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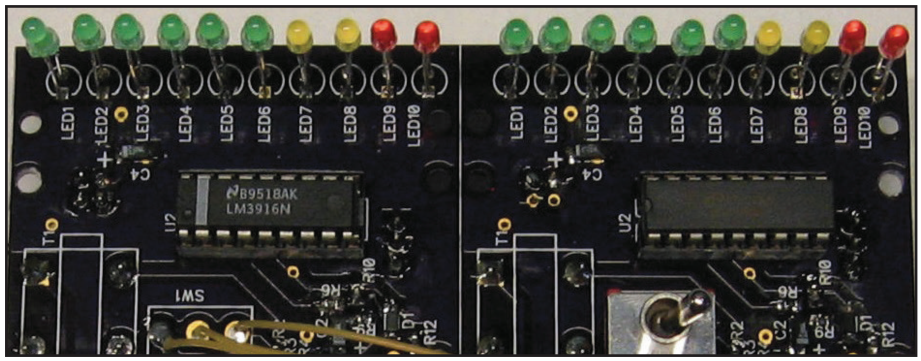

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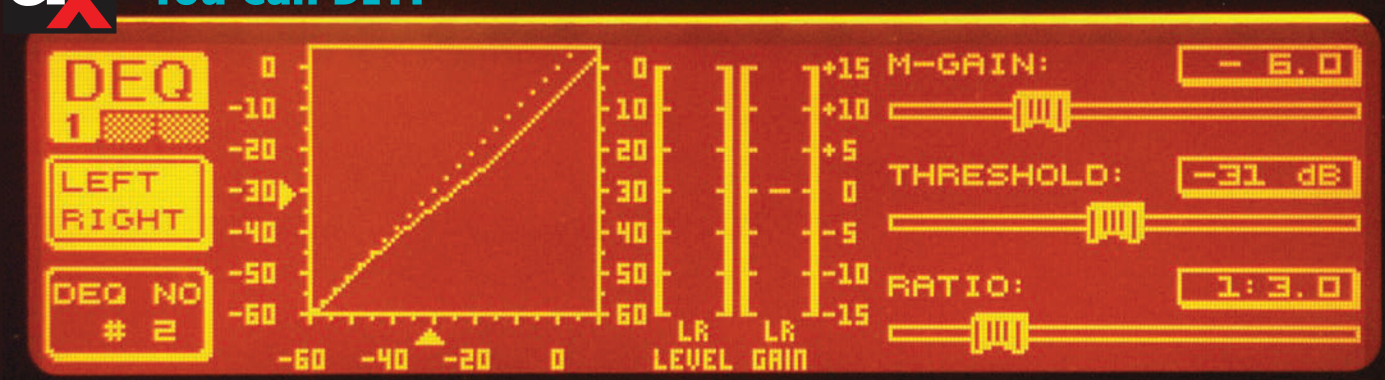

Photo 1: The DEQ function's first page details the woofer's limit over extension at low frequencies and high power.

By
Thomas Perazella
(United States)

Tips to Resurrect a Classic Speaker or Design a New System (Part 3)

Utilizing DSP-Based Processors

This third and final segment of the article series on restoring a Heil air motion transformer (AMT)-based speaker details additional problem solving methods that use DSP-based processors. The article also discusses final listening and voicing a speaker.

Although the Dayton Audio UM12-22 12" Ultimax driver has exceptional linear excursion capability, there is only so much output you can get from one 12" driver at very low frequencies. I have several CDs with extremely low frequencies that were recorded at high levels.

In particular, I have a CD from the Boston Audio Society with a recording of Camille Saint-Saëns's Organ Symphony #3, which has very high-level 17-Hz pedal notes at several places in the recording. If you attempt to play this at realistic levels with any ordinary speaker, you will surely drive the woofer out of its linear excursion range and possibly cause the voice coil to bottom out. Normally, the solution to this problem would be to reduce the overall levels. However, that will result in a low overall volume that is quite unlike listening to the real performance due to the loss of detail. Note, this is not a case of amplifier clipping.

The acoustical requirement to increase excursion with decreasing frequency to maintain the same output level creates a mechanical problem as the cone/voice coil assembly moves too far. If you have ever experienced this, you know drivers make a distinctive noise when striking the back plate that is quite unmusical and most disconcerting.

