Analog Bass Control for Magnepans

Improve the performance of your Magnepan sound system with this bass manager.

The Bass Management system is a standard feature of every multichannel audio/video amplifier. In a bass managed system the bass of all channels, below a crossover frequency, is combined and sent to one or more subwoofers.

The crossover frequency is variable in the range between 60 to 180Hz with only predefined values. You can set it very easily from the front panel of the amplifier or with the remote control.

Usually the slopes of the low-pass and the high-pass filters are not symmetrical. The slope for the high-pass section is 6 to 9dB/octave, and the slope for the low-pass section is about 12dB/octave.

The asymmetry of the slopes of the two sections tries to compensate somehow for the non-flat low-frequency response of the small satellite loudspeakers. But still the acoustic summation of the satellite speakers and the subwoofer cannot be flat because the exact responses of the speakers are not considered.

This article describes an analog bass management system (hereafter known as ABMS) that was designed especially to overcome all these problems. With the built-in equalizer it can correct the response of the speakers. The slopes of the high-pass and low-pass filters are 24dB/octave, which are much steeper than the slopes offered by a typical digital bass management system. The result is that the satellite loudspeaker is released from the large cone movements required for the reproduction of the low frequencies and the distortion is reduced.

THE ABMS

Figure 1 shows the block diagram of the ABMS. Only 3.1 channels are included on each PCB for the left, right, center loudspeaker, and the subwoofer. The signal in all three channels follows a similar path, so I will describe only the left channel.

After the buffering of the input, the signal is divided into two directions: the high-pass section and the low-pass section. The high-pass includes a fourth-order Linkwitz-Riley filter that is realized with two series-connected second-order filters. Then comes a volume control potentiometer for the adjustment of the level, and, finally, an equalizer, which can correct the response of the loudspeaker. Also, this stage is used as a buffer to drive the output to the power amplifier. The low-pass direction is common for all three channels.

The buffered signal from each channel is driven to a mixer circuit, and a common signal is produced. The gain of the mixer is set to less than one so that the signal is attenuated and possible

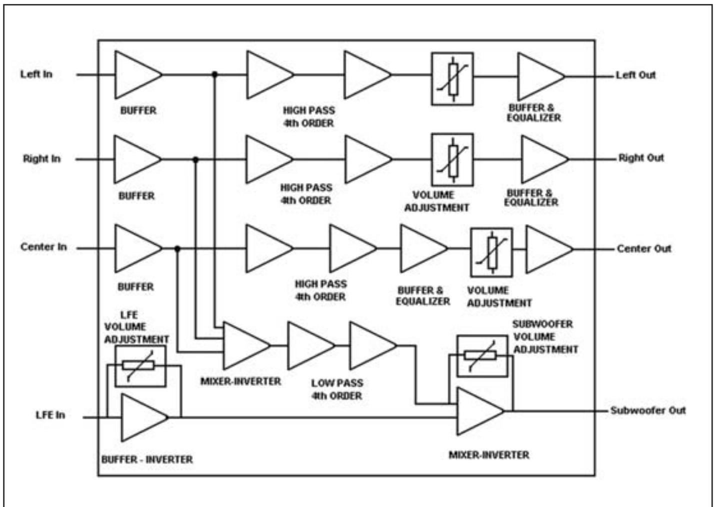

FIGURE 1: Block diagram of the Analog Bass Management System.

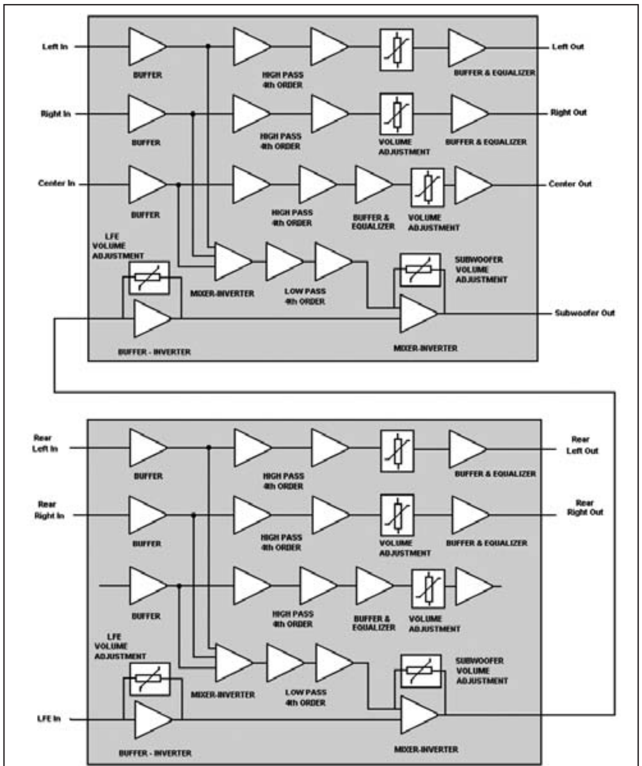

FIGURE 2: Block diagram for a complete 5.1 channel ABMS.

overload is avoided. After the mixer, the signal is driven to a fourth-order Linkwitz-Riley low-pass filter realized with two second-order stages.

