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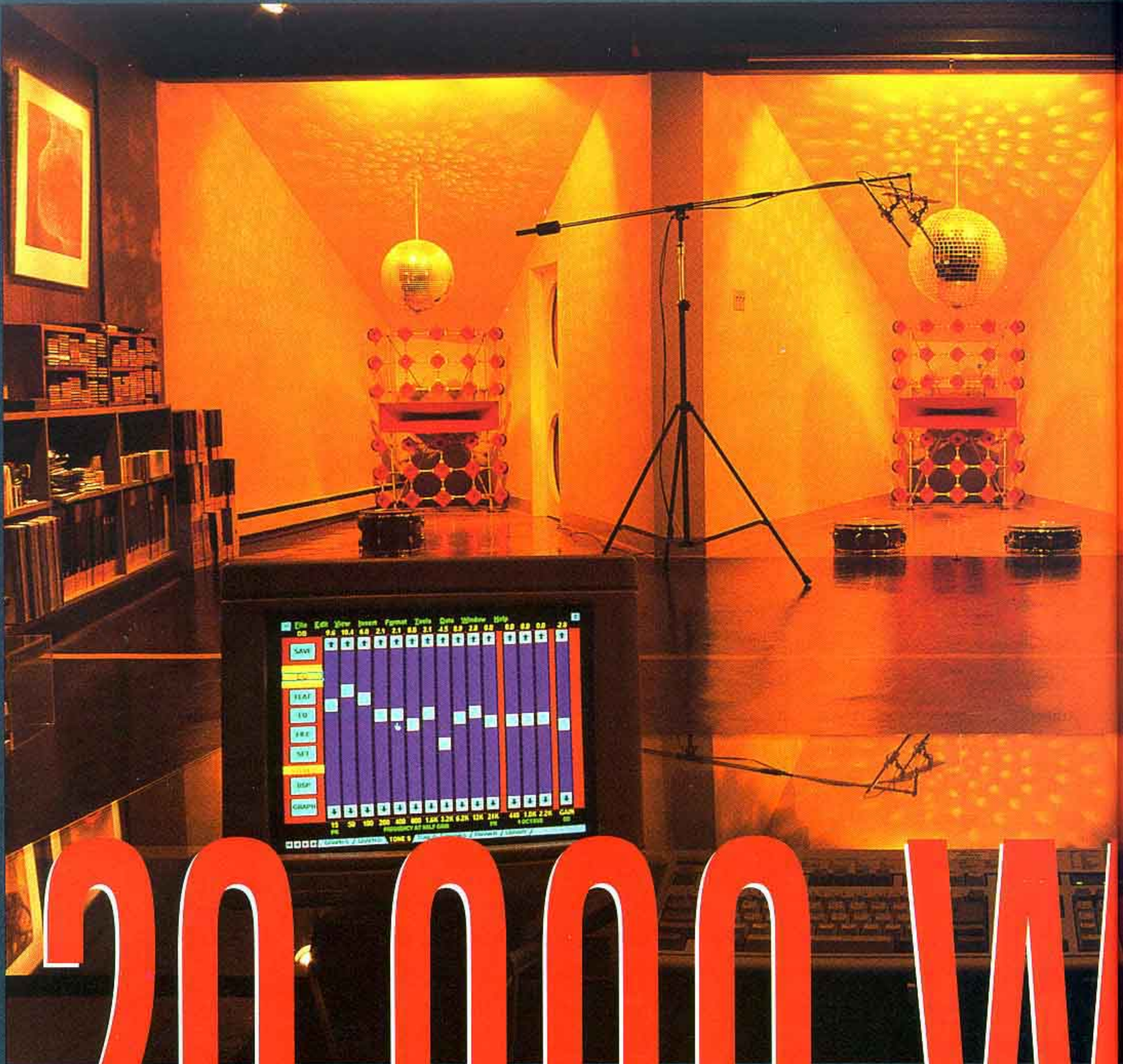