To eliminate the problem, I used the Behringer DEQ2496 high-precision digital 24-bit/96-kHz EQ/RTA mastering processor. This DEQ2496's function is controlled on three menu pages. The first menu enables you to set the amount of modification you make to the gain and is indicated in tenths of a decibel. The second menu enables you to set the attack and release times and change the trigger point. The third menu enables you to choose the correction mode and parameters.

You can use the DEQ2496 functions to determine several key signal modifiers and set specific levels where action is taken to modify the gain applied to the signal. The control parameters include the adjusted signal's modified gain level, the threshold where the modification begins, the ratio or rate that the change occurs, the time delay after the trigger level occurs for action to be taken (Attack), the time delay after the signal falls back below the threshold for the modification to cease (Release), and the type of action.

Controlling driver excursion is another excellent use of the DEQ function. In this case, I used a high-pass function with a –6-dB gain reduction, a –31-dB threshold level, and a 1:3 ratio (see **Photo 1**).

Since this iteration of the DEQ function was set for a lower frequency then longer attack and release times could be used. They were 34.86 and 304.2 ms, respectively (see **Photo 2**).

On menu page three, I selected the L12 mode for the high-pass function with a 12-dB/octave slope. At first, Behringer's nomenclature is confusing, since we always think of a high- or low-pass filter as having unity gain except for the modified areas, which have their gain lowered. However, in this function, you can decrease or increase the gain.

The nomenclature makes more sense if you think of increasing the gain. In this case, increasing the gain in the lower frequency range would create a "low-pass" device. However, I was lowering the gain in the low frequencies, making the "low-pass" function actually more consistent with generally accepted notions of a high pass.

Still confused? Fortunately, the parameters are graphically represented so what you select is immediately apparent. I chose a 39.9-Hz frequency. **Photo 3** shows the resulting correction.

Testing after I implemented this DEQ function resulted in a normal listening level for the "Organ Symphony #3," with no ugly noises coming from the woofer during the passages with the heavy pedal notes. The audible improvements this function makes are substantial, enabling you to enjoy the music without annoying distractions. Although the real answer to accurately reproducing these types of recordings requires several drivers and a lot of amplifier power, I find the ability to use normal speakers and amplifiers to achieve most of what the music contains astounding. You have to hear the results to believe it. You do make compromises, but they are a fraction of what you would have to do without DEQ.

Voicing the Speaker

For the sake of this project, I stretched the traditional definition of speaker voicing to encompass adjustments that affected many segments of the chain. The common definition of speaker voicing covers crossover points, crossover slopes, and drive levels to individual drivers. Box size, speaker positioning, and baffle shapes are also part of the equation.

I have been asked several times why you have to voice a speaker. Many people think when they purchase a speaker the manufacture has already done that job. If you build your own speakers and you chose drivers that are flat, you may think you only have to adjust drive levels and crossover points.

Photo 2: The DEQ function's second page provides input screens showing the attack and release times.

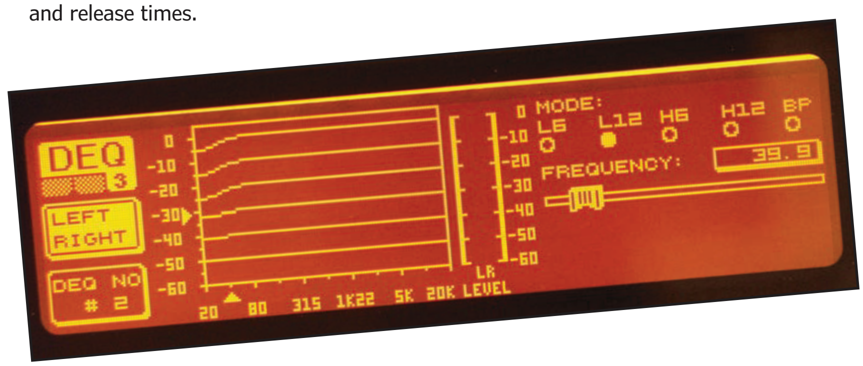

Photo 3: The DEQ function's third page numerically and graphically shows the mode employed and slope with the turnover frequency.

