Equalize The Bass Response Of Your Room

Discover how equalization can affect low-frequency reproduction in a room.

By George Danavaras

I spent a lot of time during the last years studying the influence of the listening room to the reproduction of the low frequencies. I read several articles and performed many measurements in my listening room, trying first to understand this influence and then to eliminate it.

This article describes the results of this study and also presents a general method to equalize the low-frequency response of a room using a simple analog electronic equalizer with notch filters. I must admit that these days digital technology could provide more flexible solutions for the bass equalization of a room. But before you decide to follow the digital way, you can try this method, which is simple and cheap—yet effective—and offers many benefits.

BASS REPRODUCTION

It is true that in most rooms the quality of the bass reproduction is determined more by the room than the loudspeaker itself. There are many papers that examine the room's influence on the reproduction of low frequencies, particularly articles by Floyd E. Toole[1,2,3].

Usually, the following three categories affect the room's influence:

1. The dimensions of the room
2. The position of the loudspeaker and the listener in the room
3. The sound absorption and reflection of the surfaces of the room

There is not much you can do about the first point except if you're currently building the room. For the second and third point, you should follow the general rules as given[1], although due to the long wavelengths, sound absorption cannot help much in the low-frequency range.

Of course, it is not always easy to move the loudspeakers or the listening position to the points that have less influence on the reproduction. Sometimes it is just not possible. For these cases the electronic equalization becomes a very good solution.

WHAT TO EQUALIZE

The most important issue when you want to equalize the low-frequency reproduction of a room is to know what exactly should be equalized. There is much discussion about this matter, but here is, in summary, my personal recommendation about this subject:

1. I strongly recommend the equalization of the low frequencies (below 150 to 200Hz). When applied properly, this can help dramatically the clarity of the reproduction.

2. Only a small area of the room, which is usually chosen around the listening position, can be properly equalized. The equalization in the response in one specific place usually makes the response in the other places of the room worse. I concluded this only after many tries to equalize my listening room using simple analog equalization.

3. You should correct only the large peaks of the response. Avoid the equalization of the notches that exist in the room response. For the equalization of the peaks use simple analog notch filters.

FIGURE 2: The response of the notch filter.

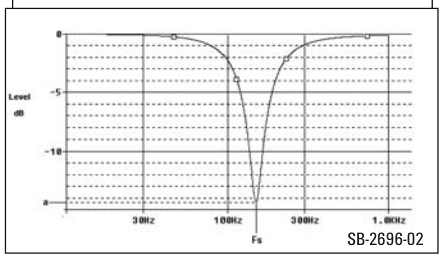


FIGURE 1: The notch filter topology and definitions.

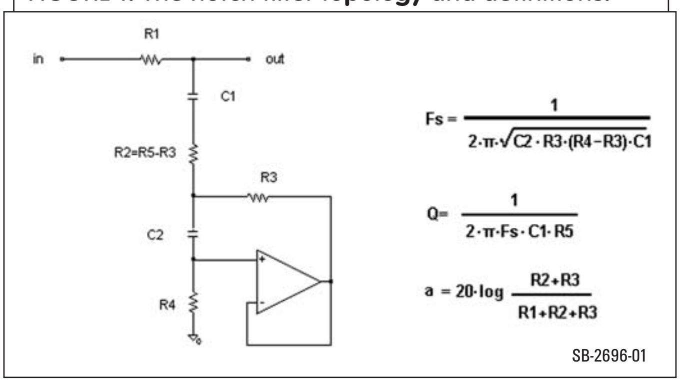
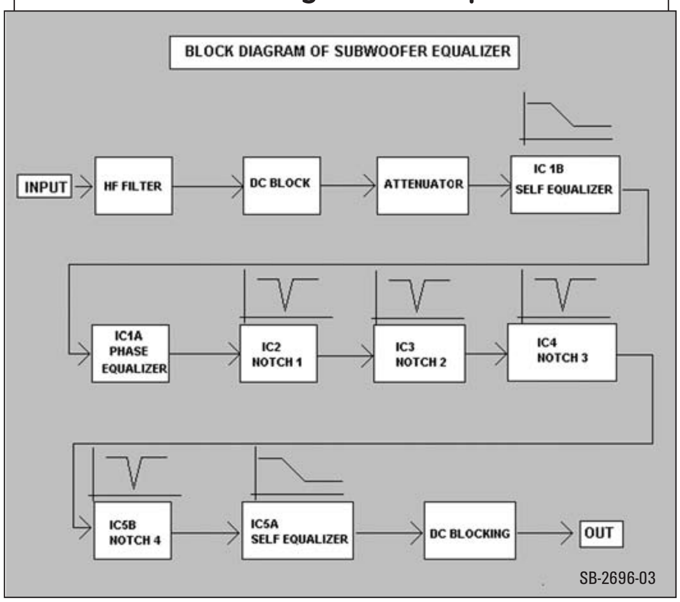