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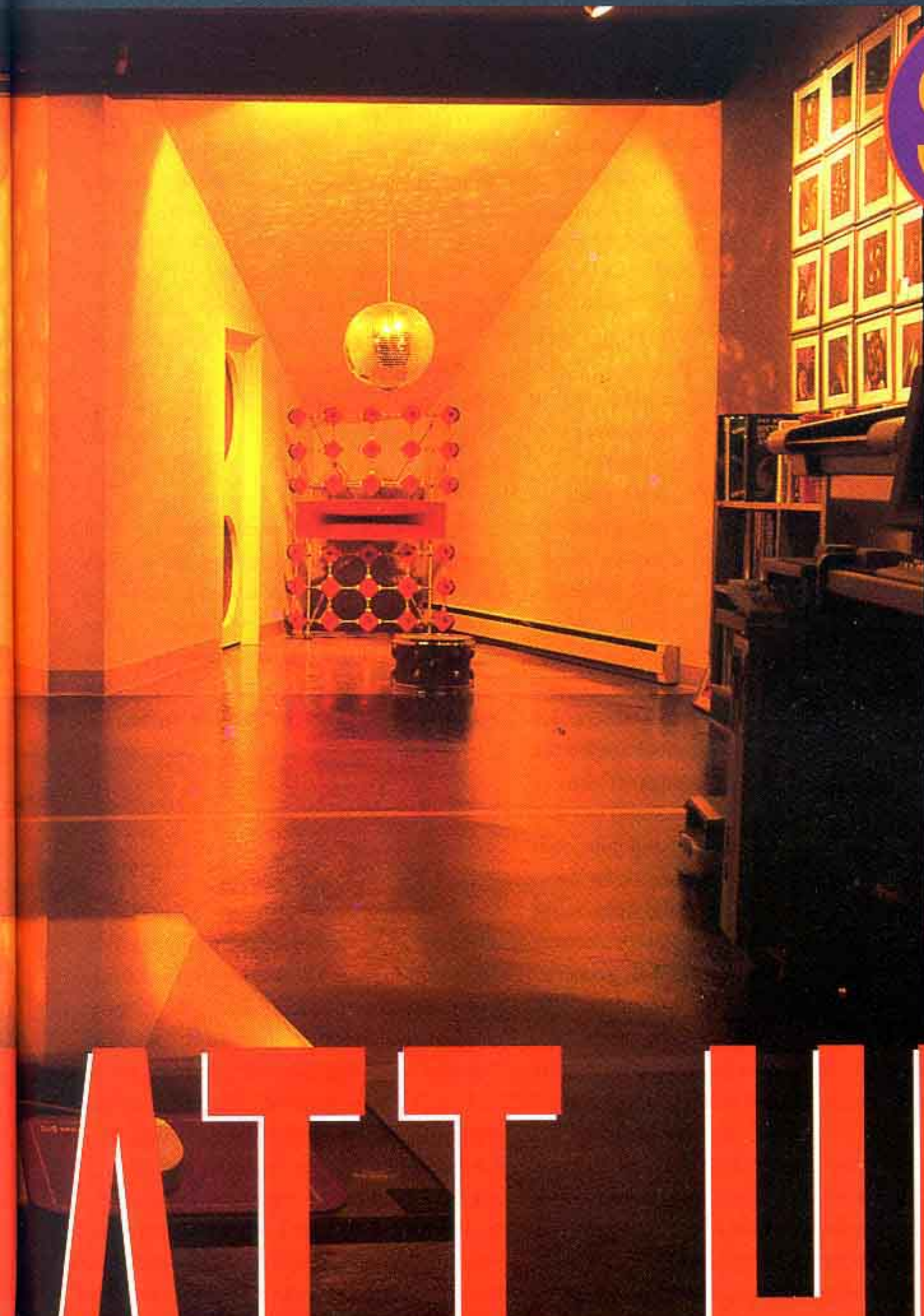

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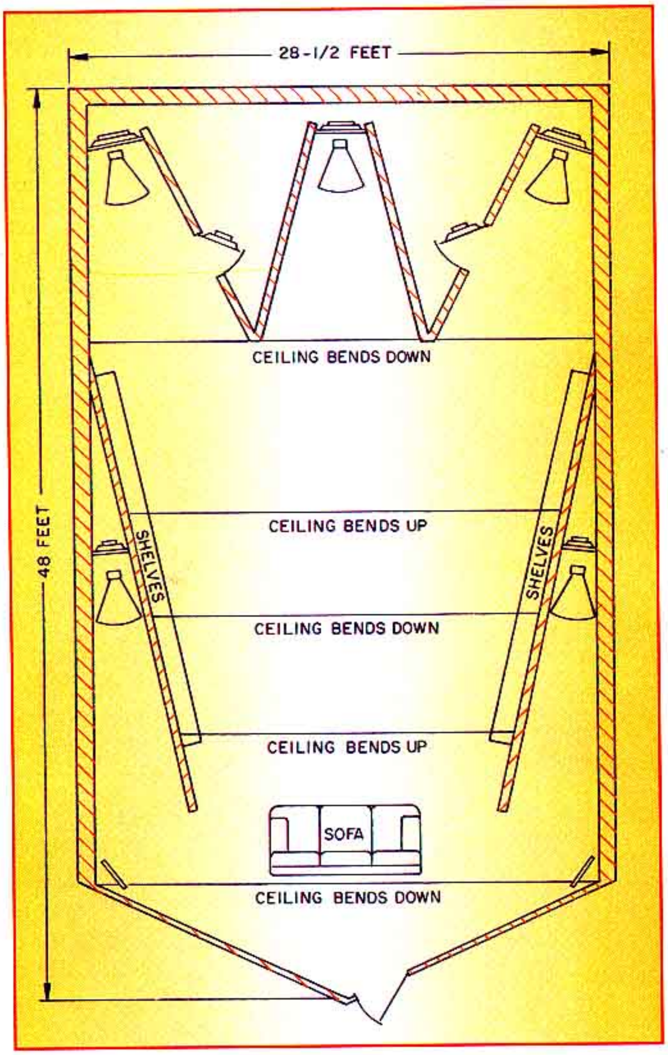
RICHARD S. BURWEN

Since 1981, I have recorded classical and jazz music, both in my studio and in concert halls and clubs, using the Sony PCM-F1 digital format and, more recently, also using DAT and MiniDisc. I have never found a piece of program material—mine or anyone else's—that could not benefit from a bit of equalization. Every decent recording I have made over the past 20 years was equalized during recording. When I play these tapes today, they sound even better with further equalization. The amount of equalization I use varies dramatically with the program material. Different musical selections recorded at the same concert may require different equalization due to orchestral imbalance or changes in seating arrangements.

