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# YAMAHA DSP-3000 DIGITAL SOUND FIELD PROCESSOR

**Manufacturer's Specifications**

**Analog Inputs and Outputs:** 2.5 V rms maximum.

**Analog Output Gain:** 0, +0.5 dB.

**Digital Input and Output Levels:** 0.5 V peak-to-peak.

**Sampling Frequencies:** 32, 44.1, and 48 kHz, with automatic selection.

**Video Input and Output Levels:** 1 V peak-to-peak.

**A/D Converter:** 16-bit linear quantization with 48-kHz sampling frequency, independent stereo channels, and internal dither circuitry.

**D/A Converter:** 18-bit (Main) and 16-bit (Effect) quantization.

**Processing Programs:** 35 preset and 20 user-set.

**Harmonic Distortion:** 0.002% on Main outputs and 0.005% on Effect outputs with analog input; 0.003% on Main outputs and 0.005% on Effect outputs with digital input.

**Frequency Response:** 10 Hz to 100 kHz for Main and 20 Hz to 20 kHz for Effect with analog input; 20 Hz to 20 kHz, ±0.5 dB, for both with digital input.

**S/N Ratio:** 110 dBA Main and 94 dBA Effect with analog input; 110 dBA Main and 105 dBA Effect with digital input.

**Channel Separation:** 80 dB at 1 kHz with analog input, 90 dB with digital input.

**Power Requirements:** 120 V a.c., 60 Hz.

**Power Consumption:** 45 watts.

**A.c. Outlet (Switched):** 300 watts maximum.

**Dimensions:** 17⅛ in. W × 3¾ in. H × 13⅞ in. D (43.5 cm × 9.55 cm × 35.2 cm).

**Weight:** 21.1 lbs. (9.6 kg).

**Price:** $1,899.

**Company Address:** 6660 Orangethorpe Ave., Buena Park, Cal. 90620.

For literature, circle No. 90

When Yamaha introduced the DSP-1 digital sound field processor, I was among many that marveled (June 1987) at what it accomplished for the listening experience. Because of the great sophistication of that unit, I forecast (to myself) that the next unit would be less complex at a lower price. The DSP-3000, however, is more sophisticated in a number of respects, and the price is roughly twice as high.

Let's take a look at the features with attention to the changes made. The new Yamaha processor offers 20 sound fields with a total of 35 variations. There are 17 new environments, including concert halls sampled in several countries. There also are two new presence modes and a new surround program. There are four new movie-theater modes which simulate the effects of commercial movie theaters. The self-descriptive program names are "Adventure," "Classic," "Musical," and "Standard."

The master volume control changes output level on all channels simultaneously with the use of the remote control or a rocker-type switch on the front panel. (The DSP-1 requires external means such as the Yamaha MVS-1 for such control.) An internal pink-noise generator can be switched on for setting system balances. This is a great convenience when setting up, and it is always available if a recheck is needed.

There is a video input/output loop to superimpose program parameters and other function readouts on the screen of a TV monitor. There are menu for this program to set the preferred type and time-duration of display and one of the nine background colors. If video is fed in, the background color disappears, and the display is superimposed in white. The DSP-3000 can select among two analog and a direct digital input. The digital input allows direct effect processing of CD or DAT signals and eliminates one stage of analog-to digital conversion. There are also the obligatory tape recorder input/output connections and monitor switch.

Yamaha's proprietary Hi-bit, floating 18-bit approach combines with a dual-converter configuration for improved signal-to-noise ratio and dynamic range, and lower distortion when the direct digital input is used. The digital processing of the DSP-3000 uses four-times oversampling digital filters for improved time-base resolution, phase coherence, and transient response. The four effects channels and the two main channels use one filter each. The main-channel filters are activated only when the digital input is used.

The new processor has a front-panel program-stepping switch which provides some convenience. The remote control selects any basic program directly and allows making the great majority of possible changes from the listening position. The DSP-3000 contains stored acoustic data based on a number of different performance environments. An original Yamaha VLSI (Very Large Scale Integrated) circuit chip, operating in real time, calculates dozens of discrete early reflections based upon this data. Each of the Yamaha YM-3818 VLSI chips used in the DSP-3000 incorporates a high-speed multiplier and an adder and subtractor. These enable the DSP-3000 to produce up to 88 discrete reflections, 22 for each of the four effect channels. Figure 1 is a block diagram of the processor.

The digitally processed delays create time lags between the sound arrivals from the main speakers and the arrivals

**Fig. 1—Block diagram. Note that all surround processing takes place in the digital domain.**

from the effect speakers. These delays, in the relatively small listening room, are the same as those between the direct sound and the reflections from the walls in a concert hall or other venue. The generated sound field removes the boundaries of the home listening room, as it were, and replaces them with the characteristics of the performance hall. The processor offers a wide variety of possible fields by providing control over many of the parameters involved in the synthesis. It is easy to vary such things as "liveness," initial time delay, and reverberation level over wide ranges for the most satisfying home listening experience.