Unfortunately, neither of these statements is true. First, I have never seen drivers with totally flat response. Second, radiation pattern differences will affect the final balance at the listening position. Third, microphone frequency, pattern differences, and their placement during the recording have major effects on the sound. Fourth, mastering decisions can vastly affect tonal balance. Fifth, speaker placements in the room and room effects also have major impacts on the final sound.

One of the easiest ways to demonstrate how microphone choices affect the final sound is to play track 5 of the first *Stereophile* test CD. J. Gordon Holt, *Stereophile's* founder, sat at a desk and read one of his articles into 18 different microphones, one at a time. He did not change anything else in the recording or subsequent processing between the sections recorded with different microphones. You will find it difficult to believe it is the same person speaking. The differences are dramatic. If just the microphone choice makes such a difference, imagine what some of the other factors can do to the resulting sound. You can either ignore the differences or try to compensate for them.

It is important to remember that if the recording process significantly modifies the original sound,

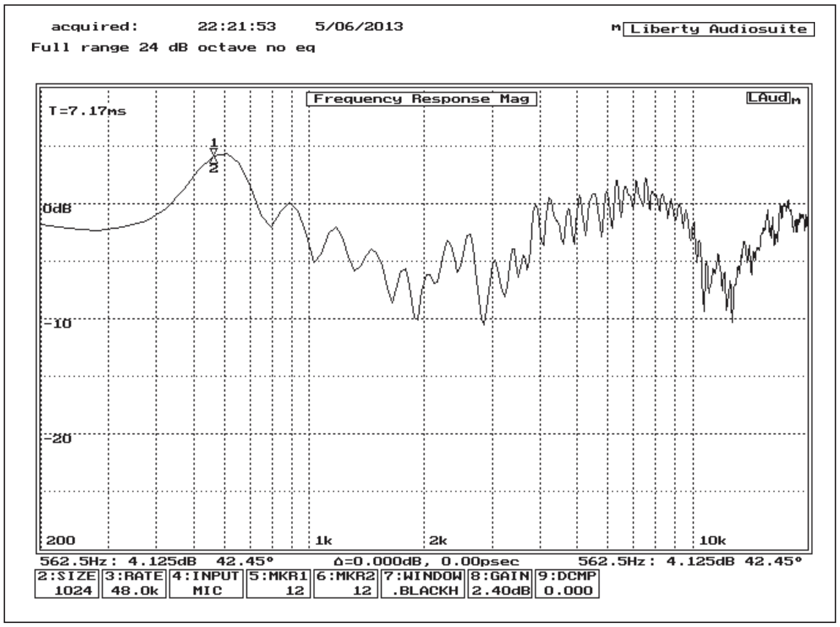

Figure 1: The completed speaker's quasi-anechoic response is shown after the appropriate drive levels were set with 24-dB/octave slopes, but before any EQ was done.

there is no practical way to entirely retrieve the original. What you can do is make the resulting sound more to your liking.

I began the voicing process by trying to get the speaker as flat as possible on axis with quasi-anechoic testing. Then, I made adjustments to the balance at the listening position. In the second phase, some of the adjustments begin to correct for room and radiation pattern effects.

A common trap when voicing a speaker is to try to eliminate all room effects. It is just not possible. One of the worst problems is cancellations at certain narrow and sharp frequency bands. They are caused by energy at that frequency, reflecting off multiple surfaces. This results in the combinations of reflections being out of phase, causing a drop in levels. As you put more energy into the problem frequency, the reflected energy from various surfaces gets equally greater and the cancellations still occur. When you do this, you also throw the balance off for all the other non-canceling positions in the listening area. Probably the best way to avoid this situation is to move the speakers or the listening position and to judiciously use room treatments.

During the speaker voicing process use a flexible, low-distortion, wide-dynamic range equalizer. Prior to digital signal processing high-quality equalizers were expensive. Now, it is possible to get a digital equalizer that offers more flexibility than ever at prices that are within almost every budget range. In addition to equalization, some of them also include signal generators, real-time analyzers, compressors, limiters, dynamic EQ, and more. For years, I have been using Behringer equalizers with great success in all my systems. With this project, I used the Behringer DEQ2496 mentioned earlier.

