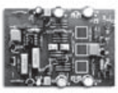


Back

A Flexible Subwoofer Amplifier, Pt.1

This four-part series explains how to build a full-featured subwoofer amp that will enhance your audiophile experience. Part 1 focuses on theoretical aspects.

By Rudy Godmaire

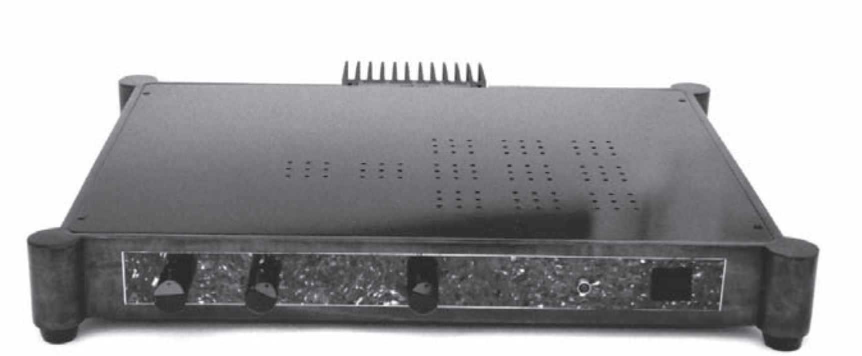


PHOTO 1: The subwoofer amplifier.

This journey started four years ago when I needed a new amplifier to power my passive subwoofer. At first, I intended to integrate an amplifier module to the three-channel home theater amplifier I was building at that time. Though the idea of a 3.1 channel amplifier was seductive, my chassis was lacking space to accommodate this fourth channel. Building a separate unit then became the only option.

At the same time, I was reading about op amps and active filters. I was eager to learn about their various uses in audio applications. As I was learning quickly, I soon realized that my amplifier could greatly benefit from a custom low-pass active filter and an all-pass active filter. This led me to design the full-featured subwoofer amplifier I describe here.

Most of the theoretical background behind my design comes from the study of three books. The first one is called *The Analysis and Design of Linear Circuits*[1] and provided me with all the fundamentals I needed to know about op amps and simple active filters. Though seemingly academic, most of the book is quite easy to understand due to the pedagogical approach.

The second one, *National Semiconductor Audio/Radio Handbook*[2], contains a wealth of interesting information. Most notably, chapter 5 titled "Floobydust" is of special interest for the DIYer. This is where I picked up the schematic of my third-order Butterworth low-pass active filter that I will describe later. The title of the third book speaks for itself: *Op Amp Applications*[3]. I believe that this reference book, edited by Walter G. Jung, is a must have for any DIYer using op amps in his projects.

DESIGN OBJECTIVES

Adjustments: Here is a key word you must consider if you want to achieve harmonious integration between the subwoofer and the main loudspeakers. Among the parameters to consider, cutoff frequency, phase alignment, and sound level are certainly the most important. These features are all part of the design illustrated in the block diagram (**Fig. 1**).

I explain each of these blocks from a theoretical perspective in Part 1. In the following parts, I will discuss the practical aspects of the project, namely, the making of the PCBs (Part 2) and the construction of the subwoofer amplifier (Part 3). The fourth and last part will address the performance of the subwoofer amplifier, including some measurements and listening results. As a "bonus feature," I will also explain how to build two preamplifiers derived from the same circuitry.

NON-INVERTING BUFFERS & INVERTING SUMMER

This stage of the circuitry deals with the input signals. As shown in **Fig. 2**, you can connect three sources to the amplifier. In my case, I wanted to use my subwoofer amplifier for both a home theater system and a stereo system. Be-

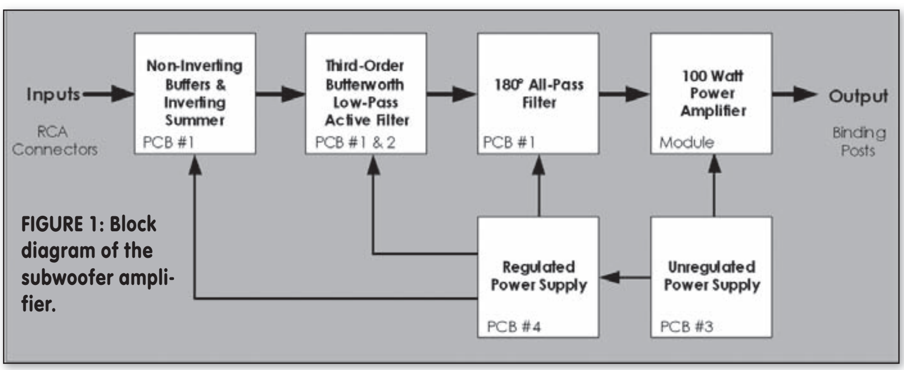


FIGURE 1: Block diagram of the subwoofer amplifier.