## Control Layout

The push on/off "Power" switch is at the lower left of the front panel. Above it is a display panel that extends from the left end almost to the middle. At the left of this display is the receptor for the remote control. To the right of that are the red-LED "Mute" indicators for "Main" (top) and "Effect" (bottom). The LEDs are not large, but the mute status can be seen at least 25 feet away. The separation between them prevents confusion as to which function is muted. Just to the right of the LEDs are the yellow annunciators for "Preset" (top) and "User Prog" (bottom). They are not easily read at a distance, but relative position shows which function is being used.

Further to the right is the bright, yellow-LED program-number display. The numbers are large enough to be read over any normal listening-room distance. They immediately dispel any doubt about which program is in use. Last in the display panel is the large, 2-line by 16-character LCD display. Its alphanumeric characters are gray on a white background and are quite easy to read at normal distances. The

**Four-times oversampling digital filters are used for better phase coherence, time-base resolution, and transient response.**

default mode shows the program name on the top line and the first changeable parameter below. Pushing buttons on the remote control causes this display to report, at least momentarily, what has happened. I will give more detail on this very useful feature later on.

Just to the right of the display panel are the "Input Selector" switches (from left to right, "Digital/1, 2, 1") and the "Master Volume" rocker. There are bright yellow LEDs above the left end of each Input Selector. Pushing any of these switches gets a 2- to 3-S display of the selection made. Push "Digital/1" without an actual digital signal, and the LCD display shows that the DSP-3000 has automatically switched to analog input 1, instead. (It makes this decision even if the analog 2 input was in use before.) This is a very minor perturbation, in my view, considering the advantage of the automatic decision.

The volume rocker has "Down" printed above its left end and "Up" above its right. With a push on either end, "Volume Level" is displayed with a row of up to 28 small, vertical bars on the second display line. (No bars are shown at the zero-volume setting.) The user needs to keep in mind that this horizontal bar graph shows the setting of the six-gang, motor-driven volume control: It does not show the actual signal level within the unit. I really like being able to control all output levels at the same time, and the status display makes this feature even more convenient.

Below the Input Selector switches are the "Tape Monitor" switch and the down/up "Program" rocker. A red LED indicates "On" for the monitor switch. Changing the switch position results in a momentary status display each time. Holding in the Program rocker steps the programs up or down at about three per second. Going below "1" of the "Preset" programs calls up "20" of "User Prog"; going above "20" of "Preset" calls up "1" of "User Prog." (Some presets, as we'll see later, have multiple modes, making a total of 35 preset programs.) All of the above button switches and rockers have good tactile and audible clues with actuation, although the monitor switch has a soft sound.

The back panel has 24 gold-plated phono-jack input/output connections. From the left, there are stereo (L/R) pairs for "Analog Input" (two sets for "1" and one set for "2"), "Tape" ("Tape PB" and "Rec Out"), "Main," and "Processing" ("Front" and "Rear") outputs. The two sets for Analog Input 1 allow looping the signal through to other equipment. Above the Main output jacks is a "Main Level" slide switch with "0 dB" and "−10 dB" positions; this can be an aid in getting the desired system balance. (I have been using the −10 dB setting most of the time with the original DSP-1, which I use as a reference system.)

A "Front Mix" slide switch above the "Front" jacks selects "4 ch" or "6 ch" to match the system configuration. The normal system has four separate effect channels, in addition to the two main channels. When the system will have just two effect speakers, "4 ch" is used to get a mixing of effects into the main stereo speakers. In this fashion, a good part of the created sound field is maintained even with the compromise.

In the center of the back panel, from left to right, are four "Output/Mono" jacks ("Front," "Right," "Left," and "Rear") for reinforcing the lower frequencies. Each output has a level control and a low-pass filter with slide-switch settings of "80," "150," and "5k" Hz. The pot knobs are very small, but knurling makes them easy to turn. As Fig. 1 will show, appropriate effect channel outputs are summed to feed each of the mono outputs: Front Left and Rear Left feeding Left, for example. Front, however, is also fed from the left and right main channels, as well as from Front Left and Front Right effect channels.

To the right are the "Digital" "In" and "Thru/Out" jacks. This configuration allows sending the digital signal from a CD or DAT player to other equipment, as well as to the DSP-3000. (The processor's power does have to be on for feeding through.) "Video" "In" and "Out" jacks allow similar looping through, but in this case, power does not have to be on. There is superimpose circuitry under the unit's control for TV-monitor display of programs and any other material that would appear on the front-panel LCD display. The back panel also has a switched a.c. outlet which will handle up to 300 watts; this is quite high, and much better than on many other units.