I used Liberty Instruments's application Liberty Audiosuite (LAUD) for PCs during the first phase in which quasi-anechoic measurements are made. An ACO Pacific 7012 microphone capsule feeding a model 4012 preamplifier and a PS9200 power supply/interface provided the input. To protect the PC, I fed output from the microphone interface through a custom interface I built and described in *Speaker Builder* (see Resources).

The measurement configuration consisted of one channel with the microphone located 1 m from the speaker on axis at a 44″ height (i.e., approximately ear level when seated). Using a tri-amped setup with an electronic crossover enables you to easily switch from measuring the overall speaker to separately measuring each driver by using the mute buttons on the crossover for the appropriate channels.

Using the Behringer DSC2496 Ultra Drive crossover, I selected a 24-dB/octave slope and revised 176- and 1,500-Hz crossover points. Next, I took a measurement of the complete speaker with no EQ. Note that these measurements are raw with no smoothing. **Figure 1** shows the results. The responses when using the 24- and 48-dB slopes were similar but a 12-dB slope showed a greater suckout in the mid- to high-interface region and was somewhat rougher.

The steep slopes of a 24- and 48-dB/octave iteration are more effective at eliminating interference between drivers located at different positions. When using a 12-dB/octave slope, there is more overlap between drivers with resulting cancellations and

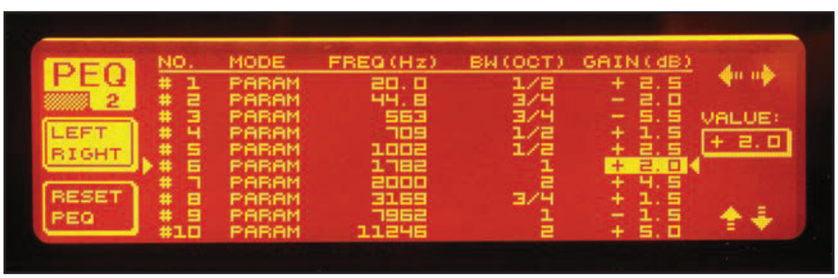

Photo 4: The final Parametric EQ in data form was applied to the completed speaker to achieve the flattest quasi-anechoic response.

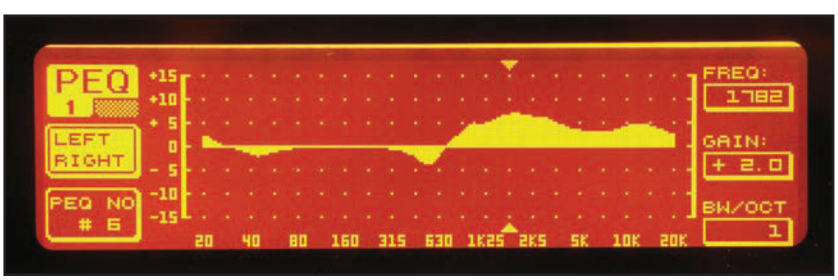

Photo 5: This is the final Parametric EQ in graphical form that was applied to the completed speaker to achieve the flattest quasi-anechoic response.

reinforcements at different positions as you move along the axis between drivers.

Next, I used the DEQ2496 to correct the frequency response. The digital electronic equalizer provides the option to use several parametric EQ channels. Being able to break away from fixed graphic type effects enables you to make finer adjustments across the entire frequency range.

For this exercise, I used the DEQ2496's 10 parametric bands to produce a fairly flat curve. **Photo 4** shows the actual applied corrections in data form. **Photo 5** shows the same EQ in graphical form. The curve's complex shape is apparent and would be extremely difficult if not impossible to achieve using passive crossover components or a standard graphic equalizer. If you look at each section's bandwidth values, you will notice they vary and are fairly broad from 0.5 to 2 octaves. This type of overlap enables you to generate a complex curve.

Overall, the equalized response at 24 dB/octave is fairly flat from the lower measurement limit of about 300 Hz to 20 kHz. **Figure 2** shows the exception, with two suckouts around 12 kHz and 15 kHz. They appear to be cavity resonances due to the Heil AMT's construction. I was able to slightly reduce the effect, but with suckouts that result from cancellations, you cannot totally remove them with more drive.

As with room problems, if you increase the drive at a suckout frequency, the higher output reflected from the housing's offending sections increase proportionately and continue to produce cancellations. The only solution is to change the physical construction that caused the interfering reflections or to add absorptive material. I added damping material inside the AMT housing but it had a limited effect. It is interesting to note the current AMT's housing has a more open design and some of the cavities have been eliminated, possibly eliminating this suckout.