cause I neither intended nor had the room to install two subwoofers for stereo operation, I needed to be able to convert the stereo signal to a monophonic one. Hence the two non-inverting buffers and the inverting summer. Should you need only a single input for whatever reason, then this part of the circuitry could be greatly simplified by keeping only one buffer and skipping the rest of the components.

It is well known that switching both signal and ground helps preserve the integrity of the signal coming from the source. When all the grounds of the RCA connectors are tied together, the common ground of the selected source and of the amplifier is now shared by all other sources connected to the amplifier. The network therefore produced may introduce distortion. This issue seems to be especially sensitive for preamplifiers in which you generally connect multiple sources.

For this reason, I chose a four-pole three-position rotary switch (S_1). This enables me to switch the signals and grounds of all my sources. The first position receives the stereo signals coming from my preamplifier. The next position is dedicated to my DVD player. Note that in this case, only two poles are connected. I left the last position free for future use.

Following the source selector S_1 are the input capacitors C_1/C_2, which contribute to the isolation of the amplifier by blocking unwanted DC voltages that might come from the source. These capacitors form a first-order high-pass filter with the input resistors R_1/R_2. The cutoff frequency of this filter is determined by:

$$f_{cHP} = 1/2\pi RC \qquad (1)$$

This filter provides a low cutoff frequency of 4Hz so that maximum gain is obtained at frequencies around 20Hz.

The two input buffers, whose main function is to isolate the amplifier from the source, provide unity gain as it should be by definition. IC_1 as well as IC_2 and IC_3 are OPA-2604 dual op amps[4]. I selected these devices throughout my project because they benefit from an excellent reputation in audio applications while being reasonably priced.

It is possible that your application requires more gain. In this case, you could replace the non-inverting buffers with non-inverting AC amplifiers as shown in **Fig. 3**. In this diagram, C_0 should remain at 0.47µF and R_2 at 100k in order to maintain the 4Hz high-pass filter. Select the value of R_1 to obtain the appropriate gain. Finally, adjust C_1 so that $R_2C_0 = R_1C_1$.

The signals emerging from the non-inverting buffers converge toward the inverting summer. The gain of the inverting summer is given by the following equation:

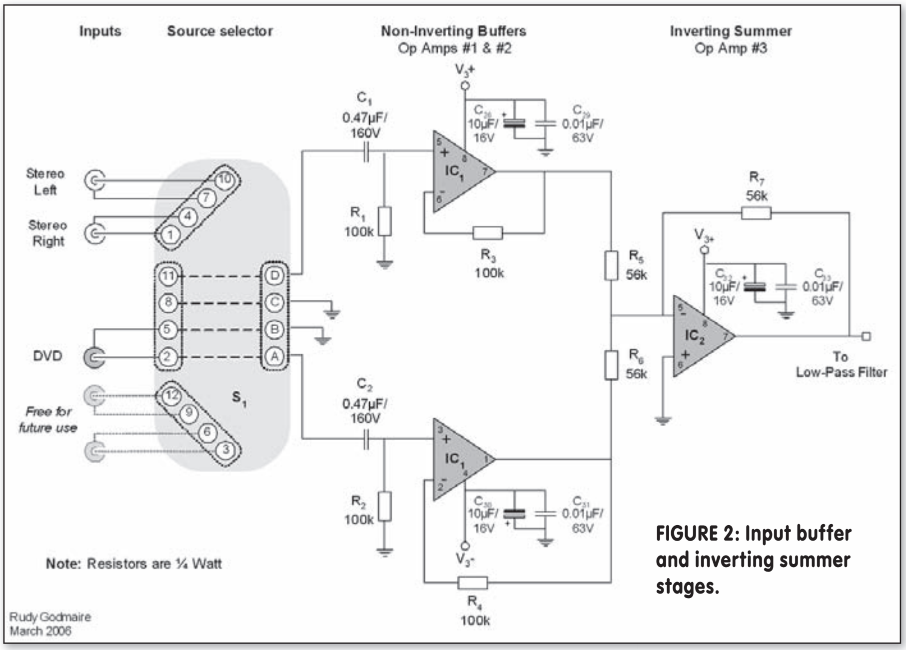
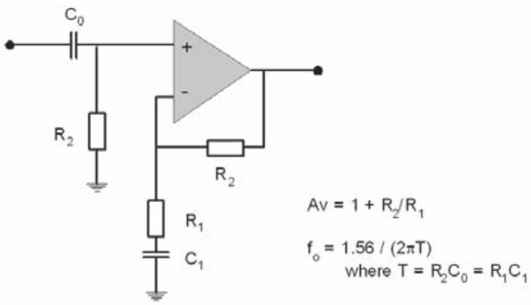