I removed the heavy top cover to get a look at the internal construction and found that two side-by-side sheet-metal covers remained. I took off the one that covered the power supply and the majority of the circuitry—mostly digital. The three Yamaha YM-3818s are quite apparent from their large size and grouping on the excellent p.c. board. The layout is very neat and clean, and parts and functions are well labelled. The transformer was hot to the touch—but not to the point of being painful—after hours of operation. It is well encased in a heavy cover, and I did not notice any ventilation paths. I could see why the transformer would be on the warm side, but I could also appreciate that the construction would minimize any coupling and radiation problems. A sheet-metal cover/shield enclosed the analog circuitry, and I did not remove it. The chassis construction was very rigid, even with the top covers removed.

**Remote Control and Programs**

Operating the Yamaha DSP-3000 is best understood by discussing the remote control, the sound-field programs, and other functions. The remote control is not heavy but it is larger than most. The wide power on/off button is the first one at the emitter/transmitting end of the control; a white-on-red label next to it catches the eye. Next is a row of three "Input" selector buttons, and then a row of three more buttons: "Memo" (labelled in red) is used for enabling the system to put user-generated parameter values into one of the user-program positions. To the right, "Preset" and "User" (in white) select the class of program. Pushing either button always gets the program that was last used under that category.

The next four rows, with five buttons each, select programs identified by name and number. Each button has a white number on its face, and above each button or group of buttons is the designation in gold lettering. The first row of five buttons are all designated "Concert Hall": "1" gets "Hall A (or B) in Europe"; "2," "Hall C (or D) in Europe"; "3," "Hall E (or F) in Europe"; "4," "Hall G (or H) in U.S.A.," and "5," "Live Concert A (or B)." The first listing, in each case, is the default choice; the Parameter decrease or increase button

The generated sound field replaces the boundaries of the home listening room with the characteristics of the performance hall.

(discussed below) is used to get the second choice for these or other programs.

Buttons "6" to "10" are in the second row: "6" selects "Opera," with "Balcony" and "Mezzanine" choices; "7," "Cathedral"; "8," "Church," and "9" and "10" select "Jazz Club" "1" and "2," respectively. Jazz Club 1 offers "Village Vanguard" and "Village Gate," based on acoustical data from those two New York City clubs. Jazz Club 2 has "Cellar Club" ("small and cozy") and "Cabaret" ("fuller, richer sound").

The third row ("11" to "15") selects "Chamber," for chamber music, and "Rock Cnct," which provides "The Roxy Theatre" of Los Angeles and "Arena." Next is "Disco," with "New York" and "Tokyo" based on locations in those cities. "Pavilion" is for recreating the sound field of a multi-purpose enclosed pavilion, and "Stadium" selects the sound fields of "Anaheim Stadium" and "Bowl."

The fourth and last row of program buttons ("16" to "20") includes: "Presence A (or B)" for a close-up effect; "Surround A (or B)" for a feeling of being surrounded by performers and the sound; "Movie Theater" "1" and "2" ("18" and "19," respectively), which are synthesized modes for "Adventure" and "Standard" ("1") and "Musical" and "Classic" ("2") movies. Last is the standard Dolby Surround mode, labelled with the double-D symbol plus "Sur."

Beneath these program selection buttons is a row of four white-labelled "Parameter" buttons: "Down," "Up," "Dec," and "Inc." "Down" and "Up" move selection in the parameter menu. "Dec" and "Inc" decrease or increase the value of the selected parameter. Below this row, on the left side of the control, is the "Title Edit" button, which selects the mode to generate an original title up to 16 characters long for any user program. Upper- and lower-case letters, plus numbers and symbols, are available. I didn't take advantage of this feature, but it would be very nice for some users.

The "Utility" button, next below, brings many desirable functions under its rather dull name. Two pushes, while in any program, put the display in "Bit Monitor" mode, and the level status of the incoming signal is displayed in terms of the number of bits that can be extracted from the highest levels. With the level of the source adjusted for "16 bit," the user knows that he is getting all that's possible in this regard. The lowest level indication is "<13 bit" and the highest is "Full," which calls for a reduction back down to "16 bit." "Utility" also accesses the menu for "Display Control for Superimpose" to define the TV monitor display, and enables system balancing in combination with "Preset" and the built-in pink noise source. (The measurement section of this profile will provide more details.)

To the right of "Title Edit" and "Utility" are the "Effect" level buttons: "Balance" ("Rear" and "Front") and "Level" ("Down and "Up"). A push of any of these four buttons displays the existing balance or level and the change while holding the button. The final setting is displayed for about 3 S after the button is released. Below are the two large "Master Volume" buttons, "Up" and "Down." A push of either displays "Volume Level" and its horizontal bar graph. To the left of these are the "Main" and "Effect" "Mute" buttons. As mentioned earlier, actuation of a mute mode turns on a red LED on the front panel.