FIGURE 3: The block diagram of the equalizer.



The method that I use to realize the above is as follows:

a. First I take a number of measurements in a small area around the listening position (for example, 20 to 30cm in front, rear, up, and down directions).

b. Then I look to the energy spectral decay plots of these measurements in order to identify all the frequencies that have a very long decay, which exists in the measurements of all positions. This is an easy way to identify the real resonance frequencies of the room that should be equalized and to avoid any other acoustical interference that depends on the position of the microphone.

c. Then I compute the average low-frequency amplitude response of all the places that the measurements were taken, in order to decide about the attenuation and the quality factor Q of the notch filter that I will use for the equalization.

d. The components of the notch filters are computed and the equalizer is constructed.

e. Then I repeat all the above measurements to see whether there is a need to make any corrective action and to confirm the benefits of the equalization.

THE NOTCH FILTER

The basic part of the equalizer is the notch filter. For the design of this notch filter I used the method described in Siegfried Linkwitz' site (www.linkwitzlab.com).

The topology of a notch filter with the relevant definitions for the center frequency (Fs), quality factor (Q), and the attenuation (a) is shown in **Fig. 1**. The components R3, R4, and C2 around the op amp form an electronic inductor, which resonates with the capacitor C1, and the typical notch filter response is produced as given in **Fig. 2**. The required parameters for the design of the notch filter are:

Fs = the notch frequency

Q = the width of the notch

a = the attenuation of the filter at the notch frequency

The steps to design this notch filter (see also **Fig. 1**) include:

1. Define the requested Fs, Q, and attenuation (a) of the filter.
2. Select R1.
3. Then R5 = R1 × a/(1–a) and C1 = 1/(Q × R5 × 2 × pi × Fs), where pi = 3.14159.
4. Select R4 > 20 × Q^2 × R5. Usually you would prefer an even greater value for R4.
5. Select R3 <= R5.
6. Then C2 = (Q × R5)/(2 × pi × Fs × R3 × (R4 - R3)).

For example, consider the following parameters to compute a notch filter:

Fs = 150Hz

Q = 8

a = -15dB = 0.1778

Then, according to the previous method, the steps to follow include:

1. Select R1 = 2kΩ.
2. Then R5 = R1 × a/(1 – a) = 432.5Ω.
3. And C1 = 1/(Q × R5 × 2 × pi × Fs) = 306.7nF.
4. Select R4 > 20 × Q^2 × R5 = 554kΩ. Let R4 = 560kΩ.
5. Select R3 <= R5. Let R3 = 390Ω.
6. Then R2 = R5 – R3 = 42.5Ω.
7. C2 = (Q × R5)/(2 × pi × Fs × R3 × (R4 - R3)) = 16.8nF

Usually standard values for the components are preferred, so it is necessary to modify the above values and to re-compute the filter. For this reason, I wrote an Excel sheet that you can download from the *audioXpress* website (www.audioXpress.com).