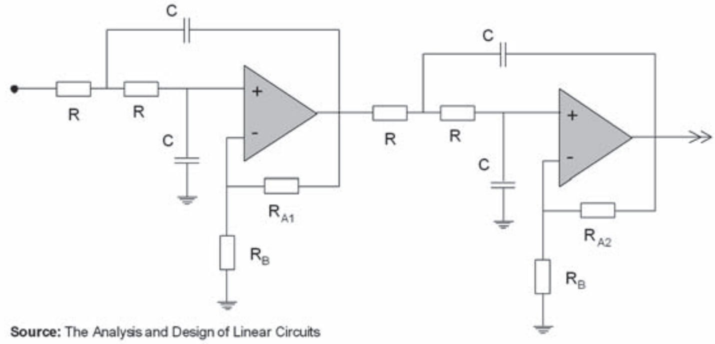
FIGURE 2: Input buffer and inverting summer stages.

FIGURE 3: Non-inverting AC amplifier.



Source: National Semiconductor Audio/Radio Handbook (1980)

FIGURE 4: General fourth-order Butterworth low-pass active filter.



Source: The Analysis and Design of Linear Circuits

$$K = K_1 + K_2$$
$$\text{where } K_1 = -R_7/R_5 \text{ and } K_2 = -R_7/R_6 \quad (2)$$

Note that the gain of the inverting summer will vary depending on the source selected. When the stereo inputs are selected the gain is –2, while it is –1 for the DVD player.

THIRD-ORDER BUTTERWORTH LOW-PASS ACTIVE FILTER

Designing this stage has been the most challenging activity of the project. My first thought was to build a fourth-order Butterworth low-pass active filter with a cascade of two op amps as in **Fig. 4**. The topology had the advantage of a very steep attenuation slope of –24dB per octave. Replacing the four resistors *R* with a four-ganged potentiometer would enable the adjustment of the cutoff frequency. However, I was not enthusiastic about the idea of using multiple cermet potentiometers in the signal path because I thought that it would degrade the signal too much.

In fact, I had another idea in mind. I wanted to produce a variable low-pass active filter that would use precision resistors instead of cermet potentiometers. A switching system would enable the simultaneous connection of the resistors and provide a few positions to change the cutoff frequency. Easier said than done; this idea quickly became very complex to implement on a PCB due to the cascaded op amps.

National Semiconductor Audio/Radio Handbook provided me with an elegant solution. The "Floobydust" chapter contains an example of a third-order Butterworth low-pass active filter that is based on a single op-amp design. **Figure 5** shows the general presentation of this filter with its equations. This was the key element I needed to design an audiophile-grade switching network.

A few sketches and days later, I found a way to implement a four-pole three-position rotary switch that would enable the selection of three different cutoff frequencies. The final circuit is depicted in **Fig. 6**.

I have selected three cutoff frequencies based on my particular needs:

- 48Hz, a custom value that was selected empirically;
- 80Hz, the THX standard;
- 100Hz, the Dolby Pro-Logic standard.

The first frequency is specially tuned for stereo music in conjunction with my own system. I determined this cutoff frequency following a process of trial and error, which you would think would be painful. But the experimentation has been made easy thanks to the use of exchangeable "chips" of resistors, which I will describe in Part 2.