My audio system has eight tone-control (equalizer) panels and remote units for recording and playback; it was the subject of the cover feature article in the April 1976 issue of *Audio*. Recently I added a new tone-control system using a digital signal processor board in a personal computer. I have found that adjusting the tonal balance (frequency response) is the single most important system parameter in achieving good sound.



My standard for fine sound is *not* the original performance and environment but maximum entertainment value. To me, music is art—and art is entertainment. Anything goes, short of grossly distorting the composer's original intent. I think tonal balance is by far the most important aspect of a sound system. Of course, you also need low distortion of all types, enough power, speakers with smooth frequency response and suitable directional characteristics, and good room acoustics—preferably live acoustics rather than lots of sound absorbing materials. Purists may disagree, but I think pleasing frequency response is what really makes a system musical and, to a large degree, accounts for the audible differences among systems.

Part of entertainment, for me, is lots of bass—not boomy, but the kind that makes your clothes flap in the breeze. I hate screechy sounds and don't care about hearing such details as resin on the bow. I like the well blended sound of a symphony orchestra in a live hall, with different notes seeming to come from different directions, but without the little noises and imperfections of the individual instruments.

Tonal balance is critical to within ±0.5 dB in the range from 200 Hz to 5 kHz. As little as 1 dB of boost from 400 Hz downward can change the "body" of a kettle drum. Tilting the high frequencies up as little as 1 dB above 1.5 kHz can increase the apparent width of the orchestra. Regardless of your own benchmark for great sound, achieving what you perceive as perfection requires precise control of the tonal balance.

Digital signal processing provides this precise tonal balance with far greater flexibility, accuracy, and channel matching than I have ever been able to achieve in 50 years of analog tone-control designs. Instead of launching directly into how I accomplished the digital equalization, let me first provide details on the system.

Sound Studio

My sound studio was designed first, and then the house was designed and built around it. As shown in the floor plan, the room is 48 feet long x 28½ feet wide; it contains five speaker horns, each 13 feet deep with a 64-square-foot mouth. Three of the horns are in the front; the other two horns are along the studio's long sides, facing the rear and delivering reflected sound. Cinder block and concrete make the walls rigid, and the heavy, wavy, plaster ceiling diffuses sound. Acoustically the room is very live. Its nonparallel surfaces produce many standing waves, closely spaced in frequency, while eliminating flutter echoes.

Altogether there are 169 woofers, midrange horns, and tweeters plus an intercom speaker. Each of the five horns contains two 16-inch Empire woofers, a midrange horn with two JBL drivers, and 30 Cerwin-Vega tweeters. In addition, the left- and right-front horns each have two 24-inch Cerwin-Vega woofers, which operate below 50 Hz, while the 16-inch woofers cover from 15 Hz to 400 Hz. The midrange horns reproduce 400 Hz to 6 kHz, avoiding crossover defects in the critical mid-frequency region.

Each woofer, each midrange, and each group of nine or 12 tweeters is driven from one channel of a modified Phase Linear 400 stereo amplifier, capable of 250 watts at the 8-ohm load impedance presented. The front-speaker enclosure room, occupying space behind and between the horns,

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contains 11 amplifiers. Six more amps are located in the side speaker horns. Electronic crossovers ahead of the 34 amplifier channels increase the effective acoustic output to that of a single 20,000-watt amplifier.

Why all that power? Because it takes most of the power available for any one horn to reproduce drums at live levels, leaving only 3 to 6 dB of headroom for the contributions of other orchestral instruments.

The crossovers, incidentally, also equalize the frequency response. I spent a year adjusting the response, in fractions of a dB, so that a tape recording of a drum set, made while the drummer actually sat in my front-center speaker horn, sounded real.

I haven't yet mentioned the lighting power. Four channels of 2,400-watt, SCR controllers light the three front horns in different colors automatically, in response to the music.

My "preamplifier" consists of 7-foot-high relay racks. I have 3½ of these racks, which contain signal-processing and tone-control equipment, mostly of my own design. The racks' bottom sections hold playback equipment, while the top sections