The final stage is again a mixer circuit, which is used to add the bass signal from the three channels and the LFE channel. The LFE input drives an inverter/buffer circuit with a potentiometer for the level adjustment of the LFE signal.

EXTENSION FOR A 5.1 OR 6.1 SYSTEM

Although the PCB includes 3.1 channels, it is very easy to construct a full 5.1 or even a 6.1 channel system using two PCBs of the ABMS. **Figure 2** shows a block diagram for a 5.1 channel system. The first PCB includes the left, right, and center-channel speakers, while the second PCB includes the rear left, rear right, and the LFE channels.

The LFE channel of the system is connected to the LFE input of the second PCB. The subwoofer output of the second PCB is connected to the LFE input of the first PCB. The subwoofer output of the first PCB is used to drive the subwoofer of the system.

THE MAGNEPAN MC1 AND CC3 LOUDSPEAKERS

The Magnepan MC1 loudspeaker is a dipole speaker using two vertically oriented drivers—one planar magneto-static driver for the low frequencies and a quasi-ribbon driver for the high frequencies.

The Magnepan CC3 loudspeaker is a similar dipole speaker with horizontal curved quasi ribbon tweeter and midrange drivers. Both speakers are rather small considering their dipole nature and are not capable of reproducing the full spectrum. A subwoofer should be used for the lower part of the audio spectrum.

According to the manufacturer, the frequency response of the MC1 is 80Hz to 20kHz, ±3dB and should be attached on the sidewall of the listening room so the reproduction of the low frequencies is enhanced.

I very much enjoy the sound of the Magnepan speakers, but my living room does not permit such an installation, so I decided to build a special constructed stand.

The stand supports the MC1 on the floor and consists of a thick MDF base and a wooden rod on which the two supporting hinged brackets that are provided by the manufacturer are installed. Using this stand, I placed the MC1 about 1m away from the sidewall (**Photo 1**).

PHOTO 1: The MC1 as installed in author's living room.

The boost of the low frequencies due to the nearby wall does not exist any more, and the lower frequencies are attenuated. Obviously the speaker in this position needs equalization, which should be applied very carefully in order to avoid any possible overload.

The CC3 due to its horizontal orientation is placed on the top of my back projection TV. This installation is more suitable for enhancement of the low frequencies reproduction, but still some equalization is necessary.

Although my multichannel amp offers the capability of simple frequency equalization for each channel, it was not possible to equalize the Magnepans correctly, so I decided to design and build this analog bass management system for these speakers. Of course, any other loudspeaker can be driven by this ABMS by simply modifying some components.

THE DESIGN PROCEDURE

The first step is to measure the MC1 frequency response, taken as the reference response for the computation of the crossover and the equalizer. You must be very careful about how you measure. I took several measurements with the loudspeaker in different places in my living room and then computed the average response of all these measurements.

For all the measurements the following conditions are valid:

a) The microphone was at a distance between 2.5 and 3m from the speaker.
b) The height of the microphone was 1m above the floor.
c) The lateral and the vertical angle between the microphone and the central axis of the loudspeaker was never more than 10°.

The average of the measurements had as a result the elimination of the room influences, which was exactly what I wanted. The average response of all these measurements is shown in **Fig. 3**, from which you can see that the MC1 response is quite flat from 1kHz to about 9kHz. Above that the loudspeaker presents a broad peak of about 7dB around 11kHz.

Below 1kHz the response gradually falls and goes down 5dB at 100Hz.

The strong peak at 90Hz is the main resonance of the membrane. Below this peak the response drops sharply. Actually, no output should be expected from this speaker below 80Hz.

The next step is to decide the crossover frequency, which involves a very difficult decision. From the response of the MC1 it seems that the crossover frequency should be much higher than 80Hz because it is not possible to expect anything from the MC1 below that frequency. From the other side you know that the frequencies reproduced by the subwoofer should not be higher than 100 to 120Hz. With this in mind, I chose a crossover frequency of 133Hz. This frequency is somehow higher than I prefer, but I think it is a good compromise.