The last two frequencies are dedicated to home theater and should fit your needs, too. But you may also want to consider a cutoff frequency of 120Hz, the upper limit of the Dolby Digital and DTS standards. These formats offer a low-frequency effect channel (LFE) that

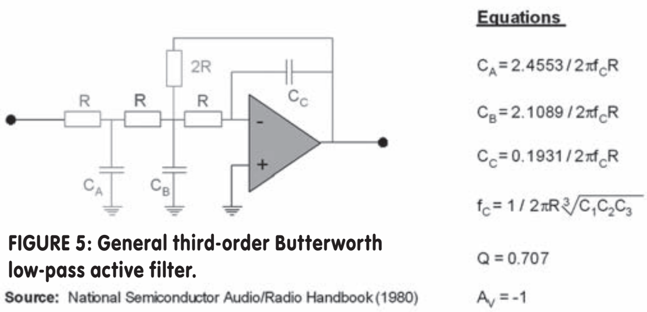
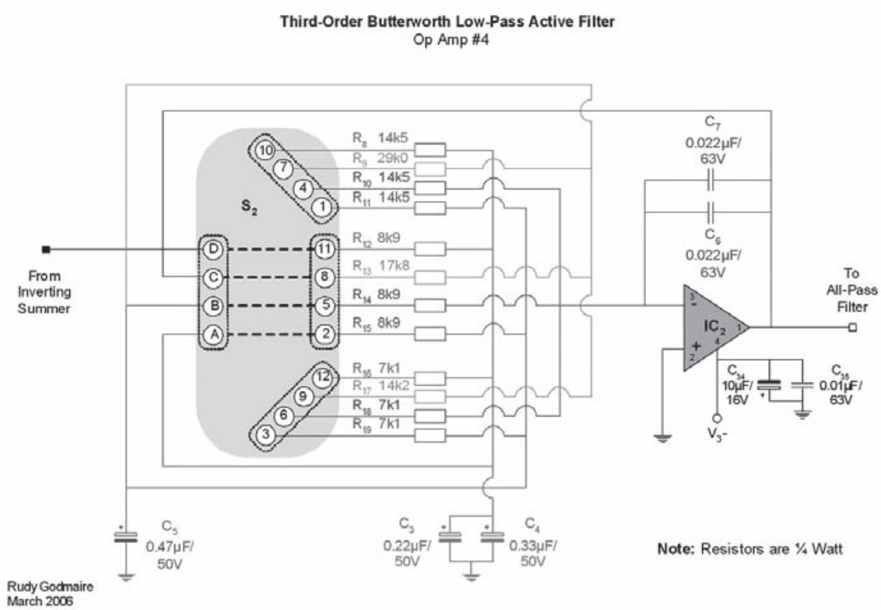


FIGURE 5: General third-order Butterworth low-pass active filter.

Source: National Semiconductor Audio/Radio Handbook (1980)

FIGURE 6: Final design of the third-order Butterworth low-pass active filter.



reproduces low bass sounds from 3 to 120Hz.

The values of the components of **Fig. 6** were established with the formulas appearing within **Fig. 5**. For your convenience, **Table 1** provides pre-calculated values for different cutoff frequencies. I recommend that you stick with the values suggested for C_A, C_B, and C_C ($C_{3, 4}$, C_5, and C_6 in **Fig. 6**) because they meet very closely the theoretical values calculated with these formulas. However, you can calculate other custom values of *R* and *2R* using the following formula:

$$R = 2.4553/2\pi f_{cLP}C_1 = 710\,497/f_{cLP}$$
where f_{cLP} is the cutoff frequency (3)

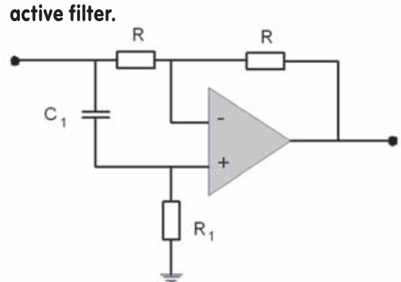
VARIABLE 180° ALL-PASS FILTER

Phase alignment is an important aspect of a successful implementation of a subwoofer. You can do this mechanically or electronically. In my opinion the best placement for a single subwoofer is right in the middle of the main loudspeakers, which enables physical alignment of the cones.

However, in most home installations this choice is not an option. The subwoofer will lie generally somewhere else in the room, which makes mechanical phase alignment impossible to achieve. This is where the variable all-pass filter finds its use.