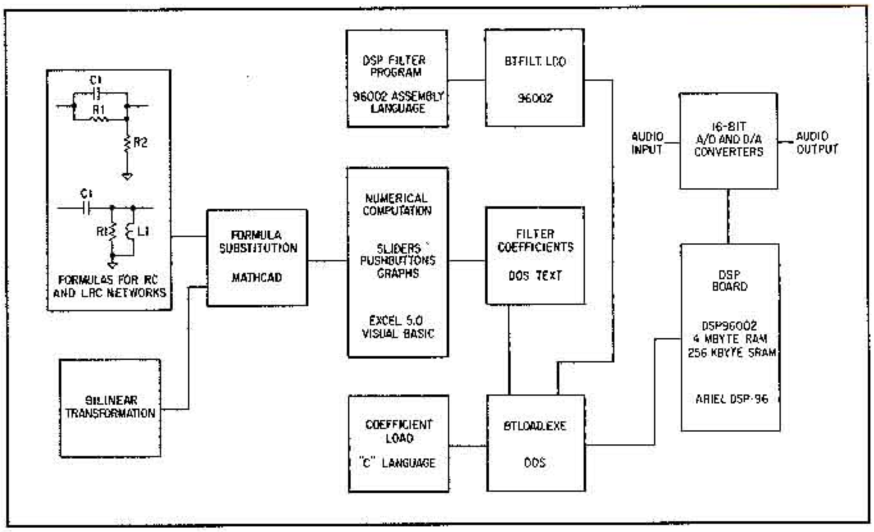

Fig. 1—*Components of the DSP software and hardware system.*

hold recording equipment that can handle live mixing of 52 channels. Each of the 28 recording inputs includes a control equalizer

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with four shelf-type tone switches and two peaking types at the ends of the audio range. The front playback section has two different tone-control panels and two remote tone-control units. The rear playback section uses only two equalizer panels.

Although I started out recording and reproducing four-channel tapes through my five speaker systems, I found in recent years that I can achieve better sound from two-channel recordings. Front sound fed into the rear speakers (directly, via five delays, and via three microphones) provides a concert-hall effect while preserving stereo imaging. The center speaker horn receives a front left/right mix at –13 dB. The new digital tone controls, which I will describe, take the place of the four front tone-control units.

How It Works

All signal processing between A/D and D/A conversion is performed digitally by an Ariel DSP-96 board in a personal computer. This board carries a daughterboard containing 16-bit A/D and D/A converters that receive analog stereo audio and deliver the processed analog stereo output. At the 48-kHz sampling rate, the Motorola DSP96002 chip can perform about 347 instructions in each 20.8-µS sample period. In response to each instruction line, the digital signal processor (DSP) can, in 60 nS, simultaneously multiply two 32-bit floating-point numbers, add and subtract two others, and move two more numbers to new locations. Many instructions are less efficient, involving only one or two operations. The 46 filters and gain changes in this stereo tone-control system use about 80% of the processor's available instructions.

For each tone control, the DSP scales the results from the last few samples and adds them, in combinations, to the value of the current sample. (Scaling means multiplication by a coefficient.) The filters are all infinite-impulse response (IIR) types, in which a portion of the output is regeneratively added to the input, closely approximating the transient, frequency, and phase responses of simple RC and LRC circuits.

Figure 1 shows the component blocks of the DSP software and hardware system. It started with the basic filter formulas for simple RC and LRC circuits. For example, the RC filter in the upper left block is a treble-boost or bass-attenuation circuit, depending on the parts values and specified gain. By substituting a simple formula, known as the bilinear transformation, for the complex frequency variable in the design formula for an analog filter, one can create a digital filter having the same frequency characteristics. Mathcad computer software from MathSoft made the algebraic substitution easy and accurate.

Instructing the DSP what to do requires three software programs, as can be seen in Fig. 1. First is a spreadsheet program using Microsoft Excel 5.0 for Windows; Excel's Visual Basic program section provides mouse-actuated sliders and pushbuttons on the screen. From the positions of the sliders, it calculates the coefficients for the corresponding digital filters, implementing the results of the formula substitution block. (Excel also makes graphs of frequency response.) The coefficients of the digital filters, each with a resolution of 10 digits after the decimal point, are exported to a DOS text file. Then another DOS program, BTLOAD.EXE, which is written in "C" language, scans the coefficients and sends them, one at a time, to the host port of the

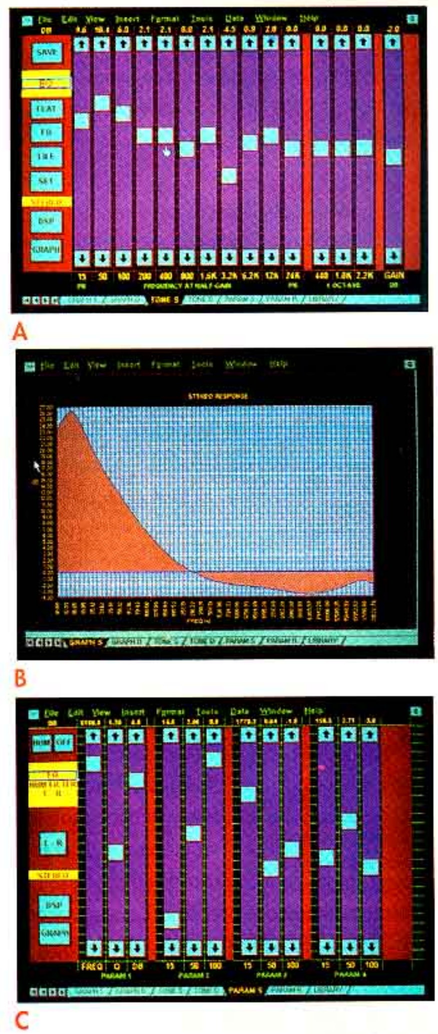

Fig. 2—*Three of seven screens: Stereo tone controls (A), frequency response graph (B), and parametric equalizers (C).*