After the download, when you open the sheet with the Microsoft Excel program, you will see two areas. On the top area you can insert the required filter values, and the program will compute the proposed network components.

You can insert the real values of the components on the bottom area, and the Excel program will compute the filter parameters. By changing the values of the components, you will realize the notch filter using standard components values.

THE ELECTRONIC EQUALIZER

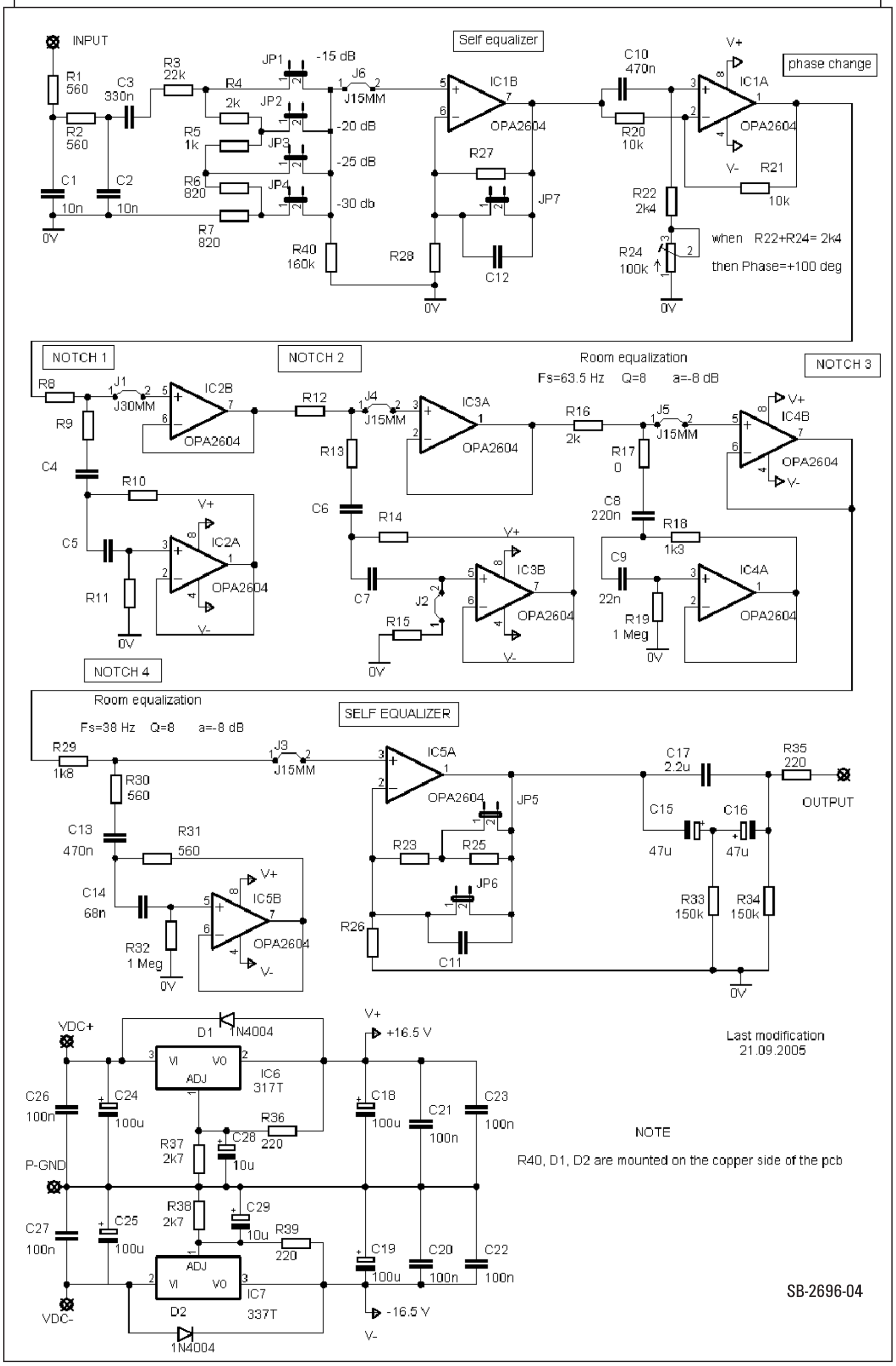
The block diagram of the equalizer is shown in **Fig. 3** and the full electronic diagram in **Fig. 4**. At the input of the equalizer, the components R1, R2, C1, C2 form a low-pass filter for the filtering of the high frequencies above the needed bandwidth. The capacitor C3 blocks any DC level and forms a high-pass filter at about 1Hz.