Figure 7 shows the general presentation of a first-order all-pass active filter. Theoretically, this circuit enables a phase shifting that goes from –180° at DC to 0° at high frequencies, all without changing the amplitude of the signal. In other words, an all-pass filter provides unity gain at all frequencies. The cutoff frequency f_{cAP} is determined by the high-pass function of C_1R_1 and is calculated using equation 1. At f_{cAP}, the phase shift of this first-order all-pass filter will be –90°.

FIGURE 7: General first-order all-pass active filter.



Source: Analog Devices, Op Amp Applications

As mentioned in a technical brief available on the website of *Maxim/Dallas Semiconductors*[5], the phase shift realized by this circuit at any given frequency can be found by:

$$\text{Phase (rad)} = \tan^{-1}\left[\frac{\frac{2?}{R_1C_1}}{?^2 - \left[\frac{1}{R_1C_1}\right]^2}\right] \quad (4)$$

where ω is the frequency in rad/s, or $2\pi f$, when *f* is in Hertz.

You can draw some conclusions from this equation. Changing the value of *R*, *C*, or ω $(2\pi f)$ will have an impact on the phase shift. You can definitely take advantage of this. Replacing R_1 of **Fig. 7** with a potentiometer will provide you with a very simple solution to control and adjust the phase shift.

Another conclusion is that when the frequency *f* of the musical signal changes, the phase shift changes as well. This tells you that perfect alignment will be reached only for a specific frequency. I analyze the effect of this in Part 4.

For now, it is important to know that

TABLE 1 Pre-calculated values of R and 2R as functions of f_{cLP}

Theoretical values			Fixed real values			S_2
f_{cLP} (Hz)	R (Ω)	2R (Ω)	C_A (μF)	C_B (μF)	C_C (μF)	Switch Position
25	25k3	56k6	0.55	0.47	0.044	
30	23k6	47k2	0.55	0.47	0.044	
35	20k2	40k4	0.55	0.47	0.044	
40	17k7	35k4	0.55	0.47	0.044	
45	15k7	31k4	0.55	0.47	0.044	
48	14k5	29k0	0.55	0.47	0.044	1
50	14k1	28k2	0.55	0.47	0.044	
55	12k9	25k8	0.55	0.47	0.044	
60	11k8	23k6	0.55	0.47	0.044	
70	10k1	20k2	0.55	0.47	0.044	
80	8k9	17k8	0.55	0.47	0.044	2
100	7k1	14k2	0.55	0.47	0.044	3
120	5k9	11k8	0.55	0.47	0.044	

TABLE 2 Some ratings of the Amplimo A120 power amplifier module

Ratings	
Output power, RMS	120W into 4Ω 100W into 8Ω
-3dB frequency range	5Hz to 40kHz
THD at 2W into 4Ω	0.08%
THD at 90% Pmax. into 4Ω	0.09%
Intermodulation distortion at 90% Pmax.	0.09%
Signal to noise ratio	107 dBA
Input voltage for Pmax. into 4Ω	1V
Input impedance	22kΩ
www.amplimo.nl	www.plitron.com

as a practical matter, a first-order all-pass active filter will generally provide an overall shifting of about 120°. This is why I decided to cascade two first-order all-pass filters in order to get at least 180°. **Figure 8** shows the circuit I developed for my amplifier. By carefully selecting the values of the components, I was able to produce a circuit that provides a means to modify the phase—within the subwoofer's frequency range—by over 180°. Because you have two cascaded first-order all-pass filters, f_{cAP} will now be at –180°.

As mentioned earlier, I replaced R_1 of **Fig. 7** with a stereo potentiometer (P_1/P_2) to enable easy adjustment of the phase. I placed the additional resistors R_{20}/R_{23} in series to prevent the signal from being shunted to ground when P_1/P_2 is positioned at 0Ω.

Finally, the all-pass filter is followed by a 10k log potentiometer (P_3) that acts as a volume control. I have decided to introduce the volume control at this stage of the circuitry because it enables the use of a mono potentiometer. You could also install it right at the beginning of the first stage in place of R_1/R_2 in **Fig. 2**. Should you prefer this configuration, you simply select a 100k stereo potentiometer.