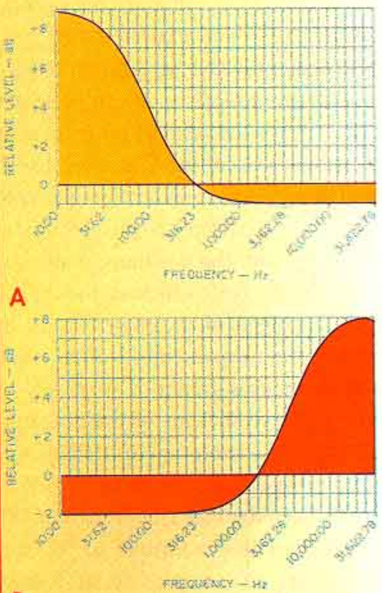

Fig. 3—*Frequency response of shelf-type tone controls: 100-Hz control at +10 dB (A) and 3.2-kHz control at +10 dB (B). Gain reduction keeps loudness nearly constant.*

DSP board. Finally, a DSP assembly-language program, BTFILT.LDD, instructs the DSP to shuffle the numbers among various registers and perform all the required multiplications, additions, and subtractions during each sample period. The net result is that each outgoing audio sample, for either the left or the right channel, winds up being a complicated function of the original incoming sample and of about 40 preceding incoming samples and 40 intermediate calculations.

Tone Controls and Buttons

The system has a total of 54 mouse-actuated sliders and 17 buttons on four display screens. Two more screens show graphs of the left-channel frequency response and the difference between the left and right channels. A seventh, library, screen stores certain button settings and the settings of all the sliders in dB as well as position. Figures 2A, 2B, and 2C show three of the seven screens. The first and principally used screen, shown in Fig. 2A, contains 15 sliders and seven buttons. These sliders are actually scroll bars, as used in familiar Windows programs. The controls in the first screen set the tone for both left and right channels, with a resolution of 0.1 or 0.2 dB and perfect digital matching.

All the tone controls are completely independent and noninteracting. If I set the 15-Hz peaking control at +30 dB and each of the shelf-type bass controls (centered at 100, 200, 400, and 800 Hz) at +15 dB, I get 90 dB of bass boost at 15 Hz! A graph of these settings actually shows a range from +69.5 dB at 15 Hz to –20.5 dB at 10 kHz, because the system automatically adjusts the gain to achieve nearly the same musical loudness with different tone-control settings. At the top of each slider is the setting, in dB.

The left group of sliders in Fig. 2A provides peaking controls at 15 Hz and 24 kHz, together with shelf-type controls operating at octave intervals. A shelf-type control produces a curve that levels off at both high and low frequencies, like the curve shown in Fig. 3A for the 100-Hz control. The name of the control shows the frequency at which half the boost or cut occurs. Each control produces symmetrical boost or cut. The 50-Hz bass control has a range of ±20 dB at d.c. (±17.2 dB at 15 Hz), adjustable in 0.2-dB steps; the 100-, 200-, 400-, and 800-Hz bass controls have a range of ±15 dB, adjustable in 0.2-dB steps. The 1.6-, 3.2-, and 6.2-kHz treble controls are each adjustable over a range of ±15 dB, in 0.2-dB steps. Figure 3B shows the response of the 3.2-kHz, shelf-type control when set for 10 dB of treble boost. Note the 2-dB reduction in gain to keep the loudness constant. All controls affect the gain in varying amounts.

The 12-kHz control, shown in Fig. 4, has half its maximum boost at 12 kHz. Although designed as a shelf type, it looks more like a peaking type with a resonance at 24 kHz. This points out one of the important differences between the analog and digital domains. A digital system cannot accurately reproduce signals above half the sampling frequency without producing aliasing or beat notes. Transforming an analog filter to a digital filter results in squeezing the high-frequency gain curve, so half the sampling frequency corresponds to infinite frequency in the analog domain. The curve, plotted by the software, does not take into account the anti-aliasing filter built into the A/D converter that cuts off high frequencies above 20 kHz with a nearly "brick-wall" response.

The 15-Hz and 24-kHz peaking filters each cover ±30 dB in 0.2-dB steps. Figure 5 shows the combined response for the 15-Hz control at +30 dB and the 24-kHz control at –30 dB. The low-frequency slope is –6 dB/octave. The high-frequency downward slope starts slowly but accelerates to 12 dB/octave at 12 kHz and a bit faster at 20 kHz. (Actually, the 24-kHz peak filter is a

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shelf type that benefits from squeezing its characteristics to below 24 kHz in the digital domain.)

The second group of controls shown in Fig. 2A consists of three one-octave peaking types, similar to those used in graphic equalizers. The center frequencies are 440, 1,000, and 2,000 Hz. As tonal balance is extremely critical in this middle frequency region, each control has small, 0.1-dB steps from –10 to +10 dB. Figure 6 shows the response of the 1-kHz control at +10 dB.