Until now you have measured the actual response of the speaker and chose the target acoustic response of the system. Next you can compute the filter so the total acoustic response of the loudspeaker and the filter will be equal to the target response. For the crossover design, I used the Orcad PSpice Student version. This is a free version of an excellent simulation program that offers many capabilities.

On the Schematic editor of the PSpice, enter the topology of the crossover filter and the topology of the equalizer. Using the special device EFREQ of the PSPICE, enter the measured response of the loudspeaker.

The full schematic of the circuit as used for the simulation with PSpice is shown in **Fig. 4** and consists of:

1) The upper left part:
   a) The high-pass 4th order Linkwitz-Riley filter section of the ABMS (with the components C1, C2, C3, C4, R1, R2, R3, R4, and the E1, E2 voltage sources).
   b) The equalizer circuit used for the correction of the MC1 response (with the components around U1).
   c) The average acoustic response of the MC1 is included in the "E_MC1_general_average" device. The device "E_MC1_gain" is used to adjust the gain so that a di-

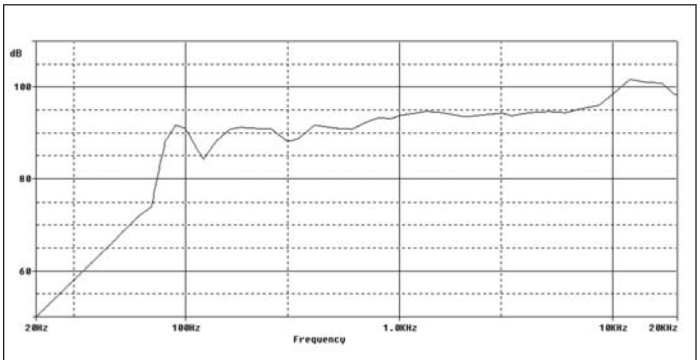

FIGURE 3: Average frequency response of the Magnepan MC1 loudspeaker.

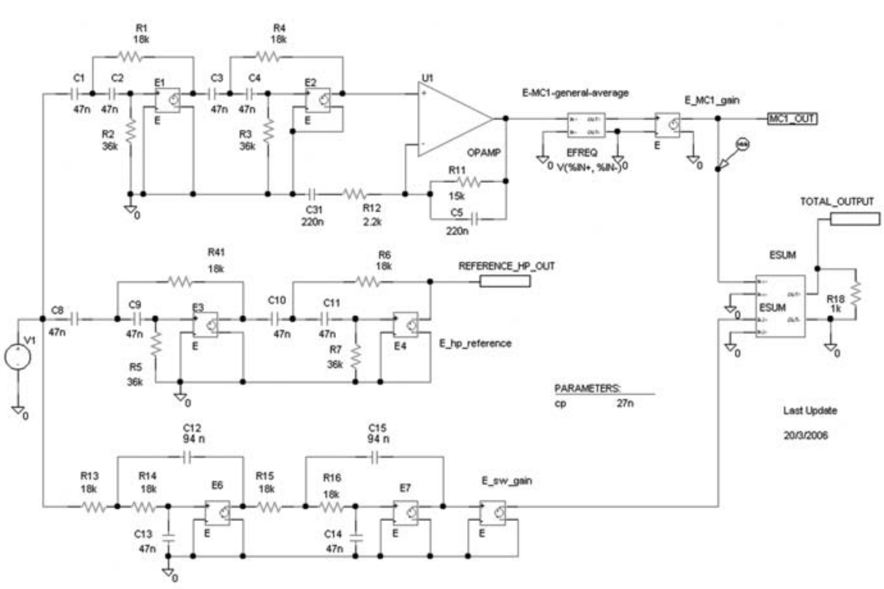

FIGURE 4: PSpice schematic for the simulation of the high-pass filter for the MC1.

rect comparison with the reference part will be easier.

At the output of this part the total acoustic response of the ABMS and the MC1 will be given.

2) The middle left part with the components C8, C9, C10, C11, R5, R6, R7, R41, and E3, E4 voltage sources. This is the reference high-pass 4^{th} order Linkwitz-Riley filter with the crossover frequency at 133Hz.
3) The low left part with the components R13-R16, C12-C15 around E6 and E7. This is the reference low-pass 4^{th} order Linkwitz-Riley filter.
4) The middle right part with the ESUM device, which is used for the summation of the responses of the MC1 and the subwoofer, and gives a view of the total output of the system.