**Measurements**

First, let me point out that all measurements were made after completing the listening tests. When I stood straight out from the DSP-3000 and pointed the remote directly at the front panel, the effective range was greater than 27 feet. At 10 to 15 feet, the remote position could be off axis up to 80° in the horizontal plane and at least 30° up or down from the horizontal axis. The pointing of the remote was actually noncritical. The LCD display could be read at 15 feet or more and up to 45° off axis horizontally. The highest contrast of the display was when looking at it in the same horizontal plane or from slightly higher. There was less contrast when viewing it from a lower angle.

The bit monitor displayed "13 bits" with 0.146 V at the input, "14 bits" with 0.295 V, "15 bits" with 0.594 V, "16 bits" with 1.214 V, and "Full" with 2.440 V. Clipping in the main output appeared with 7.34 V. These figures apply from 20 Hz to 1 kHz: With increasing frequency above 1 kHz, there was increasing reduction in the input voltage for any number of bits. By 20 kHz, for example, 16 bits was reached with 0.442 V. The reductions appeared quite acceptable in comparison with the spectral content of actual music. With a 1-kHz tone burst, it was possible to reach clipping without turning "Full" on. The clipping point, however, was greater than 10 dB above where "16 bit" appeared with the same burst.

The frequency response of the main channels was down 0.05 dB at 20 Hz and 0.4 dB at 20 kHz. The −3 dB points were at 1.7 Hz and about 80 kHz. The output levels were −0.8 dB, relative to the input for left, and −0.7 dB for right. The harmonic distortion for the main channels was 0.002% at 1 kHz. (Frequency response and distortion tests cannot be run on the effect channels because their responses are purposely modified internally. However, I heard nothing from these channels which I could classify as distortion or frequency-response errors.)

Noise in the main channels was more than 100 dBA below 1 V for any position of the volume control. The front- and rear-channel output noise was 88 dBA below 1 V with the volume control at maximum and 100 dBA below 1 V with the volume at minimum. The output impedance was 966 ohms, and channel separation was 79 dB at 1 kHz. Spectrum analysis of the six outputs showed no evidence of a 48 kHz residual or any sidebands from a high-level 1-kHz test tone. All such components were at least 87 dB below the level of the test tone.

The tracking of volume control for the two main channels was within ±0.1 dB over the range from 0 down to at least 65 dB of attenuation—much the best that I have ever seen. With a little practice, I was able to set any exact level I wanted within ±0.1 dB for up to 35 dB of attenuation. There is no need to be that precise, of course, but I have had frustrating experiences with other motor-driven pots that I could not set even roughly close. The two front-channel volumes tracked each other within 1 dB over the whole range, which is excellent. The two rear-channel volumes tracked within 1 dB for about 50 dB of attenuation, which is quite good. The effect-channel volumes tracked the main-channel volume within 1 dB for about 45 dB, which is really very good for the six sections involved.

**Obviously, you run this from your easy chair, since the remote control has 43 buttons and the faceplate carries only nine.**

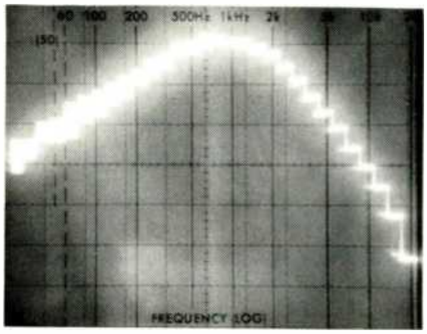


**Fig. 2—Balance test signal generated by the DSP-3000. Though called "pink noise," its frequency content is actually optimized for speaker balancing. See text.**

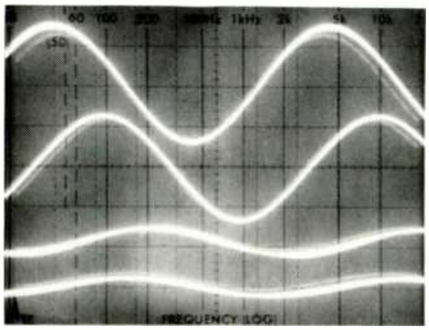


**Fig. 3—Output from effect channels in Movie Theater/Adventure mode, with 742.6-Hz test tone. Traces are (top to bottom): LF, RF, LR, and RR. Changing the test frequency would change the phase and amplitude relationships between the channels. (See text.) Vertical: 2 V/div.**

The effect-level and balance displays each had 10 vertical bars, one for each 10%. The balance display had a double bar right at the 50% point. A check of the Dolby Surround input balance demonstrated that the best setting for the minimum sound to the surround speakers with a mono input was with the Dolby input balance at 54% to the right. All of the effect levels could be changed in 1% steps.