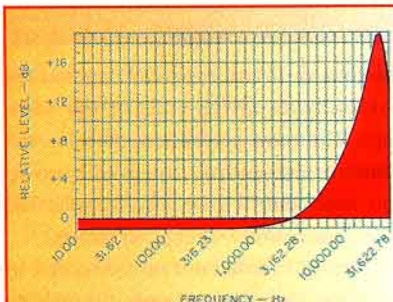

Fig. 4—*The 12-kHz shelf-type control at maximum, +20 dB. High-frequency response is squeezed to below half the sampling frequency, making it look like a peak.*

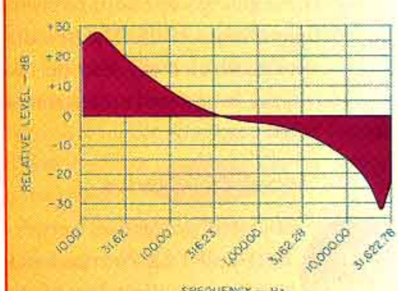

Fig. 5—*Peaking controls, 15 Hz at +30 dB and 24 kHz at –30 dB.*

Note the automatic, 2.5-dB volume reduction. The last control on the right is a gain control having a range of ±20 dB in 0.2-dB steps. It presets the gain to avoid clipping; it is not my main volume control.

At the left in Fig. 2A is a vertical array of buttons. The "Save" button retains both dB and position settings of all sliders on all four screens. Using the "Flat" and "EQ"

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buttons, I can compare my tone settings with flat, unity-gain response. The "File" button recalls the previously saved dB settings from an item on the library screen, allowing A/B comparison with new control settings. The "Set" button recalls the slider positions and dB settings for an item on the library screen.

The "Graph" button creates a plot of frequency response, shown in Fig. 2B, using the dB settings from the selected button. To make the graph, the spreadsheet calculates and adds the gains, in dB, for each of 22 filters at each of 175 different frequencies; it uses formulas as long as 140 characters. (The graphs can also be printed out.)

The "DSP" button controls the one really disappointing aspect of this digital tone-control system. This button sends the computed filter coefficients to the DSP board using the "C" and assembly programs. I had hoped each movement of the sliders could automatically update the filter coefficients in real time at the DSP, but there is too much computation, and even my new computer is too slow. It took 8 seconds on my old 33-MHz 386 machine to change the tone after actuating the "DSP" button. My new 100-MHz Pentium shortened the time to less than 2 seconds. This drawback does not prevent my enjoying the improved sound due to the new digital controls. For demonstrations, I recall library settings; if I want to remix a classical digital tape, I normally rehearse the remix before I decide on final settings.

When I want a difference between the left and right channels, I use a similar set of 15 controls on the second screen. These controls have about ° the dB range of the controls in the first screen and produce only a

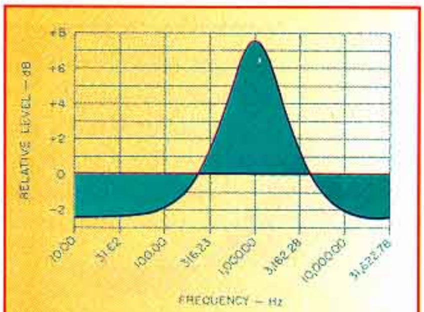


Fig. 6—*Octave control, 1 kHz at +10 dB.*

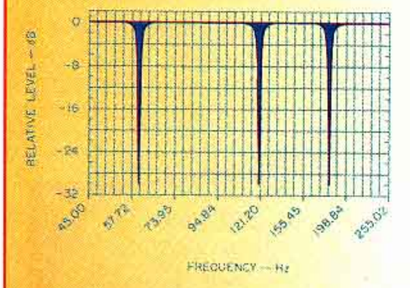


Fig. 7—*The 30-dB hum filter creates notches at 60, 120, and 180 Hz. Filter "Q" is 1,000.*

difference between the left and right channels. The "DSP" and "Graph" buttons are duplicated on this screen. (The "Graph" button on any screen actually makes two different graphs. The first shows the left-channel response; the second shows only differences between the left and right channels, which may be as small as 0.2 dB.)

The third screen, Fig. 2C, provides four parametric equalizers. These controls produce curves similar to the 1-kHz peaking control's (shown in Fig. 6) but with complete adjustability. Each equalizer has three sliders—"FREQ," "Q," and "DB." I can set the center frequency logarithmically anywhere from 10 Hz to 10 kHz, in steps of 2.3%. The "Q" slider sets the curve's sharpness anywhere from 0.2 to 20. Thus, the bandwidth of each filter can vary from five times the center frequency to 5% of that frequency. The "DB" slider sets a peak or dip up to 10 dB, in steps of 0.2 dB. Deliberate interaction between the "DB" and "Q" sliders causes some broadening of the bandwidth at maximum dB settings, so the audible bandwidth effect is nearly constant.

The fourth screen is a duplicate of the third screen and affects only the right channel. Normally an "L-R" button on the third screen causes its control settings to produce identical effects on the left and right channels. The settings on the fourth screen are ignored. By actuating the "SEP" button on the fourth screen, the third screen becomes effective only for the left channel, and the fourth screen adjusts the right channel. Each screen has duplicate "DSP" and "Graph" buttons.