I computed the initial values of the components using the relevant theory for the design of the Linkwitz-Riley filters. Then I modified the values of the components until the output of the filter at node MC1_OUT was as close as possible to the output at the node REFERENCE_HP_OUT. This was possible down to the frequency of 80Hz because, as I mentioned, below this limit the output of the MC1 falls sharply and it is not possible to compensate for the response of the loudspeaker. I followed a similar method for the design of the Magnepan CC3 crossover.

As with the MC1, I measured the response of the CC3 with the loudspeaker placed in its final position on the top of a large back projection TV. The microphone position was similar to the conditions described in the measurement of MC1.

In **Fig. 5**, the average response of all the measurements is shown. I took this response as the reference response for the design of the filter and the equalizer. The response of the CC3 is quite flat from about 300Hz to 10kHz. The main difference between the response of MC1 and CC3 exists in the low frequencies, where the CC3 is somehow better than the MC1. This is probably due to the different installation of the speaker because the size of the two speakers is similar. As with the MC1, the response below 80Hz falls sharply, meaning that you can expect no useful output.

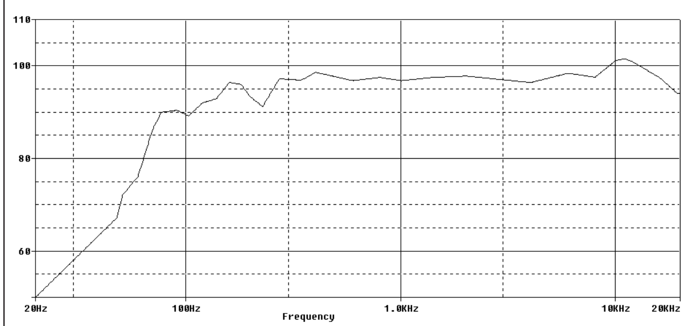

FIGURE 5: Average frequency response of the Magnepan CC3 loudspeaker.

Figure 6 shows the full schematic diagram as used for the simulation of the CC3 filter and equalizer with PSpice. The schematic consists of:

1) The upper left part, which contains:
   a) The high-pass 4th order Linkwitz-Riley filter section of the ABMS (with the components C1, C2, C3, C4, R1, R2, R3, R4, and the E1, E2 voltage sources).
   b) The equalizer circuit for the correction of the CC3 response (with the components around U1).
   c) The average acoustic response of the CC3 included in the "CC3_general_average" device.
2) The middle left part (with the components C8, C9, C10, C11, R5, R6, R7, R41, and E3, E4 voltage sources) contains the reference high-pass 4th order Linkwitz-Riley filter. The crossover frequency is 133Hz.
3) The low left part (with the components R13-R16, C12-C15 around E6 and E7) contains a reference low-pass 4th order Linkwitz-Riley filter.
4) The middle right part with the ESUM device, which is used to sum the responses of the CC3 crossover and the subwoofer in order to have a view of the total output of the system.

I computed the initial values of the components using the relevant theory for the design of the Linkwitz-Riley filters and modified the values of the components until the output of the filter was as close as possible to the output of the reference filter.

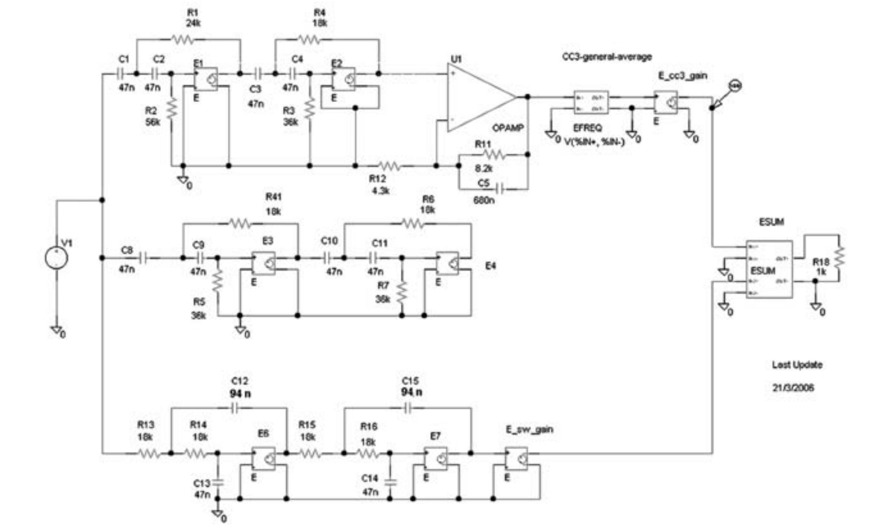

FIGURE 6: PSpice schematic for the simulation of the high-pass filter for the CC3.