The results of level tests of the DSP-3000's mono outputs were a little confusing, but the majority of times, the Left, Right, and Rear output levels were about 7 dB below the power total for the two summed channels. The Front output level, with its contributions from four channels, was more variable but usually was at least 4 dB below the total from the sources. My own judgment was that these levels might be too low to drive some amplifier/subwoofer combinations. A check of the memory function showed that effect balances and levels were saved but not the overall volume.

Figure 2 presents the ⅓-octave spectrum of the DSP-3000 test-noise output, which the owner's manual refers to as "pink." If it were truly pink, the response would be flat. (Personally, I would prefer that the noise be flat for response comparisons.) The purpose of the noise source, however, is to facilitate setting levels, so peaking the noise at frequencies where most speakers work quite well might be better, in some cases, than true pink noise. The level of the noise at the main outputs was 25 mV. Figure 3 is just one example of how the four effect-channel outputs can differ from each other. Notice that the test-tone frequency is stated quite precisely as 742.6 Hz. Just small changes in frequency caused noticeable shifts in level and relative phase in the four channels, compared to what is shown here.

Parameters for the various programs include such elements as room size, liveness, initial delay, reverberation time, reverberation level, and settings for high- and low-pass filters. Simple stepping tests demonstrated the excellent resolution of parameter values. "Room Size" is adjustable in 40 steps, from 0.1 to 4.0, and "Liveness" has a range from 0 to 10 in steps of 1 (both in arbitrary dimensions). "Initial Delay" can be set in 1-mS steps from 1 to 150 mS. "Reverberation Time" has a range from 0.3 to 10.0 S, with 0.1-S steps. "Reverberation Level" can be set from 0 to 100% in 5% steps. The high-pass filter can be set for "Thru" (flat) or in ⅙-octave steps from 32 Hz to 1.0 kHz. The low-pass filter can be set for "Thru" or in ⅙-octave steps from 1.0 to 16 kHz.

Parameter values could be stepped with a series of pushes on "Dec" or "Inc." Holding in either of these buttons caused a rapid changing in value after a second or two. All of the programs have preset values which are protected under the "Preset' function. Any combination of original and modified parameters can be saved as a "User" program. User-program memory is maintained by a special long-life backup battery which should last about five years. If the battery voltage is getting low, "** Warning ** User Mem. Error" appears in the LCD display when the unit is first turned on. Yamaha states that a qualified service center should replace this battery. They also recommend that the user fill in the manual's program parameter tables to ensure that important program information is not lost.

## Setting Up

Yamaha makes specific recommendations on the listening room and the placement of the loudspeakers. They state that the sound-field creation is best if the room is "as acoustically dead as possible," which really calls for much more surface absorption than it makes sense to have. However, the manual does mention normal means to keep the

**There are programs within programs, so 20 buttons can select 35 factory-set and 20 user-set simulated acoustical environments.**

room from being too live. For one thing, it states that the main speakers should be three to six feet from the front wall, with the front effect speakers a few feet above and behind them. However, the user's main speakers might need to be closer to the wall for good bass performance.

It is probable that most users will not be able to meet all of Yamaha's criteria. Having said that, let me reassure the reader that perfection of equipment, its arrangement, and the acoustics of the room are *not* essential for great listening. The six-channel arrangement, however, is noticeably better than the four-channel arrangement and a center speaker and subwoofer are very desirable, in my view.

Figure 4 shows the arrangement of the evaluation system that I have been using for surround-sound systems of any type. The Yamaha DSP-1 is the reference processor. To help in making comparisons, all of the in/out connections for the processor are normalled through a jack field which allows for easy insertion at all nine of the DSP-3000 inputs and outputs shown. A Yamaha AVC-50 serves as the preamplifier and the main amplifier. Other equipment includes Magnavox and Pioneer CD players, a Dual turntable, a Sanyo Beta VCR, JBL main and center speakers, a Lafayette center-channel amplifier, Dynaco effects speakers, a Triad subwoofer, and a Yamaha FM tuner, video-disc player, and effects-channel amplifier. My VHS VCR with MTS failed at the start of the evaluation, so I picked up a Realistic TV-100 TV-sound receiver at the local Radio Shack. I used a Radio Shack Archer r.f. modulator on the video output of the DSP-3000 to show the superimpose function on my TV set.

### Use and Listening Tests

As stated earlier, I did all of the listening before any measurements. The owner's manual has 64 pages of helpful and interesting information. The format is open, and the large type and illustrations make for very easy reading. However, discussions on room acoustics, speaker placement, and program parameters and their effects would benefit from more detail. The section on adding auxiliary speakers never states what the four mono output signals (Front, Right, Left, and Rear) really consist of. It would be easy to assume, for example, that Front is simply a mono summing of the main channels, but the summing also includes the front effects channels. The actual combinations are clear in the block diagram at the back of the manual, but at least a few words are needed in the earlier text.