The components R3, R4, R5, R6, R7, and R40 form a voltage attenuator. By using the jumpers JP1, JP2, JP3, and JP4, you can attenuate the input level from -15dB to –30dB in steps of –5dB. This is very useful for adjusting the level of the subwoofer to the rest of the system.

The components R27, R28, C12 around IC1B form a buffer for the previous stage and optionally a self-filter, which can boost an area of low frequencies. This kind of filter is necessary to correct the subwoofer response in a range of frequencies. If you don't need this filter, then you can set the jumper JP7 to the ON position and the stage becomes a simple buffer.

The components C10, R20, R22, R21, and the potentiometer R24 around IC1A form an all-pass circuit, which

FIGURE 4: The electronic circuit of the equalizer.



you can use to change the phase of the subwoofer signal. When R24 is set to its maximum value of 100kΩ, the phase is zero; set to its minimum value, the phase change is about +100°.

The main speakers interact with the subwoofer near the crossover frequency (because both are playing the same signal) and cancellation can occur if the difference in the phase of the two signals is around 180°. The change of the subwoofer phase can help so that a smooth integration of the subwoofer response to the rest of the system will happen.

The first notch filter is formed with R8, R9, C4, C5, R10, R11, and IC2A. The IC2B is used to buffer the first notch filter and to drive the second notch filter with the components R12, R13, R14, R15, C6, C7, and IC3B. The IC3A buffers the second notch filter and drives the third notch filter with the components R16, R17, R18, R19, C8, C9, and IC4A. The IC4B is used to buffer the third notch filter and to drive the fourth notch filter with the components R29, R30, R31, R32, C13, C14, and IC5B.

The IC5A is used to buffer the fourth notch filter and also forms a second self-filter as described above. The components C15, C16, C17, R33, and R34 form a DC blocking stage, while resistor R35 buffers the equalizer from the capacitance of the next stage. For the op amps, I used the OPA2604 from Burr-Brown.

The regulated power supply of the circuit is based on the LM317 and LM337 regulators. A small heatsink is necessary for each regulator.