Earlier, I mentioned the Eton midrange had some response irregularities as its frequency increased. **Figure 3** shows the driver's raw response when mounted in free air. Between 2,500 and 5,000 Hz, there is a significant ragged peak. It appears to be the reflected energy from the basket, which will not only change the timbre but also cause congestion. **Figure 4** shows the comparison with the same driver on the baffle with the 24-dB/octave crossover slope and the speaker's final EQ applied. The major problem in the 2.5-to-5-kHz region effectively disappears. You would not be able to achieve that result with a shallow slope in your crossover.

Music Curves

Finally, I produced a set of curves that would compensate for some of the worst deviations found in the recording/mastering process. In theory, you could produce a separate compensating curve for every cut on every record but you would quickly go crazy in the process. I have found that producing about five to 10 curves will suit the vast majority of recordings.

To test the speakers, I have a group of musical test recordings that I have used for years. Recently, I added several new selections to that list as they became available. To streamline the process, I burned four CDs with the individual tracks I wanted. The CDs included pop, rock, jazz, classical, and country selections. Some of the music also contained sounds designed to determine transient

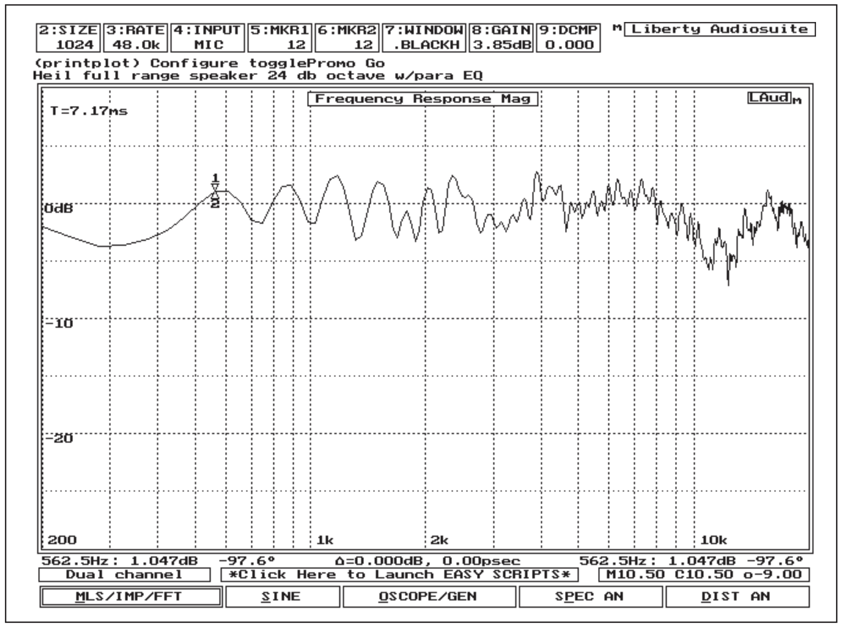

Figure 2: The completed speaker's quasi-anechoic response is shown after the appropriate drive levels were set with 24-dB/octave slopes with the final parametric EQ.