ELECTRONIC DIAGRAM

The complete electronic diagram of the ABMS is shown in Fig. 7. The topology of the left and right channel is similar, so I only describe the operation of the left channel.

Components R1, R2, and C1 at the input form a low-pass filter for the rejection of the high frequencies. No DC blocking capacitor is included at the input, so you should confirm that the previous stage is free from any DC offset. Otherwise, use a DC blocking capacitor with a value greater than 4.7µF. The IC1A buffers the input signal and drives the next stages with very low impedance.

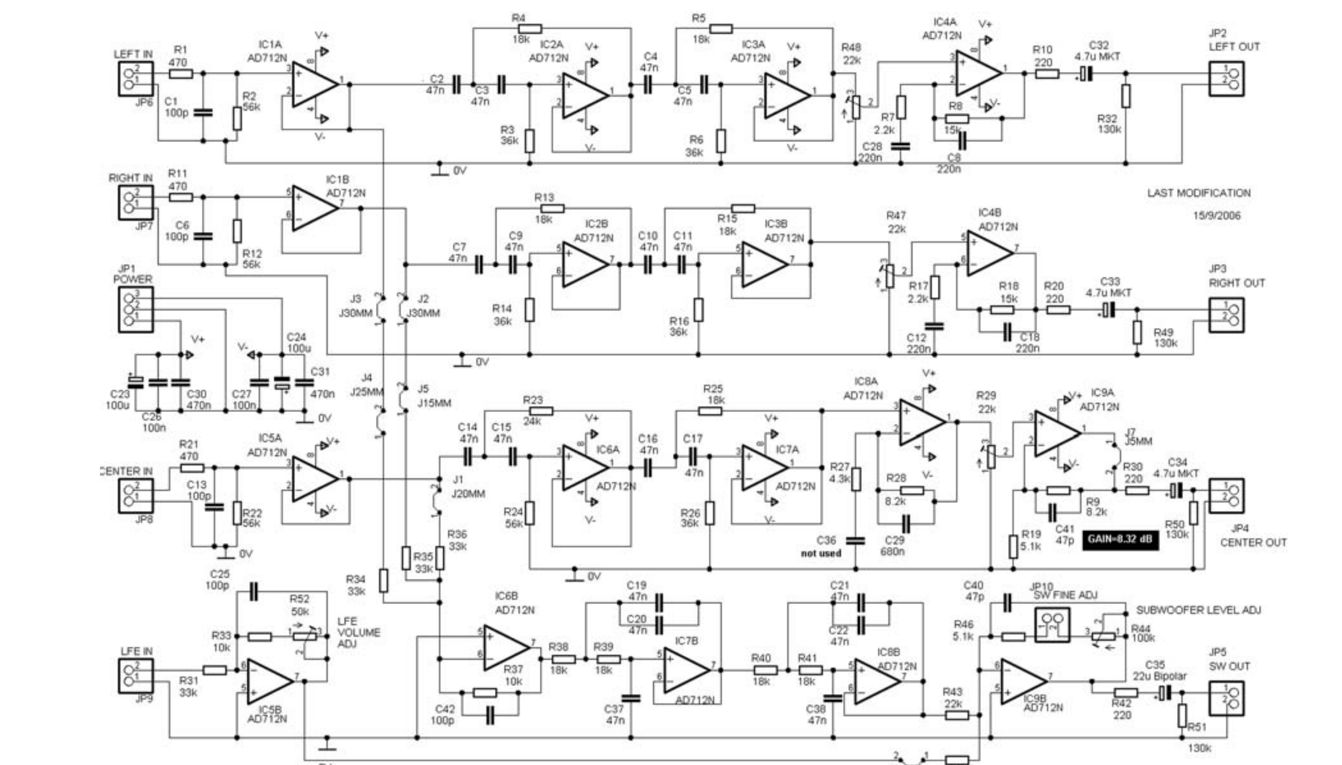

FIGURE 7: The electronic diagram of the ABMS.

Components C2, C3, R3, and R4 around IC2A form the first stage of the Linkwitz-Riley high-pass filter, and components C4, C5, R5, and R6 around IC3A form the second stage of the Linkwitz-Riley high-pass filter. The output of the IC3A drives the trimmer R48, which is used for the level adjustment.