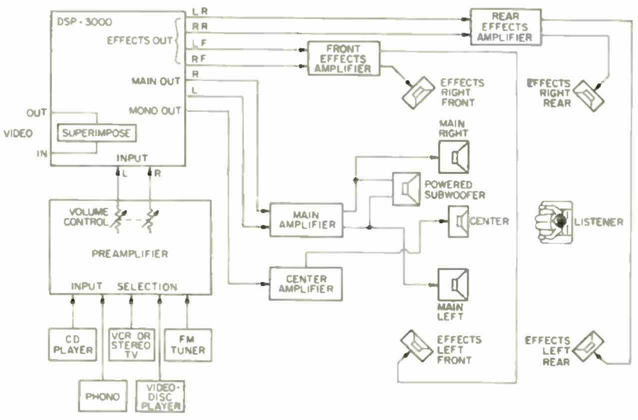



**Fig. 4—Layout of the sound system used in evaluating the DSP-3000.**

I ran through various setup operations, using the functions available on the DSP-3000 remote control. I adjusted the volume of the preamplifier to get the 16-bit display with the first source. The manual suggests that this is a one-time setting, but I checked it frequently. Source levels, even from CDs, varied greatly from one time to another. I had come to a fairly prompt conclusion: The DSP-3000 sounded quieter than the DSP-1, and yet I hadn't driven it into distortion. I believe a good part of the improvement came from being able to set exactly the level which would yield full 16-bit processing.

I used the built-in noise source to match main and effect levels, and found that I had to switch the main output to −10 dB to have the desired level range. Throughout my listening tests, I shifted effect levels and front-to-rear balance to suit. I set the operating conditions for the video superimpose, which made it easier to set parameters because of the much larger display on the TV.

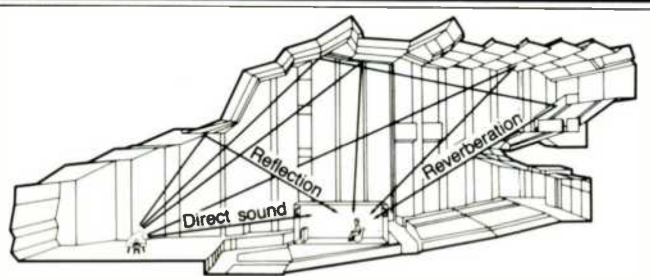
In the listening evaluation, I purposely picked sources to match the various programs, and then tried other programs if that seemed worthwhile. Unless stated otherwise, CDs were the sources.

First was the assessment of the five Concert Hall programs, each with two choices. For Berlioz's "Symphonie fantastique" with Dutoit and the Montreal Symphony (London 414203-2), I liked Halls A, B, and E in Europe and G in the U.S.A. during the first part of the listening. I ended up concluding that I really liked Hall B in Europe best of all, with Hall H in the U.S.A. in second place.

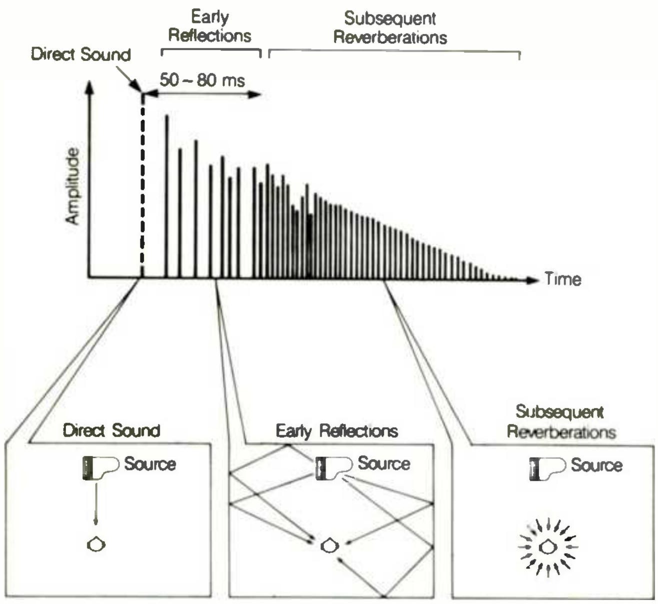
With Dvořák's "Symphony No. 9" with Solti and the Chicago Symphony (London 410116-2) I preferred Hall G in the U.S.A., but I also liked Halls B and C in Europe and Live Concert A (Program 5). Some overtures by Elgar, with Gibson and the Scottish National Orchestra (Chandos CHAN-8309), sounded best with Hall E in Europe, although Hall B and Live Concert B also were quite enjoyable. Tchaikovsky's "Serenade in C for String Orchestra" with Marriner and the Academy of St. Martin-in-the-Fields (Philips 411471-2) was a very good match for Hall C, with very satisfying sound also possible with Halls A, D, and H.