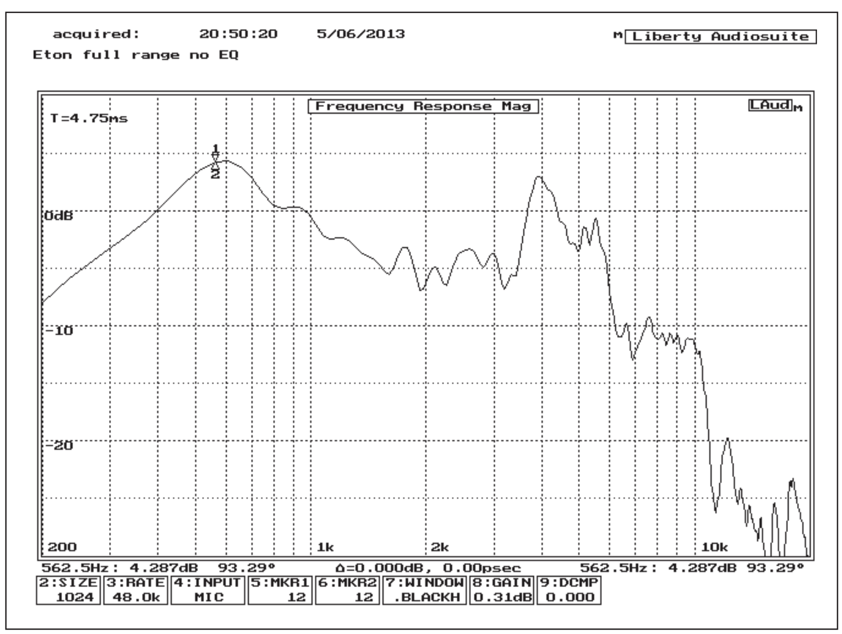

Figure 3: The quasi-anechoic response of the Eton driven full range with no EQ shows a ragged peak between 2,500 and 5,000 Hz.

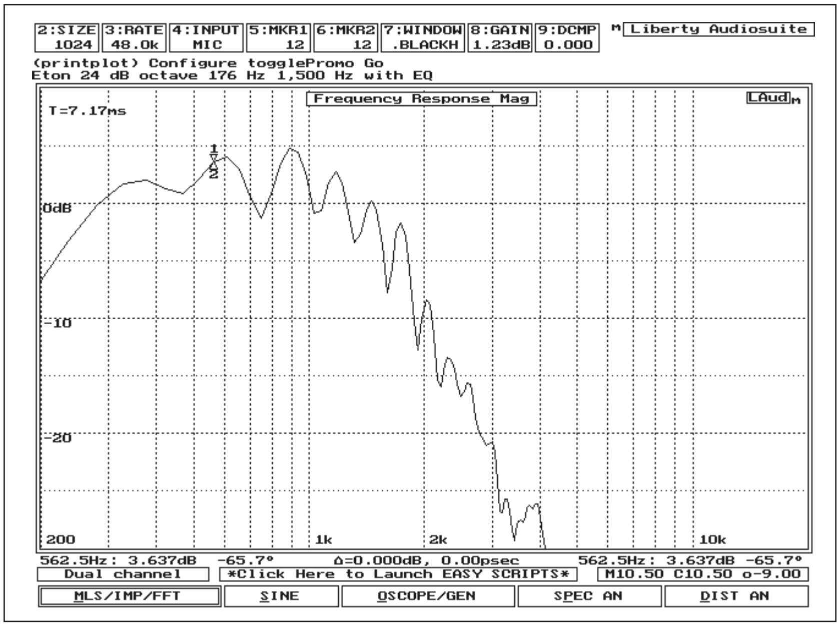

Figure 4: The Eton's quasi-anechoic response with 24-dB/octave shows crossover slopes at 176 and 1,500 Hz with compensating EQ.

response, dynamic range, frequency range, sound stage and depth, and definition.

When implementing listening tests, you can improve your results and make the process easier by finding the correct playback level for each track before it is seriously auditioned. If the level is too low, subtle details will be lost. If it is too loud, it will sound unnatural regardless of settings. I ran through each track with the base EQ, determined what I felt were the correct levels, and recorded them in a reference spreadsheet. I later used that sheet to record the final correction curve and relevant information on the piece. Inevitably when conducting listening tests with others, I always get questions about the music selections. The spreadsheet is an easy way to answer the questions and eliminate time looking for background information.

A few of the local audio club's members took part in the listening. Individual musical tastes differ, but sometimes you can ferret out a problem that all people recognize. In this test, I used the DCX2496 connected to my laptop to switch between different crossover slopes—while keeping all other parameters the same. The listeners determined that 48 dB/octave was not optimal. This surprised me because in my main system the 48-dB slope is clearly superior. It provides the greatest clarity, stage size, and smoothness. My only guess is that the controlled and consistent radiation patterns of that system's drivers lend themselves to the higher slopes.

The 24-dB/octave slope worked well. However, the 12 dB/octave had some serious shortcomings. One of the most noticeable characteristics was a shrinking of the sound stage as the slopes decreased. At the 12 dB/octave, there was a considerable reduction in the soundstage's width and depth as well as a loss of detail. I decided to stay with the 24-dB/octave slope. The 176- and 1,500-Hz crossover frequencies eliminated some of the earlier problems I noticed, so they remained.