A unique feature of the third screen is a hum filter. This filter, magnified in Fig. 7, notches out 60, 120, and 180 Hz and works in stereo. Each notch attenuates 30 dB, and the loss is only 1 dB at a point just ±3% away from the center frequency. Bass loss is negligible. Each notch filter has a "Q" of 1,000, a value that would be impractical in analog circuits.

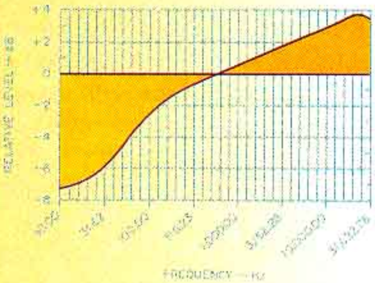
Listening Results

Program material, especially older recordings, varies in the extreme. The improved flexibility compared with my previous tone-control systems means I can make more recordings in my CD collection sound pleasant. More important, the fine adjustment capability imparts a greater degree of perfection to my best-sounding CDs and my own live recordings. Interestingly, although the midrange peaking and the parametric controls can provide any kind of a peak or dip—however broad or narrow, at any frequency—my saved control settings show that after trying these highly adjustable controls, I used them for only six out of 87 recordings or radio programs.

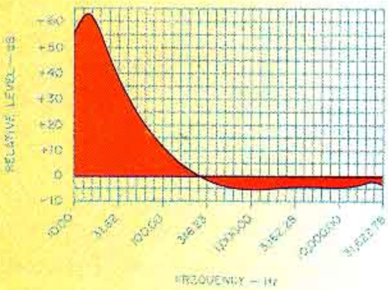
I found that once I had achieved a fairly pleasing balance on each recording, I continued to make fine adjustments, in steps of only 0.2 dB, for various controls. I do not notice the effect of moving a single tone control by 0.2 dB, but I can hear the cumulative change in balance from moving two or three mid-frequency controls by a total of 0.4 or 0.6 dB. Left/right balance is extremely critical, and I adjust the right-channel gain relative to the left in steps of only 0.1 or 0.2 dB.

It is far easier to adjust both stereo channels with a single control than with two. The gradual, completely noninteracting controls seem to produce much more desirable curves than the resonant types produced by conventional graphic equalizers. There is no problem of lack of transparency due to resistor, capacitor, or potentiometer

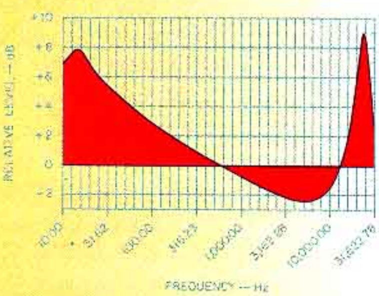
Fig. 8—*Pleasing response curves for selected program sources.*



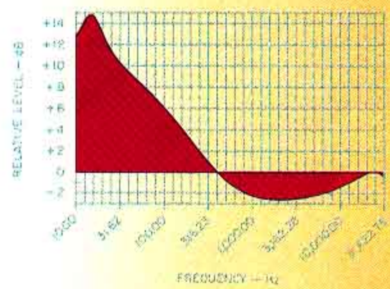
A *CNN broadcast, with a tilt up that makes voices more intelligible.*



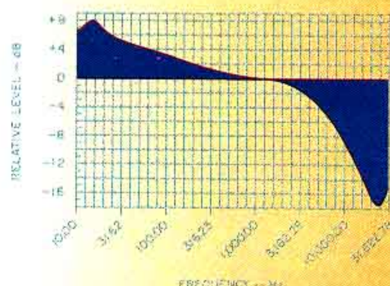
B La Traviata *on LaserDisc, where lost low bass is retrieved by 68-dB boost at 15 Hz relative to 1 kHz.*



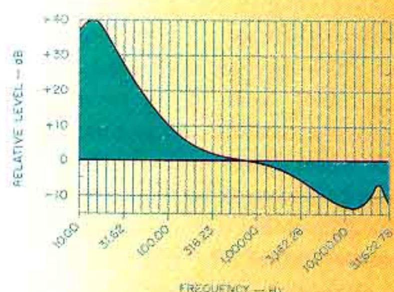
C *November 1986 concert of Civic Symphony Orchestra of Boston, on PCM-F1 format tape, where the slope is −1 dB/octave from 25 Hz to 3 kHz.*



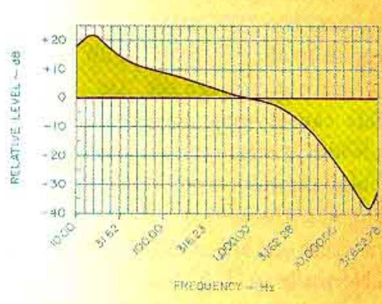
D *A passage from* Bob's Diner *(dmp CD-471), a very high-quality CD of a band led by Bob Smith.*



E *A passage from Merle Haggard's "I'm Always on a Mountain When I Fall" (from the CD of the same name, MCA MCAD-1644), where irritating high frequencies are rolled off 17 dB.*



F Moonraker, *a James Bond VHS videotape, is more exciting with 40-dB boost at 15 Hz.*



G *CBS telecast of 1994 Tony Awards, where 22-dB bass boost and 38-dB high-frequency roll-off tame sibilant voices and gross imbalance.*

tolerances, as in many conventional equalizers. I can decide on the settings of the 12- and 24-kHz controls when their effect is as little as 0.5 dB at 10 kHz. Once the controls are set close to optimum, I find that small changes of 1 or 2 dB make a noticeable difference in the apparent horizontal spread of the sound. Less high-frequency gain causes the sound to come from between the left and right speakers. More gain extends the spread a little wider than the speaker placement; too much extreme high-frequency gain may make the sound come apart or become irritating. Of course, I am using all five of my speaker systems, with nearly as much sound from the rear as from the front.