LPs were used for the assessment of "Opera/6." Puccini's "La Bohème" with Freni, Gedda, Schippers, and the chorus and orchestra of the Rome Opera House (2-Angel 4AVB-34025) sounded better with "Mezzanine." Gounod's "Faust," on the other hand, with de los Angeles, Gedda, Cluytens, and the chorus and orchestra of the National Theatre of Opera (Angel 3622) was more satisfying with "Balcony." I tried "Church/8" during the scene in the church and it didn't sound right at all. The "Soldiers' Chorus" was smoother in "Mezzanine," but there was less excitement in the singing.

**The signal-level display reads in bits, so you can optimize for low distortion with maximum S/N ratio.**




**Sound field in a typical concert hall.**




**Directional characteristics of direct sound, early reflections, and reverberation.**

I generated a user program for "Cathedral/7" by reducing reverberation time from 4.0 to 3.2 S and the initial delay from 95 to 85 mS. These changes may not seem large, but they gained important changes in the sound field. On *20 Christmas Carols* with St. George's Chapel Choir (Abbey CDMVP-827), the preset program was a very good fit—except for "Ding, Dong, Merrily on High," which benefited from the changes in the user program. It sounded even better, however, with the user version of "Church/8." Victoria's "Requiem," with The Tallis Scholars (Gimell CDGIM-012), was best overall with one of the "Cathedral" versions. I preferred Michael Murray on *The Organs at First Congregational Church, Los Angeles* (Telarc CD-80088) with the user program, but most other people preferred the preset.

My user version of "Church/8" had reverberation time reduced from 2.5 to 1.5 S and the initial delay reduced from 40 to 35 mS. The *20 Christmas Carols*, Victoria's "Requiem," and many of my own in-church recordings were very good matches to the sound fields of either the preset or user versions.

"Jazz Club 1/9" and "Jazz Club 2/10" have similar sound fields in general, but the differences can be easily heard with most music. Jennifer Warnes on *Famous Blue Raincoat* (Cypress YD-0100) matched well to "Village Vanguard," "Village Gate," and "Cellar Club," but not to "Cabaret." This CD also was good with "Rock Cnct/12/Arena" but not "Disco/13." Creedence Clearwater Revival on *Chronicle* (Fantasy FCD-CCR2-2) and Air Supply on *Love & Other Bruises* (Columbia CK 35047) sounded better with the "Jazz Club 1" choices. The former did sound good with "Jazz Club 2," but I kept switching between "Cellar Club" and "Cabaret," depending upon the tune. The Air Supply tunes were better with "Cabaret." I judged "Village Vanguard" to be the best choice overall among all programs for recorded dance music from the big band era. It might seem strange, but I thought that an NBA play-off game sounded quite good with either "Village Vanguard" or "Cellar Club."

"Chamber/11" was modified for a user program by reducing the reverberation time from 1.1 to 0.8 S. A collection of short baroque works with the Paillard Chamber Orchestra and others (Erato ECD-55018) sounded better with the user program for all of the works. I found that even if the reverberation was reduced by only 0.1 S, the change was noticeable. I came to the same conclusion with Mozart's "Eine kleine Nachtmusik" with Mackerras and the Prague Chamber Orchestra (Telarc CD-80108) and Bach's "Brandenburg Concerti" with I Musici (2-Philips 412790-2). With these CDs, however, the preset program was the better choice quite a few times. Other possible programs for this music were "Opera/6," "Jazz Club 1/9," "Jazz Club 2/10," "Rock Cnct/12," "Stadium/15," and "Presence/16." In other words, don't be afraid to try any program: There might be particular sound-field qualities that you like.

"Rock Cnct/12" was another good choice for Jennifer Warnes and Creedence Clearwater Revival, particularly "Arena." Air Supply sounded good with "The Roxy Theatre," as well. I thought "Disco/13," with its "New York" setting, was a better match to Creedence Clearwater Revival and Air Supply, but the heavier bass of the "Tokyo" position could be the preference of others.

"Pavilion/14" and "Stadium/15" were possibilities for some of the pop/rock groups, but they weren't my choices. The music of Sousa in *Peaches and Cream* with Kunzel and the Cincinnati Pops (MCD 10005) did sound quite good with both of these programs. After listening for some time, I moved the high-pass filter up to 63 Hz to reduce what sounded like a form of bass hangover.