Components R7, R8, C8, and C28 around IC4A compose the equalization filter for the correction of the frequency response of the MC1. If some other loudspeaker that has no need for equalization is driven, this stage can be used only as an output buffer by omitting components R7, C8, and C28 and by putting a short circuit in place of resistor R8. Resistor R10 buffers the output from any external capacitive load, and capacitor C32 blocks any output offset, which could produce problems in the next stage. If the next stage already has an input DC blocking capacitor, then you can replace capacitor C32 with a short circuit.

The operation of the center channel is similar to the left channel with the addition of the gain stage around IC9A. This stage has a gain of 8.32dB, useful when the sensitivity of the center loudspeaker is much lower than the sensitivity of the left/right loudspeakers. In this case you could also perform the level adjustment with potentiometers R48, R47, and R29, but I included this stage for more versatility. If you do not need this gain, you can bypass it by short-circuiting resistor R9 and omitting resistor R19.

The operation of the subwoofer channel is as follows:

From the outputs of the input buffer stages of each one of the three channels (outputs of IC1A, IC1B, and IC5A), the signal goes to the mixer amplifier IC6B. The gain of the mixer is set to –10dB in order to avoid possible overload from the summation of the three channels. The output of this mixer drives the first stage of the fourth-order low-pass Linkwitz-Riley filter, which is formed with components R38, R39, C19, C20, and C37 around IC7B.

The second stage is realized with components R40, R41, C21, C22, and C38 around IC8B. The LFE signal is connected to the IC5B buffer/inverter. Trimmer R52 adjusts the gain of the LFE channel.

The output of this stage and the output of the low-pass filter are added to the mixer amplifier IC9B. You can adjust the total gain of the subwoofer channel with trimmer R44. Additionally, you can remove jumper J10 and connect an external potentiometer in case there is a need for external level adjustment. Resistor R42 buffers the subwoofer output, and bipolar capacitor C35 blocks any offset from the signal.

THE PCB

The construction of the ABMS is very complicated to be realized on a general-purpose PCB, so I used a PCB, which I designed with the demo version of the Eagle Layout editor, which you can download for free from the Cadsoft web page (www.cadsoftusa.com). The demo version is fully operational, but includes a limitation on the maximum dimensions of the PCB. As a result, the dimensions of the PCB are 90mm × 120mm.

The placement of the components on

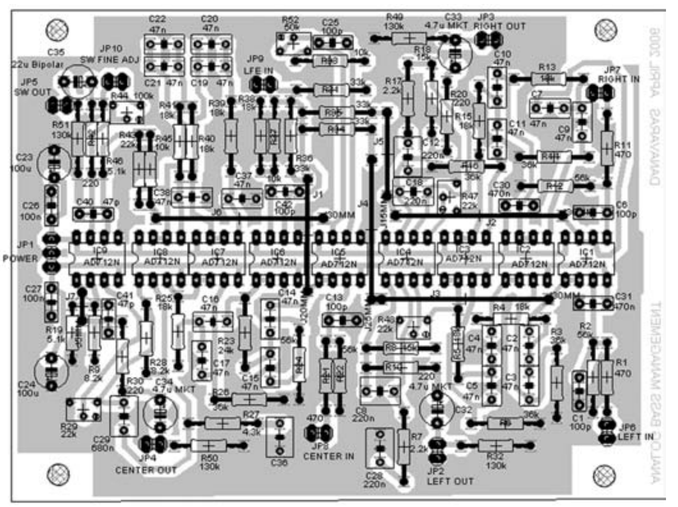

FIGURE 8: PCB layout of the components of the ABMS.

the PCB is shown in **Fig. 8**. I used pin headers for the connection of the input and output cables. I placed the trimmers that are used for the volume adjustment on the PCB, but you can use potentiometers of similar value if you need to externally adjust these. I placed all the op amps on gold-plated sockets. Place the jumpers on the PCB prior to installing all other components.

As in every filter construction, I strongly recommended you measure the value of the resistors with an ohmmeter and the value of the capacitors with a capacitance meter, at least for the components that are used for the filters and the equalizer. A tolerance of less than 1% for the capacitors will guarantee that the response of the filter will be very close to the theoretical response, which is very important.

I usually buy a large quantity of the capacitors that I need. Then I measure the value of each capacitor and choose the components with the closest value to their nominal value. This is a very good method to ensure that you have high accuracy filters for a minimum cost.

The Gerber file of this PCB is available for download at www.audioxpress.com. The file "LCR BASS MNG.zip" is a compressed file of the copper side of the PCB. When you download and unzip the file, you can use a Gerber file viewer (such as Viewmate, which is available free from www.pentalogix.com) to print the PCB on transparent paper and manufacture your own PCB. **Photo 2** shows the assembled PCB of the ABMS.