100W POWER AMPLIFIER

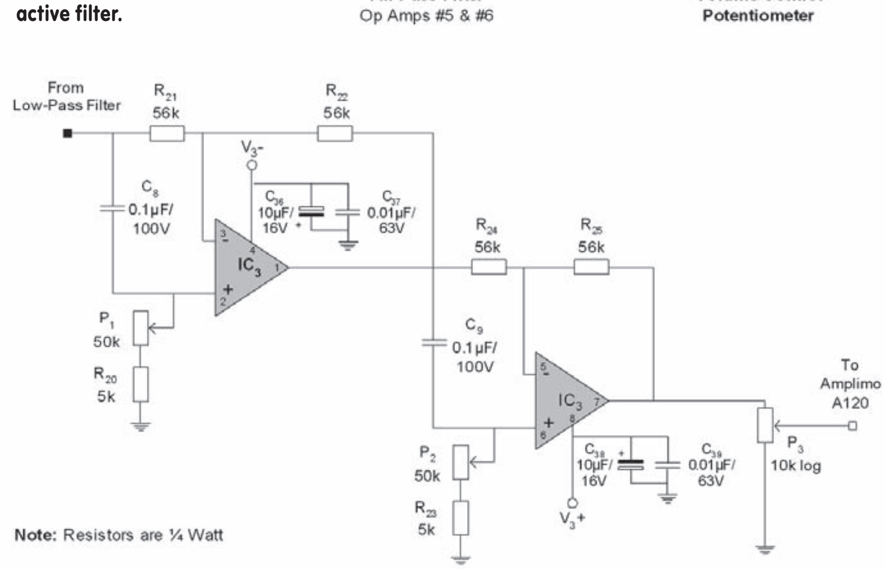
This was the easiest part of the project. Instead of building an amplifier from scratch, I decided to purchase a ready-to-use power amplifier module, the Amplimo A120, a nice solid-state amp. These modules are available either from Amplimo in the Netherlands or Plitron in Toronto. **Table 2** gathers some information available on their websites.

You might consider other solutions here. For instance, you could build an amplifier based on the well-known LM3875 power amplifier manufactured by National Semiconductor. Mike Gustafson published an interesting article[6] (*GA* 2-3-4/00) in which he compared a tube-based subwoofer amplifier to a LM3875 based solid-state amplifier. More recently, I have seen a very powerful subwoofer amplifier built around the same LM3886 in the French magazine *Led*[7]. The circuit described in the article contained eight of these ICs mounted in a push-pull configuration and produced 280W into 8Ω! The choice is up to you whether you prefer to build a unit or to use a power amplifier module such as those manufactured by Amplimo.

Figure 9 provides a schematic view of the A120 power module and its surrounding network. The amplifier is designed to operate with a relay, which performs several protection functions. One of these is to provide a one-second delay after the application of mains voltage so that the subwoofer is protected from the switch-on noise. The A120 will function without the relay, but neither the subwoofer nor the module will be protected under fault conditions.

There is also an automatic volume control feature that prevents the amplifier from clipping. Connecting the LIM

FIGURE 8: Dual first-order all-pass active filter.



pin to ground will disable this automatic volume control, and then the amplifier will clip on overload. You can install a switch between the LIM pin and the ground to activate or deactivate the feature. I chose to permanently enable this feature by leaving the LIM pin unconnected.

Note that the purpose of the LED feature is to indicate when the automatic volume control is in operation. It does not indicate when the amplifier is on or off, as you might imagine. I have decided not to connect this LED. Following the relay is a double-pole double-throw switch (S_3), which acts as a phase inverter and combines with the phase shifter to provide a 360° adjustment capability.