One of the Behringer DEQ2496's great advantages is a function called "Virtual Paragraphic Equalizer" (VPQ). The function enables you to adjust the effective Q or bandwidth of each individual frequency in the graphic equalizer from a default one-third octave to 59/3 octave in two-third octave steps, which represents one-third octave on both sides of the center frequency.

By adjusting the bandwidth to a range around 3 octaves as a start, you can move the EQ's center frequency up or down with a simple one knob adjustment while listening to the music. The function enables you to rapidly locate the center frequency of the problem compared to the stark effects created by gross high Q adjustments, which can be misleading. For example, if you have a sound that is too forward in the presence region, you can start with a 3-dB cut at a center frequency around 3 kHz and use a bandwidth of 3 (see **Photo 6**).

By adjusting one knob, you can move it back and forth in one-third octave increments until you find the spot that sounds best (e.g., 2 kHz). I did this several times with different pieces and created curves that seemed like reasonable compromises for each of them. Naming them for the problems they corrected (e.g., Pop Bright, Presence, Honky, etc.) made them easy to remember. Once you save them, you can simply turn one knob on the equalizer to recall all your adjustments. Then, you are ready to have a more pleasurable listening experience.

When you use an equalizer to make compensating curves, resist the temptation to use a lot

About the Author

Thomas Perazella is a retired IT director. He is a member of the Audio Engineering Society, the Boston Audio Society, and the DC Audio DIY group. He has authored several articles in professional audio journals.

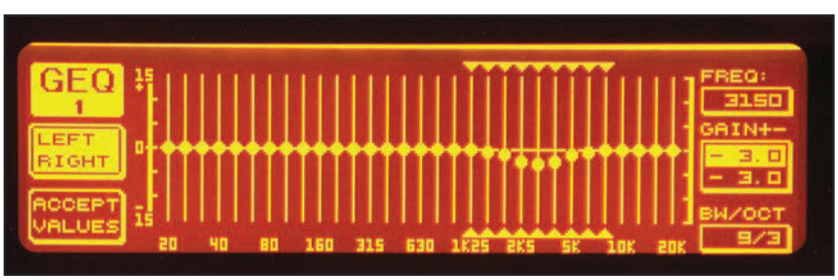

Photo 6: The Virtual Paragraphic Equalizer function permits you to set almost any bandwidth for an equalizer correction. It also enables you to move up and down the frequency scale using a single knob, while retaining that same bandwidth and levels with the test music playing.

of narrow corrections or corrections that exceed more than a few decibels. The goal is to eliminate gross problems while maintaining a realistic sound. The smoother you make the corrections the better the resulting sound. Much of the music I listen to sounds good with the basic setting, but it is helpful to correct some of the less than great recordings of music that moves me.

Final Impressions

In a nutshell, was the project successful? I believe it was. The sound is dynamic, wide and flat in frequency response, low in distortion, and provides sound stage width and depth. Transients are exceptionally good, adding to the sense of realism. Even with the basic design's limitation—using one driver for each of three frequency ranges—the addition of the new woofer plus the application of corrections provided by the digital equalizer and crossover improved the sound quality when compared to the original version.

Will this replace my primary system with huge line arrays that have the same radiation pattern and large amounts of linear volume displacement? No way! But, if I had size and cost restrictions, I could easily live with these resurrected Heil-based speakers. Long live DSP!

Resources

T. Perazella, "Interface for LAUD Measurement System," *Speaker Builder*, 1999.

C. Saint Saëns, "Organ Symphony 3," Boston Audio Society, Test CD-1, Track 3.

Stereophile, "Why Experts Disagree," Test CD 1, Track 5, http://ssl.blueearth.net/primedia/product.php?productid=2&cat=0&bestseller=Y.

Sources

7012 microphone capsule, 4012 preamplifier, and PS9200 power supply/interface
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ULTRA-DRIVE PRO DCX2496 digital loudspeaker management system and DEQ2496 processor
Behringer | www.behringer.com

UM12-22 Ultimax DVC subwoofer
Dayton Audio | www.daytonaudio.com

Liberty Audiosuite (LAUD) audio measurement and analysis system for PCs
Liberty Instruments, Inc. | www.libinst.com