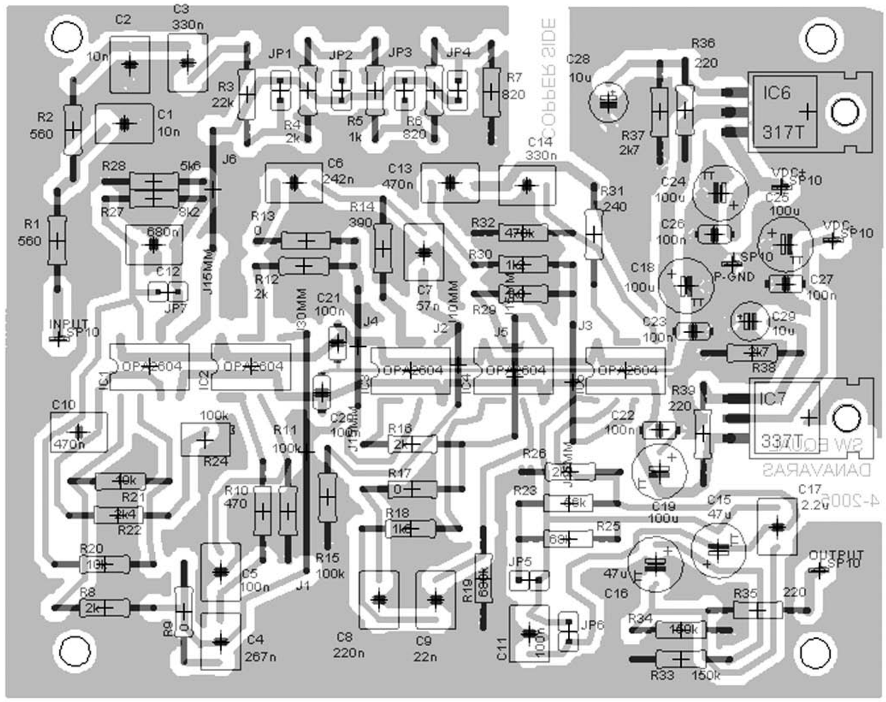
The DC unregulated voltage should be provided from an external power supply. You can use a common power supply, provided that it has an output of about ±24V DC at 500mA with low ripple.

The capacitors C24-C27 filter the input voltage, while the capacitors C18-C23 filter the output voltage. The resistors R36-R37 and R38-R39 set the output voltage at 16.5V. Diodes D1 and D2 protect the regulators during input voltage interruptions.

I also designed a single-layer 112 × 89mm PCB for the equalizer using the demo version of the EAGLE software package (www.cadsoft.de). The Gerber file of this PCB is also available for download on the *aX* website. The file SW_Processor.zip contains the compressed file SW_Processor.bot, which is the copper side of the PCB.

After downloading the file, you can use a Gerber file viewer (like View-

FIGURE 5: The layout of the components on the PCB.



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mate, which is available free from www.pentalogix.com) to print the PCB on transparent paper and manufacture your own PCB. The placement of the components on the PCB is shown in **Fig. 5**, while **Photo 1** shows the assembled PCB of the equalizer.

Please note the following two important notes when you assembly the PCB:

1. The components R40, D1, and D2 are placed on the copper side of the PCB.

2. There are two separate grounds on the PCB—one for the power supply and one for the analog filters. This flexibility could solve any existing ground-loop problems. Usually, the two grounds should be connected together either at the star ground point or directly on the PCB. Remember to connect the two grounds together; otherwise, the circuit will not work properly.

A TYPICAL EXAMPLE

I performed the following procedures for the equalization of my listening room, which measures approximately 4.5 × 10m with a height of about 2.8m. I started by taking several measurements of the impulse response of the room, with the microphone for these measurements positioned about 30cm around the ear height of the listener.

The topside of **Figs. 6–9** show the energy spectral decay as measured at four different microphone positions around the listening point. From these measurements, notice the long decays at the frequencies of 38Hz, 61.5Hz, and 66Hz that exist in all four positions. From this you can conclude that these are the main room modes that exist below 100Hz.

There are also long decays in other frequencies, but it is clear that these strongly depend on the position of the microphone, as you can see if you compare them with the measurements in the other positions. The frequencies of 61.5 and 66Hz are very near; so I decided to choose the mean value of 63.5Hz as the center frequency of the notch filter.

I determined the attenuation and the bandwidth of the notch filter at the above two center frequencies as follows:

Figure 10 is the average low-frequency ⅓ octave frequency response of the microphones for all positions. From this curve it seems that the attenuation should be about 8dB for both notch filters. I chose a value of about 8 for the quality factor Q for the notches. This is a good compromise between the sharpness of the room mode (which always demands a high Q filter) and the realization of the electronic circuit (which means that the practical realization of a high Q notch filter using electronic components is not an easy task).

So here are the two notch filters that are needed for the equalization of my room:

FIGURE 6: Energy spectral decay in position 1 without and with equalization.