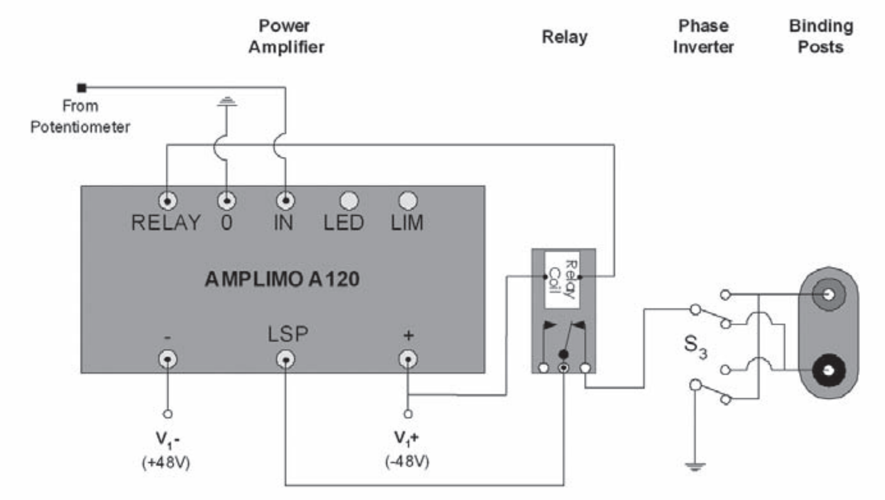
UNREGULATED POWER SUPPLY

As you can see in **Fig. 10**, this circuit is pretty straightforward. Nevertheless, I will give some explanations. I selected Z_1 in a time-delay version due to the high inrush current at power on. The 160 VA power transformer provides two secondary taps of 33V at 2.42A. This model is recommended by Amplimo and Plitron to complement the Amplimo A120. Should you use another type of power amplifier, then you will need to choose the right power transformer to fit your needs.

The full-wave bridge rectifier uses Schottky diodes (D_1 to D_4) paralleled with small value polypropylene capacitors (C_{10} to C_{13}). I selected these diodes for their low noise characteristics. You could use another type depending on your personal preferences. Make sure, however, they can pass at least 3A.

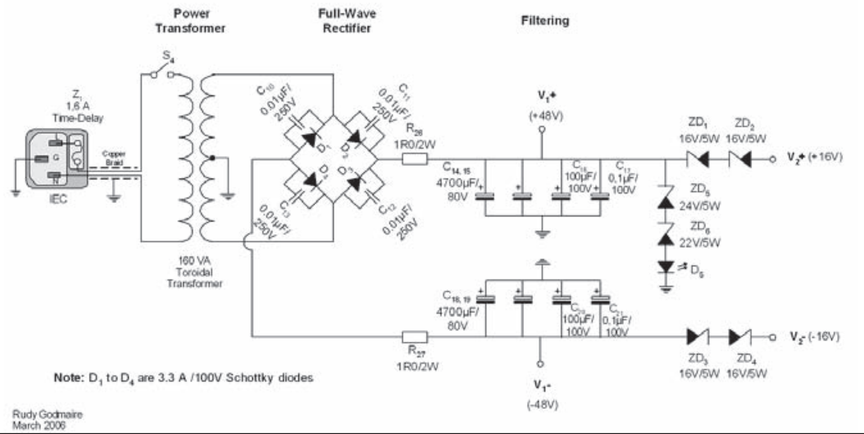
Following the full-wave rectifier, you find two power resistors (R_{26}/R_{27}), whose main purpose is to limit the inrush cur-

FIGURE 9: Amplimo A120, relay, phase inverter switch, and binding posts.



Sources: www.amplimo.nl and www.plitron.com

FIGURE 10: Unregulated power supply circuit.



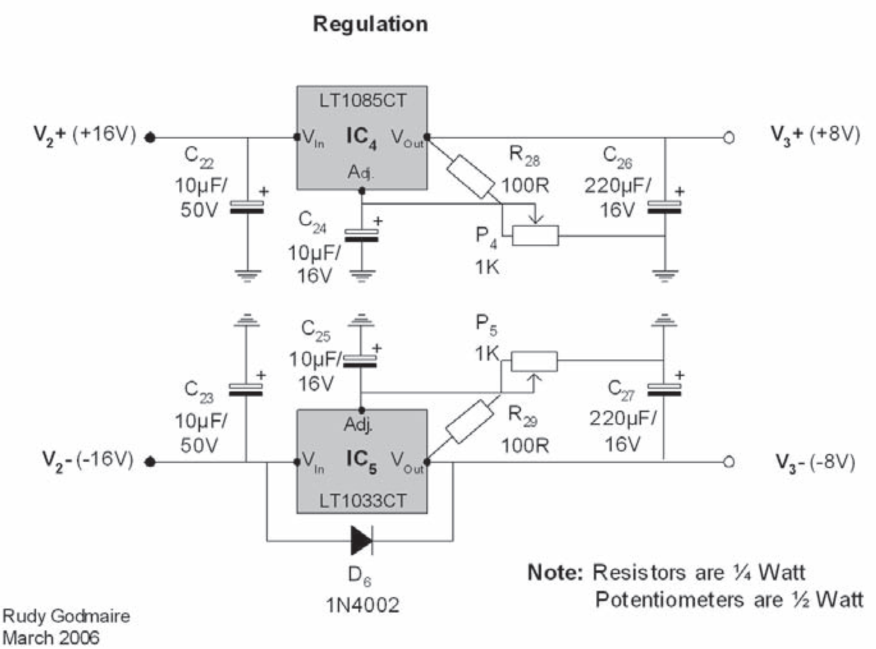
rent at power on. They also form a low-pass filter with capacitors $C_{14–17}/C_{18–21}$. The cutoff frequency of 17Hz contributes to the reduction of the 120Hz ripple voltage. Note that the 4700µF capacitors need not have 80V tolerance; 63V would be perfect. I used 4700µF/80V simply because I had these on hand. It is also important to mention that C_{16}/C_{20} and C_{17}/C_{21} are bypass capacitors that are meant to reduce ripple voltages, and also note the ESR of the large electrolytic capacitors $C_{14, 15}/C_{18, 19}$.