A regulated power supply based on the LM317T and LM337T regulators is used for the supply of the circuit. The output voltages are ±16V at about 60mA. You should place the power supply board on the same box and very close to the PCB of the ABMS.

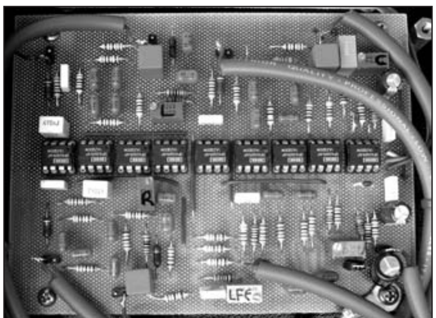

PHOTO 2: The PCB of the Analog Bass Management System.

TESTING THE ABMS

After assembling the components on the PCB, test the ABMS to verify its proper operation. You will need some basic equipment such as a low distortion audio frequency oscillator, an oscilloscope, and an AC voltage meter.

Attach an oscillator at a frequency of about 133Hz to the input of each channel of the ABMS. Then with the oscilloscope check every output to ensure that it is clean without noise or high-frequency oscillations. I chose 133Hz because with this frequency all the MC1, CC3, and SW outputs are active.

If you have a distortion analyzer then take a measurement. It is most important to verify that the filters and the equalizer have the correct frequency response. The reference response at the output of the left channel (the right channel is identical to the left channel) is shown in **Fig. 9**.

Because it is not possible to extract with accuracy the values needed for the testing, I will provide these values in dB for specific frequencies. With the oscillator connected to the left input of the ABMS, you should measure these values at the JP2 LEFT OUT connector. Similar values should be measured for the right channel also. The 0dB reference is the output signal at 20kHz.

20kHz	0dB
10kHz	+0.01dB
1kHz	+1.20dB
280Hz	+4.12dB
100Hz	-6.94dB
70Hz	-17.96dB

Follow a similar procedure for the center channel. The reference response at the output of this channel is shown in **Fig. 10.** Measure the following values in dB at the JP4 CENTER OUT connector

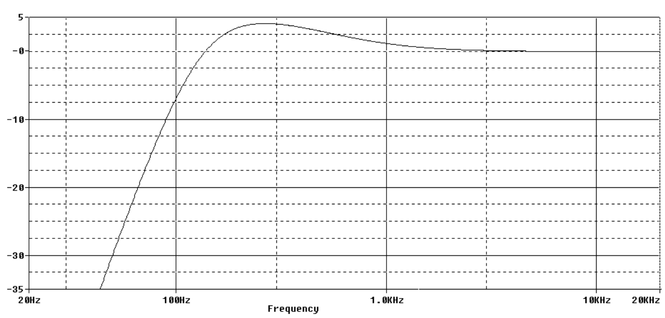

FIGURE 9: Response of the ABMS at the output of the left or right channel.

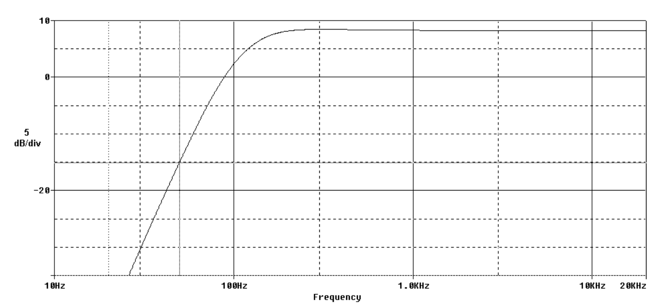

FIGURE 10: Response of the ABMS at the output of the center channel.

with an oscillator connected at the center input. The 0dB reference is again the output signal at 20kHz.

20kHz	0dB
10kHz	0dB
300Hz	+0.19dB
100Hz	-5.93dB
70Hz	-13.82dB

Finally, you should check the response of the subwoofer channel. Measure the following values in dB at the JP5 SW OUT connector when the input signal is connected only at the LEFT IN (and then at the RIGHT IN and the CENTER IN connector). The 0dB reference is the output signal at the JP5 SW OUT connector at the frequency of 25Hz.

200Hz	-15.72dB
133Hz	-6.00dB
100Hz	-2.41dB
50Hz	-0.16dB
25Hz	0dB

You should also check the response of the SW OUT with a signal connected to the LFE IN connector. In this case the response should be flat in the range 20Hz to 200Hz. If everything is according to these specifications, the testing of the ABMS is concluded.