Figures 8A through 8G show the frequency response curves corresponding to my saved control settings for various types of program material. They vary from an 11-dB upward slope for a CNN news broadcast (Fig. 8A) to an amazing 68-dB, low-bass boost (Fig. 8B) for the movie *La Traviata*,

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on LaserDisc. This Verdi opera, directed by Franco Zeffirelli in 1982, has an analog soundtrack containing a trace of low bass that can only be retrieved using that much boost. Because the 23-dB boost at 60 Hz brings up the hum level, due to the 30-foot ground path to my LaserDisc player, it is necessary to use the hum filter with this recording.

The ability to make fine settings of non-interacting, shelf-type curves allowed me to produce a very gradual downward slope of 1 dB/octave (shown in Fig. 8C) for one of my own recordings of a live concert. High frequencies above 8 kHz needed boost because the high-frequency directionality of the omnidirectional microphones, pointed at the ceiling for added reflections, caused a fall-off in the direction of the orchestra. My tone settings, however, are based entirely on what I hear.

Most curves turn up at very high frequencies, but some do not. The telecast of the 1994 Tony Awards (shown in Fig. 8G) seemed quite unbalanced, with extremely sibilant voices, causing me to roll off the high frequencies by 38 dB. Again, I preferred a smooth downward curve to a sharp cutoff or one with peaks or dips within the audible band. Based on what I hear via this extremely flexible yet finely adjustable tone-control system, it appears that, contrary to popular opinion, flat audio systems are optimal for only a tiny fraction of available program material.

Pure Sound

Purists who favor having only minimal equipment in the signal path may wonder if the sound of my system is muddied by its digital processing and the nearly 2,000 operational amplifiers in various pieces of equipment. When my system is set for unity gain and flat response, I can switch 20 or more op-amps plus the DSP system in and out of the signal path. I hear no difference when playing my cleanest recordings, and only a slight increase in noise when there is no music.

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Some of the op-amps in my system have been in use for 70 hours a week for 28 years. These discrete component devices amplify audio with low noise, low distortion, and high slew rate, and they are still in a class with today's best.

Aside from the added digitization, which can be eliminated for digital sources, DSP does tend to minimize equipment in the signal path. All that is added is pure mathematics. When enough bits are used, the mathematical errors in frequency response and channel matching can be hundreds of times smaller than those of analog circuits, producing complete transparency. Too few bits, however, can produce peculiar distortion components, beat notes, and signal-related noises.

One of the surprising results of digital filtering was the amount of round-off noise. I had read that tiny random errors from rounding off the multiplication products, from 64-bit numbers to 32-bit floating-point numbers, might result in some generated noise. I was astonished to find that with low bass boost, the noise level (peak to peak) reached 0.3 V out of 4 V full scale, only 23 dB down. It sounded like a screechy hiss that came and went at certain d.c. input levels. For a.c. input at 15 Hz, there was a fluttering effect as the signal modulated the noise. Higher frequency sine waves were cleaner, and above 50 Hz the noise was inaudible.

When I had two free hours before a scheduled demonstration, I traced the noise to the 15-Hz resonant filter. This digital filter has a denominator gain of 250,000 and tremendously amplifies the tiny round-off noise. In the remaining time before the demonstration, I was able to convert the resonant filter to a shelf type having a gain of only 500, thereby eliminating the audible noise.

Later I learned that the DSP96002 processor can compute with 64-bit accuracy instead of 32-bit simply by changing ".s" to ".x" in floating-point instructions. The complete elimination of round-off noise from all the controls (including the 15-Hz and parametric resonant types), leaving only converter noise, made a remarkable demonstration.

I would like to have a direct digital input and A/D and D/A converters whose dynamic range is 115 dB. Recently I installed three additional 40-MHz Ariel DSP96 boards. The faster DSPs allow more instructions, and they interface with two Ariel Model 656 Proport external A/D and D/A converter boxes for my front and rear channels. The Proports' differential inputs reduced hum, and the external box reduced digital noise. Still, the dynamic range is slightly less than that of the 96-dB concert recordings I have been making for the past 13 years in the PCM-F1 format.

It is inconvenient to operate the main computer from where my guests are sitting. Therefore, I have connected my laptop computer via a network coaxial cable and can use it as a remote controller from anywhere in the room.

With additional DSP96 boards, I am currently working on my second DSP project—reverberation. I want to see if I can produce an electronic "space" that sounds better than my favorite acoustic space, Jordan Hall at the New England Conservatory of Music in Boston. Now that I've seen what digital technology can do for me in other areas, I think I have a shot at it.