The red LED D_5 indicates whether the amplifier is on or off. The zener diodes $ZD_{5, 6}$ simply decrease the voltage to around 2V, within the operating range of the LED.

This power-supply circuit generates DC voltages of ±48V that directly power the Amplimo A120. These voltages are too high for the regulated power-supply circuit that follows (**Fig. 11**). This is why each rail finds two 16V/5W zener diodes ($ZD_{1, 2}/ZD_{3, 4}$) in series to bring these voltages down to ±16V. Beware that these zener diodes will become quite hot when the amplifier is in operation.

On my first prototype, I had them installed on the regulated power-supply board. Their leads conducted considerable heat to the input capacitors C_{22}/C_{23}, which was really not a good thing. This is why I moved the zener diodes to the unregulated power-supply circuit, where the components are quite rugged. The wires between ZD_2/ZD_4 and the regulated power-supply board are long enough to alleviate this heat transfer problem.

FIGURE 11: Regulated power supply.



REGULATED POWER SUPPLY

This is the last circuit of the subwoofer amplifier. Its purpose is to provide a very clean voltage source to the op amps. To fulfill this task, I selected two adjustable voltage regulators manufactured by Linear Technology: the LT1085CT *(IC_4)* and the LT1033CT *(IC_5)*. Both regulators will deliver up to 3A output current. **Figure 11** depicts the circuit of my regulated power-supply board.

As mentioned in the datasheets[8], the input capacitors C_{22}/C_{23} are required because the regulators are more than 4″ from the filter capacitors $C_{14–17}/C_{18–21}$ located on the unregulated power-supply board.

The output voltages are controlled by resistors R_{28}/R_{29} and potentiometers P_4/P_5. The following simplified formulas enable their calculation within 1% of accuracy:

$$V_{OUT^+} = 1.25V\,(1 + P_4/R_{28}) = V_3+ \quad (5)$$

$$V_{OUT^-} = 1.25V\,(1 + P_5/R_{29}) = V_3- \quad (6)$$

C_{24}/C_{25} are optional components. Bypassing the adjustment pin with a capacitor reduces the output ripple, noise, and impedance. The output capacitors C_{26}/C_{27} are required to ensure frequency compensation and stability.

Note, however, that in the case of the LT1033CT (IC_5), this capacitor will necessitate the use of an external protection diode D_6. Without this diode, the voltage regulator might be damaged by the discharge of C_{27} should the input voltage become shorted. The LT1085CT's (IC_4) internal protection is sufficient to avoid such damage as long as C_{26} is smaller than 1000µF.

You will notice in **Fig. 11** an unusual orientation for R_{28}/R_{29}. Following the recommendations of Linear Technology, I soldered these resistors very close to the output pin of the regulators. The drawing also reveals that the ground leads of P_4/P_5 are connected to the ground leads of C_{26}/C_{27} instead of being connected directly to the star ground point. These subtleties are recommended by Linear Technology to ensure the best load regulation.

One last word about the power-supply circuits: All op amps have been bypassed with 10µF/16V electrolytic capacitors and 0.01µF/63V polypropylene capacitors in order to shunt to ground any remaining ripple voltages. If you wish to learn more about this topic, I encourage you to read an interesting article[9] by Kevin Ross about bypass capacitors.

CONCLUSION

This first article in this series has explained all the theoretical aspects of my project. Part 2 will focus on making the PCBs and will support the circuits I have just described. aX

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