LEVEL ADJUSTMENTS

After successful testing, you can connect the ABMS to your system where the final level adjustments for each channel will be performed. The ABMS includes five on-board trimmers (R48, R47, R29, R44, and R52) for the level adjustment of each channel. Also, you can connect an external potentiometer on the jumper JP10 (SW FINE ADJ) for external adjustments of the subwoofer level.

Each channel should reproduce the same test signal (usually a band limited pink noise), and you should adjust the potentiometers so that the level will be equal for each channel. Almost all DVD players and audio/video amplifiers have integrated test signal generators for these level adjustments. I recommend a sound level meter such as the Radio Shack model for the adjustments.

The adjustment procedure is as follows: First set trimmers R47 and R48 at about 15kΩ. Check the balance between the left and right channel and adjust if necessary with these trimmers.

Then by using the trimmer R44, set the subwoofer level with the test signal connected on the left or right channel. Next set the center channel level using trimmer R29, and finally set the LFE channel level using trimmer R52. When the adjustment is completed, a listening test with a very well-known multichannel music piece should follow in order to verify the correct adjustment.

CONCLUSION

A final measurement of the system is shown in **Fig. 11**. In this figure three responses are shown:

a) The response of the left MC1 driven by the ABMS
b) The response of the subwoofer driven by the ABMS
c) The total acoustic response of the above

The measurement was taken at a single listening position without any spatial averaging. The subwoofer is the double dipole subwoofer as described in Nov. '04 *audioXpress*. The response of the subwoofer is equalized below 100Hz according to the method described in the article "Equalize the Bass Response of Your Room" (Aug. '06 *audioXpress*). Considering that this is a measurement in a single microphone position in a typical listening room, the response is very good.

The success of the design of the system was proven by the listening tests. The sound of the Magnepans remains very detailed and is well balanced now. The matching of the Magnepans with the double dipole subwoofer is excellent.

aX

PARTS LIST

All resistors 0.4W, 1%
All capacitors 5%, MKT, 63V except as noted

Part	Value
CAPACITORS	
C1,6,13,25,42	100p
C2-5,7,9-11,14-17,19-22,37,38	47n
C8,12,18,28	220n
C23,24	100µ/35V electrolytic
C26,27	100n
C29	680n
C30,31	470n
C32-4	4.7µ MKT
C35	22µ Bi-polar 25V
C36	not used/short-circuit
C40,41	47p
ICs	
IC1–IC9	AD712N or OPA2134
JUMPERS	
J1	20MM
J2,3,6	30MM
J4	25MM
J5	15MM
J7	5MM
PINHEADS	
JP1	1X3
JP2–10	1X2
RESISTORS	
R1,11,21	470
R2,12,22,24	56k
R3,6,14,16,26	36k
R4,5,13,15,25,35,38,39,40,41	18k
R7,17	2.2k
R8,18	15k
R9,28	8.2k
R10,20,30,42	220
R19,46	5.1k
R23	24k
R27	4.3k
R29,47,48	22k TRIMMER
R31,34-6	33k
R32,49-51	130k
R37,45	10k
R43	22k
R44	100k TRIMMER
R52	50k TRIMMER

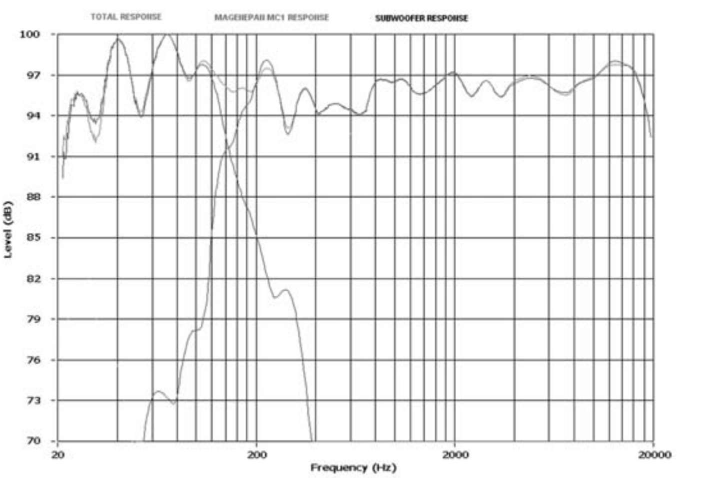

FIGURE 11: Magnepan MC1 response, double dipole subwoofer response, and the total in-room response at a single listening position using the ABMS.