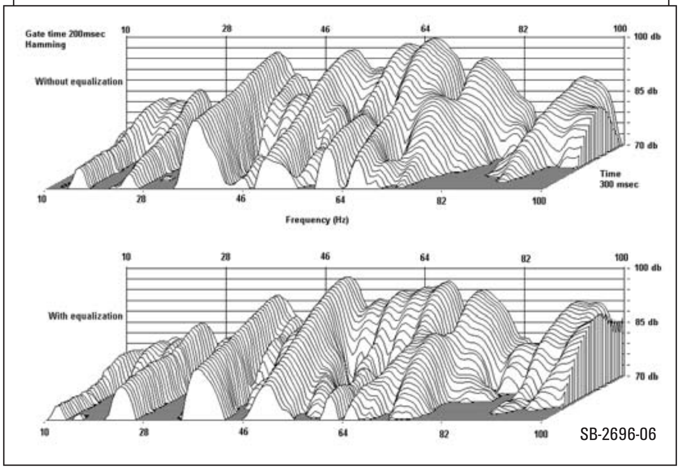


FIGURE 8: Energy spectral decay in position 3 without and with equalization.

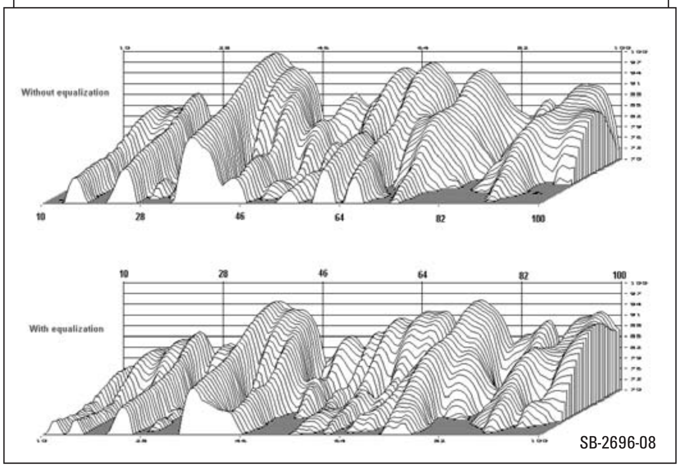


FIGURE 7: Energy spectral decay in position 2 without and with equalization.

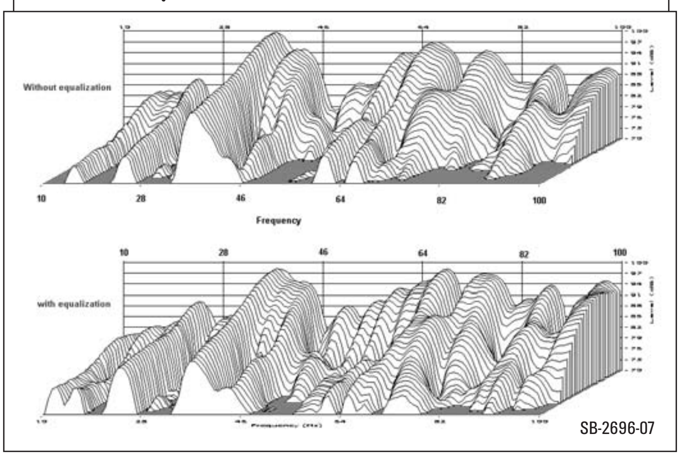
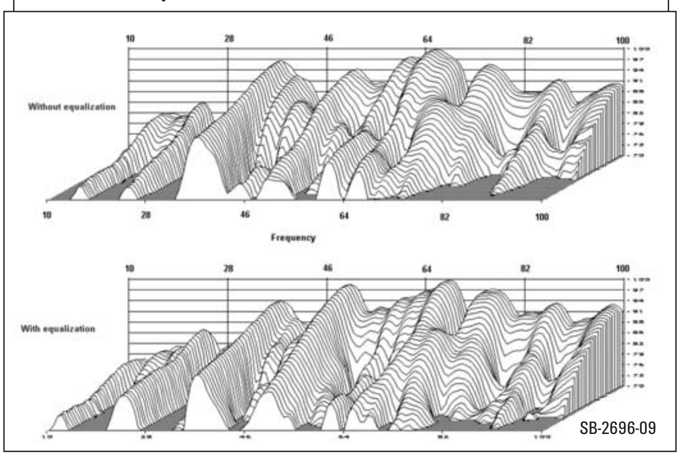