"Presence/16" is a good choice for all types of sources when an up-front sound character is wanted. It's a good compromise setting for listening to FM music programs: The effects are quite pleasurable and the announcer won't sound like he's in a garage. "Presence A" and "Presence B"

**The DSP-3000 sets new and higher performance and flexibility standards for creating realistic and exciting sonic illusions.**

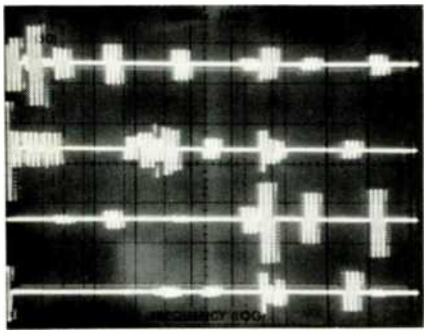


**Fig. 5—Output from effect channels in user-modified Presence A program, with 3-cycle, 700-Hz tone burst. Traces are (top to bottom): LF, RF, LR, and RR. (See text.) Vertical: 1 V/div. Horizontal: 11 mS/div.**

are usually quite different in the listening. *Kiss of the Spider Woman*, with William Hurt and Raul Julia (Showtime simulcast), had much centered dialog and I much preferred "A" over "B." *Ladyhawke*, with Matthew Broderick, Rutger Hauer, and Michelle Pfeiffer (videodisc), had more spread, but I still preferred "A." I thought that the NBA playoff game had a being-there quality with "A." In fact, I thought that this was the best choice of all for sports listening, including the announcing.

I decided to use the editing capability of the "Presence" program to create my own sound field. I put in 11 reflections each for the left and right channels. I purposely increased the angle off axis for each increase in reflection delay. I varied the levels and reversed polarity somewhat randomly. Figure 5 shows the output from the four effect channels with a 3-cycle, 700-Hz tone burst. The channel levels of the delayed bursts correspond to the levels and angles that I programmed in. The sound was smooth in character for different types of music, and quite enveloping but not very exciting after listening more than a few minutes.

"Surround/17" was very satisfactory for the two movies with either "A" or "B." *Spenser for Hire*, on ABC television, was similarly successful.

"Movie Theater 1/18" and "2/19" provided good choices for movies and TV shows. "Standard" was the best choice for *Kiss of the Spider Woman* and *Spenser for Hire*. "Adventure" was my preference for *Ladyhawke* and for the 1960 movie, *Heller in Pink Tights* with Sophia Loren and Anthony Quinn. *Lucas* (1986), with Corey Haim and Kerri Green (cable simulcast), was best with "Standard." *Kingdom of the Spiders* (1977), with William Shatner and Tiffany Bolling, was a good match for "Classic." The limitation of all of them was that the dialog seemed disembodied. It was centered, but it was also spread. I tried using "Front" to drive the center speaker, but the level was too low and the sound character wasn't what I wanted.

I did find that I could improve the dialog by reducing "C. Sptl Exps," "C. Liveness," and "C. Ini. Dly" in a variety of combinations. Unfortunately, the more dialog was improved, the poorer the background music and effects became. I added left and right connections from the main channels to the center-speaker amplifier (in mono mode) and put the voices back with the bodies. I returned the three parameters mentioned above to their preset values, getting the best results for all of the movies and TV shows.

Each main output can be Y-connected to drive both the main amplifier and a stereo amplifier with a mono function for driving both the center speaker and a subwoofer with its own low-pass filter. (Readers, please note that a Y connector *cannot* be put *across* the left and right outputs.) I do feel that the DSP-3000 lacks in not having mono center and subwoofer outputs from the main channels. Most powered subwoofers can be connected across the main speakers, so I see the missing mono center output as more of a limitation. I should note, however, that turning on the speaker of the TV set or monitor at a low level may be sufficient if the sound quality is adequate. The DSP-3000 does not have the Sound Effector programs of the DSP-1, but they have little value for normal music listening, and they have no value for movie or TV program sound.

"Dolby Surround/20" was a very good choice for Dolby-encoded movies. Although the surround channels did not match the results with other programs, there was excellent dialog centering and the voices were embodied—where they belong! For even better results, Yamaha offers the DSR-100 Dolby Pro Logic decoder, which provides the directional orientation, dialog channel, and front/rear separation of commercial theater systems. The $599 cost is high, except perhaps for confirmed movie buffs.

### Conclusions

Yamaha has added to its DSP-1 laurels by bringing out the DSP-3000. Features such as the bit monitor, the excellent displays, the direct digital input, and the noise source all contribute to the value of this superb equipment. New programs such as Opera and Movie Theater, more concert halls, jazz clubs, and all the other venues provide very worthwhile one-button choices to match specific sources. The system delivers no-fuss selection of an incredible variety of sound fields. Changing parameters is very easy for those who want to, and "Presence" offers an opportunity for involved sound-field creation. Muting the effect channels emphasizes what is lost, and collapse of the sound field to stereo is *not* pleasurable.

The Yamaha DSP-3000 is an expensive device but it is *the* premier means of enhancing the listening experience. Additional dollars would need to be spent for the effect channels equipment, but whatever is invested will bring much more than simple enjoyment. The DSP-3000 lacks the main mono center and subwoofer outputs of the DSP-1. Outside of that, the DSP-3000 sets new and higher standards in quality, performance, and flexibility in the creation of exciting, realistic sonic illusions. *Howard A. Roberson*