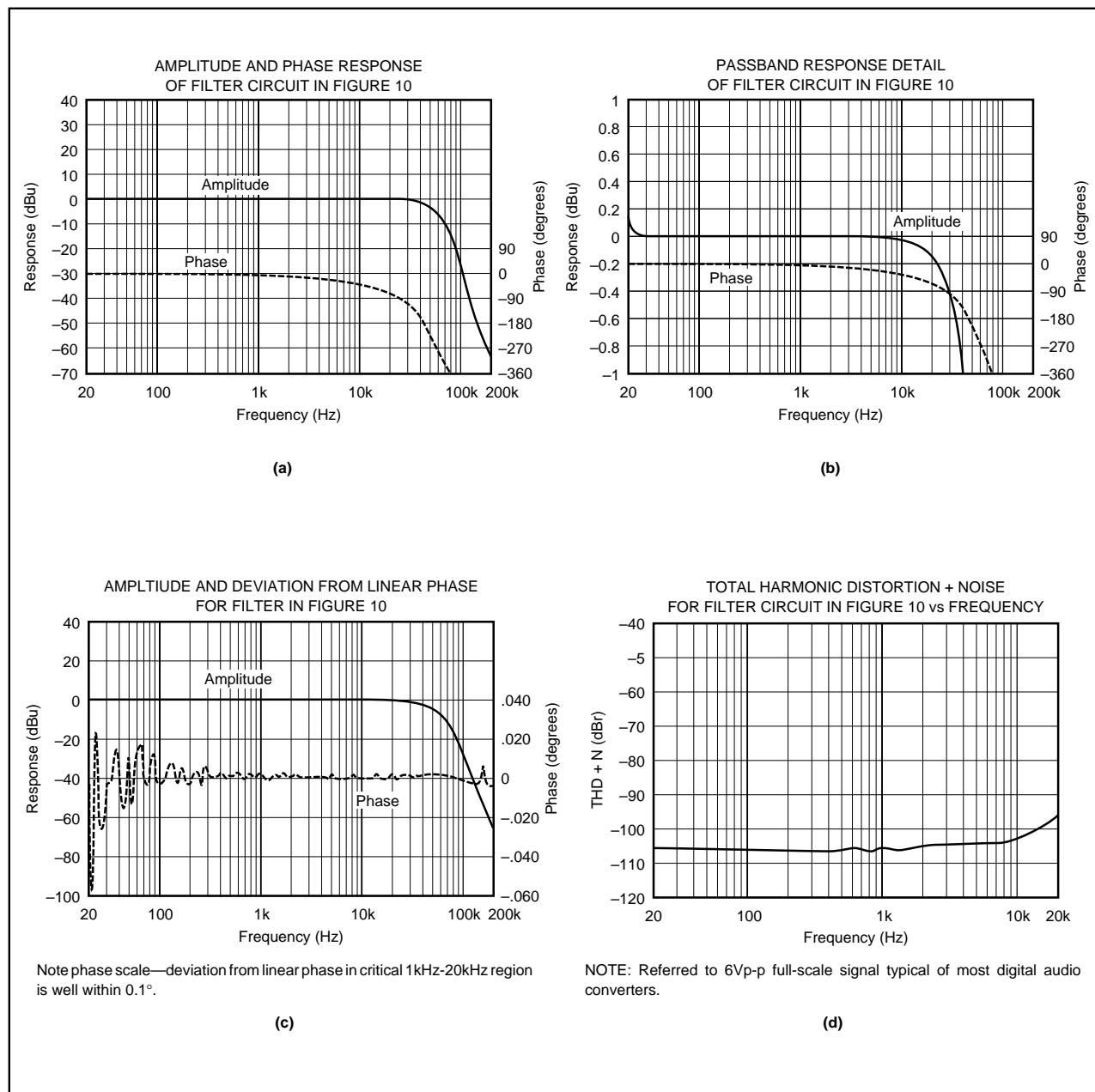


FIGURE 11. Performance Details of Figure 10 Circuit.

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