FIGURE 9: Energy spectral decay in position 4 without and with equalization.



a) Central frequency Fc = 38Hz, Q = 8, and attenuation about –8dB

b) Central frequency Fc = 63.5Hz, Q = 8, and attenuation about –8dB.

I computed the components of the filters using the method described previously, and their values are shown in the schematic of **Fig. 4** as Notch 3 and Notch 4.

After the construction and the testing of the equalizer, I again measured the energy spectral decay on the same positions as before. The results of these measurements are shown for comparison on the bottom sides of **Figs. 6–9**. From these figures you can easily see the improvement on the energy spectral decay with the equalization. The decays on the frequencies of 38 and 61.5 and 66Hz are much shorter now.

Also, the ⅓ octave average frequency response is much smoother as shown in **Fig. 11**. Comparing the responses with and without equalization, you can see that the variation in the response drops from about ±6dB without equalization to about ±3dB with equalization. This is a clear improvement.

The impulse response is also shown for both cases in **Fig. 12**. The topside is the impulse response in the listening position without equalization, and the bottom side is the impulse response at the same position with the equalization. It is easy to notice that the equalization controls much better the damping of the response.

All of this proves that the applied electronic equalization to the room response was effective. This was also confirmed by the listening tests. The improvement is dramatic. The bass reproduction is much more tight and articulate with the equalization. *aX*

FIGURE 10: Average frequency response without equalization.

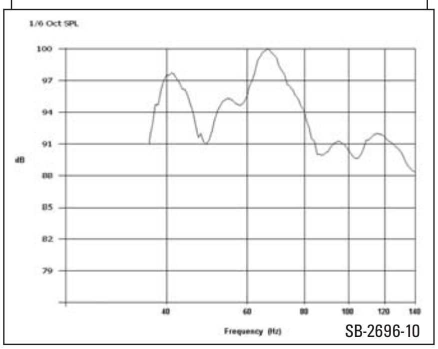


FIGURE 11: Average frequency response with equalization.

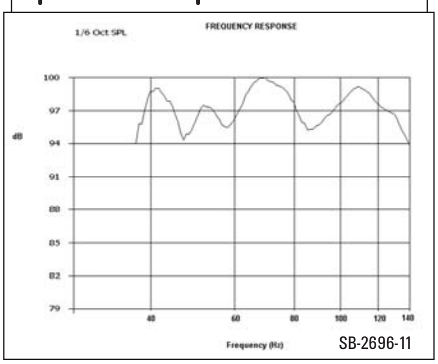


FIGURE 12: Comparison of impulse responses at the same position with and without equalization.

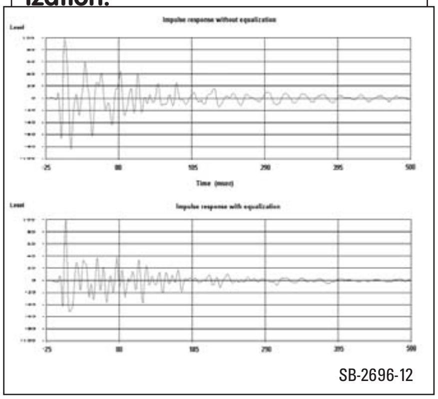


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PHOTO 1: Equalizer PCB.